SX-200 ICP

Folio Documentation

Software Release 5.0 UR2

Powering connections

IMPORTANT - PLEASE READ CAREFULLY

1. The information contained in these documents is not warranted by Mitel or any of its affiliates or subsidiaries and is subject to change without notice. Mitel assumes no liability for any errors or omissions.
2. These documents may be reproduced, either wholly or in part, solely for distribution to authorized users of Mitel's products.
3. By downloading these documents, you assume responsibility for the content they contain. To signifiy responsibility you are required to insert your company name in one of the places indicated in the document.
4. Mitel logos and wordmarks, along with legal disclaimers and copyright information, contained in these documents must NOT be removed.
5. These documents and the information they contain are provided by Mitel on an "AS IS" basis. No support is available from Mitel or its subsidiaries to assist the user with the modification, printing, distribution, or other processes pertaining to the use of these documents.

Trademarks

Mitel is a registered trademark of Mitel Networks Corporation.

Other product names mentioned in this document may be trademarks of their respective companies and are hereby acknowledged

SX-200 ICP
Release 5.0 UR2
2016

SX-200® IP Communications Platform (ICP)

- Release 5.0 UR2 -

Technical Documentation

Review Specifications

Install Hardware

Install Software

Program

Troubleshoot

Notices | Regulatory Info. | Trademarks | Version

What's New in this Release?

Release 5.0 UR2

- **Support for the new Mitel Gigabit Ethernet (GigE) Stand (V2)**

Topics changed in Release 5.0 UR2:

Peripheral Equipment

Form 38 - ACD Keys Template

Setting up the Precidia Ether232

Release 5.0 UR1

- The Mailbox Lockout feature enables a user's mailbox to be locked after three unsuccessful login attempts.
- A new embedded voice mail language, English Overlaid, provides numeric prompts ("Press 7 to play...") in place of the default alphabetic prompts ("Press P to play..."). See Form 49 (Voice Mail Options) and Form 50 (Mailboxes).

Release 5.0

- New 5540 IP Console
- New AX Controller

Topics changed in Release 5.0 for 5540 IP Console:

Install a 5540 IP Console

Peripheral Equipment

5448 PKM

Product Comparison Chart

Attendant Handset and Headset Receiver Volume

Attendant Console Set Paging
Call Logging
Call Park - Destination Phone
Call Park System Orbit
Intercom Calls
Park and Page
Record a Call
System Identifier
Whisper Announce
Form 07 - Console Assignment
Form 09 - Desktop Device Assignment
Printer/Terminal Support

Topics changed in Release 5.0 for the AX Controller:

SX-200 ICP Hardware and Software
SX-200 ICP Controllers
SX-200 AX Controller
AX Controller DSP Options
Installing the AX Controller
Integrating the AX into the IP Network
Dual Link T1/E1 Framer
T1/E1 Combo Module
Interfaces
RS-232 Ports
10/100 Base Ethernet
System Identity/Module Button
CX and AX Controller LEDs
Form 01- System Configuration
Form 04 - System Options/System Timers
Form 47 - IP Networking
Form 53 - Bay Location Assignment
Installing Mitel IP phones
North American Loss and Level Plans
PRI Support
DSP Resources
Programming DHCP Server
System Identifier
Show Device Status
Form 14 - Non-Dial-In Trunks
Form 15 - Dial-In Trunks
Form 35 - Global Find Access

Release 4.0 UR5

- New Mitel 5304 IP Phone

- New Mitel 5312 IP Phone
- New Mitel 5324 IP Phone
- Increase in the maximum number of BONS/BNIC cards from 40 to 56

Release 4.0 UR4

- A 24-Port ONSP card (PN 50005731) is available for the ASU II. The circuits on this card have additional electrical protection. In addition, each port supports 3 REN (Ringer Equivalency Number) so a call can ring up to three phones that are connected to the same port. You require a TDM license to enable each port. The 24-Port ONSP card is not supported in the Peripheral Cabinet.
- Support for Mobile Extension Calling Line ID.
- New Mitel 5330 IP Phones with backlit display.

Release 4.0 UR3

- SX-200 ICP Release 4.0 UR3 installer runs on Windows Vista.
- Support for Cordless Module and Devices on 5330 and 5340 IP Phones.
- DID server printout now prints only DID number.
- ASUInfo maintenance command now displays IDPROM for line cards on ASUII.
- SHOW-IDENTITY command's response now includes CCA number for APC installations.
- Additional 16-port ONSp card (part number 50005533) is available. The ports on the new ONSp card are protected against surge and lightning. The current ONS cards (16-port ONS card and 12 port ONS/ 4 port LS trunk Combination card) will continue to be available

Release 4.0 UR2

- Support for co-residency of Mitel Managed VPN and Unified Messaging. Refer to the product documentation for these applications on Mitel OnLine. It is important to understand the security considerations involved in running these applications as co-resident on the same server. For APC deployments, refer to article # *07-2674-00009 - Release Notes for VPN and Unified Messaging Support on the Application Processor Card (APC) for Standalone Server for SX200-ICP* available in the Mitel Knowledge Base.

What's Not Supported in this Release?

Support for the following documented products or functions has been deferred to a future release of the SX-200® ICP:

- Support for Cordless Module and Devices on 5330 and 5340 IP Phones.

Release 4.0 UR1

New Hardware Features

- Support for Mitel 5330 and 5340 IP Phones (Family 5).

Release 4.0

New Hardware Features

- New CX controller variation. The original CX controller is now the CXi. The new "switchless" CX is ideal for businesses with existing Layer 2 switches and for those that require more than 16 ports.
- CX and CXi controllers ship with 512 MB internal CompactFlash.
- New ASU II supports up to 48 ONS phones or up to 8 LS trunks depending on configuration and peripheral cards.
  - ASU II Installation
  - ASU II Programming
- Support for Quad CIM modules on new CX controllers (for ASU II support only. Bay functionality coming soon.)
- The Applications Processor Card (APC) is a PC on a compact card installed in theCX/CXi controller to host the Managed Application Server.
- Support for Mitel Gigabit Ethernet (GigE) and Wireless LAN (WLAN) Phone Stands

New Software Features

- IP phone capacity for the CX/CXi increased to 100.
- ARS entry capacity increased to 350.
- The Ring Groups feature provides the ability to ring all members of a group simultaneously.
- The Direct Inward Dialing Server application allows Hotels to automatically assign direct dial numbers for their guests.
- New DHCP Options 43 and 125 replace separate Options 128 -135 with a data string.
- New subform of Form 9 - Review (CALL FORWARD) replaces MyAdministrator Call Forwarding profile editor
- Enhanced Hotel/Motel Subattendant features for the Mitel 5340 IP Phone
- MTCE command to reset the Application Processor Card.

Discontinued or Unsupported Products

Support for the MyAdministrator software application has been discontinued.

Release 3.1

The documentation includes the following updates:

- Loop Start Trunk Measurement Tool enhancements:
  - CO Silence and Milliwatt tone numbers are programmable on Subform 13 (Trunk Circuit Descriptors Options).
  - New System Option 82 (Use HW Echo Canceller).
  - New Subform of Form 13 (Audio Configuration Table) for AMB CLASS circuits.
- New softkeys, Program Like and Range W/Keys, are available on Form 09, (Desktop Device Assignments).
- The MX Controller now supports up to 248 IP Phones when equipped with 8 DSPs for the Hospitality configuration option.

- A new Class of Service Option, 282 (Headset Hard Hold), allows headset users to place calls on hard hold while the Automatic Line Selection feature is enabled. Without COS Option 282, the system will select a new line whenever headset calls are placed on hard hold.
- New and enhanced features:
    - Paging - Telephones now supports Set Page, Group Page and All Set Page across trunks.
    - It is now possible to toggle between paging and ringing a set on Intercom Calls. Previously, set users could not return to ringing after they had switched to paging.
    - Single Button Transfer to Voice Mail is now possible while the call on Consultation hold.
    - New Attendant Single Button To Voice Mail feature.
    - Calls ringing on the console or at common alerting devices (night bells) can be picked up at any time using Trunk Answer From Any Station (TAFAS) if COS Option 281 (TAFAS Access During Day/Night Service) is enabled.
    - Display telephones no longer display MESSAGE to indicate that a voice mail message has been received. See Messaging - Call Me Back.
    - The 5485 IP Paging Unit is now accessible to stations (ONS telephones) for Paging - PA.

Release 3.0

New Hardware Features

- Support for Mitel® 5212 and 5224 Dual Mode IP Phones.
- SpectraLink NetLink h340 Wireless Telephone.

New Software Features

- The Calling Party Number Substitution feature provides calling party identification for outgoing calls for purposes of network identification and call back.
- The Direct Inward Dialing Translation feature enables calls received by dial-in trunks to be routed based on Day, Night1 and Night2 service.
- Support for Embedded PRI with Dual T1/E1 Framer (MX) or T1/E1 Combo (CX).
- The Secure Hot Swap feature enables users to switch programming between two devices.
- New options available on Form 42 (Link Descriptor Options): Protocol, Protocol Variant, Network/User, Unknown Numbering Plan, Bearer capability Voice, CLIR Voice, and Invert D Channel.
- New L2 Switch settings (CX): IGMP Snooping; Rapid Spanning Tree; STP Bridge Priority.
- New System IP settings: DiffServe Code Point; VoiceVLAN ID; Voice VLAN Priority.
- The CX now supports 802.1 p/Q prioritization for voice traffic on the default VLAN (1), or on a separate Voice VLAN. Program matching VLAN ID and priority values on Form 47, Subform 1 (System IP) , the DHCP server, and the external L2 switches.
- Line key enhancements:
    - Maximum number of line appearances is increased from 32 to 64 for DTS, CO, Logical, Multicall and Key line keys.
    - Automatic line selection is programmable using either the Line Select or Line Preference feature.
    - Call Logging is available for multiple appearances of DTS and CO line keys.
    - Forward Campon feature is now available if the forwarding destination is a Speed Call Key or System Abbreviated Dial Number.
    - Split and Hold features can now be used during conferences with multiple DTS or CO lines (two or more LS trunks).

  - If a user on a line appearance of a CO trunk exits a conference initiated by a user with an ONS set, the conference will continue without interruption.
- Subattendant functionality extended to Mitel 5020, 5220, 5220 Dual Mode, and 5224 Dual Mode IP Phones.
- New Option 16 (IP Set Voice Encryption) available on Form 04 (System Options/System Timers).
- Mitel Managed VPN now supports Traffic Shaping to ensure QoS for voice calls. With this feature, multiple sites can be interconnected using relatively low-bandwidth cable or DSL links to the Internet.
- Support for additional LS Trunk Line Length and Impedance values in Subform 13 (Trunk Circuit Descriptors Options). Also see Attenuation Levels for Short and Long CO Trunks.
- User mailboxes can now start with a 9. To prevent conflict with the Phonebook feature (Directory Dial by Name), an inter-digit timer is started every time the first digit entered is 9. If no other digits are entered before the timer expires, the user is transferred to the Phonebook.
- The time out behavior (disconnect or forward to operator) can now be selected for each Multi-level Auto Attendant menu node mailbox.
- Mitel Your Assistant now supports the YA Softphone, an IP-based software phone, and YA Lite, a free version of the product.
- MTCE command to determine the status and version number of the AMB/AOB.
- LED indication of handsfree mute varies depending whether the telephone has a MICROPHONE or MUTE key. Telephones with a MUTE key include the Mitel 5212 and 5224 IP Phones, and the TeleMatrix 3000IP Phone.
- The SX-200 ICP Web Interface can be used to program the SX-200 ICP from the IP network.
- A uniform numbering plan can be programmed for a network of systems that are interconnected by IP trunks.
- An ACD position with a Mitel IP or SUPERSET telephone may have a Programmable Key Module (PKM) associated with it.

What's Not Supported in this Release?

Support for the following documented products or functions has been deferred to a future release of the SX-200® ICP:

- Mitel TeleMatrix 3000 IP Phone.

Release 2.3

The documentation includes the following updates:

- The SX-200 ICP now supports disconnect tones for 11 Latin American countries in addition to the default disconnect tone for Canada/USA. The disconnect tone files are selected via System Option 138 (Country Variant For Disconnect Tone Control) in Form 04 (System Options/System Timers).

Release 2.2

The documentation includes the following updates:

- Mitel Line Interface Module: Provides users of 5220 and 5224 Dual Mode IP Phones with the ability to make and receive calls on an analog line. The Line Interface Module replaces the Mitel 5425 Access Module.
- DTMF tones can now be transmitted to 5485 IP Pager Unit for PA Paging.

Release 2.1

New Hardware Features

- SpectraLink NetLink e340 and NetLink i640 Wireless Telephones: Replace the discontinued Symbol MiNET Wireless phones.
- Stratum 3 Clock Module (CX)
- Support for Category 3 Ethernet cable (CX)

New Software Features

- Mitel Your Assistant
- Line Characterization Tool - milliwatt tone loopback
- Message Flash Notification for Incoming Calls (Mitel 5207 IP Phone)
- Support for SMTP Authentication
- Embedded voice mail now supports notification on every new message regardless of whether or not notification for previous messages has already been answered.

Discontinued or Unsupported Products

Symbol MiNET Wireless Telephones (replaced by SpectraLink telephones)

Release 2.0

The documentation includes the following updates:

Portfolio Migration and Line Size Expansion

- Introduction of new hardware for the SX-200® ICP: the Analog Services Unit (ASU); the Dual Link T1/E1 Framer MCC; the T1/E1 Combo MMC; and the Quad CIM MMC.
- A new form (Form 53, Bay Location) that is used to assign bay numbers to the physical connections on the SX-200 ICP Controller.
- Increased support for up to seven Peripheral Cabinets on MX controller
- DSP resource limits

  up to 8 DSPs (248 IP phones) on MX controller
  up to 5 DSPs (64 IP phones) on CX controller
- Migration path from an SX-200 EL/ML to an SX-200 ICP MX.

Voice Mail

- Mailbox key feature for embedded voice mail.
- The Unified Messaging feature package enables the SX-200 ICP to manage e-mail using SMTP (Simple Mail Transfer Protocol) and/or IMAP (Internet Message Access Protocol).

Platform Enhancements

- The LS Trunk Line Setting tool measures signals sent by the CO over LS trunks and recommends optimum circuit descriptor settings for the on-board ASU trunk cards (Analog Main Board and Analog Option Board).
- New Form 47 IP Networking subforms for MX and CX.

- A secure Telnet connection is now required to program the SX-200 ICP from the IP network. Use the Mitel Telnet client provided on the software CD-ROM, or any other Telnet client that supports SSL/TLS.
- IP Networking information is now programmed in CDE.
- The Resource Monitor checks for low and fragmented memory, and for stale tasks and components. If a problem is detected, the system generates an alarm and log record. Critical alarms cause the system to be reset.

Key Telephone System (KTS) and other Set-Related Features

- The Call Monitor feature has been extended to non-ACD usage.
- A new Class of Service Option, DTS/CO Line Transfer Call Handling for square key system applications which enables incoming calls on shared DTS/CO lines that are transferred unsupervised to another extension to move from the extension's prime line to the DTS/CO line key.
- Feature keys may be configured to park calls in one of up to 25 specific orbits using the Call Park System Specific Orbits feature.
- Feature keys may be configured to place telephones into Day, Night1, or Night2 service modes under the Night/Day Switching feature.
- The new Hold and Page feature allows telephone users to place a call on hold and quickly page the system simply by pressing the Hold key a second time.
- Support for trunk calls to use the Campon Priority Over Call Forward Busy feature.
- The ability to set the system time and date has been extended to 5020 and 5220 IP Phones
- Call forwarding of Intercom calls to phones with COS Option 278 (Intercom Mode) enabled is ignored when the forwarding destination is another extension.
- Do Not Disturb key flashes when the feature is active (requires Form 04, System Option 102 - Feature Level set to level 6).

Applications

- Support for the following applications added

  MiTAI 11.2
  My Administrator 3.1
  Mitel 6100 Contact Center Solutions (For more information about this product, see the SX-200 ICP General Information Guide)
  Mitel Teleworker Solution

CDE/Maintenance Enhancements

- Support for Mitel 5215 and 5220 Dual Mode IP Phones.
- Ping command available in Form 48 (Voice Networking) and Maintenance.
- A new display filter (SHOW ALL) has been added to Form 04 (System Options/System Timers).
- Users can switch between the CDE and Maintenance applications by pressing a softkey—no need to exit the current application to start the other.
- Support for Door Phone units with two-way communication and remote door entry capability.
- Mitel Option Code (MOC) in Form 04 programs all options and timers for the system.

What's Not Supported in this Release?

Support for the following documented and undocumented products or functions has been deferred to a future release of the SX-200® ICP:

- Mitel 6510 Unified Messaging
- TAPI™ Desktop
- BOOTP protocol for DHCP

- MyAdministrator serial connection to the SX-200 ICP
- BRI (ISDN)
- Taske™ (SMDR and RTE integration)
- Datasets used as serial ports are supported for the following Data output functions only:

SMDR	Maintenance Logs	Traffic Measurement
CDE Data Print	Hotel/Motel Wakeup	Hotel/Motel Audit
IP Traffic Measurement	ACD Agent Summary	ACD Real Time Events
ACD Group Summary		

Discontinued or Unsupported Products

The following products have been manufacture discontinued by Mitel. However, some are supported by one or more variants of the SX-200 ICP; see Product Comparison for more information.

- SUPERSET™ 3DN and SUPERSET 4DN telephones
- SUPERSET 400-series telephones
- SUPERSET PKM
- MILINK® Data Module
- LCD Console
- All Mitel Remote Access products (XpressConnect 5232i, XpressOffice Teleworker, Xpressway RLAN etc.)

Unless otherwise noted, the following products and peripheral devices are not supported by the SX-200 ICP and are not described in this infobase:

- SUPERSET 3 and SUPERSET 4 telephones
- SUPERSET 4 telephones for voice mail
- Modem Interconnect Panel
- DATASET 1102 Rack-mounted Dataset
- DATASET 2102 Rack-mounted Dataset
- DATACABINET 9000 data cabinet
- DATASHELF 9100 datashelf
- LCD Console (and Console module for Universal Card) are supported but are not described in this infobase.
- SPINE Bays
- SX-200 IP Node
- SMART-1 Device
- MITEL MyAttendant
- DATASET 1101 data cartridge
- Mitel 5424 IrDA Module
- Mitel TAPI Desktop Software
- SUPERSET DSS module (supported by the SX-200 ICP MX controller only)
- SUPERSET 4090 (supported by the SX-200 ICP MX controller only)
- Digital Line Monitor (DLM)

Changes to Documentation

Removed the LAN Design Guidelines and portions of the Product Specification section from the Technical Documentation. The Guidelines are now in an expanded Engineering Guidelines document which is provided in PDF format on the Mitel Edocs website (http://edocs.mitel.com). Gone from the Production Specification section is the non-technical content (everything except for the Engineering, Safety and Regulatory information) that is also found in the General Information Guide (GIG). A PDF version of GIG is available on the Edocs website only.

Additional Documentation

The SX-200 ICP includes the following additional printed or electronic documentation. Electronic documents are provided on the SX-200 ICP software CD and on the Mitel Edocs website (http://edocs.mitel.com). For instructions on accessing the documents, see the Introduction chapter in the Technician's Handbook.

- Technician's Handbook (print and PDF)
- Safety Instructions (print and PDF)
- General Information Guide (PDF)
- Engineering Guidelines (PDF)
- IMAT Online Help (installs with IMAT application)
- MyAdministrator Online Help (installs with MyAdministrator application) and User Guide (PDF)
- Telephone, Attendant, Subattendant, Voice Mail, Hotel/Motel Front Desk, and MyAdministrator User Guides (PDF)

Release 1.2

The documentation includes the following updates:

- Support for the 5485 IP Paging Unit
- Support for Centralized Voice mail over T1 Trunks
- New Device Types in Form 09 (5215d and 5220d) to support the DP Lite telephones. The DP Lite telephones provide the hardware to support dual mode operation (MiNET and SIP) in a future release of the SX-200 ICP (tentatively, Release 2.0).

Engineering Information

SX-200 ICP Controllers

The SX-200 ICP Controller provides the voice, signaling, central processing, and voice mail resources for the system.

There are four controllers:

- CX controller
- CXi controller
- AX controller
- MX controller

Product Comparison

The following tables provides an overview of the differences between the CX, CXi, AX, and MX controllers.

Table: Product Comparison

<u>Legend:</u>

Blank = not supported on this product variant
M = migration upgrades only
Y = supported on this product variant

	CX/CXi	AX	MX
Target Markets	Key System	PBX Hybrid	Key/PBX/Hybrid
Maximum Number of IP Phones	100	248	248
Base System Components	1 10/100 WAN port 1 10/100/1Gig LAN port 16 10/100 802.3af LAN ports (CXi only)	2 10/100 LAN ports	1 LAN port 2 CIM ports
	Internal Compact Flash (512 MB)	Internal Compact Flash (4 GB)	Internal Hard Drive
	<u>3 MMC slots:</u> • Slot 1 or 2 for T1/E1 Combo or Quad CIM • Slot 3 for Dual DSP	<u>2 MMC slots</u> • Slot 1 for T1/E1 Combo or Dual T1/E1 Framer • Slot 2 for Quad or Dual DSP	<u>3 MMC slots:</u> • Slot 1 for Quad CIM, Dual FIM, or T1/E1 Framer • Slot 2 for Quad CIM, Dual FIM, T1/E1 Framer, Quad or Dual DSP • Slot 3 for Quad or Dual DSP
	Two DSP resources on mainboard	Four DSP resources on the AX controller card	Dual DSP MMC
	1 serial port for maintenance (IP socket used for printer output)	1 serial port for maintenance (IP socket used for printer output.)	3 serial ports (maintenance, printer and alarms)
	System i-Button for Feature Options Records	System i-Button for Feature Options Records	System ID module for Feature Options Records
	Analog Main Board with 4 ONS, 6 LS trunks (RJ11), 2 PFT, Music source, 1 Pager, 2 generic relays for auxiliary bell, door sensor, or alarm		Analog Main Board with 2 ONS, 2 DNIC, 6 LS trunks (RJ77), 2 PFT, Music source, Pager and Night Bell Stratum 3 clock
	Echo Cancellation (in software on mainboard DSPs)	Echo Cancellation (in software on mainboard	Echo Cancellation (in hardware on DSP MMCs)

Table: Product Comparison

Legend:

Blank = not supported on this product variant
M = migration upgrades only
Y = supported on this product variant

		DSPs)		
Field Replaceable Units (FRUs)	Dual and Quad DSP MMC (max. 7 DSPs, including two DSPs on mainboard and single DSP on T1/E1 Combo)		Dual and Quad DSP MCC (max. 8 DSPs)	Dual and Quad DSP MCC (max. 8 DSPs)
T1/E1 Combo MMC has T1 port for T1/D4; Stratum 4 clock; single DSP (max. 1 T1/E1 Combo MMC)		T1/E1 Combo MMC has T1 port for T1/D4; Stratum 4 clock, single DSP Dual T1/E1 Framer MMC has two T1 ports for T1/D4 or PRI.	Dual T1/E1 Framer T1 has two T1 ports for T1/D4	
Internal hard disk (optional)		N/A	Internal hard disk	
Optional Analog board with 4 ONS, 6 LS trunks, 2 PFT, 1 Pager, 2 generic relays for auxiliary bell, door sensor, or alarm		N/A	Optional Analog board with 4 ONS, 6 LS trunks	
Application Processor Card (APC)		N/A	N/A	
APC Hard Drive		N/A	N/A	
Stratum 3 clock		N/A	Stratum 3 clock	
N/A		N/A	Dual FIM MMC (optional)	
Quad CIM MMC (optional)		N/A	Quad CIM MMC (optional)	
Operations and Maintenance	Software on external flash (hot swappable)		Software on internal flash	Software on external flash (hot swappable)
InstallShield for software/ database installation		InstallShield for software/ database installation	InstallShield for software/ database installation	
New install and upgrade with flash		New install and upgrade with flash	New install and upgrade with flash	
Remote upgrade via FTP server		Remote upgrade via FTP server	Remote upgrade via FTP server	
Default and Blank databases		Default, Premier and Blank databases	Default, Premier and Blank databases	
Database backup via FTP		Database backup via FTP	Database backup via FTP	
Call Control	Common to all product variants			

Table: Product Comparison

Legend:

Blank = not supported on this product variant
M = migration upgrades only
Y = supported on this product variant

Networking Support	WAN interface & services: 1 x 10/100 WAN port with DHCP, PPPoE, or static address assignment LAN interfaces & services: • Layer 2 switch supports 802.1 p/Q prioritization on a single VLAN, or on separate VLANs for voice and data (CXi only) • 16 x 10/100 802.3af LAN ports provide power to IP phones (CXi only) • 1 x 10/100/1G LAN port enables connection of expansion switches • DHCP server • Firewall features: many-to-1 NAT, unknown packet drop/reject/log, PPTP and IPSec pass-through, Port Forwarding • Router discovery and configuration • PPTP server for remote client access	LAN interfaces & services • Prioritization on a single VLAN or on a separate VLAN for voice and data • 2 x 10/100 LAN port enables connection of expansion switches •DHCP server	LAN interface & services: 1 x 10/100BaseT port • DHCP server • support for VLAN tagging
Voice mail	Embedded Voice Mail N/A N/A N/A	Embedded Voice Mail	Embedded Voice Mail ONS Voicemail DNIC Voicemail MEM (card in bays)
Nodes Supported			
Digital (TDM) Bays			Y (see Note 1)
SPINE			
NSU (PRI only)			Y
ASU	Y		Y
ASU II	Y		Y
IP Node			M (see Note 2)
Line and Trunk Interfaces Supported (See Note 3)			
ONS/CLASS	Y	Y	Y

Table: Product Comparison

Legend:

Blank = not supported on this product variant
M = migration upgrades only
Y = supported on this product variant

OPS			Y
DNIC			Y
IP	Y	Y	Y
LS/GS			Y
LS/CLASS	Y	Y	Y
DID (Analog)			Y
E&M (Analog)			Y
T1/D4	Embedded in controller	Module	Peripheral cabinet card or module
T1/PRI	Embedded in controller	Module	Peripheral cabinet card or module
IP Trunk	Y	Y	Y
Desktop Devices Supported			
SUPERSET 3DN/4DN Telephones			M
SUPERSET 400 Series Telephones			M
SUPERSET 4001 Telephone			M
SUPERSET 4015 Telephone			Y
SUPERSET 4025/4125 Telephones			Y
SUPERSET 4150 Telephone			Y
Mitel 5001 IP Phone			
Mitel 5010 IP Phone	M		M
Mitel 5020 IP Phone	M		M
Mitel 5201 IP Phone	Y		Y
Mitel 5207 IP Phone	Y		Y
Mitel 5215 IP Phone	Y		Y
Mitel 5215 IP Phone (Dual Mode)	Y		Y
Mitel 5220 IP Phone	Y		Y
Mitel 5220 IP Phone (Dual Mode)	Y		Y
Mitel 5212 IP Phone (Dual Mode)	Y		Y
Mitel 5224 IP Phone (Dual Mode)	Y		Y
Mitel 5304 IP Phone (Dual Mode)	Y	Y	Y
Mitel 5312 IP Phone (Dual Mode)	Y	Y	Y
Mitel 5324 IP Phone (Dual Mode)	Y	Y	Y
Mitel 5330 IP Phone (Dual Mode)	Y	Y	Y
Mitel 5340 IP Phone (Dual Mode)	Y	Y	Y
Mitel Gigabit Ethernet Phone Stand	Y		Y
Mitel Wireless LAN Phone Stand	Y		Y
Symbol Wireless Phones			Y
SUPERCONSOLE 1000®			Y

Table: Product Comparison

Legend:

Blank = not supported on this product variant
M = migration upgrades only
Y = supported on this product variant

5540 IP Console	Y	Y	Y
PKM12 (DNIC)			Y
PKM48 (DNIC)			Y
SUPERSET DSS Module			M
5410 PKM (discontinued)			
5415 PKM (discontinued)			
5412 PKM (IP)	Y		Y
5448 PKM (IP)	Y	Y	Y
5303 Conference Unit	Y		Y
5310 Conference Unit	Y		Y
Dataset 1103			Y
Dataset 2103			Y
Analogue Phones	Y	Y	Y
TeleMatrix 3000IP Phone	Y		Y

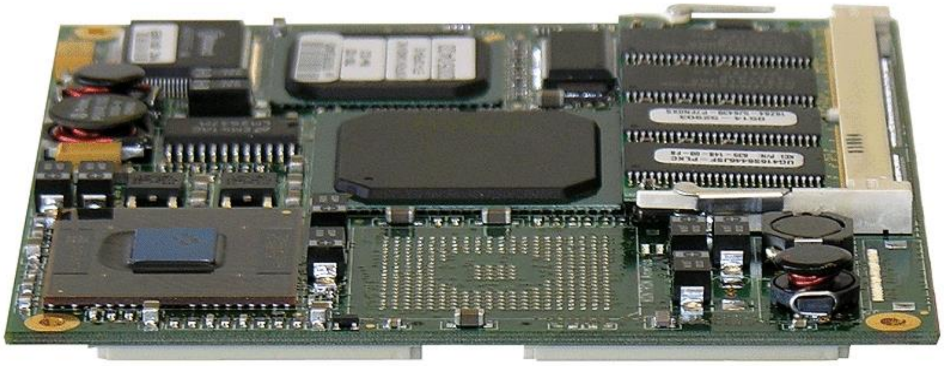
Notes:

1. Installed base to be migrated to embedded IP phone and trunk support.
2. Line and trunk interfaces on these product variants supported only with embedded cards and modules in the controller.
3. Supported by IP phones and the two embedded DNIC ports.
4. Independent and incompatible with the Automated Attendant and fax tone detection features of the SX-200 ICP embedded voice mail application.
5. As a maximum, the MX controller can be expanded to include:
   - up to seven Peripheral cabinets
   - up to four NSUs
   - up to six ASUs or six ASU IIs

Application Processor Card (CX/CXi)

The Application Processor Card (APC) is a PC on a compact card. Installed in the CX pr CXi controller, the APC hosts the Managed Application Server (MAS). For information about programming and using software blades and services, refer to the MAS and application documentation at www.mitel.com.

Application Processor Card (CX/CXi)



The APC requires a dedicated hard drive. For installation details, see Installing the APC and Installing the APC Hard Drive.

SX-200 ICP MX Controller

The SX-200 ICP MX controller, introduced in Release 1.0 (minus the MX designation) has the processing, memory, mass storage, power, and input/output capabilities to support up to 248 users. It provides 64 channels of Ethernet to Time Division Multiplexing (E2T) and 64 channels of echo cancellation.

The base MX is shipped with a Dual DSP module for baseline telephony requirements and two CIM ports for connection to Peripheral cabinets, ASUs, NSUs, and PRI cards. You can expand the capabilities of the MX by installing Mitel Mezzanine Cards (MMCs) in it. The following MMC modules are available as FRUs:

- One or Two DSP modules (Dual and Quad versions) for CLASS tone generation, conference bridges, DMTF receivers, and G.729 compression
- One or two Quad CIM modules for connecting NSUs or Peripheral Cabinets
- One or two Dual FIM modules for connecting NSUs or Peripheral Cabinets
- One or two Dual T1/E1 Framer modules for connecting T1 trunks with T1/D4 and PRI support

Note that the MX has three MMC slots, and that slot 3 (the right slot) is reserved for DSP modules (either Quad or Dual). The base MX has a Dual DSP module in slot 3. The combination of modules determine programming limits. For maximum DSP resources (eight), install two Quad DSPs. For maximum connectivity to offboard devices (any combination of Peripheral cabinets, NSUs and ASUs totalling ten), install two Quad CIMs.

The MX can ship with embedded analog capability provided by one or two analog boards:

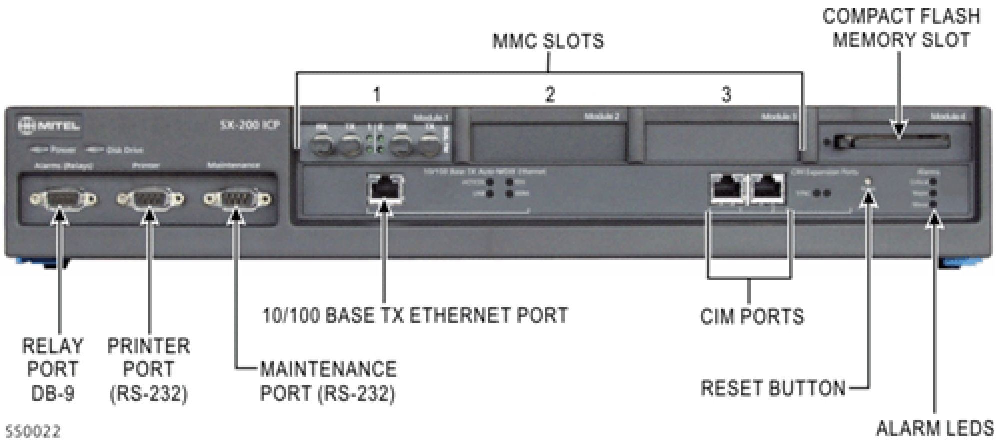
- The Analog Main Board (AMB) has the following circuits: 6 Loop Start, 2 ONS, 2 DNIC, 1 Music On Hold, 1 Paging, 1 Auxiliary Ringer
- The Analog Option Board (AOB) expands on the AMB providing an additional 6 LS and 2 ONS circuits

Front Panel

- Dry Contacts (DB-9 connector) for Alarm, Door Relay, and Auxiliary Ringer
- Two RS-232 ports (DB-9 connectors) for Printer and Maintenance terminal
- Two onboard CIM (Copper Interface Module) ports to connect Peripheral cabinets, ASUs, NSUs, and PRI cards.
- One 10/100 Ethernet connection via RJ-45 (8-pin CAT5 crossover cable)
- Expansion sites for MMC modules (T1/E1 Framer, Quad CIM, Dual FIM, Dual and Quad DSP)

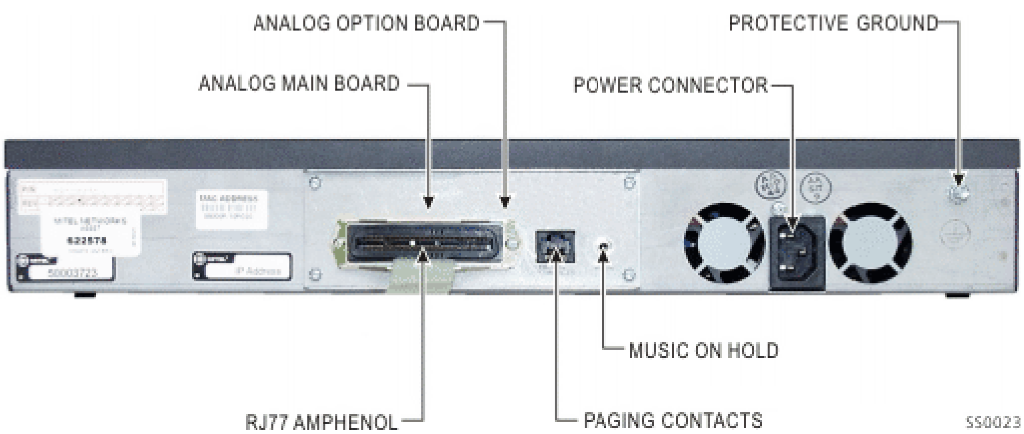
Note: Modules are purchasable options.

- CompactFlash card reader
- Status LEDs.



Back Panel

- Power connector
- Protective ground to ground the chassis
- Music on Hold port
- Paging equipment connector
- Amphenol Connector for analog trunks and ONS/DNIC stations

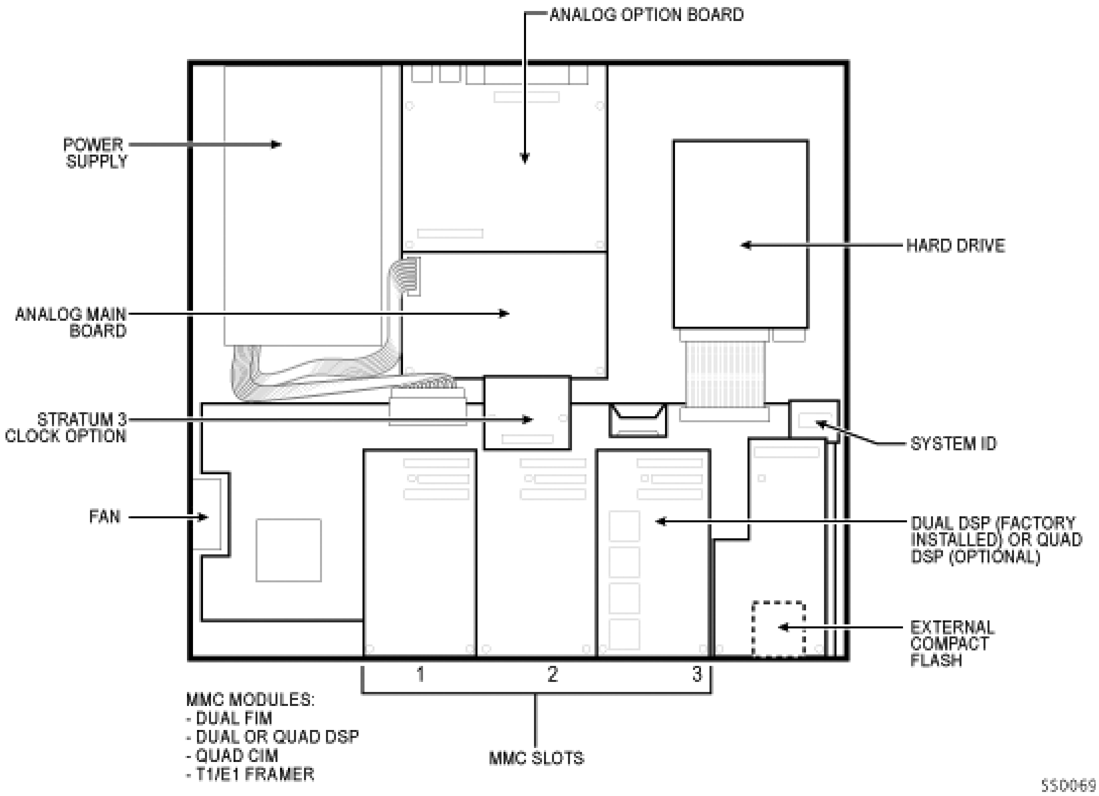


Internal Components

- Analog Main Board provides circuits for DNIC devices (2), ONS devices (2), LS CLASS trunks (6), Music on Hold (1), Paging (1), and Generic relays (1).

  Note: The Generic relay on the Analog Main Board is programmed for Night Bell use in the default database.
- Analog Option Board provides circuits for additional LS CLASS trunks (6), and ONS devices (2).
- External CompactFlash card for software installation and upgrades.
- System Hard Drive (replaces the internal CompactFlash card).
- Stratum 3 clock (optional) provides timing and synchronization for T1 trunks.

- Dual and Quad DSP modules provide resources for CLASS tone generation, Record a Call conferences, DMTF receivers, voice compression, and voice echo cancellation. The system is shipped with one Dual DSP module and can support a maximum of two (Dual or Quad) DSP modules.
- Dual T1/E1 Framer module (optional) provides two T1 trunk ports that support T1/D4. Each port has 24 B-channels.
- Dual FIM module (optional) provides two fiber optic interfaces connections to NSUs or Peripheral Cabinets.
- Quad CIM module (optional) provides four copper interfaces to NSUs or Peripheral Cabinets.
- Power supply
- Cooling fan
- Option sites
- System ID Module



Specifications

Dimensions

Height:	2.5 in. (6.35 cm) (1.5 U)
Width:	17.75 in. (45.1 cm) (19" rack mountable)
Depth:	18.5 in. (47 cm)
Weight:	15.6 lb (7.08 kg)

Environment

Table: Storage Environment

Condition	Specification
Temperature	-40º to 150ºF (-40º to +66ºC)
Humidity	15-95% Relative Humidity, non-condensing
Vibration	

0.5 g, 5 to 100 Hz, any orthogonal axis

1.5 g, 100 to 500 Hz, any orthogonal axis
Mechanical Stress

One 15.3 cm (6 in.) drop, each edge and corner adjacent to the rest face–unpackaged

One 76.2 cm (30 in.) drop, each edge and corner packaged in cardboard & foam

Table: Operational Environment

Condition	Specification
Temperature	39º to 120ºF (4º to 49ºC)
Humidity	34-95% Relative Humidity, non-condensing
Maximum Heat Dissipation - fully loaded (see Note)	170 BTUs per hour
Conversion factors: 1 watt is equal to 3.412 BTUs per hour, 1 ton of refrigeration is equal to 12,000 BTUs per hour or 3.516 Kilowatts, and 3/4 Kilowatt-hour is equal to 1 ton of refrigeration.	

Power Supply

Input / disconnect:	IEC 320 - C14 Class 1 AC Receptacle
Operation:	100-120/200-240 V ac auto selectable
Maximum power input:	100 W
AC source:	90 - 132 Vac; 47 - 63Hz in North America

SX-200 ICP AX Controller

The SX-200 ICP AX controller is ideal where a high density of analog devices is required. The AX controller has the processing, memory, mass storage, power, and input/output capabilities to support up to 248 IP users and 288 analog phones.

The base AX is shipped with four DSPs on its main board for baseline telephony requirements. You can expand the processing capabilities of the AX by installing Mitel Mezzanine Cards (MMCs) in it. The following MMC modules are available as FRUs:

- One Dual or Quad DSP modules for G.729 compression
- One T1/E1 Combo module (one DSP resource, one T1 trunk port for T1/D4 or PRI, and one stratum 4 clock)
- One Dual T1/E1 Framer modules for connecting T1 trunks with T1/D4 and PRI support

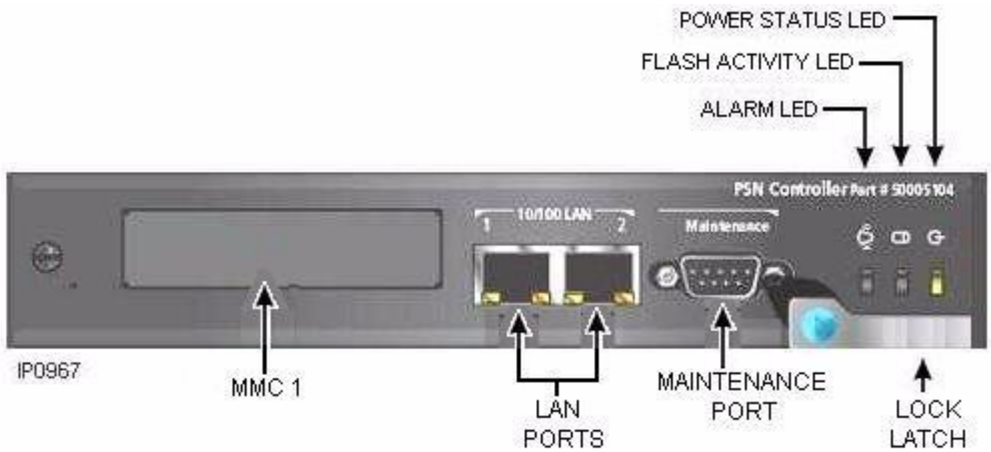
The AX has two MMC slots. A single T1/E1 Combo module or Dual Framer can be installed in slot 1. A Dual or Quad DSP can be installed in slot 2.

The AX controller also will support a second AC Power Supply Unit for redundancy.

Front Panel - AX

The controller card front panel consists of the following components:

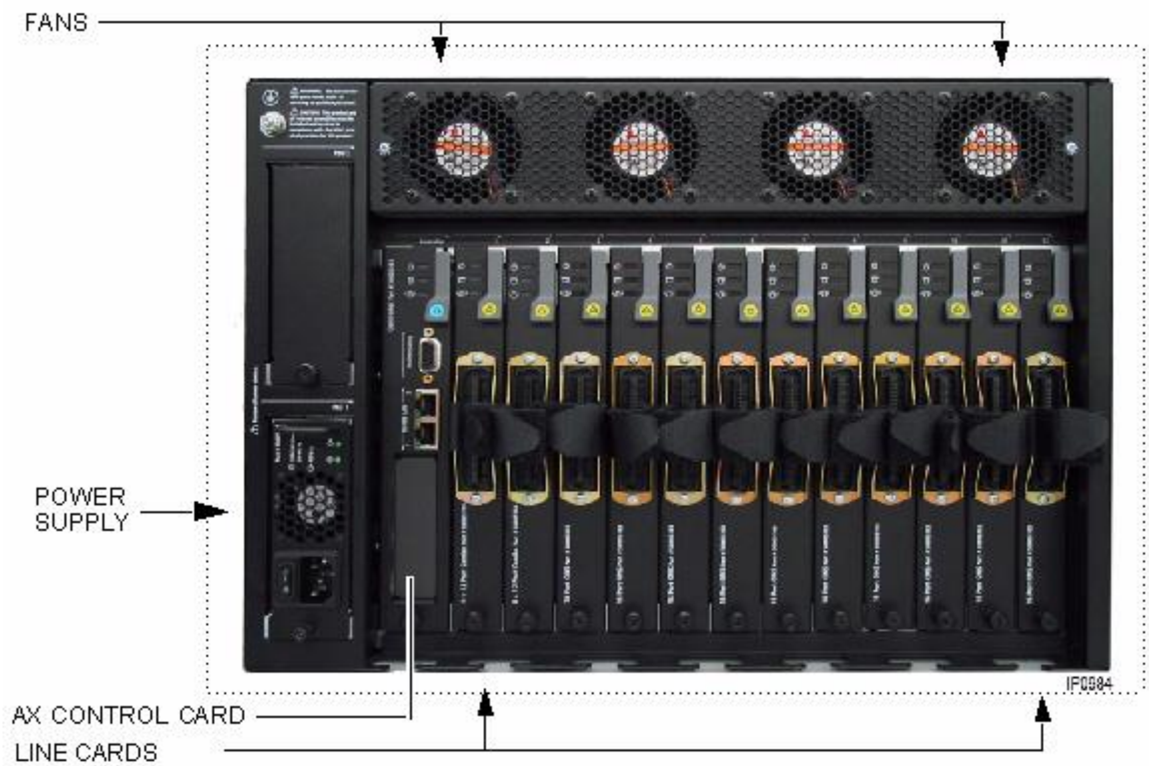
- RS-232 port (DB-9 connector) Maintenance terminal function
  (printer function available via LAN port)
- Two 10/100 Base-TX LAN ports (RJ-45 connector)
- One slot for expansion modules: T1/E1 Combo modules or Dual T1/E1Framer
- Status LEDs - Alarm, Flash Activity, Power/Status.



Rear Panel - AX

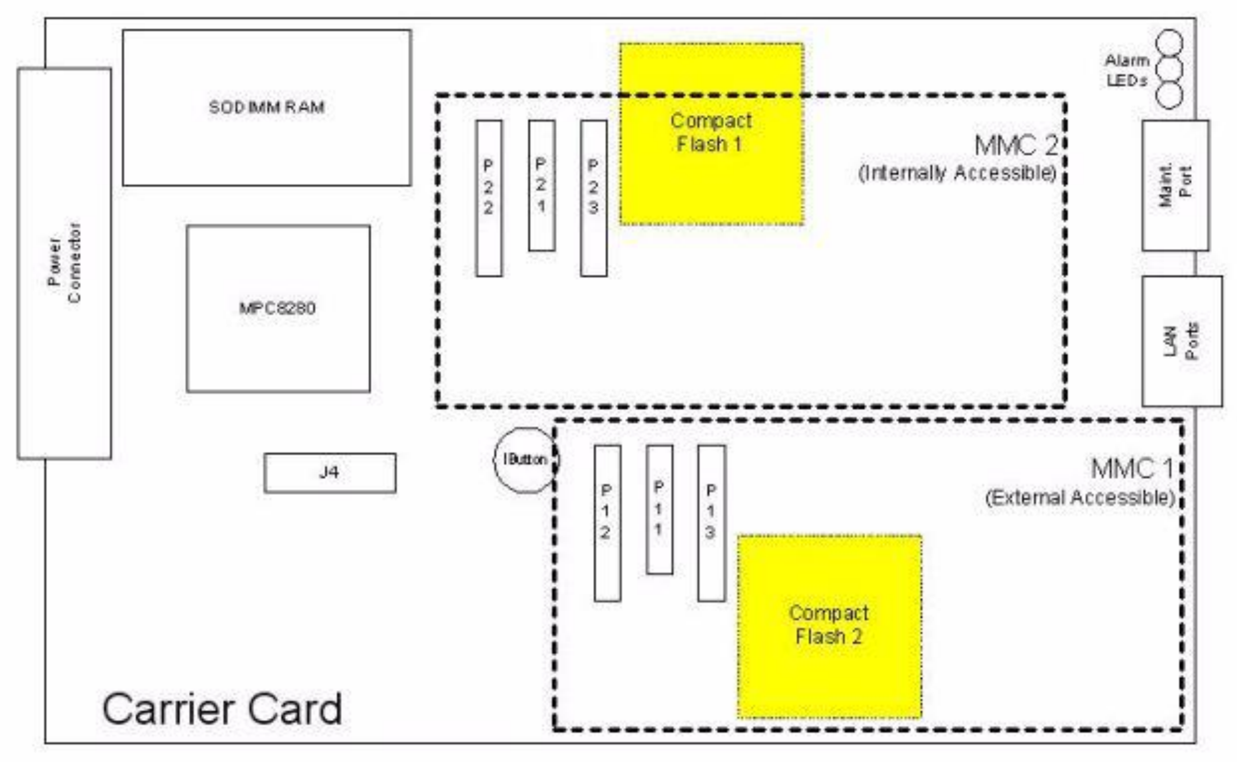
The controller card in the AX controller rear panel consists of the following components:

- One or two power supply units
- Fan complex
- Controller card (described above)
- 12 line card slots
- Protective ground



Internal Components on AX Controller Card

- Internal CompactFlash 1 for software installation and upgrades
- Internal CompactFlash 2 for system files and partitions and voicemail
- 512MB SDRAM SODIMM
- MMC slot 1 for T1/E1 Combo module, Dual T1/E1 Framer Module
- MMC slot 2 for Dual and Quad DSP
- System i-Button
- Stratum 3 clock provides timing and synchronization for T1 trunks



Specifications

Dimensions

Height: 12.25 in. (31.1 cm) (7 U)

Width: 17.76 in. (45.1 cm) (19" rack mountable)

Depth: 14.6 in. (37.1 cm)

Weight: 39.69 lb (18.0 kg)

Environment

Table: Storage Environment

Condition	Specification
Temperature	-40º to 150ºF (-40º to +66ºC)
Humidity	15-95% Relative Humidity, non-condensing
Vibration	

0.5 g, 5 to 100 Hz, any orthogonal axis

1.5 g, 100 to 500 Hz, any orthogonal axis
Mechanical Stress

One 15.3 cm (6 in.) drop, each edge and corner adjacent to the rest face –unpackaged

One 76.2 cm (30 in.) drop, each edge and corner packaged in cardboard & foam

Table: Operational Environment	
Condition	**Specification**
Temperature	39º to 120ºF (4º to 49ºC)
Humidity	5-95% Relative Humidity, non-condensing
Maximum Heat Dissipation - fully loaded (see Note)	1020 BTUs per hour (AX)
Air Flow	110 cubic feet per minute at maximum output
Acoustic Emissions	Maximum 50 dBA continuous, 75 dB intermittent (<10% duty cycle)
Conversion factors: 1 watt is equal to 3.412 BTUs per hour, 1 ton of refrigeration is equal to 12,000 BTUs per hour or 3.516 Kilowatts, and 3/4 Kilowatt-hour is equal to 1 ton of refrigeration.	

Power Supply

Input / disconnect:	IEC 320 - C14 Class 1 AC Receptacle (2 receptacles on the AX with redundant power supply)
Operation:	100-120/200-240 V ac auto selectable
Maximum power input:	300 W
AC source:	90 - 264 Vac; 47 - 63Hz in North America

SX-200 ICP CX/CXi Controller

The SX-200 ICP CX/CXi controller has the processing, memory, mass storage, power, and input/output capabilities to support up to 100 IP users. In addition to standard telephony interfaces, the CX/CXi includes a complete range of network interfaces. It has a WAN port for connection to the Internet and a Layer 2 switch (CXi only) for connection to the LAN. Each of the CXi's 16 ports provides power to IP devices in compliance with IEEE 802.3af. The CX/CXi also has an unpowered 1 Gigabit port for connection to another switch on the LAN. For security, the CXi can function as a firewall, dropping or rejecting unknown packets, allowing or disallowing IPSec and PPTP pass-through, and performing many-to-1 NAT (IP masquerading). The CX/CXi can also function as a PPTP server, enabling a remote client to access the CX/CXi on the LAN.

The base CX/CXi is shipped with two DSPs on its main board for baseline telephony requirements. You can expand the processing capabilities of the CX/CXI by installing Mitel Mezzanine Cards (MMCs) in it. The following MMC modules are available as FRUs:

- Dual DSP module (CLASS tone generation, resources for conference bridges, DMTF receivers, and G.729 compression)
- T1/E1 Combo module (one DSP resource, one T1 trunk port for T1/D4 or PRI, and one stratum 4 clock)
- Quad CIM module (four ports for peripheral connectivity)

The CX/CXi has three MMC slots. Slot 3 (the right slot) is reserved for the optional Dual DSP module. A single T1/E1 Combo module can be installed in slot 1 or 2.

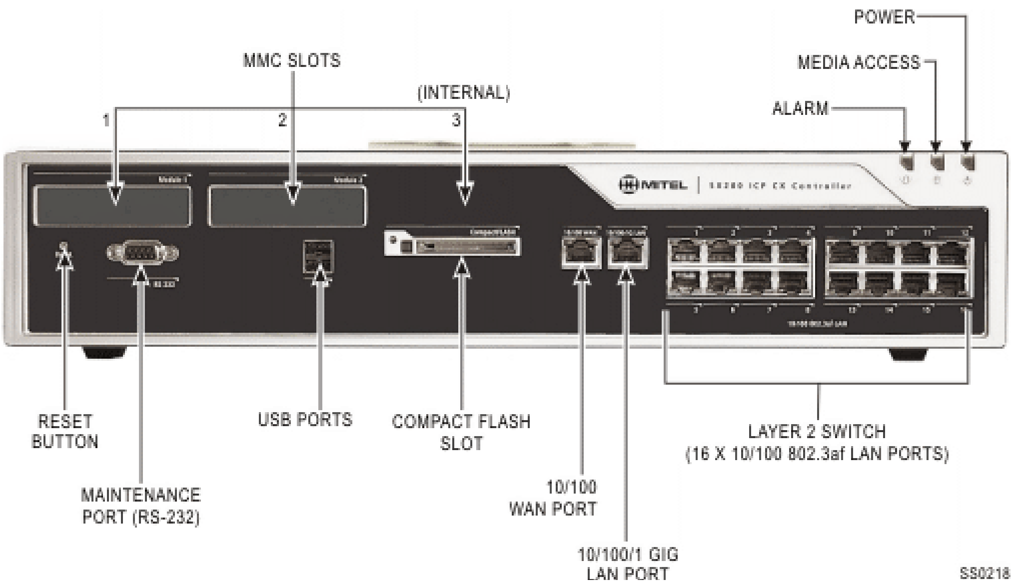
An optional Application Processor Card (APC) can be installed in the CX/CXi, enabling the system to host the Managed Application Server (MAS) and to run various applications. The MAS running on the APC serves as the system's Internet Gateway.

The CX/CXi can ship with embedded analog capability provided by one or two analog boards:

- The Analog Main Board (AMB) has the following circuits: 6 Loop Start, 4 ONS, 1 Music On Hold, 1 Pager, 2 generic relays for auxiliary bell, door sensor, or alarm
- The Analog Option Board (AOB) expands on the AMB providing an additional 6 LS and 4 ONS circuits, plus 1 Pager and 2 generic relays for auxiliary bell, door sensor, or alarm

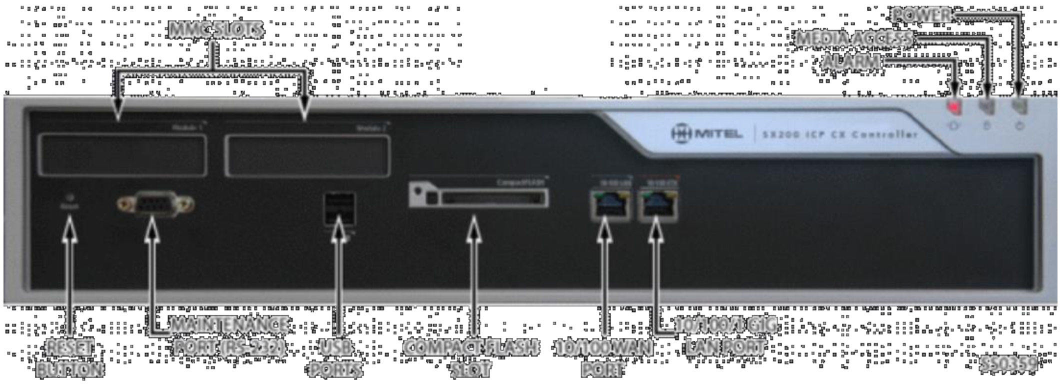
Front Panel - CXi

- RS-232 port (DB-9 connector) Maintenance terminal function (printer function available via LAN port)
- 10/100Base-TX WAN port
- 10/100/1000Base-TX LAN port
- Sixteen 10/100Base-TX 802.3af LAN ports
- Two USB 1.1 ports for future use
- Expansion sites for Dual DSP and/or T1/E1 Combo modules
- CompactFlash card reader
- Status LEDs



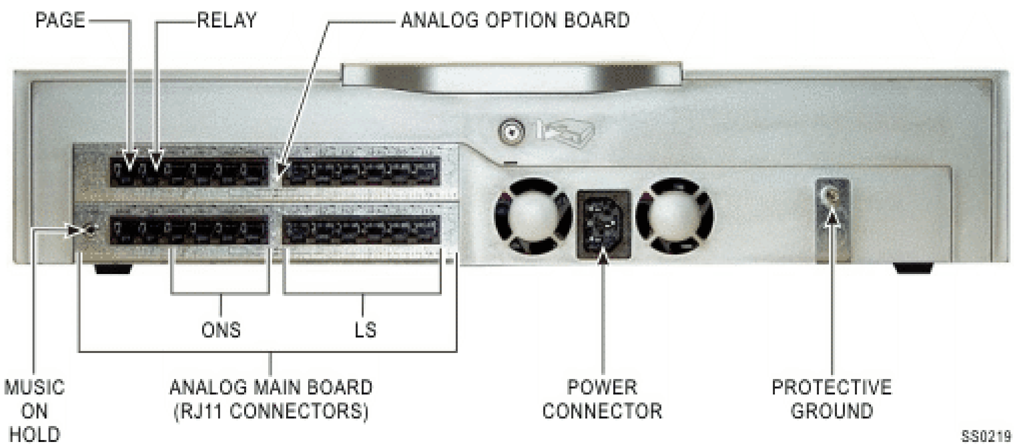
Front Panel - CX

- RS-232 port (DB-9 connector) Maintenance terminal function (printer function available via LAN port)
- 10/100Base-TX WAN port
- 10/100/1000Base-TX LAN port
- Two USB 1.1 ports for future use
- Expansion sites for Dual DSP and/or T1/E1 Combo modules
- CompactFlash card reader
- Status LEDs



Back Panel - CX and CXi

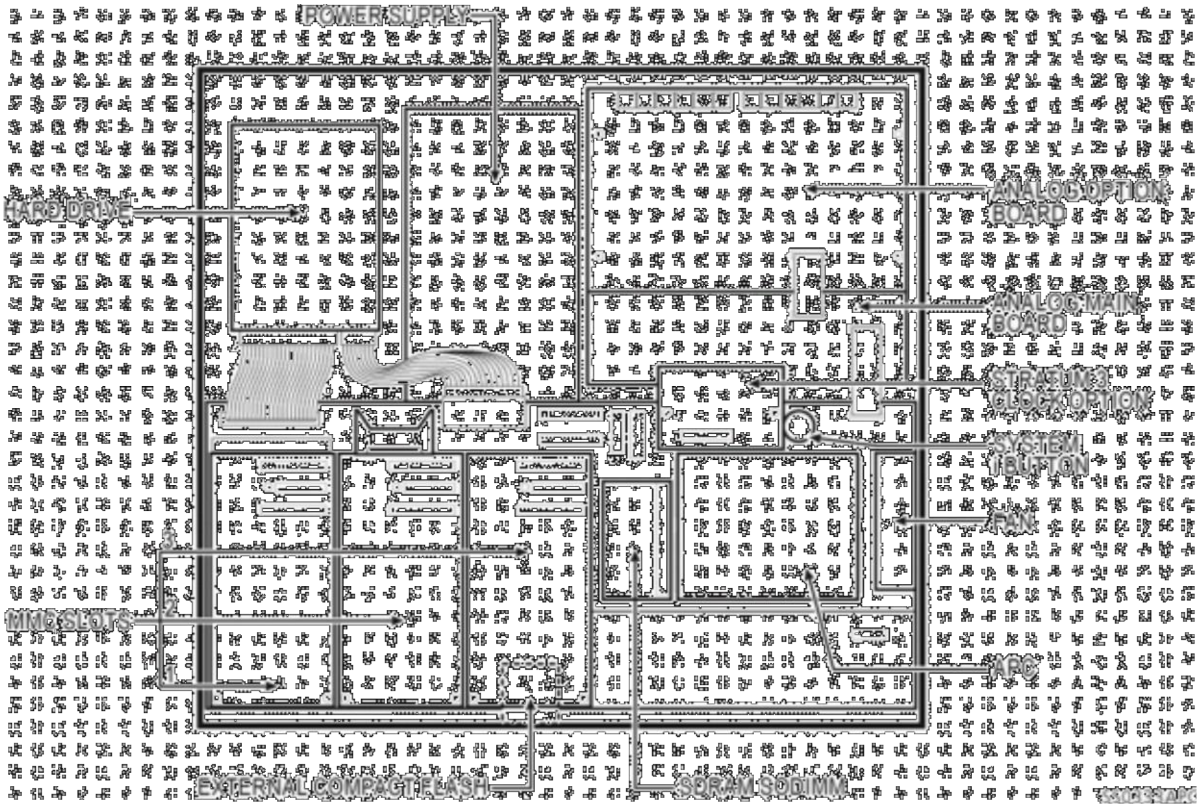
- Analog Main Board (AMB) provides six LS (RJ11), four ONS (RJ11), with two PFT ports, a MOH (3mm) port, a Paging (RJ45) port, and a relay port that supports two generic relays.
- Analog Option Board (AOB) provides an additional six LS (RJ11), four ONS (RJ11), with two PFT ports. The LS Trunk and ONS ports on the AOB have the same interface characteristics as the ports on the AMB. The PFT function is associated with ONS1/LS1 and ONS2/LS2 pair. The AOB also includes one Pager and a relay port that supports two generic relays for auxiliary bell, door sensor, or alarm.
- One AC plug site is provided for the 85-265VAC universal power supply.



Internal Components

- Analog Main Board
- Analog Option Board
- System Hard Drive
- External CompactFlash card (optional)
- 512MB SDRAM SODIMM
- Applications Processor Card (APC) (optional)
- ATX 250 Watt Power supply
- Cooling fan
- MMC slots 1 and 2 for T1/E1 Combo module
- MMC slot 3 for Dual DSP module
- 802.3af Module (required by embedded Layer 2 Ethernet switch)

- System i-Button
- Stratum 3 clock (optional) provides timing and synchronization for T1 trunks.



Specifications

Dimensions

Height: 3.5 in. (8.9 cm) (2 U)

Width: 17.75 in. (45.1 cm) (19" rack mountable)

Depth: 16.5 in. (41.9 cm)

Weight: 19.8 lb (8.98 kg)

Environment

Table: Storage Environment

Condition	Specification
Temperature	-40º to 150ºF (-40º to +66ºC)
Humidity	15-95% Relative Humidity, non-condensing
Vibration	

0.5 g, 5 to 100 Hz, any orthogonal axis

1.5 g, 100 to 500 Hz, any orthogonal axis

Mechanical Stress
One 15.3 cm (6 in.) drop, each edge and corner adjacent to the rest face –unpackaged
One 76.2 cm (30 in.) drop, each edge and corner packaged in cardboard & foam

Table: Operational Environment

Condition	Specification
Temperature	39º to 120ºF (4º to 49ºC)
Humidity	34-95% Relative Humidity, non-condensing
Maximum Heat Dissipation - fully loaded (see Note)	170 BTUs per hour
Conversion factors: 1 watt is equal to 3.412 BTUs per hour, 1 ton of refrigeration is equal to 12,000 BTUs per hour or 3.516 Kilowatts, and 3/4 Kilowatt-hour is equal to 1 ton of refrigeration.	

Power Supply

Input / disconnect: IEC 320 - C14 Class 1 AC Receptacle
Operation: 100-120/200-240 V ac auto selectable
Maximum power input: 250 W
AC source: 90 - 132 Vac; 47 - 63Hz in North America

Interfaces

The MX, CX/CXi, and AX controllers support the following interfaces as standard components or as optional modules:

Interface	MX	AX
Dual Fiber Interface Module	✓	
Quad Copper Interface Module	✓	
Dual Link T1/E1 Framer Module	✓	✓
T1/E1 Combo Module	✓	✓
Analog Main Board / Analog Option Board	✓	
RS-232 Port(s)	✓	✓
10/100Base-TX Ethernet Port	✓	✓
10/100 WAN Port		

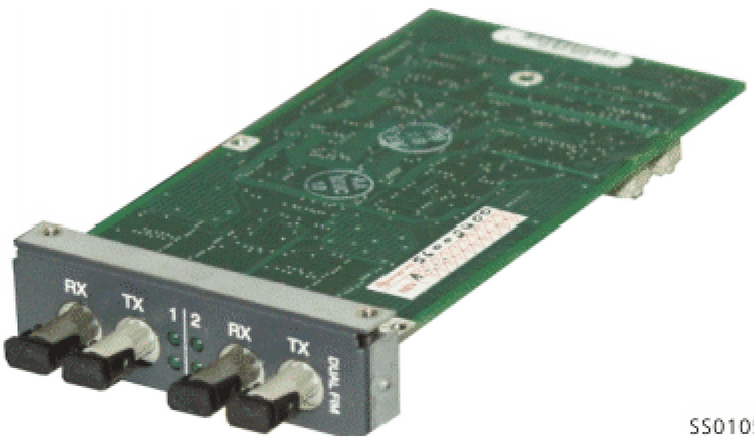
10/100 802.3af LAN Ports		
10/100/1G LAN Port		

Dual Fiber Interface Module (MX)

The Dual Fiber Interface Module (FIM) is an optional electrical/optical interface. It connects SX-200 Peripheral Cabinets or NSUs to the controller through fibre optic cable. When transmitting, the module converts electrical signals to optical signals for output over a fiber optic cable. When receiving, it converts optical signals from the cable to electrical signals.

The MX can support up to two Dual FIMs installed in MMC slots 1 and 2. The Dual FIM is available as a Field Replaceable Unit (FRU).

Dual FIM Module



SS010!

The Dual FIM is composed of a transmitter, receiver, and control section. The control section generates control signals and the transmit clocks. This section also regenerates the telephony clocks for the peripheral nodes, and provides status information for the Main Controller. Each FIM variant is identified by optical wavelength and fiber type (indicated on the FIM).

The same FIM variant must be used at each end of a fiber optic cable. However, a node (endpoint) may be equipped with different FIM variants to suit the length of each cable run.

Note: The SX-200 ICP does not support single FIM modules.

Specifications

Table: FIM Specifications for SX-200 ICP Controller (all variants)

Part Number	50001248 820nm Multi-mode	50003695 1300nm Multi-mode	50003696 1300nm Single-mode
Compatibility	Connects a SX-200 ICP to an NSU or an SX-200 Peripheral Cabinet that houses an equivalent FIM.		
Approximate maximum fiber cable run length (See Note 1)	1km (0.62 miles)	3km (1.9 miles)	14km (8.7 miles) (See Note 2)
Power consumption (Watts)	2.5		
Number of fiber links per FIM 2	2 Tx, 2 Rx		
Fiber connector type	ST (See Note 3)		

Table: FIM Specifications for SX-200 ICP Controller (all variants)			
Electrical interface for each fiber link	6 serial links		
Optical wavelength (nm)	820	1300	1300
Optical budget (See Note 4)	6 db	16 db	9 db
Data rate (Mbits/second)	16.384		
Bit rate after encoding (Mbaud)	20.48		
Fiber optic cable type	62.5/125 um Multimode or 50/125 um Multimode	62.5/125 um Multimode or 50/125 um Multimode	9/125 um Singlemode

Notes:

1. The run length is the one-way length of fiber optic cable between nodes.
2. Frame synchronization errors occur if the loop length of fiber optic cable—from controller to then NSU or Peripheral Cabinet—is between 10.0km and 10.6km (6.2 miles and 6.6 miles). Nodes are typically connected by two-strand or “paired” fiber optic cable. For paired cable, a loop length of 10.0km to 10.6 km equals a 5.0km to 5.3km cable run length.
3. ST is a registered trademark of AT&T.
4. The optical budget is the allowable loss through fiber optic cable, splices, and connectors. The optical budget applies to the run length.

LEDs

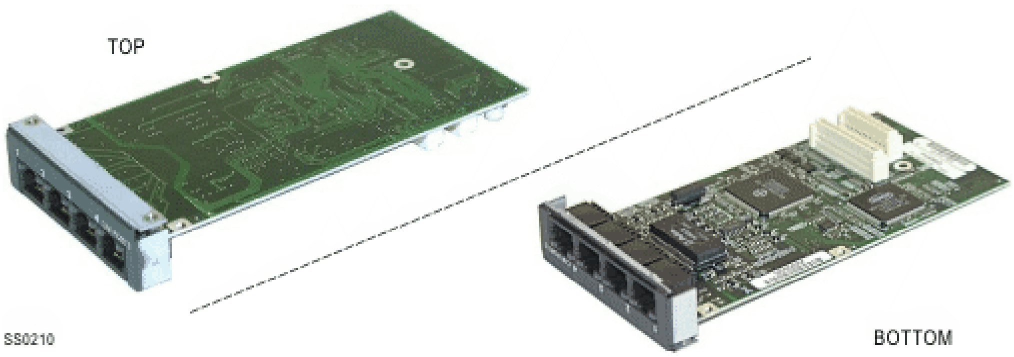
The module faceplate has LEDs that show the link status. For a description of the LEDs, see FIM LEDs.

Quad Copper Interface Module

The MX controller is equipped with an onboard dual-port Copper Interface Module (CIM). An optional Quad CIM module (available on the MX, CX and CXi controllers) provides four more ports. Up to two Quad CIMs can be installed in MMC sites 1 and 2. CIM ports can be connected to ASUs and ASU IIs (onboard CIMs only); NSUs;or Peripheral Cabinets.

Note: A single-port CIM is available for use in the SX-200 Peripheral cabinet; this module cannot be installed in an SX-200 ICP.

Quad CIM Module



Each CIM port

- provides the same interconnect functionality as the FIM but with the use of copper instead of fiber (three ST links exist)

- uses Category 5 UTP (unshielded twisted pair) crossover cable that connects to an RJ45 jack at the module faceplate
- suits co-located systems; not remote systems. The CIM supports a distance of up to 30 meters or 100 feet between cabinets
- provides LEDs that show the link status and an eight-pin modular jack that connects the copper cable.

Specifications

The CIM ports require standard 8-pin modular jacks (RJ-45) consisting of 2 balanced signal pairs on Category 5 Universal Twisted Pair (UTP) crossover cable.

Quad CIM power consumption: 1.0 W

Pinouts

- CIM Port Pinouts

LEDs

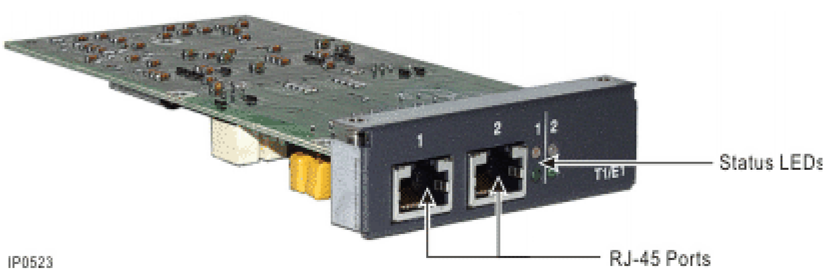
For onboard and Quad CIM Module LEDs, see CIM LEDs

Dual T1/E1 Framer Module (MX and AX)

The Dual T1/E1 Framer module has two digital trunk ports, each of which can be programmed to support either T1/D4 or PRI. Up to two modules can be installed in MMC slots 1 and 2 of the MX controller. One module can be installed in MMC slot 1 of the AX controller.The Dual T1/E1 Framer is available as Field Replaceable Unit (FRU).

Note: E1 is not supported on the SX-200 ICP.

Dual T1/E1 Framer Module



The Dual T1/E1 Framer supports the following protocols and features:

Table: Dual T1/E1 Framer Protocols and Features (Rev. 3)

Link Type	Protocol Variants	Features
PRI	- DMS 100 - DMS 250 - 4ESS - NI2 (Bellcore, 5Ess, and GTD5)	• 1.544 Mbps primary rate connectivity (2 links x 24 B-channels)

- Line encoding: AMI & ZCS, AMI & No ZCS, or AMI & B8ZS
- User/Network side programmable
- Operation mode: CSU or DSX-1
- Termination mode: NT or LT
- Framing mode:
  - PRI - Extended Super Frame
  - T1/D4 - Extended Super Frame or D4
- Generation and detection of the following status indications:
  - Loss of Signal (LOS or Yellow Alarm)
  - Alarm Indication Signal (AIS or Blue Alarm)
  - Loss of Frame Alignment
  - Excessive CRC errors

T1/D4

- Digital E&M
- Digital DID
- Digital CO

Specifications

The Dual T1/E1 Framer has two ports, each of which can be programmed to provide 24 B channels for T1/D4, or 23 B channels for PRI.

Table: Dual T1/E1 Framer Specifications (Rev. 3)

Max. Channels Provided	PRI	46 B channels per module (92 B channels with 2 modules)
T1/D4	48 B channels per module (96 B channels with 2 modules)	
Cabling	Category 5 UTP (unshielded twisted pair)	
Faceplate port	RJ45	
Data Transfer Rates	1.544 Mbps over T1 telephone trunks	

Pinouts

See T1/E1 Tip and Ring Assignments.

LEDs

See T1/E1 LEDs.

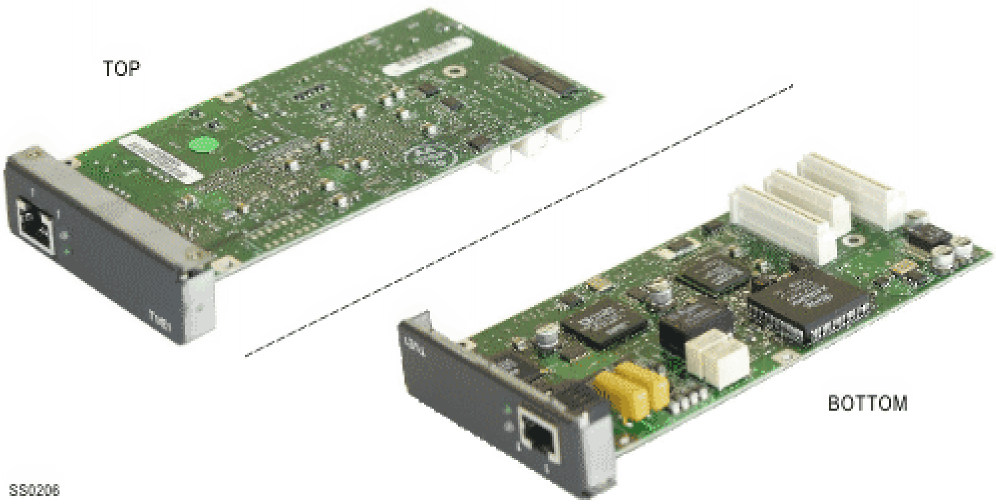
T1/E1 Combo Module (CX/CXi and AX)

The T1/E1 Combo MMC combines trunking and DSP functionality in a single card. The digital trunk port can be configured as a T1/D4 interface that provides 24 B-channels, or as a PRI interface that provides 23 B-channels. The DSP provides resources for CLASS tone generation, Record a Call conferences, DMTF receivers, voice compression, and voice echo cancellation. The module also includes a Stratum 4 clock for synchronization with the incoming digital frame rate clock.

The AX or CX/CXi can support a single T1/E1 Combo MMC installed in MMC slot 1 or 2 (slot 2 - CX/CXi only). The module is an optional Field Replaceable Unit (FRU).

Note: E1 is not supported on the SX-200 ICP.

T1/E1 Combo Module



The T1/E1 Combo Module supports the following protocols and features:

Table: T1/E1 Combo Protocols and Features

Link Type	Protocol Variants	Features
PRI	- DMS 100 - DMS 250 - 4ESS - NI2 (Bellcore, 5Ess, and GTD5)	• 1.544 Mbps primary rate connectivity (2 links x 24 B-channels) • Line encoding: AMI & ZCS, AMI & No ZCS, or AMI & B8ZS • User/Network side programmable • Operation mode: CSU or DSX-1 • Termination mode: NT or LT • Framing mode: - PRI - Extended Super Frame - T1/D4 - Extended Super Frame or D4 • Generation and detection of the following status indications: - Loss of Signal (LOS or Yellow Alarm) - Alarm Indication Signal (AIS or Blue Alarm) - Loss of Frame Alignment - Excessive CRC errors
T1/D4	- Digital E&M - Digital DID - Digital CO	

Specifications

The T1/E1 Combo MMC provides a single port which can be programmed to provide 24 B channels for T1/D4, or 23 B channels for PRI.

Table: T1/E1 Combo Module Specifications		
Max. Channels Provided	PRI	23 B traffic channels
T1/D4	24 B traffic channels	
Cabling	Category 5 UTP (unshielded twisted pair)	
Faceplate port	RJ45	
Data Transfer Rates	1.544 Mbps over T1 telephone trunks	

Pinouts

See T1/E1 Tip and Ring Assignments.

LEDs

See T1/E1 LEDs.

Analog Main Board / Analog Option Board

The Analog Main Board (AMB) and Analog Option Board (AOB) support the following:

<table>
<tr><th></th><th>CX/CXi</th><th>MX</th></tr>
<tr><td>AMB</td><td>4 ONS devices with System Fail Transfer (PFT) capability on 2 circuits</td><td>2 ONS devices with System Fail Transfer (PFT) capability on 2 circuits</td></tr>
<tr><td>--</td><td>2 DNIC devices</td><td rowspan="5"></td></tr>
<tr><td colspan="2">6 LS trunks</td></tr>
<tr><td colspan="2">1 Music-on-Hold input</td></tr>
<tr><td colspan="2">1 paging output (with relay contacts)</td></tr>
<tr><td>2 generic relays</td><td>--</td></tr>
<tr><td>AOB</td><td>4 ONS devices with System Fail Transfer (PFT) capability on 2 circuits</td><td>2 ONS devices</td></tr>
<tr><td colspan="2">6 LS trunks</td><td rowspan="3"></td></tr>
<tr><td>1 paging device</td><td>--</td></tr>
<tr><td>2 generic relays</td><td>--</td></tr>
</table>

Custom Local Area Signaling Services (CLASS) is supported on embedded LS trunks and ONS lines. CLASS allows the SX-200 ICP system to pass Calling Line ID digits and CLASS name information to display sets that support Caller ID functionality.

Note: If version 2 of the AMB or AOB is installed, the controller software must be upgraded to release 3.0 or greater. The AMB and AOB must be compatible.

Description

System Fail Transfer (SFT)

See SFT Operation.

ONS Lines

The ONS line circuits connect standard telephone sets with total combined DC resistance of the set and line not exceeding 600 Ohms. As such, the ONS line circuits are used to connect internal telephone extensions close to the system.

The ONS line circuit has the following features:

- Loop start POTS line with constant loop current mode in normal off-hook condition.
- On-hook signal transmission for CLASS function.
- Balance ringing.
- Hardware ring tip.
- 48V tip and ring DC voltage in on-hook state.
- Off-hook detection by loop current.
- Software-controlled AC line and Balance impedance setting.
- Software controlled audio gain settings in both transmit and receive path.

LS Trunk

The analog boards interface with an analog central office (CO) using -48 Vdc loop start (LS) trunks.

The LS trunk circuit has the following features:

- Ringing detection.
- On-hook signal reception for caller ID feature.
- Tip and ring voltage monitor.
- Software controlled AC line impedance setting.
- Software controlled audio gain settings in both transmit and receive path.
- Line reversal detection.

Music On Hold (AMB)

The SX-200 ICP features a mini (1/8" - 3.5mm) phono jack on the back panel of the controller providing an interface to an external music source for Music-on-Hold.

The MOH interface supports the following features:

- Transformer coupled input with DC blocking capacitor and 600 Ohms AC input impedance.
- Limited signal level going onto the PCM stream.
- Signal level overload protection as mandated by FCC part 68 on encoded analog content

Paging

The Paging interface supports the following features:

- Transformer coupled front end for audio signal with and 600 Ohms AC input impedance.

- DC load (as in an LS trunk) for 3rd party paging unit control.
- 1 FORM C relay for 3rd party paging unit control.
- Paging and answer back audio path can operate in full duplex mode.

Generic Relays

The SX-200 ICP controller provides relays for activating customer-supplied devices through a DB-9 connector (labeled Relay Port) on the front panel of the MX controller and RJ-11 connectors (labeled Relay) on the back panel of the CX controller.

Relay ports can be used for activating the following customer-supplied devices:

- Solenoid for locking and unlocking a door (Additional contacts from the RJ45 connector on the back panel connect to a door bell button or other signaling device.)
- Auxiliary Ringer (Night Bell)
- Alarm Device (typically a light with activated in response to a CRITICAL alarm condition.

All devices require a customer-supplied power source.

Specifications

ONS Lines

ONS Circuit Specifications	
Loop Detector Threshold:	2000 Ohms loop resistance
On-hook Tip/Ring Voltage:	-48 Vdc nominal
External Loop Resistance:	600 Ohms, including phone resistance
Nominal Ringing Voltage:	20 Hz sinusoidal, 55 Vrms with 10 Vdc offset
Maximum Ringer Load	2 REN for AOB and AMB (all versions of MX, and v1 to v3 of CX/CXi)
External Loop Length: 22 AWG (23 IWG) 24 AWG (25 IWG) 26 AWG (27 IWG)	3800 m (12350 ft.) with 200 Ohm phone 2300 m ((7580 ft.) with 200 Ohm phone 1500 m (4875 ft.) with 200 Ohm phone

Note: High-voltage Message Waiting lamp is NOT supported on these circuits.

ONS Transmission Parameters	
AC Impedance:	600 ohms
Balance Impedance:	600 ohms
Ringing Frequency:	20 Hz

LS Trunks

LS Trunk Signaling Protocols and Parameters	
Signaling Protocol (NA):	TIA-912-A
Min. operating loop current:	18 mA
Max operating loop current:	100 mA
Loop Current Limit:	145 mA ±15 mA
Ring detector Threshold:	16 Vrms ±3 Vrms
Dummy Ringer load:	0.65 REN
Reversal detector:	Detects CO battery polarity
Loop detect for CO disc. (no battery)	< 2V across Tip and Ring

For best performance, analog trunks to the local exchange or Central Office should operate with an attenuation in the range of 0 dB to 8 dB. Higher attenuation will degrade the signal level (both acoustic and DTMF dialing) and may make some connections—such as long-distance—difficult to use. Short loops provide the best performance. For additional information concerning analog local loop characteristics, refer to the "Analog Local Loop Characteristics" section of the Engineering Guidelines document which is provided in PDF format on the Mitel Edocs website (http://edocs.mitel.com).

The following trunk settings are recommended based on the amount of attenuation presented by the trunk line (measurable using the LS_MEASURE function in the Maintenance application):

Attenuation Levels for Short and Long CO Trunks	
Line Length (Trunk Circuit Descriptors Options Subform)	**Attenuation on Trunk Line**
Short Loop	0 dB to 3 dB
Long Loop	3 dB to 6 dB
Long Loop **(some additional signal loss will occur with lines of this length)**	6 dB to 8 dB **Caution: Near maximum attenuation.**
Long Loop **(additional signal loss will occur with lines of this length; some connections will be difficult to use)**	Greater than 8 dB **Not acceptable.**
X Long Loop **(this setting is only valid when used with AMB/AOB LS trunks)** This setting should only be used in situations where the attenuation on the trunk line is greater than 8 dB and the trunk is presenting usability problems. This connection provides additional gain with respect to the Long setting, but this additional gain is only applied on the signal transmitted from the CO to the ICP. This setting is identical to the Long setting for signals transmitted from the ICP to the CO.	Greater than 8 dB

For additional information on losses encountered in the PSTN and guidelines on connecting the SX-200 ICP to the PSTN, refer to the Engineering Guidelines document on the Mitel Edocs website (http://edocs.mitel.com).

For non-loaded loops, Bell (NA) recommends the following wire gauges:

Recommended Trunk Lengths by Wire Size

Wire Size	Length Limit
26 AWG	15kft (~5 km) ~0.53 dB/kft (~1.6 dB/km)
24AWG	18kft (~5.8 km)
22AWG	
19AWG	

Note: 26 AWG = 0.4mm
22 AWG = 0.6mm

LS Impedances

Description	Termination	Balance
600ohm	600ohms	600ohm
AT&T complex	600ohms	350 ohms + (1000 ohms//210 nF)
Centrex	600ohms	470 ohms + (800 ohms//150 nF)
DSL	600ohms	560 ohms + (1500 ohms//330 nF)
Loaded Loops (Impedance A)	N/A	N/A

Notes:

- All North American CO trunks support the 600 Ohm transhybrid balance setting and the “AT&T” complex setting.
- The 600 Ohm setting is most often used for very short connections, such as those behind a PBX, IAD or similar device.
- “AT&T” Complex matches most unloaded central office lines.It is the most commonly used setting.

Music On Hold

MOH Electrical Specifications

Input impedance	600 Ohms nominal
Gain	Fixed at -19dB A/D
Input signal range	10 to 100 mVrms
Maximum input voltage	SELV limits
Maximum input audio level without clipping	0.4 Vrms (mono input)/0.2 Vrms (each stereo input), THD @ -40 dB
Maximum PCM level	-15 dBm

Paging

Paging Electrical Specifications	
AC input impedance	600 Ohms nominal
DC load	300 to 100 Ohms for loop current 20 mA to 90 mA
Maximum DC current	90 mA with PAGE1 and PAGE2 leads
Maximum input voltage	SELV limits to all connections
A to D gain	+3 dB
D to A gain	-3 dB
D to D loss	> 25 dB @ 1 kHz
Maximum relay rating	0.3 A 125 Vac, 1 A 30 Vdc
Auxiliary paging control	1 FORM C relay contact

Generic Relays

The relay contacts are rated at 0.3 A 125 Vac, 1 A 30 Vdc peak. Normally open and normally closed contacts are provided for each relay (1 FORM C). Connection of the device must be through a customer-supplied auxiliary relay if the total current requirement exceeds the relay contact ratings.

Cabling and Cross Connections

MX controller - see Analog Main Board / Analog Option Board Tip & Ring Assignments
CXCXi controller - see Analog Main Board Pinouts and Analog Option Board Pinouts

RS-232 Ports (MX, CX/CXi and AX)

The RS-232 serial ports allow printers and maintenance terminals to be connected directly to the SX-200 ICP with DB9 connectors. The MX has three ports labeled Printer, Maintenance, and Relay. The CX/CXi and AX have a single port labeled Maintenance which can be used for maintenance functions.

The ports default to

- 9600 bits/s
- 8 data bits
- no parity bit
- 1 stop bit

Pinouts

See RS-232 Port Pinouts (MX, AX and CX)

10/100Base-TX Ethernet Port (MX and AX)

The MX provides a single LAN port and the AX provides two LAN port, which support Ethernet at either 10Base-T or 100Base-TX (both duplex and half-duplex). The port is compliant with IEEE 802.1 and requires an 8-way RJ-45 connector with Category 5 cabling.

Features

- Auto-MDIX: automatically adjusts for straight-through or crossover cables
- Port Speed Auto-Negotiation: the port will negotiate with the far end to select either 10Base-T or 100Base-TX transmission rates and half or full duplex operation

Pinouts

See 10/100Base-TX Ethernet Port Pinouts (MX/CX/CXi/AX).

LEDs

See Ethernet LEDs.

10/100 WAN Port (CX/CXi)

The CX/CXi provides a single WAN port which supports Ethernet at either 10Base-T or 100Base-TX for connection to a wide area network, an xDSL modem or a cable modem. The port is compliant with IEEE 802.3 and requires an 8-way RJ-45 connector with Category 5 cabling.

Features

- Auto-MDIX: automatically adjusts for straight-through or crossover cables
- Port Speed Auto-Negotiation: the port will negotiate with the far end to select either 10Base-T or 100Base-TX transmission rates and half or full duplex operation (or manually programmable)

Pinouts

See 10/100Base-TX Ethernet Port Pinouts (MX and CX/CXi).

LEDs

See 10/100 WAN, 10/100/1G LAN, 10/100 802.3af LAN Port LEDs.

10/100 802.1af LAN Ports (CXi)

The CXi includes a 16-port managed Layer 2 Ethernet switch. Each port supports auto-negotiation and is able to connect to 10Base-T or 100Base-TX devices and determine the appropriate data rate from information provided to it. Duplex mode and flow control are also programmable for each port. The ports use 8-way RJ-45 connectors that conform to Category 5 wiring standards. Although Category 5 cabling is recommended, Category 3 cabling can be used for limited installations. See Category 3 Cabling Guidelines for details.

The 16 ports comply with the 802.3af Power over Ethernet (PoE) specification, which enables them to deliver power to IP phones and other Ethernet devices over Category 3 or 5 cabling. As a minimum, the PoE module provides enough power for 16 IP phones, some of which may be fitted with PKMs. For details about PoE and planning a PoE installation, see the Engineering Guidelines document on the Mitel Edocs website (http://edocs.mitel.com).

The internal switch uses 802.1p/Q VLAN prioritization to ensure the quality of voice calls, forwarding packets with priority value 6 (from IP phones) ahead of packets with priority value 0 (from PCs and other IP devices). By default, all traffic is on the default VLAN (1), but if necessary, voice and data traffic can be assigned to separate VLANs.

Features

- Managed switch functions: support for 802.1p/Q prioritization on a single VLAN (1), or on two VLANs (voice and data)
- Auto-MDIX: automatically adjusts for straight-through or crossover cables on all ports
- Port speed auto negotiation: the port will negotiate with the far end device to select either 10Base-T or 100Base-TX transmission rates and half or full duplex operation (or manually programmable)
- Flow control: ensures reliable communication during full-duplex operation

Note: The 10/100 802.3af LAN port LEDs indicate Ethernet status, not power status.

Pinouts

See 10/100Base-TX Ethernet Port Pinouts (MX and CX/CXi).

LEDs

See 10/100 WAN, 10/100/1G LAN, 10/100 802.3af LAN Port LEDs.

10/100/1G LAN Port (CX)

In applications where the IP connectivity requirements exceed the capability of the CX, or of the onboard 16-port Layer 2 switch on the CXi, the 10/100/1G allows for connection of expansion switches. By attaching one 48-port switch, or two 24-port switches linked in a daisy-chain, system capacity increases from 16 to 64 ports.

The expansion Layer 2 switches can be simple or sophisticated. Simple, unmanaged switches should be installed in voice-only networks (networks that support IP phones, but not PCs). Sophisticated, VLAN-capable switches should be installed in voice and data networks (networks that support both IP phones and PCs). VLAN capability is required to prioritize voice traffic over data traffic. Optionally, the expansion switches can support the 802.3af standard, which enables them to deliver Power over Ethernet (PoE) to connected devices such as Mitel IP phones. For more information, see Install a Layer 2 Switch for Line Expansion.

The 10/100/1G LAN port is compliant with IEEE 802.3, supports auto-negotiation, and can connect to devices operating at 10BaseT, 100BaseTX, or 1000BaseT (1 Gigabit). The port uses an 8-way RJ-45 connector and Category 5 cabling.

Notes:

1. From a programming and management perspective, the 10/100/1G LAN port functions as the seventeenth port of the onboard Layer 2 switch on the CXi.
2. Although the 10/100/1G LAN port is designed for connection of an expansion Layer 2 switch, you can connect an IP phone, PC, or other Ethernet device to it. The 10/100/G port does not supply Power over Ethernet (PoE) to connected devices.
3. The 10/100/1G LAN port is the only supported method for expanding the number of switch ports. Expansion switches should not be attached to the 10/100 802.3af LAN ports of the CXi.

Features

- Managed switch functions: support for 802.1p/Q prioritization on a single VLAN (1), or on two VLANs (voice and data)
- Auto-MDIX: automatically adjusts for straight-through or crossover cables on all ports
- VLAN tagging: support for IEEE 802.1q tagging (disabled by default)
- Port speed auto negotiation: the port will negotiate with the far end to select either 10Base-T, 100Base-TX or 1000Base-T transmission rates and half or full duplex operation (or manually programmable)
- Flow control: ensures reliable communication during full-duplex operation

Pinouts

See 10/100/1000Base-TX LAN Ethernet Port Pinouts.

LEDs

See 10/100 WAN, 10/100/1G LAN, 10/100 802.3af LAN Port LEDs.

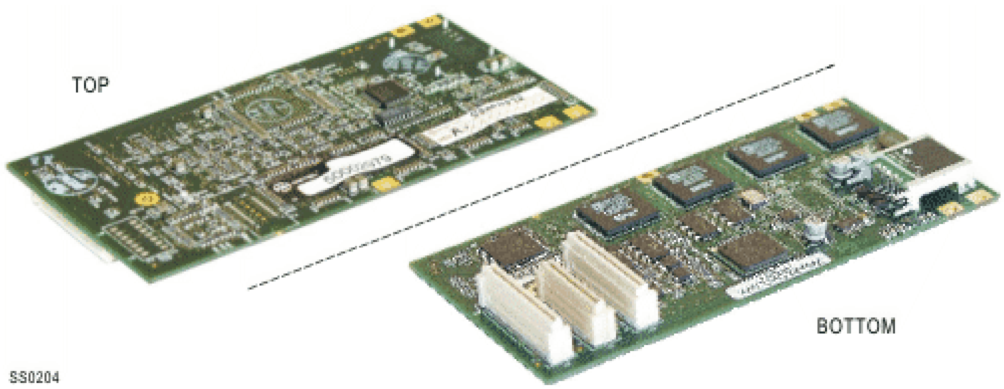
DSP Resources

The MX, CX, CXi, AX and SX-200 Peripheral Cabinet can be configured with a variety of DSP (Digital Signal Processor) resources. The MX supports up to two DSP modules (either Dual or Quad) and is shipped with a Dual DSP module. The CX/CXi has two DSPs on its main board and supports the addition of a T1/E1 Combo module and/or either Dual or Quad DSP module. The AX has four DSPs on its main board and supports the addition of a T1/E1 Combo module and/or either Dual or Quad DSP module.The SX-200 Peripheral Cabinet supports a Single DSP module.

DSPs provide the following functionality:

- CLASS tones for the ONS/CLASS circuits on the Embedded ASU module. CLASS generator resources that are assigned and released dynamically as they are required.
- Conference bridges for user conferences and the Record-a-Call feature. A Record a Call conference is between two parties and a voice mail port.
- DTMF receivers that can be used system wide.
- Voice compression resources

Quad DSP Module



DSP Configuration Options

Two tables, below, list the DSP requirements for the MX and CX/CXi controllers. The tables are guidelines only. The actual number of DSPs required will vary depending on the intended use of the system. As a rule of thumb, a 100-port or smaller system with low traffic will function with one Dual DSP. However, the same size system with heavy traffic will require additional DSP resources to keep performance from degrading. To calculate the number of DSPs that your system requires, refer to the SX-200 ICP Engineering Guidelines document which is provided in PDF format on the Mitel Edocs website (http://edocs.mitel.com).

Notes:

1. The number of conference, voice mail, and compression resources is fixed by the purchased option and the number of DSP devices available; the other values are adjustable. Additionally, the absence of a resource type, such as IP licences in the Hospitality Option (MX controller), does not mean that IP phones cannot be installed.

2. Installation of a hard drive is strongly advised for systems with more than eight voice mail ports or when Record a Call is frequently used. The drive must be purchased from Mitel (see Install FRUs for part number); drives obtained elsewhere are not supported.
3. The SX-200 ICP supports the G.711 and G.729a compression codecs which enable more devices to share available bandwidth. The G.711 PCM audio codec for 64kbits/s generally provides the best voice quality and is comparable to TDM type connections. The G.729a speech codec for 8kbps provides a good reduction in bandwidth with only minor loss in voice quality. A purchasable FOR option controls the number of G.729a codecs available to IP devices in the system. The option is purchasable in multiples of 8 to a maximum of 16 (CX/CXi) or 24 (MX). The default value is 0. The quantity entered must exactly match the quantity on the FOR sheet.
4. The base CX/CXi contains an Analog Main Board (AMB) that supports 4 ONS and 6 LS circuits. By installing an Analog Option Board (AOB), capacity doubles to 8 ONS and 12 LS circuits. The AOB does not require extra DSP resources.
5. The base MX contains an Analog Main Board (AMB) that supports 2 ONS, 2 DNIC and 6 LS circuits. By installing an Analog Option Board (AOB), capacity increases to 4 ONS, 2 DNIC and 12 LS circuits. Additional circuits are available by connecting peripheral cabinets to the MX. The AOB does not require extra DSP resources.
6. If compression is enabled on a CX/CXi with four DSPs (base system + Dual DSP), the number of conferences and voice mail ports will be reduced to 3 and 4, respectively.
7. All T1 trunk quantities include any combination of T1/D4 or T1/PRI.
8. The maximum system capacity is 672 TDM (DNIC and ONS) ports. In any option configuration, trunks may be added up to a maximum of 8 digital links (192 trunks) but only by reducing the number of digital bays (ONS and DNIC ports) connected, so that the total number of TDM ports does not exceed that shown in the table.
9. If system option 82 is disabled (Use Hardware Echo Canceller) then one DSP device is removed from the available pool. The number of TDM resources (voice mail and conference) will be reduced. This option cannot be used in a base system with compression enabled (Business Option 1) or with a large number of TDM devices (Analog Options 1 and 2).

MX Controller DSP Options

The base MX is shipped with one Dual DSP module in MMC slot 3. You can replace this module with a Quad DSP, and you can install a Dual or Quad DSP in MMC slot 2. The MX can have a minimum of 2 and a maximum of 8 DSP resources.

The table below lists the DSP requirements for each of the DSP configuration options. The requirements are guidelines only; the actual number of DSPs required depends on the intended use of the system.

	Number of DSPs Installed (MX Controller)			
Option Type	**Base Dual DSP (2 DSPs total)**	**2 Dual DSP or 1 Quad DSP (4 DSPs total)**	**1 Dual + 1 Quad DSP (6 DSPs total)**	**2 Quad DSPs (8 DSPs total)**
Business Option 1 (IP)	3 three-party conf. 4 voice mail ports 8 G.729 channels 48 IP phones 6 ONS/DNIC ports 12 LS/Class trunks	8 three-party conf. 12 voice mail ports 18 G.729 channels[1] 96 IP phones 6 ONS/DNIC ports 12 LS/Class trunks 24 T1 or 23 PRI ccts.	12 three-party conf. 18 voice mail ports 16 G.729 channels 96 IP phones 96 ONS/DNIC ports 12 LS/Class trunks 48 T1 or 46 PRI ccts.	12 three-party conf. 24 voice mail ports 24 G.729 channels 192 IP phones 192 ONS/DNIC ports 12 LS/Class trunks 96 T1 or 92 PRI ccts.
Business Option 2 (IP)	8 three-party conf. 8 voice mail ports 0 G.729 channels 48 IP phones 6 ONS/DNIC ports 12 LS/Class trunks	12 three-party conf. 18 voice mail ports 0 G.729 channels 96 IP phones 6 ONS/DNIC ports 12 LS/Class trunks 24 T1 or 23 PRI ccts.	18 three-party conf. 24 voice mail ports 8 G.729 channels 96 IP phones 96 ONS/DNIC ports 12 LS/Class trunks 48 T1 or 46 PRI ccts.	21 three-party conf. 24 voice mail ports 16 G.729 channels 192 IP phones 288 ONS/DNIC ports 12 LS/Class trunks 96 T1 or 92 PRI ccts.

	Number of DSPs Installed (MX Controller)			
Hospitality Option (IP+TDM)	8 three-party conf. 8 voice mail ports 0 G.729 channels 96 ONS/DNIC ports 12 LS/Class trunks	8 three-party conf. 12 voice mail ports 8 G.729 channels 48 IP phones 96 ONS/DNIC ports 12 LS/Class trunks 48 T1 or 46 PRI ccts.	12 three-party conf. 18 voice mail ports 16 G.729 channels 96 IP phones 192 ONS/DNIC ports 12 LS/Class trunks 48 T1 or 46 PRI ccts.	12 three-party conf. 24 voice mail ports 16 G.729 channels 248 IP phones 384 ONS/DNIC ports 12 LS/Class trunks 96 T1 or 92 PRI ccts.
Analog Option 1	2 three-party conf. 6 voice mail ports 0 G.729 channels 24 IP phones 288 ONS/DNIC ports 12 LS/Class trunks 48 T1 or 46 PRI ccts.	8 three-party conf. 18 voice mail ports 0 G.729 channels 48 IP phones 288 ONS/DNIC ports 12 LS/Class trunks 72 T1 or 69 PRI ccts.	12 three-party conf. 24 voice mail ports 0 G.729 channels 96 IP phones 288 ONS/DNIC ports 12 LS/Class trunks 72 T1 or 69 PRI ccts.	21 three-party conf. 24 voice mail ports 0 G.729 channels 192 IP phones 384 ONS/DNIC ports 12 LS/Class trunks 96 T1 or 92 PRI ccts.
Analog Option 2	2 three-party conf. 4 voice mail ports 0 G.729 channels 24 IP phones 384 ONS/DNIC ports 12 LS/Class trunks 48 or 46 PRI ccts.	10 three-party conf. 12 voice mail ports 0 G.729 channels 48 IP phones 384 ONS/DNIC ports 12 LS/Class trunks 48 or 46 PRI ccts.	12 three-party conf. 16 voice mail ports 0 G.729 channels 48 IP phones 480 ONS/DNIC ports 12 LS/Class trunks 48 or 46 PRI ccts.	21 three-party conf. 24 voice mail ports 0 G.729 channels 96 IP phones 480 ONS/DNIC ports 12 LS/Class trunks 48 or 46 PRI ccts.
Analog Option 3 (Requires Quad DSP)		8 three-party conf. 12 voice mail ports 0 G.729 channels 96 IP phones 576 ONS/DNIC ports 12 LS/Class trunks 96 or 92 PRI ccts.		

CX/CXi Controller DSP Options

The base CX/CXi has two DSPs built into its main board. You can add DSP resources to the CX/CXi by installing a single T1/E1 Combo module in MMC slot 1 or 2, and/or a Dual or Quad DSP module in MMC slot 3. The CX/CXi has a minimum of 2 and a maximum of 7 DSP resources.

The table below lists the DSP requirements for each of the DSP configuration options. The requirements are guidelines only; the actual number of DSPs required depends on the intended use of the system.

Number of DSPs Installed (CX/CXi Controller)			
Business Option 1			
Without Compression:			
Base System (2 DSPs total)	**Base System + T1/E1 Combo (3 DSPs total)**	**Base System + Dual DSP (4 DSPs total)**	**Base System + Dual DSP + T1/E1 Combo (5 DSPs total)**

Base System +Quad DSP

(6 DSPs total)	Base System +Quad DSP + T1/E1 Combo (7 DSPs total)				
• 3 three-party conf. • 4 voice mail ports • 0 G.729 channels • 24 IP phones • 8 ONS phones • 12 LS/Class trunks • 0 T1 cct	• 10 three-party conf. • 8 voice mail ports • 0 G.729 channels • 64 IP phones • 24 ONS phones • 12 LS/Class trunks • 24 T1 or 23 PRI ccts.	• 10 three-party conf. • 16 voice mail ports • 0 G.729 channels • 40 IP phones • 40 ONS phones • 16 LS/Class trunks • 0 T1 cct	• 10 three-party conf. • 16 voice mail ports • 0 G.729 channels • 80 IP phones • 150 ONS phones • 12 LS/Class trunks • 24 T1 or 46 PRI ccts.	• 10 three-party conf. • 16 voice mail ports • 0 G.729 channels • 80 IP phones • 150 ONS phones • 36 LS/Class trunks • 0 T1 ccts.	• 10 three-party conf. • 16 voice mail ports • 0 G.729 channels • 100 IP phones • 150 ONS phones • 16 LS/Class trunks • 24 T1 or 23 PRI ccts.
With Compression:					
—					

—	• 3 three-party conf. • 4 voice mail ports • 8 G.729 channels • 40 IP phones • 8 ONS phones • 12 LS/Class trunks • 0 T1 ccts	• 10 three-party conf. • 16 voice mail ports • 8 G.729 channels • 80 IP phones • 100 ONS phones • 12 LS/Class trunks • 24 T1 or 23 PRI ccts	• 10 three-party conf. • 16 voice mail ports • 8 G.729 channels • 80 IP phones • 56 ONS phones • 36 LS/Class trunks • 0 T1	• 10 three-party conf. • 16 voice mail ports • 16 G.729 channels • 100 IP phones • 130 ONS phones • 16 LS/Class trunks • 24 T1 or 23 PRI ccts

Note: Not all maximum values for lines and trunks can be realized simultaneously.

AX Controller DSP Options

The AX has four DSPs built into its main board. You can add DSP resources to the AX by installing a single T1/E1 Combo module in MMC slot 1. A Dual or Quad DSP module can be installed in MMC slot 2. The AX has a minimum of 4 and a maximum of 8 DSP resources.

Hospitality			
Base (4 DSPs)	**Base + Dual T1/E1**	**Base + T1/E1 Combo**	**Base + Dual T1/E1 +**

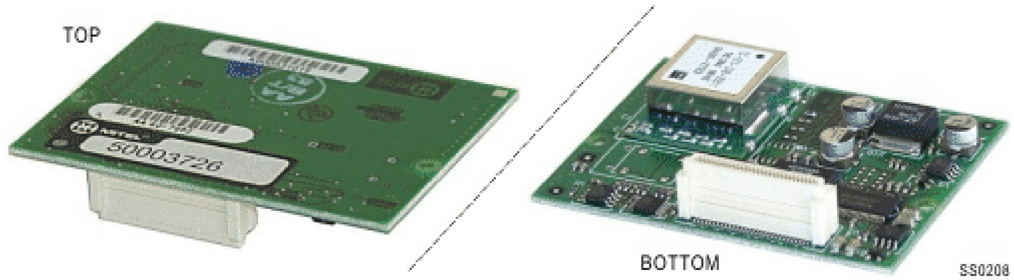
	(4 DSPs)	**(5 DSPs)**	**Quad DSP (8 DSPs)**
10 three-party conf. 20 voice mail ports 0 G.729 channels 200 IP phones 192 ONS phones 48 LS/CLASS trunks 0 T1 ccts	10 three-party conf. 20 voice mail ports 0 G.729 channels 248 IP phones 288 ONS phones 48 LS/CLASS trunks 48 T1 or 46 PRI ccts	10 three-party conf. 20 voice mail ports 8 G.729 channels 248 IP phones 288 ONS phones 24 LS/CLASS trunks 24 T1 or 23 PRI ccts	10 three-party conf. 20 voice mail ports 30 G.729 channels 248 IP phones 288 ONS phones 48 LS/CLASS trunks 48 T1 or 46 PRI ccts

Stratum Clock

If the system is connected to digital (T1 or BRI) trunks, a stratum clock is required to set the system clock and synchronize with the incoming digital frame rate clock. Two versions are available as options:

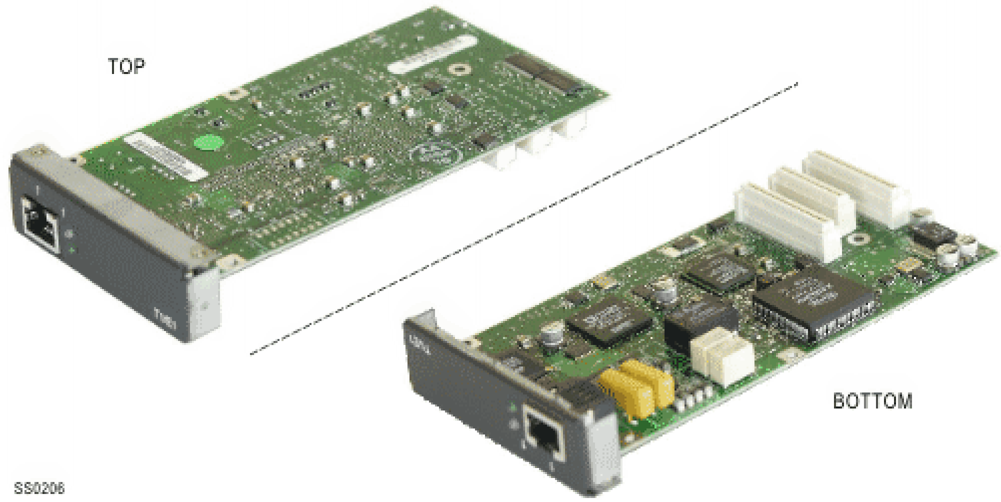
Stratum 3 Clock module (MX and CX/CXi)

Installed on the main board of the MX or CX controller, the stratum 3 clock module handles clock synthesis for most networks.



T1/E1 Combo module (AX and CX/CXi)

Installed in MMC slot 1 or 2 of the CX/CXi controller and slot 1 on the AX controller, the T1/E1 Combo module provides trunking, DSP, and stratum clock functionality in a single card. Its stratum 4 clock handles clock synthesis for some networks (including most North American networks).



System Identity Module/Button

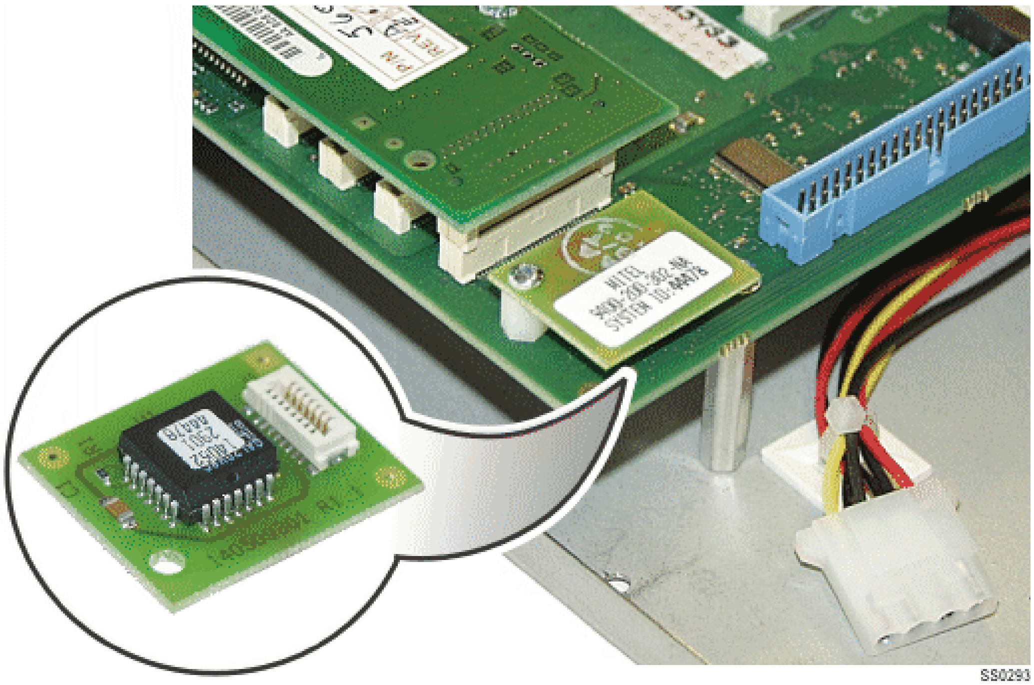
The System ID Module (MX) and System i-Button (CX/CXi or AX) enable the system options. They are located on the digital main board of the controller, and are required at power on to load software. Removal of the Module or i-Button generates a MOSS alarm.

The Module and i-Button are factory programmed with a System Identity Code. This code displays on CDE Form 04 as option 101, and must match the number printed on your Mitel Feature Options Record (FOR) sheet. The MX code is 5 numeric digits; the CX/CXi or AX code is 12 hexadecimal characters.

In the event of a non-recoverable system failure, the Module or i-Button can be moved to a new controller along with the internal media (CompactFlash or system hard drive). The new controller can then be started with the same settings, database, and options as the original system.

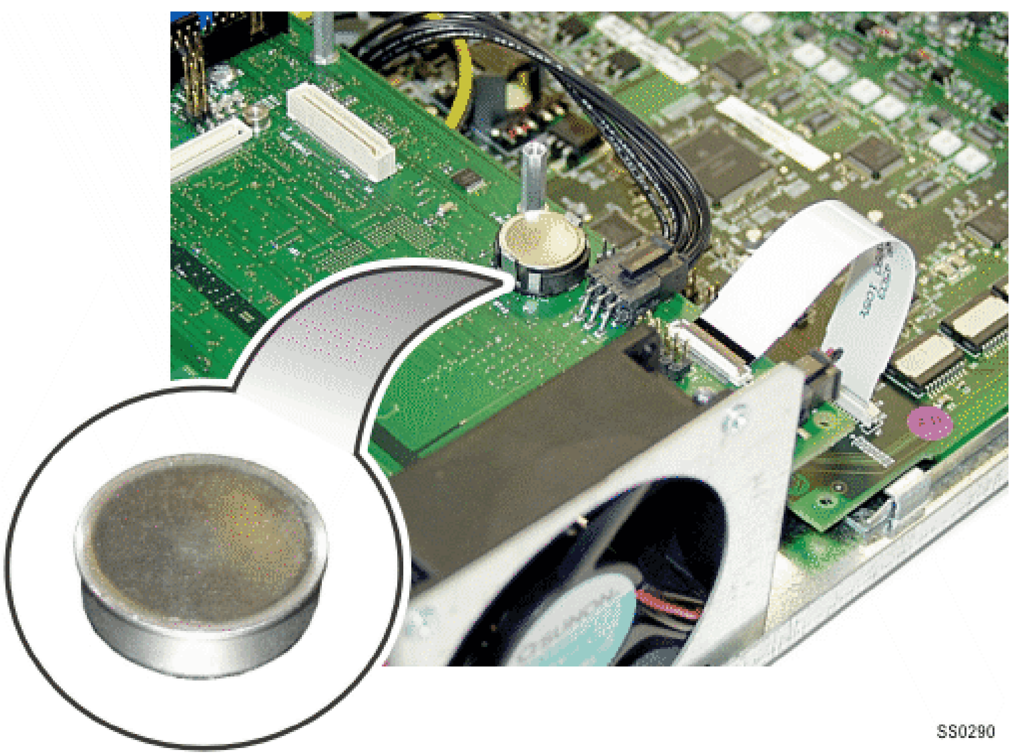
Note: EL/ML systems also contain System ID Modules; however, these cannot be installed in MX Controllers.

System ID Module (MX)



SS0293

System i-Button (CX/CXi and AX)



SS0290

Network Services Unit (NSU)

Description

The NSU provides T1 connectivity and supports up to two T1 links per unit.

Note: The NSU is supported by the SX-200 ICP MX controller only.

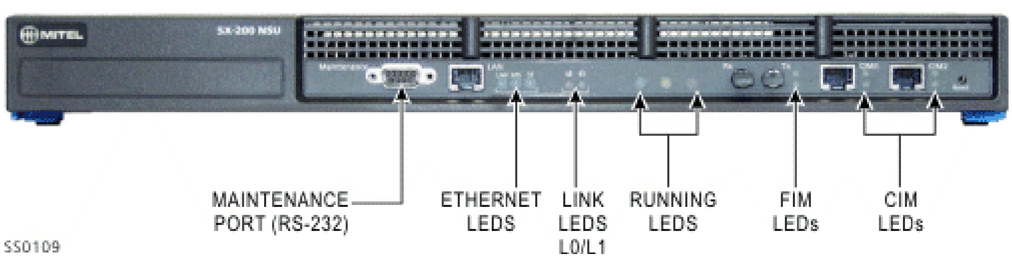
The protocols supported by the T1 interfaces are

- T1 CCS - Primary Rate ISDN (NI2_STANDARD, NI2_5ESS, NI2_GTD5), and QSIG (QSIG_ISO), DMS 100/250, 4ESS

NSUs are connected directly to the controller through a FIM or CIM cable with up to four NSUs being connected for additional digital trunk capacity.

Front panel

- RS-232 serial port (DB9 connector to a PC) for maintenance purposes such as field installation, database upgrade, access to logs, and modem connection for remote access
- Ethernet port (RJ-45 connector) for future use
- Faceplate LEDs - Miscellaneous, Link Status, and Message Link Controlled
- FIM port for fiber connection to the SX-200 ICP Controller



Rear panel

- Two T1/ E1 ports (RJ-45 connectors for T1; RJ-45 or ground and coax for E1) for network connection
- Two hybrid port status LEDs
- Two hybrid port DIP switch complexes
- Power connector
- Protective ground (for grounding the chassis).

DIP Switch Settings

NSU L0/L1 DIP Switch Settings

DIP Switch	Function	Settings	Notes
1	Tx Ground	Down: Ground Up: Floating	Set to up.
2	Rx Ground	Down: Ground Up: Floating	Set to up.

NSU L0/L1 DIP Switch Settings			
3	Not supported		Set to up.
4	T1 cable selector (T1/D4)	Down: Enabled Up: Disabled	Set to down (100 ohm impedance)
5	Not supported		Set to up
6	Line/Network Termination selector	Down: Line Up: Network	

Specifications

Dimensions

Height: 1.75 in. (4.45 cm) (1 U)

Width: 17.75 in. (45.1 cm) (19" rack mountable)

Depth: 15.5 in. (39.4 cm)

Weight: 9.4 lb (4.3 kg)

Environment

Table: Storage Environment

Condition	Specification
Temperature	-40º to 150ºF (-40º to +66ºC)
Humidity	15-95% Relative Humidity, non-condensing
Vibration	

0.5 g, 5 to 100 Hz, any orthogonal axis

1.5 g, 100 to 500 Hz, any orthogonal axis
Mechanical Stress

One 15.3 cm (6 in.) drop, each edge and corner adjacent to the rest face –unpackaged

One 76.2 cm (30 in.) drop, each edge and corner packaged in cardboard & foam

Table: Operational Environment

Condition	Specification
Temperature	39º to 120ºF (4º to 49ºC)
Humidity	34-95% Relative Humidity, non-condensing
Maximum Heat Dissipation - fully loaded (see Note)	170 BTUs per hour
Conversion factors: 1 watt is equal to 3.412 BTUs per hour, 1 ton of refrigeration is equal to 12,000 BTUs per hour or 3.516 Kilowatts, and 3/4 Kilowatt-hour is equal to 1 ton of refrigeration.	

Power Supply

Input / disconnect:	IEC 320 - C14 Class 1 AC Receptacle
Operation:	100-120/200-240 V ac auto selectable
Maximum power input:	60 W
AC source:	90 - 132 Vac; 47 - 63Hz in North America

Pinouts

See NSU Pinouts for T1 Line/Network Termination.

Analog Services Unit (ASU)

Description

The Analog Services (ASU) unit provides 24 ONS CLASS circuits. Up to three ASUs can be connected to a CX/CXi Controller and up to six can be connected to an MX controller.

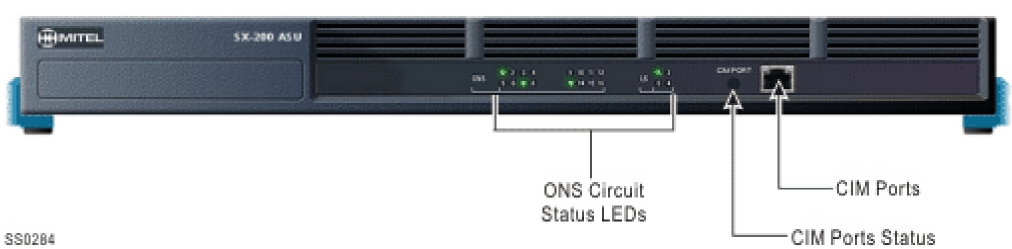
The ASU connects to the controller using a Category 5 Unshielded Twisted Pair (UTP) cross-over cable through a CIM interface.

Notes:

1. The SX-200 ICP does not support the Mitel Universal ASU. Only the SX-200 ICP ASU and ASU II (which has ONS and LS ports) are supported.
2. The ASU only supports DTMF telephones (pulse or rotary dial phones are not supported, except in TDM bays).

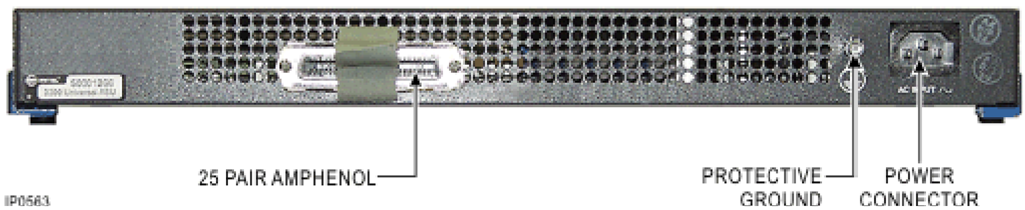
Front Panel

- 24 ONS Circuit LEDs indicate the status of the telephone circuits
- 1 CIM circuit LED indicates the status of the CIM link
- RJ-45 connector (CIM connection to the Controller).



Back Panel

- 25 pair D-type connector provides access to the ONS Tip/Ring or A/B circuits.
- Standard Male IEC AC input connector for power requirement.



Specifications

Dimensions

Height: 1.75 in. (4.454 cm) (1 U)

Width: 17.75 in. (45.1 cm) (19" rack mountable)

Depth: 15.5 in. (39.4 cm)

Weight: 10.61 lb (4.81 kg)

Environment

Table: Storage Environment

Condition	Specification
Temperature	-40º to 150ºF (-40º to +66ºC)
Humidity	15-95% Relative Humidity, non-condensing
Vibration	0.5 g, 5 to 100 Hz, any orthogonal axis 1.5 g, 100 to 500 Hz, any orthogonal axis
Mechanical Stress	One 15.3 cm (6 in.) drop, each edge and corner adjacent to the rest face –unpackaged One 76.2 cm (30 in.) drop, each edge and corner packaged in cardboard & foam

Table: Operational Environment

Condition	Specification
Temperature	39º to 120ºF (4º to 49ºC)
Humidity	34-95% Relative Humidity, non-condensing
Maximum Heat Dissipation - fully loaded (see Note)	170 BTUs per hour
Conversion factors: 1 watt is equal to 3.412 BTUs per hour, 1 ton of refrigeration is equal to 12,000 BTUs per hour or 3.516 Kilowatts, and 3/4 Kilowatt-hour is equal to 1 ton of refrigeration.	

Power Supply

Note: Connector is a Standard Male IEC-320 AC input.

Voltage Universal input design, operating input voltages from 90-132/180-264 Vac

Input Power	60 W full rated load
Frequency	AC input frequencies from 47Hz to 63Hz
Holdover	With an input voltage of 120 Vac or 240 V ac under a full rated load, the power supply outputs remain in regulation for a minimum of 16ms after loss of AC mains input voltage.
Brown-Out Recovery	Recovers from an AC input brown-out or sag condition automatically.

ASU ONS Line Specifications

Features

- Maximum of 600-Ohm external loop drive capability. This equates to approximately one mile of loop range over 26-gauge cable terminated by a 150-Ohm set. Longer loops are supported but with the characteristics as described by the next bullet item.
- Constant current design with the loop current set at 25mA. If the loop range extends pass 600 Ohm, this circuit will revert to a voltage feed of approximately 2xVdc (design dependant parameter between 22-26 Vdc). The circuit remains active and the loop current will be dependant on the external loop impedance.
- Supports a Ringing Equivalent Number (REN) of 3.
- Capable of on-hook transmission. Used in conjunction with a centralized resource to deliver calling line ID.
- Positive disconnect (removal of battery from the ring lead).

Transmission Parameters (North America)

Input Impedance:	600 ohms
Balance Impedance:	600 ohms
Digital Coding:	ITU µ-law – MT8966 CODEC

ASU ONS DC Supervision Parameters (North America)

Battery Feed:	-30Vdc feed, constant current set at 25mA +/- 1mA
Loop Resistance:	600 Ohms (includes set)
Loop Detect Threshold:	12mA
Flash Detect:	SW timed function from switch hook detector
Ground button detect threshold:	13mA Tip or Ring to ground in off hook state
Positive Disconnect:	SW timed function that breaks loop current

ASU ONS Ringing Parameters

Voltage:	65 Vrms sinewave superimposed onto –48Vdc
Frequency:	20Hz 25 Hz
Trip Battery:	
Silent interval:	-30Vdc
Ring Interval:	-50Vdc
Number of bridged ringers:	3
Max. bridged capacitance:	3uF//15kOhms
Ring Trip detect time:	
HW detector response <100ms	
HW ring trip overrides application of ringing signal	
SW ring trip response time:	Within 50ms of switch hook detect

ASU ONS Message Waiting Parameters

Voltage:	-115Vdc +/- 5V dc
Source Impedance:	Between 2k and 4K
MSW trip:	SW control, interlocks with application of ringing
Flash Rate:	Cadenced, SW controlled. 300ms on/1500ms off cont.

Cabling and Cross Connections

See ASU Tip and Ring Assignments.

Analog Services Unit II (ASU II)

Description

The ASU II supports up to 48 ONS phones or up to 8 LS trunks depending on how the unit is configured with peripheral cards:

- The **24 port ONSp** cards support:
  - 24 On-Premise Station (ONS) Lines for analog phones. The circuits on this card have

- The **16 port ONS and ONSp** cards support:
  - 16 On-Premise Station (ONS) Lines for analog phones

Note: The ONS ports on the ONSp Line Cards are augmented with protection circuitry, to allow ONS extensions to be used outside of a building, i.e. off premise. The ONSp ports meet the requirements of regulations 60950-01 and IEC 55024. Regular ONS ports are intended for on-premise applications only. ONSp ports use the same loss plan settings as regular ONS ports.

- The **4 + 12 port Combo** card supports:
  - 12 On-Premise Station (ONS) Lines for analog phones
  - Four Loop Start (LS) trunks for analog connection to a central office
  - Four System Fail Transfer (SFT) relays that provide direct connection between an analog telephone and a Loop Start trunk in the event of system or power failure.

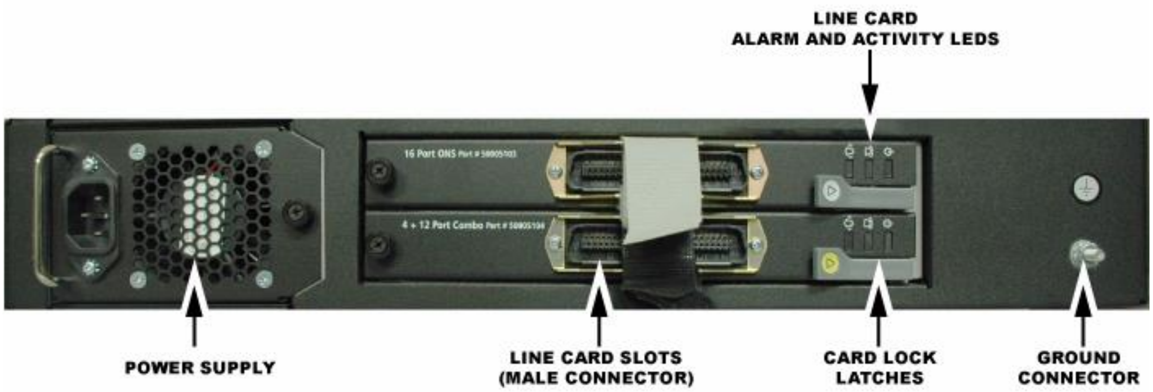
  The North American version supports Custom Local Access Signalling Services (CLASS) on the ONS circuits. CLASS allows the SX-200 ICP system to pass Calling Line ID digits and CLASS name information to display sets that suppport Caller ID functionality.

Front Panel

- Alarm, activity, and power LEDs
- CIM status LED
- One CIM port for connecting to the Controller

Back Panel

- Power supply and fan
- Two slots for peripheral line cards with amphenol connector (16 port ONS, 24 port ONSp, or 4+ 12 Combo)
- Card identifiers
- Protective ground for grounding the chassis



Specifications

Dimensions

Refer to ASU dimensions.

Environment

Refer to ASU environment

Power Supply

Refer to ASU Power Supply

ASU II ONS Line Specifications

On-Premise Station Lines

Each ONS line circuit provides:

- A Maximum of 600 ohm external loop drive capability. This equates to approximately one mile of loop range over 26-gauge cable terminated by a 150 ohm set.
- A Constant current design with the loop current set at 25 mA. If the loop range extends past 600 ohm, this circuit will revert to a voltage feed of approximately 2xVdc (design-dependent parameter between 22-26 Vdc). The circuit remains active and the loop current will be dependant on the external loop impedance.
- Support for a Ringing Equivalence Number (REN) of 2 for 16-port ONS/ONSp cards and a REN of 3 for 24-port ONSp.
- On-hook transmission capability. Use in conjunction with a centralized resource to deliver calling line ID.
- Positive disconnect (removal of battery from the ring lead).
- Ground button detection.
- Message waiting indication (dc voltage method and class message).
- A status LED.
- Low-level diagnostics.

Note: The MWI signal provided by the ASU II is intended to be used with phones that support 90V signalling.

Loop Start Trunks

Each LS trunk circuit provides:

- Loop disconnect and loop-reversal detection
- Incoming ringing detection
- A status LED
- Low-level diagnostics
- Power Cross-protection as specified by CSA/UL 950 Safety Specifications
- Lightning Protection as specified by FCC Part 68/CS-03.

System Fail Transfer

The ASU II with a Combo card has four System Fail Transfer (SFT) relays, one per LS trunk circuit.The relays are activated by power loss (power fail transfer) or by software directed transfer (SFT).

If any of the following events occur, the relays in the ASU II Combo card close:

- Failure of the controller
- Interruption of the system AC power
- Loss of the CIM link between the controller and ASU.

When the relays close, the first four ports on the ASU II Combo card are automatically connected to the four LS trunk circuits. After power is restored and communication is re-established with the controller, the relays open and the ONS and LS circuits function normally.

The four System Fail Transfer ONS ports are physically mapped to four LS trunks as follows:

LS Trunk Circuits	ONS Port Circuits
1	1
2	2

3	3
4	4

You should plan carefully where you want the four ONS ports to terminate within the network. If power is lost they will provide the only connectivity to the telephone network. The four LS trunks do not have to be dedicated to support system fail transfer. You can place them in a trunk group and use them to route calls to the central office during normal system operation.

ASU II ONS Transmission Parameters

ASU II ONS Transmission parameters are the same as those for Analog Boards. Refer to ONS Transmission Parameters.

ASU II LS Trunk Parameters

Refer to LS Trunks.

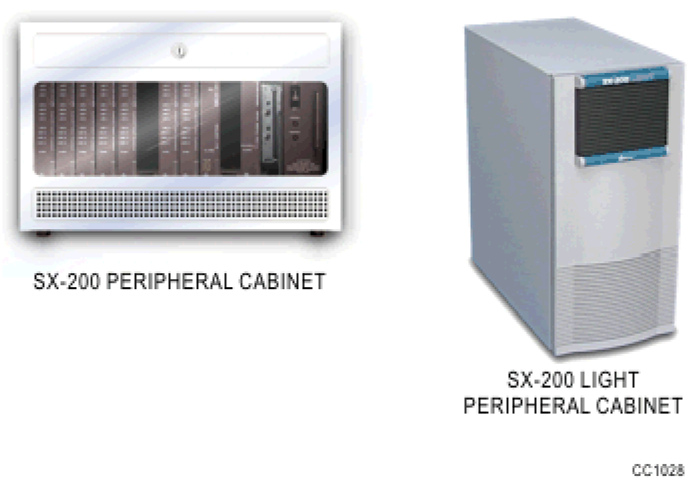
Cabling and Cross Connections

See ASU II D-Type Connector Pinout

SX-200 Peripheral Cabinets

Overview

SX-200 Peripheral Cabinets are supported on the SX-200 ICP MX only. Up to seven cabinets can be connected to the controller via CIM or FIM cables. The cabinets can be SX-200 ELx peripheral cabinets, SX-200 LIGHT peripheral cabinets, or a mix of both.

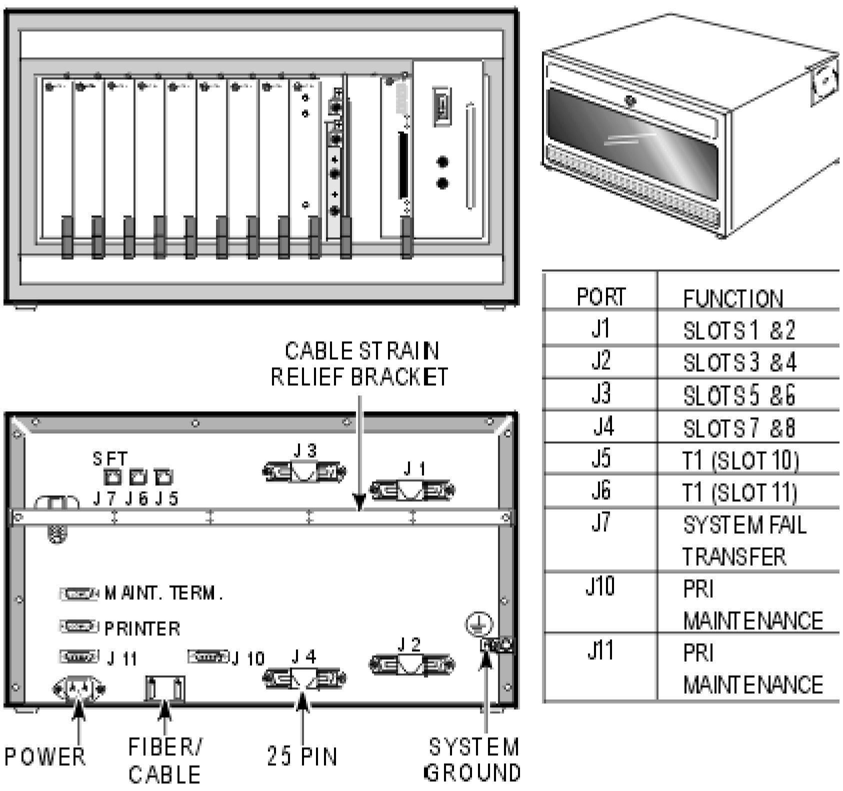


SX-200 ELx Cabinet

The SX-200 ELx peripheral cabinet can be mounted in a standard 19" rack, or they can be stacked. The cabinet is plastic and plexiglass. The door on the cabinet allows the system administrator to see the system status at a glance. The control cabinet and the peripheral bays are linked by fibre or copper cables.

Located on the rear of the SX-200 ELx cabinet are four 25 pin connectors (J1- J4 for the peripheral interface cards), three RJ-45 connectors (J5 and J6 for T1 trunks and J7 for a system fail transfer control port), a printer port, a grounding connector, a maintenance port, and two RS-232 connectors (J10 and J11 for the PRI maintenance).

SX-200 EL x CABINET



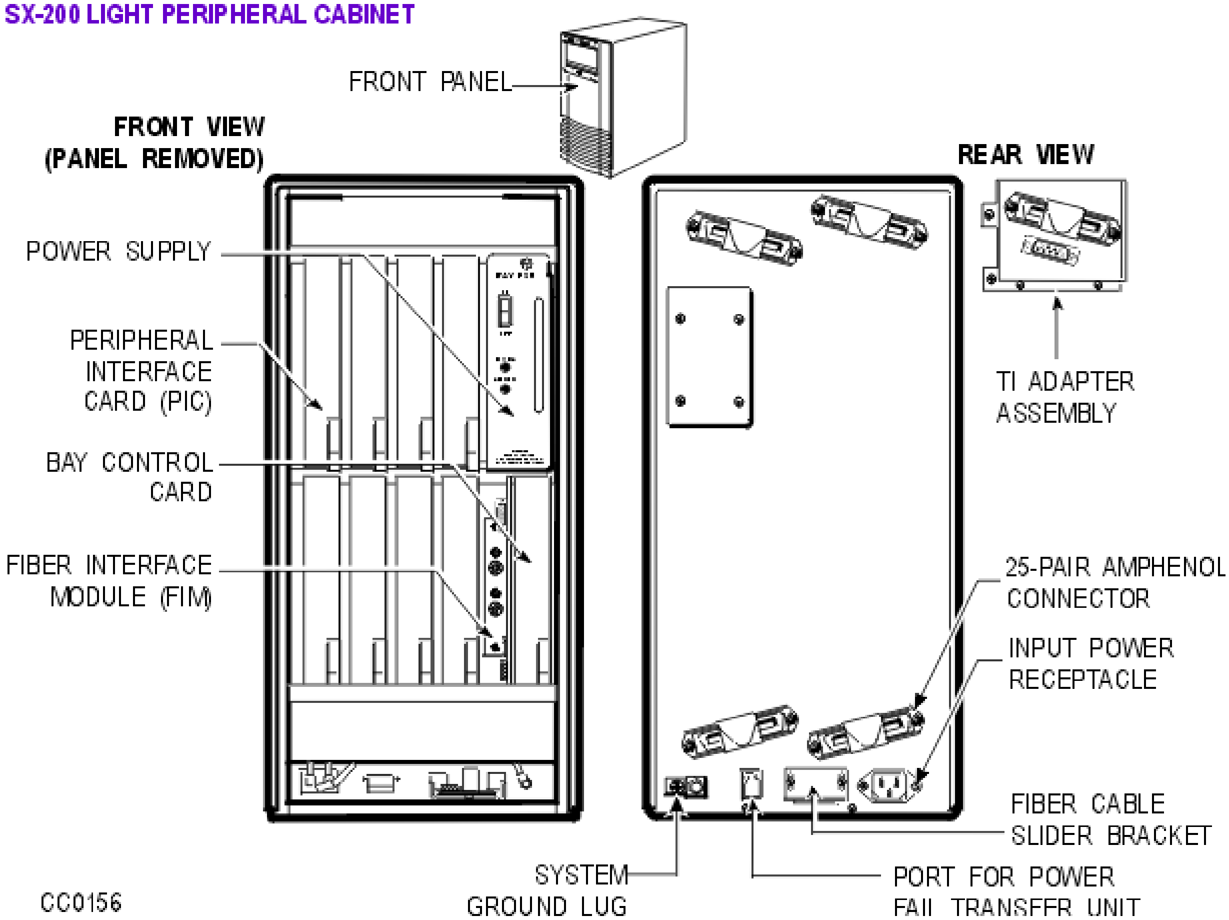
PORT	FUNCTION
J1	SLOTS 1 & 2
J2	SLOTS 3 & 4
J3	SLOTS 5 & 6
J4	SLOTS 7 & 8
J5	T1 (SLOT 10)
J6	T1 (SLOT 11)
J7	SYSTEM FAIL TRANSFER
J10	PRI MAINTENANCE
J11	PRI MAINTENANCE

The SX-200 ELx cabinet supports 12 card slots: eight slots support line and trunk cards, and four support the control cards and the FIM or CIM carrier cards.

SX-200 LIGHT Peripheral Cabinet

The SX-200 LIGHT Peripheral Cabinet is vertical as opposed to horizontal. The cabinet contains one Bay Power Supply, one Bay Control Card (with attached Peripheral FIM Carrier plus FIM), and up to eight peripheral interface cards.

SX-200 LIGHT PERIPHERAL CABINET



Peripheral Bay Power Supplies

The Bay Power Supply (BPS) is card-mounted and is located in the Peripheral cabinet. The electrical power characteristics are summarized in the following table.

Electrical Characteristics

Electrical Input Power Characteristics

	Input	Frequency	Minimum Holdover Time	Input Current
9109-008-000-SA Bay Power Supply	120 Vac	47 Hz to 63 Hz	40 ms at 120 Vac or (delivering full rated load)	Maximum of 2.5 Amps at 120 Vac
9109-008-003-NA Bay Power Supply	230 Vac	47 Hz to 63 Hz	40 ms at 230 Vac or (delivering full rated load)	Maximum of 2.0 Amps at 230 Vac

Physical Connections

The BPS connects to the backplane through a card-edge connector at the rear of the card. Also at the rear is an IEC receptacle which connects to a line cord from the system ac distribution.

Controls and Indicators

The ON/OFF switch is mounted on the front of the BPS and is used to turn the power on or off to the unit. The upper LED indicates that the BPS is operating, and the lower LED is ON when the ringing amplifier is producing power (flashing in cadence with it).

This figure shows the Bay Power Supply.

Input and Output Protection

The input to the converter is protected by a fuse, and by low voltage protection which shuts off the converter if the input voltage falls below the specified minimum. The converter will not be re-enabled until the input voltage returns to the specified minimum. The input also includes protection which limits the peak inrush current to 20 A.

Each output is protected against short circuits, overloads, and overvoltage. The overload/short circuit protection is self-resetting.

Power Fail Sense

The converter has a single alarm signal, PFS (power fail sense), which is driven low when the incoming ac falls below its minimum specified value. At this point there will be approximately 10 ms before the outputs fall out of regulation.

Peripheral Bay Circuit Card Descriptions

General

This section describes the cards that install in the SX-200 peripheral cabinets.

Bay Control Card II

Brief Description

One bay control card (BCC II), part number 9109-017-000, is required in each cabinet.

CAUTION: This card must not be inserted or removed with the power on.

Functions

The Bay Control Card II provides:

- Control of operations within the cabinet
- Monitoring of lines, trunks and other circuits within the bay; reports are sent to the SX-200 ICP controller via HDLC message links
- Ringing signal conversion.
- Connector for the SX-200 ICP controller or Peripheral FIM Carrier.

Indicators

The Bay Control Card II has Alarm LED, Tx (transmit), and Rx (receive) indicators for the HDLC message link.

Electrical Description

There are two pairs of switches on the Bay Control Card II. All four switches must be set to closed for normal operation.

Bay Control Card III

Brief Description

The design of the Bay Control Card III (BCC III) is similar to the design of the PRI card. The BCC III has three sites for add-on modules. The sites have connectors to support the modules. The BCC III has site 1 for the FIM II (CIM or Maintenance module), site 2 for the T1/E1 module, and site 3 for the DSP module (single). The FIM II sits back from the BCC III faceplate. The CIM and Maintenance module sit close to the faceplate. The BCC III determines the presence of each module by accessing the module ID information. Unlike the PRI card, the BCC III does not have an S1 switch that you must set to identify the clock source; the BCC III sets the clock source automatically with the presence of the FIM II or CIM.

Note: With software loads earlier than LIGHTWARE™ 19 Release 3.2, BCC III cards will not start properly in an SX-200 ICP MX system if they are located between Bays 8-15. To resolve this issue, relocate the BCC III card to any bay between 1-7, upgrade the software load, and then relocate the BCC III card to any between 1-15 as it should now start properly. For more information, refer to Upgrading an SX-200 LIGHT to an SX-200 EL System.

The BCC III must be installed in an SX-200 ELx (Rev 2) cabinet. The EL cabinet (Rev 1) does not supply the proper card mounting and subsequent module mounting.

CAUTION: This card must not be inserted or removed with the power on.

Functions

The BCC III has the same functional characteristics as the BCC II, plus the extra processing power for PRI plus the extra functionality that the software and the hardware installed on the BCC III provides.

The BCC III has three sites for add-on modules. The modules are T1/E1, DSP (single), FIM II, CIM, and Maintenance. The T1/E1 module provides up to 48 channels of T1/D4 connectivity that the present T1 card provides. The DSP module provides CLASS generators for ONS/CLASS sets, conference bridges for Record a Call, and 16 DTMF receivers. The FIM II and the CIM provides a connection from the BCC III in a peripheral bay to the FIM or CIM in the main control cabinet.

BCC III Faceplate

LED Indicators

The BCC III faceplate has LEDs for itself plus LEDs for the T1/E1module, FIM II or the CIM (dependant on which module is installed on the BCC). The faceplate also provides access to the connectors for the Ethernet, T1/E1 module, the FIM II, the CIM, and the Maintenance module.

A group of four LEDs near the top of the card are the status LEDS for the T1/E1 module.These LEDs show the alarm status and the line status of the two T1 links. The LEDs on the left show the alarm status, the right shows the line status.The top two LEDs are for T1 link 2; the bottom two, for T1 link 1.

A group of three LEDs in the center of the faceplate are for the BCC III. From left to right are the Ethernet LED, Tx/Rx LED (transmit and receive indicators for the HDLC message link) and System Alarm LED. The Ethernet LED indicates heavy, moderate, or no Ethernet traffic. Ethernet connectivity is not supported on the BCC III. Ethernet is used in manufacturing (software design) and not used in the field.

A group of two LEDs show the status for the FIM II or the CIM. The LED on the right is for the Tx and shows the status of the local FIM II or CIM. The LED on the left is for the Rx and shows the status for the remote FIM/FIM II or CIM.

Jacks

The BCC III faceplate also has jacks for the T1/E1module and the Ethernet. The top two RJ45 jacks are for the T1/E1 module (Link 2 on top, Link 1 on the bottom) . Directly below the three central LEDs is the RJ45 jack for the Ethernet. The Ethernet jack is used only by software design.

Openings

Two openings exist below the FIM II /CIM status LEDS for connectors to pass through to the module that is installed on site 1 of the BCC III. A FIM II uses the top opening for the AFC Msg Link Tx; the bottom one for the AFC Msg Link Rx. The CIM uses the top opening for the Tx/Rx. The Maintenance module uses the top opening for its connection.

Fiber Interface Modules (FIM and FIM II)

There are two main types of Fiber Interface Modules: FIM and FIM II. The FIM and FIM II provide a fiber optic based communications link between the SX-200 ICP and a peripheral cabinet or NSU that houses an equivalent FIM.

Fiber optic cable connects the bays of an SX-200 system . At each end of a fiber optic cable is a Fiber Interface Module. At the transmitting end, the FIM or FIM II converts electrical signals into pulses of light to be transmitted over the cable. At the receiving end, the FIM or FIM II converts the pulses of light back into electrical signals usable by the node.

The main difference between the FIM and FIM II is where they are installed. The Fiber Interface Module (FIM) sits on Control FIM Carrier cards and Peripheral FIM Carrier II cards. The FIM II is a module that resides on a PRI card, on a Bay Control Card III (BCC III), or on a Peripheral Interface Module Carrier card. The FIM II on a PRI card is an alternative to the FIM on a control carrier card. The FIM II on the BCC III (in a peripheral cabinet only), is an alternative to the FIM on the Peripheral FIM Carrier II card. The FIM II on a Peripheral Interface Module Carrier card provides connectivity when the peripheral cabinet does not have a BCC III to hold a FIM II.

The FIM and FIM II have three variants of fiber length (1 km, 5 km, and 14 km). The variants of the FIM II have the same specifications as the variants of the FIM. Each FIM variant may be identified by its optical wavelength and fiber type.

The 1 km FIM has a wavelength of 820 nm and is multi-mode.

The 5 km FIM has a wavelength of 1300 nm and is multi-mode.

The 14 km FIM has a wavelength of 1300 nm and is single mode.

On the FIM you see the ID on the FIM faceplate. On the FIM II you see the ID on the Mitel label with the marketing part number. The same variant (optical wavelength and fiber type) must be used at each end of a fiber optic cable.

Note: A FIM can be connected to a FIM II as long as both have the same optical wavelength and fiber type.

Variant 9400-300-301-NA is the onboard FIM on a Control Dual FIM Carrier card and is standard hardware. The other FIM variants are available as upgrades to suit the length of each cable run between two nodes.

Specifications

FIM and FIM II Specifications

	FIM and FIM II Variants		
Approximate maximum fiber cable run length (See Note 1)	1km (0.62 miles)	5 km (1.86 miles) (See Note 2)	14km (8.7 miles) (See Note 2)
Power consumption (Watts)	2.5	2.5	2.5
Number of fiber links per FIM	1 Tx, 1 Rx	1 Tx, 1 Rx	1 Tx, 1 Rx
Fiber connector type	ST® (See Note 3)	ST (See Note 3)	ST (See Note 3)
Electrical interface (See Note 4)	8 serial ST links	8 serial ST links	8 serial ST links
Optical wavelength (nm)	820	1300	1300
Optical budget (See Note 5)	6 db	16 db	9 db
Date rate (Mbits/second)	16.384	16.384	16.384
Bit rate after encoding (Mbaud)	20.48	20.48	20.48
Fiber optic cable type	62.5/125 um Multi-mode	62.5/125 um Multi-mode	9/125 um Single mode

Notes:

1. The run length is the one-way length of fiber optic cable between nodes. The 1 and 5 km FIMs use multi-mode cable. The 14 km FIMs use single-mode cable.

2. Frame synchronization errors occur if the loop length of fiber optic cable - from control node to peripheral node and back - is between 10.0km and 10.6km (6.2 miles and 6.6 miles). Nodes are typically connected by two-strand or “paired” fiber optic cable. For paired cable, a loop length of 10.0km to 10.6 km equals a 5.0km to 5.3km cable run length. To learn how to recognize and correct frame synchronization errors, see the Troubleshooting section.

3. ST is a registered trademark of AT&T.

4. Some channels of the electrical interface are not available.

5. The optical budget is the allowable loss through fiber optic cable, splices, and connectors. The optical budget applies to the run length.

Functions

The FIM and FIM II consist of three functional sections: transmit, receive, and control.

Electrical Description

The transmitter section accepts data in serial format. Data is converted to byte interleaved format and a checksum byte and overhead information is added along with frame synchronization. Parallel data is converted to serial form and encoded. The optical interface converts the serial data to light and transmits it on the fiber.

The receiver extracts the bit clock from the incoming serial data, converts the data to parallel, and extracts frame sync. After the data is decoded, the status information and the data are combined, frame aligned and reformatted for output.

The control section generates control signals, the transmit clock (bit clock), and regenerates the telephony clocks for the peripheral cabinet (bays) and provides status information for the control cabinet.

Cabling

Two fiber optic cables are connected to each FIM. One cable is used for Tx (transmit) and the other for Rx (receive). The cables are 62.5/125 micrometers (um) Glass Multi-mode and use ST Connector systems. The fiber optic links allow a loss of 6.5 Db.

Indicators

Two green LEDs provide a visual indication of the state of the link. One LED is associated with the local FIM and the other with the remote FIM. Usage is as follows:

OFF	Power off or held in reset
FLASHING	Out of sync
ON	In frame sync

Peripheral Interface Module Carrier Card

Description

The Peripheral Interface Module Carrier Card (PIMCC) holds a FIM II or CIM to provide the fiber or copper interface between a peripheral cabinet and the SX-200 ICP systems. The face of the CIM or FIM II module has LEDs that show the link status.

The Peripheral Interface Module Carrier card plugs into slot 12 of an SX-200 rack-mount cabinet. This card replaces the Peripheral FIM Carrier II card; a peripheral bay with a BCC II.

The card has two J3 connectors, one for the CIM and one for the FIM II. The CIM uses P4 and the FIM II uses P3.

The card also has 4 standoffs to attach the CIM or FIM II to the carrier card. Initially, the standoffs are in the CIM position. Connecting a FIM II requires you to move the standoffs back from the faceplate.

Functions

The Peripheral Interface Module Carrier card is a carrier card for a FIM II or a CIM. The card provides fiber or copper connectivity between a peripheral cabinet and a main control cabinet when the peripheral cabinet has a BCC II instead of a BCC III. If the BCC III was in the peripheral cabinet, the interface modules (FIM II or CIM) on the BCC III would provide the connectivity needed to the main control cabinet.

Peripheral FIM Carrier II

Description

The Peripheral FIM Carrier II (PFC II) provides the interface between the backplane of an SX-200 RM peripheral cabinet and its Fiber Interface Module. It plugs into slot 12.

Functions

The Peripheral FIM Carrier II provides the following functions:

- Single ended to Balanced conversion of C244 and Frame Pulse
- Acts as the FIM carrier.

Electrical Description

The FIM provides single-ended signals which the PFC converts to the Balanced signals expected by the backplane. The PFC II also converts the three Balanced DX links from the backplane to the single-ended signals expected by the FIM.

The PFC II connects to the backplane and the FIM connects to the PFC via a 64 (2X32) pin connector.

Peripheral FIM Carrier

Description

The Peripheral FIM Carrier (PFC) provides the interface between the Bay Control Card and the Fiber Interface Module for the SX-200 LIGHT Peripheral cabinet. It connects to the module position on the Bay Control Card and acts as a carrier for the Fiber Interface Module. View this figure of the Peripheral FIM Carrier, and FIM.

Functions

The Peripheral FIM Carrier provides the following functions:

- Single ended to Balanced conversion of C244 and Frame Pulse
- Acts as the FIM carrier.

Electrical Description

The FIM provides single-ended signals which the PFC converts to the Balanced signals expected by the Bay Controller. The PFC also converts the three Balanced DX links from the Bay Controller to the single-ended signals expected by the FIM.

The PFC connects to the Bay Control Card via two 96 (3X32) pin connectors, and the FIM connects to the PFC via a 64 (2X32) pin connector.

Copper Interface Module (CIM)

Description

The CIM (Copper Interface Module) is a multi-layer printed circuit board with surface mount components on the connector side of the module. The module measures 149 mm by 74 mm.

The face of this module has LEDs that show the link status (same as the FIM) and an eight-pin modular jack that connects the copper cable.

The CIM

- Provides the same interconnect functionality as the FIM but with the use of copper instead of fiber (three ST links exist)
- Uses Category 5 UTP (unshielded twisted pair) cable that connects to an 8-pin modular jack at the faceplate of the card

- Suits co-located systems; not remote systems. The CIM comes in only one variant. The CIM supports a distance of up to 30 meters or 100 feet between cabinets.

Note: Systems without data cables require Category 5 cables for the CIM. Systems with data cables from the main control cabinet to the peripheral cabinet can use these data cables for the CIM if the cable is interconnected through a punchdown at each end, as long as the distance remains within the 100 feet limit.

The CIM sits on Site 1 of the BCC III in an SX-200 peripheral cabinet or on a Peripheral Interface Module Carrier card in a rack-mount cabinet.

The CIM has one J3 connector that connects to the carrier card. Positioning is secured by the use of standoffs. Four standoff holes on the module accommodate the four standoffs on the carrier card.

Functions

The CIM provides the following functionality:

- Communication link between the control cabinet and a peripheral cabinet.

DSP Modules

Description

DSP (Digital Signal Processor) modules are available in four configurations: Single, T1/E1 Combo, Dual, and Quad. The Single DSP services an SX-200 Peripheral Cabinet and sits on Site 3 of the Bay Control Card III (BCC III). The Dual and Quad reside in the MX controller. The T1/E1 Combo resides in the CX or AX controller.

Functions

The Single DSP module provides the following functionality:

- CLASS tones for the ONS/CLASS Line card. This DSP module has 8 CLASS generator resources that are assigned and released dynamically as they are required. The DSP module on the BCC III must reside in the same bay as the ONS/CLASS Line card.
- Sixteen conference bridges for the Record a Call feature. The DSP module provides these bridges to circuits in the same bay. A Record a Call conference is between one internal party, one external party (a trunk call) and the voice mailbox. A Record a Call conference can also be setup between two internal parties and the voice mailbox.
- Sixteen DTMF receivers that can be used system wide. The DSP module replaces the Universal Card with respect to DTMF receivers.

The Dual and Quad DSP provides the functionality of the Single DSP but in greater quantities plus voice compression resources. For more information, see DSP Configuration Options.

T1/E1 Module

Description

The T1/E1 module contains two digital trunk connections that can have a T1 style interface or an E1 style interface configuration. The SX-200 ICP uses the T1 style interface configuration only. The T1/E1 module sits on site 2 of the Bay Control Card III (BCC III) and the PRI card. The T1/D4 functionality exists only when the T1/E1 module sits on a BCC III. The PRI functionality exists only when the T1/E1 module sits on a PRI card.

The SX-200 ELx cabinet supports the T1/ E1 module.

The T1/E1 module has two LT/NT connectors (jumpers), one for each T1/E1 link. These connectors are normally set to the NT position (the default setting). The LT setting is used if two systems are connected back-to-back as part of a local network.

The two links from the T1/E1 module on the BCC III occupy software slots 5 and 6 in the programming.

The T1/E1 module can be programmed for an extended superframe (ESF). The module also eliminates the need for an external CSU.

The T1/E1 module has four LEDs, two for each T1 link. Link 1 is on the bottom; link 2 is on the top.The LEDs on the right show the line status; those on the left show the alarm. The faceplate of the PRI card and the faceplate of the BCC III show the LEDs for the T1/E1 module.

Functions

The T1/E1 module provides the same functionality as the T1 card plus the following enhancements:

- Two circuits/links (24 channels per circuit) with T1/D4 functionality
- ESF/CSU functionality (enabled in Form 42).

Universal Card

Description

The Universal Card, part number 9109-005-000, interfaces up to four modules to the backplane.

Facilities

Facilities provided by the Universal Card include:

- module mounting positions (four)
- module activity LEDs (four)
- software-controlled failure alarm LED.

Physical Description

Each module has two vertical 32-pin female DIN connectors which mate to male connectors on the Universal Card. Mechanical connection to the Universal Card is assisted by a standoff.

Electrical Description

The combination of modules on a Universal Card is limited by the power available from the card. Each module has a power rating number. The total of these numbers must not exceed 10. The Universal Card is a high-power card.

Power Ratings for Universal Card Modules

Module Type	Power Rating
Empty module position	0
Music-on-Hold/Paging	1
DTMF Receiver/Relay	2

Power Ratings for Universal Card Modules	
E&M Trunk	3

Each module position is assigned card tip and ring connections as shown below:

Module Tip/Ring Assignments

Module Position	Module Tip/Ring		
1	T1/R1	T2/R2	T3/R3
2	T4/R4	T5/R5	T6/R6
3	T7/R7	T8/R8	T9/R9
4	T10/R10	T11/R11	T12/R12

DTMF Receiver/Relay Module

Brief Description

The DTMF Receiver/Relay Module facilitates the reception and decoding of DTMF dialing.

Main Components

Major components of the DTMF Receiver/Relay Module are:

- Mitel filter/codec (four)
- Mitel 8870 DTMF receiver (four)
- Parallel bus interface
- Two general purpose relays.

Facilities

Facilities provided by the DTMF Receiver/Relay Module include:

- Early line split
- Guard time circuit
- Presentation of digits on parallel bus with Data Valid signal.

Circuit Description

There are four receivers on the DTMF module. Each receiver takes its input from the incoming serial PCM audio stream and repeats this data to the outgoing serial PCM stream approximately 125 msec later. A filter/codec converts the data to analog audio which is monitored by a DTMF Receiver chip. When DTMF tones are detected, the loopback of the data to the PCM output stream is disabled (Early Line Split). The DTMF Receiver/Relay Module has a power rating of 2. Two relays are on the module; each is software controlled to provide a contact closure across one tip-ring pair.

Relays

There are two general purpose relays. When each relay closes, it connects a tip and ring pair together.

The relay contacts are rated as follows:

maximum switching voltage: 90 V
maximum carrying current: 0.5 A

Note: This relay contact may be connected only to a secondary circuit that has no direct connection to a primary circuit, and receives its power from a transformer, converter, or equivalent isolation device situated within the equipment.

E&M Trunk Module

Brief Description

The E&M Trunk Module (part number 9109-013-000) plugs into the Universal Card. It provides interface to Type 1 or Type 5 E&M trunks. The module has a power rating of 3.

Facilities

Facilities provided by the E&M Trunk Module include:

- Type 1 and Type 5 signaling
- selectable hardware gain/loss via DIL switches on the module
- selectable 600 ohm or AT&T Complex Balance Network (350 ohms + 1000 ohms in parallel with 0.21 microF)
- selectable 2- or 4-wire transmission
- on board filter/codec for analog/digital and digital/analog conversions (μlaw).

Operation

The E&M Trunk Module is set for the type of trunk in use by a set of eight DIL switches. The settings are as follows:

Function		**Switches**							
		1	**2**	**3**	**4**	**5**	**6**	**7**	**8**
PBX to Line Hardware Gain	3 dB	0	x	x	x	x	x	x	x
	-13 dB	1	x	x	x	x	x	x	x
Line to PBX Hardware Gain	- 4 dB	x	0	x	x	x	x	x	x
	-11 dB	x	1	x	x	x	x	x	x
Balance	600 ohm	x	x	1	0	x	x	x	x
	Complex	x	x	0	1	x	x	x	x
Transmission	2-wire	x	x	x	x	1	x	x	x
	4-wire	x	x	x	x	0	x	x	x
Transmission	2-wire	x	x	x	x	x	1	x	x
	4-wire	x	1	x	x	x	0	x	x

0= open, 1= closed, x= not applicable
Default setting for North America is 00101100

The E&M Trunk Module applies signals to the M lead and monitors the E lead. In the on-hook condition, the Type 1 interface grounds the M lead; an open presented to the E lead indicates idle, a grounded E lead indicates an incoming call. In the off-hook condition, the Type 1 interface applies -48 volts to the M lead; a ground sent to the E lead indicates an incoming seizure.

Music-on-Hold/Paging Module

Brief Description

The Music-on-Hold/Paging Module (part number 9109-018-000) provides an input for music-on-hold, a paging output, and a relay to switch an external paging amplifier. The module plugs into the Universal Card. The Music-on-Hold/Paging Module has a power rating of 1.

Major Components

Major components of the Music-on-Hold/Paging Module include:

- audio filter/amplitude limiter
- Mitel 8961 filter/codec
- paging driver amplifier
- paging control relay.

Electrical Description

The music input is isolated by a transformer and has an impedance of 600 ohms. The signal should be between 50 and 500 mVrms. High frequencies are attenuated and amplitude limiting is applied as required by FCC rules part 68. Amplitude limiting is applied when the signal exceeds approximately 390 mVrms.

The paging output is isolated by a transformer and has an impedance of less than 200 ohms. The output level into a 600 ohm load is typically -6 dBm (388 mVrms).

The control relay contacts are rated as follows:

maximum switching voltage	90 Vrms
maximum carrying current	0.4 Arms

Note: This relay contact may be connected only to a secondary circuit that has no direct connection to a primary circuit, and receives its power from a transformer, converter, or equivalent isolation device situated within the equipment.

DID Trunk Card

Brief Description

The DID Trunk Card (part number 9109-031-000) contains six 1-way direct inward dial circuits which provide for direct access to PBX subscriber lines from the public telephone network.

DID Trunk Cards are high-power cards. The maximum number of these cards is four per bay, providing a maximum of 24 ports.

Major Components

Major components in the DID Trunk Card are:

- Mitel 8962 Filter/Codec (one per trunk circuit)
- Feed Reversal relay (one per trunk circuit)

- Alarm LED.

Facilities

The facilities provided by each trunk circuit are:

- Trunk activity LED
- Line protection
- 2-wire / 4-wire conversion (external to internal)
- Analog-to-Digital / Digital-to-Analog conversion (µ–law)
- Immediate, Delay Dial or Wink Start supervision
- Direct Inward Dialing access to PBX subscriber lines
- Conformity with the EIA loss level plan for µ–law compatible PABXs in North America.

Operation

A trunk is idle if the resistance across Tip and Ring is 4000 ohms or more. In idle condition the PBX provides forward battery feed to the line. The Tip is grounded and the Ring is at -48 Volts.

The CO initiates a call to the PBX by terminating Tip and Ring. The supervision circuitry detects the flow of loop current and alerts the system software. The PBX signals that it is ready to receive dialing by briefly applying a battery reversal to the line. Ring is grounded and Tip is at -48 Volts. There are two types of controlled address signaling: Delay Dial and Wink Start.

A Delay Dial signal must start no later than 150 ms after trunk seizure. It is held until the PBX is ready to receive dialing. Minimum hold time is 140 ms.

A Wink Start signal must start at least 100 ms after trunk seizure. It is sent when the PBX is ready to receive dialing and can be held a maximum of 290 ms.

Where the CO does not provide controlled address signaling, the PBX must be prepared to receive dialing 65 ms after trunk seizure.

When the called station or PBX attendant answers, the PBX places battery reversal on the line for the duration of the call. The trunk then returns to the idle state.

A trunk may be busied out by the system software, which then presents an open circuit to the Tip and Ring of both the trunk and trunk card circuit. The trunks default to the busy-out state if system power fails.

Each circuit has a LED on the front panel which lights to indicate the trunk is in use. A seventh LED at the bottom of the panel lights to indicate a failure on the card.

Electrical Description

Line protection comprises high voltage varistors to energy dump ground from Tip and Ring and fusible links incorporated in the battery feed resistors. EMI is controlled by inductors in series with Tip and Ring.

The maximum loop resistance is 1800 ohms. The maximum loop length is 5850 m (19,200 ft) when using 26 AWG wire, 15,240 m (50,000 ft) when using 22 AWG wire.

The card circuitry performs 2-wire to 4-wire conversion, splitting the signal on the trunk into outgoing and incoming speech paths. The analog signal coming from the trunk is converted to Pulse Code Modulation (PCM); the signal to be sent to the trunk is converted from PCM to analog audio. These conversions are performed by a Mitel Codec chip.

Battery feed reversal and busy-out for each trunk are controlled by relays, as shown below.

Condition	Relay 1	Relay 2
Forward Feed (Idle)	ON	OFF
Reverse Feed (Talk)	ON	ON

LS/CLASS Trunk Card

Brief Description

The LS/CLASS Trunk card interfaces eight analog trunk circuits to the system. LS is the acronym for Loop Start and CLASS is the acronym for Custom Local Area Signaling Services (allows the system to receive calling Line ID digits and CLASS name on incoming CLASS trunks).

The LS/CLASS Trunk card is a low power, peripheral interface card. It can be installed into slots one to eight in a SX-200 rack mount cabinet.

Programming the card involves defining the trunk card as an "8 cct CLASS" in CDE Form 01 and programming the "8-CIRCUIT CLASS" trunk type in CDE Form 13 with the same descriptors as a "4-CIRCUIT CLASS" trunk type.

Facilities

Facilities provided by the LS/CLASS Trunk card include:

- Loop Start operation
- Forward/reverse current detectors (polarity reversal, answer supervision)
- Alarms and trunk activity indicated by an LED (a single LED for any circuit in use)
- CLASS signal reception
- Transient suppression on Tip and Ring leads.

Electrical Description

The LS/CLASS Trunk card is a low power, peripheral interface card. It can be installed into slots one to eight in a SX-200 rack mount cabinet.

Each trunk circuit has two leads, tip and ring, for connection. The tip and ring leads are protected by a PTC in series on each lead and transient voltage suppressor across tip and ring, tip to ground, and ring to ground.

Each trunk has an LED on the front faceplate of the card that lights to indicate that the circuit is busy. An LED at the bottom of the faceplate flashes to indicate a failure on the card.

Operation - Loop Start

To place an outgoing call, the trunk card places a termination across tip and ring. The CO detects the current flow and responds with dial tone. Now the user may begin to dial.

The trunk card recognizes an incoming call when it receives ringing voltage or battery reversal from the CO. The trunk card will respond by placing a termination across Tip and Ring. The trunk is released when the loop current is broken, either when the near party goes on-hook or the line is physically broken.

Custom Local Area Signaling Service (CLASS)

The CLASS modem tones are received by the trunk while in the on hook state (in the silent period of the ringing cadence between the first and second ringing burst). This tone is decoded by the caller ID decoder circuit and the message is stored in the on-board memory.

LS/GS Trunk Card

Brief Description

The Loop Start/Ground Start Trunk Card, part number 9109-011-001, interfaces six trunk circuits to the system.

Facilities

Facilities provided by the LS/GS Trunk Card include:

- Loop Start or Ground Start selectable by jumper
- M and MM signaling leads available (refer to Meter Pulse Collection)
- Trunk activity indicated by LED (one per trunk)
- Transient suppression on Tip, Ring, and signaling leads
- Alarm LED.

Electrical Description

The Loop Start/Ground Start Trunk Card mounts in any slot and interfaces six trunk circuits to the system. Each trunk circuit is programmed as loop start or ground start by a jumper clip prior to installation.

Each trunk has Tip and Ring leads and M and MM leads for additional signaling, if required. All leads are protected by varistors against transients between line and ground. There are also varistors between Tip and Ring and between M and MM. Each lead is in series with an inductor near the edge connector to reduce electromagnetic interference (EMI).

Each trunk has an LED on the front faceplate of the card that lights to indicate that the circuit is busy. An LED at the bottom of the faceplate lights to indicate a failure on the card.

Operation - Loop Start

To place an outgoing call, the trunk card places a termination across tip and ring. The CO detects the current flow and responds with dial tone. Now the user may begin to dial.

The Trunk Card recognizes an incoming call when it receives ringing voltage or battery reversal from the CO. The Trunk Card will respond by placing a termination across Tip and Ring. The trunk is released when the loop current is broken, either when the near party goes on-hook or the line is physically broken.

Operation - Ground Start

To place an outgoing call, the Trunk Card grounds the Ring lead. The CO responds by grounding the Tip lead. The Trunk then places a termination across Tip and Ring and ungrounds the Ring lead. The CO then sends dial tone to indicate that it is ready to receive dialing.

The Trunk Card recognizes an incoming call when the CO grounds the Tip lead. The CO may also send ringing voltage. The Trunk Card will respond by placing a termination across Tip and Ring. The trunk is released when the loop current is broken, either when one party goes on-hook or the line is physically broken.

Loop Start/Ground Start Card and Jumper Settings

ONS/CLASS Line Card

WARNING: Any connection of this card to an off premise application, an out of plant application, or to any other exposed plant application may result in a safety hazard, and/or defective operation, and/or equipment damage.

Brief Description

ONS is the acronym for On-Premises (interfaces standard subscriber telephones to a PBX in the same building). CLASS is the acronym for Custom Local Area Signaling Services (allows the system to receive calling Line ID digits and CLASS name on incoming CLASS trunks). The CLASS functionality is enabled by a purchasable FOR option.

The ONS/CLASS Line card

- Contains 12 line circuits. The card is a low power card and plugs into any peripheral card slot.
- Has the positive disconnect functionality. The card detects if the far end has hung up for the programmed period of time (0-5 seconds) and forces the device (fax machines or voice mail systems) to hang up. This functionality is dependant on COS Option 506, ONS Positive Disconnect.
- Is hot-swappable.
- Does not offer network side CLASS functionality. Network side CLASS functionality is provided by a LS/CLASS Trunk card in a SX-200 rackmount cabinet, or the equivalent information (Calling Line ID) from a PRI link.

Facilities

Each line circuit provides the following facilities:

- Line protection
- Positive disconnect
- Analog-to-Digital / Digital-to-Analog conversion (μ–law)
- Line circuit status monitoring
- Signaling (ringing, message waiting)
- Calling party ID data to the set while the set is in an off-hook or in an on-hook state.

Electrical Description

The following description applies to each line circuit.

A bridge rectifier provides four protection diodes for the line circuit transistors. The -28 volt line is protected by a 35 volt transorb.

The line circuit performs 2-wire to 4-wire conversion, splitting the signal on the line into outgoing and incoming speech paths. The analog signal coming from the telephone is converted to pulse code modulation (PCM); the signal to be sent to the telephone is converted from PCM to analog audio. These conversions are performed by a Mitel Codec chip.

When the telephone is off-hook, the line circuit status LED on the front panel lights. The line circuit maintains a constant 26 mA current to the telephone while the set is off-hook. Loop length is maximum 600 ohms including the telephone.

The ONS/CLASS Line Card supports the Message Waiting feature. A high voltage (-140 Vdc) is applied to the Ring terminal of the line to light a neon lamp on the subscriber's set.

Operation

When a telephone goes off-hook, the line circuit detects the flow of loop current and signals the main processor. The processor responds by connecting a DTMF receiver to the line and sending dial tone to the set. (If the telephone uses pulse dialing, the processor detects the pulses by monitoring the loop current). The user can then dial the desired number.

When a call is directed to an extension, the system applies ringing voltage to the appropriate line and monitors the loop current for an off-hook condition. When the telephone is answered, the ringing voltage is removed.

When one of the sets goes on-hook to end a call, it disconnects the call and returns the line to its idle state.

ONS/LS Combo Card

Brief Description

The ONS/LS Combo card supports 12 ONS circuits and 4 LS circuits.

Facilities and Operation

Each circuit type on the Combo card provides the facilities and operation listed under the respective single cards.

ONS/Class Cards

LS/Class Cards

Electrical Description

Circuit characteristics of the ONS/LS Combo card are similar to those of AMB/AOB.

OPS Line Card

Brief Description

The OPS Line Card, part number 9109-040-000, contains six off-premises line circuits. An Off-Premises (OPS) line circuit is used where the line goes outside the building that houses the PBX and the line may be exposed to extraneous high voltages or induced currents (lightning).

Major Components

Major components for the OPS Line Card are:

- Mitel 8962 Filter/Codec (six)
- 2-wire / 4-wire converter (six)
- Ringing relay (one per circuit)
- Alarm LED.

Facilities

Each line circuit provides the following facilities:

- Line activity LED
- Line protection

- Analog-to-Digital / Digital-to-Analog conversion (µ-law)
- Signaling (ringing).

Operation

Each circuit has a LED on the front panel which lights to indicate that the line is in use. A seventh LED at the bottom of the panel lights to indicate a failure on the card.

The line circuit applies forward battery feed to the line. The Tip is grounded and the Ring is at -48 volts. When the set goes off-hook to place a call, the system detects the loop current and responds with dial tone. Dialing may be DTMF or pulses. Dial pulses are debounced by software to assure reliable performance.

When a call is directed to the set, a relay closes and sends ringing voltage to the set. The ringing relay drops out when loop current flow indicates that the telephone has been answered (off-hook condition).

Electrical Description

Line protection comprises high voltage varistors to energy dump ground from Tip and Ring plus fusible links incorporated into the battery feed resistors. EMI is controlled by inductors in series with Tip and Ring.

The maximum loop resistance is 1800 ohms. The maximum loop length is 5850 m (19,200 ft) when using 26 AWG wire, 15,240 m (50,000 ft) when using 22 AWG wire.

The card circuitry performs 2-wire to 4-wire conversion that splits the signal on the line into outgoing and incoming speech paths. The analog signal coming from the line is converted to pulse code modulation (PCM); the signal to be sent to the line is converted from PCM to analog audio. These conversions are performed by a Mitel Codec chip.

The line circuit applies ringing voltage to the appropriate line through a relay and removes it when the telephone is answered. Answer is detected by monitoring the loop current.

Loop current is provided through a pair of 200 ohm resistors. Below 900 ohms loop resistance, active current limiting circuitry limits line power to less than 1.5 watts.

Ringing for each line is controlled by relay 2.

Condition	Relay 1	Relay 2
Idle or Talk	OFF	OFF
Ringing	OFF	ON

Digital Line Card

WARNING: Any connection of this card to an off premise application, an out of plant application, or to any other exposed plant application may result in a safety hazard, and/or defective operation, and/or equipment damage.

Brief Description

The Digital Line Card (DLC), part number 9109-012-00X, provides an interface from the system to the following:

- SUPERSET 4001 telephone

- SUPERSET 4015 telephone
- SUPERSET 4025 telephone
- SUPERSET 4125 telephone
- SUPERSET 4150 telephone
- SUPERSET 401+ telephone
- SUPERSET 410 telephone
- SUPERSET 420 telephone
- SUPERSET 430 telephone
- SUPERSET 3DN telephone
- SUPERSET 4DN telephone
- DSS/BLF Interface Unit
- DATASET
- DNIC Music-on-Hold/Paging Unit (DMP)
- SUPERCONSOLE 1000 Attendant Console

The DLC contains 12 asynchronous line circuits (Mitel Digital Network Interface Circuit), and is a low-power digital card which can plug into any slot.

CAUTION: In high traffic configurations, keep the Digital Line Card (DLC) count to a maximum of 7 per bay when using BCC II cards. If you have 8 DLCs per bay, keep the device count to 84 or less. This will avoid slow softkey response during peak traffic periods.

There are 13 LEDs on the face of the card. The top 12, one for each line circuit, light when the circuit is in use. The bottom LED on the panel lights to indicate an alarm condition within the card.

Digital Line Cards with a blue faceplate, part number 9109-012-002-NA, can be installed up to the quantity purchased as an option (programmed in CDE Form 4, System Options). Additional Digital Line Cards must be part number 9109-012-000-SA.

Facilities

Each Digital Network Interface Circuit (DNIC) provides the following facilities:

- Line protection
- Full duplex simultaneous data and voice digital transmission over a single pair of wire
- Line circuit status monitoring
- Signaling and HDLC protocol to its associated DATASET.

Electrical Description

Each Digital Network Interface Circuit (DNIC) connects a device (click here for a list) to the common circuitry on the DLC card (and then to the system). The common circuitry will be described first, followed by a description of one DNIC.

The common circuitry contains a High-level Data Link Controller (HDLC) which controls the D channel communication between each DNIC and the main controller within the system. This data is passed over one-half of a link to the main controller. The B1 and B2 channels from the DNICs are multiplexed onto one link between the DLC and the main controller. The common circuitry includes phase-lock loop circuitry to keep the DNICs in synchronization with the system clock, as well as circuits which prevent the DLC from disrupting the backplane when a card is inserted or removed. The DLC line circuits are arranged in three groups of four; at power-up, each group can be separately sequenced. There are 12 Digital Network Interface Circuits (DNIC) on the DLC card; each is connected to a separate tip-ring pair.

DNIC Description

Each DNIC connects via its tip-ring pair to a proprietary telephone, a DNIC Music-on-Hold/Paging Unit, or a DATASET which also contains a DNIC. The DNICs communicate with each other over the twisted pair at 160 Kb/s (two 64 Kb/s B1 and B2 channels and a D channel). Since the DNIC is a proprietary integrated circuit, each device connected to a Digital Line Card DNIC tip-ring pair must also contain a DNIC. The two DNICs communicate data plus voice simultaneously in full duplex over the single twisted pair between them. The twisted pair also carries the power required by the SUPERCONSOLE 1000, SUPERSET 4001, SUPERSET 4015, SUPERSET 4025, SUPERSET 4125, SUPERSET 4150, SUPERSET 401+, SUPERSET 410, SUPERSET 420, SUPERSET 430, SUPERSET 3DN, SUPERSET 4DN telephone or DSS/BLF Interface Unit, from the DNIC on the DLC. The DATASET is powered from a separate AC power supply.

Operation

The Digital Line Card communicates with a DNIC-equipped device using digital transmission techniques: a voice channel, a data channel, and a control channel. It allows simultaneous transmission of voice and data over a single twisted pair of wires. When the DLC is connected to SUPERSET 410, SUPERSET 420, or SUPERSET 430 telephones. The telephone's voice operation and the data device's data operation can both function concurrently.

Loop Length Specifications for Connections to a Digital Line Card

Rules for loop lengths between the Digital Line Card and its peripheral devices must be followed for proper operation of the device. See Loops Lengths for these rules.

T1 Trunk Card

Brief Description

The T1 Trunk Card interfaces a single T1 trunk circuit to the PBX. The T1 protocol is used primarily in North America.

Facilities

The T1 Trunk Card provides the following facilities:

- T1 Clock - System Clock Phase Comparator
- Bidirectional T1 to ST-BUS data rate and format conversion
- Line Equalization.

General Description

The T1 interface will transmit and receive 24 8-bit voice/data channels on a 4-wire digital trunk that operates at 1.544 Mb/s. The BCC performs all control functions.

To provide signaling information on the T1 line, data bits are "stolen" from each channel to provide channel associated signaling.

The T1 Trunk Card includes a phase comparator which, through the T1 Clock in the SX-200 ICP, keeps the system clock in phase with the incoming frame rate clock. The comparator prevents data losses caused by clock rate differences. Phase error is the difference between the clock rate received on the link and the clock rate generated within the system (if the T1 clock is being adjusted to the incoming clock rate). If the difference climbs by greater than 1 in a single reading then the next three readings are filtered out. If the fourth reading has climbed greater than 1 then the link is considered unstable. A maintenance log is generated and the link is no longer used as a network synchronization source.

The system supports a maximum of eight T1 links with a maximum of two T1 links per bay.

One incoming T1 trunk is selected as the primary timing source; the system locks its PCM clock and all other T1 trunk cards in the system to this incoming data stream.

The Channel Service Unit (CSU) is required for on-site T1 termination. The T1 Trunk Card includes switches to set appropriate line equalization for cable lengths up to 200 meters (655 feet) from the CSU.

An adapter fitted to the backplane connector provides a 15-pin D-Sub connector for the T1 facility and a 25-pair connector to maintain access to the adjacent odd-numbered card slot. One or two T1 Trunk Cards can be installed and require a single or dual adapter respectively.

Electrical Description

In the transmit direction, the data from the system PCM link must be converted from the Mitel ST-BUS format of 32 channels at 2.048 MBit/sec to the T1 format of 24 channels at 1.544 MBit/sec. To match the number of data channels, the T1 card skips every fourth channel on the ST-BUS links. The T1 interface circuits retime the output data to 1.544 Mbits/sec and add the framing bits.

In the receive direction, the framing bits are removed, the data is retimed to 2.048 MBit/sec and the channels are mapped onto the system PCM link in the same manner as above, with every fourth channel left empty.

Indicators

If any of the data channels on the T1 link are active, the upper front panel LED lights, giving visual indication of channel activity.

If the system is receiving a yellow alarm condition, the yellow (NO SYNC) LED lights. If transmitting a yellow alarm condition, and not receiving a yellow alarm condition, then the LED is off.

The lower red LED is the card alarm. If the system generates or receives a yellow alarm condition on the T1 link, the ALARM LED flashes.

Physical Description

The switches used (SW1) to set up proper equalization on the T1 Trunk card must be set as follows. Lengths are for cable length, not loop length.

Short Cable (under 150 feet)	S1 only - Closed
Medium Length Cable (150 to 450 feet	S2, S3, S4 only - Closed
Long Length Cable (450 to 655 feet)	S5, S6, S7 only - Closed

PRI Card

Brief Description

The PRI card provides an SX-200 ICP with Primary Rate Access (PRA) to the ISDN service provider. By interfacing the SX-200 ICP with the ISDN service provider, it allows users to access ISDN services, such as Direct Dial In (DDI), Calling Line Identification (CLID), and Call By Call Service Selection (CBC).

The PRI card, preloaded with software, comes with a T1/E1 module that supports up to two T1 links of ISDN connectivity. You use the PRI card software that accompanies the PRI card for a backup or for a upgrade.

The PRI card can originate and terminate up to 2760 calls per hour.The PRI card has the Bearer Capabilities of speech (voice) and 3.1 kHz audio.

The PRI card is "hot-swappable".

Major Components

The major components of the PRI card are

- A 33 MHz Motorola 68360 processor with an integral DRAM controller and serial communication controllers (SCCs)
- 8MB of FLASH memory for programming and database storage
- 16MB of 3.3v EDO DRAM for operating memory
- An SCC (Serial Communication Controller) for an RS232 port for maintenance (TXD and RXD only) accessible through the backplane
- A second SCC for an Ethernet port (for Mitel use only). Manufacturing loads the software on the PRI card
- Two other SCCs for HDLC controllers on two ST links
- Three sites for removable circuit modules (site #1 for the FIM II or CIM, site #2 for the T1/E1 module, Site #3 for another release)
- A T1/E1 module that has built-in CSU functionality
- A dip switch (S1 Switch) that controls the source of the clock and link signals (backplane or interface module).

General Description

The PRI card comes with a T1/E1 module (the SX-200 ICP only uses the T1 functionality). The T1/E1 module provides up to 2 links. A FIM II or a CIM (Copper Interface Module) may also reside on the PRI card. The FIM II or the CIM on the PRI card in a peripheral cabinet connects the PRI card bay to the SX-200 ICP. The FIM II or the CIM on the PRI card in a peripheral cabinet connects the PRI card bay to the SX-200 ICP controller.

A peripheral cabinet can support up to two PRI cards installed in slots 10 and/or 11.

The S1 switch on the PRI card (see illustration) controls where the clock source and link signals come from and must be set with switch 1 open and switch 2 closed.

Two PRI (T1/E1) cables use the two RJ45 jacks on the front of the PRI card to loop back to the rear of the cabinet and come out through the fiber/cable port. The top RJ-45 jack is for PRI link 2; the bottom, PRI link 1. The PRI cables then connect to the ISDN Network.

Note: The RJ45 jack for the Ethernet is used only by manufacturing.

The PRI card is also a carrier card for the FIM II or CIM. To enable the interface between a peripheral cabinet and the SX-200 ICP controller, route a cable between the PRI card bay and the back of the peripheral cabinet, then run the cable from the back of the peripheral cabinet to the FIM II or CIM in the SX-200 ICP controller.

Two RS-232 serial ports J10 and J11 connect the PRI card to IMAT on the PC for the operation, maintenance and administration for the PRI card. J10 serves the PRI card in slot 10. J11 serves the PRI card in slot 11.

Faceplate LEDS

The LEDs on the PRI card can be seen through the glass door on the front of the cabinet.

- A group of four LEDs near the top of the card are the status LEDS for the T1/E1 module.These LEDs show the alarm status and the line status of the two PRI links. The LEDs on the left show the alarm status, the right shows the line status.The top two LEDs are for PRI link 2; the bottom two, for PRI link 1.

- A group of three LEDs in the middle of the card, from left to right, show the Ethernet activity, the heart-beat of the PRI card, and the system alarm.
- A group of two LEDs show the status for the FIM II or the CIM. The LED on the right is for the Tx and shows the status of the local FIM II or CIM. The LED on the left is for the Rx and shows the status for the remote FIM/FIM II or CIM.

LED	State	Meaning	Troubleshooting
Alarm status of the PRI link	Red	Red alarm	Check PRI connection.
	Yellow	Yellow alarm	Check the PRI link with the Protocol Analyzer. Make sure the analyzer is running the correct ISDN variant.
		Yellow +Green D-channel LED= Blue alarm	Contact the Central Office that is causing the problem and tell them to turn the blue alarm off.
	Off	Layer 1 is established.	No error.
Line Status of the PRI link	Off	Layer 1 has not been established.	Wait a few minutes.
	Flashing green	Layer 1 is established, D-channel is being established.	Wait a few minutes
	Solid green	D-channel has been established.	No error.
System Alarm	Solid red	System is in self-test, the message link is down.	Normal condition on boot up. Wait 30 seconds. LED should change to the next state (below). Otherwise, check the fiber connection and/or the system programming.
	Flashing red	System is booting, the message link is down.	Normal condition after the Power-on self test is completed. Wait 20 seconds, LED should change to the next state (below). Otherwise, check fiber connection and /or system programming.
	Off	The Message link is up.	The validity of this LED is conditional upon the Heartbeat LED functioning.
Heartbeat	Flashing Green	The application is running.	No errors to troubleshoot.
	No flashing.	The application is not running.	Reset the card. If this continues, the card needs servicing.
Ethernet Activity	Solid green	Heavy Ethernet traffic, which may cause performance problems.	Ethernet connectivity is not supported on the PRI card. Ethernet is not used in the field.
	Flashing green	Moderate Ethernet traffic	
	Off	No Ethernet traffic	Normal condition presently.
FIM II/CIM status	Solid green	FIM or CIM synchronization is	Normal condition

LED	State	Meaning	Troubleshooting
		functional	
	Flashing green	FIM or CIM synchronization is not functional	Abnormal condition, check the connection or change the FIM II or CIM

Power Consumption

Restricting the number of PRI cards to one in the main control cabinet maintains the allowable power consumption for the cabinet.

Because a peripheral cabinet does not have a main control card, two PRI cards may be installed in the peripheral cabinet of a SX-200 ICP.

Modules

The PRI card has add-on modules that are designated to specific sites on the card:

- The FIM II sits on site 1 of the PRI card in a main control cabinet or in a peripheral cabinet.The FIM II in each cabinet provides a communication link between the main control cabinet and the peripheral cabinet. Three types of FIM II modules are available (1, 5, and 14 km). The FIM II modules at each end of the cable must be of the same type.
- The CIM sits on site 1 of the PRI card in a main control cabinet or in a peripheral cabinet.The CIM does not sit back from the faceplate as the FIM II does. The CIM provides the same functionality as the FIM II. There is one type of CIM.
- The T1/E1 module sits on site 2 of the PRI card. The T1/E1 module provides up to two links of ISDN connectivity. The LT/NT connectors on the module have a default setting for the user side (NT).

ISDN Protocols Supported

The PRI card supports the following ISDN Protocols:

- DMS-100
- DMS-250
- 4ESS
- NI-2 variants 5ESS, Bellcore, and GTD5

ISDN Services Supported

The PRI card supports the following ISDN network services:

- **Calling Party Number (CPN) -** This number substitutes the calling station number on outgoing calls for purposes of network identification and call back.
- **Calling Name ID (CNID)** - This ID is the incoming call name delivery per NA DMS100 custom specification or National ISDN-3 (NI3).
- **Calling Line Identification Presentation (CLIP**) - The Calling Party Number can be provided to the ISDN Network for outgoing calls or provided to the PRI card from the ISDN Network for incoming calls. This information is passed onto the system and can be used for database applications such as screen pops and for inclusion in SMDR records.
- **Calling Line Identification Restriction (CLIR) -** This feature allows users to prevent their telephone number from being presented to the called party.
- **Partial PRI Links -** The SX-200 PRI card will support COs that provide this feature.

- **Direct Dial-In (DDI) -** DDI is an ISDN option that allows direct access to a line behind a PBX through a unique directory number. This allows the dialed digits of an incoming ISDN call to be presented to the PBX. All ISDN trunks are treated as Dial-In trunks; the CO always sends digits to the PBX.
- **Call By Call Service Selection (CBC) -** This feature allows telephone users to select the ISDN network services that they wish to use on a per call basis.
- **DID Calling Party Number Forwarding -** Outgoing CPN delivers the calling party's DID number to the Network when the call has been identified as a call from a device with an associated DID number instead of delivering the main directory number associated with the PBX.
- **Equal Access to Interexchange Carriers -** The PBX provides a carrier access code which identifies to the Central Office which Interexchange Carrier is to receive the call. The PBX outpulses a digit string which includes a carrier access code, followed by an identification number, followed by the called number.
- **Min/Max** - This feature allows a customer to control incoming and outgoing call traffic. Minimums are assigned to ensure that a particular type of call (such as INWATS) always has a set number of lines available. Maximums are assigned to limit certain types of calls, i.e., OUTWATS. This ensures that resources are not used up by a single type of call. Different Min/Max databases can be created for different times of the day or for special occasions such as telethons or infomercials.
- **Auto Min/Max** - This feature provides user programmable time-of-day automatic control of Min/Max parameters.
- **NFAS** (Non-Facility Associated signaling) - NFAS allows you to use a single D-channel to handle the signaling requirements for a group of PRI links that all use the same Protocol. This feature eliminates the need to purchase a D-channel for each link. NFAS is mainly for North America.
- **D-Channel Backup -** The feature is used for signaling to establish and maintain the circuit, and to send user data. D-channel Backup provides an alternate D-channel for calls related to NFAS. If the active D-channel fails, the system switches to the backup D-channel to support call processing. This functionality is mainly for North America. NFAS is required in order to program D-channel Backup.
- **QSIG** - This feature provides the ability to connect PBXs from different vendors together to form a private network. QSIG also allows you to program Calling Name for incoming calls.
- **Remote LAN Access** - This feature provides LAN access to the wide area network (WAN) for both incoming and outgoing calls through LAN servers (routers or bridges).
- **Multiple Variants and Configurations** - This feature provides the ability to run multiple protocol variants and program multiple configurations on the two links of the PRI Gateway. The option to run multiple variants allows you to connect the PRI Gateway to two different CO switches. The option to run multiple configurations allows you to program Network-side on one link of the PRI Gateway and program User-side on the other link of the PRI Gateway.

The PRI card interfaces SUPERSET telephones, attendant consoles, and industry-standard telephones with the ISDN network.

PRI Card Options

FOR manages and controls the following options that the PRI card provides. The enabled options will be available to all the PRI cards installed within the same system but must be programmed via IMAT on the PRI cards that will use the features.

- NFAS
- D-channel backup
- Min/Max
- Auto Min/Max
- QSIG
- Remote LAN Access

PRI Maintenance

IMAT for Windows

The PRI card requires IMAT (ISDN Maintenance and Administration Tool) for programming and maintaining the PRI configuration databases. IMAT runs on Windows PC and connects to the PRI card locally through a LAN or straight-through serial connection or remotely through modems.

The IMAT software is on the PRI Card Software CD-ROM that comes with the PRI card. To install the IMAT application see the instructions on the PRI Card Software CD-ROM case. For further information, see Using the ISDN Maintenance and Administration Tool (IMAT).

Serial Port Characteristics

The serial ports (J10 and J11) on the back of the SX-200 ELx cabinet Rev 4.4 or greater (PN 9109-600-002-NA) connects the PC to the PRI card when you perform field installation, maintenance procedures, software upgrades, or want access to logs.The serial ports are terminated at the back of the cabinet by a female DB-9 connector.

The characteristics of the serial ports are as follows:

Data bits	8
Parity	None
Stop bits	1
Baud rate	38400
Error control	None
Flow control	None

Reserve Power Supply

The reserve power supply is a stand-alone Uninterruptible Power Supply (UPS) comprising a battery pack, a charger, and an inverter; it is not manufactured by Mitel Corporation. The UPS backup time is dependent upon the unit selected and the capacity of the batteries provided. The unit must meet the specifications provided below.

Please note that compliance to electrical, installation, and building codes is the responsibility of the purchaser of the equipment. Consult local municipal and electrical utility authorities before proceeding with the installation of equipment.

The UPS should be a true Uninterruptible Power Supply which always supplies the output load from its inverter. The UPS must include a reverse transfer switch to automatically bypass itself if it fails. The UPS must be capable of driving rectifier capacitor loads.

Rating	1.5 kVA minimum
Input Voltage	120 Vrms +10%,-15%, 60 Hz 15%
Output Voltage	120 Vrms +10%,-15%, 60 Hz 5%
Output Waveform	Sinewave or Quasi square wave (not square wave)
Transfer time	Less than 30 ms (includes fail detection and transfer time)
Output Receptacle	NEMA 15 A 3-pin grounded
Holdup/Recharge Times	Per customer requirements.

An Uninterruptible Power Supply (UPS) can have an external connection (from an internal relay) which provides a closed contact to remotely indicate status or condition. Conditions which may be indicated include:

- an ALARM condition is present within the UPS

- the UPS is operating from its batteries (probably because commercial ac power has been interrupted).

The relay contact may be connected to a remote alarm or to a “Contact Monitor” line circuit to promptly indicate the condition.

Refer to the manufacturer’s installation manual, which describes conditions that are indicated. Refer to the Program Features section, for a description of “Contact Monitor” line circuit operation.

Cabling Characteristics and Guidelines

Ethernet Cabling Guidelines

The SX-200 ICP transmits voice communications over a data network using Voice over Internet Protocol (VoIP). Electrical interference in the environment can reduce the quality of the voice and data signals that are transmitted over Ethernet cable. Desktop computers, printers, servers, lighting and other office devices place a high demand on the electrical infrastructure and increase the risk of electrical interference. You can minimize electrical interference and improve network efficiency by following the cabling guidelines detailed in this section and in the Engineering Guidelines document on the Mitel Edocs website (http://edocs.mitel.com).

Note: Special testing equipment is available from Mitel Product Support that can verify Ethernet cable performance and detect cable faults.

Maximum Cable Lengths

Limit the CAT 5 or CAT5e Ethernet cable runs to the lengths listed in the following table.

Table: Maximum Cable Lengths

Cabling	Maximum Length
Under Desk	3 m (10 ft)
Floor to ceiling	3.5 m (11.5 ft)
Horizontal span in ceiling	90 m (300 ft)
Equipment closet	3.5 m (11.5 ft)
Total maximum length	100 m (333 ft)

Cabling Guidelines for the Desktop

Refer to the following figure and the corresponding cabling guidelines to minimize electrical interference at the desktop.

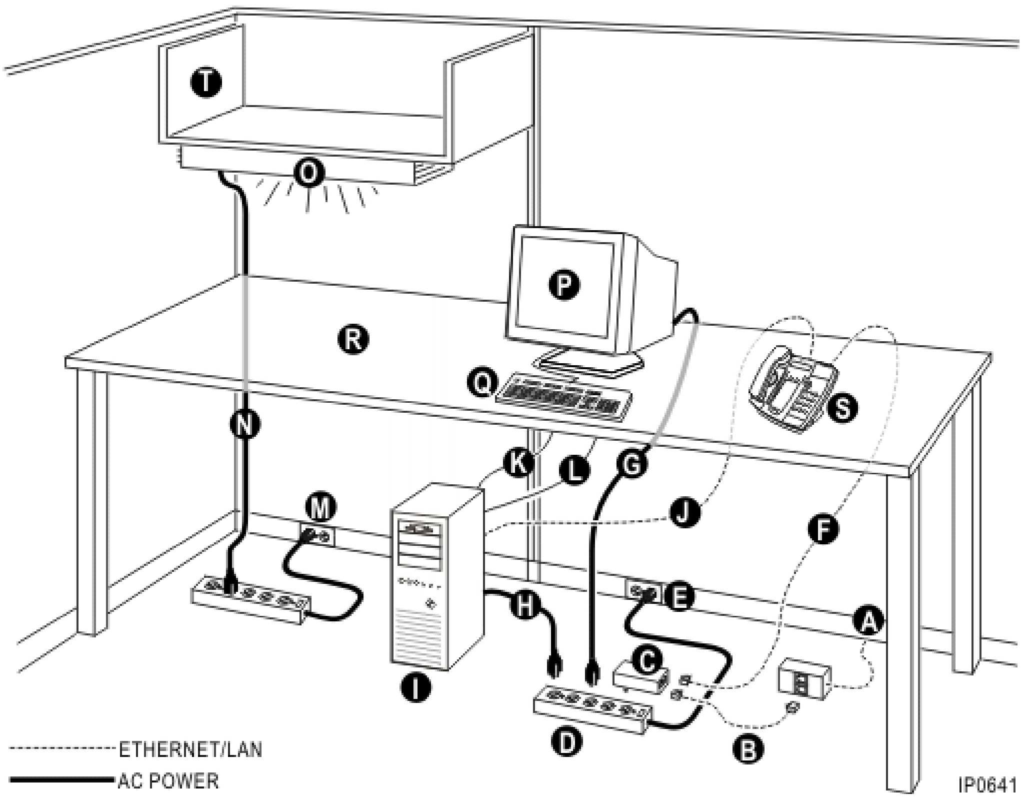


Table: Desktop Environment Cabling Guidelines

Connection		Guideline
A	Ethernet connection to network	Use CAT 5 or CAT 5e Ethernet cables and certified network connection blocks (TIA/EIA 568A). Adhere to the cable lengths listed in the table above.
B	Network to power adapter Ethernet patch cable (for local power configuration only)	

• Use CAT 5 or CAT 5e cables (certified to TIA/EIA 568A)

• Maximum length of 1.5 m (5 ft) • Route cable away from sources of interference, such as power cables		
C	IP Phone power adapter (for local power configuration only)	Plug IP Phone power adapter and the computer into the same surge suppressing power bar
D	Surge suppressing power bar	

• Recommended model is American Power Conversion “SurgeArrest”

• Route power cables away from Ethernet cables	
E	Power outlet for desktop equipment

- Use outlet to supply power to computer and IP Phone power adapter only

• Do not plug other devices such as florescent lights, coffee makers, kettles into this outlet	
F	Phone to power brick Ethernet connection

- Use CAT 5 or CAT 5e cables (certified to TIA/EIA 568A)

• Maximum length of 1.5 m (5 ft) • Route cable away from sources of interference, such as power cables	
G	Monitor power cord

- Plug into computer power bar

• Route cable away from Ethernet cables	
H	Computer power cord
I	Computer
J	Phone to computer Ethernet connection

- Use CAT 5 or CAT 5e cables (certified to TIA/EIA 568A)

• Maximum length of 1.5 m (5 ft) • Route cable away from sources of interference, such as power cables		
K	Keyboard to computer connection	Route cable as required for convenience
L	Monitor to computer connection	Route cable away from Ethernet cables
M	Power connection to auxiliary equipment	

- Use a separate power outlet for potential noise generating devices such as a lamp, coffee maker, or radio

• Route cable away from Ethernet cables • Ballast circuitry inside florescent lamps will create noise spikes on power cables when the lamps are turned off. Ensure that florescent lamps are plugged into dedicated surge suppressing power bars. Voltage limiting devices inside the power bars reduce noise spikes and reduce the risk of data errors. Some desks have power outlets that are designated for the computer and utility devices. These outlets have built-in surge protection. In this case, a power bar is not required.	
N	Florescent light power cord
O	Florescent desk light

Cabling Guidelines in the Equipment Room

Refer to the following figure and the corresponding cabling guidelines to minimize electrical interference in the equipment room.

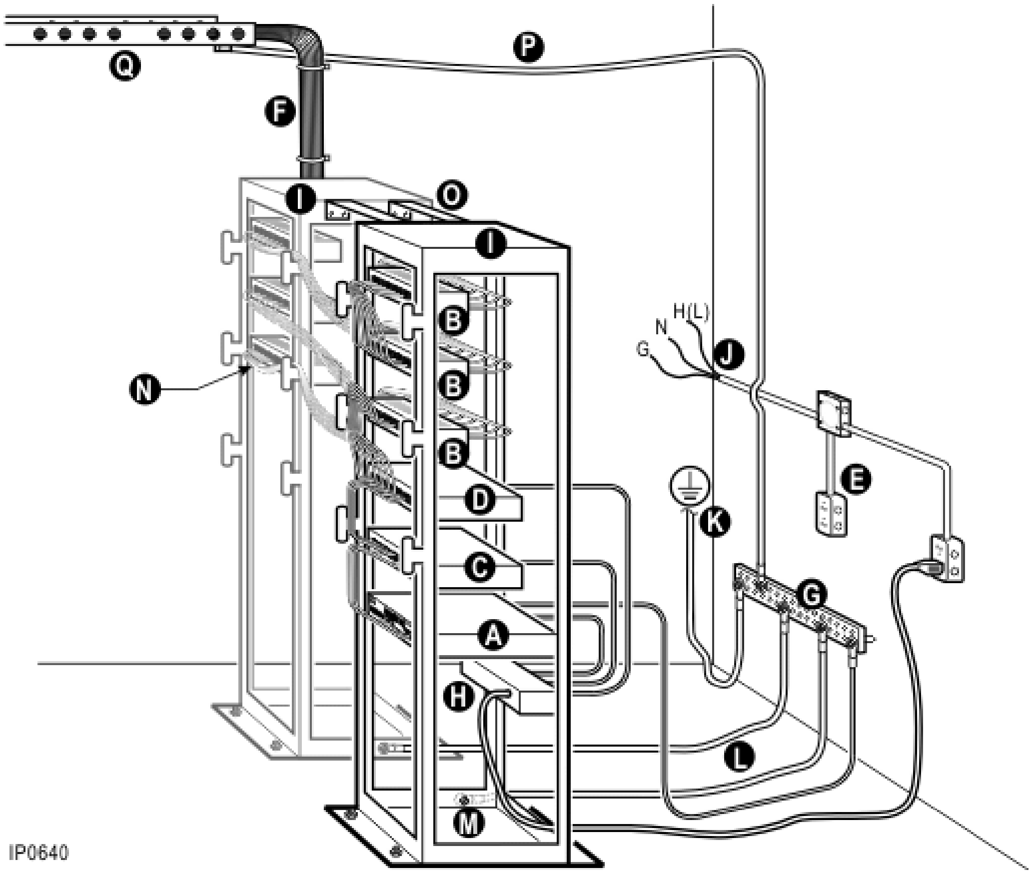


Table: Equipment Room Cabling Guidelines	
Connection	**Guideline**
A SX-200 ICP Controllers	

• Use CAT 5, CAT 5e, or CAT 6 Ethernet connector blocks and cables.

• CAT 5e or CAT6 certified cable provides better immunity to crosstalk. • Connect ground stud on back of controller to ground bar bus (G) with a dedicated ground wire.	
B	Patch panel

• Patch panels must be certified for CAT 5 cable.

• Do not use punch-down blocks that are designed for voice-grade telephony signals to interconnect 100 Mb/s Ethernet signals. • Recommended connector blocks can be obtained from www.anixter.com (part number 201011).		
C	Layer 2 switch	

- If a ground stud is provided, connect it to ground bar bus (G) with a dedicated ground wire. Use the wire gauge specified by the manufacturer.

• Powered hubs supply IP Phones with power through the Ethernet cable either through the spare wires or the signal pairs. IP Phones that have power adapters do not use powered hubs. • If a ground stud is provided, connect it to ground bar bus (G) with a dedicated ground wire. Use the wire gauge specified by the manufacturer.		
D	Powered hub	

- Powered hubs supply IP Phones with power through the Ethernet cable either through the spare wires or the signal pairs. IP Phones that have power adapters do not use powered hubs.

• If a ground stud is provided, connect it to ground bar bus (G) with a dedicated ground wire. Use the wire gauge specified by the manufacturer.		
E	Protected power system	

- Dedicate the use of the power outlets to the equipment in the equipment room only.

• Ensure that the power outlets in the equipment room are wired for 15 Amp service directly to the electrical service panel with ideally one circuit breaker per outlet. • If the site is configured with resilient IP phones, ensure that the 3300 controllers are powered by dedicated • Switching power supplies common in computers and telecommunications equipment generate noise voltages, known as harmonics. Use oversize neutral conductors to minimize harmonics. • Ensure that conduits include a dedicated copper ground		
F	Cable	Ensure that the maximum cable runs 100 m.
G	Ground bar bus	

- Use a ground bus bar that is ¼ inch thick and 2 inches wide and long enough to accommodate the grounding for all the rack-mounted equipment

- Recommended bus bar is ANIXTER part number 179639
- Mount the bus bar on the wall with insulated standoffs
- Use compression style fittings to fasten the ground wire lugs to the bar
- Connect the bus bar to the main building ground with a 6 AWG copper stranded green-colored cable.
- For grounding specifications see ITU-T K.27 “Building configurations and earthings inside a telecommunications building” and ANSI/TIA/EIA-607

H	Protected rack-mount power strip

- If you cannot provide dedicated 15 Amp power outlets for each unit in the rack, mount a surge arresting power strip on the front or rear of the rack.

- Recommended model is the Surge Arrest - Rack Mount model from American Power Conversion. The ground from the rack forms part of the shield for the power strip.
- Plastic floor type models are not recommended because they are more likely to be turned off by accident.

Caution: Power bars have a circuit breaker. If the circuit breaker is tripped due to a power surge, the power to all the outlets on the power bar is shut off. If the site supports resilient IP Phones, ensure that the controllers are plugged into different power bars.

I	Standard metal rack

- Bolt each rack securely to the floor and connect a dedicated ground wire between the frame and the ground bus bar.

- If rack-mounted equipment obtains safety ground from the metal rack, ensure that a good electrical connection is made between the rack and the cabinet metalwork. Use “star” washers to obtain a solid electrical connection to painted cabinets
- Route any power cables contained within the rack away from any UTP patch cabling

Note: Fiber optic cabling can be routed anywhere within the rack because it is not susceptible to electrical emissions.

J	AC Mains metal conduit	Metal conduit that contains power wiring must three wires for each dedicated circuit: Ground (bare), Neutral (Black), White (Hot or Line). Do not use the conduit as the ground.
K	Keyboard to computer connection	

Main ground connector must be 6 AWG stranded copper, green colored cable connected to the main building ground.

Caution: A proper ground is required for proper equipment operation and safety. A power quality engineer can provide advice on new and existing installations. Refer to G for additional information.

L	Rack grounds	Use separate wires to ground each rack to the ground bus bar.
M	Equipment grounds	Use separate wires to ground each piece of equipment to the ground bus bar. If a ground stud is provided on the back of the unit, connect it to the ground bus bar with a dedicated ground wire (use the gauge specified by the manufacturer).
N	Patch cable	

Interconnect Ethernet equipment supporting 100 Mb/s transmission with CAT 5 UTP patch cable. Label the cables and route them neatly through the channels provided in the metal rack.

Caution: Do not use voice grade twisted pair interconnect patch cable wired to standard voice grade punch-down blocks on an IDE.

O	Metal rack interconnect	The metal brackets used to connect the racks provide mechanical connection only. Use a dedicated ground wire to ground each rack separately to the ground bus bar.
P	Cable tray ground	Connect the metal racks that house the Ethernet cable to the ground bus bar to provide an effective shield against from potential noise sources such as power lines and florescent lights.
Q	Cable tray	The tray should contain Ethernet cables only. Do not mix power cables with Ethernet cables

Category 3 Cabling Guidelines (CX/CXi)

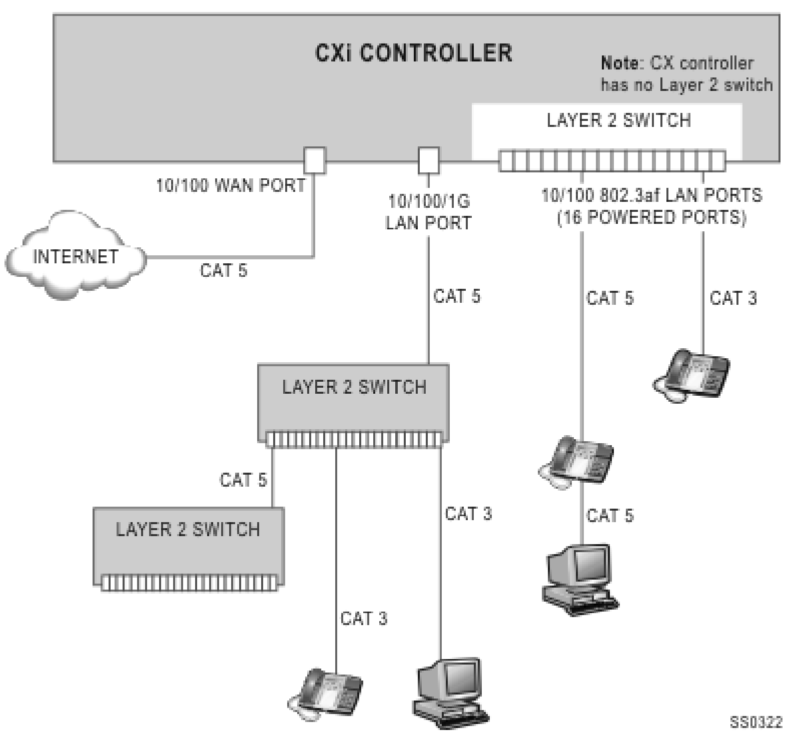
Category 5 cable is recommended for all network connections. However, the CX/CXi controller can support Category 3 cable subject to the following general guidelines:

- Category 3 cable can be used for one purpose only: to connect single Ethernet devices (IP phones or PCs) directly to the controller or a Layer 2 expansion switch. Category 5 (or better) cable is required for all other Ethernet connections, including:
  - connections from "dual devices" (IP phone and PC) to the controller or an expansion switch
  - uplink connections between switches
  - connections from the Internet to the 10/100 WAN port
  - connections to the T1/E1 Combo module
  - connections to the Quad Copper Interface Module from ASUs, NSUs, or Peripheral Cabinets
- If a connection uses Category 3 cable, restrict its port speed to 10 Mbps and half duplex. It is preferable to program the settings on the switch (onboard or expansion) rather than the connected device.
- Category 3 cable can supply Power over Ethernet (PoE) via the controller's phantom power feed.
- The cable installation must meet Category 3 standards (TIA/EIA-568).

Notes:

1. Detailed guidelines concerning the use of Category 3 cable can be found in the Engineering Guidelines document on the Mitel Edocs website (http://edocs.mitel.com). Refer to this document before starting the installation.
2. Some expansion switches may not support PoE with Category 3 cable.
3. Category 3 cable may be used for telecom installations, but not all telecom cable is Category 3-compliant.

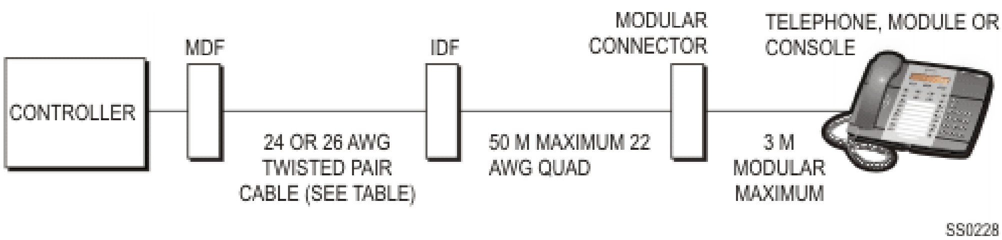
Minimum Ethernet Cabling Requirements (CXCXi)



Telephony Loop Lengths

The following rules for loop lengths between a peripheral device and a DNIC port on the controller or a Digital Line Card in a Peripheral cabinet must be followed for proper operation of the device:

Maximum loop length (twisted pair) 24 or 26 AWG	see table below.
Maximum length of quad cable (22 AWG)	50 m (160 ft)
Modular Line Cord	3 m (10 ft)



Note: Refer to the Engineering Guidelines at http://edocs.mitel.com for information on setting up analog trunks, in particular the section on "Losses in a PSTN Network."

Table: Loop Lengths for Controller Interfaces

Interface	Wire Gauge (AWG)	Loop Length
DNIC Devices (MX controller only) (Connected to Digital Telephone)	24 26	1006 m (3300 ft) 1006 m (3300 ft)
(Connected to DNIC Console)	24 26	1006 m (3300 ft) 1006 m (3300 ft)
ONS Lines	22 AWG 26 AWG	3800 m (12350 ft.) with 200 Ohm phone 1500 m (4875 ft.) with 200 Ohm phone

Table: Loop Lengths for Peripheral Devices	
Peripheral Device	**Maximum Loop Length Without Bridge Tap**
SUPERSET 4001 SUPERSET 4015 SUPERSET 401+ SUPERSET 410	1000 m
SUPERSET 4025 SUPERSET 4125 SUPERSET 4150 SUPERSET 420 SUPERSET 430 SUPERSET 4DN SUPERSET 3DN DSS/BLF Interface Unit SUPERCONSOLE 1000 Attendant Console	1000 m
Datasets 1103 and 2103	2000 m

Table: Loop Lengths for Peripheral Cabinet Interface Cards		
Card Type	**Wire Gauge (AWG)**	**Loop Length**
ONS Card	22 24 26	3560 m (11700 ft) 2250 m (7400 ft) 1400 m (4600 ft)
OPS Card	22 24 26	15240 m (50,000 ft) 11520 m (37800 ft) 5850 m (19,200 ft)
COV Card	22	

24 26		
2000 m (6600 ft)		
1500 m (5000 ft) 1000 m (3300 ft)		
E & M Trunk		
22		
24 26		
2715 m (8900 ft)		
1708 m (5600 ft) 1068 m (3500 ft)		
DID Trunk Card - CO Trunk resistance	na	2240 ohms
LS/GS Trunk Card - CO Trunk resistance	na	1600 ohms
LS/CLASS II Module - CO Trunk resistance	na	1600 ohms
T1 Trunk (cable lengths not loop lengths for 22 gauge wire) S1 only closed: 0 - 45.8 m (0 - 150 ft) S2, S3, S4 closed: 45.8 - 137.3 m (150 - 450 ft) S5, S6, S7 closed: 137.3 - 200 m (450 - 655 ft)	22	These are cable lengths, not loop lengths. Set DIP switches on T1 Trunk card for correct equalization depending on cable length between the T1 Trunk and the Channel Service Unit (CSU).

General

This section outlines the signaling and supervision parameters of the SX-200 ICP system.

The standard range of tones are available from the Digital Signal Processor:

- 12 DTMF sets of tones, as listed in DTMF Tone Parameters.
- A set of call progress tones as listed in Call Progress Tones and Ringing Cadences, which form part of the country's Audible Tone Plan.
- One ringing tone of 20 Hz.

The system is capable of accepting and repeating signals from telephones which have the parameters shown in DTMF Tone Parameters and Dial Pulse Reception Limits.

Note: Only TDM bay are capable of interpreting dial pulses.

Where any of the frequencies shown in Call Progress Tones and Ringing Cadences are present at the system input, any other single frequency (200 - 3400 Hz) should be a minimum of 40 dB below the signal frequency. DTMF pulses are registered in the presence of precise dial tone at a level of -10 dBm.

The system gives the following output signal conditions:

- Dial Pulse Conditions:

  Pulse Rate : 9 to 11 pps
  Break Interval : 58% to 62%
  Interdigit Time : 800 ms.

Note: Only TDM bay are capable of interpreting dial pulses.

- DTMF Dialing Conditions for North America:

  Frequency Deviation : ± 1%
  Tone Duration : greater than 90 ms
  Interdigit Time : greater than 100 ms
  Level, low group : greater than -4 dBm
  Level, high group : greater than - 4 dBm
  Level, DTMF signal : less than -1 dBm
  Level, third Harmonic : better than 40 dB Frequency below DTMF signal
  Twist : 0 dB.

DTMF Tone Parameters

DTMF Tone Parameters

	High Frequency (Hz)		
Low Frequency (Hz)	**1209**	**1336**	**1477**
697	1	2	3
770	4	5	6
852	7	8	9
941	*	0	#

Frequency deviation: ± 1.5 %
Signal interval (2 frequency): 40 ms (min)
Per frequency, minimum level: -17 dBm on line circuit
Twist, maximum (at -10 dBm): +4 to -8 dB (high frequency relative to low frequency)

Dial Pulse Reception Limits

Dial Pulse Reception Limits

Note: Only TDM bay are capable of interpreting dial pulses.

Parameter	Min	Max
ONS Line:		
Pulse Rate	8 pps	12 pps
Break Duration	58%	64%

Interdigit Time	300 ms	15 s
OPS Line:		
Pulse Rate	8 pps	12 pps
Break Duration	42%	84%
Interdigit Time	300 ms	15 s

Loss and Level Plan

North American Loss and Level Plans

The purpose of a transmission loss and level plan is to provide an acceptable transmission level to all subscribers in the telephone network. The AX, CX/CXi and the MX use a loss level plan that is TIA/EIA-912 compliant.

PBX Loss Plan

The SX-200-ICP uses a loss level plan that is compliant with the TIA/EIA-912-A standard. This standard uses a nomenclature for naming interfaces that is different than what was used in the past for the SX-200-ICP. The following table lists the various interface circuits supported by the SX-200-ICP, provides the EIA/TIA-912-A naming convention and the legacy naming convention. The following table should be used to determine the TIA-912-A circuit types that these satellite circuits have been remapped to, grey shading indicates circuits that are no longer supported

TIA-912 Interface Circuit Reference	Mitel Interface Circuit Reference	Description
ONS On Premise Station	ONS	

On Premise Station, this circuit is used to connect an industry-standard telephone to the ICP when the telephone is located on premise.

CAUTION: ONS circuits must not be used to support OPS connections since the ONS circuit does not provide protection against hazardous voltages.	
ONS On Premise Station	OPSS

Off Premise Station Short, this circuit is used to connect an industry-standard telephone to the ICP when the telephone is located off premise and is within 2 kilometers of the ICP. For safety reasons the OPS line card must be used to provide protection against hazardous transients.

	CAUTION: ONS circuits must not be used to support OPS connections since the ONS circuit does not provide protection against hazardous voltages.
	OPS Off Premise Station

OPSL

	Off Premise Station Long, this circuit is used to connect an industry-standard telephone to the ICP when the telephone is located off premise and is further than 2 kilometers away from the ICP. For safety reasons the OPS line card must be used to provide protection against hazardous transients.
OPS Off Premise Station	

OPSLS

	Off Premise Station Long Satellite, this circuit is no longer supported, selecting OPSLS through CDE will yield an OPS setting.	
DGS Digital Station	DNIC	**Digital Network Interface Circuit,** this circuit is used to connect Mitel proprietary DNIC telephones to the ICP.
FXO Foreign Exchange Office	ACO	**Analog Central Office,** this circuit is used to connect the ICP to a Central Office via an analog trunk. This circuit should be used for analog trunks that present between 3 dB and 8 dB of line loss.
FXD Foreign Exchange Office Digital	ATO	**Analog Toll Office,** this circuit is used to connect the ICP to a Central Office via an analog trunk. This circuit should be used for analog trunks that present between 0 dB and 3 dB of line loss.
ATT Analog Tie Trunk	ATT	**Analog Tie Trunk,** this circuit is used to provide a connection between this PBX and another PBX, over a 2 or 4 wire analog trunk.
ATT Analog Tie Trunk	SATT	**Satellite Analog Tie Trunk,** this circuit is no longer supported, selecting SATT through CDE will yield an ATT setting.
ATT Analog Tie Trunk	SATTS	**Satellite Analog Tie Trunk Short,** this circuit is no longer supported, selecting SATTS through CDE will yield an ATT setting.

DAL Digital Access Line	DCO	**Digital Central Office,** this circuit is used to connect the ICP to a Central Office or another PBX via a digital trunk.
DAL Digital Access Line	DTO	**Digital Toll Office,** this circuit is no longer supported, selecting DTO through CDE will yield a DAL setting.
DAL Digital Access Line	IST	**ISDN Trunk,** this circuit is used to connect the ICP to a Central Office via a digital trunk.
DAL Digital Access Line	DTT	**Digital Tie Trunk,** this circuit is used to provide a connection between this PBX and another PBX over a digital trunk.
DAL Digital Access Line	SDTT	**Satellite Digital Tie Trunk,** this circuit is no longer supported, selecting SDTT through CDE will yield a DTT setting.
DAL Digital Access Line	SDTTS	**Satellite Digital Tie Trunk Short,** this circuit is no longer supported, selecting SDTTS through CDE will yield a DTT setting.
DAL Digital Access Line	SDTTO	**Satellite Digital Tie Trunk with OPS,** this circuit is no longer supported, selecting SDTTO through CDE will yield a DTT setting.
DAL Digital Access Line	SDTTOS	**Satellite Digital Tie Trunk with OPS Short,** this circuit is no longer supported, selecting SDTTOS through CDE will yield a DTT setting.
FXD Foreign Exchange Office Digital	LSC	**Loop Start Class,** this circuit applies to the LS/CLASS card on the Spine Bay, the loss levels for this circuit are set to comply with FXD. Note: the Spine Bays are not currently supported on the SX-200-ICP.
ONS On Premise Station	iONS	

IP On Premise Station, this circuit is used to connect an industry-standard telephone to the ICP when the telephone is located on premise. This circuit type is located on the ASU which is designed to comply with IP connected half-channel loss plan.

CAUTION: ONS circuits must not be used to support OPS connections since the ONS circuit does not provide protection against hazardous voltages.

ONS On Premise Station	abONS

Analog Board On Premise Station, this circuit is used to connect an industry-standard telephone to the ICP when the telephone is located on premise. This circuit type is located on the Analog Main Board and the Analog Option Board.

CAUTION: ONS circuits must not be used to support OPS connections since the ONS circuit does not provide protection against hazardous voltages.		
FXO Foreign Exchange Office	abACO	**Analog Board Analog Central Office,** this circuit is used to connect the ICP to a Central Office via an analog trunk. This circuit should be used for analog trunks that present between 3 dB and 8 dB of line loss. This circuit type is located on the Analog Main Board and the Analog Option Board.
FXD Foreign Exchange Office Digital	abACOs	**Analog Board Analog Central Office Short,** this circuit is used to connect the ICP to a Central Office via an analog trunk. This circuit should be used for analog trunks that present between 0 dB and 3 dB of line loss. This circuit type is located on the Analog Main Board and the Analog Option Board.
	abACOhl	**Analog Board Analog Central Office High Loss,** this circuit is used to connect the ICP to a Central Office via an analog trunk. This circuit should be used for analog trunks that present more than 8 dB of line loss. This circuit type is located on the Analog Main Board and the Analog Option Board.
DGS Digital Station	IPSET	**IP Set,** this circuit is used to connect IP telephones to the ICP, this circuit is also used to provide a connection between this PBX and another PBX over an IP trunk.

The following Table lists the station circuits supported on the SX-200-ICP, the hardware associated with these circuits and if applicable CDE settings.

SX-200-ICP Station Circuits	Mitel Interface Circuit Reference	TIA-912 Interface Circuit Reference	Hardware & Circuit Descriptor Information
On Premise Station	ONS	ONS	**SX-200-ICP Peripheral Unit** ONS CLASS Line Card
Off Premise Station Short	OPSS	ONS	

SX-200-ICP Peripheral Unit
OPS Line Card

CDE Form # 3 has the following specific settings: Option Number 402 Long Loop (Off Extensions Premise Only) = Disabled			
Off Premise Station Long	OPSL	OPS	

SX-200-ICP Peripheral Unit
OPS Line Card

CDE Form # 3 has the following specific settings: Option Number 402 Long Loop (Off Extensions Premise Only) = Enabled			
SS4xxx Digital Sets	DNIC	DGS	**SX-200-ICP Peripheral Unit** DNI Line Card
On Premise Station	iONS	ONS	Analog Service Unit (ASU)
On Premise Station	abONS	ONS	**SX-200-ICP** Analog Main Board & Analog Option Board
IP Set	IPSET	DGS	**SX-200-ICP** LAN Interface

The following Table lists the trunk circuits supported on the SX-200-ICP, the hardware associated with these circuits and if applicable CDE settings.

SX-200-ICP Trunk Circuits	Mitel Interface Circuit Reference	TIA-912 Interface Circuit Reference	Hardware & Trunk Descriptor Information
Analog Central Office (For long ACO trunks)	ACO	FXO	

SX-200-ICP Peripheral Unit

LS/GS Trunk Card CDE Form # 13 has the following specific settings: Line Length = Long **LS/CLASS Trunk Card** CDE Form # 13 has the following specific settings: Line Length = Long **DID Trunk Card** CDE Form # 13 has the following specific settings: Line Length = Long

Analog Toll Office (For short ACO trunks)	ATO	FXD

SX-200-ICP Peripheral Unit

LS/GS Trunk Card CDE Form # 13 has the following specific settings: Line Length = Short **LS/CLASS Trunk Card** CDE Form # 13 has the following specific settings: Line Length = Short **E&M Trunk Module** CDE Form # 13 has the following specific settings: Line Length = Short

Analog Tie Trunk	ATT	ATT

SX-200-ICP Peripheral Unit

DID Trunk Card CDE Form # 13 has the following specific settings: Line Length = Short **E&M Trunk Module** CDE Form # 13 has the following specific settings: Line Length = Short

Digital Central Office	DCO	DCO

SX-200-ICP Peripheral Unit

T1/E1 Module CDE Form # 13 has the following specific settings: Line Length = Long **T1 Trunk Card** CDE Form # 13 has the following specific settings: Line Length = Long ***SX-200-ICP MX*** **Dual T1/E1 MMC** CDE Form # 13 has the following specific settings: Line Length = Long ***SX-200-ICP CX*** **T1/E1 Combo MMC** CDE Form # 13 has the following specific settings: Line Length = Long		
ISDN Trunk	IST	DAL
SX-200-ICP Peripheral Unit		
PRI Line Card BRI Line Card Network Service Unit (NSU)		
Digital Tie Trunk	DTT	DAL
SX-200-ICP Peripheral Unit		

T1/E1 Module CDE Form # 13 has the following specific settings: Line Length = Short **T1 Trunk Card** CDE Form # 13 has the following specific settings: Line Length = Short ***SX-200-ICP MX*** **Dual T1/E1 MMC** CDE Form # 13 has the following specific settings: Line Length = Short ***SX-200-ICP CX*** **T1/E1 Combo MMC** CDE Form # 13 has the following specific settings: Line Length = Short		
Analog Central Office	abACO	FXO
SX-200-ICP **Analog Main Board & Option Board** CDE Form # 13 has the following specific settings: Line Length = Long		
Analog Central Office Extra Long	abACOhl	
SX-200-ICP **Analog Main Board & Option Board** CDE Form # 13 has the following specific settings: Line Length = X Long		
Analog Central Office Short	abACOs	FXD
SX-200-ICP **Analog Main Board & Option Board** CDE Form # 13 has the following specific settings: Line Length = Short		

Tone Levels

All of the tones used by the SX-200 ICP are provided by the Digital Signal Processor in the ICP and presented on the PCM bus for use by various interfaces.

Most call progress and DTMF tones require that the receive gain pad in the line or trunk circuit interface codec be set to a specific value for the duration of the tone. The interface settings for the call progress tones are outlined in Table: Interface Level (dBm). The DTMF levels and interface settings are outlined in Table: DTMF Levels- Trunk Interface Into 600 Ohms and Table: Interface Level (dBm) respectively.

Port-to-Port Losses

North America has stipulated requirements concerning acceptable transmission performance for telephone systems. The loss plan matrix provides the correct electrical losses in decibels (dB) for each connection to meet the TIA-912 requirements.

Loss plans have a direct effect on the acoustic levels provided at the set. It is generally desirable to achieve the same relative loudness levels for all standard and proprietary telephones for a specified loss plan, taking into account loop lengths, transmission format (analog or digital), different transducers in use, line/trunk impedances, and terminating impedances.

The values in the loss plan table are in decibels (dB) and represent the voice path loss from a particular port to a particular port. Note that since the matrix is a loss plan a positive number represents a loss and a negative number represents a gain.

An example of using the loss plan matrix is as follows:

While referring to the Loss Plan Matrix, suppose you wish to know what the port-to-port loss is for an abONS to an abACOs connection and an abACOs to an abONS connection.

1. Look up abONS in the left hand Port column and follow this row across to the abACOs column, the port-to-port loss from abONS to abACOs is 3 dB.
2. Look up abACOs in the left hand Port column and follow this row across to the abONS column, the port-to-port loss from abACOs to abONS is 3 dB.

TIA-912 Name			ONS	OPS	ONS	DGS	ATT	DAL	DAL	DAL	FXO	FXD	DGS	ONS	ONS	FXO	FXD
Mitel Name		ONS	OPSL	OPSS	DNIC	ATT	DCO	IST	DTT	ACOB	ATOB	IPSET	IONS	ABONS	ABACO	ABACOS	
		↑	-	-	-	-	-	-	-	-	-	-	-	-	-	-	
ONS	ONS	→	6	3	6	0	3	0	0	0	0	3	0	6	6	0	3
OPS	OPSL	→	3	0	3	0	3	0	0	0	0	0	0	3	3	0	0
ONS	OPSS	→	6	3	6	0	3	0	0	0	0	3	0	6	6	0	3
DGS	DNIC	→	9	6	7	0	3	0	0	0	0	3	0	9	9	0	3
ATT	ATT	→	3	0	3	0	0	0	0	0	0	0	0	3	3	0	0
DAL	DCO	→	9	6	7	0	3	0	0	0	0	3	0	9	9	0	3
DAL	IST	→	9	6	7	0	3	0	0	0	0	3	0	9	9	0	3

DAL	**DTT**	→	9	6	7	0	3	0	0	0	0	3	0	9	9	0	3
FXO	**ACO**	→	0	0	0	-6	0	-3	-3	-3	0	0	-3	0	0	0	0
FXD	**ATO**	→	3	0	3	-3	0	-3	-3	-3	0	0	-3	3	3	0	0
DGS	**IPSET**	→	9	6	7	0	3	0	0	0	0	3	0	9	9	3	6
ONS	**iONS**	→	6	3	6	0	3	0	0	0	0	3	0	6	6	0	3
ONS	**abONS**	→	6	3	6	0	3	0	0	0	0	3	0	6	6	0	3
FXO	**abACO**	→	0	0	0	-6	0	-6	-6	-6	0	0	-6	0	0	0	0
FXD	**abACOs**	→	3	0	3	-3	0	-3	-3	-3	0	0	-3	3	3	0	0
	abACOhl	→	-3	-3	-3	-9	0	-9	-9	-9	0	0	-9	-3	-3	0	0

Interface Levels

Interface Level (dBm)			
Mitel Circuit Reference	**Dial Tone**	**Reorder/ Busy**	**Ringback**
ONS	-23	-27	-22
OPSL	-20	-24	-19
SS4xx	-20	-24	-19
ATT	-20	-24	-19
DCO	-20	-24	-19
ACO	-20	-24	-19
ATO	-20	-24	-19
IPSET	-20	-24	-19
iONS	-23	-27	-22
iACOs	-23	-27	-22
abONS	-23	-27	-22
abACO	-23	-27	-22
abACOs	-23	-27	-22

DTMF Levels - Trunk Interface Into 600 Ohms

DTMF Levels - Trunk Interface Into 600 Ohms
Nominal level - low group: -6 dBm
Nominal level - high group: -4 dBm
Nominal level - frequency pair: -2 dBm
Nominal twist: 2 dBm

Call Progress Tones and Ringing Cadences

This Part describes the different call progress tones and ringing cadences that are available to support system requirements.

Tone Plan

The following table identifies the tones that are generated by the system in North America.

North America Tone Generation

Tone	Frequency 1	Frequency 2	Level With No Interface Gain/Loss
Ringing	20 Hz	---	90 Vrms
Dial tone	350 Hz	440 Hz	See Interface Level dBm table
Busy tone	480 Hz	620 Hz	
Ringback tone	440 Hz	480 Hz	
Special Busy	440 Hz	---	

Note: Ringing voltage is measured at source.

Ringer Cadencing

The following table identifies the ringer cadencing that is provided by the system in North America.

North America Ringer Cadencing

Discriminating Ring (Option 17) enabled and Discriminating Ring Always (Option 18) disabled			
ONS/OPS Internal (standard)	**ONS/OPS External (discriminating)**	**SUPERSET Telephones Internal (standard)**	**SUPERSET Telephones External (discriminating)**
.9 on 3.1 off	.350 on .200 off	1 on 3 off	.400 on .200 off
repeating	.350 on 3.1 off	repeating	.400 on 3.0 off
	repeating		repeating

Traffic Considerations

For details regarding traffic considerations and traffic engineering, refer to the Engineering Guidelines on the Mitel Edocs website at http://edocs.mitel.com.

Uninterruptible Power Supplies (UPS)

The SX-200 ICP requires a reserve power supply for the control cabinet and digital peripheral cabinets comprising of a battery pack, a charger, and an inverter. The UPS backup time is dependent upon the unit selected and the capacity of the batteries provided. The unit must be able to provide 115 Vac at 15 A and the following amount of power:

MX controller: 60 W
CX/CXi controller: 250 W (assuming that all 16 ports are feeding power to the telephones)

Any UPS may be used with the system, provided that it meets the requirements specified above. Since these are available from third-party suppliers (Mitel does not manufacture UPS units) no installation or troubleshooting procedures for the UPS are provided in this document. Reference should be made to the documentation provided by the manufacturer of the UPS for installation and troubleshooting information.

System Fail Transfer

SFT Operation

In the event of a system failure, the SX-200 ICP controller switches two phones on ONS circuits at bay/slot/circuit locations 1/13/3 and 1/13/4 to Central Office (CO) trunks at locations 1/13/7 and 1/13/8 respectively.

The CX/CXi controller provides additional SFT capability on AOB ONS circuits 1/13/19 and 1/13/20, which switch over to trunk circuits 1/13/13 and 1/13/14 respectively.

The following conditions will cause system fail transfer:

- Commercial power failure (if no reserve power supply is used)
- Failure of the SX-200 ICP controller.
- Controller reset.

When the relays close, the ONS ports are automatically connected to the LS trunk circuits. After the power is restored and communication is re-established with the controller, the relays open and the ONS and LS circuits function normally again.

When a system or network failure occurs, the controller checks for active calls on the ONS lines and will only switch the ONS phones when they are free. When the network recovers, the controller checks for loop current on the CO lines (active calls) and will only switch to normal operating mode when no loop current is detected. This prevents the dropping of any on-going connection through the CO line during the recovery of the system.

To prevent dial pulses or hook flashes from triggering a switch to normal mode, loop current breaks shorter than 560 ms are ignored.

You should plan carefully where you want the two ONS ports to terminate within the network. If power is lost they will provide the only connectivity to the telephone network.

The LS trunks do not have to be dedicated to system fail transfer support. You can place them in a trunk group and use them to route calls to the central office during normal system operation.

Peripheral cabinets can be provisioned with a stand-alone, wall-mounted SFT device that can control six circuits, and up to four SFTs can be daisy-chained together to provide security for 24 internal extensions. For more information, see Install a System Fail Transfer Unit.

Calls in progress when SFT occurs will be dropped; however, calls made while in SFT mode will not be dropped when the system returns to normal operation, but will terminate normally at the end of the call.

System features are not available while SFT is in effect.

IEEE 802.3af Power Over Ethernet Standard (PoE) Advertisements

Power Class Advertisements

The following table can be used to determine Power Class Advertisements a phone will use:

Power Class Advertisements	
Device	**Class Advertised**
5201	0
5207	0
5010	0
5020	0
5020 + 5310 Conference Unit (Conference unit is powered with AC adapter 24 VDC)	0
5020+ PKM(s) (PKMs are powered with AC adapter 24 VDC)	0
5212	2
5215	0
5215 Dual Mode	2
5220	0
5220 + 5310 Conference Unit (Conference unit is powered with AC adapter 24 VDC)	0
5220+ PKM(s) (PKMs are powered with AC adapter 24 VDC)	0
5220 dual mode	2

Power Class Advertisements	
Device	**Class Advertised**
5220 dual mode + 5412 PKM	3
5220 dual mode + 5448 PKM	3
5220 dual mode + 5412 PKM + 5448 PKM	3
5220 dual mode + 5448 PKM + 5448 PKM	3
5220 dual mode + 5310 Conference Unit + Saucer	3
5220 dual mode + LIM	2
5224	2
5304	2
5312	2
5324	2
5330	2
5540	2
Telematrix 3000IP	2

Safety Instructions

READ ALL INSTRUCTIONS BEFORE ATTEMPTING TO INSTALL OR USE THIS PRODUCT

Introduction

WARNING: Read all instructions before attempting to install or use this product.

CAUTION: Changes or modifications not expressly approved by Mitel could void the user's authority to operate the equipment.
Introduction

These instructions are intended as a general guide to provide basic installation information which is necessary for the proper and safe functioning of this equipment. Only trained, qualified service personnel shall install or maintain this product.

WARNING: Failure to follow all instructions may result in improper equipment operation and/or risk of electric shock.

- Read and understand all instructions. Keep these instructions with the equipment.

- Do not attempt to install or service this equipment unless you are skilled in the installation and maintenance of electronic telecommunication equipment and have successfully completed specific training for this equipment.
- This product must be installed and serviced in accordance with this document and the information contained in the SX-200 ICP Technical Documentation.
- Follow all procedures outlined in the SX-200 ICP Technical Documentation in the sequence that is given.
- Install all assemblies using the procedures described in the Installation Information section.
- Configure this product with only the assemblies specified and in the locations stated in the SX-200 ICP Technical Documentation.
- Replace all guards or barriers. Close and screw the lids at the completion of installation or before returning the equipment to service.
- Grounding circuit continuity is vital for safe operation of telecommunication equipment. Never operate telecommunication equipment with the grounding conductor or AC power cord disconnected.
- Ensure grounding conductor and the AC power cord is installed before connecting telecommunication cabling to any system.
- Unplug all cabinets from the ac mains during servicing. To reduce static susceptibility, follow the steps listed below when servicing a cabinet:

  (a) Always attach the wrist strap from the cabinet being serviced.

  (b) Immediately place any item removed from a cabinet into an anti-static bag.
- Ensure the AC power outlet is installed near the equipment and that it is easy to access.

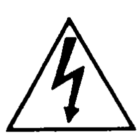
Use of Notices

The following information provides an explanation of the notices which appear on the product for this product.

DANGER	Danger indicates an imminently hazardous situation which, if not avoided, will result in death or serious injury.
WARNING	Warning indicates a potentially hazardous situation which, if not avoided, could result in death or serious injury.
CAUTION	Caution indicates a potentially hazardous situation which, if not avoided, may result in minor or moderate injury and/or damage to the equipment or property.

Use of Symbols

The following information provides an explanation of the symbols which appear on the product:

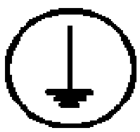


DANGEROUS VOLTAGE

The lightning flash with arrowhead symbol, within an equilateral triangle, is intended to alert the user to the presence of uninsulated “dangerous voltage” within the product’s enclosure that may be of sufficient magnitude to constitute a significant risk of electric shock to persons.

INSTRUCTIONS

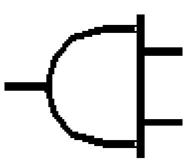
The exclamation point within an equilateral triangle is intended to alert the user to the presence of important operating and maintenance (servicing) instructions in the literature accompanying the product.



PROTECTIVE GROUNDING TERMINAL

The ground symbol within a circle identifies the terminal which is intended for connection to an external protective conductor. This connector must be connected to earth ground prior to making any other connections to the equipment.

General Installation Summary

Power Source	This product is intended to operate from an electrical branch circuit source rated at 120/230 volts RMS and 60 /50 Hz.
Equipment Location	This product shall only be situated in a clean and dry environment in accordance with the environmental and other installation requirements specified in the SX-200 ICP Technical Documentation.
	This product may not be provided with a power cord. Provision of a power cord (or replacement of a damaged/defective power cord) shall be in compliance with the electrical codes, in force, in the region of installation. The 3 conductor flexible cord shall have a voltage and current rating not less than the rating marked on this product. The polarized attachment plug shall have a current rating not less than 125 percent of the rating marked on this product. The flexible cord shall have a usage rating for floor mounted products (a typical example in the US would be an “S” or “SJ” rated cord). The cord shall not be longer than 2.4 meters.
MMC Modules	For an explanation of the steps required to install MMC Modules, refer to the procedure specified in the SX-200 ICP Technical Documentation.

Fuse and Component Replacement

WARNING: **Unauthorized repair of this product may result in a fire or shock hazard, and/or defective operation and/or equipment damage. Do not repair or replace components on circuit card assemblies or other parts of this equipment unless**

there is a specific description of the procedure provided in this set of technical practices. Return all inoperative assemblies to an authorized Mitel agent for repair.

Fuses identified with an electrical rating (voltage, current, type) shall be replaced with only the same type and rating. Never replace fuses with devices having different electrical ratings. Only those fuses installed in fuse-clips or fuse-holders shall be replaced in the field as directed by instructions in the SX-200 ICP Technical Documentation. Do not replace or attempt to bypass soldered in fuses on circuit card assemblies.

Refer to the SX-200 ICP Technical Documentation for information on the proper method of troubleshooting and servicing of this product.

Identification and Location of Circuit Card for Installation

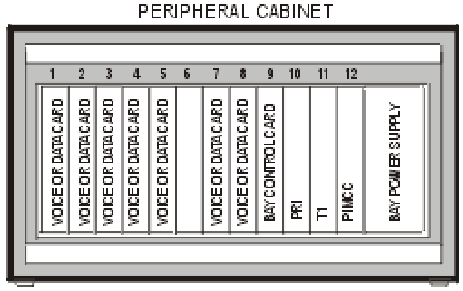
The mechanical design consists of an enclosure, a card cage, and an interconnecting backplane which define the arrangement and position of circuit card assemblies. Installation of the circuit card assemblies is performed by sliding it into the appropriate slot in the card cage.

The assembly physically interconnects into the system by first positioning it in front of the appropriate slot and then sliding the card along the card cage guides until the card is firmly seated into the mating connector on the backplane. Most circuit cards are installed in this manner. Refer to the following Figures for specific locations of the assemblies.

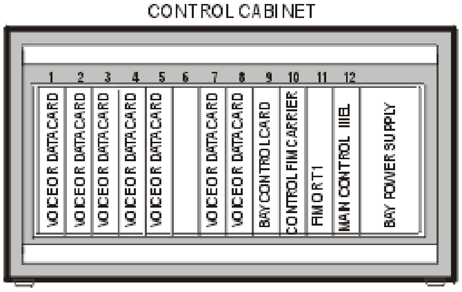
SX-200 EL CIRCUIT CARD LOCATIONS

SLOTS	CARDS
1 - 8	VOICE OR DATA CARD (LINE, ANALOG TRUNKS, BRI DIGITAL TRUNKS)
9	BAY CONTROL CARD (BCC II OR BCCIII)
10, 11	T1 CARD, CONTROL DUAL OR TRIPLE FIM CARRIER CARD, PRI CARD, CONTROL TRIPLE CIM CARD
12	MAIN CONTROL CARD, PERIPHERAL FIM CARRIER II CARD, PERIPHERAL INTERFACE MODULE CARRIER CARD

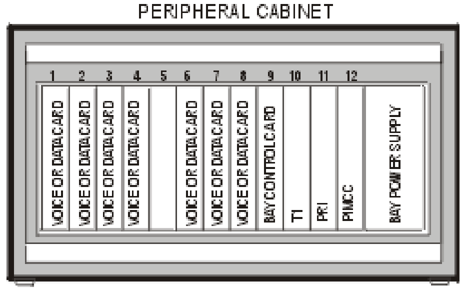
PERIPHERAL CABINET



CONTROL CABINET



PERIPHERAL CABINET



CC0922

Equipment Grounding

Redundant and independent equipment grounding conductors (see Note 1) are to be installed between the product and the wiring system ground.

One of the equipment's grounding conductors shall be an insulated grounding conductor (see Note 2) that is not smaller in size and is equivalent in insulation material and thickness to the grounded and ungrounded branch circuit supply conductor, except that it is green, with one or more yellow stripes. The grounding conductor is to be installed as part of the circuit that supplies that product or system and is to be connected to ground at the service equipment.

The other conductor (see Note 3) shall comply with the general rules for grounding contained in Article 250 of the National Electrical Code, NFPA 70, or section 10 of the Canadian Electrical Code, CSA C22.1, but shall not depend on the cord and plug of the product.

Notes:

1. There are two grounding conductors required to be installed with this equipment. One ground conductor is provided as part of the three wire 15 A branch circuit from which the product derives AC power. The other ground conductor is the supplementary or telecommunications ground.

The SX-200 ICP Peripheral Cabinets require separate and independent equipment grounding conductors. The supplementary ground is required because the cabinet contains telecommunications interfaces that connect to exposed or outside plant leads. These generally include LS/GS and DID trunk cards and OPS line cards.

The power cord for this product should only be replaced with one having the same number of conductors, gauge, insulation and usage ratings.

The telecommunications ground conductor shall be installed before installing other telecommunications wiring to the system. Multi-cabinet system installations may share a common ground conductor. Refer to the Installation Procedures section in the SX-200 ICP Technical Documentation for specific instructions for correct system grounding.

2. This grounding conductor is provided as part of the AC power cord-set provided with the equipment. The size of this conductor is allowed as stated in the National Electrical Code (NEC) in the United States NFPA/ANSI 70 section 250-95, Exception No. 1 which provides for compliance through section 240-4, Exception No. 1.

3. This grounding conductor is referred to as the telecommunications ground or supplementary ground as permitted in section 250-91 (c) of the NEC. This shall be an insulated #6 AWG, green or green and yellow striped wire which is to be connected to the protective grounding stud within the cabinet. The following symbol is located adjacent to the stud to identify the connection point for the grounding conductor:

Installation of Telecommunication Wiring

Telecommunication wiring to this product shall conform to all applicable safety and electrical wiring regulations. Installation of telecommunication wiring shall be performed following precautions in accordance with standard industry practice. The precautions to be followed include:

1. Never install telephone wiring during a lightning storm.
2. Never install telephone jacks in wet locations unless the jack is specifically designed for wet locations.
3. Never touch uninsulated telephone wires or terminals unless the telephone line has been disconnected at the network interface.
4. Use caution when installing or modifying telephone lines.
5. The SFT (System Fail Transfer) output connector shall not be connected to an off-premise application.

WARNING: **Never look directly into a fiber optic source or a fiber cable energized by a source.**

Avoid touching the ends of raw Fiber Optic Cables. Slivers can puncture the skin and cause irritation.

Any fiber cabling used to connect to a Peripheral Bay must not be encased in a metal sheath so as to ensure isolation between the two nodes. Conduit may be used to encase fiber cabling as long as the conduit does not contact the Peripheral Bay.

WARNING: Any connection of the assemblies listed below to an outside plant lead, an off-premise application or any other exposed plant application may result in a fire or shock hazard, and/or defective operation and/or equipment damage

Description / Common Name
POWER FAIL TRANSFER (PFT) CCT
MUSIC ON HOLD CCT
PAGING CCT
ONS LINE CARD CCT
DNIC MODULE CCT

Examples of installations which shall not be permitted for connection to these interfaces are those which:

(a) Require protectors in accordance with the National Electrical Code for the United States, NEC, NFPA / ANSI 70, Article 800-30, or,

(b) Are “Exposed Plant” as defined in the Canadian Electrical Code - CSA C22.1, paragraph 60-100 which states; “Exposed plant means where any portion of the circuit is subject to accidental contact with electric lighting or power conductors operating at a voltage exceeding 300V between conductors or is subject to lightning strikes.”

Regulatory Notices

NOTICE TO CANADIAN CUSTOMERS

NOTICE: This equipment meets the applicable Industry Canada Terminal Equipment Technical Specifications. This is confirmed by the registration number. The abbreviation, IC, before the registration number signifies that registration was performed based on a Declaration of Conformity indicating that Industry Canada technical specifications were met. It does not imply that Industry Canada approved the equipment.

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an approved method of connection. The customer should be aware that compliance with the above conditions may not prevent degradation of service in some situations.

Repairs to certified equipment should be coordinated by a representative designated by the supplier. Any repairs or alterations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request the user to disconnect the equipment.

Users should ensure for their own protection that the electrical ground connections of the power utility, telephone lines, and internal metallic water pipe system, if present, are connected together. This precaution may be particularly important in rural areas.

CAUTION: Users should not attempt to make such connections themselves, but should contact the appropriate electric inspection authority, or electrician, as appropriate.

NOTICE: The Ringer Equivalence Number (REN) for this terminal equipment is 1.0. The REN assigned to each terminal equipment provides an indication of the maximum number of terminals allowed to be connected to a telephone interface. The termination on an interface may consist of any combination of devices subject only to the requirement that the sum of the Ringer Equivalence Numbers of all the devices does not exceed five.

Industry Canada Notice IC: 173A-KTSICPA

This Class A digital apparatus meets all requirements of the Canadian Interference-causing Equipment Regulations.

NOTICE TO U.S. CUSTOMERS

This SX-200 ICP system equipment complies with Part 68 of the FCC rules. On the rear of this equipment is a label that contains, among other information, the FCC registration number and ringer equivalence number (REN) for this equipment. If requested, this information must be provided to the telephone company.

The FCC Registration Number for the system equipment is: US:BN2KF10BKTS or US:BN2MF10BKTS.

Port	FIC	SOC	REN	Jack
Loop Start	02LS2	N/A	1.0B	RJ21X
DS-1 (T1Module)	04DU9.1SN	6.0Y	N/A	RJ48C
Primary Rate ISDN Card	04DU9.1SN	6.0Y	N/A	RJ48C

The REN is useful to determine the quantity of devices you may connect to the telephone line. Excessive RENs on the telephone line may result in the devices not ringing in response to an incoming call. In most, but not all areas, the sum of the RENs of all devices connected to one line should not exceed five (5.0). To be certain of the number of devices you may connect to your line, as determined by the total RENs, contact the local telephone company.

If the PBX equipment causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. But if advance notice isn't practical, the telephone company will notify the customer as soon as possible. Also, you will be advised of your right to file a complaint with the FCC if you believe it is necessary.

The telephone company may make changes in its facilities, equipment, operations or procedures that could affect the operation of the equipment. If this happens, the telephone company will provide advance notice in order for you to make necessary modifications in order to maintain uninterrupted service.

If the equipment is causing harm to the telephone network, the telephone company may request that you disconnect the equipment until the problem is resolved.

For information concerning repairs that can be made by the customer, refer to Troubleshoot and Maintain in the Mitel Technical Documentation that is shipped with the equipment.

This equipment may not be used on public coin phone service provided by the telephone company. Connections to party lines service is subject to state tariffs. (Contact the state public utility commission, public service commission or corporation commission for information).

WHEN PROGRAMMING EMERGENCY NUMBERS AND (OR) MAKING TEST CALLS TO EMERGENCY NUMBERS

1. Remain on the line and briefly explain to the dispatcher the reason for the call.
2. Perform such activities in the off-peak hours, such as early mornings or late evenings.

ALLOWING THIS EQUIPMENT TO BE OPERATED IN SUCH A MANNER AS TO NOT PROVIDE FOR PROPER ANSWER SUPERVISION IS A VIOLATION OF PART 68 OF THE FCC's RULES.

Proper Answer Supervision is when

a) This equipment returns answer supervision signals to the PSTN when DID calls are
   - Answered by the called station
   - Answered by the attendant
   - Routed to a recorded announcement that can be administered by the CPE user
   - Routed to a dial prompt.

b) This equipment returns answer supervision on all DID calls forwarded to the PSTN.

Permissible exceptions are

- A call is unanswered
- A busy tone is received
- A reorder tone is received.

This equipment is capable of providing users access to interstate providers of operator services through the use of access codes. Modification of this equipment by call aggregators to block access dialing codes is a violation of the Telephone Operators Consumers Act of 1990.

FEDERAL COMMUNICATIONS COMMISSION (FCC) NOTICE

Note:

This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

Declaration of Conformity

Company: Mitel Networks Corporation
Model Numbers: Mitel Networks 3300/SX-200 ICP Controller
ICP Controller
ID#: US: BN2KF10BKTS

Samples of this product have been found to comply with the following standards:

- **FCC 47 CFR Part 68 (USA)**

This device complies with the applicable technical requirements of Part 68 of the Federal Communications Commission Rules and any applicable technical requirements referenced by this document.

This product, and all of its associated peripheral devices, complies with the requirements of 68.316 & 68.317.

Responsible party:

Mr. Ed
Silberhorn

U.S. General Counsel &
Corporate Secretary

Mitel Networks Inc.
205 Vanburen St.
Suite 400
Herndon Va. 20170

Signing Authority:

Mr. Greg Slingerland
Manager,
Regulatory Approvals
Mitel Networks Corporation

Mitel Networks Corporation
360 Legget Dr. P.O. Box 13089
Ottawa, Ontario, Canada
K2K 2W7

Mr. Greg Slingerland

13-May-2004
Date:

Declaration of Conformity

Company: Mitel Networks Corporation
Model Numbers: Mitel Networks 3300/SX-200 ICP Controller
ICP Controller
ID#: US: BN2MF10BKTS

Samples of this product have been found to comply with the following standards:

- **FCC 47 CFR Part 68 (USA)**

This device complies with the applicable technical requirements of Part 68 of the Federal Communications Commission Rules and any applicable technical requirements referenced by this document.

This product, and all of its associated peripheral devices, complies with the requirements of 68.316 & 68.317.

Responsible party:

Mr. Ed Silberhorn

U.S. General Counsel &
Corporate Secretary

Mitel Networks Inc.
205 Vanburen St.
Suite 400
Herndon Va. 20170

Signing Authority:

Mr. Greg Slingerland
Manager,
Regulatory Approvals
Mitel Networks Corporation

Mitel Networks Corporation
360 Legget Dr. P.O. Box 13089
Ottawa, Ontario, Canada
K2K 2W7

Mr. Greg Slingerland

Date: 13-May-2004

Install System

Overview

The equipment to be installed to assemble the system is shown in the Equipment Installed in the SX-200 ICP System table below. Some of the equipment listed as requiring installation may have been shipped already installed. If that is the case, the equipment should be removed and reseated during the installation procedure.

Pre-installation Preparations

Parts and Equipment

You need the following items and information:

Tools

- Static strap
- Phillips screwdriver (#1 and #2)

IMPORTANT: Use proper fitting screwdrivers to prevent damaging components and fasteners.

System Hardware and Software

- An SX-200 ICP Controller
- Optional hardware (DSP Modules, system hard drive, APC, APC Hard Drive, FIM/CIM modules etc.)
- SX-200 ICP software
- A Layer 2 Ethernet switch (a Managed switch is required if network includes PCs and other data devices)
- Mitel IP phones
- CompactFlash memory card for software installation

IMPORTANT: Use only Mitel-supplied CompactFlash cards. 256 MB is the minimum card size required. DO NOT partition nor copy files to the card before proceeding with the software installation.

Cables and connectors

- Category 5 (CAT5) cable for all LAN devices
- (IP phones and computers)
- CAT3 cable for any analog phones connected to the system
- RJ45 cable and connectors
- RJ45 crossover (patch) cable
- A power cable for the SX-200 ICP Controller (supplied)
- Crossover CIM cables if connecting to Peripheral Cabinet(s), NSUs, or ASUs
- FIM cables if connecting to Peripheral Cabinets, or NSUs

IMPORTANT: Cable plugs must meet FCC Rules part 68 subpart F for dimensions and registration. Use of non-conforming plugs can cause intermittent connections.

PC requirements

- Windows NT/98/2000/ME/XP/Vista PC or laptop
- Internet Explorer version 5.5 with service pack 2, or version 6 (recommended) for client-side rendering and 128 bit encryption (required for access to Mitel Online)
- Network Interface Card: Full Duplex 10/100M (we recommend 100M)
- for direct access, a serial cable to connect the PC to the SX-200 ICP Controller
- for remote or LAN-based connections, a secure Telnet client that supports SSL/TLS (Mitel Telnet client version 1.0.0.1 or later recommended), or a web browser (Internet Explorer 6 or Mozilla Firefox) to access the SX-200 ICP Web Interface

Note: Secure Telnet requires Windows NT/2000/ME/XP. Windows 98 is not supported.

- FTP Server—used for software upgrades, database backups, and uploading maintenance logs
- CompactFlash Reader with Read/Write capability

Line requirements

- LS/CLASS lines
- ONS/CLASS lines
- PRI-T1 lines (Dual FIM Module and Network Services Unit or Peripheral cabinet and PRI card required)

Note: Refer to the Engineering Guidelines on the Mitel Edocs website (http://edocs.mitel.com) for information on Losses in a PSTN Network.

LAN requirements

- A subnet
- SMTP Server IP address for forwarding voice mail to e-mail and for e-mail notification of 911 calls and system alarms (requires Advanced certification to implement)
- Customer data network information—for example, DNS server information (requires Advanced certification to implement)
- Router if using IP trunking or connecting to another network (requires Advanced certification to implement)

Other

- Feature codes and extension number plans
- A list of customer-purchased options
- Power for the system
- An uninterruptible power supply (recommended)
- Power source with surge protection for IP Phones
- A power cable for the SX-200 ICP Controller (supplied)
- Music on Hold source (radio, tape player etc.)
- (Optional) External Paging amplifiers and speakers.
- (Optional) SMDR printer

Site Requirements

The SX-200 ICP Controller must be located in an area that is

- Dry and clean
- Easily accessible

Do not locate the system near

- Sprinkler systems, sweating pipes, steam pipes, or steam vents
- Corrosive fumes or exhaust from machinery
- Electronic equipment that generates strong radio frequency fields (such as a radio or television receiver)
- Equipment that generates strong magnetic fields that can corrupt data on hard disks or floppy disks
- Reproducing machines.

System Power

Power wiring to the system must conform to the requirements of the local electrical code. The system power is supplied from a commercial AC power supply. Each unit is powered individually.

The system power must meet the following requirements:

- The units may share a branch circuit that is dedicated to one system only; the circuit must not be shared with any other equipment.
- Each receptacle of the branch circuit should have a rating of 120 V, 60 Hz, 15 A (or 230 V, 50 Hz, 8 A for 230 V systems)
- If the total power requirements of the entire system exceed the rating of one branch circuit, individual dedicated branch circuits may be installed for each unit.
- Each unit must have one three-wire power receptacle with the ground wire connected to the ground of the electrical system. Do not attempt to defeat the grounding conductor. The receptacle must not be controlled by a switch.
- The location of the power receptacle must be accessible so that the system can be unplugged during maintenance; however, you must ensure that the power cord does not present a hazard to users or pedestrians and that it is protected from accidental removal.

Note: To prevent accidental removal, attach a warning tag to the plug end of the cord.

Installing an MX Controller

The SX-200 ICP MX controller is shipped alone or in a package with the components listed in the table below. The system software and a default database are installed at the factory. Optional components (DSP, hard drive, etc.) are field-installed.

Note: None of the packages include power supplies for the IP phones; they must be ordered separately.

System Packages (MX)

The MC Controller is sold as a package that includes the components shown in the table below. None of the packages include power supplies for the phones; they must be ordered separately.

Table: MX Controller Packages

	Basic Business - Voice Only	Premier Business - Voice and Data	Basic Controller
LS/CLASS circuits (see Note 1)		6	
ONS circuits		2	
DNIC circuits		2	
Voice mail ports		4	
DSPs		1 Dual	

IP Phones	Seven 5207s One 5220	Four 5220s	None
PKMs	One 12-Button PKM		
Licenses (see Note 2) IP Phone Voice Mailbox TDM ACD Agent IP Channel	20 20 32 None None	8 8 32 5 2	
Software Options (see Note)	1 Digital Link Voice Mail Softkey	1 Digital Link Voice Mail Softkey 2nd Port on IP Phones Record A Call	

Notes:

1. High-voltage Message Waiting lamp is NOT supported on these circuits.
2. Licences and software options are listed on FOR Option sheet included with the system software package.

The following table contains the standard hardware components common to all SX-200 MX ICP system packages plus the optional components (not including phones or other peripherals) requiring on-site installation. Some of the equipment listed as requiring installation may have been shipped already installed. If that is the case, the equipment should be removed and reseated during the installation procedure.

Equipment Installed in the MX Controller (all packages)

Table: Equipment Installed in the MX Controller

Standard Components
SX-200 ICP MX controller
Includes:
Dual DSP MMC - CompactFlash card (internal and external) - Analog main board - Two onboard Copper Interface Module (CIM) ports to connect to ASUs
Optional Components

DSP Modules	Available in Dual or Quad MMC configurations for a maximum of eight DSP per system.
Dual T1/E1 Framer Module	Up to two modules can be installed, each of which has two T1 digital trunk ports that provide 24 B-channels for T1/D4.
Dual Fiber Interface Module	Up to two modules can be installed to provide connectivity to Peripheral Cabinets or NSUs.
Quad Copper Interface Module	Up to two modules can be installed to provide connectivity to Peripheral Cabinets or NSUs. (The two onboard CIM ports provide connectivity to external ASUs.)
Analog Option Board	Provides six additional LS/CLASS and two ONS/CLASS circuits.
Hard drive	

Replaces the standard 256 MB internal CompactFlash card to provide more storage capacity for voice mail messages and recorded calls.

NOTE: Only hard drives purchased from Mitel should be installed in the SX-200 ICP; drives obtained elsewhere are not supported.	
Stratum 3 Clock Module	Required for digital trunks
Network Services Unit (NSU)	Supports digital trunk protocols for ISDN PRI (NI2_STANDARD, NI2_5ESS, NI2_GTD5), and QSIG (QSIG_ISO), DMS 100/250, 4ESS
SX-200 ELx cabinet	

Includes:

- Bay Control Card (BCC II or BCC III) - Bay Power Supply - Peripheral FIM Carrier II (Peripheral cabinet only) - PRI card - Up to eight digital peripheral cards
SX-200 FD cabinet

Includes:

- Bay Control Card - Bay Power Supply - Peripheral FIM Carrier (Peripheral cabinet only) - up to eight digital peripheral cards	
Dual FIM Module	Provides connectivity to Peripheral Cabinets and/or an NSU

Configuration Rules

Each MX controller can be configured according to the following rules.

Cabinet Configuration Rules

The MX controller can be expanded to include:

- up to seven SX-200 Peripheral cabinets which provide 672 TDM ports for ONS, OPS, DID, T1, PRI/T1
- up to four Universal NSUs which provide eight PRI Links (192 PRI/T1 trunks)
- up to six ASUs or six ASU IIs which provide 288 ONS/CLASS circuits

FIM and CIM Assignments for Each Cabinet Type

Fiber interface module and copper interface module carrier cards are configured in the following cabinets. The CIM supports a distance of up to 30 meters or 100 feet between cabinets.

Fiber and Copper Interface Modules supported by SX-200	
Card	**Cabinet**
Peripheral FIM Carrier II	SX-200 RM peripheral cabinet
Peripheral FIM Carrier	SX-200 LIGHT peripheral cabinet
PRI Card (FIM II or CIM)	ELx peripheral cabinet
Bay Control Card III (FIM II or CIM)	(ELx peripheral cabinet)

Typical Installation Sequence

Follow this typical sequence when installing a system.

- Prepare for Installation
- Install the SX-200 ICP Controller and Layer 2 switch
  - (Optional) Install the system software and a database
  - Configure the controller
- (Optional) Install SX-200 Peripheral Cabinets
  - Unpack and Inspect Cabinet
  - Open and Close Front Door
  - Install Peripheral Cabinet (cabinets are pre-configured with Control cards)
  - Verify Ground Connection
  - Install Circuit Cards and Modules
- (Optional) Install NSU
- (Optional) Install ASU/ASU II
- (Optional) Install Wireless Devices
- Connect the Controller to the LAN
- Put System Into Service
  - Connect Optical Fiber Cables between Nodes
  - Connect Cables Between System and Cross-connect Field
  - Install the Flash Memory Card into the controller.
  - Connect power to the controller.
  - NSU, ASU/ASU II and Peripheral Cabinet Power On
  - Enable FOR Options.
  - Check System Initialization

Notes:

1. Installation of a hard drive is strongly advised for systems with more than eight voice mail ports or when Record a Call is frequently used. The drive must be purchased from Mitel (see Install FRUs for part number); drives obtained elsewhere are not supported.
2. Premier Business systems require a change from the factory-installed default database to the Premier database. The change should be made BEFORE enabling the FOR options included with the systems. For more information, see Installing an Alternate database.

To install the SX-200 ICP MX

1. If your system includes optional controller hardware, install according to the instructions indicated in the following table:

Component	**Installation Instructions**
DSP Modules	Installing a DSP Module
Analog Option Board	Installing the Analog Options Board
Dual FIM Module	Installing a Dual FIM Module
Stratum Clock Module	Installing the Stratum Clock Module
Quad CIM Module	Installing a Quad CIM Module
Dual T1/E1 Module	Installing a Dual T1/E1 Module
Hard drive	Installing the System Hard Drive

2. If your system includes expansion nodes, install them according to the instructions in the following table:

Node	**Installation Instructions**
NSU	Installing an NSU
ASU	Installing an ASU
ASU II	Installing an ASU II
SX-200 Peripheral Cabinets	Installing SX-200 Peripheral Cabinets

3. Wall mount the units, rack mount them, or place them on a desk or shelf. See Wall Mounting and Rack Mounting

   Note: The NSU and ASU are NOT wall-mountable.
4. Connect the ground stud on the rear panel of the controller to a hard-wired ground using 18 AWG (0.75mm 2/) gauge wire. The wire must have green or yellow insulation. Crimp the wire to the ground source.
5. Connect a PC to the maintenance port on the controller. See Serial Connection to the Controller.
6. Set up an FTP server (required for backups and software upgrades). See Setting up an FTP Server.
7. Do one of the following:
   - If you are NOT installing software and a database in the controller, power up the system. See System Initialization.
   - If you are installing software and database in the controller, see Installing the System Software on the External CompactFlash card.

Install Optional Hardware

Installing the System Hard Drive

The SX-200 ICP is shipped with an internal CompactFlash card which provides 256 MB of memory for system software and database storage. If you need increased storage capacity for voice mail messages and recorded calls, you can replace the card with a Mitel-supplied system hard drive.

IMPORTANT: Installation of a hard drive is strongly advised for systems that have more than eight voice mail ports or when Record a Call is frequently used.

Partitions on an internal hard drive should be organized as follows:

Partition Name	Partition Size	Capacity Used
/boot	1 GB	approximately 1.5 MB
/db	256 MB	approximately 2 MB
/vmail	13 GB	35 MB (see note above)
/spare	remainder of disk	none

The following procedure formats the new media, and then copies the contents of the internal CompactFlash to it.

Use this procedure to upgrade systems only. To replace faulty cards or drives, use the media replacement procedure.

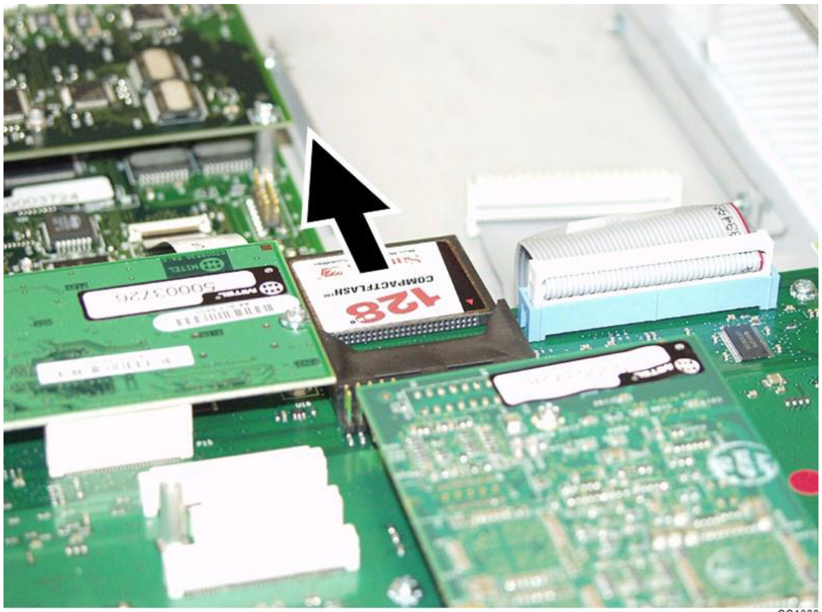
Notes:

1. The drive must be purchased from Mitel (see Install FRUs for part number); drives obtained elsewhere are not supported.
2. The hard drive must be configured to operate as a Master. See the manufacturer's documentation for the required jumper settings.

To install an optional System Hard Drive:

1. Establish a serial connection to the SX-200 ICP.
2. As a precaution, perform a full database backup. Skip this step if the system is new and has no database changes to preserve.
3. If an external CompactFlash card is inserted in the controller, remove it.

   Note: Removing the external CompactFlash card changes the bootline on the internal flash to boot from the front flash the next time the system is powered up.
4. Use the System > Restart > Shutdown command in Maintenance to stop the system.
5. When prompted on the PC, power down the controller.
6. Remove the cover from the controller.
7. Remove the internal CompactFlash card. Keep it on hand.


CC1033

8. Install the system hard drive as follows:
   a) Remove the drive from its packaging and set the jumpers on the drive to the Master setting.
   b) Insert the hard drive as shown in the figure below.
   c) Connect the power and IDE cables to the corresponding connectors on the hard drive and main board. (The cables are keyed for proper connection.)
   d) Secure the hard drive plate to the controller using the screws provided.

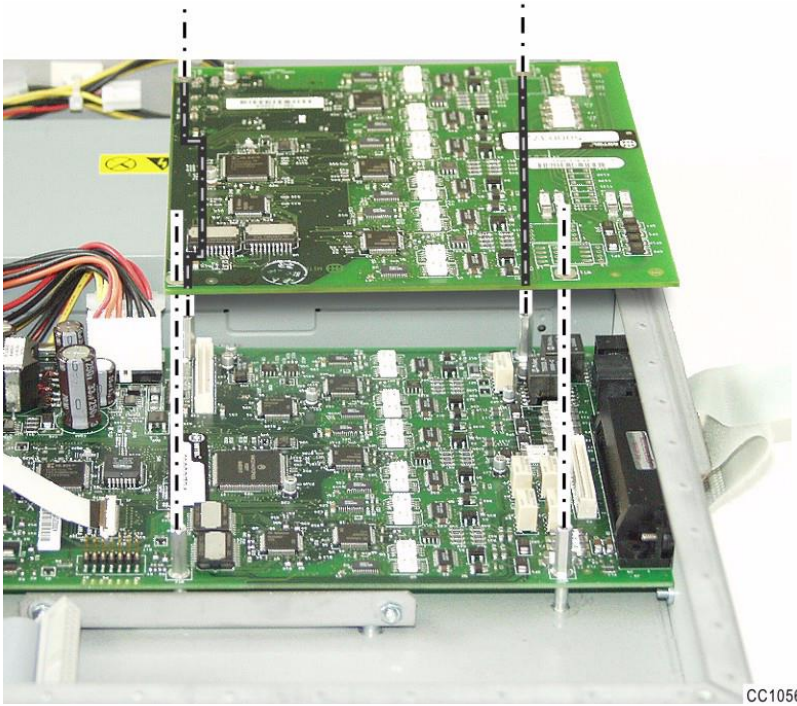

CC1035

9. Replace the cover.
10. Insert the internal CompactFlash card previously removed into the external card slot.
11. Reconnect the power cord to the controller.
12. Re-establish a serial connection to the controller and wait while the new media is formatted and the contents of the CompactFlash card copied to it.
13. When prompted, press return four times to log in to CDE/Maintenance.
14. Verify that the phones are working and that calls can be made.

15. Remove the CompactFlash card from the external slot.

Installing the Analog Option Board

1. Unplug the power cord from the controller.
2. Remove the cover of the controller.
3. Remove the Clock Module (if installed) by reversing the installation procedure.
4. Attach the standoffs as shown in the following figure.



CC1056

4. Insert the new Analog Option Board onto the Analog Main Board. Ensure it is well-seated in the socket.
5. Attach the screws.
6. Re-install the Clock Module (if it was removed).
7. Replace the cover.
8. Program circuits for phones and trunks in CDE.

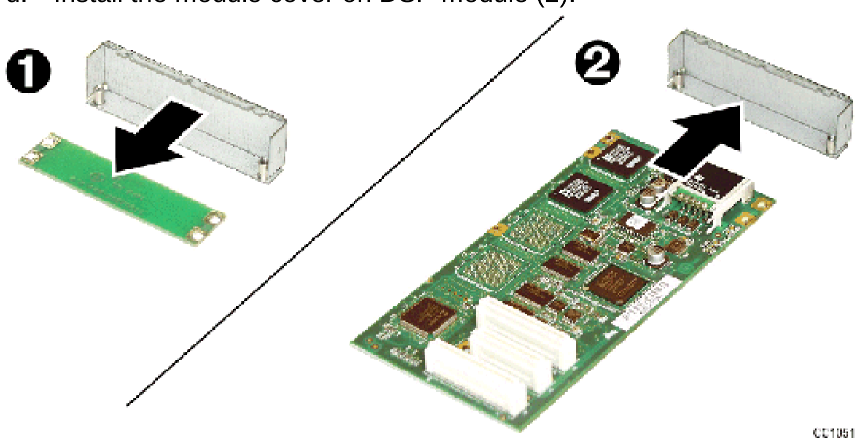
Installing a DSP Module

The basic SX-200 ICP MX controller is shipped with one Dual DSP installed Module Slot 3. The system can be upgraded to include modules as shown in the following figure.

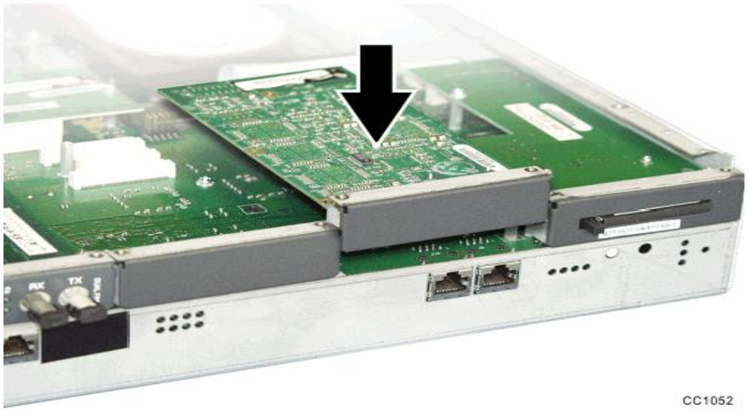
See DSP Configuration Options for information on determining DSP requirements.

To install the optional DSP Modules:

1. Unplug the power cord from the controller.
2. Remove the cover of the controller.
3. Repeat the steps below for each DSP module you install:
   a. Remove the DSP module from its packaging.
   b. Remove the blank module cover at the front of the controller, and insert the DSP module in the appropriate slot.
   c. Remove the small PCB (1).
   d. Install the module cover on DSP module (2).



   e. Insert the DSP module in the appropriate slot.

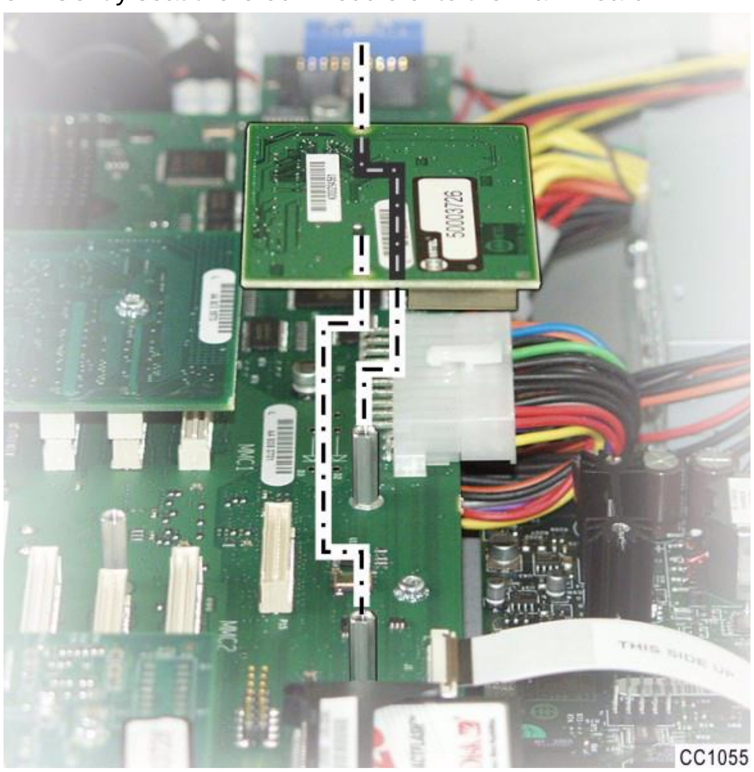


f. Secure the DSP module to the controller using the screws provided with the module.

4. Replace the cover.

Installing the Stratum Clock Module

1. Unplug the power cord from the controller.
2. Remove the cover of the controller.
3. Gently seat the Clock Module onto the Main Board.



4. Attach the screws.
5. Replace the cover.
6. To check whether the system recognizes the Stratum clock card, log into Maintenance and use the System > Show Identity command. It should show the clock as ST3.

Installing a Dual FIM Module

The Dual FIM Module provides connectivity to a Peripheral Cabinet and/or to an NSU. The MX can support up to two Dual FIMs installed in MMC slots 1 and 2.

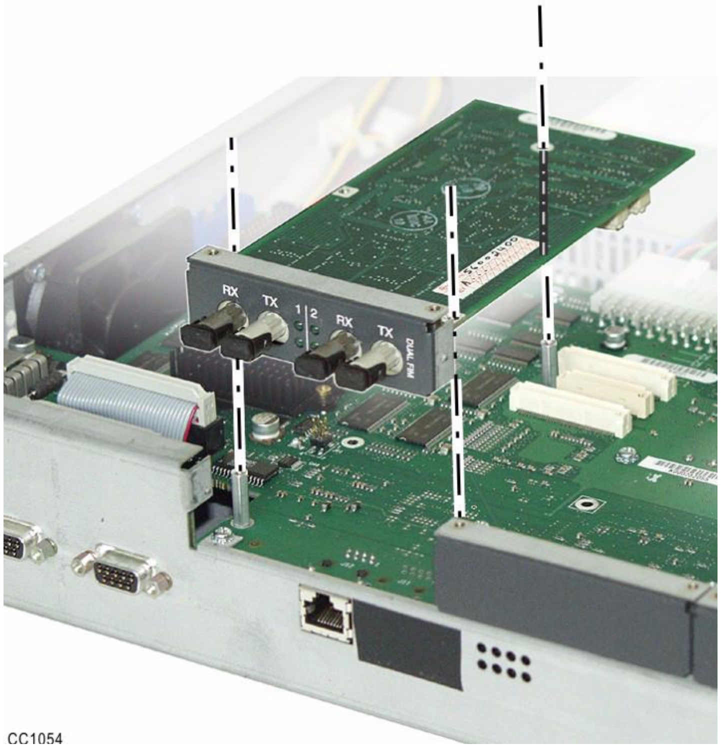
There are three fiber length variants of the FIM Module: 1, 5, or 14 km. Both ends must use the same variant.

Notes:

1. The NSU supports the 1 km variant only.
2. The SX-200 ICP does not support single FIM modules.

To install the FIM II Module:

1. Unplug the power cord from the controller.
2. Remove the cover of the controller.
3. Insert the Dual FIM Module into Module slot 1 or 2 on the Main Board connector.



3. Attach the screws.
4. Replace the cover.
5. Assign bay numbers to the module in Form 53, Bay Location..

Installing a Quad CIM Module

The optional Quad CIM module has four ports that provide connectivity to Peripheral Cabinets and/or NSUs using Category 5 UTP copper cabling. Up to two Quad CIM Modules can be installed in Module slots 1 and 2 of the MX.

To install the Quad CIM Module:

1. Unplug the power cord from the controller.
2. Remove the cover of the controller.

3. Insert the Quad CIM Module into Module slot 1 or 2 on the Main Board connector.

4. Attach the screws.
5. Replace the cover.
6. Assign bay numbers to the module in Form 53, Bay Location.

NOTE: If an ASU II is configured on a Quad CIM in the MX platform, it must be configured in port 4 first and then in any of the other connectors.

Installing a Dual T1/E1 Framer Module

The Dual T1/E1 Framer module has two digital trunk ports, each of which can be programmed to support either T1/D4 or PRI. Up to two modules can be installed in MMC slots 1 and 2 of the MX controller.

To install a Dual T1/E1 Framer Module:

1. Unplug the power cord from the controller.
2. Remove the cover of the controller.
3. Insert the Dual T1/E1 Framer into Module slot 1 or 2 on the Main Board connector.

4. Attach the screws.
5. Replace the cover.
6. Program the module in CDE:
   - Assign the module a bay number in Form 53, Bay Location.
   - Program the T1 link; see Trunk Support - T1.

Mounting the MX Controller

The MX is wall and rack mountable.

Wall Mount

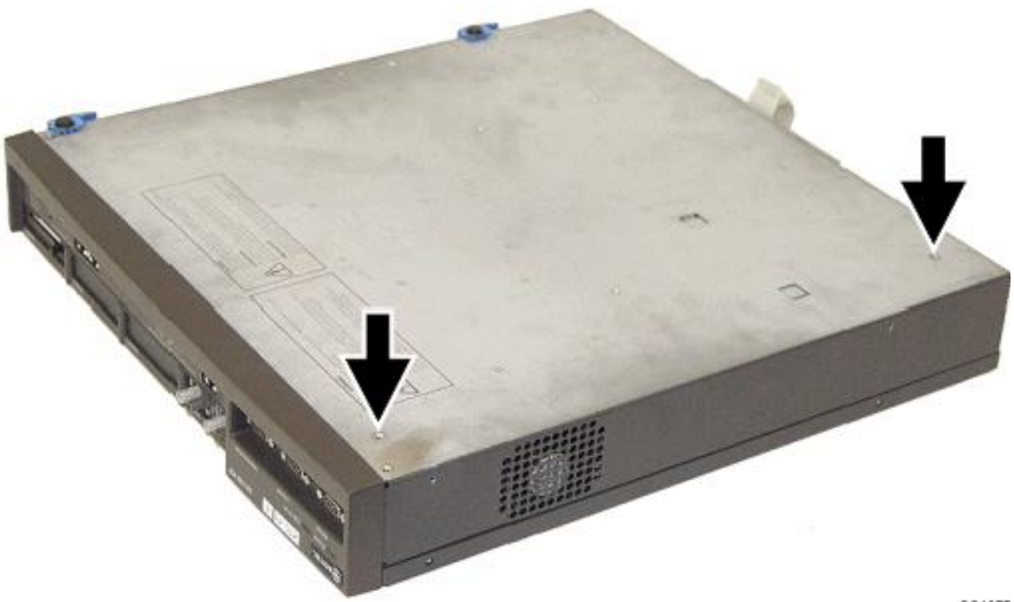
Use the supplied brackets and the screws (6-32 X 3/16) to wall mount.

CAUTION: **Make sure the wall material is capable of supporting the weight of the unit. It is recommended that you mount the supplied bracket onto a wall that has wooden studs with 16" centers. Mitel is not responsible for units damaged as a result of improper wall mounting.**

1. Turn the Controller upside down.
2. Locate the two holes on the bottom of the Controller that are shown in the following figure.

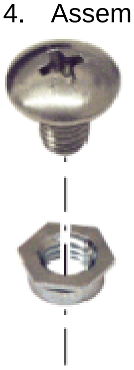
CC1078

3. Remove the two feet as shown below.



CC1077

4. Assemble the two supplied screws and two nuts as shown below.

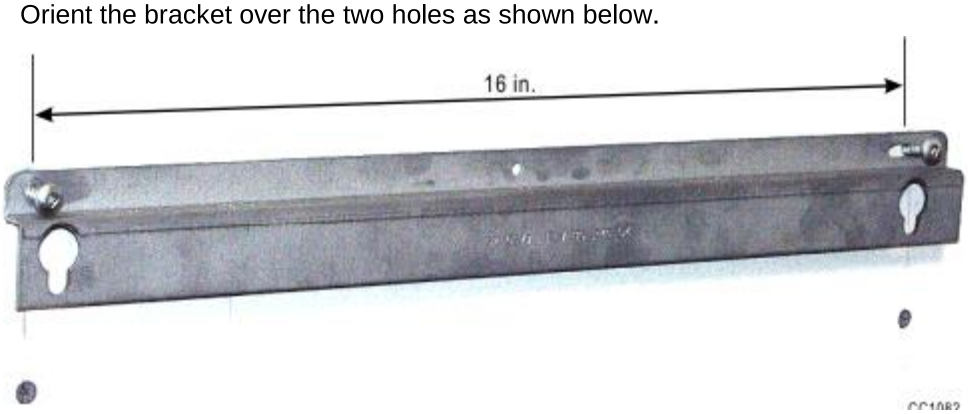


CC107!

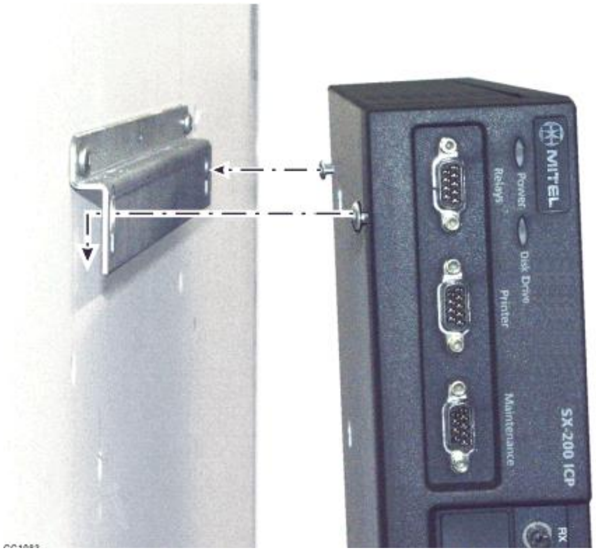
5. Screw the assembled nuts and screws into the holes as shown below.

6. Screw the supplied small bracket onto the bottom of the Controller as shown.

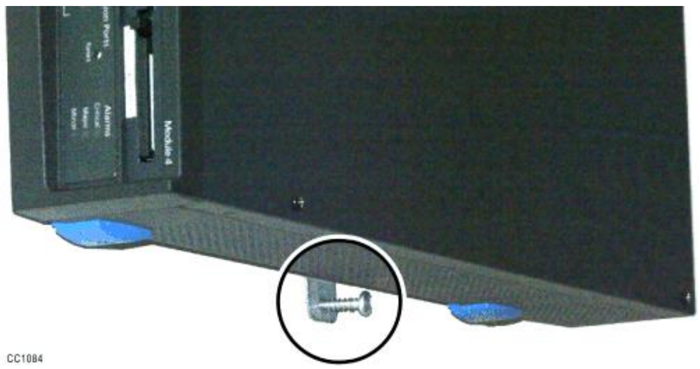
7. Mount the bracket onto the wall.
   a. Pre-drill two pilot holes into two wall studs with 16" centers.
   b Orient the bracket over the two holes as shown below.



   c. Insert a screw into the hole on the left side of the bracket.
   d. Insert a screw into the hole on the slot on the right side of the bracket.
8. Hang the Controller onto the mounted bracket as shown below.



9. Insert a screw into the bottom bracket to stabilize the Controller as shown below.



Rack Mount

Use the Mitel Rack Mount Kit (Part Number: 50004150) to rack mount the MX.

CAUTION: When installing the system in an enclosed rack, you MUST provide adequate ventilation to ensure that the maximum ambient temperature inside the rack does not exceed 40°C/104°F.

CAUTION: Ensure that a hazardous condition does not result from any uneven mechanical loading.

CAUTION: When using the system in a rack, you should consider the connection of the equipment to the power supply circuit and the effect that overloading of circuits might have on overcurrent protection and supply wiring. When addressing this concern, refer to the system's ratings label.

1. Attach the brackets to the rack.
2. Slide the unit into the brackets.
3. Secure the unit to the brackets using the supplied thumbscrews. The screws fasten to the underside of the unit and fit into the notch on the bracket.

Feeding Power to the Phones

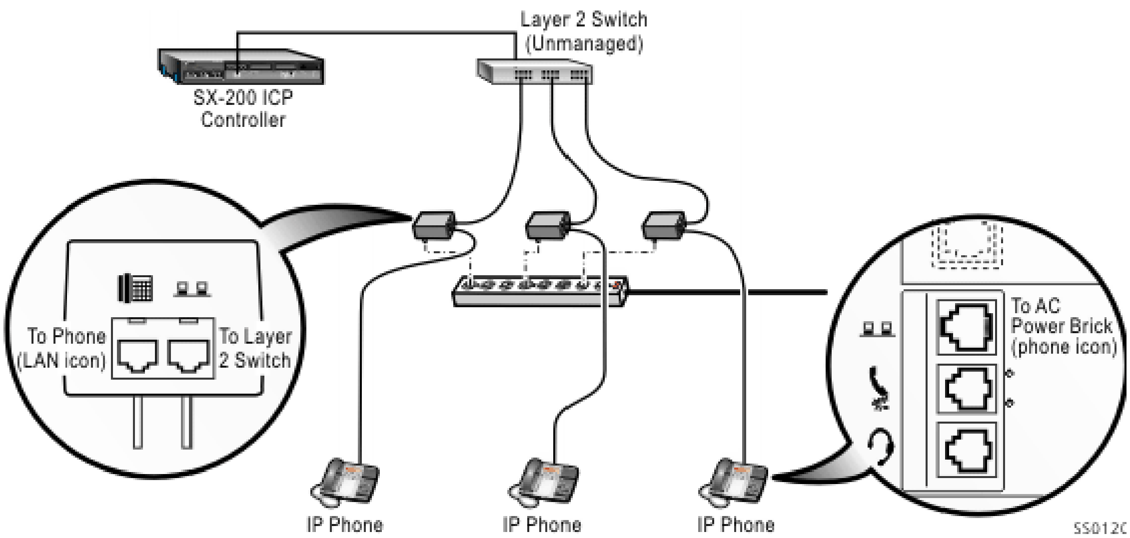
The IP Phones require power. This can be provided by

- an external supply such as a 24-volt adapter or 48-volt power brick

  The 5000 series IP Phones require a 24-Volt adapter that plugs into the back of the phone.

  The 5200 and 5300 series IP Phones require a 48-Volt power brick. The power brick connects to the Layer 2 switch and the IP Phones via Ethernet CAT5 cables as shown in the following illustration.

Note: Some 5200-series IP Phones (not the 5207 or 5215) can also be powered by a 24-Volt adapter.



- an in-line power unit (such as the PowerDsine 24PT In-line Power Unit)
- a Layer 2 switch with integral power feed. (For information on planning a Power-over-Ethernet (PoE) installation, see the Engineering Guidelines document on the Mitel Edocs website at http://edocs.mitel.com.)

The 24-volt adapter, power brick and PowerDsine unit are available for purchase from Mitel. See Ordering Information for part numbers.

Power backup to the IP Phones, the SX-200 ICP, and the Ethernet switches is required for the phones to work during a power failure.

CAUTION: Ensure that the powered cable from the inline power adapter is installed in the proper connector on the IP Phone. DO NOT plug it in to the connector that may be on the IP Phone that is designed for a PC or other Ethernet devices (Layer 2 port).

Install an Ethernet Switch

You must connect all IP devices to a Layer 2 Ethernet Switch. The type of Ethernet Switch required depends on the number of IP Phones you need to install.

Powered switches can be used. (For information on planning a Power-over-Ethernet (PoE) installation, see the Engineering Guidelines document on the Mitel Edocs website at http://edocs.mitel.com.)

Hubs should not be used. They do not provide the bandwidth or performance of an Ethernet Switch. Hubs are also prone to collisions, which can increase the likelihood of poor voice quality.

When selecting the size of Ethernet Switch, consider if growth is needed. Also consider whether or not data will be needed in the future and what level of maintenance is required.

Special cables may be required, and therefore the physical location of the switches needs to be considered. For example, do the phones connect directly to the switches or is there a central patch panel that uses structured wiring? For larger installations, central patch panels make maintenance easier; however it requires additional patch panels and cables.

Ethernet Switch Specifications

The Ethernet Switch specifications below are based on the following assumptions:

- SX-200 ICP limited size, typically less than 60 users
- Voice ONLY network (any data is on a completely separate network)
- LAN environment only

Specifications:

For best performance, select an Ethernet switch that has the following properties:

- Type: Should be a Switch such as a Layer 2. Hubs are NOT recommended
- Number of ports. Five port and eight port devices are readily available commercially from a number of vendors. They are suited for smaller installations. Several switches can be connected together to provide more ports. Larger units that have 24 or more ports are also available. They have more features, but are also more expensive.
- Speed: 10/100BaseT with autosensing ports
- One or more ports with a cross-over is desirable, but not essential, reducing the need for cross-over cables
- Autosensing of cross-over is desirable, but not essential. It removes the need for cross-over cables
- The switch used at the Distribution layer should be able to handle at least 4000 Packets per Second, based on a 60-user system, and normal business use. Use with ACD will require higher packet rates, typically above 6000 PPS.
- Indication of active link is desirable (know if device is working and cable is correct)
- Indication of link speed is desirable (is it 10BaseT or 100BaseT)
- The ports should employ CAT5 or better connections.

To install an Ethernet Switch in a small installation:

1. Connect the Ethernet Switch to the SX-200 ICP Controller Ethernet Port with an Ethernet cable.
2. Connect the Ethernet Switch power cord to a power source.

To install an Ethernet Switch in a larger installation:

- If you are connecting several switches together, connect them in a tree-type structure. Daisy-chaining a number of switches together is not recommended because all switches become involved in connections from one end of the chain to another. Layering reduces this unnecessary traffic.

Connect ONS/CLASS, DNIC, and LS/CLASS circuits

After connecting the LS trunks, use the LS Measurement tool in Maintenance to determine the Circuit Descriptor settings to program for optimum audio quality.

1. Connect a breakout box to the 25-pin Amphenol connector at the back of the SX-200 ICP Controller.

Note: ONS/CLASS lines do not support high-voltage message-waiting lamps.

2. Assign the ONS and DNIC phones to the breakout box lines according to the Analog Main Board / Analog Option Board Tip & Ring Assignments.

Installing a CX or CXi Controller

The SX-200 ICP CX/CXi controller is shipped alone or in a package with the components listed in the table below. The system software and a default database are installed at the factory. Optional components (DSP, system hard drive, APC, APC hard drive, etc.) are field-installed.

CX and CXi Installation Procedures

The following installation procedures are common to the CX, CXi, and MX controllers:

- Installing the System Software on the External CompactFlash Card
- Program the Customer Data Entry Forms
- Installing IP Phones

All other installation procedures are unique to the CX/CXi.

System Package

The CX/CXi controller is sold as a package that includes the components shown in the table below.

Table: CX/CXi Controller System Package

LS/CLASS circuits	6
ONS circuits	4
Voice mail ports	4
DSPs	2 DSPs on main board
IP Phones	Four 5207
Licenses (see Note) IP Phone Voice Mailbox	 10 10
Note: Licences are listed on Feature Options Record sheet included with the system software package.	

Equipment Installed in the CX and CXi Controllers

Table: Equipment Installed in the CX and CXi Controller Systems

Standard Components
SX-200 ICP CX and CXi controller Include: 2 DSPs on digital main board - CompactFlash card (internal and external) - analog main board (AMB) with 6 LS and 4 ONS
Optional Components

DSP Modules	Available in Dual DSP and T1/E1 Combo MMC configurations for a maximum of five DSP resources per system (including 2 DSPs on digital main board).
Analog Option Board	Provides an additional six LS/CLASS and four ONS/CLASS circuits.
Stratum 3 Clock Module	Required for digital trunks
System Hard drive	

Replaces the standard 256/512 MB internal CompactFlash card to provide more storage capacity for voice mail messages and recorded calls.

NOTE: Only hard drives purchased from Mitel should be installed in the SX-200 ICP; drives obtained elsewhere are not supported.	
Quad Copper Interface Module	Up to two modules can be installed to provide connectivity to Peripheral Cabinets or NSUs. (The two onboard CIM ports provide connectivity to external ASUs.)
Application Processor Card	The APC is a PC on a compact card that holds the Mitel Managed Application Server.
APC Hard Drive	A separate hard drive is required to support the APC.
T1/E1 Combo MMC	Combines T1 trunking and DSP functionality in a single module. Also includes a stratum 4 clock.

Configuration Rules

Each CX/CXi system can be configured according to the following rules.

The SX-200 ICP CX/CXi controller can be expanded to include:

- up to 8 ONS/CLASS circuits
- up to 12 LS/CLASS circuits
- a single T1 digital trunk port which provides 24 B-channels for T1/D4
- up to 16 voice mail ports
- up to 100 IP phones.

Typical Installation Sequence

Follow this typical sequence when installing a system.

- Prepare for Installation
- Install the SX-200 ICP Controller
  - Install optional controller hardware
  - Install software and database (optional)
  - Enable FOR Options.
  - Configure the controller
  - Connect phones and lines

- Integrate the controller into the IP network
    - LAN interface: program IP addresses, DHCP server, and internal Layer 2 switch
    - WAN interface (if used): program IP address and firewall settings
    - Expansion Layer 2 switch (if used): connect to SX-200 ICP Controller and program
- Put system into service
    - Connect power to the controller.

To install the SX-200 ICP CX/CXi controller

1. If your system includes optional controller hardware, install according to the instructions indicated in the following table:

Component	Installation Instructions
System Hard drive	Installing the System Hard Drive
APC Hard Drive	Installing the APC Hard Drive
DSP Modules	Installing a DSP Module
Analog Option Board	Installing the Analog Option Board
Application Processor Card	Installing the APC Card
Stratum Clock Module	Installing the Stratum Clock Module
Quad CIM MMC	Installing the Quad Copper Interface Module
T1/E1 Combo Module	Installing the T1/E1 Combo Module

4. Wall mount the units, rack mount them, or place them on a desk or shelf. See Wall Mounting and Rack Mounting
5. Connect the ground stud on the rear panel of the controller to a hard-wired ground using 18 AWG (0.75mm 2/) gauge wire. The wire must have green or yellow insulation. Crimp the wire to the ground source.
6. Connect a PC to the maintenance port on the controller. See Serial Connection to the Controller.
7. Set up an FTP server (required for backups and software upgrades). See Setting up an FTP Server.
8. If you are NOT installing software or optional hardware in the controller, power up the system. See System Initialization.

Install Optional Hardware

Installing the System Hard Drive

The CX and CXi are shipped with an internal CompactFlash card which provides 256 MB of memory for database storage. If you need increased storage capacity for voice mail messages and recorded calls, you can replace the card with a Mitel-supplied system hard drive.

IMPORTANT: Installation of a hard drive is strongly advised for systems that have more than eight voice mail ports or when Record a Call is frequently used.

Partitions on an internal hard drive should be organized as follows:

Partition Name	Partition Size	Capacity Used
/boot	1 GB	approximately 1.5 MB

/db	256 MB	approximately 2 MB
/vmail	13 GB	35 MB (see note above)
/spare	remainder of disk	none

The following procedure formats the new media, and then copies the contents of the internal CompactFlash to it.

Use this procedure to upgrade systems only. To replace faulty cards or drives, use the media replacement procedure.

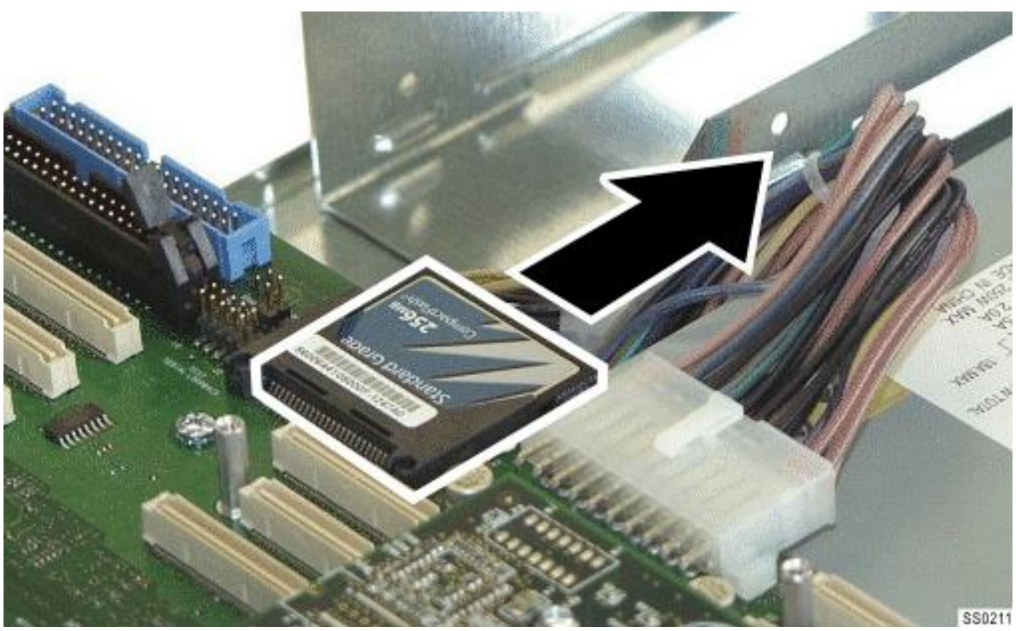
Notes:

1. The hard drive must be purchased from Mitel (see Install FRUs for part number); drives obtained elsewhere are not supported.
2. The hard drive must be configured to operate as a Master. See the manufacturer's documentation for the required jumper settings.

To install an optional System Hard Drive:

1. Establish a serial connection to the SX-200 ICP.
2. As a precaution, perform a full database backup. Skip this step if the system is new and has no database changes to preserve.
3. If an external CompactFlash card is inserted in the controller, remove it.

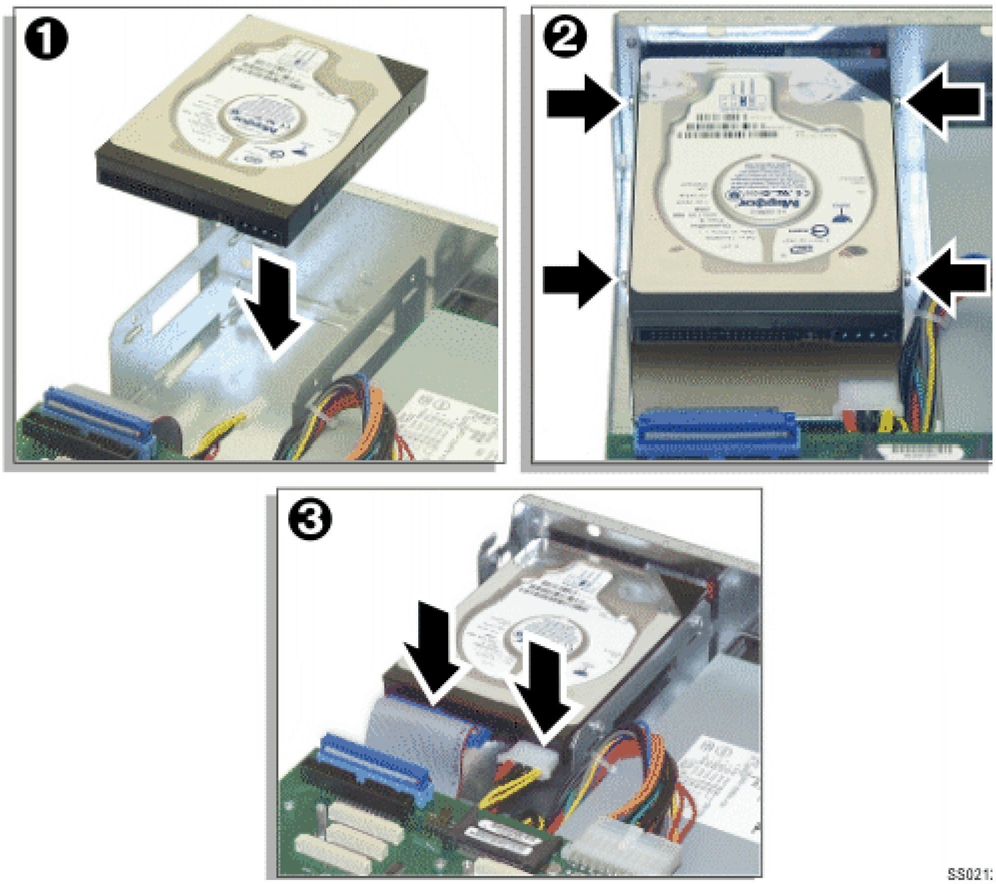
   Note: Removing the external CompactFlash card changes the bootline on the internal flash to boot from the front flash the next time the system is powered up.
4. Use the System > Restart > Shutdown command in Maintenance to stop the system.
5. When prompted on the PC, power down the controller.
6. Remove the cover from the controller.
7. Remove the internal CompactFlash card. Keep it on hand.



8. To install the system hard drive:

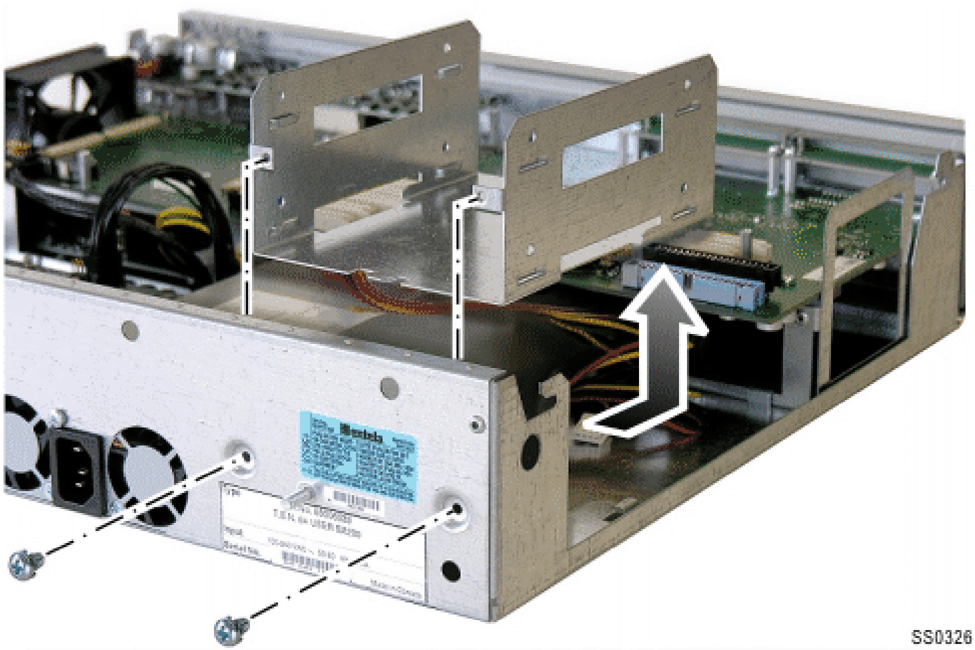
- Set the jumpers on the drive to the Master setting.
- Lower the drive onto the bracket (1).
- Secure the hard drive to the sides of the bracket with the screws provided (2).
- Connect the power and IDE cables to the corresponding connectors on the hard drive and main board (3). The cables are keyed for proper connection.

Note: Connect the system hard drive IDE cable to the main board connector labeled MPC8270 HARDDRIVE.



SS021:

11. Replace the controller cover.
12. Insert the internal CompactFlash card previously removed into the external card slot.
13. Restore power to the controller.
14. Re-establish a serial connection to the controller and wait while the new media is formatted and the contents of the CompactFlash card copied to it.
15. When prompted, press return four times to log in to CDE/Maintenance.
16. Verify that the phones are working and that calls can be made.
17. Remove the CompactFlash card from the external slot.



Installing the Analog Option Board (AOB)

Note: If installing version 2 of the AMB/AOB, upgrade the controller software to release 3.0 or greater.

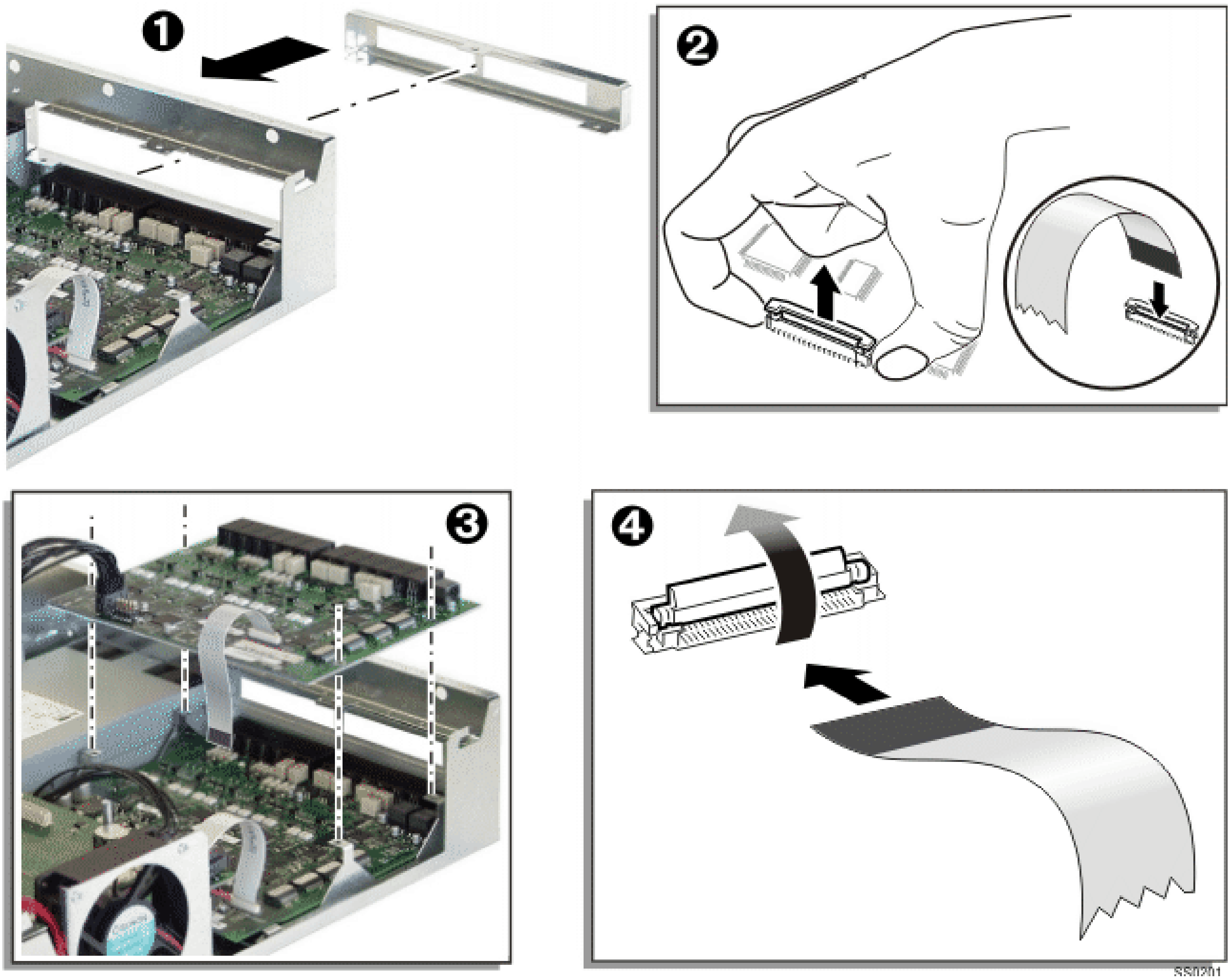
AMB/AOB compatibility is as follows:

AMB Version	must use AOB version:
1	1
2	2
3	2

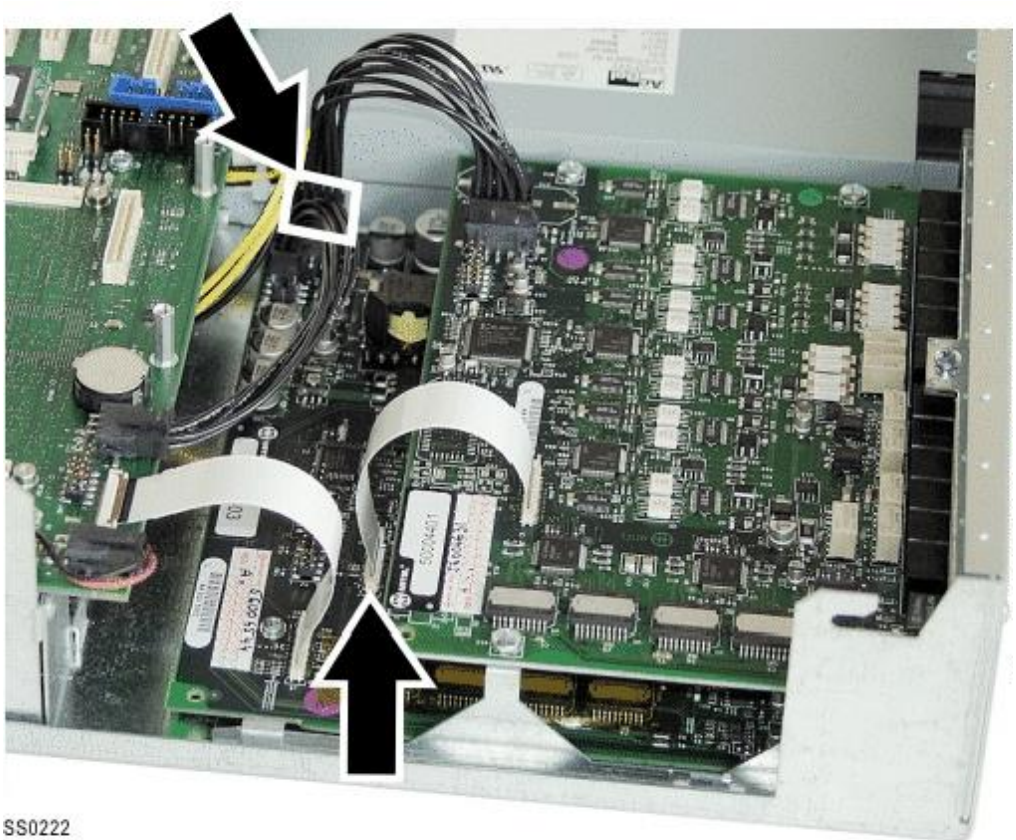
The AOB version 2 kit includes a new power cable.

To install the Analog Option Board:

1. Unplug the power cord from the controller.
2. Remove the controller cover.
3. Remove the blanking panel from the back of the controller.
4. Insert the replacement faceplate into the back of the controller and attach it with screws (1).



5. Place the AOB on a flat surface.
6. Attach the flex and power cables. To attach the flex cable, lift up on the tabs at the end of the connector to loosen it, insert the cable observing that the cable is facing in the direction indicated on the label, and then press down on the tabs to tighten connector (2).
7. Lower the AOB into place and attach it with the supplied screws (3).
8. Connect the other ends of the flex cable and power cable where indicated in the following figure. The horizontal flex cable connector is hinged: flip up to loosen it, insert the cable, and then press down to tighten (4).



SS0222

4. Replace the cover.
5. Reconnect the power cord to the controller.

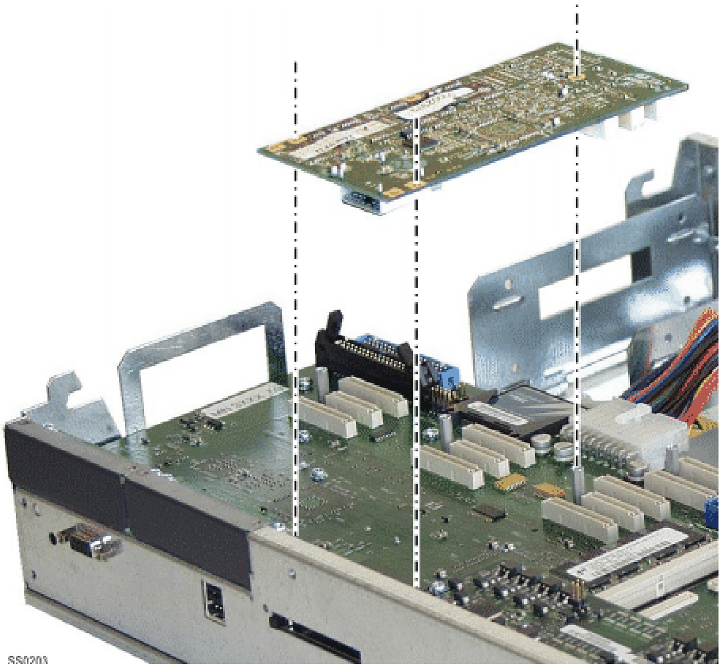
Installing a DSP Module

The CX and CXi can be upgraded by installing a Dual or Quad DSP module in MMC slot 3.

See DSP Configuration Options for information on determining DSP requirements.

To install the Dual DSP Module:

1. Unplug the power cord from the controller.
2. Remove the controller cover.
3. Insert the Dual DSP module in MMC slot 3 on the Main Board connector.

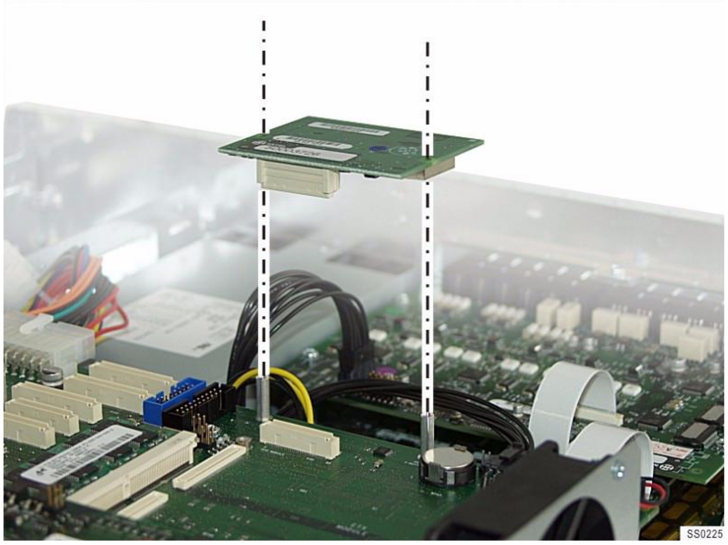


4. Secure the Dual DSP module to the controller using the screws provided.
5. Replace the cover.
6. Reconnect the power cord to the controller.

Installing the Stratum Clock Module

To install the Stratum Clock Module:

1. Unplug the power cord from the controller.
2. Remove the cover of the controller.
3. Gently seat the Clock Module onto the Main Board.



4. Attach the screws.
5. Replace the cover.

6. Reconnect the power cord to the controller.
7. To check whether the system recognizes the Stratum clock card, log into Maintenance and use the System > Show Identity command. It should show the clock as ST3.

Installing the T1/E1 Combo Module

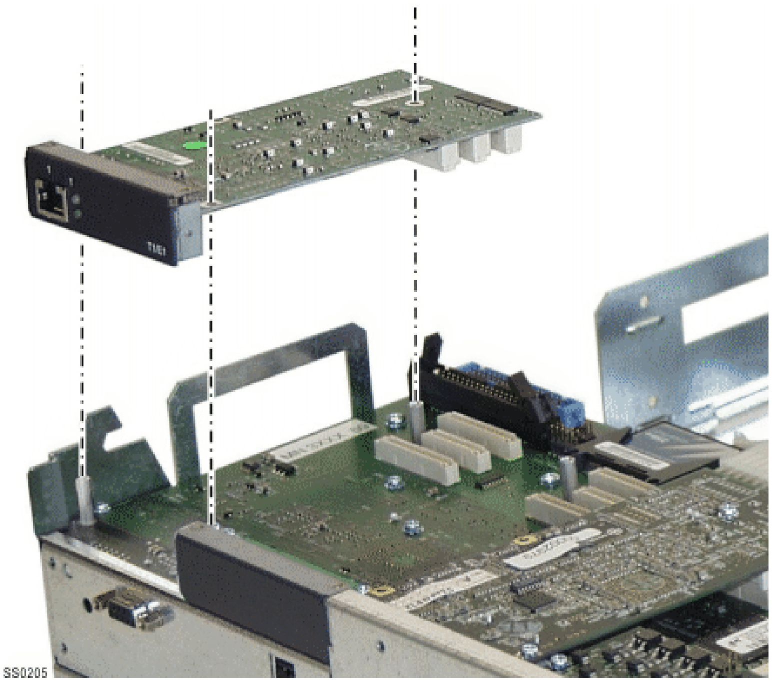
The T1/E1 Combo MMC combines trunking and DSP functionality in a single card. The digital trunk port can be configured as a T1 interface that provides 24 B-channels, or as a PRI interface that provides 23 B-channels. The module also includes a Stratum 4 clock. The CX/CXi supports a single T1/E1 Combo module installed in MMC slot 1 or 2.

Note: Newer T1/E1 Combo cards (called T1/E1 Combo II) have two RJ-45 ports. The second port is not functional for the SX-200 ICP CX controller.

See DSP Configuration Options for information on determining DSP requirements.

To install a T1/E1 Combo Module:

1. Unplug the power cord from the controller.
2. Remove the cover.
3. Insert the T1/E1 Combo module into MMC slot 1 or 2 on the Main Board connector.



4. Secure the T1/E1 Combo module to the controller using the screws provided.
5. Replace the cover.
6. Reconnect the power cord to the controller.
7. Program the module in CDE:
   - Assign the module a bay number in Form 53, Bay Location.
   - Program the T1 trunk card in Form 01, System Configuration.
   - Program the T1 link; see Trunk Support - T1.

Installing a Quad CIM Module

The optional Quad CIM module has four ports that provide connectivity to ASUs using Category 5 UTP copper cabling. Up to two Quad CIM Modules can be installed in Module slots 1 and 2 of the controller.

To install the Quad CIM Module:

1. Unplug the power cord from the controller.
2. Remove the cover of the controller.
3. Insert the Quad CIM Module into Module slot 1 or 2 on the Main Board connector.

SS0223

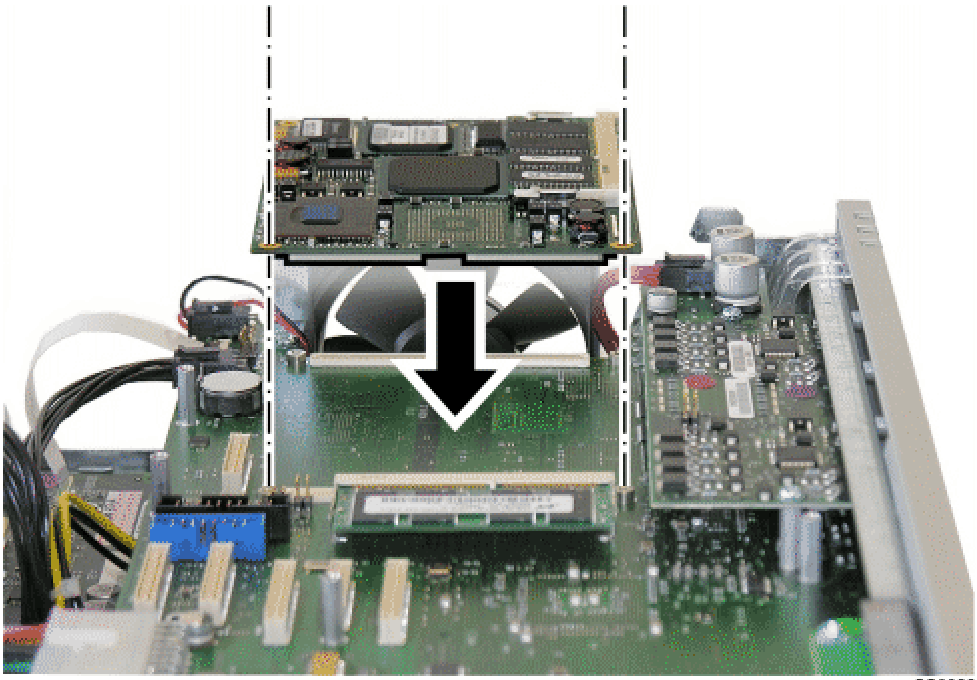
4. Attach the screws.
5. Replace the cover.
6. Assign bay numbers to the module in Form 53, Bay Location.

Installing the APC

The optional Application Processor Card (APC) allows the system to host the Mitel Managed Application Server (MAS).

To install the APC:

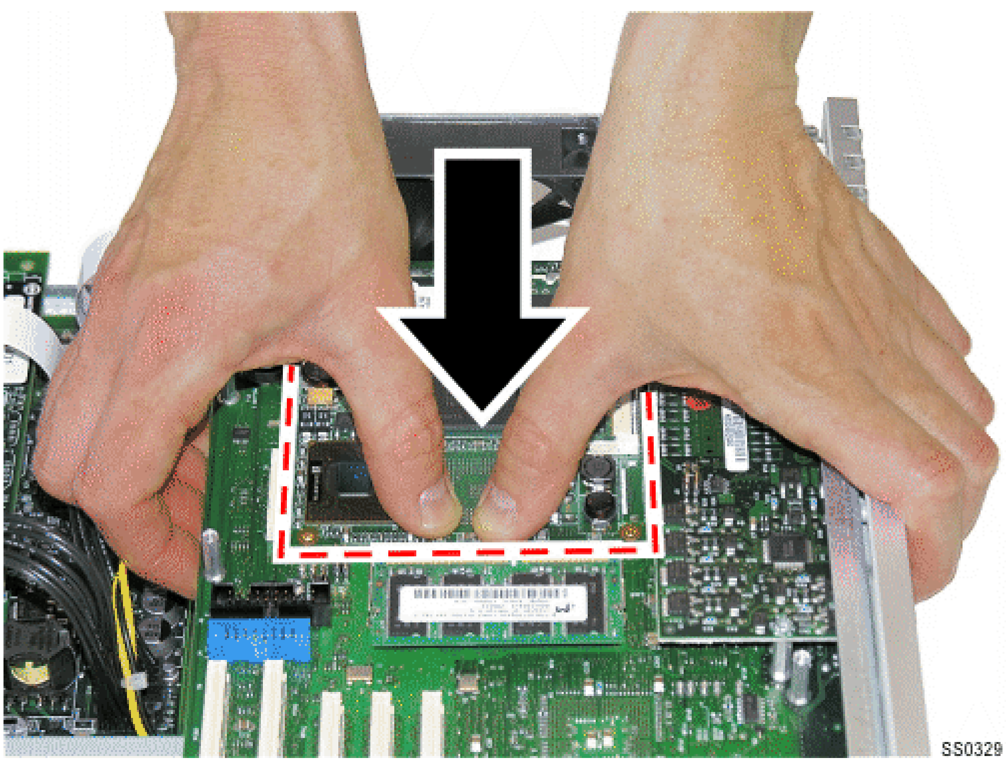
1. Unplug the power cord from the controller.
2. Remove the cover from the controller.
3. If a Stratum Clock module is installed, remove it.
4. Place the APC on the main board connectors. The connectors are spaced irregularly to assist in alignment.



SS0328

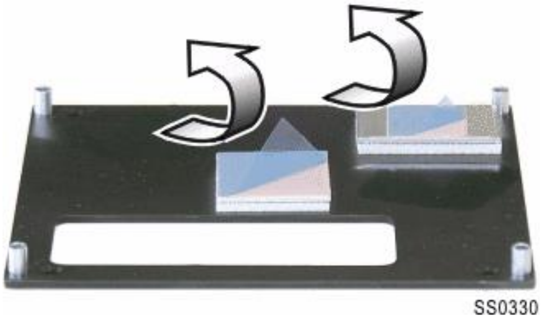
5. To seat the connectors, grasp the main board with your fingertips and press down firmly on the middle of one side of the APC with your thumbs as shown. Repeat for the other side of the APC next to the fan. You should hear and feel the connectors seating themselves.

Note: To ensure that the APC is seated securely, press down on the APC over each of the four connectors, one at a time. Once the APC is properly installed, the four corners of the APC will rest against the standoffs located at each corner of the APC.

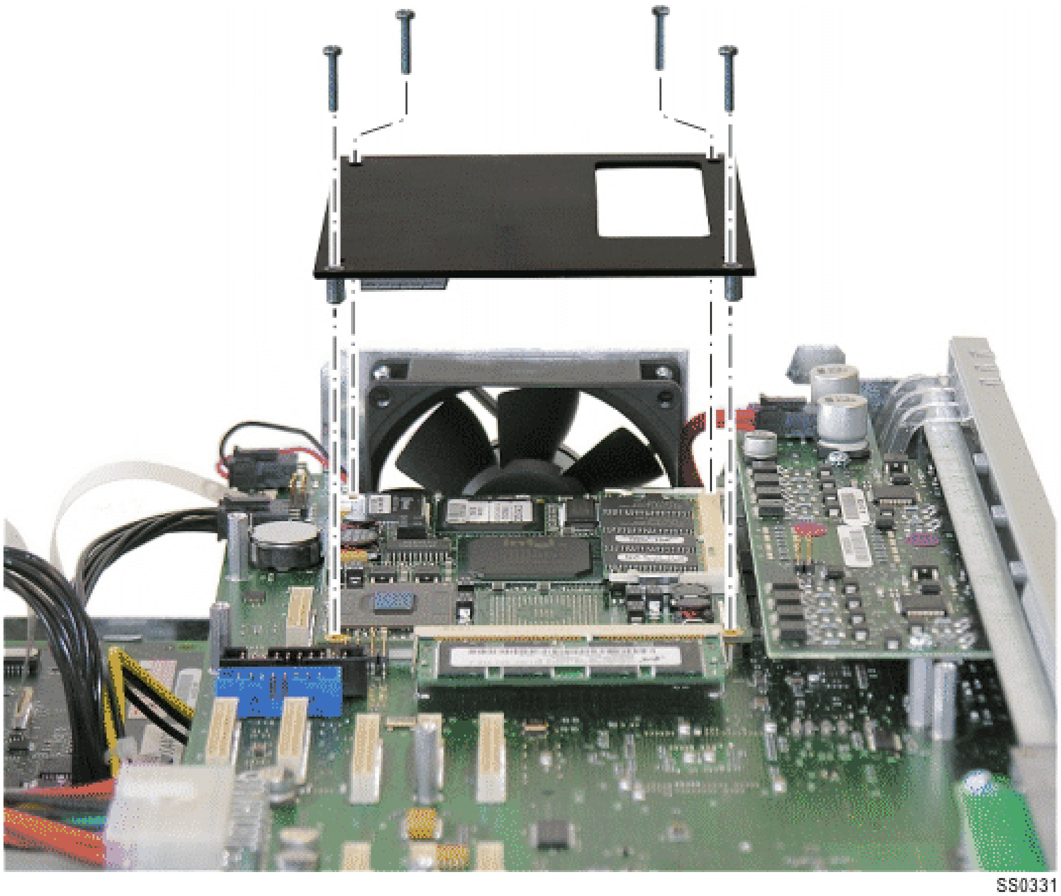


SS0329

6. Prepare the heat spreader for installation by removing the protective strips from the adhesive heat pads.



7. Place the four screws (supplied) in the heat spreader, and lower the heat spreader onto the APC. Align the cutout on the heat spreader above the memory modules on the APC.



8. Tighten the screws in an alternating pattern until they are snug. Do not over-tighten.
9. Replace the Stratum Clock module (if removed).
10. Install the APC Hard Drive.
11. Replace the controller cover.
12. Reconnect the power cord to the controller.
13. Enable system option 83 (Internet Gateway) in Form 04, System Options/System Timers.

Note: To install MAS software on the APC, see the SX-200 ICP Technician's Handbook or the MAS Installation and Administration Guide.

Installing the APC Hard Drive

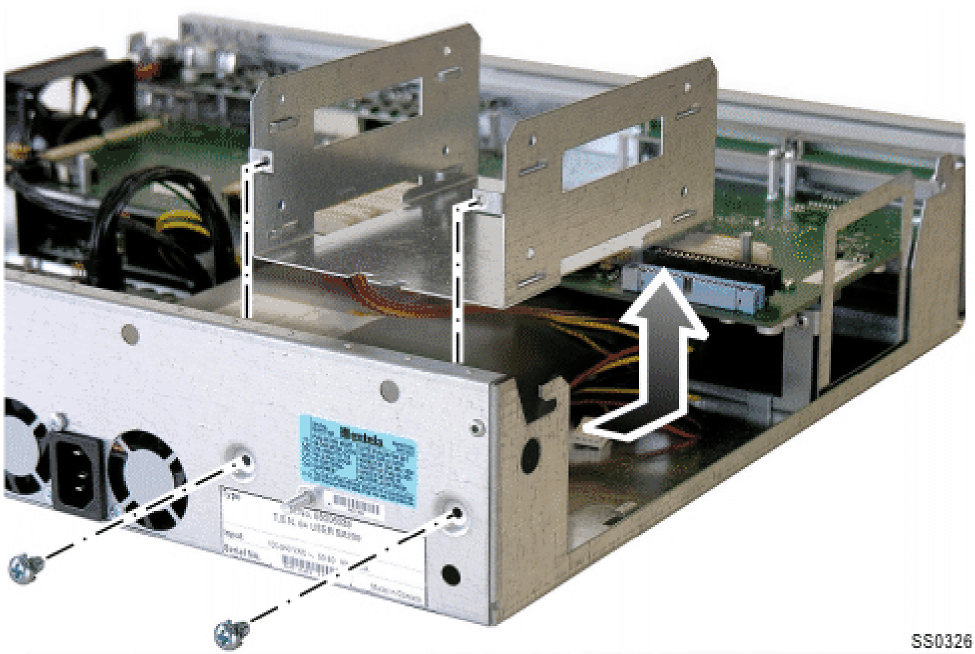
If you are installing the optional Application Processor Card, then you must also install a hard drive to support the APC's operating system and applications.

Notes:

1. The hard drive is supplied with the APC (see Install FRUs for part number); drives obtained elsewhere are not supported.
2. The hard drive must be configured to operate as a Master. See the manufacturer's documentation for the required jumper settings.

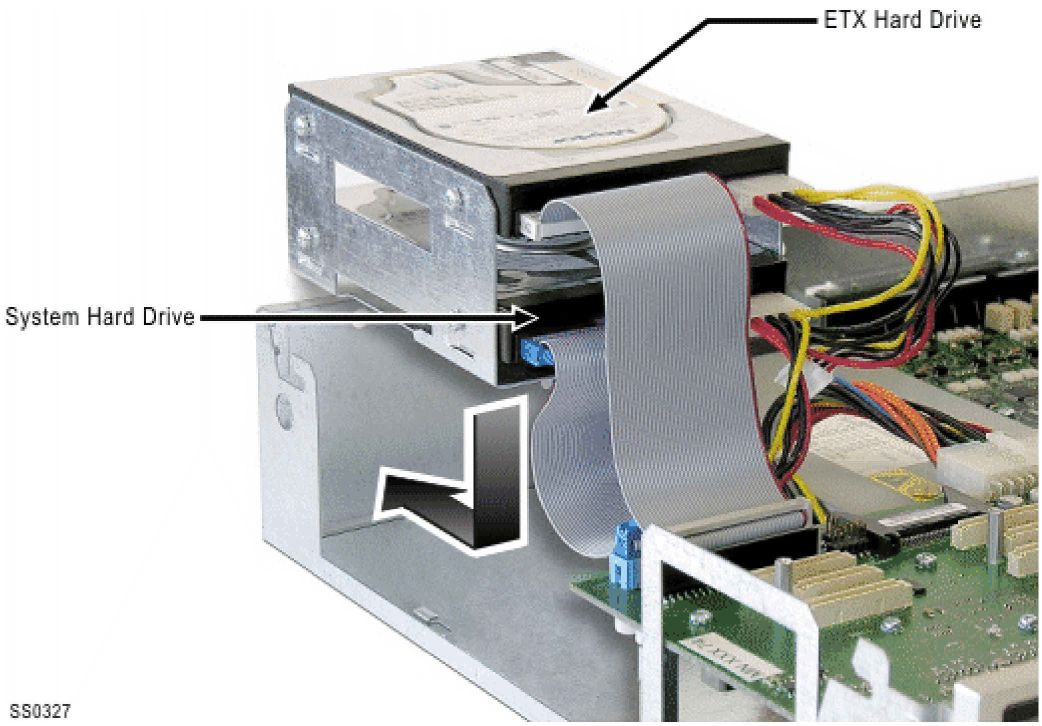
To install the APC hard drive:

1. Unplug the power cord from the controller.
2. Remove the cover from the controller.
3. Remove the screws connecting the bracket to the back of the controller, then slide the bracket forward and remove it.



4. If a System hard drive is already installed in the upper bracket position, unscrew it from the bracket and move it to the lower bracket position. The power and IDE cables can remain connected.
5. Lower the APC hard drive into the upper bracket position.
6. Secure the hard drive(s) to the bracket with the screws provided.
7. Connect the power and IDE cables to the corresponding connectors on the hard drive(s) and main board. The cables are keyed for proper connection.

Note: Connect the System hard drive IDE cable to the main board connector labeled MPC8270 HARDDRIVE. Connect the APC hard drive IDE cable to the main board connector labeled ETX HARDDRIVE.



8. Slide the bracket back into the chassis, then fasten the screws connecting the bracket to the back of the controller.
9. Replace the controller cover.
10. Restore power to the controller.

Mounting the CX/CXi Controller

The CX/CXi is wall- and rack-mountable.

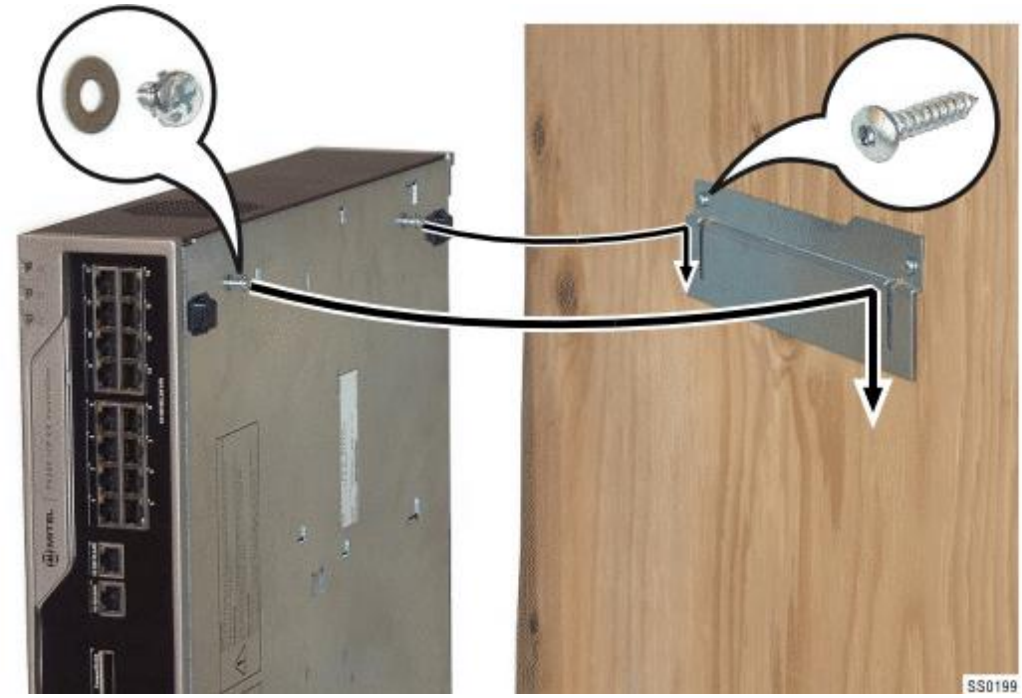
Wall Mount

Use the supplied bracket, screws (6-32 X 3/16), and washers to wall mount.

CAUTION: **Make sure the wall material is capable of supporting the weight of the unit. It is recommended that you mount the supplied bracket onto a backer board of 3/4" plywood that is securely fastened to the wall studs. DO NOT mount the bracket directly onto drywall (sheetrock). Mitel is not responsible for units damaged as a result of improper wall mounting.**

1. Turn the Controller upside down.
2. Locate the two holes on the bottom of the Controller.
3. Assemble the two supplied screws and washers, then screw them into the holes on the bottom of the Controller.
4. Mount the bracket onto the plywood backer board as follows:
   - Pre-drill two pilot holes spaced 11.25" (28.58 cm) apart into the backer board.
   - Orient the bracket over the two holes as shown below.
   - Insert a screw into the hole on the left side of the bracket.
   - Insert a screw into the hole on the slot on the right side of the bracket.
5. Secure the backer board to the wall studs.

6. Hang the Controller onto the bracket as shown below.



Rack Mount

Use the supplied brackets and screws (10-32 X 1/2) to rack mount.

CAUTION: When installing the system in an enclosed rack, you MUST provide adequate ventilation to ensure that the maximum ambient temperature inside the rack does not exceed 40°C/104°F.

CAUTION: Ensure that a hazardous condition does not result from any uneven mechanical loading.

CAUTION: When using the system in a rack, you should consider the connection of the equipment to the power supply circuit and the effect that overloading of circuits might have on overcurrent protection and supply wiring. When addressing this concern, refer to the system's ratings label.

1. Attach the brackets to the rack using the four supplied screws. Do not tighten the screws.
2. Slide the controller into the brackets.
3. Secure the controller in place by tightening the screws.

Install a Layer 2 Switch for Line Expansion

The following is a summary of the steps required to install expansion switches for the CX and CXi. For more information, see the Engineering Guidelines document on the Mitel Edocs website (http://edocs.mitel.com).

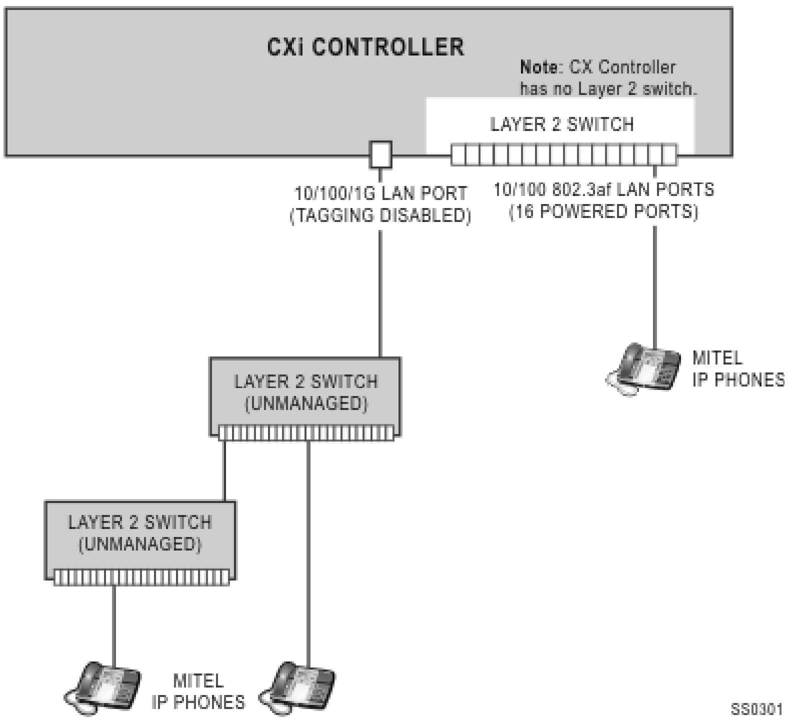
Voice Only Implementation

For a voice only IP network (i.e. a network with no PCs), system capacity can be expanded to a maximum of 100 users by connecting an expansion Layer 2 switch to port 17 (labeled 10/100/1G LAN) of the CX/CXi. Use one 48-port switch or two 24-port switches connected in a daisy chain. As a minimum, the expansion switches should support 10/100 BaseT; preferably, they should support 10/100/1000 BaseT.

Because some programming may be necessary (such as the port speeds), the expansion switches require a management interface.

Notes:

1. Connect a single expansion Layer 2 switch to port 17 (labeled 10/100/1G LAN). If using two 24-port switches, connect the second switch to the first in a daisy chain. Do not connect expansion switches to ports 1 to 16 (labeled 10/100 802.3af LAN).
2. MITEL telephones require power, which they can receive from an adapter or power brick, or from a powered Ethernet connection. Ports 1 to 16 of the CX provide Power over Ethernet (PoE), as do some expansion switches. Port 17 does not provide PoE. For details about PoE and planning a PoE installation, see the Engineering Guidelines document on the Mitel Edocs website (http://edocs.mitel.com).
3. Category 5 cable is required for the uplink connection between the expansion switches and the Controller, and is recommended for all other Ethernet connections. Category 3 cable can be used to connect single IP Phones directly to the expansion switches or to the Layer 2 switch of the CXi. For details, see Category 3 Cabling Guidelines.



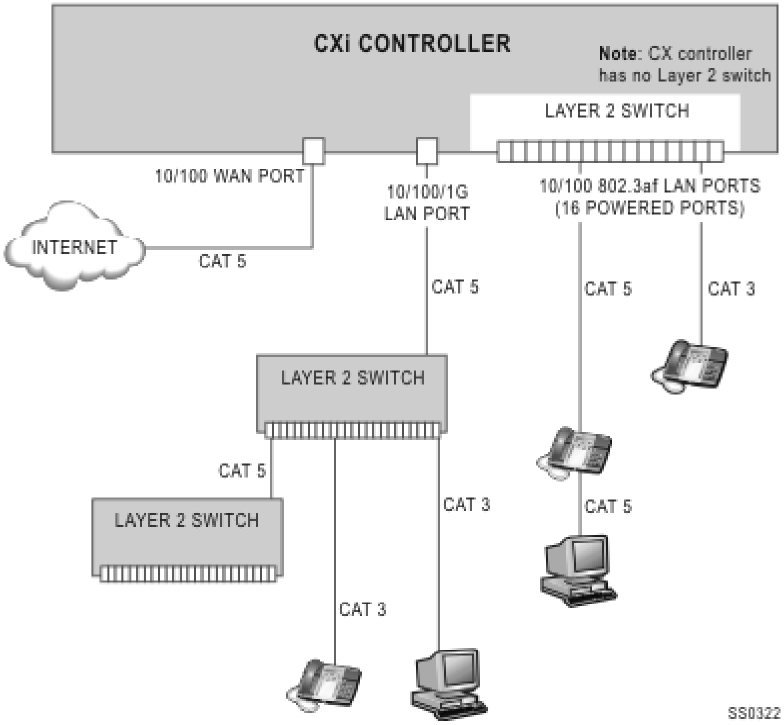
Voice and Data Implementation

For a voice and data IP network (i.e. a network with IP phones and PCs), system capacity can be expanded to a maximum of 100 IP phones and 64 PCs by connecting an expansion Layer 2 switch to port 17 (labeled 10/100/1G) of the CX/CXi. Use one 48-port switch or two 24-port switches connected in a daisy chain. As a minimum, the expansion switches should support 10/100 BaseT; preferably, they should support 10/100/1000 BaseT. The expansion switches must be manageable, and must adhere to the 802.1 p/Q VLAN standard.

The CX/CXi uses VLAN prioritization to ensure quality of service for voice calls. Normally, only one VLAN is required, the default VLAN (1), which carries both voice and data traffic. Optionally, the system can be programmed to support a Voice VLAN in addition to the default VLAN. Depending on this setup, voice packets from Mitel IP Phones can belong to either the default VLAN or the Voice VLAN. Regardless of their VLAN membership, voice packets always receive priority over data packets.

Notes:

1. Connect a single expansion Layer 2 switch to the 10/100/1G LAN port only. If using two 24-port switches, connect the second switch to the first in a daisy chain. Do not connect expansion switches to the 10/100 802.3af LAN ports.
2. VLAN Tagging must be enabled on all "trunk" links which connect the expansion switch(es). For two expansion switches, you need to enable VLAN tags on:
   - 10/100/1G LAN port of CX/CXi (port 17)
   - Switch port on first expansion switch which connects to port 17
   - Switch port on first expansion switch which connects to second switch
   - Switch port on second expansion switch which connects to first switch
3. If the parameter "Tag VLAN 1 on Trunk Port (17)" is enabled, the 10/100/1G LAN port applies 802.1p/Q VLAN tagging for VLAN 1 to packets that are already on VLAN 1. The port maintains tagging for packets that belong to the Voice VLAN. For Voice VLAN programming, see Subform 47 - System IP.
4. For the CXi, the expansion switches must be on the same VLAN that you have programmed for the internal Layer 2 switch.
5. For the CXi, once programmed on Subform 47 - System IP, all ports of the internal Layer 2 switch are on the same VLAN.
6. To maximize the bandwidth capability of the 10/100/1G LAN port, connect it to the highest speed port on the first expansion switch, preferably a 1G port.
7. There are two ways to connect IP devices (PCs) to the voice and data network: directly to a switch or indirectly through a dual-port IP phone.
8. Dual-port phones use the same port speed as the connected PCs. For this reason, PCs with 100 Mbps Ethernet cards are recommended.
9. Mitel telephones require power, which they can receive from an adapter or power brick, or from a powered Ethernet connection. The 10/100 802.3af LAN ports of the CX/CXi provide Power over Ethernet (PoE), as do some expansion switches. The 10/100/1G LAN port does not provide PoE. For details about PoE and planning a PoE installation, see the Engineering Guidelines document on the Mitel Edocs website (http://edocs.mitel.com).
10. Category 5 or better cable is recommended for all Ethernet connections in a mixed voice and data environment. However, Category 3 cable can used used to connect single Ethernet devices (IP phones or PCs) to the network. For more information, see Category 3 Cabling Guidelines.
11. For more information on voice and data networking, see the Engineering Guidelines document on the Mitel Edocs website (http://edocs.mitel.com).



Connect ONS/CLASS and LS/CLASS circuits

Connect the LS CLASS lines and any ONS phones to the line ports on the back of the controller using standard 4-conductor cables with RJ-11 connectors. Refer to the following table for circuit assignments and default numbering for ONS phones.

After connecting the LS trunks, use the LS Measurement tool in Maintenance to determine the Circuit Descriptor settings to program for optimum audio quality.

Table: ONS and LS Circuit Assignments

	Circuit	PLID	Default DN/Line Number		Circuit	PLID	Default DN/Line Number
Analog Main Board	ONS 1	1/13/3	200	Analog Option Board	ONS 1	1/13/19	--
ONS 2	1/13/4	201	ONS 2	1/13/20	--		
ONS 3	1/13/5	202	ONS 3	1/13/21	--		
ONS 4	1/13/6	203	ONS 4	1/13/22	--		
LS 1	1/13/7	1	LS 1	1/13/13			
LS 2	1/13/8	2	LS 2	1/13/14			
LS 3	1/13/9	3	LS 3	1/13/15			
LS 4	1/13/10	4	LS 4	1/13/16			
LS 5	1/13/11	5	LS 5	1/13/17			

LS 6	1/13/12	6	LS 6	1/13/18	

Notes:

1. Tip and Ring correspond to pins 3 and 4 of the RJ11 connector respectively.
2. ONS/CLASS ports do not support high-voltage message waiting lamps.
3. Trunks circuits located at 1/13/7 and 1/13/8 on the AMB and 1/13/13 and 1/13/14 on the AOB are System Fail Transfer (SFT) trunks. In the event of a system or power failure, the trunks connect to ONS circuits 1/13/3 and 1/13/4 on the AMB and circuits 1/13/19 and 1/13/20 on the AOB respectively.
4. For SX-200 ICP CX/CXi systems, Analog Option Board ONS PLIDs 1/13/19, 1/13/20, 1/13/21, and 1/13/22 can not be programmed as both stations and data stations in CDE Form 09 - Desktop Device Assignments and CDE Form 12 - Data Assignment at the same time. As soon as one of these PLIDs is programmed in one of these two forms, it will no longer appear in the other form. For example, programming PLID 1/13/20 in Form 09 as a station means that it will no longer appear in Form 12. Programming PLID 1/13/21 in Form 12 as a data station means that it will no longer appear in Form 09. This does not apply to SX-200 ICP MX systems.
5. If an SX-200 ICP CX/CXi database with ONS PLIDs 1/13/19, 1/13/20, 1/13/21, and 1/13/22 programmed as Door Relays is restored to an SX-200 ICP MX system, the PLIDs will not normally be listed in Form 09. Performing a search using the FIND EXT feature for extensions assigned to those PLIDs will, however, display them in Form 09.
6. ONS ports are not designed with the necessary safety protection for off premise connections. ONS ports must not be used to connect to off premise phones.

Installing an AX Controller

The SX-200 ICP AX controller is shipped in a package with the components listed in the table below. The system software and a default database are installed at the factory. Optional components are field-installed.

AX Installation Procedures

The following installation procedures are common to the AX, CX, CXi, and MX controllers:

- Installing the System Software on the External CompactFlash Card
- Program the Customer Data Entry Forms
- Installing IP Phones

All other installation procedures are unique to the AX controller. For additional information, refer the *SX-200 CX/CXi and AX Technician's Handbook*.

System Package

The AX controller is sold as a Hospitality hardware package that includes the components shown in the table below.

Table: AX Hospitality Hardware Package

AX Controller	1
24 Port ONS Card	4
4 LS + 12 ONS Port Combo Card	1
Dual T1/E1 Module	1
Power Cables (3 pack)	1
5540 IP Console	1

Note: The AX Hospitality Hardware package is primarily designed to be used in conjunction with the Hospitality Software Package (54003441).

Table: AX Controller System Package

LS/CLASS circuits	4
ONS circuits	108
Voice mail ports	20
DSPs	4 DSPs on Controller Card
IP Phones	one 5540 IP Console
Licenses IP Phone Voice Mailbox TDM Devices ACD Agent	 8 100 130 5

Equipment Installed in the AX Controller

Table: Equipment Installed in the AX Controller

Standard Components	
SX-200 ICP AX controller	
Include:	
Four DSPs on the mainboard - CompactFlash card (internal)	
Optional Components	
Dual T1/E1 Framer Module	One module can be installed, each of which has two T1 digital trunk ports that provide 24 B-channels for T1/D4
DSP Modules	Available in Dual DSP or Quad DSP and T1/E1 Combo MMC configurations for a maximum of 8 DSP resources per system (including 4 DSPs on digital main board).
T1/E1 Combo MMC	Combines T1 trunking and DSP functionality in a single module. Also includes a stratum 4 clock.

Configuration Rules

Each AX system can be configured according to the following rules.

The SX-200 ICP AX controller can be expanded to include:

- up to 288 ONS/CLASS circuits
- up to 48 LS/CLASS circuits
- up to two T1/E1 digital trunk ports which provides 24 B-channels each for T1/D4
- up to 20 voice mail ports
- up to 248 IP phones.

Typical Installation Sequence

Follow this typical sequence when installing a system.

- Prepare for Installation
- Install the SX-200 ICP Controller
  - Install optional controller hardware
  - Install software and database (optional)
  - Enable FOR Options.
  - Configure the controller
  - Connect phones and lines
- Integrate the controller into the IP network
  - LAN interface: program IP addresses, DHCP server, and internal Layer 2 switch
  - Expansion Layer 2 switch (if used): connect to SX-200 ICP Controller and program
- Put system into service
  - Connect power to the controller.

To install the SX-200 ICP AX controller

1. If your system includes optional controller hardware, install according to the instructions indicated in the following table:

Component	Installation Instructions
DSP Modules	Installing a DSP Module
T1/E1 Framer Module	Installing a Dual T1/E1 Framer
T1/E1 Combo Module	Installing T1/E1 Combo Module

4. Rack mount them, or place them on a desk or shelf. See Rack Mounting
5. Connect the ground stud on the rear panel of the controller to a hard-wired ground using 18 AWG (0.75mm 2/) gauge wire. The wire must have green or yellow insulation. Crimp the wire to the ground source.
6. Connect a PC to the maintenance port on the controller. See Serial Connection to the Controller.
7. Set up an FTP server (required for backups and software upgrades). See Setting up an FTP Server.
8. If you are NOT installing software or optional hardware in the controller, power up the system. See System Initialization.

Install Optional Hardware

Installing T1/El Combo Module or T1/E1 Dual Framer

The T1/E1 Combo MMC combines T1/D4 or PRI trunking and DSP functionality in a single card. It also includes a Stratum 4 clock and installs in Module Slot 1. The T1/E1 Dual Framer module provides T1/D4 and PRI Trunking.

Note: Replacement T1/E1 combo cards may have two RJ45 ports.The second port is not functional for the SX-200 controller. Follow the instructions below to install either card type.

To install a T1/E1 Combo Module or T1/E1 Dual Framer:

1. Switch the power off using the power button on the power supply. If a second power supply is installed, turn the power off on the second supply as well.
2. Remove the Controller card from the chassis.

   Remove the blanking plate (or the old MMC) from the controller by removing the screws that hold the standoffs to the controller. (The screws are on the back side of the controller card.)

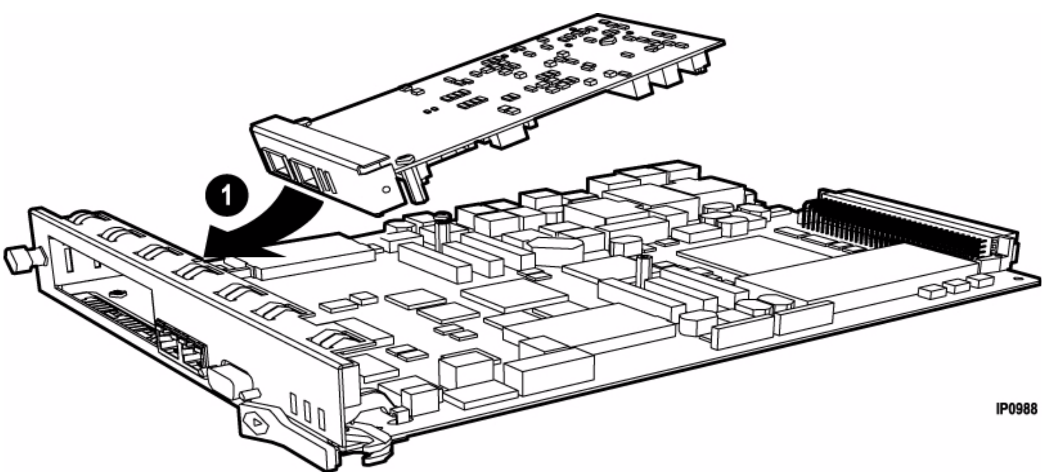
   Back off the controller faceplate screw nearest the MMC slot a couple of turns (because the screw interferes with the removal/insertion of T1/E1).

   Slide the blanking plate out of the opening from the back of the controller faceplate.

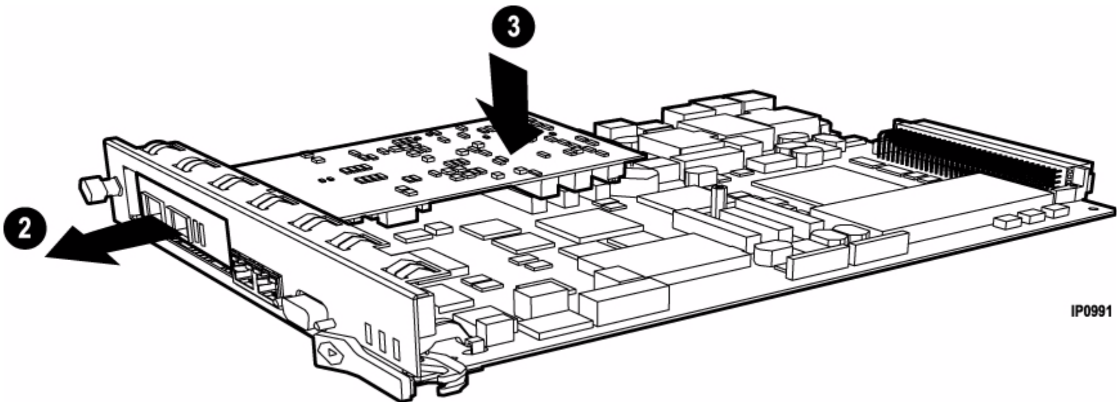
   Remove the two standoffs (closest to the face plate) from the blanking plate (or old MMC). Retain the standoffs and screws.

   Fasten the standoffs to the front of the new MMC.
3. Proceed with extreme care to avoid damaging components on the controller card.
4. Insert the T1/E1 Combo Module or T1/E1 Dual Framer into Module Slot 1.

   Carefully slide the MMC face plate under the lip of the controller face plate. See figure below.



   Do not push the MMC past the controller face plate as shown below.



5. Re-install and/or re-tighten the screws.
6. Re-insert the controller card.
6. Connect from the T1 line from the service provider to the RJ45 connector on the T1/E1 Combo Module. See T1/E1 Combo Module Tip/Ring for connector pinouts.
7 Program the module in CDE.
   - Assign the module a bay number in Form 53, Bay Location.
   - Program the T1 trunk card in Form 01, System Configuration.
   - Program the T1 link; see Trunk Support - T1.

Installing a DSP Module

The basic SX-200 ICP AX has four DSPs built into the motherboard. Two or four more can be added by installing either a Dual or Quad DSP Module in Module Slot 2. For information on determining DSP requirements, see “DSP configurations”.

To install the optional DSP Module:

1. Switch the power off using the power button on the power supply. If a second power supply is installed, turn the power off on the second supply as well.
2. Remove the controller card from the chassis.
3. Remove the DSP module from its packaging.
4 Insert the DSP module in Module Slot 2.
5 Secure the DSP module to the controller using the screws provided with the module.
6 Re-insert the controller card.

Installing a Dual T1/E1 Framer Module

The Dual T1/E1 Framer module has two digital trunk ports, each of which can be programmed to support either T1/D4 or PRI. Only 1 module can be installed in MMC slot 1 of the AX controller.

To install a Dual T1/E1 Framer Module:

1. Switch the power off using the power button on the power supply. If a second power supply is installed, turn the power off on the second supply as well.
2. Remove the controller card from the chassis.
3. Insert the Dual T1/E1 Framer into Module slot 1 on the Main Board connector.
4. Attach the screws.
5. Re-insert the controller card.
6. Program the module in CDE:
   - Assign the module a bay number in Form 53, Bay Location.

- Program the T1 link; see Trunk Support - T1.

Installing the System Software (Optional Initial Installation)

The SX-200 ICP is shipped from the factory with the system software and a default database. Perform this procedure only if you are doing one of the following:

- are upgrading the system software on site
- require a language other than the default English for voice mail prompts
- require a second language for bilingual voice mail operation
- are replacing the internal CompactFlash card or installing a hard drive
- are re-initializing the controller by re-installing the system software

Note: The initial power-up and the reset in this procedure will each take 5 to 10 minutes.

Note: For details on installing the AX, refer the the *SX-200 CX/CXi and AX Technician's Handbook*.

To install an "Initial" software load on a CompactFlash card

Note: For details on AX Installation, see *Installing AX Controller Software* in the *SX-200 ICP CX/CXi and AX Technician's Handbook*.

1. Insert the 256 MB, Mitel-supplied CompactFlash card into the CompactFlash reader connected to your PC.
2. Run the SX-200 ICP SETUP program from the supplied CD-ROM.
3. In the Installation software, select **Initial [Compact Flash Card] Installation**.
4. (Optional) Select the additional language(s) that you want to install.

Note: The additional languages enable the embedded voice mail system to operate with bilingual prompts. Bilingual prompts are a purchasable FOR option.

Note: It is not possible to install all three languages as the system only supports a maximum of two languages, one of which must be English.

5. Select the drive letter assigned to the CompactFlash reader.

   If the flash drive cannot be selected from the list, choose another drive letter, and then manually change it to the flash drive letter.
6. Select Format to format the CompactFlash card.

Note: When formatting the CompactFlash card, specify FAT as the file system.

IMPORTANT: Use only Mitel-supplied CompactFlash cards. 256 MB is the minimum required card size. DO NOT partition nor copy files to the card before proceeding with the software installation.

7. Select the database to be installed from, either **Premier** (MX only), **Default**, **Blank** or **Custom**.

Note: Selecting the **Custom** database allows customers to navigate through their file system to select a database of their own (CX/MX controllers with release 2.0 or later software only).

   a. If a **Custom** database is selected, click **Browse** to locate the database, or click **Next** to use a default database.
   b. Navigate to the database file and select it.
   c Click **OK**.

8. Click **Next** to begin installing the software on the CompactFlash card.
9. Click **Finish** to end the software installation.
10. Wait for the flash card reader to finish writing to the card before ejecting it.

Note: To avoid ending the writing operation too soon, DO NOT click **STOP** prior to **EJECT**.

11. Remove the card from the reader, and then insert it into the CompactFlash slot on the front of the SX-200 ICP Controller.
12. Log into Maintenance.
13. Reboot the SX-200 ICP by pressing SYSTEM > RE_START > RESTART_SYSTEM.

    The system will boot from the flash card and run the install utility. The install utility will launch automatically and the system will restart automatically after completing the installation.

    After the boot process is complete, the following message appears:

    "Press Return Four Times to Log In"
14. Press **Enter** four times quickly on the keyboard to launch the CDE program.
15. Follow the instructions in the next procedure Programming the Controller IP Address and DHCP Settings if using non-default IP settings.

Note: The CompactFlash card configured by this procedure can be used for either an SX-200 ICP CX/CXi or an SX-200 ICP MX system.

Software Upgrades

To upgrade SX-200 ICP controller software, or to migrate software from from an EL/ML system to an SX-200 ICP MX, refer to the following maintenance procedures:

- Upgrading using the External CompactFlash Card
- Upgrading Remotely by FTP
- Upgrading from SX-200 ICP Release 1.x to Release 2.x
- Migrating an SX-200 EL/ML System to an SX-200 ICP MX.

For details on upgrading the AX, refer the the *SX-200 CX/CXi and AX Technician's Handbook*.

Programming the Controller IP Address and DHCP Settings

The IP addressing and DHCP settings for the SX-200 ICP are set at the factory (see table below). Use the following procedure to change the defaults if required. The procedure can be used for "voice-only" and "voice and data" implementations. For more information about programming the controller to support voice and data (IP phones and PCs), see Integrating the MX into the IP Network and Integrating the CX into the IP Network.

To set the SX-200 ICP Controller IP address:

1. If you have not already done so, establish a serial connection to the Maintenance port on the SX-200 ICP Controller.
2. In Form 47 (IP Networking), Subform 01 (System IP), enter the System IP settings.
3. In Form 47 (DHCP Server), Subform 02 (DHCP Parameters), enter the DHCP settings. See the table below for required settings.
4. Reset the controller.

Note: You may use the internal or an external DHCP Server. The controller is shipped with the DHCP server Enabled. Ensure the LAN Administrator enters the options in the following table.

DHCP options 128-133 used to configure Mitel IP endpoints have been reclassified as public options by the Internet Engineering Task Force (see RFC 2133 and RFC 3925). To comply with the change, Mitel uses options 125 or 43 depending on the server's ability to support them and on administrator preference. (The embedded server supports both options with 125 as the default.) The old options can still be used to provide backward compatibility with IP sets that have yet to be upgraded with firmware that supports the new options. After the upgrade, the old options may be removed to prevent future conflicts with standard use, or other vendors' use of these options.

Table: Internal DHCP Server default settings (shipped enabled)		
Subnet IP	**Network mask**	**Range (Scope)**
192.168.1.0	255.255.255.000	192.168.1.10 - 192.168.1.250
DHCP Options for Release 4.0 and later:		
Option #	**Option Name**	**Value**
3	Router	192.168.1.1
125 (New for Rel 4.0)	Vendor Specific	

ASCII string. Old options 128 - 133 are replaced by tags included in the configuration string:

id:ipphone.mitel.com
TFTP Server = **sw_tftp**
SX-200 IP Addr = **call_srv**
IP Phone Analyzer = **ipa_srv**
VLAN ID = **vlan**
VLAN Priority = **l2p**
Diffserv Codepoint = **dscp**
HTTP Proxy Server = app_proxy*
*not used on the SX-200 ICP

DCHP Options prior to Release 4.0:		
128	IP Phone TFTP Server	192.168.1.2
129	SX-200 ICP IP Address	192.168.1.2
130	DHCP Server Identifier	MITEL IP PHONE
132-VLAN ID (CX/CXi only)	Numeric	1
133-VLAN Priority (CX/CXi only)	Numeric	6

New DHCP Options

In Release 4.0, DHCP Options 128 - 133 are replaced by a series of "tags" that form an ASCII data string in Option 125 (default for Mitel-specific options) and Option 43 (for Mitel- and non-Mitel-specific options.) :

Option 125: Use this option to enter Mitel-specific options into the form to automatically create the ASCII configuration string. Press the **NUMERIC** softkey and enter the Mitel Vendor ID **1027**.

Option 43: Use this option to enter Mitel and non-Mitel-specific options to manually create the ASCII configuration string. Vendor ID is optional but, if entered, must be an **ASCII** string. (For example, the Mitel Vendor ID string is **ipphone.mitel.com**.)

The default configuration string for the internal DHCP Server is:

id:ipphone.mitel.com;sw_tftp=192.168.1.2;call_srv=192.168.1.2;vlan=1;l2p=6

Programming the Customer Data Entry Forms

The CDE Forms are factory-set with default values that make it easier and faster to program the system. The defaults allow you to install the system, connect IP phones and two analog terminals (phone, fax, or modem), and then make extension-to-extension calls without any additional programming. You will also be able to receive fax and modem calls, but will have to program ARS to make external calls.

SX-200 ICP Premier Business systems (applies to MX controller only) require the Premier database which is provided on the system software CD. Before enabling FOR Options in these systems, replace the factory-installed default database with the Premier database. Doing so prevents conflicts with Option 114 (Maximum IP Sets). The conflict is caused by the different number of IP phones programmed in the two databases: twenty in the default database versus eight in the Premier database. For more information about the database and how to install it, see Installing an Alternate Database.

The CDE Forms are factory-set with default values that make it faster and easier to program the system. The defaults allows you to install the system and connect up to 20 IP phones and two analog terminals (phone, fax, or modem) and make extension-to-extension calls without doing any programming. You will also be able to receive fax and modem calls, but will have to program ARS to make external calls.

The default numbering plan uses three-digit extension numbers. If you require four-digit extension numbers, you will have to reprogram Forms 9, 17, and 50. Or, you can use the blank database provided on the SX-200 ICP software CD. For more information, see Installing an Alternate Database.

SX-200 ICP Premier Business systems require the Premier database which is provided on the SX-200 ICP software CD. For more information about the database and how to install it, see Installing an Alternate Database.

Use the CDE terminal to do the programming.

IMPORTANT: Certain options in Form 04, System Options/System Timers, could cause unexpected behaviours in system operation if changed from their factory-set (default) values. See System Options to Avoid for more information.

Integrating the MX into the IP Network

This section is for technicians who are installing the MX controller for voice and data.

IMPORTANT: Do not attempt the procedures in this section until you have successfully completed the SX-200 ICP Advanced Installation and Maintenance Course.

IP Networking Requirements

Adding PCs and data devices to a voice-only LAN or installing the SX-200 ICP into an existing data LAN requires careful planning. Refer to the Engineering Guidelines document on the Mitel Edocs website (http://edocs.mitel.com) for detailed information on network planning and configuration.

Plan the Local Area Network

To maintain optimum voice quality, it is important to recommend that voice and data traffic be segregated as much as possible. There are three methods of segregation, depending on the existing LAN configuration:

- Run Voice and Data on separate physical networks
- Run Voice and Data on separate Virtual LANs (MX only)
- Use a separate subnet for voice traffic

The following is a summary of network planning guidelines. For more information, see the Engineering Guidelines document on the Mitel Edocs website (http://edocs.mitel.com).

- Use Ethernet switches instead of hubs. It is recommended that each ICP Controller and associated IP phones be configured in their own subnet. This configuration prevents an IP phone from registering with the wrong ICP Controller.
- Use Full Duplex Fast Ethernet where possible between switches.
- Choose between two options for the IP Phones. One: Provide one switched Ethernet drop to the desktop for the PC/IP Phone. In this case, the PC connects to the network through the IP phone. Two: Provide two switched Ethernet drops, one for the PC and one for the IP phone. For optimum voice quality, VLANs are highly recommended where only one Ethernet drop to the desktop is used.
- Install the SX-200 ICP in its own broadcast domain along with the majority of its IP phones.
- Make sure that there is only one active Dynamic Host Configuration Protocol (DHCP) server within the Broadcast Domain. The SX-200 ICP includes a DHCP server.
- Place functional groups on the same switches and/or subnets to reduce traffic across backbones and routers.
- Use Layer 2 Switches (port-based) to set the voice VLAN default for the MX controller and to modify the data format to add 802.1p/Q (tagging). IEEE 802.1p/Q priority is supported at the phones but not supported on the SX-200 ICP.
- Ensure that when VLANs (802.1p/Q) are used with IP Phones, the Layer 2 switch ports that are connected to IP phones have the port configured to accept data already tagged for the voice VLAN and a default or native VLAN (typically set to 1) to accept untagged data from an attached PC and convert it to tagged data.
- Do not mix data and voice with the same priority. Set any queues and priority so that voice has a high priority and data has a low priority.
- Ensure that the phones and SX-200 ICP are not connected to the network in such a way that the connection could introduce a network loop. For mobility of IP phones, it is recommended that Layer 2 switch ports, configured for operation with the SX-200 ICP and IP telephones, have "spanning tree protocol" disabled and/or configured to "Portfast."

This system supports G.711 voice encoding and G.729 voice compression between IP Phones. G.727 compression is a purchaseable FOR option -- System Option 120 (Number of Compression Resources).

Plan the Wide Area Network

If you plan to connect to a WAN through a Layer 3 switch or router

- Activate DHCP forwarding (DHCP Helper) on the router to forward a DHCP request across a Layer 3 switch or router. Alternatively, use a secondary DHCP server on the remote side of the Layer 3 device. This secondary DHCP server must programmed with the same options as the main DHCP server for the phones to operate properly.
- Configure the routers to prioritize voice traffic based on Type Of Service (TOS) using techniques such as Weighted Fair Queuing (WFQ) to ensure optimum performance and voice quality when using IP phones across routed links, IP phones are preset to precedence 5 with Minimum Delay. A high priority queue has settings 4, 5, 6, 7; A low priority queue has settings 0, 1, 2, 3. PCs should use low priority. On slow WAN links between routers, set the Maximum Transmittable Unit (MTU) appropriately for the speed of the WAN link (a value of 500 is recommended for links using T1/E1 type seeds).
- Be aware that not all routers or Layer 3 switches will recognize 802.1 p/Q VLAN information. In such a situation, you may not be able to run multiple VLAN/subnets on one physical port, and it will be necessary to provide a separate physical connection for each VLAN and subnet in use.
- Enable Internet Control Message Protocol (ICMP) Redirect on your Layer 3 switches and default gateway. The redirect will forward messages to the correct LAN segment and reduce broadcast messages and LAN usage.
- Set the WAN link at the minimum recommended speed of 1.544 Mbps (T1) with Frame Relay, HDLC, PPP, and compressed PPP.

- Do not use more than 70% bandwidth if the Layer 3 switch or router can support TOS and queue priority. If the Layer 3 switch or router cannot support TOS queue, do not use more than 40% bandwidth. The usage of the link must be determined before installing IP phones to determine the level of operation possible. A heavily used link may require an upgrade or implementation of TOS and MTU (500) settings.

For further considerations pertaining to the implementation of WANs, see the Engineering Guidelines document on the Mitel Edocs website (http://edocs.mitel.com).

Plan IP Address Requirements

You need IP addresses for

- The MX controller
- Each IP phone (a range of IP addresses assigned by the DHCP Server or statically assigned)
- A router or gateway (if using)

The controller uses the following reserved IP addresses:

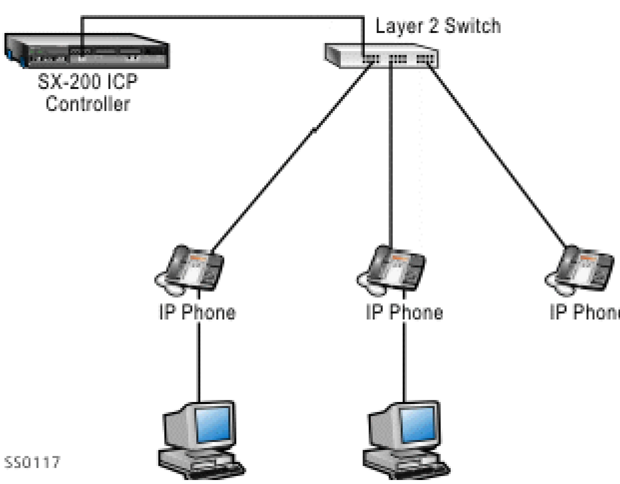
192.168.10.1 - 192.168.10.255
192.168.11.1 - 192.168.11.255
192.168.12.1 - 192.168.12.255
192.168.13.1 - 192.168.13.255

Ensure no other devices on the network use IP addresses within that range.

Basic PC Networking

The dual-port IP Phones (5010, 5020, 5212, 5215, 5220, 5224, 5312, and 5324) provide an inexpensive way to network a small number of PCs.

Figure: Basic PC Network



Enabling the (2nd) Port on IP Phones

IMPORTANT: To ensure optimum network performance, DO NOT connect servers to the 2nd port on IP phones.

1. Form 04, System Options/System Timers
   - Enable System Option 131, PC (2nd) Port on IP Phone.
2. Form 03, COS Define

- Enable Option 280, PC (2nd) Port on IP Phone.

Virtual LANs (VLANs)

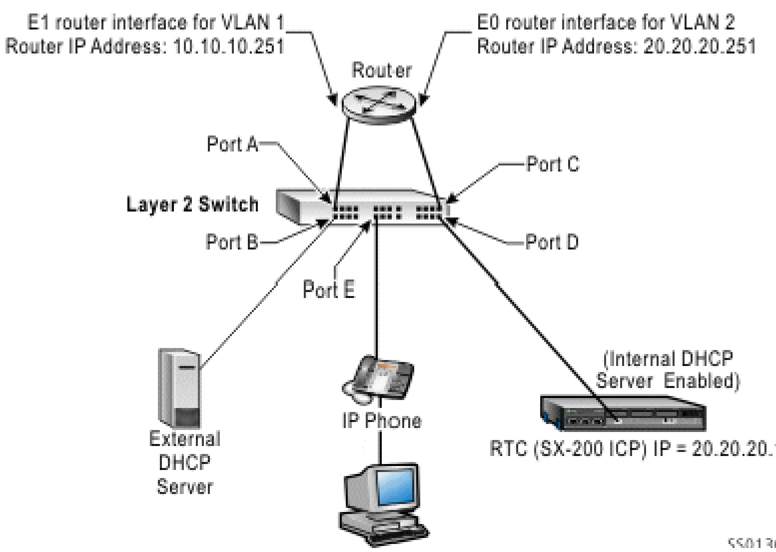
This section shows examples of the three most common, converged voice and data networks for an SX-200 ICP.

Configuration 1: One DHCP Server per VLAN

Configuration 2: One External DHCP Server for Two VLANs

Configuration 3: Router on a Stick (one router interfacing multiple VLANs)

Configuration 1: One DHCP Server per VLAN



DHCP Server Settings

The following table shows the DHCP settings programmed for this configuration. See Configuring a Windows 2000 DHCP Server for information on programming SX-200 ICP DHCP settings on a Windows 2000 DHCP server.

Setting	DHCP Server on VLAN 1 (IP: 10.10.10.2) Scope 1	Internal DHCP Server on Controller Scope 1
DHCP	10.10.10.10 to 10.10.10.100	20.20.20.10 to 20.20.20.100
Subnet	255.255.255.0	255.255.255.0
Option 03 - Router	10.10.10.251	20.20.20.251
Option 125 - New for Release 4.0	See Note 1	See Note 1
Option 128 - (TFTP Server IP Address)	20.20.20.1	20.20.20.1
Option 129 - (RTC IP) (SX-200 ICP IP address)	20.20.20.1	20.20.20.1
Option 130 - IP Phone DHCP Server (MITEL IP PHONE)	MITEL IP PHONE	MITEL IP PHONE

Setting	DHCP Server on VLAN 1 (IP: 10.10.10.2) Scope 1	Internal DHCP Server on Controller Scope 1
Option 132 - VLAN ID	2	2
Option 133 - Priority	6	6

NOTE 1: In Release 4.0, DHCP Options 128 - 133 are replaced by a series of "tags" that form an ASCII data string in Option 125 (default for Mitel-specific options) and Option 43 (for Mitel- and non-Mitel-specific options.) In this example, the ASCII data strings in Option 125 would look like this:

id:ipphone.mitel.com;sw_tftp=20.20.20.1;call_srv=20.20.20.1;vlan=2;l2p=6

Layer 2 Switch Settings

The following two tables show settings on a Cisco and an HP Layer 2 switch for the Configuration 1 example. These settings also apply for the other network configuration examples.

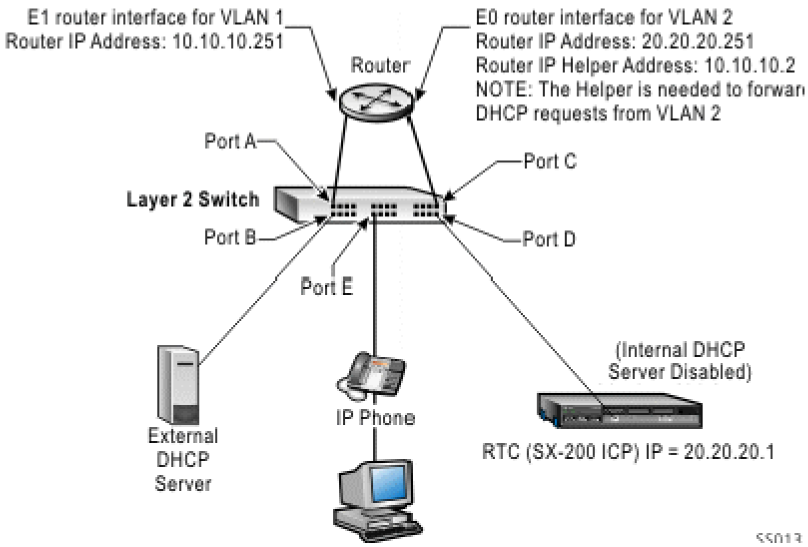
Table: Cisco Layer 2 Switch Settings for All Configuration Examples

Port	Use	Command
A	Access port for VLAN 1	None (by default, all ports belong to VLAN 1)
B		
C	Access port for VLAN 2	L2switch(config-if)#switchport mode access L2switch(config-if)#switchport access vlan 2
D		
E	Trunk port with Dot1q for IP Phone	L2switch(config)#interface fast 0/5 L2switch(config-if)#switchport mode trunk L2switch(config-if)#switchport trunk encapsulation dot1q

Table: HP Layer 2 Switch Settings for All Configuration Examples

Port	Use	Command
A	Access port for VLAN 1	VLAN 1 = untagged VLAN 2 = NO
B		
C	Access port for VLAN 2	VLAN 1 = NO VLAN 2 = untagged
D		
E	Trunk port	VLAN 1 = untagged VLAN 2 = tagged

Configuration 2: One External DHCP Server for Two VLANs



To configure the SX-200 ICP system to use an external DHCP server (for example, Windows NT server or Windows 2000 server) through the Layer 2 switch port, you must disable the DHCP server that is built into the SX-200 ICP. The local phones and PCs on the SX-200 ICP LAN will then be able to receive IP addresses from the external server.

DHCP Server Settings

The following table shows the DHCP settings programmed for this configuration. See Configuring a Windows 2000 DHCP Server for information on programming SX-200 ICP DHCP settings on a Windows 2000 DHCP server.

	DHCP Server on VLAN 1 (IP: 10.10.10.2)	
Setting	**Scope (Range) 1**	**Scope (Range) 2**
DHCP	10.10.10.10 to 10.10.10.100	20.20.20.10 to 20.20.20.100
Subnet	255.255.255.0	255.255.255.0
Option 03 - Router	10.10.10.251	20.20.20.251
Option 125 - New for Release 4.0	See Note 1	See Note 1
Option 128 - (TFTP Server IP Address)	20.20.20.1	20.20.20.1
Option 129 - (RTC IP) (SX-200 ICP IP address)	20.20.20.1	20.20.20.1
Option 130 - IP Phone DHCP Server (MITEL IP PHONE)	MITEL IP PHONE	MITEL IP PHONE
Option 132 - VLAN ID	2	2
Option 133 - Priority	6	6

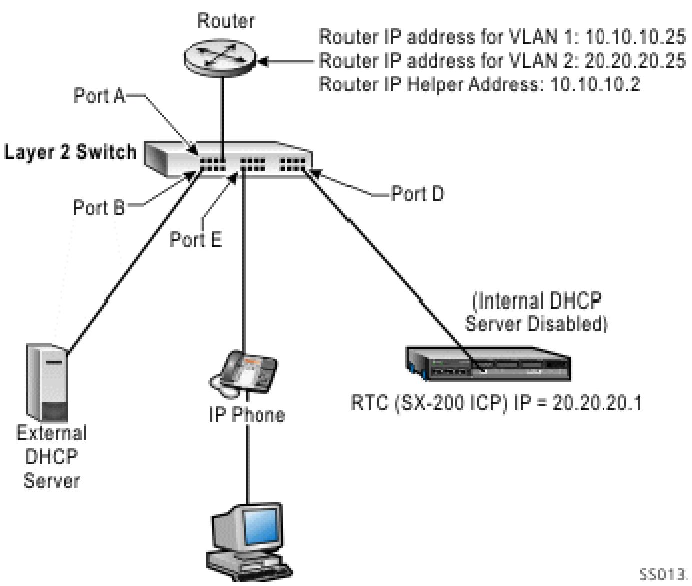
NOTE 1: In Release 4.0, DHCP Options 128 - 133 are replaced by a series of "tags" that form an ASCII string in Option 125 (default for Mitel-specific options) and Option 43 (for Mitel- and non-Mitel-specific options). In this example, the ASCII data strings in Option 125 would look like this:

id:ipphone.mitel.com;sw_tftp=20.20.20.1;call_srv=20.20.20.1;vlan=2;l2p=6

Layer 2 Switch Settings

Use the same settings as those for the Configuration 1 example.

Configuration 3: Router on a Stick



DHCP Server and Layer 2 Switch Settings

Use the same settings as those for the Configuration 1 example.

Configuring a Windows 2000 DHCP Server (prior to Release 4.0)

You can modify a Windows 2000 DHCP server to support IP Phones. A DHCP server must be configured to support the IP Phones for each subnet. The following items must be configured:

- TFTP Server IP address (System IP Address of SX-200 ICP)
- IP address of the RTC card
- Mitel tag “MITEL IP PHONE”

To modify a Windows 2000 DHCP Server prior to Release 4.0:

1. On the Start menu, point to **Programs**, then **Administrative Tools**, and click **DHCP**.
2. Highlight the Server name and point to **Action**, then click **Set Predefined Options**.
3. On the Predefined Options and Values window, click **Add**.
4. On the Option Type window, set the following:
   Name: IP phone TFTP Server IP Address
   Type: IP Address
   Code: 128

Click **OK**.

Value: enter the System IP Address of SX-200 ICP.

Click **OK**.

5. On the Predefined Options and Values window, click **Add** again.
6. On the Option Type window, set the following:

   Name: RTC IP Address
   Type: IP Address
   Code: 129

   Click **OK**.

   Value: enter the System IP Address of SX-200 ICP.

   Click **OK**.
7. On the Predefined Options and Values window click **Add** again.
8. On the Option Type window, set the following:

   Name: IP Phone DHCP Server
   Type: String
   Code: 130

   Click **OK**.

   Value: MITEL IP PHONE.

   Click **OK**.
9. If using VLANs for Quality of Service (QoS), set the following on the Option Type window:

   Name: VLAN ID
   Type: Hex LONG (32 bit word)
   Code: 132

   Click **OK**.

   Value: enter a numeric value for the VLAN.

   Click **OK**.

Note: The server will automatically convert the numeric value to hex.

10. If using VLANs for Quality of Service (QoS), set the following on the Option Type window:

    Name: Priority
    Type: Hex LONG (32 bit word)
    Code: 133

    Click **OK**.

    Value: enter a value from 1 to 7. Mitel recommends 6 for IP phones.

    Click **OK**.

Note: The server will automatically convert the numeric value to hex.

11. Highlight the Scope which contains the IP range for the IP Phones and select **Scope Options**.
12. On Action, click **Configure Options**.
13. On the general window, Select option **003 Router** and enter the IP address of the default Gateway and then click **Add**.
14. Click **OK**.

Configuring a Windows 2000 or Windows 2003 DHCP Server for Release 4.0 and later

Neither Windows 2000 nor Windows 2003 support DHCP options 124/125. Options 60/43 must be used instead.

Note: Options 128-133 used in previous ICP releases are required to provide backward compatibility with IP sets that have yet to be upgraded with firmware that supports options 124/125 or 60/43. After the upgrade, the old options may be removed to prevent future conflicts with standard use or with other vendors' use of these options.

To create Options 60/43 on a Windows 2000 or Windows 2003 DHCP server:

1. Upgrade the SX-200 ICP to Release 4.0.
2. Upgrade the IP Phone firmware.
3. Start DHCP Manager.
4. In the console tree, click the applicable DHCP server branch.
5. Right-click the server, then click **Define Vendor Classes** followed by **Add**.
6. In the New Class dialog box, type "Mitel Vendor Class" or other name for the new option.
7. Type "ipphone.mitel.com"in the right side of the text box under **ASCII**.
8. Enter a null terminator (0x00) at the end of the hex string under **Binary**.
9. Click **OK** and then click **Close**.
10. On the **Action** menu, select **Set Predefined Options**.
11. In the **Predefined Options and Values** dialog box, select the Mitel Vendor Class from the Option class list. Click **Add**.
12. In the **Option Type** dialog box, enter the following:

Field Name	What to enter...
Name	Mitel Option
Data type	String
Code	001

13. Click OK.
14. In the **Predefined Options and Values** dialog box, select **001 Mitel Option** and the Option name and enter the Mitel Information Data string with the appropriate values for <IP address> and <N>:

    id:ipphone.mitel.com;sw_tftp=<IP address>;call_srv=<IP address>;dscp=<N>;vlan=<N>;l2p=<n>
15. Click **OK**.
16. Add the Mitel option to the DHCP scopes that require it, modifying the ID string accordingly.

Integrating the AX into the IP Network

For details on integrating the AX into the IP Network, refer the the *SX-200 CX/CXi and AX Technician's Handbook*.

Integrating the CXi into the IP Network

This section is for technicians who are installing the CXi controller in a standalone network that is dedicated to voice traffic, or in a network (either new or existing) that combines voice and data traffic.

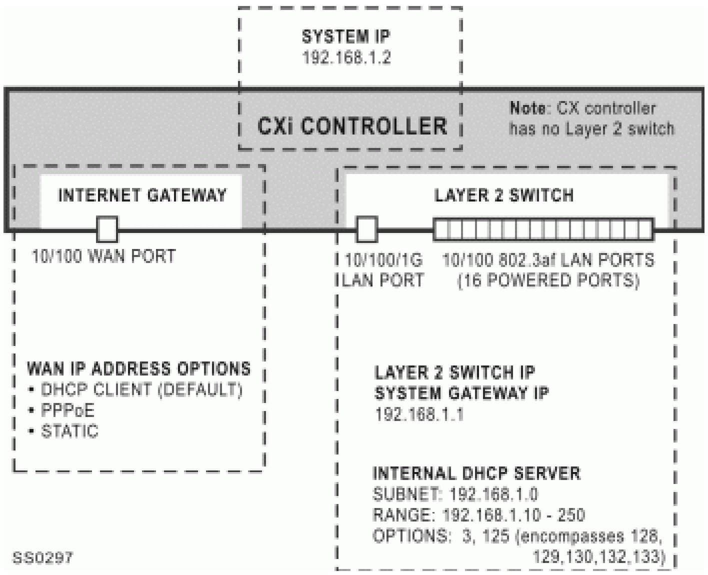
CAUTION: Do not attempt the procedures in this section until you have successfully completed the SX-200 ICP Advanced Installation and Maintenance Course.

Note: IP Networking Requirements for the CX are the same as those for the MX.

Default IP Network Settings

The IP addressing information for the SX-200 ICP is set at the factory (see the illustration and tables below). If you are integrating the CXi into a network that is dedicated to voice traffic, then no changes should be required. If you are integrating the CXi into a network that combines voice and data traffic, then you may need to change the default settings on the CXi or on devices that are already deployed. This must be done prior to connecting the CXi to an existing network in order to prevent IP addressing conflicts.

Note: The CXi controller requires two IP addresses on the same subnet on the LAN. Both addresses must be configured.



CXi Controller IP Address Assignment Defaults

Table: IP Address Assignment Defaults

Interface Name	Function	Default	Comment
Internet Gateway	• Provides access to Internet through WAN interface • Provides NAT and firewall functionality	DHCP client	

This interface can receive its IP address by DHCP, PPPoE, or static assignment. Alternatively, this interface can be programmed to support the optional Application Processor Card (APC).

Before this interface can be programmed or used, the Internet Gateway license must be enabled on Form 04 (Option 83).			
System IP	Network location point that provides call control functionality for IP phones. Also provides management access to the	192.168.1.2	Before connecting the CXi to the LAN, confirm that the IP address for this interface does not conflict with an IP address assigned to another device on

	system.		the LAN. If it does, change one or both addresses. This may require programming the CXi with an address in a new subnet. (See Note)
Layer 2 Switch IP	• Provides 16 port Layer 2 switch on LAN interface • Capable of VLAN prioritization on the default VLAN (1), or on a VLAN dedicated to voice traffic • Default System Gateway IP address, allowing connection to the Internet	192.168.1.1	As above.
System Gateway IP	Default gateway to other networks.	192.168.1.1 (same as Layer 2 Switch IP)	

If there is only one subnetwork on the LAN, enter the IP address of the Layer 2 Switch.

If there are multiple subnets, enter the IP address of the router connected to the local subnet. In all cases, DHCP server option 3 should be the same as the System Gateway IP to ensure that all DHCP clients use the proper default gateway.

Note: Often, the internal network setup determines whether the System and Layer 2 Switch IP addresses require updating. If the internal network is a "voice-only" network, where the CXi provides services to IP phones but not PCs, then the addresses can usually be left unchanged. But if the internal network is a "voice and data" network, where the CXi provides services to both IP phones and PCs, then the addresses may require updating. This is especially true if the CXi is being integrated into an existing data network. .

CX/CXi Controller Internal DHCP Server Defaults

DHCP options 128-133 used to configure Mitel IP endpoints have been reclassified as public options by the Internet Engineering Task Force (see RFC 2133 and RFC 3925). To comply with the change, Mitel uses options 125 or 43 depending on the server's ability to support them and on administrator preference. (The embedded server supports both options with 125 as the default.) The old options can still be used to provide backward compatibility with IP sets that have yet to be upgraded with firmware that supports the new options. After the upgrade, the old options may be removed to prevent future conflicts with standard use, or other vendors' use of these options.

Table: Internal DHCP Server default settings

Subnet IP	Network mask	Range (Scope)
192.168.001.0	255.255.255.000	192.168.1.10 - 192.168.1.250
DHCP Options		
Option #	**Option Name**	**Value**
3	Router	192.168.1.1
125	Vendor Specific	ASCII configuration string (see note 4)
128	IP Phone TFTP Server	192.168.1.2
129	SX-200 ICP IP Address	192.168.1.2
130	DHCP Server Identifier	MITEL IP PHONE
132	VLAN ID	1
133	VLAN Priority	6
134	DiffServ Code Point	44 (upgrades) 46 (new installations)

Notes:

1. The internal DHCP server on the CX/CXi is enabled by default.
2. You can use an external DHCP server. However, it is recommended that only one DHCP server is enabled on the LAN. Make sure that it is programmed to provide addresses and options to all devices that require them. For example, if you use an external DHCP server, program it with settings from this table.
3. For releases prior to 4.0, Options 132 and 133 are required for voice and data implementations. If only one VLAN is required, use the default values. If a separate voice VLAN is required, change options 132 and 133 to match the Voice VLAN settings programmed in Form 47, Subform 1 (System IP). In both setups, the internal L2 switch prioritizes voice packets ahead of data packets.
4. NEW DHCP Options for Release 4.0:
   Option 125: Use this option to enter Mitel-specific options in a form to create the ASCII configuration string.

   Option 43: Use this option for non Mitel- and Mitel-specific options that must be entered manually to create the configuration string.

The default configuration string is:
id:ipphone.mitel.com;sw_tftp=192.168.1.2;call_srv=192.168.1.2;vlan=1;l2p=6;dscp=44 (or 46).

IP Network Requirements

The following is a summary of network planning guidelines for the CXi. For more information, see the Engineering Guidelines document on the Mitel Edocs website (http://edocs.mitel.com).

LAN Requirements (Voice Only or Voice and Data)

These conditions must be met before the Layer 2 Switch of the CXi can be connected to the LAN. The conditions apply to all implementations (voice only, and voice and data implementations).

- A subnet with IP addresses reserved for the following:
  - the CXi system
  - the internal Layer 2 switch

  - the system gateway - IP address of a router on the LAN, the APC (if installed), or the L2 switch of the CXi
  - static or DHCP-assigned addresses for IP phones (DHCP-assigned is recommended)
- a system hostname
- a Dynamic Host Configuration Protocol (DHCP) server within the Broadcast Domain. The CXi includes a DHCP server that is capable of assigning options to clients (Mitel IP Telephones).
- Ethernet cabling to connect the equipment together
- Optional:
  - a router on the same subnet as the CXi
  - up to two Layer 2 switches (unmanaged for a voice only implementation; managed and VLAN-capable for a voice and data implementation)
  - SMTP server address to support e-mail forwarding of voice mail and logs
  - IMAP server address to support Standard Unified Messaging
  - DNS server address; external DHCP server information
  - one PC per IP phone (phones must be dual-port models; PCs must have 10/100BaseTx Ethernet cards)

Note: LAN traffic requires a system gateway. If the LAN port connects directly to a single subnet, the L2 switch of the CXi is the system gateway. If the LAN port connects to a router leading to additional subnets, the router is the system gateway.

WAN Requirements (Voice and Data)

These conditions must be met before the Internet Gateway of the CXi can be connected to the Internet for a voice and data implementation. The conditions do not apply if the Internet Gateway is disconnected.

- System option 83 (Internet Gateway) must be enabled in CDE form 04.
- Static, PPPoE, or DHCP IP address assignment details for the Internet Gateway (WAN Interface):

Static	PPPoE client	DHCP client
- IP address - Subnet Mask - Default Gateway		
- User name		
- Password		
- Client name		
- Client ID or MAC address (as required by ISP)		

- Optional:
  - Port forwarding configuration details (i.e., list of IP address and port number for services on internal network that are to be made available to external network)

Plan IP Address Requirements

You need IP addresses for

- The CXi controller
- Each IP phone (a range of IP addresses assigned by the DHCP Server or statically assigned)

- A router or gateway (if using)

The controller uses the following reserved IP addresses:

192.168.10.1 - 192.168.10.255
192.168.11.1 - 192.168.11.255
192.168.12.1 - 192.168.12.255
192.168.13.1 - 192.168.13.255

Ensure no other devices on the network use IP addresses within that range.

IP Network Configuration Options

Choose between three IP network configuration options:

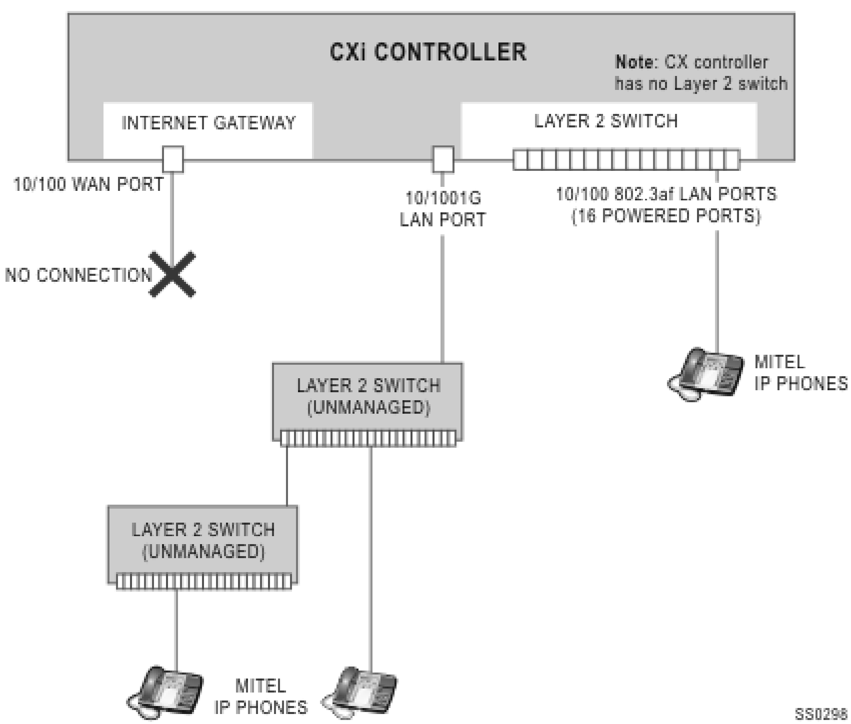
- Voice Only: The CXi provides services for IP phones on a private network.
- Voice and Data (LAN interface): The CXi provides services for IP phones and PCs on a private network.
- Voice and Data (WAN interface): The CXi provides services for IP phones and PCs on a private network, plus a firewall-protected connection to the Internet.

Voice Only Network

The configuration is intended for new implementations, where the voice network is physically separated from the data network. This configuration uses the internal LAN interface (Layer 2 switch) of the CXi. The WAN interface is left disconnected from the Internet.

To implement the system, simply plug your IP phones into the LAN ports of the CXi; the default IP network settings will enable the phones to function.

As a standalone system, the CXi supports up to 16 users. More users (maximum 100) can be added by connecting expansion Layer 2 switches to the CXi.



Configuration Procedure (Optional)

- Updating the System IP Information (required only if the Default IP Network Settings are unsatisfactory)

Notes:

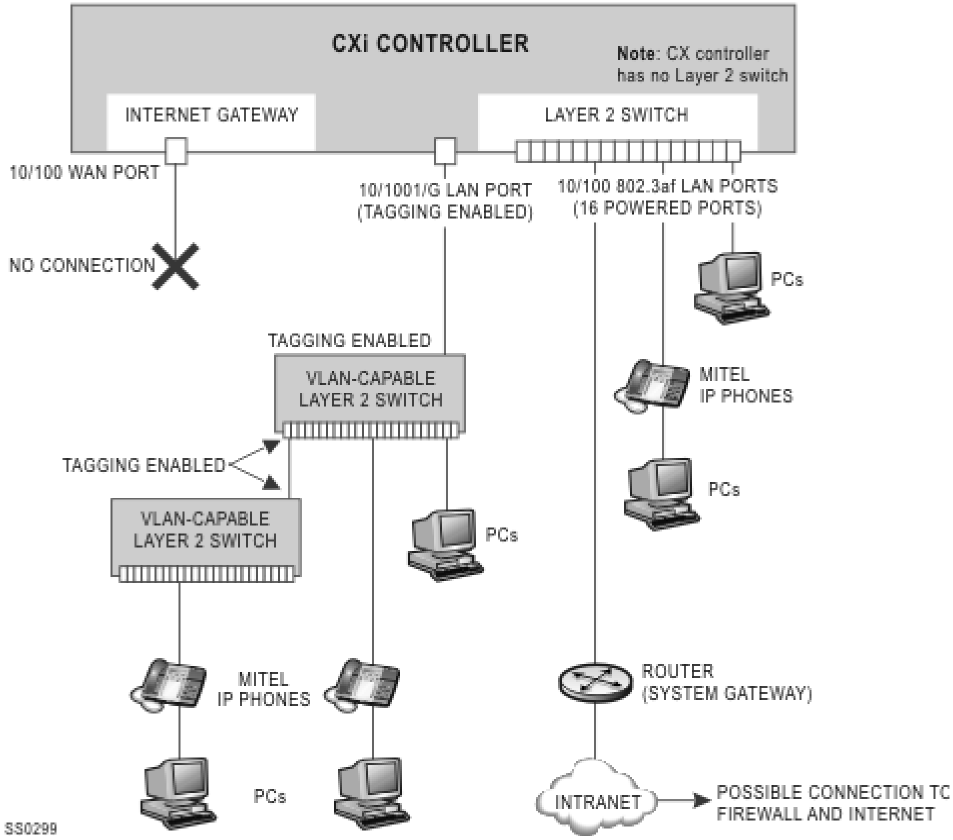
1. If you are Updating the System IP Information , make sure that the System Gateway IP and the Layer 2 Switch IP are identical.
2. Up to two expansion switches can be connected to the CX/CXi. For configuration instructions, see Install a Layer 2 Switch for Line Expansion (Voice Only).
3. The WAN port can be enabled to allow external management access. For configuration instructions, see Establishing a Management Connection from Internet.

Voice and Data Network (LAN Interface)

This configuration is intended for implementations where voice and data traffic is combined on the internal IP network. This configuration uses the internal LAN interface (Layer 2 switch) of the CXi. The WAN interface is left disconnected from the Internet.

Depending on the existing network configuration, programming changes may be required to prevent IP address conflicts, and to allow traffic to be routed between the local and remote subnets. After programming is complete, up to 16 users, each with an IP phone and a PC, can be connected to the CXi. System capacity can be increased to a maximum of 100 users by adding expansion switches.

In this configuration, the CXi uses VLAN prioritization to ensure quality of service for voice calls. Because packets from Mitel IP phones are assigned a high VLAN priority, the internal Layer 2 switch of the CXi routes them ahead of packets from other devices, which are assigned a low priority. Most expansion switches will do the same, provided that they are VLAN-capable. VLAN information (ID and priority) is distributed to Mitel IP phones by the DHCP server. The system can be programmed to perform prioritization on the default VLAN, or on a separate Voice VLAN.



Configuration Procedures

- Updating the System IP Information (required only if the Default IP Network Settings are unsatisfactory)
- Programming the DHCP Server (Local or Remote)
- Programming the Internal Layer 2 Switch
- Programming Router Controls
- Programming the Network List
- Enabling the (2nd) Port on IP Phones

Notes:

1. Up to two expansion switches can be connected to the CX/CXi. For configuration instructions, see Install a Layer 2 Switch for Line Expansion (Voice and Data).
2. Dual-port phones use the same port speed as the connected PCs. For this reason, PCs with 100 Mbps Ethernet cards are recommended.
3. This configuration supports the connection of IP trunks through the LAN interface, provided that NAT is not used on the LAN.
4. This configuration can provide a connection to other networks, including the Internet, with the addition of a router on the LAN.
5. The WAN port can be enabled to allow external management access. For configuration instructions, see Establishing a Management Connection from Internet.

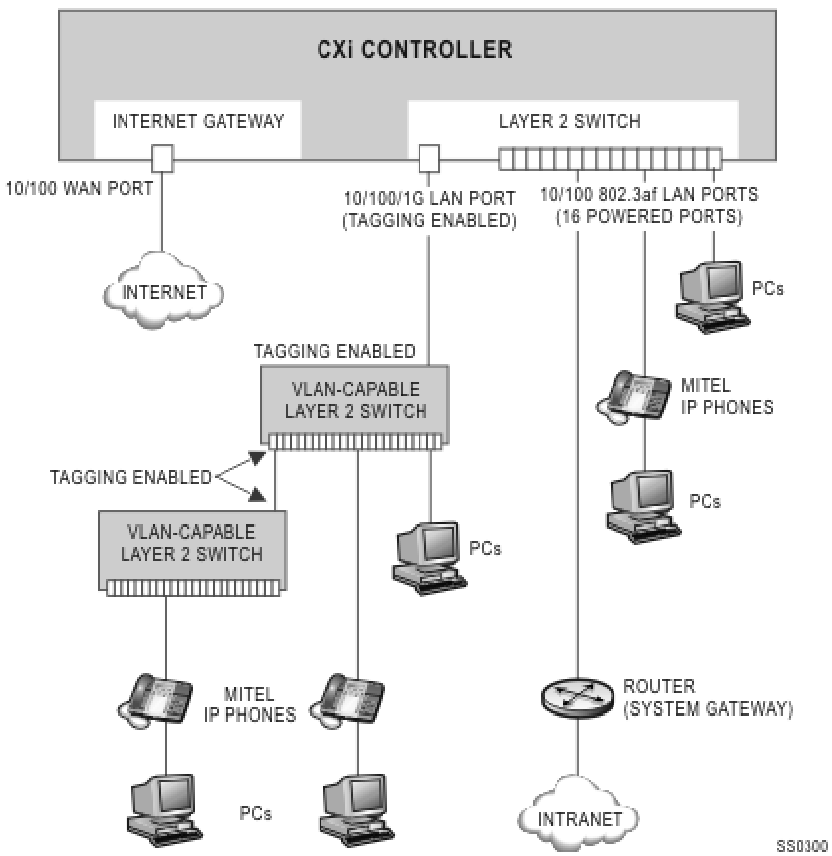
Voice and Data Network (WAN Interface)

With this configuration, the CXi provides connectivity to the internal voice and data network though its LAN interface (Layer 2 switch), and to the Internet through its WAN interface (Internet Gateway).

The primary function of the Internet Gateway is to link the internal and external networks. In this role, it performs many-to-one NAT, converting private IP addresses on the LAN to a single public IP address on the WAN interface. IP Port Forwarding is included as a programmable feature, enabling external traffic to reach internal services or machines. The Internet Gateway can also provide firewall functionality, logging unknown packets and then either dropping or rejecting them. If VPN tunnels are in use, the Internet Gateway can perform IPSec and PPTP pass-through for outbound connections, and can function as a PPTP server for a single inbound client.

IP phones and PCs for 16 users can be connected to the LAN ports of the CXi. System capacity can be increased to a maximum of 100 users by adding expansion switches.

In this configuration, the CXi uses VLAN prioritization to ensure quality of service for voice calls. Because packets from Mitel IP phones are assigned a high VLAN priority, the internal Layer 2 switch of the CXi routes them ahead of packets from other devices, which are assigned a low priority. Most expansion switches will do the same, provided that they are VLAN-capable. VLAN information (ID and priority) is distributed to Mitel IP phones by the DHCP server. The system can be programmed to perform prioritization on the default VLAN, or on a separate Voice VLAN.



Configuration Procedures

- Updating the System IP Information (required only if the Default IP Network Settings are unsatisfactory)
- Programming the DHCP Server (Local or Remote)
- Programming the Internal Layer 2 Switch
- Programming Router Controls

- Programming the Network List
- Programming the Internet Gateway (WAN Interface)
- Programming the Firewall
- Programming Port Forwarding
- Programming PPTP Remote Access
- Enabling the (2nd) Port on IP Phones

Notes:

1. Up to two expansion switches can be connected to the CX/CXi. For configuration instructions, see Install a Layer 2 Switch for Line Expansion (Voice and Data).
2. Dual-port phones use the same port speed as the connected PCs. For this reason, PCs with 100 Mbps Ethernet cards are recommended.
3. IP trunking is supported through the LAN interface, provided that NAT is not used on the LAN. IP trunking is not supported through the WAN interface because it performs NAT.
4. This configuration can provide a connection to other subnets on the LAN with the addition of a router.
5. The Internet Gateway of the CXi does not provide web or email content filtering, or antivirus protection. Purchase the 6040 Office Server Suite to perform these services.

IP Network Configuration Procedures

Updating the System IP Information

Use this procedure to view and update the basic system information for the CXi. These settings apply to the LAN interface of the CXi.

1. Go to form 47 and select subform 01, System IP.
2. Program:
    - **System IP Address**: This address provides call control functionality for IP phones and management access to the system through a CDE Maintenance session. This address must be on the same subnet as the Layer 2 Switch IP Address, and must not conflict with the DHCP Server configuration (static or range address). Default is 192.168.1.2.
    - **System Netmask**: Enter the network mask of the system IP address. Default is 255.255.255.0.
    - **System Gateway IP**: Enter the IP address of the default gateway to the Internet. If there is only one subnetwork on the LAN, enter the IP address of the CXi's Layer 2 Switch, or the IP address of the APC, if installed. If there are multiple subnets, enter the IP address of the router connected to the local subnet. In all cases, DHCP server option 3 should be the same as the System Gateway IP to ensure that all DHCP clients use the proper default gateway. Default is 192.168.1.1.
    - **L2 Switch IP Address**: Enter the IP address of the Layer 2 switch on the LAN interface of the CXi. This address must be on the same subnet as the System IP Address, and must not conflict with the DHCP Server configuration. Default is 192.168.1.1.
    - **Hostname**: Enter the system hostname, which is required for e-mail notification. It must be a valid domain name, 49 characters or less, that has been registered in your DNS or listed in your e-mail server's Hosts file.
    - **FTP Server**: Enter the IP address of the FTP server used to install software in the CXi controller or to upload Maintenance logs.
    - **FTP Username**: Enter the name used to access the FTP server.
    - **FTP Password**: Enter the password used to access the FTP server. Required entry.
    - **DiffServ Code Point:** Enter the Differentiated Services Code Point Value (0-63) for voice streaming and signaling. This value should match the value programmed in the dscp tag of DHCP option 125 or 43. The default is 44 (for upgrades) or 46 (for new installations).

- **Voice VLAN ID:** Enter the voice VLAN membership number (1-4093) of the embedded L2 switch ports. This value should match the Voice VLAN ID programmed on external L2 switches as well as the value programmed in the vlan tag of DHCP option 125 or 43. The default voice VLAN ID is 1.
- **Voice VLAN Priority:** Enter the voice VLAN priority (0-7) for expedited forwarding of traffic. The controller will apply the voice priority to traffic originating from the System IP address. This value should match the Voice VLAN priority programmed on external L2 switches as well as the value programmed in the l2P tag of DHCP option 125 or 43. The default voice VLAN priority is 6.

Notes:

1. Reset the system after changing System IP Address, System Netmask, L2 Switch IP, or System Gateway IP. If you program a new System IP Address, use it to access the system over Ethernet.
2. The System IP Address and Netmask must not be in the subnet range specified in Form 47, DHCP Parameters.
3. IP configuration settings (IP addresses and FTP server address) return to their default values when an initial software installation is performed.
4. IP configuration data is retained when moving the internal media (CompactFlash or hard drive) and System ID or i-Button from one controller to another.

Programming the DHCP Server

Using the Local DHCP Server

Use this procedure to program the LAN interface on the AX or CX/CXi to support a DHCP address range, or up to 11 static IP addresses. Once the programming is complete, IP phones and PCs located on the same LAN subnet as the AX or CX/CXi will be able to obtain IP addresses from the DHCP server.

The internal DHCP server is shipped enabled on the AX or CX/CXi, with options that apply globally to all DHCP subnets, address ranges and static IP addresses. For a list of default settings, see CX/CXi Controller Internal DHCP Server Defaults.

Note: It is recommended that only one DHCP server be enabled on the internal network. Make sure that it is programmed to provide addresses and options to all devices that require them.

Add the DHCP address range or static IP address

1. Go to form 47 and select subform 02, DHCP Server.
2. Press the ADD softkey and program:
   - **SUBNET NAME**: Enter the name of the DHCP subnet.
   - **IP ADDRESS**: Enter the IP address of the subnet served by the DHCP server.
   - **BIT MASK**: Enter the subnet mask of the subnet served by the DHCP server.
3. Select the new DHCP subnet; then press the EXPAND softkey.
4. Select the NEW RANGE softkey to add a DHCP address range, or select NEW STATIC to add a static address.

   ADDRESS RANGE:
   - **Name**: The name of the DHCP address range.
   - **Range Start**: The starting IP address.
   - **Range End**: The ending IP address.
   - **Clients Class ID must match name**: Yes or No (default).
   - **Lease Time (minimum 30 minutes)**: Lease time in minutes, hours, days, weeks, or months.

   STATIC IP:
   - **Name**: The name of this DHCP client.
   - **IP address**: The IP address.

- **MAC Address or Client ID**: MAC address or Client ID of this DHCP client. Do not enter both values.

Program the DHCP common (global) options

1. Go to form 47 and select subform 02, DHCP Server.
2. Press the COMMON OPT softkey and program:

Option	Value Required
3-Router	Enter the IP address of the default gateway to the Internet, which can either be the CXi itself (L2 Switch IP), a router on the LAN, or the Application Procesor Card (if installed).
6-Domain Name Server	

For an existing network that already has DNS server(s), enter their IP addresses (use a comma or space to separate multiple entries).

For a new network, enter the L2 Switch IP Address of the CXi, which will resolve names in the Internet DNS system provided that the WAN port is enabled.
43-Vendor Specific

ASCII configuration string. DHCP options 128 - 133 are replaced by tags included in an ASCII string.

Select the MITEL or NON-MITEL softkeys to add DHCP options by manually creating the configuration string.
125-Vendor Specific

ASCII configuration string. DHCP options 128 - 133 are replaced by tags included in an ASCII string.

Select the MITEL softkey to add Mitel-specific options into a form that creates the ASCII configuration string. Select the NON-MITEL softkey to add non-Mitel options by manually creating the configuration string. The default string for this option is: id:ipphone.mitel.com;sw_tftp=192.168.1.2;call_srv=192.168.1.2;vlan=1;l2p=6

DHCP Options prior to Release 4.0:	
128-IP Phone TFTP Server	System IP Address of the controller.
129-IP Phone service provider	System IP Address of the controller.
130-Mitel Identifier	MITEL IP PHONE (all caps).
132-VLAN ID	VLAN ID for the voice VLAN (32 bit word)
133-VLAN Priority	VLAN Priority (values are 1 - 7; Mitel recommends 6)
134-DiffServ Code Point	44 (upgrades) 46 (new installations)

Notes:

1. In addition to programming COMMON OPTions that apply to all DHCP subnets, you can also program SUBNET OPTions for a subnet, RANGE OPTions for an address range, or STATIC OPTions for a static IP address.
2. Prior to Release 4.0, Options 132 and 133 control VLAN tagging for voice packets (packets from Mitel IP phones), and are required for voice and data implementations. If only one VLAN is required, use the default values. However, if a second "voice" VLAN is implemented, program its unique VLAN ID and Priority. These values must match the Voice VLAN settings programmed on Form 47, Subform 1 (System IP) and on the external L2 switches. For Release 4.0 and later, VLAN tagging for voice packets is controlled in the Option 125 (or Option 43) data string using the "vlan" and "l2P" tags.

Using a Remote DHCP Server

Use this procedure to program an external DHCP server to support a DHCP address range. Once the programming is complete, IP phones and PCs will obtain IP addresses from the external DHCP server rather than the CX/CXi controller.

The Managed Application Server can be configured as a DHCP server; it provides Mitel telephone DHCP options on its IP Phone Support panel.

Note: It is recommended that only one DHCP server be enabled on the internal network. Make sure that it is programmed to provide addresses and options to all devices that require them.

1. Go to form 47 and select subform 02, DHCP Server.
2. Disable the internal DHCP server by pressing the DISABLE or DELETE softkey for each DHCP subnet.
3. Add the following options to the external DHCP server:

ID	Name	Format	Value	Scope
3	Router	IP Address	Enter the IP address of the default gateway to the Internet, which can either be the CXi itself (L2 Switch IP), a router on the LAN, or the Application Procesor Card (if installed).	Global
6	Domain Name Server	IP Address		

For an existing network that already has DNS server(s), enter their IP addresses (use a comma or space to separate multiple entries).

For a new network, enter the L2 Switch IP Address of the CX/CXi, which will resolve names in the Internet DNS system, provided that the WAN port is enabled.	Global	
43	Vendor Specific (New for Release 4.0)	ASCII string

ASCII configuration string. DHCP options 128 - 133 are replaced by tags included in an ASCII string.

Select the MITEL or NON-MITEL softkeys to add DHCP options by manually creating the configuration string.		
125	Vendor Specific(New for Release 4.0)	ASCII string

ASCII configuration string. DHCP options 128 - 133 are replaced by tags included in an ASCII string.

Select the MITEL softkey to add Mitel-specific options into a form that creates the ASCII configuration string. Select the NON-MITEL softkey to add non-Mitel options by manually creating the configuration string. The default string for this option is: id:ipphone.mitel.com;sw_tftp=192.168.1.2;call_srv=192.168.1.2;vlan=1;l2p=6				
DHCP Options prior to Release 4.0				
128	TFTP Server	IP Address	Enter the System IP Address of the controller.	Global
129	IP Phone service provider	IP Address	Enter the System IP Address of the controller.	N/A
130	DHCP Server Identifier	ASCII	MITEL IP PHONE (all caps)	Global
132	VLAN Identifier	Hex Long	Enter the VLAN ID for the voice VLAN (32 bit word)	Global
133	VLAN Priority	Hex Long	Values are 1 - 7; Mitel recommends 6	Global
134	Differentiated Services Code Point Value	Numeric	Values are 0 - 63. Program 44 for upgrades and 46 for new installations.	Global

4. Upgrade the IP Phones/firmware to 2.0.0.18 or later.
5. Program your DHCP server with the following new DHCP options:

   - 43 Vendor Specific Information (for non Mitel-specific options, or for Mitel-specific options that you will enter manually.)

   OR

   - 125 Vendor Specific Information (for Mitel-specific options that you will enter using a form.)

Data string format is as follows:
id:ipphone.mitel.com;sw_tftp=10.37.189.11;call_srv=10.37.189.11,10.37.190.11;vlan=1056;l2p=6;dscp=46

6. Delete Options 128 -133.

Note: IP sets require a firmware upgrade to support the new DHCP options introduced in Release 4.0; otherwise the sets will fail to boot. For the sets to boot initially, DHCP options 128 -133 must be present in the DHCP server. After the sets have booted, options 128 - 133 may be removed to avoid future conflicts with standard use or with other vendors' use of these options.

Note: Prior to Release 4.0, Options 132 and 133 control VLAN tagging for voice packets (packets from Mitel IP phones), and are required for voice and data implementations. If only one VLAN is required, use the default values. However, if a second "voice" VLAN is implemented, program its unique VLAN ID and Priority. These values must match the Voice VLAN settings programmed on Form 47, Subform 1 (System IP) and on the external L2 switches. For Release 4.0 and later, VLAN tagging for voice packets is controlled in the Option 125 (or Option 43) data string using the "vlan" and "l2P" tags.

Programming the Internal Layer 2 Switch (CXi)

Use this subform to program the global settings for the managed Layer 2 Ethernet switch. The first 16 ports are the powered 10/100Base-T/TX LAN ports, and the 17th port is the unpowered 10/100/1GBase-T LAN port. The powered ports are designed for connection of IP phones but can also support PCs. The unpowered port is designed for connection to another Layer 2 switch, but it can also support an IP phone or PC.

The internal switch uses VLAN prioritization to ensure the quality of voice calls, routing packets from IP phones ahead of data packets. Programming is required only if an expansion switch is connected. For details, see Install a Layer 2 Switch (Voice and Data).

1. Go to form 47 and select subform 03, L2 Switch.
2. Program:
   - **Tag VLAN 1 on Trunk Port (17)**: Select the ENABLE softkey to enable 802.1p/Q VLAN tagging for VLAN 1 on the 10/100/1G LAN port. Enable VLAN tagging on the port only if a VLAN-capable switch is connected to the port. By default, this parameter is disabled all VLAN tags are removed before being forwarded by the switch.

Note: This field applies to untagged and VLAN 1 tagged packets only; voice VLAN tagging (if enabled) is always preserved. Program the Voice VLAN ID and Priority on the System IP Information form.

   - **IGMP Snooping:** The purpose of IGMP (Internet Group Multicast Protocol) snooping is to restrain multicast traffic in a switched network. When the feature is enabled (the default), multicast frames are sent only to devices that request them. When the feature is disabled, multicast frames are sent to all devices. Disabling IGMP snooping may cause problems with network or device performance.

   Rapid Spanning Tree (RSTP): The Rapid Spanning Tree Protocol (RSTP) enables loop-free, redundant bridging paths by controlling the forwarding state of links between switches. Wherever two paths to a given destination exist, RSTP eliminates the possibility of a loop by forwarding on only one of the paths. The other path is used only in the event of a link or interface failure. Enable RTSP if multiple Ethernet connections are being used to connect the internal L2 switch to external IP device. Disabled by default.

   STP Bridge Priority [0-15] (*4096): Enter the Spanning Tree Bridge priority. The priority is a step value of 0-15 which results in a multiple of 4096 (0x1000) from zero to fifteen (0-15). A higher value means lower priority and less chance of becoming the Root Bridge in the Spanning Tree network. The default is 15 (61440). Note: The user should fully understand RSTP before updating this parameter.
3. To update the Layer 2 switch port settings, press the ETHER PORTS softkey and program:
   - **State**: Enables / disables the port. All ports are enabled by default.
   - **Speed**: Select the port speed - AUTO (default), 10, or 100. For 1000 Mb (gigabit) speed operation on port 17, set the speed to Auto, and it will auto-negotiate for the gigabit speed. All ports default to AUTO.
   - **Duplex Mode**: Select the duplex mode - AUTO DUPLEX (default; mode negotiated between the two endpoints), FULL DUPLEX (allows simultaneous transmission in both directions), HALF DUPLEX (transmits in one direction at a time). All ports default to AUTO DUPLEX.
   - **Flow Control**: Select the flow control mode - AUTO FLOW (default; flow control negotiated between the two endpoints), ENABLE FLOW (forces the port to use flow control), DISABLE FLOW (disables flow control for the port). All ports default to AUTO FLOW.
   - **Power over Ethernet**: The PoE mode, either AUTO (automatically negotiated) or disabled. Ports 1 to 16 default to AUTO. Port 17 does not support PoE. PoE can be left enabled even if the

connected device does not require PoE support. For diagnostic information, see the PoE Status report in Maintenance.

Notes:

1. There is a slight delay before configuration updates take effect. For example, the port will continue to receive power for 10 to 15 seconds after Power over Ethernet is disabled.
2. Use AUTO for Speed, Duplex, and Flow Control; program manual settings only if autonegotiation fails. For port 17 (the 1G port), Speed can be set to 10, or 100, or AUTO (1000 Mbps is always autonegotiated).

Programming Router Controls

If the LAN is divided into multiple segments, the CXi can be programmed as an ICMP router discovery server. In this role, it exchanges "ICMP Router Discovery" messages with hosts on the network. This process enables the hosts to dynamically discover the CXi and other routers on the network, and to switch to a backup router in the event of a network failure.

In addition to enabling the ICMP router discovery server, it may also be necessary to configure the CXi with a Network List containing routes to destination networks on the LAN.

1. Go to form 47 and select subform 04, Routing.
2. Program:
   - **ICMP ROUTER DISCOVERY SERVER:** Enables / disables the ICMP Router Discovery server on the LAN interface of the CXi. When enabled, the server periodically sends router advertisement messages to the network. It also responds to router discovery messages sent from hosts. Enabled by default.

Programming the Network List

The Network List consists of routes to destination networks on the LAN. When the CXi receives a packet from a host on the LAN, it checks the table. If a route is found, the CXi forwards the packet to the system gateway, which is either the Cxi itself (i.e, when System Gateway IP = L2 Switch IP), the APC (MAS), or a router connected to the local subnet. If no route is found, the CXi forwards the packet to the Internet through its WAN interface.

The administrator can add up twenty routes to the Network List. The list includes three default routes to private subnets (RFC 1918 networks):

- 172.016.0.0 / 255.240.0.0
- 192.168.0.0 / 255.255.0.0
- 10.0.0.0 / 255.0.0.0

1. Go to form 47 and select subform 04, Routing.
2. Press the NETWORK LIST softkey. Each route consists of a destination IP address and netmask for a remote subnet on the LAN.
4. Press ADD to add a route to the list; press DELETE to remove a route from the list.

Programming the Internet Gateway (WAN interface)

The WAN interface can obtain an IP address by DHCP or PPPoE, or be programmed with a static IP address and default gateway. Alternatively, the WAN interface can be programmed to support the optional Application Processor Card (APC). The WAN interface performs many-to-one NAT.

1. Go to form 04 and enable system option 83 (Internet Gateway).
2. Go to form 47 and select subform 05, Internet Gateway.
3. Enable the WAN State.
4. Select a WAN IP Method by pressing the PPPoE, DHCP CLIENT, STATIC IP, or APC softkey. Default is DHCP CLIENT. If APC is selected, the APC (Managed Application Server software) will

function as the Internet Gateway instead of the WAN interface of the SX-200 ICP. Program the Internet connectivity settings (static IP, PPPoE, or DHCP client) on the MAS.

5. Program a WAN IP Method by pressing the PPPoE_CFG, DHCP_CFG, or STATIC_CFG softkey.

   PPPoE Configuration:
   - **User name**: Enter the PPPoE user name specified by your ISP. The characters ', ", `, and (space) are invalid.
   - **Password**: Enter the PPPoE user name specified by your ISP. The characters ', ", `, and (space) are invalid.

   DHCP Client Configuration:
   - **Client name**: Enter the client name specified by your ISP.
   - **Client ID / Ethernet Address**: Enter the client ID or Ethernet (MAC) address specified by your ISP.
   - **Password**: Use the softkeys to select an authentication method, either Ethernet or Client ID.

   Static Configuration:
   - **IP Address**: Enter the static IP address of WAN interface.
   - **Subnet Mask**: Enter the subnet mask of WAN interface.
   - **Default Gateway**: Enter the external default gateway address.

Notes:

1. IP trunking is not supported through the WAN interface, or through any other interface that performs NAT.
2. All three WAN IP Methods (PPPoE, DHCP, and Static IP) can be programmed, but only one WAN IP Method can be enabled.
3. If the WAN interface is programmed as a DHCP client, it may be necessary to renew the DHCP lease to obtain an IP address. To do this, disable the WAN State and then enable it.
4. Speed, duplex mode, and flow control are automatically negotiated by the WAN interface. If manual settings are required, install a programmable switch between the cable modem/DSL router and the WAN interface.
5. If PPPoE or DHCP address assignment fails, the system generates a MAJOR alarm that is recorded in the maintenance log. Assigning an IP address to the WAN interface clears the alarm.
6. If the CXi contains the optional APC (MAS), enable the WAN State and select APC as the WAN IP method. Program the Internect connectivity settings (static IP, PPPoE, or DHCP client) on the MAS.

Programming the Firewall

The firewall examines all packets destined for the internal network. Unless they are part of an existing connection, or match specific TCP or UDP port programmed for forwarding, the packets are declared "unknown" and then either dropped or rejected. Unknown packets are also logged.

The firewall can be configured to allow outbound VPN tunnels by enabling PPTP and IPSec pass-through, and inbound connections to the LAN by programming Port Forwarding.

1. Go to form 47 and select subform 06, Firewall.
2. Program:
   - **Logging State**: Select the ENABLE softkey to enable logging of unknown packets. Enabled by default.
   - **PPTP Pass-through**: Select the ENABLE softkey to enable outbound PPTP VPN tunnels to pass through the firewall from the LAN to the Internet. Enabled by default.
   - **IPSEC**: Select the ENABLE softkey to enable IPSec outbound VPN tunnels to pass through the firewall from the LAN to the Internet. Enabled by default.
   - **Action for Unknown Packets**: Press a softkey:
   - **DROP**: Unknown packets are discarded without sending a reply to the sender. Default setting.
   - **REJECT**: Unknown packets are discarded and the firewall sends an “ICMP port unreachable“ error message to the sender.
   - **TCP do not log ports**: Program a list of TCP ports that do not generate logs when they receive packets. By default, TCP ports 135, 137-139 are not logged.

- **UDP do not log ports**: Program a list of UDP ports that do not generate logs when they receive packets. By default, UDP ports 137-139, 520 are not logged.

To obtain logs, see Sending Logs to an E-mail Address or FTP Server.

Programming Port Forwarding

Use this procedure to program the firewall forwarding table to allow external traffic to reach resources on the internal network. The firewall forwarding table can contain up to 40 entries. Each entry consists of a protocol (TCP or UDP) and port number combination on the Internet Gateway (WAN interface), and an IP address and port number combination on the internal network. Ports can be entered a single number or as ranges of numbers. Port Forwarding is also known as NAT Redirect. See IP Port Usage for a list of port numbers used by the SX-200 ICP.

1. Go to form 47 and select subform 06, Firewall.
2. Press the FIREWALLTBLE softkey.
3. Press the ADD softkey.
4. Select the WAN interface protocol by pressing the TCP or UDP softkey, and then program:
   - **Start Port**: Specify the start of the port number range on the WAN interface.
   - **End Port**: Specify the end of the port number range on the WAN interface.
   - **Internal IP**: Enter the IP address on the internal network to which the external packets are sent.
   - **Int. Start Port**: Specify the start of the port number range on the LAN interface.
   - **Int. End Port**: Specify the end of the port number range on the LAN interface.

Notes:

1. Each protocol/port number combination on the WAN interface must be unique.
2. Some protocols (for example, FTP) cannot be forwarded with IP Port Forwarding.
3. The size of the port number ranges on the two interfaces must match. For example, if the WAN interface has a three-number range, then the LAN interface must have a three-number range.

Programming PPTP Remote Access

Use this procedure to program the Internet Gateway as a PPTP server for a remote client on the Internet.

1. Go to form 47 and select subform 07, PPTP REMOTE ACCESS.
2. Program:
   - **Dial In Username**: Username that the server uses to authenticate the remote client. The characters ', ", `, and (space) are invalid.
   - **Dial In Password**: Password that the server uses to authenticate the remote client. Use a combination of upper and lower case letters, punctuation, and numbers (for example, luv2sk!). The characters ', ", `, and (space) are invalid.
   - **PPTP LAN Client IP Address**: IP address that the remote PPTP client uses on the LAN. The IP Address must be on the same subnet as the System IP address. It cannot conflict with System IP address, L2 Switch IP address, and DHCP Server configuration. If PPTP Access is enabled, the IP address cannot remain blank or zero. The default is blank.
   - **PPTP State**: Enables / Disables the PPTP server. Disabled by default.

Enabling the (2nd) Port on IP Phones

Dual-port IP phones (5312, 5324, 5212, 5224, 5215, 5220, 5010, and 5020) support the connection of a single PC through their 2nd Ethernet port. The PC requires a 100Base-Tx Ethernet card.

IMPORTANT: To ensure optimum network performance, DO NOT connect servers to the 2nd port on IP phones.

1. Form 04, System Options/System Timers
   - Enable System Option 131, PC (2nd) Port on IP Phone.
2. Form 03, COS Define
   - Enable Option 280, PC (2nd) Port on IP Phone.

Networking Mitel IP-PBXs

IP trunks allow you to interconnect multiple Mitel SX-200 ICP, SX-200 IP Nodes and 3300 ICP systems in a Wide Area Network (WAN). The IP trunks carry voice and signal messages through the Ethernet switch to the WAN.

IMPORTANT: Do not attempt the procedures in this section until you have successfully completed the SX-200 ICP Advanced Installation and Maintenance Course.

The CX/CXi supports up to 16 IP trunks, the MX up to 30, on a single "virtual" IP Trunk card programmed in Bay 1 (default), Slot 6.

Notes:

1. The SX-200 ICP functions as an end node in a network, which means that it can be connected to only one other network node as shown in the following figure.
2. Do not use the WAN interface of the CX/CXi controller, or any other NAT-enabled interface, for IP trunking.
3. This configuration requires dedicated, managed links to the Internet and is intended for implementations with more than 24 IP Phones. For implementations with fewer than 24 IP Phones, cable or DSL links to the Internet can be used if the IP trunks are carried within IPSec VPN tunnels with traffic shaping features. The Mitel Managed VPN is recommended

Figure: IP Trunking Example (MX and CX controllers)

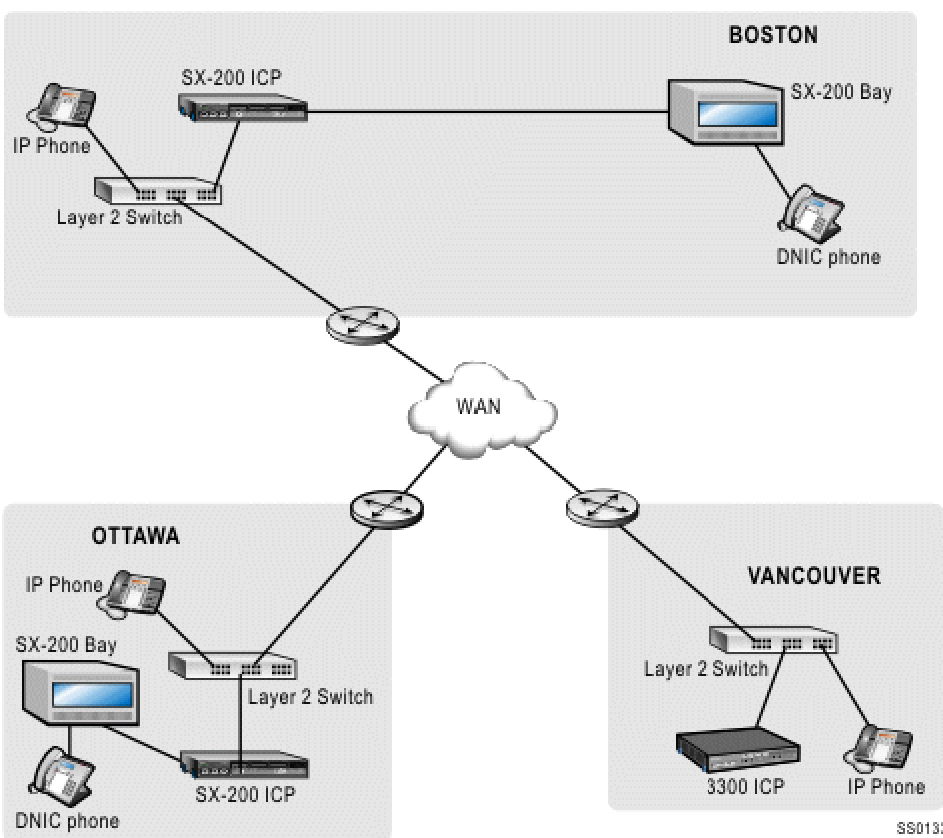
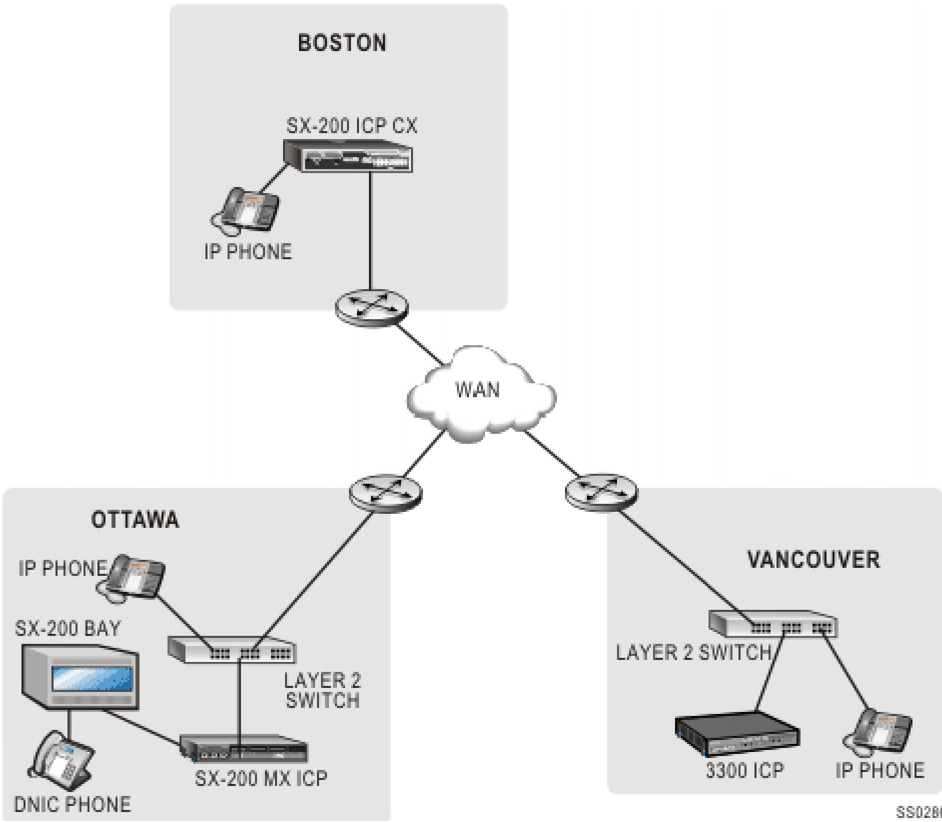


Figure: IP Trunking Example (CXi controller)



To program IP trunks:

1. Form 48, Voice Networking
   - Enter the IP Node Number and IP Address of each IP-PBX and specify the maximum number of calls allowed to each.

     The example below shows the Form 48 programming for a network of three IP-PBXs. Each IP-PBX (node) has a unique IP Node number that all the other IP-PBXs must have in their programming. The local site is always the IP bay (default is bay 1); the other sites have no Bay association.

   Notes:

   - MAX CALLS must be greater than zero (the default); otherwise, no calls are allowed to the node.
   - Each IP-PBX in the network must be assigned a unique IP Node Number, which can be any number in the range 1 - 255. (Note: A maximum of 100 IP nodes can be assigned.)

IP NODE NUM	IP BAY	IP ADDRESS	MAX CALLS	COMMENTS
1	1		20	Local Site
2		010.037.184.015	20	Branch Site 1 (3300)
3		010.037.024.210	20	Branch Site 2 (200ICP)

1-	2-	3-INSERT	4-	5-
6-QUIT	7-	8-	9-	0-

2. Form 03, COS Define
   - Enable the following COS options

     801, Incoming Trunk Call Rotary
     802, Limited Wait For Dial Tone
     811, ANI/DNI/CLASS Trunk.
   - It is also recommended that you enable COS option 702 to prevent calls from being denied access to trunks when the SMDR output is stopped. If you want to capture incoming SMDR, enable COS options 806 and 814.

     Note: COS number 7 has the default IP Trunk settings.
3. In Form 04, System Options/System Timers, enable option 86, PRI Card **-** QSIG, to allow Calling Name and Number to be transmitted over IP trunks.
4. In Form 13, Trunk Circuit Descriptors
   - Set a descriptor to T1 E&M in the Options subform. Descriptor 7 in the default database has the correct settings for IP trunks. If selecting another descriptor, set options to the default values, except:

     set Incoming Start Type to **WINK**
     set Outgoing Start Type to **WINK**
     set QSIG Supplementary Services to **YES**
     set DTMF to **NO**.
5. Form 15, Dial-in Trunks
   - In the CDN field, enter the descriptor number that defines the T1 E&M trunk specified in Form 13.
6. Form 16, Trunk Groups
   - All IP Trunks must be programmed in the same group. Incoming IP trunk calls will select the first trunk programmed. To avoid collisions with outgoing IP trunk calls, program the trunks in reverse order in the trunk group.
7. Form 23, Route Definition
   - Program route definitions
   - In the Show IP subform, specify the routes of IP Trunk Groups.
8. If you have not already done so, program the following Forms:
   - Form 26, ARS: Digit Strings.
   - Form 22, ARS: Modified Digit Table.
9. If the connection to the PSTN is through embedded PRI (Dual T1/E1 Framer or T1/E1 Combo module), program Calling Party Number (CPN) substitution in the following Forms

- Form 54, Calling Party Number
- Form 19, Call Rerouting Table
- Form 22, ARS: Modified Digit Table

Notes:

- In Form 22, for all SX-200 ICP controllers in the network, program a modified digit entry with CPN enabled for external calls (calls destined to the PSTN through the embedded PRI). Program another entry with CPN disabled for internal network calls.
- The SX-200 ICP controllers must be running R3.0 or later software to support CPN substitution with embedded PRI.
- If a 3300 ICP, NSU, or PRI card is the gateway to the PSTN, then CPN substitution is programmed in the normal manner using IMAT, and the SX-200 ICP controllers must be programmed with a modified digit entry with CPN disabled to access the PSTN.
- A 3300 ICP cannot send CPN substitution numbers to an SX-200 ICP that has embedded PRI and is the gateway to the PSTN.
- SMDR logs collected on the originating switch will capture the dialed CPN and the calling party's primary access code.

Uniform Numbering Plan

Optionally, you can configure the nodes in your network with a uniform numbering plan. For example, you can program nodes A, B, and C to share extensions 1000 to 1999.

To enable a uniform plan between nodes that are connected by IP trunks:

1. Set Present Node ID over IP Trunking to **YES** in the Options subform of Form 13, Trunk Circuit Descriptors.
2. Program the Node ID (Feature 34) in Form 02, Feature Access Codes.
3. On each node, program ARS leading digits containing node numbers for all other nodes in the network. This eliminates the need for users to insert digits when they return external calls from other nodes.

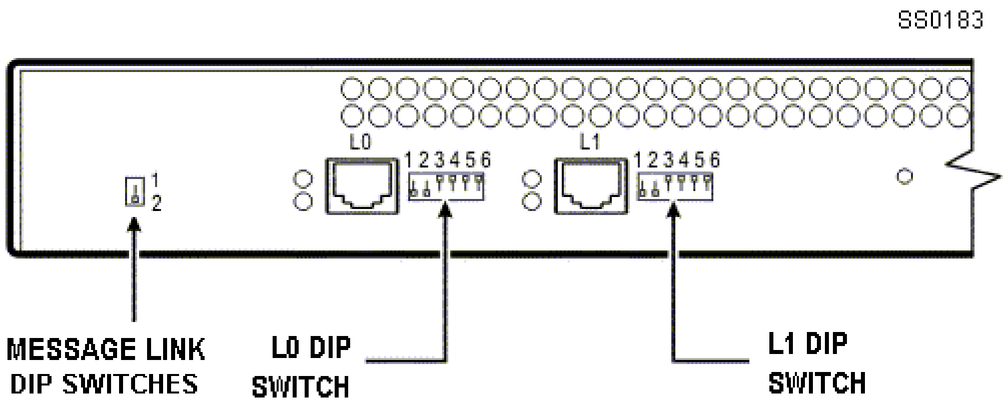
Installing an NSU

The SX-200 ICP supports the direct connection (via FIM or CIM) of four NSUs; the units cannot be daisy-chained.

1. Mount and secure the NSU in the desired location.

Note: Unlike the SX-200 ICP controller, the NSU is not wall-mountable. It is, however, rack-mountable.

2. Set the L0/L1 port DIP switches to the appropriate termination mode and impedance. The default is network termination mode.



3. Set both of the Message Link DIP switches to the down position. Note: The left switch is partially hidden by the rear panel of the NSU.
4. Connect a fiber optic cable between the NSU FIM and the controller FIM.
   OR
   Connect a Cat 5 Crossover cable between the left-hand CIM connector on the NSU and a controller CIM.

Note: The right-hand CIM connector on the NSU has been disabled.

5. Connect the NSU L0 and/or L1 port to the remote system (the PSTN or another system). See Pinouts for T1 Line/Network Termination for the pinout.
6. Install IMAT.
7. Connect power to the NSU.

Note: The NSU software will not load until the NSU programming is complete (see Step 8 and Step 9 below).

8. Using the CDE Forms, program the NSU settings on the controller; see Programming the NSU in CDE.
9. Program the NSU using IMAT.
10. After programming the NSU via both SX-200 ICP CDE and IMAT, plug the T1 cable from the demarcation point for the T1 provided by the Carrier to either L0 or L1 on the back of the NSU. Each connector (L0 or L1) has LED indicators beside the connector to indicate sync or not. For example:
    - Red LED indicates no sync (check connection or switch 6 is in wrong position).
    - Flashing green LED indicates synch but D-channel is not synchronized (check programming (see table below) on IMATs to ensure correct protocol).
    - Solid green LED indicates that D-channels and B-channels are all in sync and PRI trunks on NSU are ready to process calls.

Programming the NSU in CDE

1. Form 53, Bay Location Assignment
   - Assign a bay number to the CIM or FIM ports used to connect each NSU or PRI card to the controller.
2. Form 01, System Configuration
   - Program the bay(s) assigned in Form 53 as ISDN nodes.
   - Program a T1 Trunk Card in Slot 6 and 8 (assuming both spans will be programmed)
2. Form 03, Class of Service (COS)
   - Create a separate COS for ISDN trunks.
   - Enable the following COS options for the ISDN Trunks:

246 – SMDR – Extended Record
701 – No Dial Tone
702 – SMDR – Overwrite Buffer. (Required if option 806 SMDR – Record Incoming Calls enabled)
801 – Incoming Trunk Call Rotary
802 – Limited Wait for Dial Tone
811 – ANI/DNIS Trunk
814 – SMDR – Record ANI/DNIS

- For extensions (sets and stations), the following COS options must be set:

COS Option	Setting	Notes
236 – Outgoing Trunk Callback	Disable	
237 – Outgoing Trunk Campon	Disable	
502 – Display ANI/DNIS Information and possibly 613 – Display ANI information only	Enable	Must have 502 enabled for 613 to function.
503 – Enable Calling Name Display	Enable	
702 – SMDR Overwrite Buffer	Enable	Required if option 806 SMDR – Record Incoming Calls enabled

3. Form 04, System Options/System Timers
   - The following table shows the FOR options required for the NSU:

System Options	Setting	Notes
Option 48 – Limited Wait for Dial Tone	1-5 secs	Set to 1 sec. Default = 5 sec.
Option 96 – Number of Links	2	Purchase 1 for each link used. Default = 0
Option 86 - QSIG	Optional	Enabling Option 86 displays the caller's name on the called party's telephone. The option is not required by DMS100 Central Offices as the calling name is sent automatically.
Option 91 – NFAS	Optional	If Option 91 is to be disabled, Option 92 must be disabled first. Not required if using a single link PRI.
Option 92 – D-Channel Backup	Optional	If Option 92 is to be enabled, Option 91 must already be enabled. Not required if using a single link PRI
Option 94 – Min/Max	Optional	If Option 94 is to be disabled, Option 95 must be disabled first.
Option 95 – Auto Min/Max	Optional	If Option 95 is to be enabled, Option 94 must already be enabled.

4. Form 13, Trunk Circuit Descriptor

- Assign a T1 trunk circuit descriptor. To receive CLID or DDI digits, the PRI trunk must be a non-DISA trunk that is programmed with a T1 E&M trunk circuit descriptor. The Debounce timer must be set to 100 milliseconds and the Incoming Start type and Outgoing Start type must be set to WINK.

5. Form 13, Options Subform
   - Set the DTMF option to NO.
6. Form 15, Dial-In Trunks

   PRI trunks are Dial-In Trunks and are not normally used in a Non-Dial-in configuration
   - Search for the correct Bay/Slot/Circuit and enter the following for each trunk:

     COS number – see previous requirements
     COR number
     Tenant number – unless site has tenanting set to 1
     N – number of incoming DID digits – usually 4
     M – number of digits to absorb – usually 0
     X – digits to be inserted (up to 2 digits)
     CDN – Circuit Descriptor Number
     Trunk Number – Enter a trunk number (1-200)
7. Form 16, Trunk Groups
   - Locate an empty Trunk Group and add the PRI trunks (trunk numbers programmed in Form 15) to the Trunk Group. (Program them in reverse order to prevent contention with incoming calls from the telco)
   - Add a name to the Trunk Group for future reference.
   - Set Group Type to Terminal (recommended) or Cyclic.
   - Enable SMDR (outgoing only) if required.
8. Form 42, T1 Link Descriptor
   - Create a Link Descriptor for the PRI trunks using the recommended settings in the following table:

Alarm debounce timer (300 –3200 ms)	2500
Line Coding (AMI, AMI&ZCS, B8ZS)	B8ZS
Line Build Out (0, -7.5, -15. –22.5 dB)	0 dB
Line Length (max 132, 265, 398, 533 or 655)	266-398
Framing (D4 or ESF)	D4
Slip rate – maintenance limit (0 – 9000) /24 hrs	255
Slip rate – service limit (0 – 9000) /24 hrs	7000
Slip rate – network sync limit (0 – 9000) /24 hrs	7
BER – maintenance limit (10**-n , n = (3,4,5,6)) / hour	4
BER – service limit (10**-n , n = (3,4,5,6)) / hour	3
Framing losses – maintenance limit (0 – 9000) /24 hrs	255
Framing losses – service limit (0 – 9000) /24 hrs	9000
RTS timer – service limit exceeded (1- 255 min)	30
RTS timer – net slip limit exceeded (1- 255 min)	30
RTS timer – after alarm (0 – 300 sec)	10

9. Form 43, T1 Link Assignment
   - Assign the Link Descriptor to the PRI trunks (referencing the Bay and Slot location). For example, if the first CIM connector is used to connect to the NSU, the Link Descriptor is programmed against Bay 2, Slots 6 and 8.
10. Form 44, Network Sync
    - Enter the links according to their Bay/Slot/Circuit in the order that you want them to be used as the network sync source. Typically, COs are the first choice for a sync source. When using both PRI and TI trunks, make the PRI trunk the clock source.
11. Form 22, Modified Digit Table
    - DID Calling Party Number to the network interacts with the current SX-200 ICP system networking functionality and ignores programmed node-IDs in the NSU or PRI card. The Node ID Information Element ("*8") for Analog Networking, if programmed in the Modified Digit Table, will be ignored by the NSU or PRI card. If *6 is programmed in the digit modification table the DID calling extension number is sent to the NSU or PRI card and can be sent to the network as the calling party. The extension must belong to a block of DID numbers purchased from the Network provider.

    The following table shows how Analog Networking and Call-by-Call information co-exist in the Modified Digit Table

Entry	Quantity to Delete	Digits to be Inserted	Comments
01	1	*4000*6*4*04	ISDN CxC and DID

In the "Digits to be Inserted" field, the definition of characters are:

*4 No SMDR for further modified digits
000 Call-by-Call digits, (speech, default O/G, clid)
*6 Send caller ID (DID to network)
*4 Start SMDR again for further modified digits
*04= Inserts Caller's Name (for calls over ISDN Trunks)

12. Complete ARS programming
    - ARS programming must match with the service selection programming that is entered through the ISDN Maintenance Administration Tool (IMAT) computer.
13. IMAT Programming
    - If you are installing a PRI/QSIG NSU, program the NSU using IMAT software. See Programming with IMAT.

    Note: The NSU software will not load until the NSU programming is complete.
14. Continue with the final step of the procedure, Installing an NSU.

Programming with IMAT

The IMAT software is used to program hardware and call characteristics for the NSU. IMAT is also used to install software upgrades and back up the NSU database.

Notes:

1. IMAT is not needed to program Embedded PRI (Dual Link T1/E1 Framer MMC or T1/E1 Combo MMC) in the controller.
2. You must have IMAT software version 7.3 or greater.

Maintenance activities include access to the following maintenance information:

- a list of all software files and versions on the NSU
- log messages that contain a history of activities and the status of faults
- a database.

This section contains the following:

- Installing IMAT Software
- Using IMAT

You can install IMAT on the Maintenance PC or on its own PC (IMAT PC).

Note: If using a dial-up connection to the NSU, we strongly suggest that you use a non-networked computer (a Windows 95/98 networked computer may have difficulties communicating with Dial-up Networking).

Overview

1. Install IMAT on the PC.
2. Establish a direct cable, modem, or LAN connection to the NSU.

Install IMAT on the PC

1. Close all the applications running on the PC.
2. Insert the SX-200 ICP software CD-ROM in the CD drive.
3. Open the Tools folder, then the Disk 1 folder.
4. Double-click the Setup.exe file. The installation program starts. Follow the prompts to install IMAT.

Program IP Settings

Collect the following information:

- The Static IP Address that will be programmed on the NSU. Choose an IP Address that will not conflict with the IP Phone range, which starts at the default 192.168.1.20 and continues to 192.168.1.250. For example, you might choose 192.168.1.15 as the static IP address.
- The gateway address (if there is a gateway on your network)
- The subnet mask (the default setting is 255.255.255.0 which is represented in hexidecimal as ffffff00)

To program a static IP address on the NSU:

1. Ensure the NSU is disconnected from the SX-200 ICP Controller and that no power is applied to the NSU. Re-apply power only after completing CDE and IMAT programming.
2. If you have not already done so, connect a serial cable from a com port on your PC to the Maintenance port on the NSU.
3. Launch a Hyperterminal session on the PC (38400, 8, N, 1, no flow control).
4. Apply power to the NSU by connecting the power cord.

   Messages are displayed on the PC while the NSU is booting up.
5. When the PC displays "Press any key to stop this autoboot", press any key on the PC keyboard. If you do not stop the autoboot on time, remove the power cord from the NSU and repeat this step.

   The PC displays “[MC269AA Boot]”
6. Type ‘c’ on the PC keyboard.

   The PC displays "boot device".
7. Press the Enter key until “inet on ethernet” is displayed.
8. Enter the IP Address of the NSU and Subnet mask as shown in the following example:

   192.168.1.2:ffffff00 (where ffffff00 is the subnet mask 255.255.255.0)

Note: There are 6x f's and 2 x 0's

9. Press the Enter key until the PC displays “gateway inet (g):”
10. Enter the IP Address of the Gateway as shown in the following example:

    192.168.1.1

11. Press the Enter key until the PC displays “[MC269AA Boot]”
12. Enter "@" to continue the bootup or power down the NSU.

Connect a Windows PC to the NSU

Only the direct cable connection is described here. For information on connecting to the NSU via modem and or LAN, see the IMAT Online Help.

To Install a Direct Connection Device Driver in Windows

By default, Windows does not support a direct cable connection. You must add a device driver. Windows takes the information from a Mitel file and creates the driver called NT Direct Connection.

1. Use the instructions for your Windows operating system to install a new modem.
2. Browse to find the install file located at: c:\Program Files\Mitel\Imat.
3. Select a COM port.
   Windows creates a driver called NT Direct Connection.
4. Set the following parameters:
   - Maximum speed: 38 400
   - Only connect at this speed enabled
   - Data bits: 8
   - Parity: None
   - Stop bits: 1
   - Mode: Auto Answer (in Advanced Settings).
   - Error control: Clear if you have a direct connect cable.
   - Flow Control: Clear if you have a direct connect cable.
5. To connect the PC to the NSU, connect an RS-232 straight DTE serial cable between the NSU’s serial port and the PC’s serial port.

Note: If you are connecting through a modem, use a null modem adapter on the NSU side.

To create a Dial-Up Network Connection on Windows 95/98

You might need to do this procedure twice to create two Dial-up Networking connections: one for on-site direct access, and one for remote modem access.

1. Click the Start button, select Programs, Accessories, then Dial-Up Connections.
2. Double-click Make New Connection.
3. Enter an appropriate name for the connection (for example, Direct for direct connections, Remote or a customer’s name for remote connections) and click Next.
4. Enter an Area Code and Telephone Number and select a Country Code from the drop-down list. Click Next.

Note: Even though it is not needed for a direct connection, Windows requires that you enter this information.

5. Click Finish.
6. Right-click your new connection icon and click Properties.
7. Click Configure. Ensure the fields are set as follows:
   - Data bits: 8
   - Parity: none

Note: If you are creating a direct connection, make sure NT Direct Connection is listed in the drop-down list in the Make a New Connection window.

- **For a direct connection**:
- Maximum speed: 38400

- Check: only connect at this speed
- Select wait for dial tone before dialing
- Select cancel the call time at 60 seconds.
- Click Advanced and turn off error control and flow control
- **For a remote connection**:
- Stop bits: 1
- Click Advanced and turn on error control and select Compress data.
- Turn on flow control and select Hardware.

8. Click OK.
9. Select Server Types tab and make sure that PPP: Windows, WindowsNT3.5, Internet or PPP:Internet appears in the Type of Dial-Up Server field.
10. In the Advanced Options field, select Log onto Network and Enable software compression.
11. Make sure that only TCP/IP is selected in the Allowed network protocols field.
12. Select the Scripting tab and enter:
    c:\program files\mitel\Imat\pridun.scp.
13. Click OK.

To Create a Dial-Up Network Connection on Windows 2000:

You may need to follow this procedure twice to create two Dial-up Networking connections: one for on-site direct access, and one for remote modem access.

1. Click the Start button, select Programs, Accessories, Communications, then Dial-Up Connections.
2. Double click Make New Connection, and then click Next.
3. Select Dial-up to the Internet, and then click Next.
4. Select I want to set up my Internet connection manually, or I want to connect through a local area network (LAN). Click Next.
5. Select I connect through a phone line and a modem, and then click Next.
6. Use the COM Port that has been configured as a NULL Modem connection: 38400, 8, none, 1.
7. In the Choose Modem box, from the drop-down list select Communications cable between 2 computers. Click Next.
8. Clear the box Use area code and dialing rules, and then click Advanced.
9. For the Connection type, select PPP (Point to Point Protocol).
10. For the Logon procedure, select Use logon script, and then click Browse. Select the appropriate script c:\program files\mitel\Imat\pridun.scp, click OK, then Next.
11. In the Internet account logon information box, leave the username and password fields blank and then click Next.
12. Dialog boxes appear that warn you that you will not be able to connect to your Internet service provider without your user name and your password. Disregard these warnings and click Yes on these boxes to continue.
13. Enter the Connection name, and then click Next.
14. In the box to set up an Internet mail account, select No, and then click Next.
15. De-select the option to connect to the Internet immediately, then click Finish.
16. In the Network and Dial-up Connections window, right-click on the new DUN connection, point to Properties, then click Configure.
17. From the Maximum speed (bps) list, select 38400 for the baud rate.
18. Click OK until you exit the windows.

Start IMAT

1. Launch IMAT (ignore any message about username configuration error).
2. In the File menu, select Connect to Remote Site.
3. Choose the Connection Medium: Serial or Ethernet Network Card.
4. If you selected Ethernet Network Card, enter the IP Address of the Remote Site.

   If you selected Serial, select the connection name you entered when you created the dial-up network connection.
5. For the Ethernet Remote System, select PRI Card / Universal NSU.
6. Click Connect.

   IMAT displays a confirmation of the connection (for example "Connected over Ethernet").
7. Press Okay
8. In the File menu, select Load > Database > Sources > PRI Card /NSU. This loads the IMAT with the database from the NSU.
9. Press Load.

   IMAT provides a confirmation of the connection (for example "Database received")
10. Press Okay.
11. In the File menu, select Load > SW Versions > Sources > PRI Card /Universal NSU. This loads the IMAT with the Software version from the NSU.
12. Press Load.

    IMAT provides a confirmation of the connection (for example "Software Version received")
13. Press Okay.
14. Verify that the SW Version of the NSU matches the SW version listed in the Release Notes delivered with the latest SX-200 ICP load:
    - In the Maintenance menu, select Software versions. Verify the package version that is listed against the Release Notes.
    - If the package version and the Release Notes don't match, select SW Upgrade From IMAT (the NSU load is on the CD).
    - After the Upgrade is completed, select Remote Site Reset.
15. Program the NSU Config for the links provided by your PRI provider:
    - In the Config menu, select Site Options.
    - Confirm the System Type: Universal NSU.
    - Confirm the Connected Platform: SX200 EL/ML/ICP.
    - If the following options are required, enable them: Qsig, Network Side Interface.
    - In the Config menu, select PRI Link Characteristics.
    - Verify BOTH links for Protocol type: DMS250, DMS100, 4ESS, NI2, QSIG.
    - Verify BOTH links for Physical type: T1/CSU or T1/DSX1.
    - Verify BOTH links for Characteristics: line coding, line length, framing, invert data, invert D channel.
    - To get a view all of your settings, from the Config menu select View Database.

Save a Database

1. In the File menu, select Save, then Database.
2. In the Files box, type in the Destination of the database (if you have changed the database you can save it back to the PRI/Universal NSU and/or you can save a copy of the database to your PC) and give it a file name.
3. Click Save.

Note: If you save the database changes to the NSU, you must reboot the NSU. You can do this by selecting Maintenance -- Remote Site Reset. (This should be done after hours or when there is no traffic on the NSU.)

Convert a Database

If you are loading a database with a software version that is earlier than the current IMAT version, IMAT displays the Local Database Load window, which allows you convert the ISDN database to a newer version.

1. In the Local Database Load window, select the new database version from the Database Version drop-down menu.
2. Click OK.

Close IMAT

Always save any open databases before closing IMAT.

Upgrading or Re-installing NSU Software

Use the IMAT application to upgrade the software. For more information on upgrading releases, refer to IMAT online Help.

To upgrade an NSU to a new version of software or re-install the software:

1 Ensure that the NSU card is running by checking its status LEDs. See NSU LEDs.

2 Connect a computer to the RS-232 serial port of the NSU.

3 Run IMAT.

4 Click on File/Connect to connect to the card.

5 Download the software from Mitel Online or insert the software CD-ROM in the computer's CD-ROM drive.

6 Click on Maintenance/Software Upgrade and proceed with the software upgrade or re-installation.

7 Click on Maintenance/Remote Site Reset to reset the NSU.

Installing an ASU

The SX-200 ICP (MX) supports up to six offboard ASUs; the CX supports up to three (all must be programmed as node type ASU BAY). They can be connected with the controller running as there is no requirement to power down in order to make a connection.

Note: The ASU can be located up to 30 meters (100 feet) away from the SX-200 ICP.

1. Mount the ASU.

NOTE! If an ASU is configured on a Quad CIM in the CX/CXi platform, ports 1 to 3 of the Quad CIM must be used.

2. Connect a Cross-over Category 5 cable with RJ-45 connector to the CIM port on the ASU and a free onboard CIM port on the controller.
3. Complete telephony cabling for the ASU.
4. Connect power to the ASU.
5. Once the CIM link synchronizes, the CIM LEDs turn on. The SX-200 ICP detects the ASU and polls it to verify that the firmware revisions are up to date. If the firmware is out-of-date, the controller will

automatically upload the current firmware load. Once this has been completed, the ASU will begin functioning immediately.

6. Using the CDE forms, program the ASU as described below.

Programming the ASU in CDE

To program the ASU:

1. In CDE Form 53 - Bay Location Assignment, assign a bay number to the CIM ports used to connect the ASU to the controller.
2. In CDE Form 01 - System Configuration, program the bays assigned in step 1 as node type **ASU BAY**.
3. Program ONS devices in CDE Form 09 - Desktop Device Assignments (additional configuration in CDE Form 03 - COS Define, CDE Form 05 - Tenant Interconnection Table, CDE Form 19 - Call Rerouting Table, and/or CDE Form 30 - Device Interconnection Table is optional).

Installing an ASU II

The SX-200 ICP CX/CXi controller supports up to three ASU IIs via Quad CIM card(s). The MX controller supports up to six ASU IIs using a combination of CIM expansion ports and Quad CIMs. The ASU II is progammed as node type ASU BAY and can be connected with the controller running as there is no requirement to power down to make a connection.

To install the ASU II:

1. Mount the ASU II.

NOTE! If an ASU is configured on a Quad CIM in the MX platform, it must be configured in port 4 first and then in any of the other connectors. Failure to do so results in non-functional ASUs.

NOTE! If an ASU is configured on a Quad CIM in the CX/CXi platform, ports 1 to 3 of the Quad CIM must be used.

2. Connect a Cross-over Category 5 cable with RJ-45 connector to the CIM port on the ASU II and a free CIM port on the controller.
3. Complete telephony cabling for the ASU II.
4. Connect power to the ASU II.
5. Once the CIM link synchronizes, the CIM LEDs turn on. The SX-200 ICP detects the ASU II and polls it to verify that the firmware revisions are up to date. If the firmware is out-of-date, the controller will automatically upload the current firmware load. Once this has been completed, the ASU II will begin functioning immediately.
6. Using the CDE forms, program the ASU II as described below.

Programming the ASU II in CDE

To program the ASU II:

1. In CDE Form 53 - Bay Location Assignment, assign a bay number to the CIM ports used to connect the ASU II to the controller.

NOTE! If an ASU II is configured on a Quad CIM in the MX platform, it must be configured in port 4 first and then in any of the other connectors. Failure to do so results in non-functional ASUs.

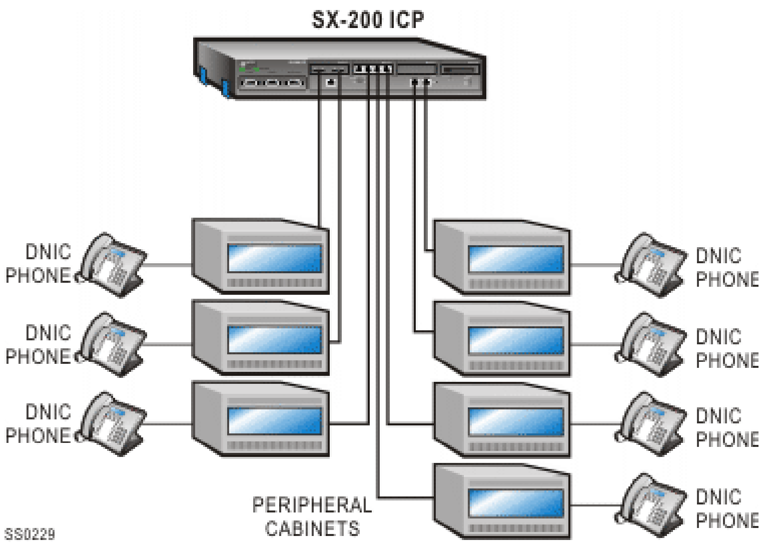
2. In CDE Form 01 - System Configuration, program the bays assigned in step 1 as node type **ASU BAY**. Program the line cards in Slots 1 and 2 <u>only</u> as **ONS LINE** (when selected displays "ONS CLASS CARD") or **LS/ONS COMBO**.

3. Program **ONS** devices in CDE Form 09 - Desktop Device Assignments (additional configuration in CDE Form 03 - COS Define, CDE Form 05 - Tenant Interconnection Table, CDE Form 19 - Call Rerouting Table, and/or CDE Form 30 - Device Interconnection Table is optional).
4. Program **LS** trunk devices in CDE Form 14 - Non-Dial-In Trunks or CDE Form 15 - Dial-In Trunks.

Installing SX-200 Peripheral Cabinets

Install an SX-200 Peripheral Cabinet

The following chart lists the steps required to unpack, inspect, and install a Peripheral Cabinet. Up to seven Peripheral Cabinets can be connected to the SX-200 ICP MX controller (requires 2 Quad CIM modules). Connected cabinet types can be any mixture of SX-200 ELx, EL, ML or LIGHT.



CAUTION: Do not open or unpack any printed circuit card cartons at this time.

Chart: Unpack, Inspect, and Install an SX-200 FD Cabinet

Step	Action	Comments
	Remove Cabinet Front Carton	
1.	Set aside all other packaged items for later unpacking and installation.	
2.	Unpack carton and check contents against packing list.	
3.	Remove the cabinet from the carton.	
	Inspect for Transit Damage	
4.	Visually check that the cabinet and all attached equipment is undamaged.	
	Open Cabinet Front Door	
5.	Pry open the two latches on the right-hand side of the black cover panel on the front door, and swing the cover panel off to	

Chart: Unpack, Inspect, and Install an SX-200 FD Cabinet		
Step	**Action**	**Comments**
	the left.	
6.	Loosen the two screws under the black panel to release the front door panel.	
7.	Pull the top of the front door out from the cabinet and lift it off the lower front cabinet rail.	
	Inspect Cabinet Interior	
8.	Make sure that all connector contacts are free of foreign matter.	
9.	Remove and reseat preconfigured cards.	
10.	Check that all cable connectors are seated firmly and are undamaged.	
11.	Inspect backplane for damage.	
	Close Cabinet Front Door	
14.	Lower the front door panel down over the front cabinet rail and position it on the front of the cabinet so that the two screw slots are aligned.	
12.	Tighten the two screws holding the door panel on the front of the cabinet.	
13.	Place the black cover panel over the appropriate opening, first catching the latch on the left hand side, and then the two latches on the right hand side.	
	Position Cabinet	
14.	Put the cabinet in its assigned position.	
15.	Verify that the site and power requirements have been met.	

Notes:

1. When positioning the cabinet, allow room for any PBX accessories (such as a UPS) or peripherals (such as a maintenance terminal).
2. Arrange for the power requirements for these peripherals and accessories. These power requirements are described in the documentation provided with the items.
3. An Uninterruptible Power Supply (UPS) is recommended.

Install an SX-200 Rack Mounted Cabinet

The following chart lists the steps required to unpack and inspect the SX-200 RM Peripheral Cabinet.

CAUTION: Do not open or unpack any printed circuit card cartons at this time.

Chart: Unpack, Inspect, and Install an SX-200 RM Cabinet		
Step	**Action**	**Comments**

Chart: Unpack, Inspect, and Install an SX-200 RM Cabinet

Step	Action	Comments
	Remove Cabinet from Carton	
1.	Set aside all other packaged items for later unpacking and installation.	
2.	Unpack carton and check contents against packing list.	
3.	Remove the cabinet from the carton.	
	Inspect for Transit Damage	
4.	Visually check that the cabinet and all attached equipment is undamaged.	
	Open Cabinet Front Door	
5.	Unlock the front door.	
6.	Pull the top of the front door out from the cabinet and lift it off the lower cabinet rail.	
	Inspect Cabinet Interior	
7.	Remove metal shipping brace (2 screws). Tighten BPS thumbscrew.	Retain shipping brace for use as a cable strain relief.
8.	Make sure that all connector contacts are free of foreign matter.	
9.	Remove and reseat preconfigured cards.	
10.	Check that all cable connectors are seated firmly and are undamaged.	
11.	Inspect backplane for damage.	
	Close Cabinet Front Door	
12.	Lower the front door panel down and position it over the front cabinet rail.	
13.	Lock the door onto the cabinet.	
	Position Cabinet	
14.	Put the cabinet in its assigned position.	
15.	Verify that the site and power requirements have been met.	
16.	Mount the shipping brace across the back of the cabinet, using either center pair of screws. Attach	

Chart: Unpack, Inspect, and Install an SX-200 RM Cabinet

Step	Action	Comments
	cables to the brace to provide strain relief.	
	Attach Rack-mount Hardware to Cabinet	A single cabinet can be supported on the wall with the SX-200 Wall Mount Brackets.
17.	Attach a rack-mount bracket to each side of the cabinet using 2 screws for each.	Screws are in plastic bag with key (skip these two steps if not rack mounting the cabinet).
18.	Install the cabinet in the rack using 4 screws for each bracket.	
	Stack non-Rack-mount Cabinet	
19.	Install additional cabinets above the first cabinet.	**CAUTION: A maximum of three cabinets may be stacked vertically.**

Notes:

1. When positioning the cabinet, allow room for any system accessories (such as a UPS) or peripherals (such as a maintenance terminal).

2. Arrange for the power requirements for these peripherals and accessories. These power requirements are described in the documentation provided with the items.

3. An Uninterruptible Power Supply (UPS) is recommended.

Wall Mounting an SX-200 Peripheral Cabinet

The SX-200 Wall Mount Bracket lets you support a single SX-200 ELx cabinet on the wall. These brackets are for the horizontal rackmount cabinet, not the vertical cabinet. The cabinet can be installed at eye level to improve working conditions. The bracket also has two push buttons that allow you to extend the cabinet away from the wall, thereby giving you plenty of room to access the back of the cabinet.

The package contains a template/instruction sheet, 1 set of brackets with mounting arms, and 16, ¾ inch, #10 wood screws.

The installation requires a ¾ inch (or thicker) piece of plywood secured on the wall.

1. Shut the system power off.
2. Use the template to mark the mounting hole locations on the plywood. Ensure that the leveling line is horizontal.
3. Identify the L and R (left and right) markings on the brackets.
4. On the plywood, mount the L bracket on the left side and the R bracket on the right side.
5. Push in the locking pins and pull the mounting arms completely out of the brackets. Note that there are two locking locations on the brackets.
6. If the SX-200 ELx cabinet has mounting ears, undo the 4 screws (1/2 inch, #8/32) and detach the ears from the cabinet. These screws will be used to attach the mounting arms to the cabinet. The mounting ears can be kept for rack mounting in the future.
7. Identify the L and R markings on the mounting arms. The mounting arms are marked L and R to match the brackets.
8. Attach the mounting arms to the cabinet observing the Left and Right markings.

9. Lift the cabinet, align the mounting arms to the brackets, and insert the arms into the bracket till the locking pin engages in the forward position.
10. Attach all cables to the cabinet.
11. Push in the locking pins and push the mounting arms in till the locking pin engages in the rear position.
12. Dress the cables if necessary. The installation is now complete.

Verify Ground Connections

The chart Verify Ground Connections checks ground requirements.

Canada: CEC - CSA C22.1 - Canadian Electrical Code, section 10.

TIA/EIA Standard TIA/EIA-607 - Commercial Building Grounding and Bonding Requirements for Telecommunications.

Refer to Testing Ground Paths for detailed information on grounding.

Each cabinet must derive ground from a Telecommunications Grounding Busbar which connects separately to the building grounding electrode system. Each cabinet must use a #6 AWG insulated green wire to connect to the Telecommunications Grounding Busbar.

The cabinet and associated ducting hardware must not be exposed to any ground source other than that described above. AC service wires bringing ac to the cabinet must not share an enclosure or raceway with other system grounds, dc power distribution wires, or signaling wires. All sources of external ground (for example, system signaling ground to the approved ground source) must connect only to the system single point ground.

Note: Before installation, an insulated #6 AWG green ground wire must be connected between the equipment location and an approved ground (metallic cold water pipe where it enters the building, or equivalent.

Chart: Verify Ground Connections

Step	Action	Comments
	Measure Voltage	
1.	Ensure that equipment power switches are in the OFF position	
2.	Verify the wiring of the ac outlet.	Use a locally approved plug-in circuit tester to confirm that the ground connection is good.
3.	Plug the ac line cord into the IEC (ac) receptacle on the cabinet.	230 Volt systems require the local procurement of a power cord for this product that shall be in compliance with the electrical codes in force within the region of installation.
4.	Plug the ac line cord into the building ac receptacle.	
5.	Measure the ac voltage between the PBX ground lug and the “approved” ground wire.	The #6 AWG green ground wire should be connected to an approved ground but not to the PBX.
6.	If the voltage is greater than 1.0 V, locate another approved ground.	The customer’s engineering support group is responsible for providing the approved ground.
7.	If the voltage is less than 1.0 V, measure the resistance to confirm continuity between the ac ground and the telecommunications ground	

Chart: Verify Ground Connections

Step	Action	Comments
	(both connect to the same building grounding electrode conductor).	
8.	Unplug the ac line cord from both the cabinet and the building ac receptacles.	
	Connect Ground	
9.	Connect the verified ground wire to the common ground terminal on the rear of the cabinet.	

Connect Cables (Fiber and Copper) between Cabinets and the SX-200 ICP MX Controller

In the SX-200 ICP system, optical fiber cables and copper cables connect the SX-200 cabinets to an MX controller.

The grade or type of cable used must be suitable for the installation (light or heavy duty, plenum, outdoor, etc.). Consult local building codes and your cable supplier.

Fiber Cable Handling Guidelines

The following basic guidelines and precautions for the handling of fiber cable and connectors apply to installation or servicing of the system.

- Cleanliness of the connector ferrule (or tip) is important for error-free transmission. Never touch the tip of a fiber connector.
- Always place the dust caps on the connectors and cables immediately after disconnecting. Ferrule tips on connectors can be cleaned with isopropyl alcohol.
- Fiber-optic cables are often more easily installed and pulled than copper because of their light weight and flexibility. However, care must be taken not to exceed specifications for minimum bend radius and maximum tensile strength.
- Optical repeaters cannot be used to increase the distance between cabinets.
- Procedures for repairing, splicing, or assembling fiber cables are available from the fiber component manufacturer, many of whom offer training courses for this purpose.

Fiber and Copper Cable Installation

When you have completed the following chart, optical fiber or copper cables will be connected between each Peripheral bay and the SX-200 ICP MX controller.

WARNING: **Fiber optic sources may emit intense light which can damage the retina. Never look directly into a source or into the end of a fiber energized by a source.**

When working with raw fiber optic cable be careful of fiber ends or slivers that can puncture the skin and cause irritation

Any fiber cabling used to connect the SX-200 ICP Controller to a Peripheral Bay must not be encased in a metal sheath so as to ensure isolation between the two nodes. Conduit may be used to encase fiber cabling as long as the conduit does not contact the Peripheral Bay.

Chart: Connect Cables Between the SX-200 ICP Controller and Peripheral Cabinets

Step	Action	Comments
	Run Cable between Devices	
1.	Run each cable between the SX-200 ICP Controller and the peripheral bay for which it is intended, following the handling guidelines listed in this section, and those specified by the cable manufacturer.	
	Connect Cables to the SX-200 ICP Controller	
2.	If using fiber, at the peripheral cabinet, remove the dust caps from the 2 connector ferrules at one end of the optical fiber cable.	Each cable has 2 connectors at each end.
3.	If using fiber, remove the dust caps from the Transmit (TX) and Receive (RX) connector ferrules on the 200 ICP Controller.	The TX ferrule is light colored and the RX ferrule is dark colored.
4.	If using fiber, at the front of the SX-200 ICP Controller, attach the two connectors of one cable end to the two FIM connectors. If using copper, attach the Category 5 UTP crossover cable (TX and RX pairs reversed) to the 8-pin modular jack for the CIM on the faceplate front of the 200 ICP Controller.	When the cabinet is powered up, the TX cable may be determined by using an optometer. When the fiber cables are connected correctly, both LEDs on the FIM will remain steadily lit (not flashing). If the FIM LEDs continue to flash after connecting the cable at both ends, switch the TX and RX connectors on one FIM. For information on FIM/FIM II/CIM LED indicators, see Troubleshooting section.
	Connect Cables to Peripheral Cabinet	
5.	At the rear of the Peripheral cabinet, raise the fiber cable slider bracket. Loosen the slider bracket screws, if necessary.	
6.	Route the fiber/copper cable through the slider bracket opening, and run it forward across the cabinet floor.	
7.	Unseat the carrier card for the FIM/ FIM II or CIM.	Follow general procedures for handling circuit cards.

The SX-200 ICP system uses the Peripheral FIM Carrier Card II in slot 12 to hold the FIM. FIM II and CIM modules are installed on the PRI card (slots 10 or 11) and the BCC III (slot 9) in the SX-200 ELx cabinet.		
8.	If using fiber, remove the dust caps from the Transmit (TX) and Receive (RX) connector ferrules on the FIM/FIM II.	
9.	If using fiber, remove the dust caps from the 2 connector ferrules at the peripheral bay end of the optical fiber cable.	The TX ferrule is light colored and the RX ferrule is dark colored.
10.	Plug the SX-200 ICP Controller TX cable into the TX connector on the Peripheral FIM Carrier II card. Do the same with the RX cable and the RX connector on the Peripheral FIM Carrier II card If using copper, plug the other end of the copper cable from the control cabinet into the modular jack for the CIM on the BCC III or on the PRI card in the peripheral cabinet or the CIM on the Peripheral Interface Carrier Module Card	**NOTE**: BCCIII card will not come up when you have an onboard CIM module and you are connected through Peripheral Carrier card.
11.	At the rear of the cabinet, lower the slider bracket so that the foam strip rests on the fiber/copper cable. Be sure to tighten the screws after closing the bracket.	

Connect Cables Between the SX-200 ICP MX and Cross-Connect Field

The following chart describes how to connect cables at the connection blocks, run cables from the cross-connect field to the cabinet(s), attach cable connectors to the cabinet plugs, and connect cables at the cross-connect field.

Chart: Connect Cables Between the SX-200 ICP and Cross-Connect Field

Step	Action	Comments
	Install Connection Blocks	
1.	Install required connection blocks at cross-connect field.	
	Connect Cables	

Chart: Connect Cables Between the SX-200 ICP and Cross-Connect Field

Step	Action	Comments
2.	Connect 25-pair cables to connection blocks.	Cross-connection tables are shown in the Cabling and Cross Connections section.
3.	Mark each cable connector or plug with its corresponding cabinet plug number.	
	Run Cables	
4.	Run the 25-pair cables between the cross-connect field and the cabinet.	
	Attach Cabinet Cable Connectors	
5.	Attach each cable connector to its cabinet plug, and tighten connector retaining screw or strap.	
6.	Install split ferrites around each 25-pair cable which exits the system. Locate the ferrites approximately 4 inches from the case of the connector (see this Figure). This step does not apply to the SX-200 EL.	The split ferrites supplied consist of two ferrite cores contained in a hinged plastic housing (see this Figure).
	Cross Connect Cables at Cross-connect Field	
7.	Cross connect station lines, CO trunks, other trunks, and equipment to cross-connection blocks for cables from the SX-200 ICP.	Tip and Ring assignment tables in identify which equipment connects to each connection block pin.

Figure: Ferrite Installed on a Cable

Power Up Peripheral Cabinet(s)

The following chart describes how to power up a peripheral cabinet.

Chart: Power Up Peripheral Cabinet

Step	Action	Comments
	Power Up Cabinet	
1.	Make sure that all cards are in place and well seated, and that all cables within the cabinet (including optional UPS in the peripheral bay) are connected.	It is recommended that a UPS be used with all cabinets of a system.
2.	Plug the line cord into the ac receptacle.	If the bay has a UPS, the line cord should be plugged into the UPS, and the UPS line cord into the ac receptacle, according to manufacturer's instructions.
3.	Turn on UPS (if present), and then open the peripheral bay front door panel, if not already opened.	
4.	Turn Bay Power Supply ON.	

5.	Close the peripheral bay front door panel.	

Installing and Replacing Circuit Cards and Modules in Peripheral Cabinets

Unpack and Handle Printed Circuit Cards and Modules

The chart Unpack and Handle Printed Circuit Cards describes how to properly unpack and handle printed circuit cards.

CAUTION: Read these instructions carefully and follow them when performing procedures described in following sections. Power must be off when cards are being removed.

Cards that are not correctly packed in antistatic packaging when returned will not be covered by any warranty.

All cabinets must be unplugged from the ac mains during servicing. To reduce static susceptibility, always attach the wrist strap from the cabinet being serviced, and immediately place any item removed into antistatic packaging.

Chart: Unpack and Handle Printed Circuit Cards and Modules

Step	Action	Comments
	When Unpacking	
1.	Make sure the system ground is connected.	All cabinets must be unplugged from the ac mains during servicing. To reduce static susceptibility, always attach the wrist strap from the cabinet being serviced, and immediately place any item removed into antistatic packaging.
2.	Put on the antistatic wrist strap when unpacking and handling cards.	The antistatic wrist strap must be connected to the cabinet chassis, which must be connected to an approved ground to provide protection from static discharges.
3.	Remove the card from its Mitel packaging.	The card is packed in antistatic packaging.
4.	When you are ready to put the card in its slot, open the antistatic packaging, and remove the card.	The card should stay protected in the bag until it is installed.
	When Handling Cards	
5.	Keep the packaging and antistatic bag in case the card has to be returned.	Handling the card faces or components may cause damage.
6.	Do not touch the gold edge connectors.	
7.	Avoid contact with any exposed electrical connections.	

	Identifying Card or Module Positions	**CAUTION:** Do not install cards or modules yet.
8.	Identify the appropriate slots for digital control cards in the peripheral cabinets. The System Configuration and/or the Tip and Ring forms identify which card type goes into each slot.	Peripheral Bay Control Card: semicircle Peripheral Bay Power Supply: right triangle
9.		

Identify the appropriate peripheral cabinet card slot for digital peripheral cards.

The T1 Trunk card plugs into slot 10 or 11 but appears logically as slot 5 or 6 respectively.

Install high-power digital peripheral cards only in upper slots (FD cabinets only):
- DID Line Card
- Universal Card
- T1-DS1 Trunk Card
- OPS Line Card

Install low-power digital peripheral cards in any slots. The SX-200 RM cabinets do not have high power and low power designations. Slots 1 to 8 can accept any peripheral interface card, except T1 trunk. A maximum of four high power cards is allowed. The system accepts the first four high power cards it detects, beginning at slot 1. T1 Trunk cards can be installed to slots 10 and 11 only, and require corresponding slots 5 or 6 to be left empty. The PRI card (unlike the T1 trunk card) is not classed as a high power card. Because the PRI card is a separate bay, the PRI card is not included in the count for the four high power cards.

10.	If necessary, move the standoffs on the BCC III to accomodate the FIM II.	The BCC III standoffs are in the CIM postion as a default.

11.	Add the required modules on the circuit cards.	The BCC III, the PRI card and the FIM Carrier cards hold add-on modules.
	Installing Cards or Modules	**CAUTION:** Do not install cards or modules yet.
12.		

If installing control cards, make sure power is off.

If installing peripheral cards in an operating system, power may be left on.	Cards that must not be inserted while system power is on carry a caution notice.	
13.	Set switches on those circuit cards requiring it.	Instructions for setting switches are included in card installation charts for: - Bay Control Card II - Universal Card - T1-DS1 Trunk Card - PRI Card
14.	Lift the card extractor, and slide the card into the slot. Press on the extractor after it mates with the notch in the shelf to seat the card firmly.	Each digital peripheral card and module has a card extractor to help seat the card firmly in the backplane. The extractor is also used to provide leverage to remove cards or modules.
	Removing Cards or Modules	
15.	If you are removing cards from an operating system, turn power off, if possible.	**CAUTION:** Cards that must not be removed while the system power is on are listed in Step 10.
16.	Make sure the system ground is connected.	
17.	Put on the antistatic wrist strap when removing and repacking cards. Connect the antistatic wrist strap to the system chassis, which must be connected to an approved ground to provide protection from static discharges	All cabinets must be unplugged from the ac mains during servicing. To reduce static susceptibility, always attach the wrist strap from the cabinet being serviced, and immediately place any item removed into antistatic packaging.
18.	Remove the card by using the extractor as a lever and pulling the card towards you.	The card extractor provides leverage to remove the card.
	Repacking Cards and Modules	
19.	Use the packaging retained after unpacking.	Use original or similar packaging material.
20.	Put a card or module in antistatic packaging as soon as you remove it from its slot.	Place suspected faulty cards into antistatic packaging to prevent further damage.

21.	When you are finished, replace the antistatic wrist strap in the cabinet. Replace any guards and/or covers which have been removed.	
22.	If you have powered down the system, power it up again.	

Install a Bay Control Card

The following chart describes installation of a Bay Control Card (BCC).This installation pertains to the BCC II and the BCC III. The BCC is NOT a "hot-swappable card", that is, the power must be off when putting the card into or out of the cabinet.

Notes:

1. A BCC performs lower-level real-time control tasks for a digital peripheral bay such as scanning for activity on ONS ports, controlling CO trunks and E&M trunks, tone cadencing, and communicating to consoles.
2. One BCC is installed in each bay. **Note**: BCC III cards must be installed in an SX-200 ELx (Rev 2) cabinet. The EL cabinet (Rev 1) does not supply the proper card mounting and subsequent module mounting.

Chart: Install a Bay Control Card

Step	Action	Comments
	Unpack, Inspect, and Install	
1.	Wearing the antistatic wrist strap, unpack and inspect the BCC.	Follow general procedures for handling circuit cards.
2.		

For a BCC II, set all 4 switches (2 pairs) on the circuit card to the closed position (for normal operation).

A BCC III has no switches.	See this Figure for switch locations on the BCC II.
3.	

For a BCC III, attach the required modules. See the following installation procedures for the modules: FIM II, CIM, Maintenance module, T1/E1 module, and DSP module (single).

A BCC II has no modules.	A BCC III in a peripheral cabinet (SX-200 ELx cabinet) can have a FIM II or a CIM, a T1/E1 module, and a DSP module (single); a Maintenance module is not necessary. The BCC III has three module sites. Site 1 supports a FIM II or a CIM (connectivity between the peripheral cabinet and the main control cabinet) Site 2 supports a T1/E1 module (T1/D4 functionality) Site 3 supports a DSP module (single) (CLASS functionality and Record a Call). Remember that slots 5 and 6 must be empty if both T1 links on the T1/E1 module are programmed. If a card is installed in slots 5 or 6, the T1 link will be ignored.	
Caution: Make sure that the power is off, before you install the BCC.		
4.	Insert the BCC II or the BCC III card into slot 9 of the rack mount cabinet.	The peripheral cabinets and the control cabinet require a BCC.

Install Modules on the BCC III

A BCC III in a peripheral cabinet (SX-200 ELx cabinet only) can have a FIM II or a CIM, a T1/E1 module, and a DSP module (single); a Maintenance module is not necessary.

The following procedures describe how to install these modules on the BCC III:

- FIM II
- CIM
- Maintenance module
- T1/E1 module
- DSP module (single)

Installing a FIM II on the BCC III

The FIM II sits on site 1 of the BCC III recessed back from the faceplate.

1. On the BCC III, remove the standoffs that are in positions WTA1, WTB1, WTC1, and WTD1 (screws are on both ends of the standoff).The default position for the site 1 standoffs is the CIM position.
2. On the BCC IIII, install the four standoffs in positions WTA11, WTB11, WTC11, and WTD11.
3. Line up the FIM II so the J3 connector on the FIM II connects to P311 on the BCC III.
4. Make sure that these standoff holes on the FIM II align with the stated standoffs on the BCC III:
   WT5 hole with WTB11 standoff
   WT2 hole with WTC11 standoff
   WT1 hole with WTD11 standoff

WT4 hole with WTA11 standoff.

5. Fasten the FIM II onto the BCC III with the screws on the standoffs.

Installing a CIM or Maintenance module on the BCC III

The CIM supports a distance of up to 30 meters or 100 feet between cabinets. The CIM uses copper cable, Standard Category 5 UTP (unshielded twisted pair) with RJ45 connectors. Only one twisted pair cable is required for copper installation; the cable has both a TX and RX pair inside. Refer to the CIM Ports table that shows the pinouts for the copper cable.
Systems without data cables require Category 5 cables for the CIM. Systems with data cables from the main control cabinet to the peripheral cabinet can use these data cables for the CIM if the cable is interconnected through a punchdown at each end, as long as the distance remains within the 100 feet limit.

The CIM or Maintenance module sits on site 1 of the BCC III close to the faceplate.

1. On the BCC III, remove the screws on the standoffs WTA1, WTB1, WTC1, and WTD1. The default position for the site 1 standoffs is the CIM position.
2. Line up the CIM or Maintenance module so the J3 connector on the module connects to P31 on the BCC III.
3. Make sure that these standoff holes on the module align with the following standoffs on the BCC III:
   WT5 hole with WTB1 standoff
   WT2 hole with WTC1 standoff
   WT1 hole with WTD1 standoff
   WT4 hole with WTA1 standoff.
4. Fasten the module onto the BCC III with the screws on the standoffs.

Installing a T1/E1 module on the BCC III

The T1/E1 module sits on site 2 of the BCC III close to the faceplate. The two LT/NT connectors (jumpers) on the T1/E1 module are normally set to the NT (Network Termination) setting. This is the default setting on the T1/E1 module.

Note: The LT/NT connector is set to the LT (Line Termination) setting when two systems are connected back-to-back as part of a local network. In this scenario, one of the connectors is set to LT. This setting swaps the transmit and receive pairs on its connector so that the transmit of the NT port goes to the receive of the LT port and vice versa.

1. On the BCC III, remove the screws on the standoffs WTA2, WTB2, WTC2, and WTD2.
2. Line up the T1/E1 module so the J1 connector on the module connects to P12 on the BCC III and the J3 connector on the module connects to P32 on the BCC III.
3. Make sure that these standoff holes on the module align with the following standoffs on the BCC III:
   WT5 hole with WTB2 standoff
   WT2 hole with WTC2 standoff
   WT1 hole with WTD2 standoff
   WT4 hole with WTA2 standoff.
4. Fasten the module onto the BCC III with the screws on the standoffs.

The bottom connector is T1 Link 1 and occupies slot 5 in software programming;
the top connector is T1 Link 2 and occupies slot 6. Ensure that cabinet slots 5 and 6 are empty

when you program both T1 links from the T1/E1 module. If there is a card in slot 5 or 6, the T1 link will be ignored.

Installing a DSP module (single) on the BCC III

The DSP module sits on site 3 of the BCC III.

1. On the BCC III, remove the screws on the standoffs WTA3, WTB3, WTC3, and WTD3.
2. Line up the DSP module so the following J connectors on the module connect to the stated P connectors on the BCC III:
   J2 connects to P23
   J1 connects to P13
   J3 connects to P33.
3. Make sure that these standoff holes on the DSP module align with the following standoffs on the BCC III:
   WT3 hole with WTC3 standoff
   WT2 hole with WTB3 standoff
   WT4 hole with WTD3 standoff
   WT1 hole with WTA3 standoff.
4. Fasten the module onto the BCC III with the screws on the standoffs.

Install a Peripheral Interface Module Carrier Card

Before installing the Peripheral Interface Module Carrier Card (PIMCC) into a cabinet, you must install a FIM II or a CIM onto the PIMCC. The following procedures are listed below:

- Installing a FIM II on the Peripheral Interface Module Carrier Card
- Installing a CIM on the Peripheral Interface Module Carrier Card
- Installing the Peripheral Interface Module Carrier Card

Caution: Do not remove the PIMCC with the Bay Power on.

Installing a FIM II on the Peripheral Interface Module Carrier Card

The FIM II sits recessed back from the faceplate of the Peripheral Interface Module Carrier card.

1. On the Peripheral Interface Module Carrier card, remove the standoffs that are in the CIM position (screws are on both ends of the standoffs).
2. On the Peripheral Interface Module Carrier card, install the four standoffs in the positions WTA3, WTB3, WTC3, and WTD3.
3. Line up the FIM II so the J3 connector on the FIM II connects to P3 on the Peripheral Interface Module Carrier card.
4. Fasten the FIM II onto the Peripheral Interface Module Carrier card with the screws on the standoffs.

Installing a CIM on the Peripheral Interface Module Carrier Card

The CIM supports a distance of up to 30 meters or 100 feet between cabinets. The CIM uses copper cable, Standard Category 5 UTP with RJ45 connectors.

The CIM sits close to the faceplate of the Peripheral Interface Module Carrier card.

1. On the Peripheral Interface Module Carrier card, remove the screws on the standoffs.
2. Line up the CIM so the J3 connector on the CIM connects to P4 on the Peripheral Interface Module Carrier card.
3. Make sure that these standoff holes on the module align with the following standoffs on the Peripheral Interface Module Carrier card:
   WT5 hole with WTB4 standoff
   WT2 hole with WTC4 standoff
   WT1 hole with WTD4 standoff
   WT4 hole with WTA4 standoff.
4. Fasten the module onto the Peripheral Interface Module Carrier card with the screws on the standoffs.

Installing the Peripheral Interface Module Carrier card

This card with the CIM or FIM II provides connectivity to the main control cabinet when the peripheral cabinet has a BCC II.

1. Follow general procedures for handling circuit cards.
2. Install the CIM or the FIM II onto the Peripheral Interface Module Carrier card.
3. Attach the antistatic wrist strap.
4. Insert the Peripheral Interface Module Carrier card into slot 12 of a peripheral rack-mount cabinet.

Install a FIM Carrier Card and FIM

Fiber Interface Modules (FIMs) are installed only in a multi-cabinet system. Refer to the following chart.

Chart: Install FIM Carrier and FIM into a Cabinet

Step	Action	Comments
1.	Follow general procedures for handling circuit cards.	
2.	Attach the antistatic wrist strap.	
	Install FIM onto FIM Carrier (all variants)	
3.	Unpack the FIM from its antistatic packaging, and inspect the FIM for damage.	
4.	Install the FIM onto its FIM Carrier, connecting it firmly with the DIN connector on the Carrier.	
5.	Fasten the FIM to the FIM Carrier with 2 screws.	

	Install Peripheral FIM Carrier II Card	
6.	Unpack the Peripheral FIM Carrier II card from its antistatic packaging, and inspect it for damage.	The Peripheral FIM Carrier II card is only used in an SX-200 RM Peripheral Cabinet.
7.	Install the FIM onto the Peripheral FIM Carrier II card.	Refer to this Figure.
8.	Install the Peripheral FIM Carrier Card II and FIM into slot 12 of a SX-200 RM Peripheral cabinet.	
	Install Peripheral FIM Carrier on the BCC	
9.	Unpack the Peripheral FIM Carrier card from its antistatic packaging, and inspect it for damage.	The Peripheral FIM Carrier card is only used in an SX-200 LIGHT Peripheral Cabinet.
10.	Install the FIM onto the Peripheral FIM Carrier card.	Refer to this Figure.
11.	Position the Peripheral FIM Carrier on the component side of the BCC and align its connectors.	Refer to this Figure.
12.	Snap the Peripheral FIM Carrier into place on the BCC.	
13.	Slide the Bay Control Card into its Peripheral bay slot.	

Install a Universal Card and Modules

The following chart describes how to install a Universal Card and its modules.

Note: Up to four modules can be mounted on a Universal Card provided that the total power rating is 10 or less.

DTMF Receiver/Relay Module - has four dual-tone multi-frequency (DTMF) receiver circuits and two relays for night bells or alarms. (power rating = 2)

Music/Paging Module - provides one 600-ohm balanced audio (music) input, a 200-ohm output to a paging amplifier, and one relay contact for controlling an amplifier. (power rating = 1)

E&M Trunk Module - Interfaces the PBX to one standard E&M trunk. (power rating = 3)

Chart: Install Universal Card and Modules

Step	Action	Comments
	Unpack and Inspect	
1.	While wearing the antistatic wrist	Follow general procedures for handling circuit cards.

Chart: Install Universal Card and Modules

Step	Action	Comments
	strap, unpack the Universal Card (s), standoffs and module(s).	
2.	Inspect for loose or missing components, and for damage.	This Figure shows a Universal Card and modules.
	Verify Power Ratings	
3.	Verify the power ratings for each Universal Card.	If the total power rating exceeds 10, the system indicates an alarm and ignores the Universal Card at power up.
	Install Standoffs	
4.	Install the plastic standoffs on the Universal Card (where required to mate with the modules). Do not attempt to install standoffs in holes that are obstructed by components or wire.	This Figure shows typical module installation.
	Install DTMF/Receiver Module	
5.	Insert each DTMF/receiver module into its assigned location. Two relays (not associated with the DTMF receivers) can be used to control system functions such as night bells or alarms. Each relay provides a contact closure across a Tip-Ring pair. Ratings: 90 V rms at 0.1 A 48 V dc at 0.5 A	The number of modules is determined by calculations in the Engineering Information section. Main Control cards may also contain receivers. **CAUTION:** This relay contact may be connected only to a secondary circuit that has no direct connection to a primary circuit, and which receives its power from a transformer, converter, or equivalent isolation device situated within the equipment.
6.	Press the module until it snaps into its standoffs. Make sure that connectors are properly seated.	
	Install Music/Paging Module	Module specifications are shown in the Music-on-Hold/Paging Module Specification table.
7.	Insert each music/paging module in its assigned location. Music/paging equipment is: - outside the PBX - should be in an environment specified by the suppliers - connected to the PBX through the cross-connect field	**CAUTION:** The relay contact is only used to control the paging amplifier. It may be connected only to a secondary circuit that has no direct connection to a primary circuit, and which receives its power from a transformer, converter, or equivalent isolation device situated within the equipment.
8.	Press the module until it snaps into its standoffs. Make sure that connectors	

Chart: Install Universal Card and Modules

Step	Action	Comments
	are properly seated.	
	Install E&M Trunk Module	
9.	Set the eight DIP switches for the type of trunk in use.	Refer to the E&M Trunk Module Switch Settings table.
10.	Insert the trunk module in its assigned location.	Interfaces to Type 1 and Type 5 E&M trunks.
11.	Press the module until it snaps into its standoffs. Make sure that connectors are properly seated.	
	Install Universal Card	
12.	Insert the Universal Card in its assigned slot in the bay.	A Universal Card can plug into any upper peripheral slot in a digital bay.

Music-on-Hold/Paging Module Specifications

Table: Music-on-Hold/Paging Module Specifications

Music-on-Hold Input	input impedance input level	600 ohms -6 dBm
Paging Output	output impedance (low) output level into 600 ohms	200 ohms -6 dBm
Relay Contact	maximums	90 V rms at 0.1 A 48 V dc at 0.5 A

E&M Trunk Module Switch Settings

Table: E&M Trunk Module Switch Settings

Function	Switches							
	1	2	3	4	5	6	7	8
PBX to Line Hardware Gain								
3 dB	0	X	X	X	X	X	X	X
-13 dB	1	X	X	X	X	X	X	X
Line to PBX Hardware Gain								

Table: E&M Trunk Module Switch Settings

Function	Switches							
-4 dB	X	0	X	X	X	X	X	X
-11dB	X	1	X	X	X	X	X	X
Termination								
600 ohm	X	X	1	0	X	X	X	X
Complex	X	X	0	1	X	X	X	X
Transmission								
2-wire	X	X	X	X	1	X	X	X
4-wire	X	X	X	X	0	X	X	X
Signaling								
Type 1	X	X	X	X	X	1	X	X
Type 6	X	X	X	X	X	0	X	X

0 = open, 1 = closed, X = not applicable

Install Line (ONS, OPS, and Digital) and DID Trunk Cards

The following chart applies to the ONS/CLASS, OPS, and Digital Line Cards, and the DID Trunk Card. It describes unpacking, inspecting, and installing a line or trunk card.

Notes:

1. The ONS/CLASS Line Card interfaces up to 12 standard telephones (rotary or DTMF) or ONS Voice mail ports. The ONS/CLASS Line Card ports can interface alternate music sources for the Automatic Call Distribution feature. An alternate music source must be either an FCC Part 68 and Industry Canada CS-03 approved recorded announcement device connected to an ONS circuit, or another source connected through an FCC Part 68 and Industry Canada CS-03 approved voice coupler or voice connecting arrangement to an ONS circuit.
2. The OPS Line Card contains six off-premises line circuits.
3. The Digital Line Card is a 12-circuit card that interfaces the PBX to the following peripherals: SUPERSET 4000 series telephones, SUPERSET 400 series telephones, SUPERSET 3DN telephones, SUPERSET 4DN telephones, DSS/BLF Interface Unit, DNIC Music-on-Hold/Pager Unit (DMP), SUPERCONSOLE 1000 Attendant Console, DNIC voice mail, and the DATASET 1103 and 2103 products. See Engineering Information for the loop length rules applying to the installation of the above listed peripheral devices.
4. The DID Trunk Card interfaces to six one-way direct inward dial circuits.

WARNING: **Any connection of an ONS or Digital Line Card to an off-premise application, an out-of-plant application, or any other exposed plant application may result in a safety hazard and/or defective operation and/or equipment damage.**

Chart: Install Line or DID Trunk Cards		
Step	**Action**	**Comments**
	Unpack and Inspect	
1.	Wearing the antistatic wrist strap, unpack and inspect the line or trunk card.	Follow general procedures for handling circuit cards.
	Install the Line or Trunk Card	
2.		

Slide the line or trunk card into its slot.
Cards with squares on their faceplates are high-power digital peripheral cards. The DID Trunk Card is a high-power digital peripheral card.

The ONS/CLASS Line Cards and Digital Line Cards are low-power digital peripheral cards.

Up to four DID cards can be installed in each digital peripheral bay.

CLASS functionality requires the ONS/CLASS Line card to be installed in a SX-200 ELx cabinet. The ONS/CLASS card must be in the same cabinet as a BCC III with a DSP module (single). CLASS functionality also requires the purchasable FOR Options 102 (Feature Level) and 89 (CLASS Functionality for ONS Sets). See CLASS for Analog Sets.

Install a LS/CLASS Trunk Card

The following chart describes how to install a LS/CLASS Trunk card in a peripheral cabinet. The LS/CLASS Trunk card provides interfaces to eight central office trunks. Each trunk circuit operates as a loop start, not ground start.

Note: The tip and ring assignments for the LS/CLASS Trunk card differ from the LS/GS card. Please refer to the SX-200 ICP Tip and Ring Assignments table.

Chart: Install LS/CLASS Trunk Card		
Step	**Action**	**Comments**
	Identify Trunk Circuits	
1.	Identify trunk circuits by bay, slot, circuit and type (loopstart only)	

This is available from the configuration information prepared by the customer service representative.

There are eight trunk circuits per card.		
	Unpack and Inspect	
2.	Wearing the antistatic wrist strap, unpack and inspect the card	Follow general procedures for handling circuit cards.
	Install LS/CLASS Trunk Card	
3.	Slide the LS/CLASS Trunk card into its slot.	The card is a low powered Peripheral Interface Card and can be installed in any slot 1-8 in a SX-200 rack mount cabinet.

Install an LS/GS Trunk Card

The following chart describes how to install a loop start/ground start (LS/GS) trunk card. The LS/GS Trunk Card provides interfaces to six central office trunks. Each trunk circuit can operate as loop start or as ground start.

Chart: Install LS/GS Trunk Card

Step	Action	Comments
	Identify Trunk Circuits	
1.	Identify trunk circuits by bay, slot, circuit and type (loopstart or groundstart).	This is available from the configuration information prepared by the customer service representative.
	Unpack and Inspect	
2.	Wearing the antistatic wrist strap, unpack and inspect the LS/GS Trunk Card and jumpers.	Follow general procedures for handling circuit cards.
	Set LS/GS Trunk Card Jumpers	
3.		

Set the jumpers in position for each of the six trunks on the card.

Position the marked end to G for a ground start or L for a loop start trunk.	Shown in this Figure.	
	Install LS/GS Trunk Card	
4.	Slide the LS/GS Trunk Card into its slot.	

Install a T1-DS1 Trunk Card into a Peripheral Cabinet

The T1-DS1 Trunk Card interfaces to a single T1 trunk circuit. SX-200 RM cabinets do not use a T1 adapter.

WARNING: Incorrect installation may result in a shock hazard and/or defective equipment operation.

Chart: Install a T1-DS1 Trunk Card into an SX-200 RM Cabinet

Step	Action	Comments
	Unpack and Inspect	
1.	While wearing the antistatic wrist strap, unpack and inspect the T1-DS1 Trunk Card.	Follow general procedures for handling circuit cards.
	Set Switches	If you are reseating a previously installed card, switches may already be set.
2.		

Set DIP switches (for appropriate line equalization) according to cable length from the CSU.

0 to 149 ft:	S1 CLOSED;	S2-8 OPEN
150 to 449 ft:	S2-4 CLOSED;	S1, S5-8 OPEN
450 to 655 ft:	S5-7 CLOSED;	S1-4, S8 OPEN

Note:
Length is distance between T1-DS1 Trunk card and channel service unit (CSU), not loop length.

Note:
The CSU is provided by the telephone company.

	Install T1-DS1 Trunk Card(s)	
3.	Insert the T1-DS1 trunk card into slot 10. Connect T1 trunk cable to connector J5 on the back of the cabinet.	PIC slot 5 must be left vacant when a T1 card is installed in slot 10.
4.		

Insert the T1-DS1 trunk card into slot 11.

Connect T1 trunk cable to connector J6 on the back of the cabinet.	PIC slot 6 must be left vacant when a T1 card is installed in slot 11.

	Install T1 Trunk Cable From CSU	
5.	Connect the shielded T1 trunk cable from the CSU to each active T1 connector on the back panel. Connect the T1 trunk cable shield to the T1 ground stud on the back panel of the cabinet.	

Each T1 trunk card requires a cable.

Note: T1 cables must be shielded. Attach the T1 cable to the strain relief bar on the back of the cabinet.

Install a T1-DS1 Trunk Card and T1 Trunk Adapter into an SX-200 FD Cabinet

SX-200 FD cabinets use a T1 adapter with a straight-out T1 connector. SX-200 RM cabinets do not use a T1 adapter.

Notes:

1. The T1 -DS1 Trunk Card interfaces to a single T1 trunk circuit.
2. The T1 Trunk Card kit includes:
   - T1-DS1 Trunk Card
   - T1 Trunk Adapter assembly
   - hardware kit
3. The hardware kit contains parts to accommodate manufacturing variants; all the hardware parts will not be used in any particular installation.

WARNING: Incorrect installation may result in a shock hazard and/or defective equipment operation.

CAUTION: Failure to remove metal housing before installing the T1 Adapter onto a Control or Peripheral Cabinet may result in equipment damage and/or electrical shock.

Chart: Install a T1-DS1 Trunk Card into an *S*X-200 RM Cabinet

Step	Action	Comments
	Unpack and Inspect	
1.	While wearing the antistatic wrist strap, unpack and inspect the T1-DS1 Trunk Card.	Follow general procedures for handling circuit cards.
	Set Switches	If reseating a previously installed card, switches may already be set.
2.		

Set DIP switches (for appropriate line equalization) according to cable length from the CSU.

0 to 149 ft:	S1 CLOSED;	S2-8 OPEN
150 to 449 ft:	S2-4 CLOSED;	S1, S5-8 OPEN
450 to 655 ft:	S5-7 CLOSED;	S1-4, S8 OPEN

Note:
Length is distance between T1-DS1 Trunk card and channel service unit (CSU), not loop length.

Note:
The CSU is provided by the telephone company.

Install T1 Trunk Adapter Assembly.

A dual T1 adapter may be installed on connector J5, and T1 trunk cards may be installed in slots 5 and 6.	Install part number 9400-100-302-NA for a single T1 in slot 6, or part number 9400-100-304-NA for two T1 cards in slots 5 and 6.	
3.	At the backplate connector associated with the T1-DS1 Trunk Card, remove the strain relief assembly and the hex nuts or screws from the connector.	If connector J5 protrudes through the backplate, install the four very short standoffs from kit into the four screw holes in the backplate (See the Install a T1 Trunk Adapter Assembly (Triple Play) Figure and the Install a Dual T1 Trunk Adapter Assembly (Triple Play) Figure).
4.	Plug the T1 Trunk Adapter Assembly connector into the backplate connector. Align the 4 screw holes on the housing with holes (or standoffs) on backplate. Fasten using 4 screws and washers from kit.	Refer to the Install a T1 Trunk Adapter Assembly (Triple Play) Figure and the Install a Dual T1 Trunk Adapter Assembly (Triple Play) Figure.
	Install T1-DS1 Trunk Card(s)	
5.	Insert the T1-DS1 trunk card in a high power slot.	First T1 card goes in slot 6 of the peripheral cabinet.
	Install T1 Trunk Cable From CSU	
6.	Connect the shielded T1 trunk cable from the CSU to the T1 Trunk Adapter.	Each T1 trunk card requires a cable.

Install a PRI card

Before You Begin

Installation Pre-requisites

The PRI card requires

- Slot 10 or slot 11 in an SX-200 ELx cabinet Rev 4.4 or greater (PN 9109-600-002-NA). The SX-200 ICP must have a Stratum 3 clock module.
- ISDN Maintenance and Adminstration Tool (IMAT) installed on a PC with Windows 98, or Windows 2000 Professional operating system. IMAT provides the communication link between the card and the computer.
- Purchasable FOR option - Number of Links, where the number of links equals the number of links that will be activated on the PRI card(s) being installed.

Connecting a Laptop Computer to the PRI Card

To connect a laptop computer to the PRI card:

1 Install IMAT.
Refer to instructions on the PRI card software CD-ROM case. Use the directory IMAT_install. To install IMAT on a PC with Win95/98 or Windows 2000 Professional operating system, double-click on the setup.exe file and follow the installation steps.

2 Connect the serial port of the PC to the serial port of the PRI card. You can use a regular (straight) 9-pin to 9-pin female-to-male cable or modems for remote access. If you are connecting a modem to the PRI card, you must use a null modem adapter.

Note: A networked computer running Win95/98 has difficulties communicating using Dial-Up Networking. It is strongly suggested that a non-networked computer be used. Use a null modem adapter if connecting to the card through a modem.

3 If you have not already done so on the laptop computer, install a NT Direct Connect modem type. You may also wish to install a modem for remote connection.

4 On the laptop, create a new Dial-Up Networking entry. DIAL-up networking establishes a data link with the PRI card. The user may need to download DIAL-up networking upgrade v1.3 from Microsoft in order to connect to the PRI card.

Installing a NT Direct Connection Device Driver

To install and configure NT Direct Connection Device Driver:

1. On the **Start** menu, point to **Settings**, and then click **Control Panel**.
2. Double-click the **Modems** icon.
3. From the **Modem Properties** window, click **Add**.
4. When the **Install New Modem** window appears, ensure that **Don't detect my modem, I will select from a list** is checked. Click **Next**.

5. A list of modem manufacturers and models is now displayed. Click **Have disk** for the direct connect cable.
   You may also install a modem for remote connection.

6.

Enter *c:\Program Files\Mitel\Imat* or insert the PRI CD-ROM and enter the CD-ROM drive letter and *:\Support\gateway.inf* in the **Copy manufacturer's files from:** field in the **Install From Disk** dialog box and click **OK** for the direct connect cable.

Note: If you have an IMAT version earlier than Release 6, enter *c:\mitel\isdn\imat* instead of *c:\Program Files\Mitel\Imat.*
If you are installing a modem, follow the manufacturer's installation instructions.

7. Click **Next** to select the NT Direct Connection.

8. Select COM 1 or COM 2, whichever is appropriate for your PC settings and click **Next**.

9. Click **Finish** on the *Your modem has been set up successfully* window.

10. Select your country, area code, and dial-out digit in the **Location Information** window. Click **Next**.

11. Click **Finish** on the **Install New Modem** window.

12.

In the **Modem Properties** window
- Maximum speed: **38400**
- Check: **only connect at this speed**
- Data bits: **8**
- Parity: **none**
- Stop bits: **1**
- Mode: **auto answer**

In the **Advanced Settings** window, for a direct connect cable:
- Turn off: **error control**
- Turn off: **flow control**

In the **Advanced Settings** window, for modem:
- Turn off: **error control**
- Turn on: **flow control** and select **Hardware**

13. Close the **Control Panel** window.

Configuring a Modem for the PRI Card

To configure a modem for the PRI T1 card

1.

Set the DIP switches on the modem for initial setup, according to the manufacturer's instructions.

For example, on a US Robotics modem, use the following settings:

Function	Switch	Position
DTR override	1	Off
Numeric results	2	Off
Display result codes	3	Off
Echo offline commands	4	On
Auto answer on first ring	5	On
Carrier detect normal	6	On
Load NVRAM defaults	7	On
Smart mode	8	Off

Note: In the US Robotics modem switches, the up position is on, and the down position is off.

2. Connect your PC to the external modem, launch a Hyperterminal session, and set the serial port settings to
   - Speed: **38400**
   - Data bits: **8**
   - Parity: **none**
   - Stop bits: **1**

3.

Enter the modem command settings.

For example, on a US Robotics modem, the command settings are as follows:

Function	Command
Factory settings	AT&F1
Modem ignores RTS	AT&R1
Lock the modem/com port to this speed	AT&B1
Selects power-on-reset default configuration	AT&Y0
Write to NVRAM register set 0	AT&W0

4. Power down the modem.

5.

Change the settings of the DIP switches on the modem for normal operation.

For example, on a US Robotics modem, use the following settings:

Function	Switch	Position
Suppress result codes	3	On
No echo, offline commands	4	Off
Dumb mode	8	On

Creating Dial-Up Networking

For computers with Windows 98 operating systems, create two connections; one for direct connect and one for remote connect.

If you are using a Windows 2000 Professional Operating system, refer to the procedure To Create Dial-Up Networking connection with Windows 2000 Professional.

To create a Dial-Up Networking connection for direct connect:

Note: This procedure applies to computers with Windows 98 operating systems.

1. On the **Start** menu, point to **Programs**.
2. Point to **Accessories**, then click **Dial-up Connections**.
3. Double-click the **Make New Connection** icon in the **Dial-Up Networking** window.
4. At the **Make New Connection** window, type in a <*name*> for the connection and select **NT Direct Connection** to bind it to the Phonebook entry. Click **Next**.
5. Enter the **Area Code** and **Telephone Number** and select a **Country code** from the drop down list. Click **Next** and then click **Finish**.
6. In the **Dial-Up Networking** window, right-click the new entry and click **Properties**.
7.

Click **Configure**:
- Maximum speed: **38400**
- Check: **only connect at this speed**

Under the **Connection** tab:
- Data bits: **8**
- Parity: **none**
- Select: **wait for dial tone before dialing**
- Select: **cancel the call time at 60 sec**.

8.

Click **Advanced**:
- Turn off: **error control**
- Turn off: **flow control**

Click **OK**.

9. Click the **Server Types** tab:
- Type of Dial-Up Server: **PPP: Windows, Windows NT3.5, Internet,** or **PPP:Internet**
- Advanced Options: select **Log onto Network** and **Enable software compression**
- Allowed network protocol: **TCP/IP** only

10.

Click the **Scripting** tab, and In the **Filename** field, type ***c:\Program Files\Mitel\Imat\pridun.scp***.

Note: If you have an IMAT version earlier than Release 6, enter *c:\mitel\isdn\imat* instead of *c:\Program Files\Mitel\Imat.*

11. Click **OK**.

To create a Dial-Up Networking connection at the IMAT PC modem:

Note: This procedure applies to computers with Windows 98 operating systems.

1. On the **Start** menu, point to **Programs**.
2. Point to **Accessories**, then click **Dial-up Networking**.
3. Double-click the **Make New Connection** icon in the **Dial-Up Networking** window.
4. At the **Make New Connection** window, type in a <*name*> for the connection and select your PC modem to bind it to the Phonebook entry. Click **Next**.
5. Enter the **Area Code** and **Telephone Number** to dial into the PRI T1 card. Select a **Country code** from the drop down list. Click **Next** and then click **Finish**.
6. In the **Dial-Up Networking** window, right-click the new entry and click **Properties**.
7. Click **Configure**:
   Under the **Connection** tab:
   - Data bits: **8**
   - Parity: **none**
   - Stop bits: **1**
8.

Click **Advanced**:
- Turn on: **error control** and select **Compress data**
- Turn on: **flow control** and select **Hardware**

- Ensure that **Modulation** is set to **Standard**

Click **OK**.

9. Click the **Server Types** tab:
   - Type of Dial-Up Server: **PPP: Windows, Windows NT3.5, Internet or PPP:Internet**
   - Advanced Options: select **Log onto Network** and **Enable software compression**
   - Allowed network protocol: **TCP/IP** only
10. Select the **Scripting** tab, and In the **Script File Filename** field, type ***d:\support\pridun.scp***. (if your CD-ROM drive letter is D:\).
11. Click **OK**.

To create a Dial-Up Networking connection with Windows 2000 Professional:

Typically, you will want to create two Dial-up Networking connections, one for the on-site direct access, and one for the remote modem access.

1. On the **Start** menu, point to **Programs,** point to **Accessories**, click **Communications,** and then click **Dial-Up Connections.**.
2. Double-click **Make New Connection,** and then click **Next.**
3. Select **Dial-Up to the Internet** and then click **Next.**
4. Select **I want to set up my Internet connection manually** or **I want to connect through a local area network**. Click **Next**.
5. Select **I connect through a phone line and a modem**, and then click **Next**.

6. Use the Com Port that has been configured as a NULL Modem connection: 38400, 8, none, 1. In the Choose Modem box, from the drop-down menu select **Communications cable between 2 computers**. Click **Next**.
7. Deselect the box **Use area code and dialing rules**, and then click **Advanced**.
8. For the Connection Type, select **PPP (Point to Point Protocol)**. For the Logon procedure, select **Use logon script**, and then click **Browse**. Select **pridun.scp or gateway.scp**. Click **OK**, and then **Next**.
9. In the Internet account logon information box, leave the username and password fields blank and then click **Next**.
10. Dialog boxes appear warning you that you will not be able to connect to your internet service provider without your user name and your password. Disregard these warnings and click **Yes** on these boxes to continue.
11. Enter the Connection name and then click **Next**.
12. In the box to set up an internet mail account, select **No** and then click **Next**.
13. De-select the option to connect to the internet immediately, and then click **Finish**.
14. In the Network and Dialog Connections window, right-click on the new DUN connection, point to **Properties**, and click **Configure**.
15. From the maximum speed (bps) drop-down list, select **38400** for the baud rate.
16. Click **OK** until you exit the windows.

Using the ISDN Maintenance and Administration Tool (IMAT)

IMAT provides an interface with the PRI card, through which you can perform daily operations plus perform the following tasks:

- Software upgrades
- Editing database files
- Collecting the maintenance logs from the PRI card
- Displaying the maintenance logs and Min/Max logs
- Initiating a reset on the PRI card
- Activating Min/Max profiles.

To log on to an IMAT computer:

1. If the computer is running, from the **Start** menu, click **Shut Down**.
2. Select **Close all programs and log on as a different user** or **Restart your computer**.
3. Launch IMAT from the desktop.
4. Proceed to **Installing and Configuring Software** or **Connect to Remote Site.**

To exit the IMAT application:

1. **Save** any open databases.
2. On the **File** menu, click **Exit**.

Accessing the PRI Card Maintenance Window

The PRI card maintenance window provides an interface with the PRI card. This interface is used primarily for debugging purposes. Logs and debugging information appear on the maintenance window in real time.

To access the PRI card maintenance window, use a communications package (for example, Hyperterminal) on the WIN95/98 or Windows 2000 Professional PC that is connected to the PRI card through the serial port. You can use a regular (straight), 9-pin to 9-pin female-to-male serial cable to connect the PC to the PRI card.

To access the PRI card's maintenance window

1. Connect the modem or the straight-through cable to the PRI card.
2. For VT100 terminal emulation (dumb terminal):
   - Run a communications package such as Hyperterminal™.
   - Ensure that the settings are 8 bits, no parity, no flow control,1 stop bit, 38400 baud rate.

   The following commands will be available from the PRI card maintenance window:
   -Option +/- cause: Turns on/off output of cause codes
   -Option +/- dispcall: Turns on/off calling/called party number display
   -Option state: Shows the states of the links from the PRI card perspective
   -Option state_help: Displays the state leter commands

Troubleshooting the Optioning Alarm

If the PRI card does not receive an optioning message from the system in 30 seconds, then an optioning alarm is sent to the PBX, a log is generated on the PRI card and the PRI card waits with the D-channels not in service. During this time, you can install a new database to the PRI card and reboot the PRI card.

Options between the PRI card and the PBX will not correspond if you program an option in IMAT that was not purchased. If options do not match with D-channel backup, QSIG, NFAS, and/or Network Side, then the following occurs:

- The D-channels are held down
- An optioning alarm is sent to the PBX
- The PRI card generates a log.

You can connect to the PRI card, determine the problem and reload a new PRI card database. You must then reset the PRI card.

If the options do not match with Min/Max or Auto Min/Max, the Min/Max datatbases will not take effect, and an entry indicating that the feature was selected but not purchased will appear in the PRI card maintenance log. An optioning alarm does not occur in this case. You select the Min/Max and Auto Min/Max options in the Site Options window.

Installing the PRI card

Install the PRI card in the following sequence:

- Configure the SX-200 CDE Forms to support the PRI card. See PRI support.
- Install the PRI card.
- Configure the IMAT database.
- Connect the computer to the PRI card.
- Save the IMAT database on the PRI card.
- Connect the PRI card to the ISDN network.
- Test the PRI card.

Configuring the CDE Forms

See PRI support in the Features section.

Installing the PRI card

To install the PRI card:

1. Attach the anti-static strap to your wrist.

   Caution: You can only install the PRI card in a SX-200 ELx cabinet. Inserting the PRI card in a peripheral interface card slot in any other SX-200 cabinet will cause the power supply to fail.

   Note: The SX-200 ICP must have a Stratum 3 clock module.

2. Unpack and inspect the PRI card (the PRI card comes with a T1/E1 module).Follow general procedures for handling circuit cards. The P12 and P32 connectors are for the T1/E1 module.The P311 connector is for the FIM II. The P31 connector is for the CIM. A FIM II or CIM is necessary only if the PRI card is installed in a peripheral cabinet.

   Note: You may change the LT/NT termination type (the transmit and the receive signals) by changing the stuffing of the jumper blocks on the LT/NT connectors on the T1/E1 module. Two LT/NT connectors serve the two T1 links on the T1/E1 module. The LT is for the network side and the NT is for the user side (the default setting).

3. If you are installing the PRI card into a peripheral cabinet (slot 10 or 11) place a FIM II on the PRI card and then connect a fiber to the FIM II.
   Remove the screws from the top of the standoffs.
   Connect the J3 connector on the FIM II to the P311 connector on the PRI card.
   Fasten the FIM II using the four standoffs.
   If you choose to use a CIM instead of a FIM II on the PRI card, then move the standoffs to the forward position. This procedure is the same as installing a CIM on a BCC III. Use the P31 connector and connect the appropriate copper cable (Standard Category 5 UTP with RJ45 connectors) from the main control cabinet to the CIM. Only one twisted pair cable is required for copper installation; the cable has both a TX and RX pair inside. Refer to the CIM Ports table that shows the pinouts for the copper cable.

   Note: If you are installing the PRI card in a main control cabinet, you have the option of using the PRI card in the control cabinet as a Control Dual FIM Carrier Card. You can connect a peripheral bay to the main control cabinet using a CIM or FIM II on the PRI card in the main control cabinet.
   When you want the PRI card to act as a Control Dual FIM Carrier Card, make sure to program System Options 71 and 72 in CDE Form 04 to reflect this.

4. Set the S1 switch on the PRI card to control the source of the clock and frame pulse signals (backplane or interface module). For a PRI card, using a backplane source in a main control cabinet, set the S1 switch to show 1 closed and 2 open. For a PRI card in a peripheral cabinet using a FIM II or CIM source, set the S1 switch to show 1 & 2 closed. The default setting is the setting for a PRI card without an interface module (FIM II or CIM).

   Note: If the S1 switch has positions 1 & 2 open, the PRI card will not function

5. In the SX-200 ELx cabinet Rev 4.4 or greater (PN 9109-600-002-NA) install the PRI card into slot 10 or 11. The SX-200 ICP must have a Stratum 3 clock module. Screw the PRI card in to ensure that the card is firmly seated in the cabinet. Only one PRI card is allowed in a main control cabinet. Up to two PRI cards are allowed in a peripheral cabinet.

   Note: If the SX-200 ELx cabinet is Rev 4.3 or less, the cabinet can support a PRI card if you use an RS-232

Adapter (PN 9109-632-001-NA) to connect the cabinet to the PC.

Note: Install the PRI card in a peripheral cabinet for a SX-200 ICP system with six or seven bays. This avoids the PRI card using the slot for the Control Triple FIM Carrier card in the main control cabinet

Caution: You can only install the PRI card in a SX-200 ELx cabinet. Inserting the PRI card in a peripheral interface card slot in any other SX-200 cabinet will cause the power supply to fail.

6.

Route the PRI cable from the Fiber/Cable Port at the rear of the cabinet to the front of the cabinet and connect the PRI cable to the appropriate RJ 45 connector on the faceplate of the PRI card.

If the PRI card is in a peripheral cabinet, loop back the fiber/cable to the rear of the cabinet, bring the fiber/cable through the Fiber/Cable Port and connect the fibe/cable to the FIM or CIM on the FIM or CIM carrier card in the main control cabinet.

7. Connect the PRI cable(s) to the ISDN Network.

8. Connect the PRI card to the computer. Use J10 if the PRI card sits in slot 10; J11 for slot 11.

Configuring the IMAT database

To create and configure a new PRI database:

1. Run the IMAT application.
2. From the **File** menu, select **New Database**.
3. Select the **ISDN/PRI System** and with the drop drown menu select **PRI Card Release 8.**
4. Click on **OK**.
5. Click on the **Config/System Configuration/Site Options** menu item to enable site specific options. Select **PRI Card** for the System Type, **SX-200 ICP** for Connected Platform, and then select the options. Click **Update** and then **Close**.
6. Click on the **Config/System Configuration/PRI Link Characteristics** menu item to program PRI Link Characteristics.
7. Click on **Config/Incoming Call Characteristics** to program the **Incoming Call Characteristics**.
8. Click on **Config/Outgoing CPN Setup** to program the **Outgoing CPN Substitutions Setup**. (optional)
9. Program the **Outgoing Call Characteristics**. See Digit Modification.
10. Program a **Min/Max** database (optional).
11. **Save** the database.

Note: For more information, see the IMAT online Help.

Connecting the Computer to the PRI Card

To connect the computer to the PRI card:

1. Connect the serial cable from the computer's COM port to the serial port on the rear of the cabinet (J10 or J11).
2. On the **File** menu, click **Connect to Remote Site**.
3. In the **Dial-Up Entry** box, select the appropriate settings for the PRI card.

   Note: The PRI card does not require a password.

4. Click **Connect**.
5. In the **Connected to remote site** window, click **OK**.

   Note: A networked computer running Win95/98 has difficulties communicating using Dial-Up Networking. It is strongly suggested that a non-networked computer be used.

Saving the IMAT database onto the PRI card

To save the IMAT database onto the PRI card:

1. Log on to the IMAT PC.
2. On the **File** menu, click **Connect to Remote Site**.
3. In the **Dial-Up Entry** box, select the appropriate PRI Card settings.
4. Click **Connect**.
5. On the **File** menu, point to **Save**, and then click **Database**.
6. Select the connected ISDN interface from the **Destination** list.
7. Click **Save**.

Connecting the PRI card to the ISDN network

To connect the PRI card to the network:

1. Obtain the ISDN PRI cables (Category 5 cable with a RJ-45 connector at the PRI card end).
2. Plug the ISDN PRI cable into the PRI port(s).
3. Plug the ISDN PRI cable(s) into the network terminating equipment.

Note: A dual-port card needs two ISDN PRI cables if you are using both ports.

Testing the PRI card

If the installation is successful, the PRI card will boot up, configure itself with the default database, and communicate with the PBX.

Note: You may look at the log messages while the card boots up using a terminal emulator program on the PC.

To test the PRI card:

1 Inspect the LEDs on the faceplate to verify that the card is functioning. See Faceplate LEDs.

2 Open the PBX maintenance window.

3 Verify that the trunks associated with the PRI channels are all in **IDLE** state.

Default PRI Card Configuration

The PRI card comes pre-configured with the following database.

PRI Configuration Default Values	
Parameter	**T1 Default Setting**
PBX variant	SX-200 ICP
Maximum number of links programmed	2
ISDN protocol variant	NI2 Bellcore
Incoming calling ID (CPN) presentation [CLIP]	ON
Incoming called party number (CPN) presentation [DDI]	ON
Outgoing default calling party number (CPN) [CLIP]	Blank
Outgoing bearer capability selection	Fixed - voice
Outgoing HLC selection	n/a
Outgoing NSF selection	Fixed - default
Outgoing calling party number restriction [CLIR]	Fixed - allow for voice
Min/Max	None

Upgrading or Re-installing PRI Software

Use the IMAT application to upgrade the software. For more information on upgrading releases, refer to IMAT online Help.

To upgrade a PRI card to a new version of software or re-install the software:

1 Ensure that the PRI card is running by checking the card's PBX status LEDs. See the PRI card faceplate and Faceplate LEDs.

2 Connect a computer to the RS-232 serial port of the PRI card.

3 Run IMAT.

4 Click on File/Connect to connect to the card.

5 Download the software from Mitel Online or insert the software CD-ROM in the computer's CD-ROM drive.

6 Click on Maintenance/Software Upgrade and proceed with the software upgrade or re-installation.

7 Reset the PRI card.

Resetting the PRI card

There are three methods for resetting the PRI card. Do one of the following:

1. Enter the **LOAD <plid>** command in the maintenance window of the PBX.
2. Reset the PRI card by unseating and reseating the card.
3. From IMAT, click on the Maintenance/Remote Site Reset menu item.

Installing Peripherals

The following table lists optional equipment that can be installed on the SX-200 ICP. Not all variants of the SX-200 ICP controller support the devices listed.

A full list of Mitel peripherals is in the Field-Replaceable Units section (see the Part Numbers for Peripheral Equipment).

Table: Peripheral Devices Installed on the SX-200 ICP

Peripheral Device	Chart/Section
SUPERCONSOLE 1000 Attendant Console	Chart
SUPERCONSOLE 1000 installed as a maintenance console	Chart
5540 IP Console	Chart
Mitel IP Phones	Connect the IP Phones
SUPERSET telephones 400- and 4000-series telephones	Chart
Programmable Key Modules	Chart
System printer	Chart
Night Bell	Chart
Paging equipment	Chart
Music-on-Hold equipment	Install Music-on-Hold Equipment
Door Phone equipment	Chart
Alarm Device	Chart
SUPERSET Interface Module	Chart
5421 and 5422 Interface Module	Chart
Mitel 5303 and 5310 Conference Units	Install a Mitel Conference Unit
DSS/BLF Interface Unit	Install a DSS/BLF Interface Unit

System Fail Transfer Unit (Systems with Peripheral Cabinets only)	Install a System Fail Transfer Unit
Music on Hold Equipment and DNIC Music-on-Hold/Pager Unit (DMP)	Install Music on Hold Equipment

Install a SUPERCONSOLE 1000 Attendant Console

General Description

The SUPERCONSOLE 1000 attendant console is used to perform call handling functions as well as some maintenance and administrative functions (such as moves and changes). The 4-line by 80-character alphanumeric display shows source and destination information, time and date information, call waiting information, and station information (such as COS and COR values). Macros can be programmed to facilitate the transfer of calls to voice mail, recover calls released to the wrong extension, dial frequently called numbers using one button.

The console has

- Fourteen hardkeys
- Four programmable firmkeys (for access to purchased options such as Hotel/Motel)
- Ten softkeys
- A dial pad (for both alphabetic and numeric input)
- Backlit display
- Volume controls
- Integral handset
- Connector for a headset
- An RS-232 serial printer port.
- A PKM port for the connection of up to two PKM 48 devices.

WARNING: Any connection of this console to an off premise application, an out of plant application, or to any other exposed plant application may result in a safety hazard, and/or defective operation, and/or equipment damage.

Notes:

1. The SUPERCONSOLE 1000 Attendant Console uses a two-wire connection to connect to one DNIC circuit in the controller or on a Digital Line Card or DNIC Module (Tip to Tip and Ring to Ring) in a Peripheral Cabinet. The Digital Line Card must be in a high-powered slot. Four consoles are allowed per digital line card.
2. The system maximum is 11 consoles (applies to MX systems with Peripheral Cabinets).
3. The SUPERCONSOLE 1000 Attendant Console can be used in the maintenance terminal port, by changing system default programming. Refer to the Installation Information section.
4. The password must be numeric only to use the console as a maintenance terminal.
5. The console operates in either English, French, or Spanish.
6. The console has a programmable printer port.

Environmental Specifcations

	Temperature	Humidity
Operating Environment	32° to 86°F (0° to 30°C)	20% to 80% RH, non-condensing

Shipping/Storage Environment	-4° to 140°F (-20° to 60°C)	10% to 70% RH, non-condensing

Installation

Chart: Install a SUPERCONSOLE 1000 Attendant Console

Step	Action	Comments
1.	Inventory, Unpack, Inspect	
	Check off received items against packing list and equipment list.	
	Use only the Mitel AC adapter (PN 700063020) to power the console.	
	This adapter powers the back-lit display, the printer port and the optional PKM 48 devices. The console supports PKM 48 devices or a printer, not both.	
2.	Unpack cartons.	
3.	Assemble Console	
	Connect the headset (or handset) to its cord.	
4.	Plug the cord into the jack at the side of the console.	The two jacks are connected in parallel; either can be used.
5.	Position Console	
	Put the console in its assigned position.	
	Maximum loop length is 1000 m (3300 ft).	
6.	Plug one end of the console modular cord into the jack at the back of the console, and the other end into its assigned modular telephone jack.	The console jack is marked LINE PORT.

7\.

Connect Console

Connect the modular telephone jack to the cross-connect field.

The green lead connects to a DNIC port Tip circuit.

The red lead connects to a DNIC port Ring circuit.

8.	Connect the cross-connect field to a DNIC port on the SX-200 ICP controller or SX-200 Peripheral cabinet.	In a busy environment , we suggest using the first DNIC port on the card for the console and second DNIC port for the DSS/BLF Interface Unit.

9\.

Connect the Printer (Optional)

Connect the printer to the RS-232 printer connector on the back of the console.	Maximum 50 ft between the printer and the SUPERCONSOLE 1000 Attendant Console printer port.	
10.	Set the printer baud rate..	Maximum baud rate is 2400 baud.
11.		

Cross Connect

At the distribution frame, cross connect the cable from the modular jack to the cable from the DNIC port.

Refer to the Installation Information section, for cross-connect cabling information.

12\.

Program the Console

Program a DNIC circuit in Form 01, System Configuration. Assign an extension number, COS, COR, and Tenant for each console bay/slot/circuit entry in Form 07, Console LDN Assignment. Assign a directory number and label to each required console LDN key in Form 08, Console LDN Assignment.	

RS-232C Connector

A printer can be attached to the SUPERCONSOLE 1000 Attendant Console via its RS-232C port connector which allows direct connection to most printers. The RS-232C interface is programmable from 110 to 2400 baud.

The RS-232C connector (J3) is configured such that the console is the data communications equipment (DCE) to allow direct connection to most serial printers. The RS-232C Port Pin Configuration - DCE Table lists the RS-232C pins recognized by the console's printer port.

RS-232C Port Pin Configuration - DCE

Pin	Designation	Status	Console Printer Port
1	Frame Ground	- -	Connected
2	Tx Data	Input	Supported
3	Rx Data	Output	Supported
4	RTS (Request to Send)	Input	Supported
5	CTS (Clear to Send)	Output	Supported
6	DSR (Data Set Ready)	Output	*
7	Digital Ground	- -	Connected
8	DCD (Data Carrier Detect)	Output	*
20	DTR (Data Terminal Ready)	Input	Supported
Note:	* DSR and DCD signals are wired together, driven from a single RS-232C output.		

Install a SUPERCONSOLE 1000 Attendant Console as a Maintenance Console

The following chart describes installation of a SUPERCONSOLE 1000 Attendant Console as a maintenance console and connection of an optional printer to it.

Chart: Install a SUPERCONSOLE 1000 Attendant Console as a Maintenance Console

Step	Action	Comments

Chart: Install a SUPERCONSOLE 1000 Attendant Console as a Maintenance Console		
Step	**Action**	**Comments**
	Connect Console	
1.	Connect the console to its assigned modular wall jack.	A DNIC circuit is required for a SUPERCONSOLE 1000 Attendant Console.
2.	Use only the Mitel AC adapter (PN 700063020) to power the console.	This adapter powers the back-lit display, the printer port and the optional PKM 48 devices.
	Connect Printer (Optional)	
3.	Connect the printer to the RS-232 printer connector on the back of the console. Maximum 50 ft between the printer and the printer port on the SUPERCONSOLE 1000 Attendant Console.	**Pin** **Signal** 1 frame ground 2 transmit data 3 receive data 4 ready to send 5 clear to send 6 data set ready 7 signal ground 8 carrier detect 20 data terminal ready
4.	Set the printer baud rate.	Maximum baud rate is 2400 baud.

Install a 5540 IP Console

General Description

The 5540 IP console is used to perform call handling functions as well as some maintenance and administrative functions (such as moves and changes). The 4-line by 80-character alphanumeric display shows source and destination information, time and date information, call waiting information, and station information (such as COS and COR values).

The console has

- Fourteen hardkeys
- Ten softkeys
- A dial pad
- Backlit display
- Volume controls
- Contrast controls
- Integral handset
- Connector for a wired headset
- Connector for cordless headset
- A PKM port for the connection of up to two PKM 48 devices.

WARNING: Any connection of this console to an off premise application, an out of plant application, or to any other exposed plant application may result in a safety hazard, and/or defective operation, and/or equipment damage.

Notes:

1. The system maximum is 11 consoles.
2. The 5540 IP Console can be used as a maintenance terminal by changing system default programming. Refer to the Installation Information section.
3. The password must be numeric only to use the console as a maintenance terminal.
4. The console operates in either English, French, or Spanish.
5. The console does not have a printer port. Printing is requested through the network to which the console is connected.

Environmental Specifcations

	Temperature	Humidity
Operating Environment	32° to 86°F (0° to 30°C)	20% to 80% RH, non-condensing
Shipping/Storage Environment	-4° to 140°F (-20° to 60°C)	10% to 70% RH, non-condensing

Installation

Refer to the *5540 IP Console Installation Guide* for full instructions.

Chart: Install 5540 IP Console

Step	Action	Comments
1.	Inventory, Unpack, Inspect	
	Check off received items against packing list and equipment list.	In the absence of Power-over-Ethernet (POE), use a qualified Mitel DC adapter (PN 50005301) to power the console. Power is provided for the back-lit display and the optional PKM 48 devices.
2.	Unpack cartons.	
3.	Assemble Console	
	Connect the headset (or handset) to its cord. For cordless headset operation, refer to the *5540 IP Console User Guide*.	
4.	Plug the cord into the jack at the side of the console.	The two jacks are connected in parallel; either can be used.

5.		
Position Console		
Put the console in its assigned position.		
6.	Plug the LAN cable into the LAN port on the back of the console. Plug the other end of the LAN cable into the network jack or DC adapter.	The console jack is marked.
7.		
Program the Console		
Assign an extension number, COS, COR, and Tenant for each console bay/slot/circuit entry in Form 07, Console LDN Assignment. Assign a directory number and label to each required console LDN key in Form 08, Console LDN Assignment.		

Installing Mitel IP Phones and IP Consoles

IP Phone and IP Console Installation are almost the same. The term IP Phone will refer to both IP Phone and IP Console in this section.The term IP Console will be used only when applicable.

Connect each IP phone to the Ethernet Switch.

Powering IP Phones differs between the SX-200 ICP MX, CX/CXi and AX controllers. The CXi controller contains an L2 switch that supplies power to the phones while the MX, AX and CX provide power using an external supply (such as a 24-volt adapter or 48-volt power brick), an in-line power unit (such as the PowerDsine 24PT In-line Power Unit), or a Layer 2 switch with integral power feeds. For more information on powering IP phones with an MX or CX controller, refer to the Feeding Power to the Phones section and to the Engineering Guidelines document on the Edocs website (http://edocs.mitel.com).

After you connect an IP phone to an Ethernet port, you can connect another IP device, such as a PC, to the Ethernet port on the IP phone. This allows you to have two IP devices connected to one Ethernet drop.

Note: Use of the second IP port is a FOR option feature (System Option 131, COS Option 280).

Registering the MAC Addresses of IP Phones

Before you can use the IP phones, you must register their MAC addresses with the controller. You can use CDE or the phones themselves to complete this task.

Using CDE to Register

This procedure uses CDE to manually add the MAC address of each IP set or IP Console to the SX-200 ICP database.

Before you begin

- Ensure you have the MAC address for each phone you intend to register in CDE.

- Program the internal DHCP server on the SX-200 ICP.

To register a MAC address using CDE (IP Phones)

1. In CDE Form 09 (Desktop Device Assignments):
   - select a free PLID
   - program the Extension Number, Device Type, Tenant Number, COS, COR, and Name
   - enter the MAC Address

Note: For the MAC Address, enter letters and numbers only. Upper and lower case letters are permitted, but not symbols such as "-." Do not confuse the phone's serial number with its MAC address.

2. Use CDE to program all other station features (for example, Interconnect Restriction and Set Key Assignments).
3. Connect the IP phone to an RJ-45 Ethernet port on the LAN.

   The set registers with the SX-200 ICP. If the phone is a 5201 IP Phone, its Message Waiting lamp is extinguished after registration is complete (20 - 40 seconds).

To register a MAC address using CDE (IP Console)

1. In CDE Form 07 (Console Assignment):
   - select a free PLID
   - program the Extension Number, Tenant Number, COS and COR.
   - press the Show MAC softkey and enter the MAC Address

Note: For the MAC Address, enter letters and numbers only. Upper and lower case letters are permitted, but not symbols such as "-." Do not confuse the phone's serial number with its MAC address.

2. Connect the IP Console to an RJ-45 Ethernet port on the LAN. The port should be POE enabled.

   The console registers with the SX-200 ICP.

Using a Phone to Register

This procedure registers the MAC address of IP sets or IP Consoles with the SX-200 ICP database. You do not need to register the IP telephones if you entered the MAC address of each IP telephone in CDE Form 09, Desktop Device Assignments, and for IP Consoles in CDE form 07, Console Assignments.

Note: 5201, 5215, and 5010 IP phones will fail to register on a system that has a Default or Premier database (MX only) because of the line appearances programmed on keys 8 and 10—keys that exist on the 5207 but not on the 5201, 5215 or the 5010. To register these phones, first delete the line appearances in Form 09 (the 5201 must have all appearances deleted) or follow the procedure for replacing a registered phone and re-programming the circuit.

Before you begin

- Ensure you have the access code—if you programmed one—for set registration in CDE Form 02, Feature Access Codes.
- For IP Phones - Ensure the directory number and device type is programmed in CDE Form 09, Desktop Device Assignments.
- For IP Consoles - Ensure the directory number is programmed in CDE Form 07, Console Assignments.

To register a MAC address using an IP phone

1. Connect the IP phone to an RJ-45 Ethernet port on the LAN.
2. Press * during power-up (to clear any directory number in memory).

   The phone displays "Enter the PIN number." On 5201 IP Phones, the Message Waiting lamp is lit.
3. At the IP phone, enter the IP Set Registration PIN access code (default, ***) followed by the IP Phone extension number. For example:

***100

Note: The directory number must be in the database; otherwise, the phone will fail to register.

4. Press Superkey (or Hold for 5201 and 5207 IP phones, Hold1 for 5540 IP Console).

   The set registers its MAC address with the SX-200 ICP.

Note: Use CDE Forms to program all other station features (for example, Class of Service, Interconnect Restriction, Set Key Assignments, and Class of Restriction).

Assigning Static IP Addresses to IP Phones (Optional)

To navigate through the set prompts:

- Use the Volume Down key to move to the next field.
- Use the Volume Up key to go back to the previous field.
- Use the * key to move backwards within a field (to correct an error).
- Use the # key to insert a decimal character and move to the next IP address field.

To set static IP address on the IP telephones:

1. Ensure that the set you want to program is not connected to the network and there is no power supplied to the set.
2. Hold down the Volume Up key for 3 seconds and at the same time, plug the set Ethernet cable and power into the set to display the STATIC IP SETUP MODE.

   The SET STATIC IP PARAM? prompt appears.
3. To begin entering static IP address information, press #.

   Wait a few seconds until the USE PRESENT SETTINGS prompt appears.
4. Press # to enter IP data. To revert back to DHCP from static parameters, press *.

   The INPUT VLAN ID prompt appears.
5. If you are using VLANS, enter the VLAN ID that will be inserted into packets sent by the phone. Then press the Volume Down key to move to the next field. If you are not using VLANS, enter nothing and press the Volume Down key.

   The INPUT PRIORITY prompt appears.
6. Enter 6 if you are using priorities, or leave the field blank. Press the Volume Down key to move to the next field.

   The INPUT IP ADDRESS prompt appears.
7. Enter the customer-supplied static IP address (for example: 10.30.27.191).

Note: Enter two-digit portions of IP addresses as three-digit numbers with leading zeros. For example, enter '10.12.67.30' as '010.012.067.030.

Note: If you see the INVALID IP ADDRESS message, press the Volume Up key to return to the field with the incorrect entry. Edit the entry.

   The INPUT PDA ADDRESS prompt appears.

8. Enter the customer-supplied static IP PDA address. Press the Volume Down key to move to the next field.

   The INPUT SUBNET MASK prompt appears.
9. Enter the subnet mask and press the Volume Down key to move to the next field.

   The INPUT DEFAULT GATEWAY prompt appears.
10. Enter the IP address of the Router and press the Volume Down key to move to the next field.

    The INPUT RTC ADDRESS prompt appears.
11. Enter the IP address of the RTC and press the Volume Down key to move to the next field.

    The INPUT TFTP ADDRESS prompt appears.

12. Enter the IP address of the TFTP server that is used to download the main-load and boot-load images. Press the Volume Down key to move to the next field.

    The INPUT DNS ADDRESS appears.

13 Enter the IP address of the server that will be used during Web browsing to resolve host names into IP addresses. Press the Volume Down key to move to the next field.

    The INPUT WINS ADDRESS prompt appears.

14. Enter the Windows Internet Name Services (WINS) server IP address. For more information about this address, consult your Microsoft Windows documentation. Press the Volume Down key to move to the next field.

    The INPUT PROXY ADDRESS prompt appears.

15. Enter the IP address of the proxy server. Press the Volume Down key to move to the next field.

    The INPUT PROXY PORT prompt appears.

16 Enter the Proxy Port number and press the Volume Down key to move to the next field.

    The TECHNICIAN IP ADDR? prompt appears.

17. To enter the IP address of the debugging utility, press #, enter the address and then press the Volume Down key to move to the next field. If you do not want to enter this information, press * .

    The STORE IN NVRAM? prompt appears.

18. To store parameters in non-volatile RAM, press # . This ensures that your static settings will be used when the set is powered-up and when the FLASH software is upgraded. If you do not want to store the static settings in NVRAM, press *.

The set reboots and applies the new static IP data.

Removing Static IP Addresses on the IP Sets

To return to using DHCP when static parameters were previously enabled:

Plug the set cable and power into the set while holding down the Volume Up key for 3 seconds to display the STATIC IP SETUP MODE.

At the USE PRESENT SETTINGS screen, select * = DISABLE to revert back to DHCP from static parameters.

Program Features for each Phone

Before you begin

- Ensure that the appropriate FOR options are specified in CDE Form 04, System Options.
- Ensure that the IP phones have IP addresses acquired via any of the following:
  - External DHCP server
  - Internal DHCP server (enter the addresses in Form 47, DHCP Parameters)
  - Static IP address programmed on the phone (see Assigning Static IP addresses to IP Phones)

  Once the IP address are acquired, register the phones with the system by using the PIN registration procedure or by entering the MAC address in Form 9 (Form 7 for IP Consoles).
- Ensure that IP Telephone Registration access codes and Set Replacement Access Codes are specified in Form 02, Feature Access Codes.
- Ensure that Class of Service, Interconnect Restriction, and Intercept Handling is programmed for each device.

To program features for IP phones:

- Program the appropriate features for each IP phone.

Install a SpectraLink Wireless Telephone

Description

See SpectraLink Wireless Telephones in the Program Features section.

Installation

The chart below summarizes the configuration requirements for all SpectraLink and other associated third-party devices. For detailed instructions, refer to the following SpectraLink documentation on the SX-200 ICP software CD:

- NetLink SVP Server - Installation, Setup, and Maintenance
- NetLink Open Applications Interface (OAI) Gateway - Installation
- NetLink e340/h340/i640 Wireless Telephone; Mitel SX-200 ICP with Mitel 5220 IP Phone emulation - Setup and Administration
- NetLink Wireless Telephone Access Point Compatibility

See also the documentation supplied with the Access Points selected for the installation.

Chart: SpectraLink Installation Checklist

Notes:

1. The menus and options in the examples may vary slightly depending on the version of software in the device or the device type.
2. Settings not explicitly configured should be left at their default values.

Device	Parameter	Requirement	Example
Netlink Telephones	ESS ID	Ensure that ESSID matches Access Point ESSID	ESS ID -> Static Entry -> 123456
License Mgmt	Ensure that selected license setting allows for TFTP Server IP Address Input.	License Mgmt -> Set Current -> Type 014	
Security	Ensure that selected security matches Access Point Security Settings.		

Security -> WEP -> Authentication -> Open System

Security -> WEP -> WEP On/Off -> WEP On
Security -> WEP -> Key Information -> Default Key -> 1
Security -> WEP -> Key Information -> Key Length -> 40-Bit
Security -> WEP -> Key Information -> Key #1 -> 1111111111
Security -> WEP -> Key Information -> Key #2 -> 2222222222
Security -> WEP -> Key Information -> Key #3 -> 3333333333
Security -> WEP -> Key Information ->

Key #4 -> 4444444444 Security -> WEP -> Rotation Secret -> 1 **NOTE:** Please be aware that once a WEP Key has been entered, it will not be displayed when re-accessing that Key Information menu. CAREFULLY enter the required Key to prevent typos.		
IP Addresses	Ensure that the defined IP addresses match the host Access Point subnet settings using either Static or DHCP configuration.	

Static settings:

IP Addresses -> Static IP -> Phone IP -> 192.168.1.101 IP Addresses -> Static IP -> TFTP Server IP -> 192.168.1.2 IP Addresses -> Static IP -> Default Gateway -> 192.168.1.1 IP Addresses -> Static IP -> Subnet Mask -> 255.255.255.000 IP Addresses -> Static IP -> SVP IP Addr IP Addresses -> Static IP -> RTC IP Addr -> 192.168.1.2 **DHCP settings**: 3 - Default Gateway IP - default is 192.168.1.1 66 - SpectraLink firmware TFTP server IP - default is 192.168.1.2 129 - SX-200 ICP IP (RTC IP) - default is 192.168.1.2 130 - DHCP Server Identifier - default is MITEL IP PHONE 151 - SVP server IP (used for voice prioritization)		
Access Point (Symbol)	ESS ID & IP Addresses	Ensure that the correct IP settings for the Access Point, as well as the desired Net_ID (ESS ID), are configured correctly.

IP Address - 192.168.0.25

Gateway IP Address - 192.168.1.1 DNS IP Address - 192.168.0.200 Net_ID (ESS) - 123456 Additional DNS - 192.168.0.201	
Special Functions	Ensure that the desired wireless security settings are defined correctly.

Configure Authentication and Encryption -> Pre-shared Key -> Enabled

Configure Authentication and Encryption -> WEP -> 40 bit Configure Authentication and Encryption -> Configure WEP/KeyGuard -> Encryption Key ID -> 1

Configure Authentication and Encryption -> Configure WEP/KeyGuard -> WEP/KeyGuard Key Maintenance -> Key 1 -> 11111 11111 Configure Authentication and Encryption -> Configure WEP/KeyGuard -> WEP/KeyGuard Key Maintenance -> Key 2 -> 22222 22222 Configure Authentication and Encryption -> Configure WEP/KeyGuard -> WEP/KeyGuard Key Maintenance -> Key 3 -> 33333 33333 Configure Authentication and Encryption -> Configure WEP/KeyGuard -> WEP/KeyGuard Key Maintenance -> Key 4 -> 44444 44444

Set System Configuration	Ensure that Access Control is enabled in order to make the wireless network more secure.	Set System Configuration -> Access Control -> Allowed
Set Access Control List	Ensure that devices allowed on the wireless network have their MAC Address entered correctly in the list.	Set Access Control List -> Address Type -> Individual -> Add-[F2] -> 08:00:0F:01:02:03
Netlink SVP Server	SVP-II Configuration	Ensure that the proper settings for the NetLink wireless phones SpectraLink Voice Priority management are entered correctly

SVP-II Configuration -> Phones per Access Point -> 4

SVP-II Configuration -> SVP-II Master -> 192.168.0.20 SVP-II Configuration -> First Alias IP Address: -> 192.168.0.21 SVP-II Configuration -> Last Alias IP Address: -> 192.168.0.24 **NOTE:** Please ensure that the defined Alias IP Addresses Range does not overlap with any other device, including NetLink Wireless IP Phones, located on the same subnet.

Network Configuration	Ensure that the correct IP Settings for the SVP Server are configured correctly.

Network Configuration -> IP Address -> 192.168.0.20

Network Configuration -> Subnet Mask -> 255.255.255.000 Network Configuration -> Default Gateway -> 192.168.0.1 Network Configuration -> SVP-II TFTP Download -> Netlink IP

IMPORTANT: Ensure that the Access Point(s) in use is/are on the same subnet as the SVP Server. Each subnet must have its own SVP server:

Install a SUPERSET 400 or SUPERSET 4000 series telephone

Description

The SUPERSET 400 and 4000 series telephones are digital sets that connect to DNIC circuits in the SX-200 ICP controller or Digital Line cards installed in peripheral cabinets.

Note: The 400 series telephones have been discontinued but are still supported on the MX controller only.

Environmental Specifications

	Temperature	Humidity
Operating Environment	32° to 122°F (0° to 50°C)	0% to 90% RH, non-condensing
Shipping/Storage Environment	-13° to 158°F (-25° to 70°C)	0% to 90% RH, non-condensing

Installation

WARNING: Any connection of a SUPERSET 400 or SUPERSET 4000 set to an off-premise application, an out-of-plant application, or to any other exposed plant application may result in a safety hazard, and/or defective operation, and/or equipment damage.

CAUTION: Do not connect SUPERSET 400 or SUPERSET 4000 series telephones in parallel, to standard lines, or as power fail transfer extensions. Do not use a hand test telephone (butt-in) to check a SUPERSET 400 or SUPERSET 4000 series telephone line (its digital line card does not have a loop detector).

Chart: Install a SUPERSET 400 or SUPERSET 4000 series telephone	
Step	**Action**
1.	

Complete the cabling and the following CDE programming:

Enable the phones in Form 04 (System Options/System Timers) Assign the phones to system in Form 09 (Desktop Device Assignments) Select COS Options in Form 03 (COS Define).	
2.	Connect the handset cord to the handset and to the telephone.

3.	Write the telephone's extension number on the identification card and install the card and plastic lens on the telephone.
4.	For multiline sets, write the line appearance numbers and features on the designation card and install the card and plastic lens on the telephone.
5.	Connect the line cord to the telephone and the telephone jack.

Install a Programmable Key Module

Description

A Mitel Programmable Key Module (PKM) adds up to 96 programmable keys to a Mitel telephone or SUPERCONSOLE 1000 or a 5540 IP Console.

Table: PKM Models

Model	Number of Keys	Connects to	Number of PKMs that can be attached
PKM 48	48	SUPERSET 4025 SUPERSET 4125 SUPERSET 4150 SUPERCONSOLE 1000	2
PKM 12	12	SUPERSET 4025 SUPERSET 4125 SUPERSET 4150	1
5415 PKM	48	5020 IP	2
5410 PKM	12	1	
5448 PKM	48	5220 IP 5224 IP 5324 IP	2
5412 PKM	12	1	

Requirements

Interface Modules/Units

- The PKM 48 and PKM 12 require a SUPERSET Interface Module (SIM1 or SIM2) in the attached phone.
- The SUPERCONSOLE 1000 with part numbers 9189-000-300 and 9189-000-301 can directly connect up to two PKM 48 devices. The older models require a Direct Station Select/Busy Lamp Field (DSS/BLF) Interface Module to associate up to two PKM 48 devices with the attendant console.
- The PKM 5415 and 5410 require a Mitel 5421 Interface Module in the attached phone.

- The 5448 PKM and 5412 PKM require a Programmable Key Module Interface Module (5422 PKM IM) in the attached phone.

Power

The IP PKM Interface Module (IM) supplies the power to the 5412 (IP PKM 12) or the 5448 (IP PKM 48). There are different powering requirements depending on which phone the IP PKM 12 and IP PKM IM is attached to, as follows:

5220: The DC adapter (24 v) that powers the host 5220 IP Phone powers the IP PKM IM and the IP PKM 12. The DC adapter is always required to power the IP PKM IM when used on a 5220 IP Phone.

5224/5235/5324: The IP PKM IM supports Power over Ethernet (PoE) from the Phone to power the IP PKM 12. The DC adapter (24 v) is not required when using a PKM 12 on a 5224/5235/5324 IP Phone that is powered by PoE.

Table: Power Requirements

Host	Adapter Connects to...	Adapter Included?	Voltage	Part Number
4025	SIM 1/SIM 2	No	12V	50000690 50002790*
4125	Phone (see Warning)	Yes		
4150				
5020 IP	Phone	No	24V	
5220 IP				
5224 IP				
SC1000				

Console (Backlit version - PN 9189-000-300/1only)

or DSS/BLF Interface Unit

No

Yes	12V	700063021

* Universal model

Environmental Specifications

	Temperature	Humidity
Operating Environment	32° to 95°F (0° to 35°C)	0% to 90% RH, non-condensing
Shipping/Storage Environment	-13° to 158°F (-25° to 70°C)	0% to 90% RH, non-condensing

Installation

WARNING **Any connection of this set to an off premise application, an out of plant application, or to any other exposed plant application may result in a safety hazard, and/or defective operation, and/or equipment damage.**

Chart: PKM Installation Instructions

Step	Action
Attaching a PKM to a Telephone	
1.	Before installing a PKM, associate it with the host telephone in CDE Form 9. The PKM will not function until it is programmed in CDE.
2.	Install the appropriate interface module in the telephone. For SUPERSET telephones, see Install a SUPERSET Interface Module; for IP telephones, see Installing a 5421 or 5422 Interface Module.
3.	Attach the 12 button PKM to the host telephone: a. Remove the two housing screws on the underside of the host telephone. b. Attach the PKM to the telephone using the screws supplied with the PKM; see the illustrations for the PKM 12 and the 5412 PKM.
4.	Ensure that the SUPERSET 4025, SUPERSET 4125, or 4150 telephone has the latest firmware upgrade. Use the Show Status command in maintenance to check.
5.	Write the extension numbers of the line appearance keys and the functions of the feature keys on the key overlay, position the key overlay on the PKM and install the plastic lens.
6.	Connect the telephone to the system.
7.	Connect the appropriate AC adapter to the interface module or to the underside of the telephone itself as indicated in the above table. Ensure that the adapter supplies the correct voltage (12V or 24V).

WARNING:
Never plug a power supply into the interface module installed in a SUPERSET 4125 or SUPERSET 4150 telephone. A SUPERSET 4150 would lose the full duplex functionality of the speaker phone if the power supply is plugged into the SUPERSET interface module (SIM1 or SIM2).

8.	Plug one end of the modular cable into the jack located on the underside of the PKM.
9.	Plug the other end of the modular cable into the jack on the telephone's interface module. The PKM key 1 temporarily flashes. Continual flashing of the PKM key 1 indicates incorrect installation.
10.	

Route the cables as shown in the following illustrations:

48 button PKMs

SUPERSET 4025

SUPERSET 4125 and 4150

5020, 5220, 5224 and 5324IP Phone

12 button PKMs

SUPERSET 4025

SUPERSET 4125 and 4150

11.

Program the keys as described in the telephone user guide. Some features can only be programmed through CDE.

See the following illustrations for the key ID numbers required when programming using CDE:

48 button PKM key IDs

12 button PKM key IDs

Attaching a PKM to a DSS/BLF Interface Unit

1.	Refer to DSS/BLF Interface Unit.

Attaching a Second PKM 48

1.	Ensure that the telephone or DSS/BLF Interface Unit and first PKM 48 are configured and powered as described above.
2.	Plug one end of the modular cable into the jack on the underside of the first PKM.

3.	Plug the other end of the modular cable into the jack on the underside of the second PKM. The PKM key 1 temporarily flashes. Continual flashing of the PKM key 1 indicates incorrect installation.
4.	Route cables and program the second PKM as described above.
5.	Write the extension numbers of the line appearance keys and the functions of the feature keys on the key overlay, position the key overlay on the PKM, and install the plastic lens.
6.	Position the PKMs next to the telephone or console on a flat surface.

Visual Indication of Error Conditions

The following table provides some basic troubleshooting information for PKMs.

PKM Error Conditions

Symptoms	Error Condition	Possible Corrective Action
PKM key 1 (lower left hand side) continues to flash.	**IP Phones:** The host phone is not registered with the controller. **DNIC Phones:** The DNIC port is not in synchronization.	Make sure the IP Phone is registered. Check the interface module (SIM1, SIM2 or AIM), the MILINK cable, the PKM, and the telephone set. For DNIC phones, check that the firmware for the telephone set is Firmware 5.0 or higher.
PKM keys do not function as required.	The PKM is not communicating.	Verify that the PKM is programmed correctly in CDE. Form 9 should have asterisks preceding the host phone or DSS/BLF Interface Unit that are programmed with the PKM.
PKM keys flash erratically and do not function as required.	The wrong type of power adapter is connected.	Connect the appropriate AC adapter (either 12 or 24 volts) as indicated in the Power Requirements table.
PKM keys do not flash at all during setup.	The connections are incorrect or the equipment may be faulty.	Verify that the host phone or the DSS/BLF Interface Unit is functioning.

Check that the correct installation ports are used for the power adaptor. See for PKM Power Requirements. Check that the second PKM is plugged into the bottom port of the first PKM.

Install a SUPERSET Interface Module

Description

The SUPERSET interface modules install inside SUPERSET 4025, SUPERSET 4125, and SUPERSET 4150 telephones. Two types of SUPERSET interface modules are available: the SUPERSET Interface Module 1 (SIM 1) and the SUPERSET Interface Module 2 (SIM 2).

The SIM 1 allows you to connect your telephone to a PKM 12 or to one or two PKM 48 devices.

The SIM 2 allows you to connect your telephone to a PKM 12, or to one or two PKM 48 devices, or to one or two PKM 48 devices and to a two-wire analog device such as an analog telephone, fax machine, or modem (to a maximum loop length of 50 feet and a ringer load of up to 2 REN). The module allows simultaneous use of both the 4000 series telephone and the analog peripheral.

The SIM 2 allows an analog device to be connected to the port on the set while having its own directory number, Class of Service, and Class of Restriction. Both devices share the same 2-wire physical connection. The SIM 2 is programmed in a form for data devices, but is actually a voice device that connects to other voice devices.

The system will recognize the SUPERSET interface module only if it is both programmed and installed.

Message Waiting is not supported by sets connected to a SUPERSET interface module. Devices with a Z-type REN are not to be connected to a SUPERSET interface module. The analog device must signal using DTMF tones; devices that use dial pulse signaling are not supported.

Installation

WARNING: Any connection of this device to an off premise application, an out of plant application, or to any other exposed plant application may result in a safety hazard, and/or defective operation, and/or equipment damage.

CAUTION: For a SUPERSET interface module to work correctly with a SUPERSET 4000-series telephone, the SUPERSET 4000-series telephone must have a Boot Revision of 4.2 or greater, and the SUPERSET 4000-series telephone must be upgraded to the latest firmware available from the system maintenance firmware upgrade command. The Boot Revision is displayed on the set briefly after the set is connected.

The following chart lists the steps required to install SUPERSET interface modules (SIM 1 or SIM 2).

Chart: SUPERSET Interface Module Installation/Removal Instructions

Step	Action
	Caution: Only Mitel certified personnel are authorized to install SUPERSET interface modules. Handle the modules by the edges of the printed circuit boards or the jacks. Do not touch components or circuitry.
SUPERSET Interface Module Installation	

Chart: SUPERSET Interface Module Installation/Removal Instructions

Step	Action
1.	Unplug the set from the system and upgrade the firmware of the set.
2.	With a Philips screwdriver, remove the cover from the underside of the telephone. Retain the cover and the screw.
3.	Holding the module by the PKM jack (the jack nearest the power adapter jack), set the module squarely into the exposed bay with the jack(s) nearest to the front of the set (the hinged part of the stand). See illustration.
4.	

In the plastic cover shipped with the module, use your thumb to snap out the plastic PKM jack punch-out (the punch-out nearest the small, square power-adapter hole).

For SIM2 only, use your thumb to also snap out the plastic ONS jack punch-out.	
5.	Place the cover shipped with the module over the module and apply equal pressure to all sides of the cover until the module clicks into place.
6.	Fasten the module cover with the screw shipped with the module.
7.	Reconnect the set to the system.
8.	

For SUPERSET 4025 telephones only, plug the unpowered power adapter into the interface module's power jack.

For SUPERSET 4150 and SUPERSET 4125 telephones only, plug the unpowered power adapter into the telephone itself. **Caution: Never plug a power supply into the interface module installed in a SUPERSET 4125 or a SUPERSET 4150 telephone.**	
9.	Plug the power adapter into a power source.
10.	Verify that CDE (CDE Form 09) for the interface module is complete.
SUPERSET Interface Module Removal	
1.	Unplug the telephone from the system and all power adapters and cables from the interface module jacks.
2.	Unplug the power adapter from its power source.

3.	With a Philips screwdriver, remove the module cover from the underside of the telephone. Store the cover and screw.
4.	Grasp the interface module by the PKM jack and gently wiggle the module as you pull it from the set.
5.	Replace the telephone's original cover and fasten it with the screw removed in Step 3.

Install a 5421 or 5422 Interface Module

Descriptions

The Mitel 5421 Interface Module (IM) provides connectivity between a Mitel 5020 IP Phone and as many as two Mitel 5415 Programmable Key Modules (PKM) or a single Mitel 5410 PKM. The 5422 IM provides connectivity between a Mitel 5220 IP Phone, 5224 IP Phone, or a 5324 IP Phone and as many as two Mitel 5448 Programmable Key Modules (PKM) or a single Mitel 5412 PKM.

Installation

The Interface Modules include an installation guide (available on edocs.mitel.com).

The following chart lists the steps required to install a 5412 or 5422 interface modules:

Chart: Interface Module Installation/Removal Instructions	
Step	**Action**
	Caution: **Only Mitel certified personnel are authorized to install 5421 and 5422 interface modules. Handle the modules by the edges of the printed circuit boards or the jacks. Do not touch components or circuitry.**
5421 and 5422 Interface Module Installation	
1.	Unplug the phone from the system.
2.	

With a Philips screwdriver, remove the cover from the underside of the telephone. Retain the cover and the screw.

NOTE: With the 5422 PKM IM, punch out the PC port tab before installing the IM in a 5220 IP Phone.	
3.	Place the 5422 PKM IM and apply equal pressure to all sides of the cover until the 5422 PKM IM clicks into place. See illustration.

4.	Fasten the 5422 PKM IM cover with the supplied screw.
5.	Place the cover shipped with the module over the module and apply equal pressure to all sides of the cover until the module clicks into place.
6.	Fasten the module cover with the screw shipped with the module.
7.	

Reconnect the phone to the system and plug the power adapter into the phone.

NOTE: LAN-powered 5220 IP Phones still require a DC adapter when using a PKM.	
8.	Verify that CDE Form 09 for the interface module is complete.
5421 and 5422 Interface Module Removal	
1.	Unplug the telephone from the system and all power adapters and cables from the interface module jacks.
2.	Unplug the power adapter from its power source.
3.	With a Phillips screwdriver, remove the screw from the 5422 PKM IM from the under side of the phone. Store the screw. Grasp the 5422 PKM IM and gently pull the 5422 PKM IM from the phone.
4.	Replace the phone's original cover and fasten it with the screw removed in Step 2 of installing a 5422 PKM IM.
5.	Reconnect the system and interface cables to the phone.

Install a Mitel Conference Unit

Description

The SX-200 ICP supports the 5303 and the 5310 conference units.

The 5303 is an analog unit that connects to an ONS port on the controller to provide full-duplex audio with echo cancellation and noise reduction. The 5310 conference unit is installed on a 5020 IP Phone, 5220 IP Phone, 5220 IP Phone (Dual Mode), 5224 IP Phone (Dual Mode), 5324 IP Phone (Dual Mode), 5330 IP Phone, and 5340 IP Phone.

Installation

The conference units include an installation and user guide (also available on edocs.mitel.com).

Note: The 5310 Conference Unit requires a separate side unit except for use on the 5324 IP Phone, which requires an interface module.

Programming

- The 5303 requires an ONS station extension number programmed in Form 9 (Desktop Device Assignments).
- The 5310 does does not require any programming on the SX-200 ICP controller.

Conditions

- The 5310 may not work properly with a Dual Mode 5220 phone that is below Rev D.8. Please use the normal repair process to get sets upgraded to Rev D.9 or higher.

Install a Mitel Line Interface Module

Description

The Line Interface Module (previously the Mitel 5425 Access Module) provides users of Dual Mode 5220, Dual Mode 5224, Dual Mode 5324, 5330, and 5340 IP Phones (Dual Mode) with the ability to make and receive calls on an analog line in either of the following modes:

- Line Interface Module Mode: user can access the analog line at any time by pressing the programmed Line Interface Module key.
- Fail-over Mode: user can access the analog line when the IP connection has failed.

Users can place Emergency Calls to the Local Emergency Number using the analog line in either mode.

Note: Handsfree and on-hook dialing are not supported on the Line Interface Module line.

Install a DNIC Music-on-Hold/Pager Unit (DMP)

Description

The DNIC Music-On-Hold/Pager unit (DMP) interfaces an SX-200 ICP DNIC port to the following external equipment:

- External music source for Music-on-Hold
- External paging amplifier (with or without answerback capability)
- Up to two night bells
- An external alarm

The unit is powered by the SX-200 ICP and does not require a separate power source. A single 25 pair Amphenol connects to the SX-200 ICP via the main distribution frame. A single LED indicator provides basic status information. The unit can be wall-mounted next to the SX-200 ICP.

Each Music-on-Hold/Pager Unit supports a single paging zone. If more than one paging zone is required, additional Music-on-Hold/Pager Units can be added as required.

Music-On-Hold/Pager Unit

LED Indicator

The LED indicates the following states:

Music-On-Hold/Pager Unit LED Indicator	
Indication	**Status**
OFF	No power from SX-200 ICP DNIC
Flashing (2 Hz - On 250 ms, Off 250 ms.)	Loss of synchronization with SX-200 ICP System, the MOH/Pager Unit may be faulty
On Solid	MOH/Pager Unit is operating
Winking (2 Hz - ON 50 ms, Off 1 sec.)	MOH/Pager Unit is operating and the paging amplifier is being accessed

Interface Technical Specifications

MOH Input

The MOH input is a transformer coupled input with impedance of 600 ohms. The gain is fixed at -4 dB A/D. To meet signal power limits applied to such interfaces by FCC part 68 and Industry Canada CS03, the audio level at the digital point will be limited to -15 dBm0 and the frequency response is rolled off (3 dB point) at approximately 3 Khz. The amplitude limiting begins to affect the input signal when it exceeds approximately 100 mVrms. Input signals should be in the range of 10 to 100 mVrms.

Paging Input/Output

The paging interface is a transformer coupled loop start trunk with an impedance of 600 ohms. The DC termination is activated by a relay when the paging amplifier is being accessed. The gain is fixed at -4 dB A/D and +3 dB D/A.

When used with an “answer back” paging amplifier, this interface must meet signal power limits applied to such interfaces by FCC part 68 and Industry Canada CS03, the audio level at the digital point will be limited to -15 dBm0 and the frequency response is rolled off (3 dB point) at approximately 3 Khz. The amplitude limiting begins to affect the input signal when it exceeds approximately 100 mVrms, therefore, answerback signals should be in the range of 10 to 100 mVrms.

Relays

Relays are provided for paging, night bells, alarms, and door lock solenoids. The contacts are rated at 0.1 A @90 Vac or 0.5 A @48 Vdc.

Paging Control Relay. Both normally open and normally closed contacts are provided to control the external paging amplifier. These relay contacts are switched whenever the paging amplifier is being accessed.

Night Bell Relays. Two independent, normally open relays are provided to control external night bells.

Alarm Relay. One normally open relay is provided to control external devices such as an alarm.

Wiring and Cross Connections

The figure below shows the wiring required for the Music-on-Hold/Pager unit.

Music-on-Hold/Pager Unit Connections

Installation

See Connecting a Music-on-Hold/Paging module or unit.

Install a DSS/BLF Interface Unit

Description

The Direct Station Select/Busy Lamp Field (DSS/BLF) Interface Unit allows you to associate up to two PKM 48 devices with an attendant console. The DSS/BLF Interface unit needs to be programmed in CDE to associate with the PKM 48. Each PKM 48 provides the attendant console user with 48 DSS/BLF keys.

The DSS/BLF Interface Unit uses a separate line connection to a DNIC port. You attach the PKM 48 devices to the DSS/BLF Interface Unit and associate the PKM 48 devices with the attendant console through Customer Data Entry (CDE). You can attach up to two PKM 48 devices to the DSS/BLF Interface Unit, for a maximum of 96 DSS/BLF keys.

Attendant consoles excluding SUPERCONSOLE 1000 attendant consoles with PN 9189-000-300 and PN 9189-000-301 require the DSS/BLF interface to associate the PKM 48 devices.

Installation

1. Ensure that the DSS/BLF Interface Unit and PKM 48 devices are programmed (see Programming).
2. Position the DSS/BLF Interface Unit and PKM 48 devices near the attendant console. If possible, place the DSS/BLF Interface Unit out of sight (for example, under the desk).
3. Plug one end of the DNIC line cable into the LINE jack located on side panel of the DSS/BLF Interface Unit. Plug the other end into the telephone wall-jack. See the DSS/BLF Console Interface Box Figure.

Note: The 2.13 m (7 ft.) DNIC line cable and the 5 m (16.4 ft.) DSS/BLF-to-PKM cable are shipped with the DSS/BLF Interface Unit.

4. Plug the power adapter jack into the power input on the side panel of DSS/BLF Unit. Plug the power adapter into the AC power source.
5. Plug one end of the 5 m (16.4 ft.) DSS/BLF-to-PKM cable into the PKM jack on the DSS/BLF Interface Unit. Plug the other end of the cable into jack located on the underside of the first PKM.

Note: The PKM LEDs will temporarily flash and then go out. If the LEDs continue to flash, check that the cable connections and programming are correct.

6. If a second PKM is required, plug the 0.6 m (2 ft.) PKM modular cable in the jack on the underside of the first PKM. Then, plug the other end of the cable into the jack on the underside of the second PKM.

Note: The PKM LEDs will temporarily flash and then go out. If the LEDs continue to flash, the installation or programming is incorrect.

7. Verify the operation of the DSS/BLF keys by placing a call from each key.

Note: The DSS/BLF keys will not function until they are programmed in CDE.

Install a Door Phone and Door Opener

Description

Customer-provided door phone units (up to three) can be connected to ONS ports to provide two-way communication with designated extensions. Door entry is controlled by the general-use relays in the SX-200 ICP which are actuated at the designated extension by pressing a feature key or dialing a feature access code.

The relay contacts are rated at 90mA @60 Vac or Vdc peak and are normally open.

Installation

Chart: Install a Door Phone

Step	Action	Comments
1.	Install the Door Phone according to the manufacturer's instructions.	
2.	Connect the Door Phone to an ONS circuit. For ONS Tip & Ring assignments see: **MX Controller-** Analog Main Board / Analog Option Board Tip & Ring Assignments **CX/CXi Controller** - Analog Main Board Pinouts and Analog Option Board Pinouts	Door phones should not be assigned to plids associated with power or system fail transfer circuits as the phone would become the answering point for any incoming calls in the event of a failure.
3.	Connect the general-use relays in the controller to the door lock solenoid. **MX Controller -** Connect the door lock solenoid to a Relay port. For pinouts, see Generic Relay Pinouts (MX Controller). **CX/CXi Controller -** Connect the door lock solenoid to a Relay port. Each port has two relays. For pinouts, see Analog Main Board Pinouts and Analog Option Board Pinouts. When the Open Door feature key or MESSAGE key is pressed, or when the Open Door feature access code is dialed at the door phone answering position, these two pins are shorted which unlocks the door. **Relay Contact Ratings:** 90mA @60 Vac or Vdc peak and are normally open.	
4.	Program the Door Phone feature in CDE.	For programming details, see Door Phone/Open Door.

Install a System Printer

System Printer Port (MX)

The following chart describes how to install a system printer.

Chart: Install a System Printer

Step	Action	Comments

Chart: Install a System Printer

Step	Action	Comments
	Connect System Printer	
1.	Connect one end of a standard 25-pin RS-232 cable to the RS-232 port of the printer (or similar output device).	
2.	Connect the other end of the 25-pin RS-232 cable to a 25-to-9 pin RS-232 adapter arrangement (connector adapter or cable adapter). Then connect the adapter arrangement to the Printer port on the front panel of the SX-200 ICP.	See Printer port pinouts in the previous table.
3.	Set up the printer data characteristics in Maintenance.	8 data bits 1 start bit 1 stop bit space parity 300-1200 baud (system default baud rate is 300 baud)
4.	In CDE Form 34, Directed IO, define the Printout Types that are to be delivered to the system printer port.	
	Power Up	
5.	Plug in the printer and turn it on.	

Dataset Printer Port (MX)

The following steps describe how to program a dataset printer port.

1. Form 11, Data Circuit Descriptor
   - Define a circuit descriptor to match characteristics of device type. Refer to Form 04, System Options and Timers on page 133 for typical circuit descriptor.
2. Form 12, Dataset Assignment
   - Assign the type of data device for the dataset PLID. Available types are DSCONS Console (output only, maximum 2400 baud) and 1103/2103 Standalone dataset.
   - Assign a Tenant, Extension number, COS, COR, and circuit descriptor number.
3. Form 34, Directed I/O
   - Define printout type to be delivered to the dataset.

IP Printer Port

Data for the functions listed below can be output through an IP socket in the controller to a telnet-enabled application for printing.

- SMDR
- CDE Data Print
- ACD Real Time Events
- ACD Agent Print

- Maintenance Logs
- Traffic Measurement
- IP Traffic Measurement
- ACD Group Summary
- Hotel/Motel Audit
- Hotel Motel Wakeup

1. Form 11 - Data Circuit Descriptor
   - Define a circuit descriptor to match characteristics of device type. Refer to Subform 11, Data Circuit Descriptor Options for typical options.
2. Form 12 - Data Assignment
   - Program a SOCKET type data device to an available PLID in the following range:

PLID	Port	PLID	Port
1/13/19	61319	1/13/24	61324
1/13/20	61320	1/13/25	61325
1/13/21	61321	1/13/26	61326
1/13/22	61322	1/13/27	61327
1/13/23	61323	1/13/28	61328

   Note: Port 61328 (PLID 1/13/28) is the default printer port on the CX/CXi controller. (You must Telnet to the socket number, not directly to the extension.)
   - Assign a Tenant, Extension number, COS, COR, and circuit descriptor number.
3. Form 34, Directed I/O
   - Define printout type to output.

Install Night Bell

Connecting a Night Bell to the Controller Relay Port describes how to connect a night bell to the SX-200 ICP controller.

Connecting a Night Bell to the Controller ONS Port describes how to connect a night bell to the SX-200 ICP controller.

The following installation procedures apply to systems with peripheral cabinets.

- Install Night Bell (Direct Connect) describes how to install a Direct Connect night bell
- Install Night Bell (Auxiliary Relay) describes how to install an Auxiliary Relay night bell. Both apply to systems with peripheral cabinets

Notes:

1. Incoming and internal calls can be directed to a common alerting device (bell). The bell is activated by relay contacts terminating at the DB-9 connector on the front of the SX-200 ICP controller or by the DTMF receiver/relay module on the universal card installed in a peripheral cabinet. The calls can be answered from the attendant console, or from any station with Trunk Answer From Any Station (TAFAS) feature access assigned to it.
2. The night bell, the auxiliary relay (normally open), and independent ringing supply are customer-supplied.

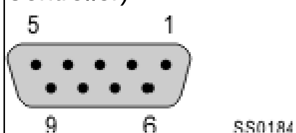
   The night bell must be a type of unit which does not require ringing voltage (a “Buzzer” style unit) from the system. A power source other than the SX-200 ICP is required.
3. Direct Connect Method: Night bells can be connected directly if the total current requirement does not exceed the relay contact ratings.
4. Auxiliary Relay Method: Night bells must be connected through an auxiliary relay if the total current requirement exceeds the relay contact ratings.
5. All wiring must be done in accordance with local electrical codes.

Connecting a Night Bell to the Controller Relay Port

Chart: Connect Night Bell to Controller Relay Port		
Step	**Action**	**Comments**
	Install Night Bell	
1.	Install the bell according to manufacturer's instructions and environment specification.	
	Connect Night Bell	
1.	Connect one side of the night bell through the cross-connect field to the power source.	
2.		

Connect the relay contacts on the controller to the night bell.

MX Controller - If using the default database, connect pin 7 of the relay contact from the DB-9 connector to ground and pin 8 to the other side of the night bell. For the relay pinouts and bay/slot/ circuit assignments, see Relay Connector Pinouts (MX Controller).



CX/CXi Controller - Connect the night bell to a Relay port on the back of the controller. The default database is programmed to use the relay on the Analog Main Board. For relay pinouts and bay/slot/ circuit assignments, see Analog Main Board and Analog Option Board Pinouts (CX/CXi Controller).

Relay contact ratings:
90mA @60 Vac or Vdc peak.

Connection of the bell or alarm device must be through an auxiliary relay if the total current requirement exceeds the relay contact ratings. See this Figure for relay connections.

	Program Night Bell in CDE	
1.	The default database contains the required CDE programming for Night Bell operation. If using a different database, see Night Bells.	

Connecting a Night Bell to the ONS Port

Chart: Connect Night Bell to Controller ONS Port

Step	Action	Comments
	Install Night Bell	
1.	Install the bell according to manufacturer's instructions and environment specification.	
	Connect Night Bell	
1.	Connect both tip and ring to the Night Bell.	
	Program Night Bell in CDE	
1.	Locate a spare ONS bay/slot/circuit in Form 9 that does not have a phone programmed. Note the number.	
2.	In Form 18, select one of the Night Bells and fill in the bay/slot/circuit number from Form 9.	
3.	Tab over to the Ext # field and enter an extension number. Cross-connect the Night Bell to the bay/slot/circuit in Step 2.	
4.	Program the trunks to ring the extension number of the Night Bell assigned in Form 18. (Step 2)	When this number is called, the Night Bell will ring and anyone with the TAFAS COS option enabled will be able to answer.

Installing the Night Bell with Direct Connect (Peripheral cabinets only)

Chart: Install Night Bell (Direct Connect)

Step	Action	Comments
	Install Bell	
1.	Install quipment according to manufacturer's instructions and environment specification.	
2.	Install DTMF Receiver/Relay	
3.	Connect Night Bell	
4.	Connect one side of the night bell through the cross-connect field to the power source.	See this Figure for relay connection.

Connect one side of the relay contact from the DTMF receiver/relay module to ground and the other side of the relay contact to the other side of the night bell.

Note: Relay contacts may be connected only to a secondary circuit that has no direct connection to a primary circuit, and receives its power from a transformer, converter, or equivalent isolation device situated within the equipment.	**Relay contact ratings:** 90 Vrms at 0.1 A 48 Vdc at 0.5 A **Program Night Bell in CDE** See Night Bells.	

Installing the Night Bell with Auxiliary Relay (Peripheral cabinets only)

Chart: Install Night Bell (Auxiliary Relay)

Step	Action	Comments
	Install DTMF Receiver/Relay module	
1.	Ensure that the DTMF receiver/relay module is installed on the Universal Card.	
	Install DNIC Music-on-Hold Paging Module	
2.	Ensure that the DNIC Music-on-Hold Paging module is installed and connected to a DNIC circuit.	
	Install Bell	
3.	Follow manufacturer's instructions to install the bell.	Install equipment in the environment specified by the manufacturer.
	Connect Auxiliary Relay	
4.	Connect one side of the auxiliary relay coil through the cross-connect field to the customer supplied power source. See the Music on Hold Unit Night Bell Configuration Figure.	
	Connect one side of the relay contact from the DTMF receiver/relay module to ground and the other side of the relay contact to the other side of the auxiliary relay coil. **Relay contact ratings:** 90 Vrms at 0.1 A 48 Vdc at 0.5 A	**Note:** Relay contacts may be connected only to a secondary circuit which has no direct connection to a primary circuit, and which receives its power from a transformer, converter, or equivalent isolation device situated within the equipment.
	Connect Bell to Auxiliary Relay	
5.	Connect one side of the bell to one side of the auxiliary relay contact.	
6.	Connect the other side of the bell to one side of the independent ringing source.	
7.	Connect the other side of the auxiliary relay	

Chart: Install Night Bell (Auxiliary Relay)		
Step	**Action**	**Comments**
	contact to the other side of the independent ringing source. **Program Night Bell in CDE** See Night Bells.	

Install Paging Equipment

Paging equipment may be connected to an SX-200 ICP via the Paging port on the back of the unit, a Mitel 5485 IP Paging Unit, a Music-on-Hold/Paging module on the Universal Card installed in a peripheral cabinet or via a DNIC Music-on-Hold/Pager Unit (DMP).

Notes:

1. The paging equipment is customer-supplied.
2. The system supports up to nine separate paging zones. Each zone requires a paging circuit in the controller, a 5485 IP Paging Unit, a paging module or a DMP.

Chart: Install Paging Equipment		
Step	**Action**	**Comments**
1.	**Connect Paging Equipment to the SX-200 ICP Controller**	

MX Controller: Connect the paging equipment through the cross-connect field to RJ-45 connector on the Controller rear panel. See Paging Relay Pinouts.

CX/CXi Controller - Analog Main Board and Analog Option Board Tip & Ring Assignments

See this Figure paging equipment connections.

Install equipment in the environment specified by the manufacturers.

2.	Program the Paging feature.	
1.	**Connect Paging Equipment to the 5485 IP Paging Unit** Use a standard Ethernet cable to connect the 5485 IP Paging Unit to the Layer 2 switch.	

2.	Use a 25-pair twisted-pair cable to connect the paging equipment to the 5485 IP Paging Unit. See the Installation Guide supplied with unit for detailed instructions.	

The 5485 IP Paging Unit uses only 3 pairs from the 25-pair cable. The wire color code and signal for each connector pin are as follows:

PIN	Color	Signal
9	BR/R	Audio output, Positive
34	R/BR	Audio output. Negative
36	BK/BL	Relay Closure (Normally closed)
37	BK/O	Relay Closure (Normally open)
11	BL/BK	Page Control input
12	O/BK	

See this Figure for paging equipment connections.

Install equipment in the environment specified by the manufacturers.

3.	Program the Paging feature.	
	Connecting Paging Equipment using the Music-on-Hold/Paging module or DMP	
1.		

Install the Music-on-Hold/Paging module or DMP

Universal Card: Ensure that the Music-on-Hold/Paging module is installed.

DMP: Ensure that the DNIC Music-on-Hold/Pager unit is installed.

2.	Install the Paging Equipment	
	Follow manufacturer's instructions to install the paging equipment. See this Figure for the module or this Figure for the DMP.	Install equipment in the environment specified by the manufacturers.
3.	Connect Paging Equipment	

Universal Card: Connect the output from POA and POB on the Music-on-Hold/Paging module through the cross-connect field to the paging equipment.

DMP: Connect the output from PAGE+ and PAGE- on the Music-on-Hold/Paging unit through the cross-connect field to the paging equipment.	**Output impedance (low):** 200 ohm **Output level into 600 ohm:** -6 dBm	
4.		

Universal Card: Connect the relay contacts, normally open or normally closed and common (or as specified in the documentation with the paging amplifier), from the Music-on-Hold/Paging unit (PKA and PKB) to the control circuit of the paging equipment.

DMP: Connect the appropriate relay contacts from the Music-on-Hold/Pager unit to the control circuit of the paging equipment.

Note:
Relay contacts may be connected only to a secondary circuit which has no direct connection to a primary circuit, and which receives its power from a transformer, converter, or equivalent isolation device situated within the equipment.

Relay contact ratings: 90 Vrms at 0.1 A 48 Vdc at 0.5 A		
5.	Program the Paging feature.	

Install an Alarm Device

The controller provides relays through the DB-9 connector on the front panel for activating an alarm device (typically a lamp) in response to a CRITICAL alarm condition.

The relay contacts are rated at 90mA @60 Vac or Vdc peak and are normally open. Connection the alarm device must be through an auxiliary relay if the total current requirement exceeds the relay contact ratings.

The alarm device requires an independent power supply

All equipment (alarm device, power supply, auxiliary relay.) are customer-supplied.

Chart: Connect Alarm Device

Step	Action	Comments
	Install Alarm Device	
1.	Install alarm device and power source according to manufacturer's instructions and environment specification.	
	Connect the Alarm Device to the Controller	
1.		

MX Controller: Connect the alarm device to a generic relay through the RJ45 connector on the back of the controller. For pinouts, see Generic Relay Pinouts (MX Controller).

CX/CXi Controller: Connect the alarm device to a generic relay on the back of the controller. For pinouts, see Analog Main Board Pinouts and Analog Option Board Pinouts.	**Relay contact ratings:** 90mA @ 60 Vac or Vdc peak

Program the Alarm Device in CDE

1. In Form 18 (Miscellaneous System Ports), assign Major Alarm to one of the following PLIDs

MX Controller 1/13/29/1 1/13/29/2 1/13/30/1 **CX Controller** 1/13/30/3 1/13/30/4 1/13/31/3 1/13/31/4	**Note:** The default database for the MX has PLID 1/13/29/1 programmed for Night Bell use. For the CX, the default is PLID 1/13/30/4.

Install Music-on-Hold Equipment

An external music source can be connected to the system via the Music-On-Hold connector on the back panel of the SX-200 ICP controller, or by using a Music-on-Hold/Paging module on the Universal Card, or by using the Music-on-Hold/Pager unit (DMP). Music-on-Hold is used with the camp-on and hold features. Callers on hold hear music instead of no sound; camp-on callers hear music instead of busy tone.

Connecting a music source to the Music-On-Hold jack on the controller

Any music source (radio, CD player, etc.) with a mini (1/8" - 3.5 mm) phono plug may be connected to the Music-On-Hold connector. Use the device's volume control to adjust volume levels as required.

Input signals should be in the range of 10 to 100 mVrms. Any DC voltage applied to the input shall be less than 50 VDC.

The default database contains the necessary CDE programming for the Music-on-Hold port. If a different database is used, see Music-on-Hold (MOH) for programming requirements.

Note: Powering down the SX-200 ICP redirects the MOH source to the paging output. To stop the music from being heard over the pager, power down both the MOH source and paging amplifier before powering down the SX-200 ICP.

Connecting a Music-on-Hold/Paging module or unit (applies to system with peripheral cabinets)

The following chart describes how to install a Music-on-Hold/Paging module/unit and a music source in systems with peripheral cabinets.

For CDE programming, see Music-on-Hold (MOH).

Notes:

1. This equipment also provides the background music played through the speakers of the Mitel telephones while the set is idle (background music feature).
2. The music source is customer-supplied, and can be a tape recorder, radio, or other music source.

Chart: Install a Music-on-Hold/Paging module/unit and a music source

Step	Action	Comments
1.	**Install the Music-on-Hold/Paging module** **Universal Card:** Ensure that the Music-on-Hold/Paging module is installed. **DMP:** Ensure that the Music-on-Hold/Pager unit is installed. See this Figure for the module wiring connections or this Figure for the DMP wiring connections.	Applies to systems with peripheral cabinets.
2.	**Install Music Source** Follow manufacturer's instructions to install the music source.	Install equipment in the environment specified by the manufacturers.
	Connect Music Source **Universal Card:** Connect the music source through the cross-connect field to MIA and MIB on the Music-on-Hold/Paging module. **DMP:** Connect the music source through the cross-connect field to MOH+ and MOH- on the Music-on-Hold/Paging unit. See the Music on Hold Unit Wiring Schematic Figure.	**Input impedance:** 600 ohm **Input level:** -6 dBm
	Program the Music on Hold feature See Music-on-Hold (MOH).	

Install a System Fail Transfer Unit

Description

The SFT maintains telephone service in the event of system failure (such as a power outage). When the system goes into SFT mode, the SFT unit connects up to six internal POTS telephone extensions directly to the CO, bypassing the SX-200 ICP completely.

The SFT is an optional, standalone, wall-mounted device that connects to the system's peripheral cabinet or main distribution frame (MDF). Each SFT can control six circuits, and up to four SFTs can be daisy-chained together for each zone, providing security for 24 internal extensions.

The SFT switches to SFT mode under the following conditions:

- Failure of the system power converter
- Failure of the system main control (in a redundant system, both main control planes must fail, causing a critical alarm to all zones)
- Interruption of the system AC power
- Failure of the peripheral switch controller (zone)
- Loss of the fiber link between the main control and peripheral cabinets.

Power Supply

All power for the SFT unit is provided from the -48 Vbat source on the system. A source of -12 V powers the electronic circuitry on the card. This supply is derived from the -48 V input and powers all the SFT circuitry except the transfer relays. The relays are powered by a transistor-regulated -41V source, also derived from the -48 Vbat input. Thus, in the event of Vbat varying between the standard -42.5 V to -56.5 V, the current drain remains constant.

Transfer Relays

Each circuit in the SFT uses a four form C relay to transfer between normal and SFT modes of operation.

Loop Detector

When a transfer relay enters SFT mode, the loop detector connects in series with the loop between the extension and CO trunk facility. This circuit prevents the extension from returning to normal operating mode before an SFT mode call is completed. When the SFT mode call is completed, the extension is returned to normal operating mode.

SFT Control Leads

The transfer control sensor on the SFT senses a loop closure across the SFT and SFT return (SFTR) leads. When a loop closure is sensed, the power to the relays is removed, the relays are released, and all circuits enter the transferred state.

Power Consumption

The total current drain for the SFT is typically 80 mA.

Power Dissipation (watts)

Power Supply	TYP. (Watts)	TYP+20%
-48Vbat	3.18	3.81
@Vbat=-56 V	3.71	4.45

Installation

Before You Begin

You need gather the following tools and parts to install the SFT:

Tool	Use

Tool	Use
Wire cutters	To cut frame jumpers
Punch-down tool	To terminate frame jumpers on the MDF
25-pair cable	To connect the SFT to the MDF
Quad cable	To connect the SFT to the MDF
Two 1 inch (2.54 cm) self tapping screws	To attach the mounting bracket to the wall
6-pin RJ-11 cable	To connect the SFT unit J1 connector to the system (MDF or peripheral cabinet)

Mount the SFT

To mount the SFT

1.

Choose a place to mount the SFT. The location should be:

- in the vicinity of the peripheral cabinet (SX-200)
- on a flat, solid surface (such as a wall)
- in an area that is subject to the same temperature and humidity ranges as the PBX.

Note: You can mount the unit vertically or horizontally.

2. Remove the two rubber feet from the mounting holes at the top of the SFT.

3. Attach the mounting brackets to the SFT by inserting the plastic push rivets.

4. Using two 1-inch (2.54-cm) self-tapping screws, attach the SFT to a flat surface.

Connect the Cables

There are three female connectors located on the rear panel of the SFT, labeled J1, J2 and J3:

- J1 connects the SFT to the system with an RJ-11 cable
- J2 connects the SFT to another SFT with an RJ-11 cable
- J3 connects the SFT to the MDF with an Amphenol cable.

As you proceed with installation, keep the following points in mind:

- Ensure SFT extensions and the associated trunk facilities are assigned to the same zone
- Ensure SFT lines and trunks are connected to the SFT unit so that when a transfer occurs, the stations and trunks are:
  - compatible (for example, DTMF telephones)
  - connected to trunks capable of accepting DTMF

- equipped with a ground button, or loop/ground converter for SFT stations assigned to ground start trunks.

Connecting J1

J1 is a 6-pin 4T RJ-11 connector that transmits the -48V power and the transfer control pair (SFT & SFTR) from the PBX to the SFT. To complete this connection, use the RJ-11 cable supplied with the unit.

The J1 connector is assigned as follows:

SFT		**PBX SX-200**
J1 (Input)		Peripheral Node
Pin	Signal Name	Make a direct connection by using an RJ11 cable.
1	SFT	
2	SFTR	
3	-48 VRET	
4	-48 Vbat	

Connecting J2

J2 is a 6-pin 4T RJ-11 connector that transmits the -48V power and the transfer control pair (SFT & SFTR) from the SFT to another SFT. Up to four SFTs can be daisy-chained in this manner.

The J2 connector is assigned as follows:

SFT		**SFT**
J2 (Output)		J1 (Input)
Pin	Signal Name	Daisy-chain to a maximum of four other SFT units.
1	-48 Vbat	
2	-48 VRET	
3	SFTR	
4	SFT	

Connecting J3

J3 is a 25-pin Amphenol connector that provides TIP and RING connections to the six circuits on the SFT. All TIP and RING pairs assigned to the SFTs are connected through the MDF.

The J3 connector is assigned as follows:

Pin	**Signal Name**	**Pin**	**Signal Name**

1/26 2/27	T/R EXT 1 T/R LC 1	3/28 4/29	T/R TC 1 T/R CO FAC 1
5/30 6/31	T/R EXT 2 T/R LC 2	7/32 8/33	T/R TC 2 T/R CO FAC 2
9/34 10/35	T/R EXT 3 T/R LC 3	11/36 12/37	T/R TC 3 T/R CO FAC 3
13/38	NC		
14/39 15/40	T/R EXT 4 T/R LC 4	16/41 17/42	T/R TC 4 T/R CO FAC 4
18/43 19/44	T/R EXT 5 T/R LC 5	20/45 21/46	T/R TC 5 T/R CO FAC 5
22/47 23/48	T/R EXT 6 T/R LC 6	24/49 25/50	T/R TC 6 T/R CO FAC 6
Where T/R = Tip/Ring, Ext = Extension, LC = Line Circuit, TC = Trunk Circuit, CO FAC = Central Office Facility, NC = No Connection			

Cabling and Cross Connections

This section gives the Tip-Ring Assignment tables for cabling and cross connecting the SX-200 ICP. It also provides the DB-9 connector pinouts for the Relay Port on the front panel, the RJ45 pinouts for the Paging contacts on the back panel, and the NSU pinouts for T1 Line/Network Termination.

Amphenol Connector Gender

Table: 25 Pair Amphenol Connector Gender

Component	Connector
PER Node	Female
MX Controller	Female
Music-on-Hold/Pager	Female
ASU	Female
ASU II cards	Male

MX Controller

Table: Cable Terminations (MX Controller)	
Table	Analog Main Board / Analog Option Board Tip/Ring Assignments
Table	Dual T1/E1 Framer Module Tip/Ring Pinouts
Table	CIM Port Pinouts (Onboard and Quad CIM MMC)
Table	Maintenance Terminal and Printer Port Pinouts
Table	Generic Relay Pinouts - Front Panel DB-9 Connector
Table	Paging Relay Pinouts - Back Panel RJ-45 Connector
Table	RS-232 Port Pinouts
Table	10/100Base-TX Ethernet Port Pinouts
Table	10/100/1000Base-T Ethernet Port Pinout
NSU and ASU/ASU II	
Table	NSU Pinouts for T1 Line/Network Termination
Table	ASU Tip/Ring Assignments
Table	ASU II Tip/Ring Assignments
Peripheral Cabinets	
Table	Universal Card Tip/Ring Assignments J1, J2, J3, and J4
Table	SX-200 Peripheral Cabinet Tip/Ring Assignments (High-power Slots) J5 and J9
Table	SX-200 Peripheral Cabinet Tip/Ring Assignments (Low-power Slots) J7 and J11
Table	T1 Trunk Port (J5 and J6)
Table	SFT Port (J7)
Table	Maintenance Module Port
Table	USOC Connector Pinouts
Table	Music-on-Hold/Pager Unit Pinouts

Analog Main Board / Analog Option Board Tip/Ring Assignments (MX Controller)

Table: Analog Main Board / Analog Option Board Tip/Ring Assignments (MX Controller)

Note: The Analog Main Board / Analog Option Board comprise the IP bay for the controller. By default, the IP bay is bay number 1.

Pins	Pairs	Circuit type	Bay/Slot/Circuit	Comments
1/26	W-BL / BL-W	ONS/CLASS	1/13/3	Default Extension numbers: 200

2/27	W-O / O-W	ONS/CLASS	1/13/4	Default Extension number: 201
3/28	W-G / G-W	ONS/CLASS	1/13/5	Available as upgrade option
4/29	W-BR / BR-W	ONS/CLASS	1/13/6	Available as upgrade option
5/30	NOT USED			
6/31	R-BL / BL-R	DNIC	1/13/1	Sub Attendant Default Extension number: 199
7/32	R-O / O-R	DNIC	1/13/2	SUPERCONSOLE 1000 Default Extension number: 198
8/33 - 10/35	NOT USED			
11/36 - 16/41	BK-BL / BL-BK	LS/CLASS	1/13/7 - 1/13/12	Trunks at locations 1/13/7 and 1/13/8 are the System Fail Transfer trunks. They connect to ONS phones located at 1/13/3 and 1/13/4 respectively.
17/42 - 22/47	Y-O / O-Y	LS/CLASS	1/13/13 - 1/13/18	Available as upgrade option
48/23 - 50/25	NOT USED			

Dual T1/E1 Framer Module Tip/Ring Assignments (MX Controller)

Table: Dual T1/E1 Frame Module Tip/Ring Assignments (MX Controller)

Connector Pin	Signal	NT/LT Setting (software selectable)	
NT (Default)	**LT**		
1	--	Rx Ring	Tx Ring
2	--	Rx Tip	Tx Tip
3	N/C	--	--
4	--	Tx Ring	Rx Ring
5	--	Tx TIp	Rx Tip
6	N/C	--	--
7	N/C	--	--
8	N/C	--	--

CIM Port Pinouts (MX Controller)

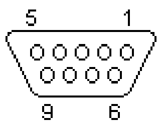
Table: Port Pinouts for CIMs

PIN	Signal
1	RX +
2	RX -
3	TX +
4	Not Used
5	Not Used
6	TX -
7	Not Used
8	Not Used

Note:	The CIM ports are the same whether the card has a control end connection or a peripheral end connection. Connections from the control cabinet to the peripheral cabinet require Category 5 UTP (unshielded twisted pair) cable, with TX and RX pairs reversed--i.e., a crossover cable.

Relay Connector Pinouts (MX Controller)

Table: Generic Relay Pinouts - Front Panel DB-9 Connector (MX Controller)



Pin	Bay/Slot/Circuit	Function
1	1/13/30/1 (Assigned for Night Bell use in the default database)	Generic Relay Contact (normally open)
2	Generic Relay Contact Return	
3		Not Used
4	1/13/29/2	Generic Relay Contact (normally open)
5	Generic Relay Contact Return	
6		Not Used
7	1/13/29/1	Generic Relay Relay Contact (normally open)
8	Generic Relay Relay Contact Return	
9		Not Used
Note: PLID 1/13/30/1 is programmed for Night Bell use in the default database.		

Paging Relay Pinouts - Back Panel RJ-45 Connector (MX Controller)

Table: Paging Relay Pinouts - Back Panel RJ- 45 Connector (MX Controller)

Back Panel RJ-45 Connector

Pin	Signal
1	Paging relay contact A (normally closed)

Table: Paging Relay Pinouts - Back Panel RJ- 45 Connector (MX Controller)

Back Panel RJ-45 Connector

Pin	Signal
2	Paging relay common
3	Paging relay contact B (normally open)
4	Paging wire 1 (pager audio)
5	Paging wire 2 (pager audio)
6	Factory test control (not for customer use)
7	Unused
8	Unused

RS-232 Port Pinouts (MX, AX, and CX Controller)

Table: RS-232 Asyncronous Serial Port Pinouts (MX Controller)

Connector Pin	Signal Name	Abbreviation
1	Data Carrier Detect	DCD
2	Receive Data	RXD
3	Transmit Data	TXD
4	Data Terminal Relay	DTR
5	Ground	GND
6	Data Set Ready	DSR
7	Request to Send	RTS
8	Clear to Send	CTS
9	Ring Indication	RI

10/100Base-TX Ethernet Port Pinouts (MX/CX/CXi/AX Controller)

The 10/100Base-TX connectors conform to EIA/TIA-568 and can either transmit (TX) or receive (RX).

Table: Ethernet Port Pinouts (MX/CX/CXi/AX Controller))

Connector Pin	Signal
1	RX +
2	RX -
3	TX +
4	Not Used
5	Not Used
6	TX -
7	Not Used
8	Not Used

T1 Termination Mode Pinouts (ASU and NSU)

Table: Pinouts for T1 Line/Network Termination (ASU and NSU)

Pin	Line Termination Mode	Network Termination Mode
1	Tx Ring	Rx Ring
2	Tx Tip	Rx Tip
3	Unused	
4	Rx Ring	Tx Ring
5	Rx Tip	Tx Tip
6	Unused	
7	Unused	
8	Unused	

ASU Tip/Ring Assignments

Table: ASU Tip/Ring Assignments

Note: Connection of the Tip and Ring (A and B) leads of the ONS lines are through a 25 pair female D-type connector.

Pin	Pair	Signal
1/26	W-BL / BL-W	ONS Tip 1 / Ring 1
2/27	W-O / O-W	ONS Tip 2/ Ring 2
3/28	W-G / G-W	ONS Tip 3 / Ring 3
4/29	W-BR / BR-W	ONS Tip 4 / Ring 4
5/30	S-W / W-S	ONS Tip 5 / Ring 5
6/31	R-BL / BL-R	ONS Tip 6 / Ring 6
7/32	R-O / O-R	ONS Tip 7 / Ring 7
8/33	G-R / R-G	ONS Tip 8 / Ring 8
9/34	BR-R / R-BR	ONS Tip 9 / Ring 9
10/35	S-R / R-S	ONS Tip 10 / Ring 10
11/36	BK-BL / BL-BK	ONS Tip 11 / Ring 11
12/37	O-BK / BK-O	ONS Tip 12 / Ring 12
13/38	G-BK / BK-G	ONS Tip 13 / Ring 13
14/39	BR-BK / BK-BR	ONS Tip 14 / Ring 14
15/40	S-BK / BK-S	ONS Tip 15 / Ring 15
16/41	BL-Y / Y-BL	ONS Tip 16 / Ring 16
17/42	O-Y / Y-O	ONS Tip 17 / Ring 17
18/43	G-Y / Y-G	ONS Tip 18 / Ring 18
19/44	BR-Y / Y-BR	ONS Tip 19 / Ring 19
20/45	S-Y / Y-S	ONS Tip 20 / Ring 20
21/46	BL-V / V-BL	ONS Tip 21 / Ring 21
22/47	O-V / V-O	ONS Tip 22 / Ring 22
23/48	G-V / V-G	ONS Tip 23 / Ring 23
24/49	BR-V / V-BR	ONS Tip 24 / Ring 24

25/50	V-S	N/C

ASU II 25-Pair Male D-Type Connector Pinout

Table: ASU II Connector Pinout

Pin	Color Code	24 Port ONSp	Bay/Slot/ Circuit	16 Port ONS	Bay/Slot/ Circuit	4 + 12 Port Combo	Bay/ Slot/ Circuit
26/1	W/BL, BL/W	ONS Tip/Ring 1	n / 1 / 1	ONS Tip/Ring 1	n / 1 / 1	ONS Tip/Ring 1	n / 1 / 1
27/2	W/O, O/W	ONS Tip/Ring 2	n / 1 / 2	ONS Tip/Ring 2	n / 1 / 2	ONS Tip/Ring 2	n / 1 / 2
28/3	W/G, G/W	ONS Tip/Ring 3	n / 1 / 3	ONS Tip/Ring 3	n / 1 / 3	ONS Tip/Ring 3	n / 1 / 3
29/4	W/BR, BR/W	ONS Tip/Ring 4	n / 1 / 4	ONS Tip/Ring 4	n / 1 / 4	ONS Tip/Ring 4	n / 1 / 4
30/5	W/S, S/W	ONS Tip/Ring 5	n / 1 / 5	ONS Tip/Ring 5	n / 1 / 5	ONS Tip/Ring 5	n / 1 / 5
31/6	R/BL, BL/R	ONS Tip/Ring 6	n / 1 / 6	ONS Tip/Ring 6	n / 1 / 6	ONS Tip/Ring 6	n / 1 / 6
32/7	R/O, O/R	ONS Tip/Ring 7	n / 1 / 7	ONS Tip/Ring 7	n / 1 / 7	ONS Tip/Ring 7	n / 1 / 7
33/8	R/G, G/R	ONS Tip/Ring 8	n / 1 / 8	ONS Tip/Ring 8	n / 1 / 8	ONS Tip/Ring 8	n / 1 / 8
34/9	R/BR, BR/R	ONS Tip/Ring 9	n / 1 / 9	ONS Tip/Ring 9	n / 1 / 9	ONS Tip/Ring 9	n / 1 / 9
35/10	R/S, S/R	ONS Tip/Ring 10	n / 1 / 10	ONS Tip/Ring 10	n / 1 / 10	ONS Tip/Ring 10	n / 1 / 10
36/11	BK/BL, BL/BK	ONS Tip/Ring 11	n / 1 / 11	ONS Tip/Ring 11	n / 1 / 11	ONS Tip/Ring 11	n / 1 / 11
37/12	BK/O, O/BK	ONS Tip/Ring 12	n / 1 / 12	ONS Tip/Ring 12	n / 1 / 12	ONS Tip/Ring 12	n / 1 / 12
38/13	BK/G, G/BK	ONS Tip/Ring 13	n / 1 / 13	ONS Tip/Ring 13	n / 1 / 13	N/C	
39/14	BK/BR, BR/BK	ONS Tip/Ring	n / 1 / 14	ONS Tip/Ring	n / 1 / 14	N/C	

		14		14			
40/15	BK/S, S/BK	ONS Tip/Ring 15	n / 1 / 15	ONS Tip/Ring 15	n / 1 / 15	N/C	
41/16	Y/BL, BL/Y	ONS Tip/Ring 16	n / 1 / 16	ONS Tip/Ring 16	n / 1 / 16	N/C	
42/17	Y/O, O/Y	ONS Tip/Ring 17	n / 1 / 17	N/C		N/C	
43/18	Y/G, G/Y	ONS Tip/Ring 18	n / 1 / 18	N/C		N/C	
44/19	Y/BR, BR/Y	ONS Tip/Ring 19	n / 1 / 19	N/C		N/C	
45/20	Y/S, S/Y	ONS Tip/Ring 20	n / 1 / 20	N/C		N/C	
46/21	V/BL, BL/V	ONS Tip/Ring 21	n / 1 / 21	N/C		LS Ring/Tip 1	
47/22	V/O, O/V	ONS Tip/Ring 22	n / 1 / 22	N/C		LS Ring/Tip 2	
48/23	V/G, G/V	ONS Tip/Ring 23	n / 1 / 23	N/C		LS Ring/Tip 3	
49/24	V/BR, BR/V	ONS Tip/Ring 24	n / 1 / 24	N/C		LS Ring/Tip 4	
50/25	V/S, S/V	N/C		N/C		N/C	

SX-200 RM Peripheral Interface Card Slot Assignments

The SX-200 RM cabinets do not have high power and low power designations. Slots 1 to 8 can accept any peripheral interface card, except T1 trunk. A maximum of four high power cards is allowed. The system accepts the first four high power cards it detects, beginning at slot 1. T1 Trunk cards can be installed to slots 10 and 11 only, and require corresponding slots 5 or 6 to be left empty. The PRI card (unlike the T1 trunk card) is not classed as a high power card. Because the PRI card is a separate bay, the PRI card is not included in the count for the four high power cards.

Table: SX-200 RM Peripheral Interface Card Tip/Ring Assignments

J1, J2, J3, and J4			Lead Designation								
Slot/ Plug	**Pin**	**Pair**	**Cct**	**ONS or DLC**	**Cct**	**LS/GS**	**Cct**	**LS/ CLASS**	**Cct**	**Cct**	**OPS or DID**
	26	W-BL	1	T1	1	T1	1	T1	1	1	T1
	1	BL-W		R1		R1		R1			R1
	27	W-O	2	T2		MM1	2	T2	2		
2-J1	2	O-W		R2		M1		R2			
or	28	W-G	3	T3	2	T2	3	T3	3	2	T2
4-J2	3	G-W		R3		R2		R3			R2
or	29	W-BR	4	T4		MM2	4	T4	4		
6-J3	4	BR-W		R4		M2		R4			
or	30	W-S	5	T5	3	T3			5	3	T3
8-J4	5	S-W		R5		R3					R3
	31	R-BL	6	T6		MM3			6		
	6	BL-R		R6		M3					
	32	R-O	7	T7	4	T4	5	T5		4	T4
	7	O-R		R7		R4		R5			R4

Table: SX-200 RM Peripheral Interface Card Tip/Ring Assignments

J1, J2, J3, and J4			Lead Designation								
Slot/ Plug	**Pin**	**Pair**	**Cct**	**ONS or DLC**	**Cct**	**LS/GS**	**Cct**	**LS/ CLASS**	**Cct**	**Cct**	**OPS or DID**
	33	R-G	8	T8		MM4	6	T6			
	8	G-R		R8		M4		R6			
	34	R-BR	9	T9	5	T5	7	T7		5	T5
	9	BR-R		R9		R5		R7			R5
	35	R-S	10	T10		MM5	8	T8			
	10	S-R		R10		M5		R8			
	36	BK-BL	11	T11	6	T6				6	T6
	11	BL-BK		R11		R6					R6
	37	BK-O	12	T12		MM6					
	12	O-BK		R12		M6					
	38	BK-G	SPARE								
	13	G-BK	SPARE								
	39	BK-BR	1	T1	1	T1	1	T1	1	1	T1
	14	BR-BK		R1		R1		R1			R1
	40	BK-S	2	T2		MM1	2	T2	2		
1-J1	15	S-BK		R2		M1		R2			
or	41	Y-BL	3	T3	2	T2	3	T3	3	2	T2
3-J2	16	BL-Y		R3		R2		R3			R2
or	42	Y-O	4	T4		MM2	4	T4	4		
5-J3	17	O-Y		R4		M2		R4			
or	43	Y-G	5	T5	3	T3			5	3	T3
7-J4	18	G-Y		R5		R3					R3
	44	Y-BR	6	T6		MM3			6		
	19	BR-Y		R6		M3					
	45	Y-S	7	T7	4	T4	5	T5		4	T4
	20	S-Y		R7		R4		R5			R4
	46	V-BL	8	T8		MM4	6	T6			
	21	BL-V		R8		M4		R6			
	47	V-O	9	T9	5	T5	7	T7		5	T5
	22	O-V		R9		R5		R7			R5
	48	V-G	10	T10		MM5	8	T8			
	23	G-V		R10		M5		R8			
	49	V-BR	11	T11	6	T6				6	T6
	24	BR-V		R11		R6					R6
	50	V-S	12	T12		MM6					
	25	S-V		R12		M6					

Table: SX-200 Universal Card Tip/Ring Assignments

J1, J2, J3, and J4			Lead Designation					
Slot/ Plug	**Pin**	**Pair**	**Cct**	**MOH/ Pager**	**Cct**	**DTMF Relay**	**Cct**	**E&M Trunk**
	26	W-BL	1	MIA1	1		1	T1
	1	BL-W		MIB1				R1

Table: SX-200 Universal Card Tip/Ring Assignments								
J1, J2, J3, and J4			Lead Designation					
Slot/ Plug	Pin	Pair	Cct	MOH/ Pager	Cct	DTMF Relay	Cct	E&M Trunk
	27	W-O		POA1		K5A1		TR1
2-J1	2	O-W		POB1		K5B1		RR1
or	28	W-G		PKA1		K6A1		E1
4-J2	3	G-W		PKB1		K6B1		M1
or	29	W-BR	2	MIA2	2		2	T2
6-J3	4	BR-W		MIB2				R2
or	30	W-S		POA2		K5A2		TR2
8-J4	5	S-W		POB2		K5B2		RR2
	31	R-BL		PKA2		K6A2		E2
	6	BL-R		PKB2		K6B2		M2
	32	R-O	3	MIA3	3		3	T3
	7	O-R		MIB3				R3
	33	R-G		POA3		K5A3		TR3
	8	G-R		POB3		K5B3		RR3
	34	R-BR		PKA3		K6A3		E3
	9	BR-R		PKB3		K6B3		M3
	35	R-S	4	MIA4	4		4	T4
	10	S-R		MIB4				R4
	36	BK-BL		POA4		K5A4		TR4
	11	BL-BK		POB4		K5B4		RR4
	37	BK-O		PKA4		K6A4		E4
	12	O-BK		PKB4		K6B4		M4
	38	BK-G	SPARE					
	13	G-BK	SPARE					
	39	BK-BR	1	MIA1	1		1	T1
	14	BR-BK		MIB1				R1
	40	BK-S		POA1		K5A1		TR1
1-J1	15	S-BK		POB1		K5B1		RR1
or	41	Y-BL		PKA1		K6A1		E1
3-J2	16	BL-Y		PKB1		K6B1		M1
or	42	Y-O	2	MIA2	2		2	T2
5-J3	17	O-Y		MIB2				R2
or	43	Y-G		POA2		K5A2		TR2
7-J4	18	G-Y		POB2		K5B2		RR2
	44	Y-BR		PKA2		K6A2		E2
	19	BR-Y		PKB2		K6B2		M2
	45	Y-S	3	MIA3	3		3	T3
	20	S-Y		MIB3				R3
	46	V-BL		POA3		K5A3		TR3
	21	BL-V		POB3		K5B3		RR3
	47	V-O		PKA3		K6A3		E3
	22	O-V		PKB3		K6B3		M3
	48	V-G	4	MIA4	4		4	T4
	23	G-V		MIB4				R4
	49	V-BR		POA4		K5A4		TR4
	24	BR-V		POB4		K5B4		RR4
	50	V-S		PKA4		K6A4		E4

Table: SX-200 Universal Card Tip/Ring Assignments								
J1, J2, J3, and J4			Lead Designation					
Slot/ Plug	Pin	Pair	Cct	MOH/ Pager	Cct	DTMF Relay	Cct	E&M Trunk
	25	S-V		PKB4		K6B4		M4

SX-200 Peripheral Cabinet Tip/Ring Assignments (High-power Slots)

Table: SX-200 Peripheral Cabinet Tip/Ring Assignments (High-power Slots)											
J5 and J9 Bay 2 slots 5-8			Lead Designation								
Slot/ Plug	Pin	Pair	Cct	ONS or DLC	Cct	LS/GS	Cct	LS/ CLASS	Cct	Cct	OPS or DID
	26	W-BL	1	T1	1	T1	1	T1	1	1	T1
	1	BL-W		R1		R1		R1			R1
	27	W-O	2	T2		MM1	2	T2	2		
	2	O-W		R2		M1		R2			
	28	W-G	3	T3	2	T2	3	T3	3	2	T2
6-J5	3	G-W		R3		R2		R3			R2
or	29	W-BR	4	T4		MM2	4	T4	4		
8-J9	4	BR-W		R4		M2		R4			
	30	W-S	5	T5	3	T3			5	3	T3
	5	S-W		R5		R3					R3
	31	R-BL	6	T6		MM3			6		
	6	BL-R		R6		M3					
	32	R-O	7	T7	4	T4	5	T5		4	T4
	7	O-R		R7		R4		R5			R4
	33	R-G	8	T8		MM4	6	T6			
	8	G-R		R8		M4		R6			
	34	R-BR	9	T9	5	T5	7	T7		5	T5
	9	BR-R		R9		R5		R7			R5
	35	R-S	10	T10		MM5	8	T8			
	10	S-R		R10		M5		R8			
	36	BK-BL	11	T11	6	T6				6	T6
	11	BL-BK		R11		R6					R6
	37	BK-O	12	T12		MM6					
	12	O-BK		R12		M6					
	38	BK-G	SPARE								
	13	G-BK	SPARE								
	39	BK-BR	1	T1	1	T1	1	T1	1	1	T1
	14	BR-BK		R1		R1		R1			R1
	40	BK-S	2	T2		MM1	2	T2	2		
	15	S-BK		R2		M1		R2			
	41	Y-BL	3	T3	2	T2	3	T3	3	2	T2
5-J5	16	BL-Y		R3		R2		R3			R2
or	42	Y-O	4	T4		MM2	4	T4	4		
7-J9	17	O-Y		R4		M2		R4			

Table: SX-200 Peripheral Cabinet Tip/Ring Assignments (High-power Slots)

J5 and J9 Bay 2 slots 5-8			Lead Designation								
Slot/ Plug	**Pin**	**Pair**	**Cct**	**ONS or DLC**	**Cct**	**LS/GS**	**Cct**	**LS/ CLASS**	**Cct**	**Cct**	**OPS or DID**
	43	Y-G	5	T5	3	T3			5	3	T3
	18	G-Y		R5		R3					R3
	44	Y-BR	6	T6		MM3			6		
	19	BR-Y		R6		M3					
	45	Y-S	7	T7	4	T4	5	T5		4	T4
	20	S-Y		R7		R4		R5			R4
	46	V-BL	8	T8		MM4	6	T6			
	21	BL-V		R8		M4		R6			
	47	V-O	9	T9	5	T5	7	T7		5	T5
	22	O-V		R9		R5		R7			R5
	48	V-G	10	T10		MM5	8	T8			
	23	G-V		R10		M5		R8			
	49	V-BR	11	T11	6	T6				6	T6
	24	BR-V		R11		R6					R6
	50	V-S	12	T12		MM6					
	25	S-V		R12		M6					

SX-200 Peripheral Cabinet Tip/Ring Assignments (Universal Card Modules)

Table: SX-200 Peripheral Cabinet Tip/Ring Assignments for Universal Card Modules

J5 and J9 Bay 2 slots 5-8			Lead Designation					
Slot/ Plug	**Pin**	**Pair**	**Cct**	**MOH/ Pager**	**Cct**	**DTMF Relay**	**Cct**	**E&M Trunk**
	26	W-BL	1	MIA1	1		1	T1
	1	BL-W		MIB1				R1
	27	W-O		POA1		K5A1		TR1
	2	O-W		POB1		K5B1		RR1
	28	W-G		PKA1		K6A1		E1
6-J5	3	G-W		PKB1		K6B1		M1
or	29	W-BR	2	MIA2	2		2	T2
8-J9	4	BR-W		MIB2				R2
	30	W-S		POA2		K5A2		TR2
	5	S-W		POB2		K5B2		RR2
	31	R-BL		PKA2		K6A2		E2
	6	BL-R		PKB2		K6B2		M2
	32	R-O	3	MIA3	3		3	T3
	7	O-R		MIB3				R3
	33	R-G		POA3		K5A3		TR3
	8	G-R		POB3		K5B3		RR3
	34	R-BR		PKA3		K6A3		E3

Table: SX-200 Peripheral Cabinet Tip/Ring Assignments for Universal Card Modules

J5 and J9 Bay 2 slots 5-8			Lead Designation					
Slot/ Plug	**Pin**	**Pair**	**Cct**	**MOH/ Pager**	**Cct**	**DTMF Relay**	**Cct**	**E&M Trunk**
	9	BR-R		PKB3		K6B3		M3
	35	R-S	4	MIA4	4		4	T4
	10	S-R		MIB4				R4
	36	BK-BL		POA4		K5A4		TR4
	11	BL-BK		POB4		K5B4		RR4
	37	BK-O		PKA4		K6A4		E4
	12	O-BK		PKB4		K6B4		M4
	38	BK-G	SPARE					
	13	G-BK	SPARE					
	39	BK-BR	1	MIA1	1		1	T1
	14	BR-BK		MIB1				R1
	40	BK-S		POA1		K5A1		TR1
	15	S-BK		POB1		K5B1		RR1
	41	Y-BL		PKA1		K6A1		E1
5-J5	16	BL-Y		PKB1		K6B1		M1
or	42	Y-O	2	MIA2	2		2	T2
7-J9	17	O-Y		MIB2				R2
	43	Y-G		POA2		K5A2		TR2
	18	G-Y		POB2		K5B2		RR2
	44	Y-BR		PKA2		K6A2		E2
	19	BR-Y		PKB2		K6B2		M2
	45	Y-S	3	MIA3	3		3	T3
	20	S-Y		MIB3				R3
	46	V-BL		POA3		K5A3		TR3
	21	BL-V		POB3		K5B3		RR3
	47	V-O		PKA3		K6A3		E3
	22	O-V		PKB3		K6B3		M3
	48	V-G	4	MIA4	4		4	T4
	23	G-V		MIB4				R4
	49	V-BR		POA4		K5A4		TR4
	24	BR-V		POB4		K5B4		RR4
	50	V-S		PKA4		K6A4		E4
	25	S-V		PKB4		K6B4		M4

SX-200 Peripheral Cabinet Tip/Ring Assignments (Low-power Slots)

Table: SX-200 Peripheral Cabinet Tip/Ring Assignments (Low-power Slots)

J7 and J11 Bay 2 slots 1-4			Lead Designation					
Slot/ Plug	**Pin**	**Pair**	**Cct**	**ONS or DLC**	**Cct**	**LS/GS**	**Cct**	**LS/CLASS**

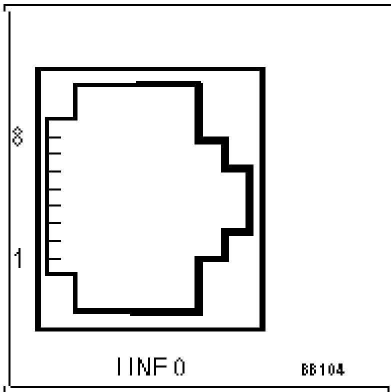
Table: SX-200 Peripheral Cabinet Tip/Ring Assignments (Low-power Slots)

J7 and J11 Bay 2 slots 1-4			Lead Designation					
Slot/ Plug	Pin	Pair	Cct	ONS or DLC	Cct	LS/GS	Cct	LS/CLASS
2-J7 or 4-J11	26	W-BL	1	T1	1	T1	1	T1
	1	BL-W		R1		R1		R1
	27	W-O	2	T2		MM1	2	T2
	2	O-W		R2		M1		R2
	28	W-G	3	T3	2	T2	3	T3
	3	G-W		R3		R2		R3
	29	W-BR	4	T4		MM2	4	T4
	4	BR-W		R4		M2		R4
	30	W-S	5	T5	3	T3		
	5	S-W		R5		R3		
	31	R-BL	6	T6		MM3		
	6	BL-R		R6		M3		
	32	R-O	7	T7	4	T4	5	T5
	7	O-R		R7		R4		R5
	33	R-G	8	T8		MM4	6	T6
	8	G-R		R8		M4		R6
	34	R-BR	9	T9	5	T5	7	T7
	9	BR-R		R9		R5		R7
	35	R-S	10	T10		MM5	8	T8
	10	S-R		R10		M5		R8
	36	BK-BL	11	T11	6	T6		
	11	BL-BK		R11		R6		
	37	BK-O	12	T12		MM6		
	12	O-BK		R12		M6		
	38	BK-G	SPARE					
	13	G-BK	SPARE					
1-J7 or 3-J11	39	BK-BR	1	T1	1	T1	1	T1
	14	BR-BK		R1		R1		R1
	40	BK-S	2	T2		MM1	2	T2
	5	S-BK		R2		M1		R2
	41	Y-BL	3	T3	2	T2	3	T3
	16	BL-Y		R3		R2		R3
	42	Y-O	4	T4		MM2	4	T4
	17	O-Y		R4		M2		R4
	43	Y-G	5	T5	3	T3		
	18	G-Y		R5		R3		
	44	Y-BR	6	T6		MM3		
	19	BR-Y		R6		M3		
	45	Y-S	7	T7	4	T4	5	T5
	20	S-Y		R7		R4		R5
	46	V-BL	8	T8		MM4	6	T6
	21	BL-V		R8		M4		R6
	47	V-O	9	T9	5	T5	7	T7
	22	O-V		R9		R5		R7
	48	V-G	10	T10		MM5	8	T8
	23	G-V		R10		M5		R8

Table: SX-200 Peripheral Cabinet Tip/Ring Assignments (Low-power Slots)								
J7 and J11 Bay 2 slots 1-4			**Lead Designation**					
Slot/ Plug	**Pin**	**Pair**	**Cct**	**ONS or DLC**	**Cct**	**LS/GS**	**Cct**	**LS/CLASS**
	49	V-BR	11	T11	6	T6		
	24	BR-V		R11		R6		
	50	V-S	12	T12		MM6		
	25	S-V		T12		M6		

T1 Trunk Port

Table: T1 Trunk Port (J5 and J6)

Pin	**Signal**
1	RxRing
	
2	RxTip
3	nc
4	TxRing
5	TxTip
6	nc
7	nc
8	nc

SFT Port

Table: SFT Port (J7)

Pin	**Signal**

1	nc
2	-48 V SFT
3	-48 R SFT
4	SFT0R
5	SFT0
6	nc

Maintenance Terminal and Printer Ports

Table: Maintenance Terminal and Printer Ports

Pin	Signal
1	DTR *
2	RXD
3	TXD
4	DTR *
5	GND
6	DSR *
7	RTS **
8	CTS **
9	Not connected

Note: The RS-232 ports (Maintenance, Printer, J10, J11) are all configured as DTE type. All backplane connections use DB-9 connectors. The pins with * and ** are connected together on PRI maintenance ports (slots 10 and 11).

Maintenance Module Port

Table: Maintenance Module Port

Pin	Signal
1	RTS *
2	CTS *
3	TXD
4	GND
5	RXD
6	DSR **
7	DTR **
8	DCD **

Notes: The pins with * and ** are connected together.

Maintenance Module RS-232

Table: Maintenance Module RS-232 Cable

D-Sub Connector Pin	Signals	DIN Connector Pin
2	RXD	5
3	TXD	3
4	DTR	7
5	GND	4*
6	DSR	6
7	RTS	1
8	CTS	2
		8*

Notes:

1. The pins with * are connected together.
2. Pin 1 and 9 of the D-Sub connector are not used.

USOC Connector PINs

Table: USOC Connector Pin Designation

		Connector Type		
		RJ21X	RJ2EX	RJ2GX
26	W-BL	T	T	T
1	BL-W	R	R	R
27	W-O	T	E	T1
2	O-W	R	M	R1
28	W-G	T	T	E
3	G-W	R	R	M
29	W-BR	T	E	T
4	BR-W	R	M	R
30	W-S	T	T	T1
5	S-W	R	R	R1
31	R-BL	T	E	E
6	BL-R	R	M	M
32	R-O	T	T	T
7	O-R	R	R	R
33	R-G	T	E	T1
8	G-R	R	M	R1
34	R-BR	T	T	E

Table: USOC Connector Pin Designation

		Connector Type		
		RJ21X	RJ2EX	RJ2GX
9	BR-R	R	R	M
35	R-S	T	E	T
10	S-R	R	M	R
36	BK-BL	T	T	T1
11	BL-BK	R	R	R1
37	BK-O	T	E	E
12	O-BK	R	M	M
38	BK-G	T	T	T
13	G-BK	R	R	R
39	BK-BR	T	E	T1
14	BR-BK	R	M	R1
40	BK-S	T	T	E
15	S-BK	R	R	M
41	Y-BL	T	E	T
16	BL-Y	R	M	R
42	Y-O	T	T	T1
17	O-Y	R	R	R1
43	Y-G	T	E	E
18	G-Y	R	M	M
44	Y-BR	T	T	T
19	BR-Y	R	R	R
45	Y-S	T	E	T1
20	S-Y	R	M	R1
46	V-BL	T	T	E
21	BL-V	R	R	M
47	V-O	T	E	T
22	O-V	R	M	R
48	V-G	T	T	T1
23	G-V	R	R	R1
49	V-BR	T	E	E
24	BR-V	R	M	M

Table: USOC Connector Pin Designation

		Connector Type		
		RJ21X	**RJ2EX**	**RJ2GX**
50	V-S	T	SPARE	
25	S-V	R	SPARE	

Notes:

1. RJ21X is a standard trunk.
2. RJ2EX is a two-wire E&M trunk.
3. J2GX is a four-wire E&M trunk.

Music on Hold/Pager Pinouts

Table: Music-on-Hold/Pager Unit Pinouts

Wire	Signal	Color	Description
32	MOH-	R-O	Music-on-Hold Inputs
7	MOH+	O-R	
34	PAGE-	R-BR	Paging Input/Output
9	PAGE+	BR-R	
36	PAGEREL(NC)	BK-BL	Page Relay, Normally Closed contact
11	PAGEREL(C)	BL-BK	Page Relay Common
37	PAGEREL(NO)	BK-O	Page Relay, Normally Open contact
12	PAGEREL(C)	O-BK	Page Relay Common
39	NIGHTBELL1-	BK-BR	Night Bell 1 relay contacts
14	NIGHTBELL1+	BR-BK	
41	NIGHTBELL2-	Y-BL	Night Bell 2 relay contacts
16	NIGHTBELL2+	BL-Y	
45	ALARM-	Y-S	Alarm relay contacts
20	ALARM+	S-Y	
50	RING	V-S	Connection to DNIC Line
25	TIP	S-V	Connection to DNIC Line

CX/CXi Controller

Table: Cable Terminations	
Table	Analog Main Board Pinouts
Table	Analog Option Board Pinouts
Table	T1/E1 Combo Module Tip/Ring Pinouts
Table	RS-232 Maintenance Port Pinouts
Table	10/100Base-TX Ethernet Port Pinouts
Table	10/100/1000Base-T Ethernet Port Pinouts
Table	ASU II 25-Pair Male D-Type Connector Pinout

Analog Main Board Pinouts (CX/CXi Controller)

Table: Analog Main Board Pinouts (CX Controller)

Port	Bay/Slot/Circuit	Pins	Signal
Music on Hold	1/13/29/1	1	Common
		2	MOH IN 1
		3	MOH IN 2
Pager Relay 1	1/13/30/2	1	Normally Closed
		2	Common
		3	Ring
		4	Tip
		5	Normally Open
		6	
Generic Relay 1 Generic Relay 2	1/13/30/3 1/13/30/4 (See Note 3)	1	Relay 2 Normally Closed
		2	Relay 2 Common
		3	Relay 1 Common
		4	Relay 1 Normally Open
		5	Relay 2 Normally Open
		6	Relay 1 Normally Closed
ONS 1	1/13/3	3 / 4	Ring / Tip
ONS 2	1/13/4	3 / 4	Ring / Tip

Table: Analog Main Board Pinouts (CX Controller)

Port	Bay/Slot/Circuit	Pins	Signal
ONS 3	1/13/5	3 / 4	Ring / Tip
ONS 4	1/13/6	3 / 4	Ring / Tip
LS 1 (See Note)	1/13/7	3 / 4	Ring / Tip
LS 2 (See Note)	1/13/8	3 / 4	Ring / Tip
LS 3	1/13/9	3 / 4	Ring / Tip
LS 4	1/13/10	3 / 4	Ring / Tip
LS 5	1/13/11	3 / 4	Ring / Tip
LS 6	1/13/12	3 / 4	Ring / Tip

Notes:

1. The Music on Hold signal pins (IN 1 and IN 2) connect to the left and right channels of a 3.5 mm stereo jack.
2. LS/CLASS trunks at locations 1/13/7 and 1/13/8 are the System Fail Transfer trunks. They connect to ONS phones located at 1/13/3 and 1/13/4 respectively.
3. PLID 1/13/30/4 is programmed for Night Bell use in the default database.

Analog Option Board Pinout Assignments (CX.CXi Controller)

Table: Analog Option Board Pinout Assignments (CX/CXi Controller)

Port	Bay/Slot/Circuit	Pin	Signal
Pager Relay 2	1/13/31/2	1	Normally Closed
		2	Common
		3	Ring
		4	Tip
		5	Normally Open
		6	
Generic Relay 3 Generic Relay 4	1/13/31/3 1/13/31/4	1	Relay 4 Normally Closed
		2	Relay 4 Common
		3	Relay 3 Common
		4	Relay 1 Normally Open
		5	Relay 4 Normally Open
		6	Relay 3 Normally Closed

Table: Analog Option Board Pinout Assignments (CX/CXi Controller)

Port	Bay/Slot/Circuit	Pin	Signal
ONS 1	1/13/19	3 / 4	Ring / Tip
ONS 2	1/13/20	3 / 4	Ring / Tip
ONS 3	1/13/21	3 / 4	Ring / Tip
ONS 4	1/13/22	3 / 4	Ring / Tip
LS 1 (See Note)	1/13/13	3 / 4	Ring / Tip
LS 2 (See Note)	1/13/14	3 / 4	Ring / Tip
LS 3	1/13/15	3 / 4	Ring / Tip
LS 4	1/13/16	3 / 4	Ring / Tip
LS 5	1/13/17	3 / 4	Ring / Tip
LS 6	1/13/18	3 / 4	Ring / Tip
Note: LS/CLASS trunks at locations 1/13/13 and 1/13/14 are the System Fail Transfer trunks. They connect to ONS phones located at 1/13/19 and 1/13/20 respectively.			

T1/E1 Combo Module Tip/Ring Assignments (CX/CXi and AX Controller)

Table: T1/E1 Combo Module Tip/Ring Assignments (CXCXi and AX Controller)

Connector Pin	Signal	NT/LT Setting (software selectable)	
NT (Default)	LT		
1	--	Rx Ring	Tx Ring
2	--	Rx Tip	Tx Tip
3	N/C	--	--
4	--	Tx Ring	Rx Ring
5	--	Tx TIp	Rx Tip
6	N/C	--	--
7	N/C	--	--
8	N/C	--	--

10/100/1000Base-T Ethernet Port Pinouts (CX/CXi Controller)

10/100/1000Base-T connectors can both transmit (TX) and receive (RX).

Table: Ethernet Port Pinouts (CXi Controller)

Connector Pin	Signal
1	TX/RX 0 +
2	TX/RX 0 -
3	TX/RX 1 +

Table: Ethernet Port Pinouts (CXi Controller)

Connector Pin	Signal
4	TX/RX 2 +
5	TX/RX 2 -
6	TX/RX 1 -
7	TX/RX 3 +
8	TC/RX 3 -

10/100Base-TX Ethernet Port Pinouts (CX/CXi Controller)

10/100Base-TX connectors can either transmit (TX) or receive (RX).

Table: Ethernet Port Pinouts (CX/CXi Controller)

Connector Pin	Signal
1	RX +
2	RX -
3	TX +
4	Not Used
5	Not Used
6	TX -
7	Not Used
8	Not Used

RS-232 Maintenance Port Pinouts (CX/CXi Controller)

Table: RS-232 Maintenance Port Pinouts (CX/Cxi Controller)

Connector Pin	Signal Name	Abbreviation
1	Data Carrier Detect	DCD
2	Receive Data	RXD
3	Transmit Data	TXD
4	Data Terminal Relay	DTR
5	Ground	GND
6	Data Set Ready	DSR
7	Request to Send	RTS
8	Clear to Send	CTS
9	Ring Indication	RI

ASU II 25-Pair Male D-Type Connector Pinout

Table: ASU II Connector Pinout

Pin	Color Code	24 Port ONS	PLID	16 Port ONS	PLID	4 + 12 Port Combo	PLID
26/1	W/BL, BL/W	ONS Tip/Ring 1	n 1 x 1	ONS Tip/Ring 1	n 1 x 1	ONS Tip/Ring 1	n 1 x 1
27/2	W/O, O/W	ONS Tip/Ring 2	n 1 x 2	ONS Tip/Ring 2	n 1 x2	ONS Tip/Ring 2	n 1 x2

28/3	W/G, G/W	ONS Tip/Ring 3	n 1 x 3	ONS Tip/Ring 3	n 1 x 3	ONS Tip/Ring 3	n 1 x 3
29/4	W/BR, BR/W	ONS Tip/Ring 4	n 1 x 4	ONS Tip/Ring 4	n 1 x 4	ONS Tip/Ring 4	n 1 x 4
30/5	W/S, S/W	ONS Tip/Ring 5	n 1 x 5	ONS Tip/Ring 5	n 1 x 5	ONS Tip/Ring 5	n 1 x 5
31/6	R/BL, BL/R	ONS Tip/Ring 6	n 1 x 6	ONS Tip/Ring 6	n 1 x 6	ONS Tip/Ring 6	n 1 x 6
32/7	R/O, O/R	ONS Tip/Ring 7	n 1 x 7	ONS Tip/Ring 7	n 1 x 7	ONS Tip/Ring 7	n 1 x 7
33/8	R/G, G/R	ONS Tip/Ring 8	n 1 x 8	ONS Tip/Ring 8	n 1 x 8	ONS Tip/Ring 8	n 1 x 8
34/9	R/BR, BR/R	ONS Tip/Ring 9	n 1 x 9	ONS Tip/Ring 9	n 1 x 9	ONS Tip/Ring 9	n 1 x 9
35/10	R/S, S/R	ONS Tip/Ring 10	n 1 x 10	ONS Tip/Ring 10	n 1 x 10	ONS Tip/Ring 10	n 1 x 10
36/11	BK/BL, BL/BK	ONS Tip/Ring 11	n 1 x 11	ONS Tip/Ring 11	n 1 x 11	ONS Tip/Ring 11	n 1 x 11
37/12	BK/O, O/BK	ONS Tip/Ring 12	n 1 x 12	ONS Tip/Ring 12	n 1 x 12	ONS Tip/Ring 12	n 1 x 12
38/13	BK/G, G/BK	ONS Tip/Ring 13	n 1 x 13	ONS Tip/Ring 13	n 1 x13	N/C	
39/14	BK/BR, BR/BK	ONS Tip/Ring 14	n 1 x 14	ONS Tip/Ring 14	n 1 x 14	N/C	
40/15	BK/S, S/BK	ONS Tip/Ring 15	n 1 x 15	ONS Tip/Ring 15	n 1 x 15	N/C	
41/16	Y/BL, BL/Y	ONS Tip/Ring 16	n 1 x 16	ONS Tip/Ring 16	n 1 x 16	N/C	
42/17	Y/O, O/Y	ONS Tip/Ring 17	n 1 x 17	N/C		N/C	
43/18	Y/G, G/Y	ONS Tip/Ring 18	n 1 x 18	N/C		N/C	
44/19	Y/BR, BR/Y	ONS Tip/Ring 19	n 1 x 19	N/C		N/C	
45/20	Y/S, S/Y	ONS Tip/Ring 20	n 1 x 20	N/C		N/C	
46/21	V/BL, BL/V	ONS Tip/Ring 21	n 1 x 21	N/C		LS Ring/Tip 1	
47/22	V/O, O/V	ONS Tip/Ring 22	n 1 x 22	N/C		LS Ring/Tip 2	
48/23	V/G, G/V	ONS Tip/Ring 23	n 1 x 23	N/C		LS Ring/Tip 3	
49/24	V/BR, BR/V	ONS Tip/Ring 24	n 1 x 24	N/C		LS Ring/Tip 4	
50/25	V/S, S/V	N/C		N/C		N/C	

CIM Port Allocation

The CIM ports require standard 8-pin modular jacks (RJ-45) consisting of 2 balanced signal pairs on a Category 5 Universal Twisted Pair (UTP) crossover cable. The twisted pairs are arranged as: 1,2: 3,6; 4,5; 7,8. Each tied pair is connected to a 75 ohm resistor. The following table lists the signal for each pin:

Table: CIM Port Allocation

Pin Number	Signal

Table: CIM Port Allocation

Pin Number	Signal
1	RX+
2	RX-
3	TX+
4	-
5	-
6	TX-
7	-
8	-

Testing Ground Paths

Introduction to Ground Paths

A ground path protects people and equipment from harm by safely redirecting unwanted electrical current to a predetermined ground/earth location.

Mitel recommends that you provide two types of protective ground paths to the SX-200 ICP. The two ground path types are

- safety ground
- system ground.

Safety Ground

The safety ground absorbs dangerous voltages that come in contact with the SX-200 ICP cabinet. The safety ground connects to the SX-200 ICP through the ground termination of the system’s three-prong-power-cord.

System Ground

The system ground provides a stable ground reference for voltages used by the system. The system ground attaches to the SX-200 ICP through a separate ground wire connected directly to a system cabinet ground-stud. The recommended wire size for the system ground is 6 AWG (American Wire Gauge).

The system ground also provides an additional safety ground as it can typically carry more current than the safety ground.

Typical Ground Path Connections

In most buildings the cold water system provides the ground source.

The cold water pipe must provide a metallic connection all the way back to the building entry point. To be an effective ground, the water pipe must be buried in moist earth to a depth of at least 3 meters (10 feet).

Figure: Typical Ground Path Connections

Ground Connections

The best ground connection to the cold water system is on the street-side of the building before the water meter.

If you connect to the cold water system on the building side of the water meter you must ensure there is a metallic strap around the water meter. The metallic strap connects to the same water pipe on the street-side of the building and must have conductivity characteristics equal to the cold water pipe.

Any connection to the cold water system must be a solid metal-to-metal contact. When you install a ground connection scrape all paint and corrosion away from the ground point and securely clamp the ground wire to the exposed metal surface.

Note: The water in the pipe is not part of the grounding system.

Problems Caused by Improper Grounding

Improper grounding can disrupt system operation in three ways:

- Misinterpretation of central office trunk signaling (DC differential)
- Disruption of logic circuit operation (AC differential)
- Interference in the audio (RF differential).

Misinterpretation of Central Office Trunk Signaling (DC Differential)

The SX-200 ICP uses Direct Current (DC) signals to supervise the state of Central Office (CO) trunk circuits and compares these signals to known voltage reference points. If the CO and the SX-200 ICP have different ground reference voltages each can fail to recognize signals sent by the other. For example:

- Ground start trunks may not seize when the SX-200 ICP grounds the ring lead
- CO may not release trunks when the SX-200 ICP removes its termination.

Disruption of Logic Circuit Operation (AC Differential)

Low frequency Alternating Current (AC) ground differentials can disrupt the operation of SX-200 ICP logic circuits. These circuits control the operation of the system and when damaged cause incorrect operation or system failures.

Interference in the Audio (RF Differential)

Alternating Current (AC) ground differentials at radio frequencies can cause audio interference and possibly disrupt SX-200 ICP logic circuits.

Testing the Safety and System Ground Paths

To test the safety and system ground paths you measure the Alternating Current (AC) voltage and Resistance present in the ground loop.

You must perform the following Ground Path Tests:

- Ground Path AC Voltage Test
- Ground Path Resistance Test.

Ground Path AC Voltage Test

Safety Ground To Water Pipe

This test measures the presence of AC voltage in the metallic loop.

If the system ground and safety ground both connect to a building ground your measurement is the metallic loop from the SX-200 ICP chassis, to the electrical panel ground to the cold water pipe and back through the system ground wire.

1. Disconnect power to SX-200 ICP.
2. At the PBX system connect the cabinet's chassis ground to the electrical panel ground. The wire should be a minimum 10 AWG and not longer than 15 meters.
3. At the SX-200 ICP, disconnect the ground wire from the system's ground-stud.
4. Connect the voltmeter as shown in the Ground Path AC Voltage Test - Safety Ground To Water Pipe Figure or the Ground Path AC Voltage Test - Safety Ground To Earth Figure.
5. Start with your meter set for high AC voltages and adjust it down until you get a reading.
6. You should get a reading of 1 Vac or less.

If your reading is greater than 1 Vac check to see if your electrical panel ground connects to the building ground. Depending on local utility regulations the connection usually exists between the electrical panel ground and a cold water pipe entering the building. If this connection is present, try an alternate ground point(s) and measure the AC voltage again.

Figure: Ground Path AC Voltage Test - Safety Ground To Water Pipe

Safety Ground To Earth

If the safety ground and system ground do not connect at the electrical panel your measurement indicates the AC voltage differential between protective earth and the building ground.

Figure: Ground Path AC Voltage Test - Safety Ground To Earth

Ground Path Resistance Test

Safety Ground To Water Pipe

With your meter connected in the same way as you did for the AC Voltage Test, follow these steps to test the resistance in the ground path.

1. Set your meter to OHMS at the highest scale
2. Measure the resistance between the SX-200 ICP chassis safety ground and the ground wire that provides the system ground. Adjust your meter down until you get a reading.
3. The resistance between the two grounds should be less than 5 OHMS. If not, try alternate ground point(s) and repeat the test.

Figure: Ground Resistance Test - Safety Ground To Water Pipe

If the system ground and safety ground both connect to a cold water pipe or other building ground your measurement indicates the resistance of the metallic loop from the SX-200 ICP chassis, to the electrical panel ground, to the cold water pipe and back through the SX-200 ICP system ground wire.

Note: This test assumes there is good conductance from the street-side of the water meter to ground. If you are not sure of your connection to ground then perform the CO/PBX Ground Differential Test described later in this document.

Safety Ground To Earth

If the safety ground and system ground do not connect at the electrical panel your measurement is the resistance of the ground loop from the SX-200 ICP chassis, through the safety ground conductor to protective earth, through the earth to the building ground, and back through the SX-200 ICP ground wire.

Figure: Ground Path Resistance Test - Safety Ground To Earth

Testing the Central Office (CO) and SX-200 ICP Ground Potential

If the SX-200 ICP experiences trunk lock-ups or trunk seize failures then perform the CO/ICP Ground Differential Test.

What Will The Test Determine?

This test determines if the Direct Current (DC) ground potential between the selected building ground point and the Central Office (CO) ground point are within acceptable limits.

What Do I Test?

With this test you measure two currents:

- "Loop" current
- "Ring" current.

A calculation using both measured values determines the relation of the building ground potential to the CO ground potential.

How Do I Calculate The CO/ICP Ground Differential?

You divide the measured "Ring" current by the measured "Loop" current. If the CO and the SX-200 ICP ground points are the same potential the result will be 2.0, a perfect match.

A result between 1.85 and 2.15 indicate an acceptable building ground. A result outside the upper and lower limits means you must locate an alternate building ground.

Ring/Loop = 1.85 to 2.15

CO/ICP Ground Differential Test

Follow these steps to test the CO/ICP Ground Differential:

1. Disconnect the building ground wire from the SX-200 ICP ground-stud.
2. Disconnect a loop start or ground start trunk from the SX-200 ICP.
3. Measure the Loop current.
   - Set the meter to Milliamperes = DC and Range = 200 Milliamperes
   - Connect the meter between the Tip and Ring trunk leads
   - For a ground start trunk, apply the building ground momentarily to the ring side of the trunk. This will signal the CO to complete the loop and provide DC loop current
   - Allow sufficient time for the current to stabilize and record the loop current
   - DC Loop Current = ________________________________

Figure: Measure The "Loop" Current

4. Measure the Ring current:
   - Set the meter to the same settings

  - Connect the meter between the Ring trunk lead and the building's open-ended ground wire. Be sure you disconnect the building ground wire from the SX-200 ICP ground-stud

5. Allow sufficient time for the current to stabilize and record the Ring current.

   DC Ring current = ________________________

Figure: Measure The "Ring" Current

6. Calculate the CO/ICP ground potential.
   - Divide the Ring current value by the Loop current value

     Ring/Loop = 1.85 to 2.15
   - An ideal ground result is 2.0.
   - An acceptable ground will give a result between 1.85 to 2.15
   - If the result falls outside this range you must find an alternative building ground for your SX-200 ICP ground connection.

Special Accessories Required for Compliance with FCC Rules Part 15

The following special accessories are used with the SX-200 ICP. Refer to the appropriate section for special accessory installation information.

Special Accessory	Special Accessory Installation Information
Split Ferrite Block Chart	Connect Cables Between SX-200 ICP and Cross-Connect Field

Install FRUs

Ordering Information

Ordering information for the Mitel SX-200 ICP is divided into the following tables.

- SX-200 ICP Hardware & Software
- Peripheral Interface Cards and Modules Table lists digital cards and modules required for SX-200 Peripheral Cabinets
- Digital Control and Digital Services Cards and Modules, and Spares Table lists spares for SX-200 Peripheral Cabinets components
- Peripheral Equipment Table lists peripheral equipment

Warranty

Mitel PBX communications system is warranted against defective material and workmanship. Equipment requiring service or repair during the warranty period must be packaged in accordance with the Repack Equipment For Shipment Chart and returned prepaid to the supplier. Repaired or replacement equipment is returned to the customer, prepaid by Mitel.

Spares Level

Mitel recommends that the minimum sparing level be one replacement unit for any part installed in the field, including Flash memory cards, for the first system installed. This would be 100% spares for the first system installed. As the number of installed systems increases, the sparing level should decrease to 10% of the installed units.

Hardware & Software

Table: Part Numbers for SX-200 ICP Hardware and Software

Common Name	Part Number	Comments
SX-200 ICP MX Controller	50004357	With internal hard drive
SX-200 ICP CX Controller	50005382	With internal CompactFlash (512MB)
SX-200 ICP CXi Controller	50005381	With internal CompactFlash (512MB)
SX-200 ICP AX Controller Chassis	50006161	With Music on Hold Input Adapter (56009491)
SX-200 AX Controller Card	50006181	With internal CompactFlash (4GB), 512MB Sodimm RAM and i-Button
SX-200 NSU	500034993	Network Services Unit (NSU) provides T1 connectivity.
SX-200 ASU	50001267	Analog Service Unit (ASU) with 24 ONS Ports.
SX-200 ASU II	50005105	Analog Service Unit (ASU) with up to 48 ONS Ports or a combination of ONS and LS.
512 MB CompactFlash	50005551	Internal CompactFlash.
24-port ONSp Card	50005731	ASU II option card
16-port ONS Card	50005103	ASU II option card
16-port ONSp Card	50005533	ASU II option card with lightning protection
4 + 12 Port Combo Card	50005104	ASU II option card
CompactFlash Memory Interface MMC	50003727	Carrier card for CompactFlash memory card.
External CompactFlash Memory card (blank)	50004154	256MB
DSP Modules	50003728 50002979	

Dual DSP MMC
Quad DSP MMC

Expansion of three-party and Record a Call conferences, G.729 codes, TDM (non-IP) devices and Digital trunks. See DSP Resources.		
T1/E1 Combo MMC (original)	50004402	Combines a T1 digital trunk port and DSP functionality in one card. Also includes a stratum 4 clock.
T1/E1 Combo MMC II	50005160	Combines a T1 digital trunk port and DSP functionality in one card. Also includes a stratum 4 clock.**Note**: this card has two RJ-45 ports but the second is not functional on the CX controller.
Dual FIM 820NM - 1 km Multi-Mode MMC	50001248	Supports Peripheral cabinets, PRI Cards, and NSUs.
Dual FIM 1300NM - 5 km Multi-mode MMC	50003695	
Dual FIM 1300NM - 14 km Single-mode MMC	50003696	
SX-200 ICP Dual FIM Interface Module	52001391	
Quad CIM MMC (CX, CXi, MX)	50004451	Supports Peripheral cabinets, PRI Cards, and NSUs.
Dual T1/E1 Framer MMC	50003560	Two digital trunk ports. Each port can be configured as a T1 (1.544 Mbits/sec) interface.
Application Processor Card (APC Solution)	51010725	Embedded PC that hosts the Mitel Managed Application Server.
APC Hard Drive (media kit)	50005413	Hard drive and mounting hardware for APC installations.
Fan (MX)	50003884	
Fan (AX)	50005471	
Power Supply (MX)	50003885	
Power Supply (AX)	50005182	
Stratum 3 Clock Module (MX or CX)	50003726	Required to support digital trunks.
Analog Main Board (MX)	50003724	6 LS/CLASS, 2 ONS/CLASS and 2 DNIC circuits, 2 PFT (Power Fail Transfer) circuits, 1 MOH (Music On Hold) port to connect an external audio source, 1 Loudspeaker port to connect to an external paging system. This card installs in the MX controller.
Analog Option Board (MX)	50003725	6 LS/CLASS and 2 ONS/CLASS circuits for expansion and sits on top of the Analog Main Card in the MX controller.

Analog Main Board (CX/CXi)	50004403 (v1) 50004870 (v2) 56008157 (v3)	

6 LS/CLASS and 4 ONS/CLASS ports, 2 PFT (Power Fail Transfer) circuits, 1 MOH (Music On Hold) port, 1 Page port, 1 Relay port with 2 generic relay circuits for auxillary bell, door sensor, or alarm. This card installs in the CX controller.

Note: The AMB and AOB must be the compatible. **Note**: AMB version 3 has an extra connector on the back of the board for future use. Ignore this connector when connecting the AMB v3 to the controller.

Analog Option Board (CX/CXi)	50004401 v1 50004871 (v2)	

6 LS/CLASS and 4 ONS/CLASS ports, 2 PFT (Power Fail Transfer) circuits, 1 Page port, 1 Relay port with 2 generic relay circuits for auxillary bell, door sensor, or alarm. This card sits on top of the Analog Main Card in the CX controller.

Note: The AMB and AOB must be compatible.

R4.0 Software CD-ROM	500055463	Includes English, French, and Spanish voice mail language files
Hard drive and IDE cable	50004151	Unformatted 80 GB Hitachi
Rack mount brackets	50004150	Suitable for SX-200 ICP MX and NSU
Your Assistant Starter Kit	54001761	Software and single user license
Your Assistant Upgrade Kit	54002200	Single user license
Your Assistant Softphone	54002201	Single user license
Your Assistant Lite	54002202	Single user license

Software Options

Table: Part Numbers for SX-200 ICP Software (Feature Options Record) Options

Description	Part Number	Comments
ACD Agents (11-15)	9109-531-015-NA	
ACD Agents (1-5)	9109-531-005-NA	
ACD Agents (16-25)	9109-531-025-NA	
ACD Agents (26-50)	9109-531-050-NA	
ACD Agents (51-75)	9109-531-075-NA	
ACD Agents (6-10)	9109-531-010-NA	
ACD Agents (76-100)	9109-531-100-NA	

ACD Real Time Events	9109-532-013-NA	
Auto-Attendant	9109-532-001-NA	
Centralized Voice Mail & Attendant S/W	9109-532-006-NA	
CLASS Sets	9109-532-022-NA	
Compression Resource Licenses (2)	54000925	
Compression Resources (8)	54000890	
DID Server Application	54002761	
Digital Bay Licenses (1)	54000924	
Email (SMTP) Client	54000931	
Fax Tone Detect	9109-532-010-NA	
Feature Level 5	54000920	
Feature Level 6	54001230	
Guest Suites	54000131	
Hospitality	54001134	
Internet Gateway	54001231	
IP Network Channel (12)	54001152	
IP Network Channel	54000661	
IP Network Channel (2)	54001151	
IP Phone Licenses (4)	54000923	
IP Set (1)	54000660	
Max Digital Line Cards	9109-533-002-NA	
Max ONS Line Cards	9109-533-101-NA	
MiTAI	9109-532-002-NA	
PC (2nd) Port on IP Phones Licenses	54000929	
Phonebook licenses	54000928	
PRI Card Auto Min/Max	9109-532-016-NA	
PRI Card D Channel Backup	9109-532-018-NA	
PRI Card Min/Max	9109-532-015-NA	
PRI Card NFAS	9109-532-017-NA	
PRI Card QSIG	9109-532-020-NA	
PRI Card Remote LAN Access	9109-532-019-NA	

Record a Call	9109-532-021-NA	
Recorded Announcement Device	54000927	
Remote S/W Download	9109-532-005-NA	
Speak@Ease	54000050	
SS4000 Sets	9109-532-007-NA	
Standard Unified Messaging	50001353	
Support 3DN, 4DN and 400 series Set Types	9109-532-012-NA	
Support Softkey Access to Voicemail	9109-532-011-NA	
SX200 Max. 1 Digital Link	9109-533-201-NA	
SX200 Max. 2 Digital Links	9109-533-202-NA	
SX200 Max. 3 Digital Links	9109-533-203-NA	
SX200 Max. 4 Digital Links	9109-533-204-NA	
SX200 Max. 5 Digital Links	9109-533-205-NA	
SX200 Max. 6 Digital Links	9109-533-206-NA	
SX200 Max. 7 Digital Links	9109-533-207-NA	
SX200 Max. 8 Digital Links	9109-533-208-NA	
TDM Device License (4)	54000970	
Voice Mail Bilingual Prompts	54000950	
Voice Mail Box License(1)	54000921	
Voice Mail Boxes License (4)	54000922	
Voice Mail Personal Contacts	54000951	
Voice Mail PMS	54000930	

Peripheral Cabinets, Interface Cards, and Modules

Table: Part Numbers for Peripheral Interface Cabinets, Cards and Modules

Common Name	Part Number	Comments
ONS/CLASS Line Card	9109-110-001-NA	For Rotary and DTMF Telephones (12 circuits per card)
ONS/CLASS Line Card (Blue Faceplate)	9109-110-002-NA	For Rotary and DTMF Telephones (12 circuits per card)
OPS Line Card	9109-040-000-SA	6 OPS Circuits per card

Digital Line Card (Blue Faceplate)	9109-012-002-NA	12 DNIC Circuits per card
Digital Line Card	9109-012-001-NA	12 DNIC Circuits per card
LS/CLASS Trunk Card	50001730	8 CO Trunk Interfaces (Loop Start only)
LS/GS Trunk Card - CSA	9109-011-001-SA	6 CO Trunk Interfaces
DID Trunk	9109-031-000-SA	6 DID Circuits per card
Universal Card	9109-005-000-SA	supports: - E&M Trunk Module - Music-on-Hold/Pager Module - Receiver/Relay Module
E&M Trunk Module	9109-013-000-SA	1 E&M Trunk Circuit per Module
Music-on-Hold/Pager Module	9109-018-000-SA	Music Input, Paging Preamp Output, and Relay to control external amplifier
Receiver/Relay Module	9109-016-000-SA	4 DTMF Receivers and 2 General Purpose Relays
Mitel Express Messenger™ Card (2-port)	9109-080-001-NA	All models use the same card and the number of ports enabled is determined by the software. For example, you can upgrade a 2-port card to a 4-port card by enabling the software with a passcode. Refer to the Mitel Express Messenger System Administration Manual (PN 9109-080-005-NA) for instructions.
Mitel Express Messenger Card (4-port)	9109-080-002-NA	
Mitel Express Messenger Card (6-port)	9109-080-009-NA	
Mitel Express Messenger Card (8-port)	9109-080-008-NA	
SX-200 Per Bay Cabinet - Drk Gry	50004041	
SX-200 Per Node BCCII - Drk Gry,110V	52001550	
SX-200 Per Node BCCIII - Drk Gry,110V	52001551	
SX-200 Per Node BCCIII - Drk Gry, 220V	52001552	
SX-200 Drk Gry Rack Mount Bracket FRU	50002891	

Digital Control and Digital Services Cards and Modules, and Spares

Table: Part Numbers for Digital Control and Digital Services Cards and Modules, and Spares

Marketing Name	Part Number	Comments
Bay Control Card II	9109-017-001-SA	
Bay Control Card III	9109-117-001-NA	Requires the SX-200 peripheral cabinet
DSP Module (single)	9180-510-006-NA	
Bay Power Supply 120 V ac	9109-008-000-SA	
Bay Power Supply 230 V ac	9109-008-002-NA	
System ID Module	50002212	
Peripheral FIM Carrier II	9109-612-001-NA	
Peripheral Interface Module Carrier Card	9109-616-001-NA	Installs in any SX-200 rack-mount cabinet
FIM Module 820NM Multi-mode	9180-510-001-NA	
FIM Module 1300NM Multi-mode	9180-510-002-NA	
FIM Module 1300NM Single-mode	9180-510-003-NA	
CIM	9180-510-010-NA	Copper Interface Module
CIM Cable	9125-100-106-NA	Peripheral Interconnect Cable (15 ft.)
T1 Trunk Card	9109-021-001-NA	
T1 Trunk Card	9109-021-000-SA	Includes T1 Trunk Adapter
T1 Trunk Adapter	9400-100-302-NA	
Dual T1 Trunk Adapter	9400-100-304-NA	For 2 links
T1/E1 Module	9180-510-004-NA	Dual link, for PRI card or the BCC III
PRI Card Package	50002285	Includes: • PRI card • Dual link T1 PRI Module • software
PRI Card	9109-615-001-NA	The carrier card for the PRI
PRI Card Software	9125-070-001-NA	
Fan Assembly - SX-200 RM Cabinet	9109-631-000-NA	

Peripheral Equipment

Table: Part Numbers for Peripheral Equipment

Marketing Name	Part Number	Description
Symbol MiNET Wireless Phone	51004423	Symbol MiNET (NetVision II) Wireless Phone Kit, 11Mb, DS
Mitel 5201 IP Phone	50002815	5201 IP Phone Dark Gray FRU
Mitel 5207 IP Phone	50003812	5207 IP Phone Dark Gray
Mitel 5212 IP Phone	50004890	5212 IP Dual Port Dark Grey
Mitel 5215 IP Phone	50002817	5215 IP Dual Port Dark Gray
Mitel 5220 IP Phone	50002818	5220 IP Dual Port Dark Gray
Mitel 5220 IP Phone	50002819	5220 IP Dual Port Red
Mitel 5224 IP Phone	50004894	5224 IP Dual Port Dark Grey
Mitel 5224 IP Phone	50004895	5224 IP Dual Port Dark Red
Mitel 5304 IP Phone	51011571	5304 IP Phone Dark Grey
Mitel 5312 IP Phone	50005847	5312 IP Dual Port Dark Grey
Mitel 5324 IP Phone	50005664	5324 IP Dual Port Dark Grey
Mitel 5330 IP Phone	50005070	5330 IP Dual Port Grey
Mitel 5330 IP Phone with backlight	50005804	5330 IP Dual Port Grey
Mitel 5340 IP Phone	50005071	5340 IP Dual Port Grey
Mitel 5412 PKM (with SIM module)	50002821	5412 PKM Kit (12 Key) Dark Gray
Mitel 5412 PKM	50002822	5412 PKM FRU (12 Key) Dark Gray
Mitel 5448 PKM (with SIM module)	50002823	5448 PKM Kit (48 Key) Dark Gray
Mitel 5448 PKM	50002824	5448 PKM FRU (48 Key) Dark Gray
Mitel 5422 PKM Interface Module	50002825	5422 PKM Interface Module
Mitel Line Interface Module (LIM)	50004197	Line Interface Module (formerly 5425 Access Module)
Mitel 5303 Conference Phone	50001900	Mitel 5303 Conference Phone - Black
Mitel 5303 Conference Phone	50001903	Mitel 5303 Conference Phone - Silver
Mitel 5310/5310R Remote Control Mouse for IP Conf. Unit, Silver	50001542	Wired Remote Control Unit for use with 5310 or 5310R IP Conference Unit,

Table: Part Numbers for Peripheral Equipment

Marketing Name	Part Number	Description
		Dark Grey. Purchased separately.
5310/5310R Remote Control Mouse for IP Conf. Unit, Dark Grey	50001543	Wired Remote Control Unit for use with 5310 or 5310R IP Conference Unit, Silver. Purchased separately
Mitel 5310 IP Conference Saucer - Dark Grey - requires Side Control for 5220/5224 (below)	50004459	For use with: 5310 IP Conference Side Control (5220/5224) (50004461) or 5310 IP Conference Unit Module for 5330/5340 (50005321).
Mitel 5310 IP Conference Saucer - Silver - requires Side Control for 5220/5224 (below)	50004460	For use with: 5310 IP Conference Side Control (5220/5224) (50004461) or 5310 IP Conference Unit Module for 5330/5340 (50005321).
Mitel 5310 IP Conference Unit Side Control for 5220/5224	50004461	For use on 5220 IP Phone. Used to connect 5310 IP Conference Saucer to 5220 IP Phone. Supports 5310 IP Conference Saucer, Silver (50004460) & 5310 IP Conference Saucer, Dark Grey (50004459). 24VDC Power Adapter required to power 5310 IP Conference Side Control for 5220 IP Phone. 24V Power Adapters (50000685, 50000687 or 50000690) sold separately.
5310 IP Conference Module (5300 Series)	50005321	For use on 5330, and 5340 IP Phones. Used to connect 5310 IP Conference Saucer to IP Phone. Supports 5310 IP Conference Saucer, Silver (50004460) & 5310 IP Conference Saucer, Dark Grey (50004459). Separate Power Adapter not required to power 5310 Conference Module for 5330 and 5340. Supports Power Over Ethernet from the 5330 and 5340 IP Phones.
Mitel 5201 IP Phone	50002832	5201 IP Phone Dark Gray 5PK.

Table: Part Numbers for Peripheral Equipment

Marketing Name	Part Number	Description
Mitel Gigabit Ethernet (GigE) Phone Stand	51009841	Ethernet phone stand for 5200 series Dual Mode phones.
Mitel Gigabit Ethernet (GigE) Phone Stand (v2)	50006371	Ethernet phone stand for 5300/5200 series IP phones
Mitel Wireless LAN (WLAN) Phone Stand	51009840	Wireless phone stand with WiFi technology built in.
C7 Power Cord for GigE and WLAN stands (country-specific) **Note**: If other country-specific C7 plug types are required, they must be sourced locally.	51005172	C7 power cord with NA plug type
51000582	C7 power cord with UK plug type	
51004990	C7 power cord with Euro plug type	
Mitel Cordless Handset and Module	57005711	Cordless Module and Handset
Mitel Cordless Headset and Module	57005712	Cordless Module and Headset
Mitel Cordless Handset	50005405	Cordless Handset only
Mitel 5540 IP Attendant Console	50005811	Dark grey IP console with a tilted, back-lit LCD display, English and French text available
Mitel 5540 Lens and Label Kit	51012286	Lens and desi card for console
Mitel GN Cordless Headset Cable (for 5540)	50006212	Connects the 5540 IP Console to the GN9350e Cordless Headset. Purchased separately.
Mitel Cordless Headset	50005522	Cordless Headset only
SUPERCONSOLE 1000 attendant console (English)	9189-000-001-NA	Console with tilted LCD display (interfaces to Digital Line Card) and English documentation
SUPERCONSOLE 1000 attendant console (French)	9189-000-300-NA	Console with tilted LCD display (interfaces to Digital Line Card) and French documentation
SUPERCONSOLE 1000 attendant console	9189-000-300-NA	Light grey console with a tilted, back-lit LCD display, English and French text available
SUPERCONSOLE 1000 attendant console	9189-000-301-NA	Dark grey console with a tilted, back-lit LCD display, English and French text available

Table: Part Numbers for Peripheral Equipment

Marketing Name	Part Number	Description
Console Handset Amplifier	9189-888-001-NA	Provides volume control for the SUPERCONSOLE 1000 handset.
SUPERSET Line Cord Pack	9170-048-004-NA	Spare Line Cord for SUPERSET telephones and for SUPERCONSOLE 1000 Attendant Consoles (quantity: 10)
SUPERSET 4001 Telephone	9132-001-100-NA	Single line digital telephone (light grey)
SUPERSET 4001 Telephone	9132-001-200-NA	Single line digital telephone (dark grey)
SUPERSET 4001 Telephones	9132-001-106-NA	6-pack Single line digital telephone (light grey)
SUPERSET 4001 Telephones	9132-001-206-NA	6-pack Single line digital telephone (dark grey)
SUPERSET 4015 Telephone	9132-015-200-NA	Multi-line digital telephone (dark grey)
SUPERSET 4025 Telephone	9132-025-100-NA	Non-backlit, multi-feature, multi-line digital telephone (light grey)
SUPERSET 4025 Telephone	9132-025-102-NA	Backlit, multi-feature, multi-line digital telephone (light grey)
SUPERSET 4025 Telephone	9132-025-200-NA	Non-backlit, multi-feature, multi-line digital telephone (dark grey)
SUPERSET 4025 Telephone	9132-025-202-NA	Backlit, multi-feature, multi-line digital telephone (dark grey)
SUPERSET 4125 Telephone	9132-125-102-NA	Backlit, multi-feature, multi-line digital telephone, attaches to a PC (light grey). Includes Mitel TAPI Service Provider software on a CD ROM, an AC power adapter, and RS232 cable. **NOTE:** The SX-200 ICP does not support TAPI.
SUPERSET 4125 Telephone	9132-125-202-NA	Backlit, multi-feature, multi-line digital telephone, attaches to a PC (dark grey). Includes Mitel TAPI Service Provider software on a CD ROM, an AC power adapter, and RS232 cable. **NOTE:** The SX-200 ICP does not support TAPI.
SUPERSET 4150 Telephone	9132-150-102-NA	Backlit, full-feature, multi-line digital telephone (light grey)

SUPERSET 4150 Telephone	9132-150-202-NA	Backlit, full-feature, multi-line digital telephone (dark grey)
SUPERSET 4150 Telephone with TAPI	9132-150-302-NA	

Backlit, full-feature, multi-line digital telephone (light grey) with Mitel TAPI Service Provider software on a CD ROM, an AC power adapter, and a RS232 cable.

NOTE: The SX-200 ICP does not support TAPI.	
SUPERSET 4150 Telephone with TAPI	9132-150-402-NA

Backlit, full-feature, multi-line digital telephone (dark grey) with Mitel TAPI Service Provider software on a CD ROM, an AC power adapter, and a RS232 cable

NOTE: The SX-200 ICP does not support TAPI.	
Mitel TAPI Service Provider CD-ROM	9132-000-202-NA

Required for TAPI enabled sets.

NOTE: The SX-200 ICP does not support TAPI.		
SUPERSET Interface Module 1 (SIM 1)	9132-250-100-NA	Installs inside a SUPERSET 4025, SUPERSET 4125, or SUPERSET 4150 telephone to connect to one or two PKM 48 devices or a PKM 12.
SUPERSET Interface Module 2 (SIM 2)	9132-250-101-NA	Installs inside a SUPERSET 4025, SUPERSET 4125 or SUPERSET 4150 telephone to connect to a PKM 12 or one or two PKM 48 devices, and an analog interface
Mitel Programmable Key Module 48 (PKM 48)	9132-200-100-NA 9132-200-200-NA	Light Grey Dark Grey Provides 48 additional feature keys to a SUPERSET 4025, SUPERSET 4125, or SUPERSET 4150 telephone
Power Adapter		

9132-800-210-NA

500000690

North America AC power adapter for PKM 48, SIM 2, or DSS/BLF Interface Module (DNIC sets only)

North American 24VAC Power Adapter for 5010 and 5020 IP Phones		
-48V Power Brick	50002070	Required by 5200 and 5300 Series IP Phones

PowerDsine 24 Port In-line Power Unit	PD-6024/AC	Power source option for 5200 and 5300 Series IP Phones
Short bracket (black) for PKM 48	9132-800-209-NA	Secures one PKM 48 to its telephone (sold in packages of 5)
Long bracket (black) for PKM 48	9132-800-299-NA	Secures two PKM 48 devices to a telephone
Mitel Programmable Key Module 12 (PKM 12)	50000110 50000111 50000112 50000113	Dark Grey PKM 12 (with SIM 1) Dark Grey PKM 12. Light Grey PKM 12 (with SIM 1) Light Grey
Cord Detangler	9132-800-001-NA	10 pack
Universal Handset (dark grey)	9132-800-200-NA	10 pack without cord
Universal Handset (light grey)	9132-800-201-NA	10 pack without cord
5485 IP Paging Unit	50001754	
Plantronics Headsets	Refer to Plantronics web site for available headsets	For use with Mitel 4000 and 5000 series telephones, except the 5207. See Headset Operation for information on headset requirements for the 5207.
Wall Mounting Kit	9132-800-601-NA	For use with SUPERSET 4025 and SUPERSET 4150 telephone
SUPERSET 401+ Telephone	9113-000-002-NA	Single line digital telephone
SUPERSET 410 Telephone	9114-000-000-NA	Multi-line digital telephone
SUPERSET 420 Telephone	9115-000-000-NA	Multi-feature, multi-line digital telephone
SUPERSET 430 Telephone	9116-000-000-NA	Full-feature, multi-line digital telephone
SUPERSET 401+ Telephone	9113-502-002-NA	Single line digital telephone (dark grey)
SUPERSET 410 Telephone	9114-502-000-NA	Multi-line digital telephone (dark grey)
SUPERSET 420 Telephone	9115-502-000-NA	Multi-feature, multi-line digital telephone (dark grey)
SUPERSET 430 Telephone	9116-502-000-NA	Full-feature, multi-line digital telephone (dark grey)
International Handset for SUPERSET 400 series	9115-007-001-NA	Spare Handset for SUPERSET 400 series telephones (quantity:10)
International Handset Cord for SUPERSET 400 series	9115-010-001-NA	Handset Cord for SUPERSET 400 series telephones (quantity:10)

Static Protection Unit	9180-067-001-NA	Protects system against static discharges at stations. Installed at distribution frame. One unit handles 25 stations.
Standalone DATASET 1103	9141-110-300-NA	Standalone asynchronous DATASET
Programmable Key Module	9112-200-000-NA	
DSS/BLF Interface Unit	9180-500-001-NA	
Standalone DATASET 2103	9141-210-300-NA	Standalone synchronous/asynchronous DATASET
DIGITAL LINE MONITOR	9132-800-600-NA	with AC adapter and cables
DNIC Music-on-Hold/Pager Unit	9401-000-024-NA	connects to DNIC line circuit
Desi Cards and Lens for IP Sets and PKMs:		
• 5201 Fixed Desi Cards (25 pack)	51004388 51004537 51004397	English French Spanish
• 5201 Lens (25 pack)	51004409	
• 5205 Fixed Desi Cards (25 Pack)	51004386	
• 5205 Lens (25 Pack)	51004406	
• 5215 / 5220 Removable Desi Cards (25 Pack)	51004387	
• 5215 Fixed Desi Cards (25 Pack)	51004390 51004539 51004399	English French Spanish
• 5215 Lens (25 Pack)	51004407	
• 5215 / 5220 Removable Desi Cards (25 Pack)	51004387	
• 5220 Fixed Desi Cards (25 Pack)	51004391 51004540 51004400	English French Spanish
• 5220 Lens (25 Pack)	51004408	
• 5412 IP PKM12 Desi Cards (25 Pack)	51005211	
• 5448 IP PKM48 Desi Cards (25 Pack)	51005215	
• 5412 IP PKM12 Lens (25 Pack)	51005219	
• 5448 IP PKM48 Lens (25 Pack)	51005223	

FRU Procedures

Only persons who have successfully completed a Mitel Installation and Maintenance training course for the SX-200 ICP and SX-200 EL PBXs should perform removal and replacement procedures.

WARNING: Instructions must be followed explicitly when they involve work with and changes to the primary power supply of the unit.

Precautions

Observe the following precautions when working on the system, particularly when handling PCB cards or using test equipment to measure voltages.

- When replacing PCB cards turn power off (when possible), but maintain the ground connections to the equipment (see Note below). Power must be OFF when inserting or removing common control cards. These cards are identified with appropriate warnings on their faceplates.
- Always wear an antistatic wrist strap when handling printed circuit cards. Handle PCB cards only by the edges and avoid contact with any exposed electrical connections. When removing a new card from its package, touch the package to the cabinet frame first to release any static voltage buildup, prior to removing the card and inserting it into the equipment.
- Conductive packages (antistatic packaging) should be grounded prior to opening them to remove the contents, and similarly grounded prior to placing a card in the package. Place suspected faulty cards in conductive packages to prevent further possible damage to the cards. Cards that are not correctly packed in antistatic packaging when returned will not be covered by any warranty.

Note: All bays must be unplugged from the ac mains during servicing. To reduce static susceptibility, always attach the wrist strap from the cabinet being serviced, and immediately place any item removed from a node into antistatic packaging.

Power Down System

The following chart describes system power down procedures.

Chart: Power Down System

Step	Action	Comments
1.	**Power Down the SX-200 ICP** Disconnect the power cord from the controller.	
2.	**Power Down the NSU (if installed)** Disconnect the power cord from the NSU	
3.	**Power Down Peripheral Cabinets (if installed)** Unlock and open door.	

	Turn off Bay Power Supply switch and remove cabinet line cord from the wall outlet.	The system is now properly powered down.

Power Up System

The following chart describes system power up procedures.

Chart: Power Up System

Step	Action	Comments
1.	Plug the SX-200 ICP controller line cord into the wall outlet	
2.	Plug the NSU (if installed) into the wall outlet	
3.		

Plug Peripheral cabinet (if installed) line cord into the wall outlet. Turn on Bay Power Supply switch.

Replace any covers or barriers that were removed previously. Close and lock the door.	The system is now properly powered up, and its door is closed and locked to prevent unauthorized access to equipment.

MX Controller FRUs

Replacing the Hard Drive or Internal CompactFlash Card

Use this procedure to replace faulty internal media. To upgrade the media in a working system, follow the procedure for installing a hard drive.

IMPORTANT: Use Mitel-provided CompactFlash cards and hard drives only; see Install FRUs for part number. Cards and drives obtained elsewhere are not supported.

To replace the CompactFlash card with another CompactFlash card

1. Remove the external CompactFlash card, if inserted.
2. Power down the controller.
3. Remove the cover.
4. If replacing the CompactFlash card, remove it, install the new one, and then skip to step 6.
5. If replacing the hard drive,
   a) Unplug the ribbon cable and the power cable from the old hard drive.
   b)- Remove the old hard drive by unfastening the four small border screws; see figure.
   c) Replace the drive with the new one.

   Note: Ensure that the jumpers on the new drive are set to the Master setting.

   d)- Connect the power and IDE cables to the corresponding connectors on the hard drive and main board. (The cables are keyed for proper connection.)

6. Replace the cover.
7. Power up the controller.
8. Perform an Initial (CompactFlash Card) installation from the software CD.

   Note: When prompted for a database, select Custom to install a backup of the original database (if available).

Replacing the Analog Options Board

To replace the Analog Options Board, follow the installation procedure; Installing the Analog Options Board.

Replacing a DSP Module

To replace the DSP Module, follow the installation procedure; see Installing a DSP Module.

Replacing the Stratum Clock Module

To replace the Stratum Clock Module, follow the installation procedure; see Installing the Stratum Clock Module.

Replacing a Dual FIM Module

To replace the Dual FIM Module, follow the installation procedure; see Installing a Dual FIM Module.

Replacing a Quad CIM Module

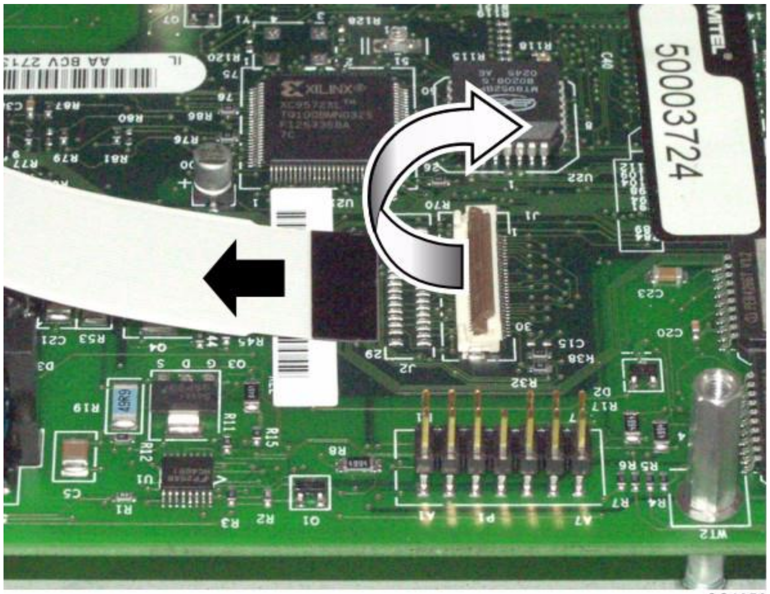
To replace the Quad CIM Module, follow the installation procedure; see Installing a Quad CIM Module.

Replacing a Dual T1/E1 Framer Module

To replace the Dual T1/E1 Framer Module, follow the installation procedure; see Installing a Dual T1/E1 Framer Module.

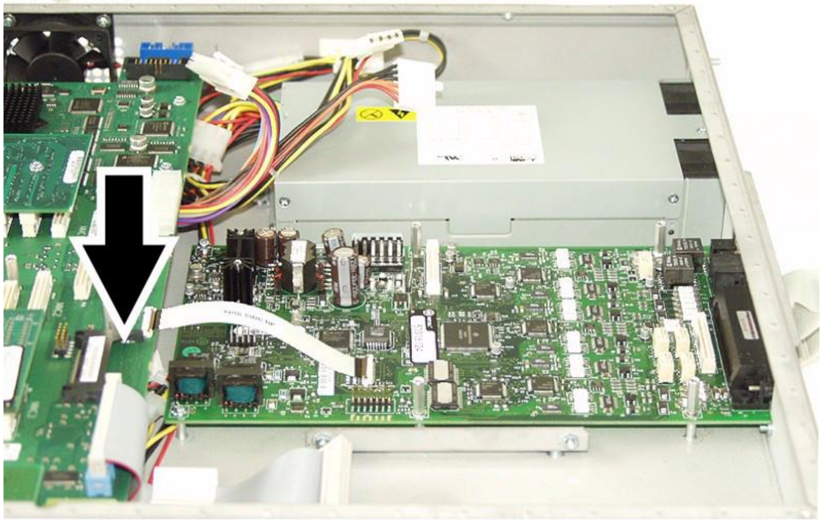
Replacing the Analog Main Board

1. Unplug the power cord from the controller.
2. Remove the cover of the controller.
3. Remove the Analog Option Board by following the installation procedure.
4. Disconnect the three Power Supply connectors.
5. Remove the ribbon cable by flipping up the clip on the connectors at each end of the cable as shown in the following figure.



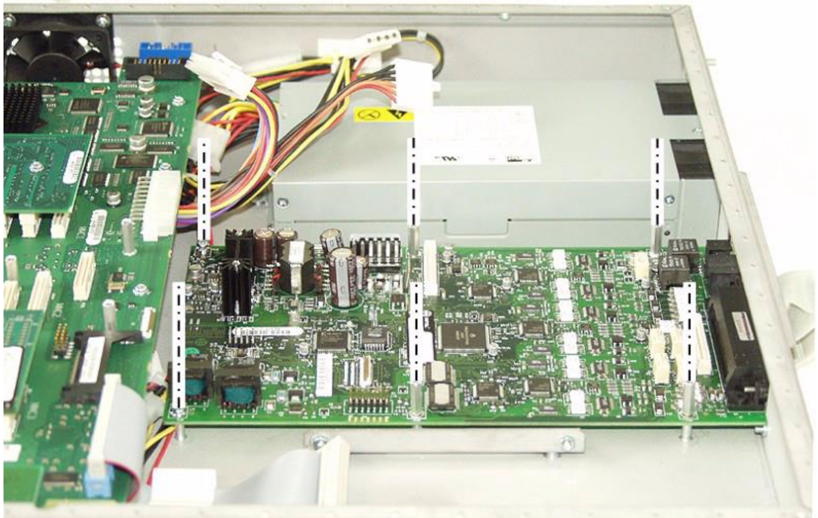
CC1058

6. Remove the internal flash card as shown in the following figure.



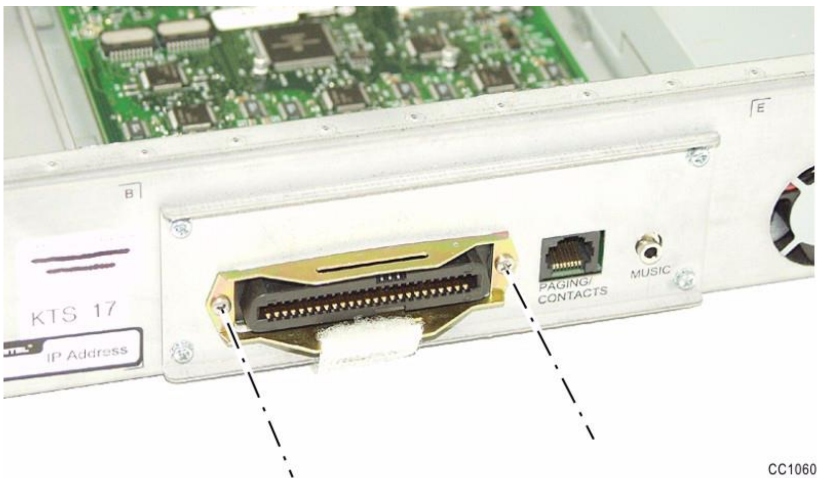
CC1057

7. Remove the four standoffs and two screws as shown in the following figure.



CC1059

8. Remove the two screws at the back of the unit as shown in the following figure.



CC1060

9. Remove the Analog Main Board.
10. Insert the new Analog Main Board.
11. Re-attach the two screws at the back of the unit.
12. Re-attach the two screws and the four standoffs.
13. Replace the flash card.
14. Replace the ribbon cable and snap the clips at both ends in place.
15. Replace the Power Supply connectors.
16. Snap the front panel back in place.
17. Replace the controller’s cover.

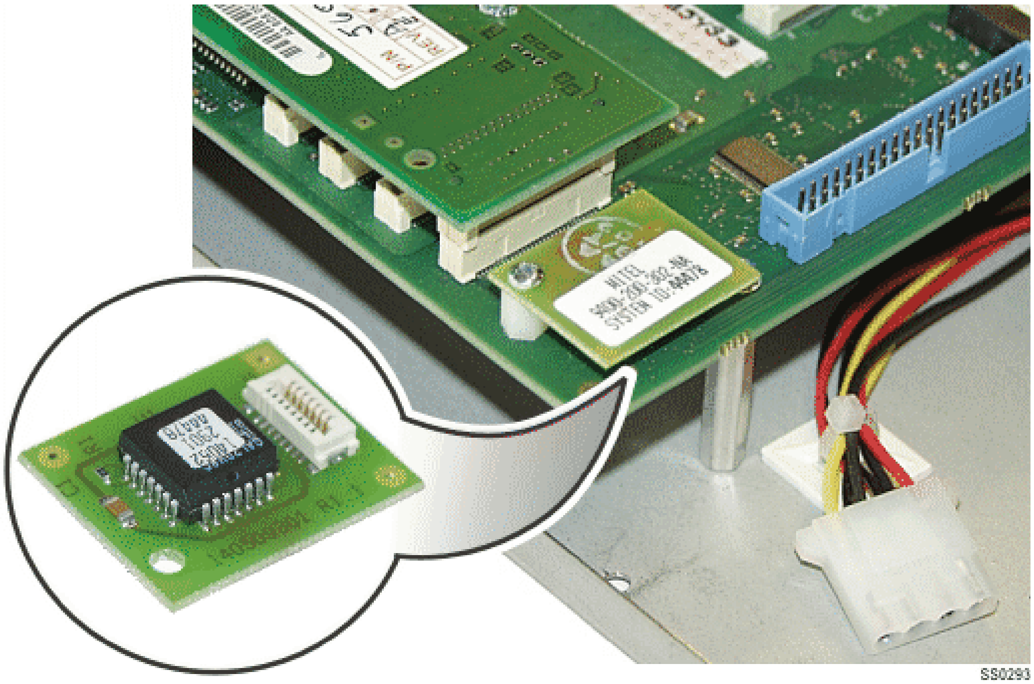
Replacing the System ID Module

In the event of a non-recoverable system failure, the System ID Module can be moved to a new controller along with the internal media (CompactFlash card or hard drive). The new controller can then be started with the same settings, database, and options as the original system.

Note: Prior to starting this procedure, remove the internal media and System ID Module from the controller that has experienced a non-recoverable failure.

To replace a System ID Module:

1. Unplug the power cord from the controller.
2. Remove the cover of the controller.
3. Replace the hard drive or internal CompactFlash card.
4. Insert the System ID Module on the main board connector.



5. Attach the screw.
6. Replace the cover.

SX-200 Peripheral Cabinet FRUs

Replace Printed Circuit Cards

The following chart describes circuit card replacement procedures.

CAUTION: Power must be off when you are removing the Bay Control Card and Bay Power Supply.
Cards that are not correctly packed in antistatic packaging when returned will not be covered by any warranty.

Chart: Replace Printed Circuit Cards

Step	Action	Comments
	Removing Cards	
1.	When removing cards from an operating system, turn power off, if possible. Power must be off when you are removing a Bay Control Card or Bay Power Supply. To remove a Bay Power Supply, refer to Chart: Replace Bay Power Supply.	**CAUTION:** Cards that must not be removed while the system power is on carry a Caution notice. These cards are: Main Control Cards, Bay Power Supply, Bay Control Card or its attached cards.

Chart: Replace Printed Circuit Cards

Step	Action	Comments
2.	Make sure the PBX ground is connected (see Note at the end of this table).	
3.	Put on the antistatic wrist strap when removing and repacking cards.	The antistatic wrist strap must be connected to the PBX chassis, which must be connected to an approved ground to provide protection from static discharges.
4.	Remove the card by using the extractor as a lever and pulling the card towards you.	Each digital peripheral card has one card extractor. The extractor helps seat the card firmly in the backplane. The extractor provides leverage to pull the card free of the backplane connector.
	Install New Card	
5.	Unpack the replacement card. If there are modules or jumpers on the original card, transfer them to the replacement card. Ensure that any switches on the replacement card are in the same position as the original card.	Retain the packaging material for the repacking of the original card for return.
6.	Install the replacement card into the card slot.	
	Repacking Cards	Packaging is shown in this Figure.
7.	Handle printed circuit cards by their edges only, except when seating connectors.	Handling the card faces or components may cause damage.
8.	Do not touch the gold edge connectors.	
9.	Avoid contact with any exposed electrical connections.	
10.	Use original or similar packaging material kept after unpacking.	
11.	Ground the antistatic packaging before putting a card in it.	
12.	As soon as you remove a card from a slot, place it in antistatic packaging.	Place suspected faulty cards into antistatic packaging to prevent further possible damage.
13.	When you are finished, replace the antistatic wrist strap in the cabinet.	
14.	If you have powered down the system, power it up again.	

Note: All bays must be unplugged from the ac mains during servicing. To reduce static susceptibility, always attach the wrist strap from the cabinet being serviced, and immediately place any item removed into antistatic packaging.

Figure: Printed Circuit Card Packaging

Replace a Bay Control Card

The following chart describes Bay Control Card (BCC) replacement.

CAUTION: Turn off power before removing the Bay Control Card.

Chart: Replace a Bay Control Card

Step	Action	Comments
1.	Follow general procedures for handling circuit cards	Given in Chart: Replace Printed Circuit Cards.
	Turn off power before removing a Bay Control Card.	
	Remove Original BCC	
2.	Attach the antistatic wrist strap and remove original BCC and any card attached to it. Disconnect the attached card.	Refer to replacement procedures for cards that are attached to the BCC.
	Unpack and Inspect New BCC	
3.	Wearing the antistatic wrist strap, unpack and inspect the BCC for damage.	Retain packaging for shipment of original unit.
	Install New BCC	
4.	Reattach the card to the BCC and slide the BCC assembly into its slot.	

Replace Bay Power Supply

The following chart describes Bay Power Supply (BPS) replacement.

Note: The BPS is an ac-to-dc convertor. The BPS faceplate is labeled “BAY PSU”. It has a ringing voltage generator used by the card slots it supports.

CAUTION: Turn off power before removing the Bay Power Supply.

Chart: Replace Bay Power Supply

Step	Action	Comments
1.	Follow general procedures for handling circuit cards.	Given in Chart: Replace Printed Circuit Cards
	Turn off power before removing a Bay Power Supply.	
	Remove Original BPS	
2.	Turn off the unit, and unplug power cord.	
3.	Loosen the thumbscrew at the upper front of the BPS to release it from the cardfile.	Remove the cover plate on the back of the node and unplug the internal AC power cord. To remove the cover plate, unscrew the screws and place them nearby.

Chart: Replace Bay Power Supply		
Step	**Action**	**Comments**
4.	Remove the BPS from its slot.	
	Unpack and Inspect New BPS	
5.	Wearing the antistatic wrist strap, unpack and inspect the BPS.	Retain packaging for shipment of original unit.
6.	Repack and return the original unit.	
	Install New BPS	
7.	Insert the BPS into its slot, and tighten the thumbscrew at the upper front of the BPS to secure it to the cardfile.	
8.	Make sure that the BPS I/O (on/off) switch is in the O (off) position.	
	Connect Power Cable	
9.	Plug the internal AC power cord into the back of the Bay Power Supply, directly under the access cutout.	
10.	Replace the access cover plate back over the cutout and replace the screws.	Power up the system (see Chart: Power Up System).

Figure: Bay Power Supply

Replace FIM Carrier or FIM

The Replace FIM Carrier or FIM Chart describes the following FIM Carrier and FIM replacement procedures.

- Peripheral FIM Carrier II and FIM in an SX-200 RM Peripheral cabinet.
- Peripheral FIM Carrier and FIM in an SX-200 LIGHT Peripheral Cabinet.

Chart: Replace FIM Carrier or FIM		
Step	**Action**	**Comments**
1.	Follow the general procedures for handling circuit cards.	
	Remove the FIM Carrier and FIM	
2.	Attach the antistatic wrist strap, and partially remove the FIM Carrier card.	For a Peripheral FIM Carrier attached to the BCC, partially remove the BCC.
3.	Identify and disconnect the Rx and the Tx fiber cables from the FIM.	
	Peripheral FIM Carrier II: Remove the Peripheral FIM Carrier II from its slot and place it on antistatic packaging.	Refer to this Figure.
	Peripheral FIM Carrier: Remove the BCC and	Refer to this Figure. Also place the BCC

Chart: Replace FIM Carrier or FIM

Step	Action	Comments
	Peripheral FIM Carrier, separate the cards, and place the Peripheral FIM Carrier on antistatic packaging.	into antistatic packaging.
	Remove Fiber Interface Module	
4.	Unfasten the FIM faceplate from the FIM carrier by removing the 2 screws.	
5.	Slide the FIM to disconnect it from the DIN connector.	
	Unpack and Inspect the Replacement FIM Carrier or FIM	
6.	Wearing the antistatic wrist strap, unpack and inspect the FIM Carrier or FIM for damage.	Retain packaging for returning original unit.
	Install FIM Onto FIM Carrier	
7.	Slide the FIM into the DIN connector on the FIM Carrier until it connects firmly.	
8.	Fasten the FIM faceplate to the FIM Carrier with 2 screws.	
	Install FIM Carrier and FIM Into Cabinet	
9.	Install the FIM Carrier and its FIM into the assigned slot.	
	Peripheral FIM Carrier II: Slide the Peripheral FIM Carrier II into its slot.	Refer to this Figure.
	Peripheral FIM Carrier: Attach a FIM Carrier to its BCC, and slide the pair into the BCC slot.	Refer to this Figure.
	Connect Fiber Interface Cable to the FIM	
10.	Remove the protective caps from the cable connectors on the cables, and from the connectors on the FIM faceplate.	
11.	Connect optical fiber cables to the positions from which they were removed.	

Figure: Peripheral FIM Carrier II and FIM

Figure: Peripheral FIM Carrier and FIM

Replace the Fan Assembly

The following chart describes replacement of the cabinet fan assembly.

Chart: Remove and Replace the Fan Assembly

Step	Action Comments
Remove Fan Assembly	
1.	Remove the two screws holding the fan assembly. The cabinet may be left powered on.
2.	Support the fan assembly and remove it from the cabinet. Retain screws.
3.	Disconnect the fan power connector.
Replace Fan Assembly	
4.	Unpack and inspect new fan assembly.
5.	Connect the cabinet fan power connector to the new fan assembly.
6.	Replace the power cable within cabinet and attach the fan assembly using removed screws.

Replace DNIC Music-on-Hold/Pager Unit

The following chart describes how to remove and replace the DNIC Music-on-Hold/Pager Unit. This procedure only interrupts system functions which are directly associated with the Music-on-Hold/Pager Unit (night bells, alarms, paging, and Music-on-Hold). The rest of the system continues to function.

Chart: Replace the Music-on-Hold/Pager Unit

Step	Action	Comments
	Prepare the Music-on-Hold/Pager Unit	
1.	Power down the external equipment including paging amplifiers, night bells, alarms etc.	
	Replace the unit	
2.	Disconnect the 25 pair Amphenol connector from the unit.	Use a small screwdriver to detach the connector locking screw.
3.	Replace the unit with a new one.	If the unit is wall mounted, loosen the mounting screws as necessary.
4.	Reattach the 25 pair Amphenol connector. The Status LED will light.	
5.	Reapply power to the external equipment. When the pager is in use, the status LED will wink.	

CX/CXi Controller FRUs

Replacing the Hard Drive or Internal CompactFlash Card

Use this procedure to replace faulty internal media. To upgrade the media in a working system, follow the procedure for installing a hard drive.

IMPORTANT: Use Mitel-provided CompactFlash cards and hard drives only (see Install FRUs for part number). Cards and drives obtained elsewhere are not supported.

To replace the hard drive or internal CompactFlash

1. Remove the external CompactFlash card, if inserted.
2. Power down the controller.
3. Remove the cover.
4. If replacing the CompactFlash card, remove it, install the new one, and then skip to step 6.
5. If replacing the hard drive
    a) unplug the ribbon cable and the power cable from the old hard drive.
    b) remove the screws connecting the bracket to the back of the controller, then slide the bracket forward and remove it. (Removal is unnecessary if the drive is installed in the upper bracket position.)
    c) replace the drive with the new one and secure it to the bracket. (If the bracket was removed, re-install it and secure it to the back of the controller). Ensure that the jumpers on the new drive are set to the Master setting.
    d) connect the ribbon cable and power cable to the new hard drive.
6. Replace the cover.
7. Power up the controller.
8. If replacing the system hard drive or the CompactFlash card (NOT the APC hard drive), perform an Initial (CompactFlash Card) installation from the software CD.

    Note: When prompted for a database, select Custom to install a backup of the original database (if available).

Replacing the Analog Option Board

To replace the Analog Option Board, follow the installation procedure; Installing the Analog Options Board.

Replacing the Stratum Clock Module

To replace the Analog Options Board, follow the installation procedure; see Installing the Stratum Clock Module.

Replacing the APC

To replace the Application Processor Card, follow the installation procedure. See Installing the APC.

Replacing a DSP Module

To replace the DSP Module, follow the installation procedure; see Installing a DSP Module.

Replacing the T1/E1 Combo Module

To replace the T1/E1 Combo Module, follow the installation procedure; see Installing a T1/E1 Combo Module.

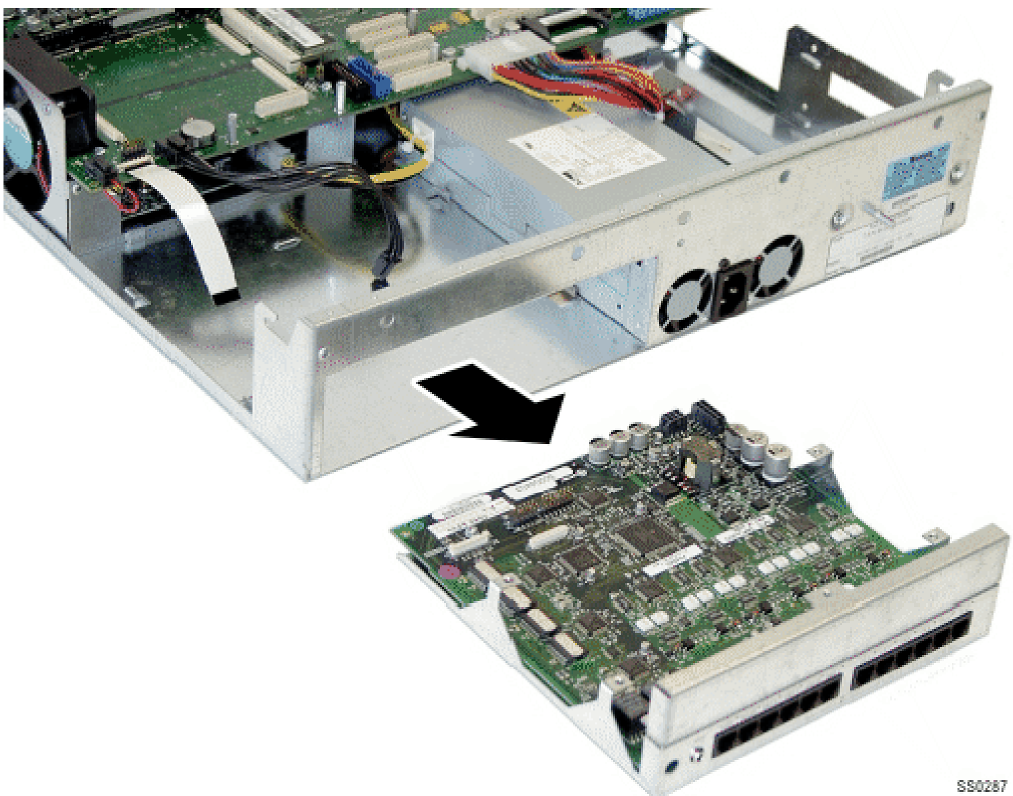
Replacing the Quad Copper Interface Module

To replace the Quad CIM, follow the installation procedure; see Installing a Quad CIM Module.

Replacing the Analog Main Board (AMB)

Note: If installing version 2 of the AMB/AOB, upgrade the controller software to release 3.0 or greater. The AMB and AOB must be the same version.

1. Unplug the power cord from the controller.
2. Remove the cover.
3. Remove the screw that secures the faceplate to the back panel of the controller.
4. Disconnect the cables on the Analog Option Board (if installed) as follows:
   - loosen the flex cable connector by pulling up on the tabs at ends of the connector, and then remove the cable.
   - disconnect the power supply cable by squeezing the retaining clip on the side of the connector and pulling upwards.
5. Disconnect the flex cable on the AMB by flipping up the clamp on the connector.
6. Disconnect the power supply cable on the AMB.
7. Slide the AMB out.



8. Insert the new AMB. **Note**: AMB v3 has an additional connector on the back of the board for future use. Ignore this connector when replacing the AMB in a CX system.
9. Replace the flex cable and secure it by pressing down on the tabs on the end of the connector.
10. Replace the power supply cable.
11. Replace the Analog Option Board (if previously removed).
12. Replace faceplate screw.
13. Replace the top cover.

Replacing the System i-Button

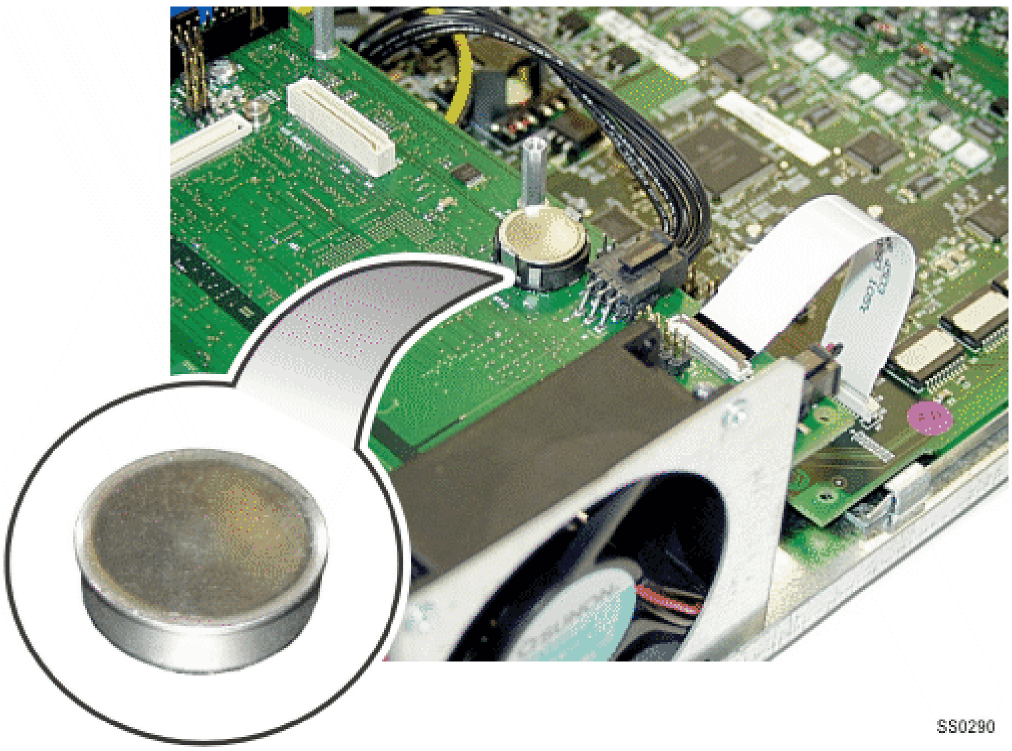
In the event of a non-recoverable system failure, the System i-Button can be moved to a new controller along with the internal media (CompactFlash card or hard drive). The new controller can then be started with the same settings, database, and options as the original system.

Note: Prior to starting this procedure, remove the internal media and System ID Module from the controller that has experienced a non-recoverable failure.

To replace a System i-Button:

1. Unplug the power cord from the controller.
2. Remove the cover of the controller.
3. Replace the hard drive or internal CompactFlash card.
4. Insert the System i-Button in the twin tab connector located on the main board.

Note: Make sure that written side of the i-Button faces down.



SS0290

5. Replace the cover.

AX Controller FRUs

For information on replacing AX Controller FRUs, refer the the *SX-200 CX/CXi and AX Technician's Handbook*.

Repack Equipment for Shipment

Complete the Repack Equipment For Shipment Chart to properly repack equipment for shipment.

Notes:

1. Do not disconnect the system ground until printed circuit cards have been properly packed. The system must be properly grounded for the wrist strap to be effective.

2. Whenever possible, re-use original packaging to repack equipment for shipment.

Chart: Repack Equipment For Shipment

Step	Action	Comments
1.	Pack all circuit cards in antistatic packaging and PC card shipping cartons.	Wear the antistatic wrist strap while handling circuit cards.
2.	Handle cards by the edges only, and follow the card handling procedures.	
3.	Wrap all items with air-cushion type material, and surround with loose paper to minimize movement within the carton.	Ensure that items within the carton cannot shift about, and will not get scratched or damaged.
4.	Repack all items carefully, and list the contents of each carton on the label.	

Program Features

2Feature Levels

The Features Options Records (FOR) option, Feature Level, provides controlled access to selected features in new software releases. Customers purchasing new systems and software upgrades will receive this option. Software received for bug fixes will NOT receive this option. The Feature Level allows our customers to access selected features in new software releases. The Feature Level option (option 102 in CDE Form 04) will be incremented for major releases of software. Feature Levels were introduced on the SX-200 EL/ML beginning with LIGHTWARE 18, Release 1.0 which was Level 1. Release 2.0 was Level 2. LIGHTWARE 19 Release 2.0 was Level 3 and Release 3.0 was 4.

SX-200 ICP, Release 1.x was Level 5 and Release 2.0 is Level 6.

The following table lists the software features that are dependant on the Feature Level option.

Software Features Dependant On Feature Levels
Feature Level 1
• Call Blocking - Subattendant
• Call Forwarding - Multiple Destinations • Call Logging • Direct Connection of PKMs to Superconsole 1000 or 5540 IP Console • Emergency Calls (911) - Detection and Reporting to Display Sets • Privacy Release - Default Setting • Wakeups - Multiple and Personal
Feature Level 2
• BRI Card Support
• Internal Number Block • ONS Positive Disconnect • Remote Printing of CDE Forms

Feature Level 3

- Single Button Transfer to Voice mail
- Tenant-based Call Forwarding
- Display Caller ID on Non-Prime Lines
- Paging for PA and Telephones
- ONS Ring Groups
- Disable Conference Beep
- Display Ringing Extension within a Pickup Group
- Sets/Consoles Paging Other Paging Groups

Feature Level 4

- ARS suppport of interchangeable area & office codes
- Auto programming of stations/sets in CDE.
- Direct ARS access for residential users
- E911/lockout notification to ONS (class) sets
- ONS as music source
- Range delete of station/set enhancement
- Range programming of trunks
- Retention of 2-party call from conference on DTS/CO line keys

Feature Level 5

- Call Park - Destination Phone
- Call Park System - Specific Orbit
- Distinctive Ring Tones
- Intercom Calls
- Park and Page
- DTS/CO Line Transfer Call Handling

Feature Level 6

- Call Monitor from
- Call Park Orbit Feature Keys
- Do Not Disturb - Flash Key when Active
- Campon Priority Over Call Forward Busy for trunk calls
- Hold and Page
- Mailbox Key
- Extension Night/Day Switching
- Time and Date Setup

Notice for Mitel Telephone SUPERKEY Operations

On a Mitel 5010, 5207, and 5215 IP Phone, or SUPERSET 4015 telephone, press the dial pad * key instead of the YES softkey and press the dial pad # key instead of the NO softkey when performing SUPERKEY operations.

On a Mitel 5212, Dual Mode 5215, Dual Mode 5220, 5224, 5330, and 5340, press the dial pad # key instead of the YES softkey and press the dial pad * key instead of the NO softkey when performing SUPERKEY operations.

Abbreviated Dial

Description

The Abbreviated Dial feature allows trunks and extensions to be accessed by dialing a 2 to 8-digit number which is then translated by the system into the actual number. The actual number can contain up to 26 digits.

The Abbreviated Dial feature can also give system-wide access to a defined set of long distance numbers, while denying general access to long-distance dialing.

Abbreviated dial numbers can also be used as dial-in trunk prefixes, as routing points for automatic call distribution (ACD) interflow, automated attendant, and as call-forwarding points.

The attendant or customer data entry (CDE) programmer can program or display system abbreviated dial numbers. These numbers can be marked as confidential to prevent them from appearing on display telephones or in SMDR reports.

Special codes can be embedded within the abbreviated dial number for different features. The following codes can be inserted into a stored number:

*3= Wait for user to manually insert digits (2 digits)
** = DTMF digit *
# = DTMF digit #
*9= 1-second pause

Example: For example, a typical number for external directory assistance is 9+1+(area code)+5551212; the area code is to be dialed manually. The number to be stored would be 91*3035551212.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Tie, DID, and DISA trunks, extensions, data devices, and consoles can access abbreviated dial numbers directly.
- The user should release or hang-up the call before the destination answers, to allow the DTMF tones to go through. If the user is slow in releasing or hanging-up the call, thereby allowing the destination to answer before the user has released or hung up, then the DTMF tones shall be lost.
- Calls can be forwarded to abbreviated dial numbers.
- No toll control is done for external calls using abbreviated dial.
- Through the Automated Attendant feature, all devices can directly dial abbreviated dial numbers.
- Confidential numbers can only be viewed from CDE or by consoles with COS Option 110 (Attendant Abbreviated Dial Confidential Number Display) enabled.
- Access to abbreviated dial numbers is available to: interflow points in ACD paths, and automated attendant default destinations.
- Abbreviated dial numbers dialed after automatic route selection (ARS) leading digits will be interpreted as ARS digits to be outpulsed rather than as an abbreviated dial number.

- If an abbreviated dial number is to appear in another speed dial number (Abbreviated Dial or Speedcall), then the abbreviated dial number index must be three digits in length (leading zeros required).

The following conditions apply to inserting pauses in speed calls:

- Pauses are not supported for digit strings that are outpulsed over rotary dial trunks.
- Inserting a pause converts all subsequent dialed digits to DTMF format.
- Each pause is 1 second in length.
- Each inserted pause occupies 2 digit positions of the speed dial number.
- Pauses can also be programmed in personal speed dial numbers.
- Pauses do not appear in SMDR records and any digits dialed after a pause do not appear in SMDR records.
- Host Command Interface (HCI) does not display pauses or any digits that are dialed after a pause.
- Speed dial digits used during talking state do not appear in SMDR records.
- The system inserts a pause after internal dialed extension number digits to allow for answer and establishment of a stable connection before sending additional digits. Insert the total number of required pauses.
- Abbreviated dial numbers can be used to program reminders or wakeup calls. The access code, room number, timer number (for example #1 for timer 1), the repeated wakeup/reminder character (an asterisk specifies a daily wakeup/reminder at the same time), and the four digit time definition (24-hour) can be entered in Form 31 and then used for an abbreviated dial. See Reminder for more details on setting reminders using the access code.

Programming

Enable COS Option 245 (Abbreviated Dialing Access) for non-console devices to access abbreviated dial directly. Consoles do not need this option enabled to access abbreviated dial directly.

Enter the desired abbreviated dial index numbers, and the digit strings to be dialed, into CDE Form 31 (System Abbreviated Dial Entry). To insert pauses in a numbers, enter ∗9 for each 1-second.

You can also enter the index numbers and digit strings from the attendant console. See Attendant Abbreviated Dial Number Entry.

Assign an access code to Feature 24 (Abbreviated Dial Access) in CDE Form 02 (Feature Access Codes).

Operation

To dial an abbreviated dial number, when dial tone is heard:

- Dial the Abbreviated Dial access code
- Dial the desired Abbreviated Dial index number (one to three digits).

Account Codes

Description

Account codes are typically used to charge the cost of outgoing trunk calls to departmental cost centers or project accounts. The account code can be optional or mandatory, and appears on all station message detail recording (SMDR) records. An account code can apply to both incoming and outgoing trunk calls.

Account codes can range from 2 to 12 digits.

For verified account codes, see Account Codes - Verified in this section.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- If verified account codes are used, account codes can be two to 12 digits in length.
- If the account code is of the variable length type, the account code digits must be followed by a "#", except where the entered code length is 12 digits.
- Rotary-dial type telephones always have a digit length of six, as they can not dial the # character.
- Users of Family 1, Family 4, and Family 5 telephones can also enter account code(s) during a call.
- A "*" or a "#" is not allowed in an account code.
- A "#" is only allowed to be dialed if the account codes are of the variable length type.
- If no SMDR buffers are available, the extension user receives busy tone when the account code entry is attempted.
- If you enable forced account codes for industry-standard telephones that also have the Direct To ARS feature enabled, users must dial an account code before they can originate a call.
- The Direct To ARS feature applies to Mitel telephones when a valid account code has been entered.
- Extension users who have Forced Account Codes enabled in their COS are not forced to enter an account code for an external call unless the SMDR feature is programmed to record the call and the account code.
- For Family 1 , Family 4, and Family 5 telephones, an account code can only be entered during a call if the call is a 2-party call, the other party being a trunk (of any type). There is one exception. An account code can also be entered by an ACD agent on such a call, who is also being silent monitored by a supervisor.
- For Family 1 , Family 4 , and Family 5 telephones, the **Account Code** softkey is present, providing that a consultation hold is not in progress, the set has completed dialing, and SMDR is enabled.

  In addition to station and Mitel telephones and consoles, DISA, Tie and DID trunks can also access account codes. If an error is made when entering a verified account code, the trunk is dropped.
- Account Code access for DID trunks is available.

Programming

To force an extension user, Tie, DID or DISA trunk to use account codes for long distance calls, enable COS Option 201 (Account Code, Forced Entry - Long Distance Calls) for the device. Refer to the Automatic Route Selection and Toll Control section, for the designation of Long Distance calls.

To force an extension user, Tie, DID or DISA trunk to use account codes for all external calls, enable COS Option 200 (Account Code, Forced Entry - External Calls) for the device's class of service.

Set Account Code Length (System Option 55) in CDE Form 04 (System Options/System Timers).

Assign a feature access code to Feature 01, Account Code Access in CDE Form 02 (Feature Access Codes).

COS Option 700 (SMDR - Does Not Apply) must be disabled in the extension, or Tie, or DISA trunk COS before the extension may use account codes.

COS Option 806 (SMDR - Record Incoming Calls) must be enabled in an incoming trunk COS to permit account codes to be used on incoming calls on the trunk to Family 1 , Family 4, and Family 5 telephones.

Enable all of required SMDR options as outlined in the Station Message Detail Recording section.

Enable System Option 36 (End Of Dial Character) for use with variable length account codes.

To provide feature key activation for account codes on Family 2 and Family 3, program an ACCOUNT CODE feature key. (See Feature Keys.)

Operation

Operation varies depending upon the type of device as described below.

At the completion of the call, an SMDR record is printed. This printout includes the time of call, trunk used, duration of call, and the account code.

Family 2 and Industry-Standard Telephones:

To access a trunk via account code entry:

- Dial the access code for account code entry.
- Dial the account code digits (if System Option 55 - Account Code Length is selected, account code digits must be followed by a “#” to indicate end of account code).
- Dial tone is returned. The account code entered now applies to the trunk call immediately made while still listening to dial tone.
- Dial the trunk call.

OR (Family 2 telephones only)

- During a call, press the ACCOUNT CODE softkey. The LCD second line shows ENTER ACCOUNT CODE:
- Enter the account code. Press the ← softkey to erase incorrect digits.
- Press SAVE softkey.

At the completion of the call, an SMDR record is printed. This printout includes the time of call, trunk used, duration of call, and the account code.

Family 3, SUPERSET 410, and SUPERSET 3DN Telephones:

To access a trunk via account code entry:

- Press the ACCOUNT CODE feature key.
- Dial the account code digits (if System Option 55 - Account Code Length is selected, account code digits must be followed by a “#” to indicate end of account code).
- Dial tone is returned. The account code entered now applies to the trunk call immediately made while still listening to dial tone.
- Dial the trunk call.

At the completion of the call, an SMDR record is printed. This printout includes the time of call, trunk used, duration of call, and the account code.

Family 1 , Family 4 , and Family 5 Telephones:

To access a trunk via account code entry:

- Dial the access code for account code entry.
- Dial the account code digits (if System Option 55 - Account Code Length is selected, account code digits must be followed by a “#” to indicate end of account code).
- Dial tone is returned. The account code entered now applies to the trunk call immediately made while still listening to dial tone.
- Dial the trunk call.

OR

- During a call, press SUPERKEY key.
- Press the NO softkey until ACCOUNT CODE? appears in the display.

- Press the YES softkey.
- Enter the account code. Press the ← softkey to erase incorrect digits.
- Press the SAVE softkey.

Family 5 - 5340 Phones:

- During a call, press the key you have programmed as Superkey.
- Press the More softkey until Acct Code Verified appears.
- Press Acct Code Verified.
- Enter the account code. Press the ← softkey to erase incorrect digits.
- Press Save.

Account Codes - Display on Phones

Description

The Display Account Codes on Phone option determines whether account code digits are displayed or replaced with asterisks on the telephone display.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

Account codes will be displayed in SMDR.

Programming

To ensure account codes are not displayed on phones, disable COS Option 279, Display Account Codes on Phone, in Form 3, COS Options.

Operation

None.

Access Codes - Global Find

Description

The Access Codes feature allows the user to view all access codes in the system. The system reports the type of device associated with the access code and its location (Bay/Slot/Circuit). The user can also query the system about a particular access code. Data in Customer Data Entry Form 35 (Global Find Access Code), is generated by the system and cannot be modified.

Conditions

None.

Programming

None.

Operation

Refer to the Program CDE section.

Account Codes - Verified

Description

The Account Codes - Verified feature helps to ensure accuracy for accounting purposes, and helps to prevent fraudulent use of Direct Inward System Access (DISA) lines and outgoing trunks. Verified account codes control access to trunks and external (DISA) access to the system by checking the dialed account code against a list of preprogrammed codes.

The caller's COS and COR can be changed (traveling class of service) when a valid account code is entered. This can give the caller access to different features and external call privileges. If the caller's COS is changed and the direct to ARS feature is enabled in the COS, the direct to ARS feature operates immediately.

Each verified account code has an active/inactive status. This allows accounts to be denied access when problems are encountered such as nonpayment of billings.

See Account Codes Verified (Special DISA), Resale Package, Trunk Operation (DISA). See also Analog Networking for the use of Verified Account Codes in networking.

Feature Availability

Click here to see a list of the phones that can use this feature..

Conditions

The following conditions apply to this feature:

- When there is a COR and COS associated with an account code, the COS and COR apply to the device for the duration of the call. Once the device has completed the call (goes on–hook), the original COS and COR are restored.
- The match for account code is attempted when the account code is completely entered. For fixed length account codes this is the account code length and for variable length account codes this is when the "#" is entered.
- The caller is given reorder tone if there is no exact match for the account code entered.
- By default, all account codes are active.

Programming

Enable System Option 05 (Verified Account Codes).

Select the number of account code digits (VARIABLE or 2 - 12 digits) via (System Option 55) (Account Code Length).

Enter the Verified Account Codes, COS, and COR into CDE Form 33 (Account Code Entry) as required.

Activate or deactivate Verified Account Codes in CDE Form 33 (Account Code Entry) as required.

Operation

See Account Codes.

Account Codes - Verified (Special DISA)

Description

Verified Account Codes can be used to replace the Direct Inward System Access (DISA) code. A caller who accesses a Special DISA trunk must dial an account code rather than the DISA code. By using a Verified Account Code, each DISA trunk can have access to its own COS options through the COS and COR associated with the account code. SMDR records each of these calls.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- There is no minimum digits dialed before the trunk is given re-order tone as is the case with regular DISA. The rules for dialing invalid verified account codes apply.
- The use of a verified account code rather than the DISA code for entry in to the system applies on an individual trunk basis. DISA trunks not selected to use the verified account code still use the DISA code.

Programming

See programming under Trunk Operation - Direct Inward System Access (DISA) and Account Codes - Verified.

Enable COS Option 808 (Special DISA) for the DISA trunk(s).

Operation

To access the feature from an external line:

- Access the system on a specified DISA trunk - ringback is heard, followed by dial tone.
- Dial the DISA Account Code - if the system verifies the account code, dial tone is returned; if not, the trunk is dropped.
- Dial the required number.

Add Held

Description

Add Held allows a user engaged in an active call on a Family 1 or Family 4 telephone to add a call that is on hold on another line to the current line.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- This feature is applicable to above telephones only.
- The set must have, in addition to its prime line, an appearance of another line.

- A private trunk line cannot be added into an established call.
- The feature can be used to add parties into a conference call or to transfer a call in progress to another line appearance.
- A conference on hold on a line cannot be added.

The party doing the add held must not have a consultation hold in progress and must not be talking to a third party.

- The console cannot be involved in the affected call.
- The line being added to can be a private trunk line only if dialing on the line (there can be no established call on the line).
- There must be less than five parties in the current call.
- The current call must not be on consultation hold.
- There must be no other parties in the current call performing an add held at the same time.

Programming

Enable COS Option 302 (Flash-in Conference) in the set's COS to allow it to add a held line while talking to another party.

Operation

Family 4 Telephones:

- Establish a call, with a call on hold on another line.
- Press the Addheld softkey. SELECT HELD LINE appears in the display.
- Press the flashing line key of the call on hold. The call on hold is added to your current call.

Family 1 Telephones:

- Establish a call, with a call on hold on another line.
- Press the **Add Held** softkey to add a call on a previously held line to the current call. The LCD display prompts the user to press the line select key associated with the call on hold, and the call on the line is added to the current call.

Analog Networking

Description

Analog Networking allows an SX-200 ICP system to send and receive caller information over a private network. Analog Networking uses the ARS Modified Digit feature to insert feature access codes and other codes (called information elements) into the outgoing digit string (refer to information elements in the Information Elements for Analog Networking Table. The information elements consist of special codes for inserting the caller's extension number, account code, and node identification. A glossary of analog networking terms can be found in the Glossary of Analog Networking Terms Table.

An SX-200 ICP system with Analog Networking can function as an end node, an intermediate node, or a hub. As an end node, network elements are transmitted into the network via DTMF digits. As an intermediate node, all information elements received by the intermediate node are passed on to the next node. As a hub, an SX-200 ICP system receives information elements to provide Calling Party Identification.

The information elements are:

Caller's Extension Number: Consoles and Mitel display telephones show the caller's extension number when a user answers a call. The caller's extension number replaces the trunk number or trunk name in the display, and the trunk number in SMDR records.

Caller's Dialed Account Code: The existing account code access code identifies the beginning of a caller's dialed account code. The account code is written into the SMDR record associated with the call. Verified account codes can be used to implement Traveling Class Marks, by providing a COS and COR with the account code, which, at the destination node, replaces the COS and COR associated with the trunk being used. The destination node processes the call using the COS and COR associated with the caller's verified account code that was passed on the trunk.

PBX Node ID: This information element is associated only with the originating node and must be imbedded with another information element because it has no access code of its own (usually it is the first digit of the extension number).

Conditions

The following conditions apply to this feature:

- In CDE Form 03 (Class of Service), enable COS Option 613 (Display ANI Information Only) in the COS of the receiving devices.
- SMDR for outgoing calls must be enabled for the account code to be sent to the destination. Refer to the Station Message Detail Recording section.
- This feature is compatible with that on the SX-2000 INTEGRATED COMMUNICATIONS System.
- The calling extension number and account code are prefixed with an appropriate feature access code.
- If data associated with an information element (such as an account code) is not available, no digits are outpulsed for that element. This could cause the call to fail at the far end because the feature access code is sent, but not the data associated with it.
- The Analog Networking feature transmits additional digits, increasing post dialing delay.
- SMDR can be instructed not to record the digit modification done for passing the network information.
- Dial pulse trunks cannot be used for analog networking because the # digit is not available on them.
- Analog networking does not work with System Option 36 (End of Dial Character #) enabled.
- The destination node trunk must be a dial-in type trunk (Tie or Special DISA).
- Only the last five digits of the calling extension number are displayed on SUPERSET display telephones and consoles.
- The # digit is required after the account code if the far end expects a variable length account code.
- The node identification provides a combined access code or calling extension number for the far end. If not associated with one of these elements, then it must match the same access code in the far end.
- If a device is forwarded to an analog network trunk, then the extension number of the forwarding telephone is sent as the calling extension number.
- On an intermediate node call (if incoming and caller's extension were received on the trunk), the caller's extension number received on the trunk is sent as the caller's extension number, even if the incoming trunk was forwarded to an outgoing trunk or routed to ARS.
- The Messaging – Call Me Back feature will function across analog networking on a remote extension only if the extension leaving the call me back message (the local extension) is assigned to a hunt group in CDE Form 17.

Programming

In CDE Form 03 (Class of Service), enable COS Option 613 (Display ANI Information Only) in the COS of the receiving devices.

In CDE Form 22 (ARS: Modified Digit Table), program the caller information elements that are to be transmitted on the trunk.

In CDE Form 02 (Feature Access Codes), assign a feature access code to Feature 39 (Analog Network Accept Caller's Extension) for PABXs that will be receiving analog network trunk calls.

To record SMDR information for incoming calls, enable COS Option 806 (SMDR - Record Incoming Calls) and COS Option 808 (Special DISA) in the COS of the destination node trunk; refer to the Station Message Detail Recording section.

Enable System Option 06 (Analog Networking SMDR) to overwrite the trunk ID information with the calling party's extension number.

Programming Example 1

This example shows how analog networking sends a calling party's extension number (Ext. 2001 dials Ext. 3001).

This particular example uses the following set up.

Form 26, ARS Digit Strings has the Leading Digit 3.

Form 26, Digit and Strings Subform has the Digits To Be Analyzed XXX (assuming a four digit extension).The Quantity to Follow is 0. The Term Type and Number is Route 1 (The digit string is directed to Route 1. Route 1 identifies a Modified Digit Entry, entry 1, in Form 22 and Form 23).

Form 22, ARS Modified Digit Table has Quantity To Delete left blank and Digits To Be Inserted 44*6# .

> 44 is the feature access code assigned to Feature 39, Analog Network Accept Caller's Extension in Form 2
>
> *6 is the "send caller's extension number" information element
>
> # is the delimiter

All the PBXs in the network have the same Feature Access Code (44) for Feature 39, Analog Network Accept Caller's Extension in Form 2.

Form 23, ARS Route Definition identifies the correct trunk group and the Modified Digit Entry of 1.

Form 15, Dial-In Trunks has N set to 0, M set to 0 and X is left blank for the system which receives the call.

Form 13, Options Subform has Far End Gives Answer Supervision status NO.

Ext. 2001 goes off-hook and dials Ext.3001.
The host system for Ext. 2001 has the Leading Digit Entry 3 (Form 26). The Digit String with 3 shows an entry of XXX and is directed to take Route 1. Route 1 identifies the correct trunk group and is directed to use the Modified Digit Entry 1. The Modified Digit Entry 1 deletes nothing and adds the string 44*6#.
The host system sees the Leading Digit 3 and waits for the next three digits (XXX).The host system then takes the trunk in the group identified by Route 1 (Form 23) and uses the Modified Digit Entry 1 which in Form 22 states to delete nothing and to add 44*6# to the beginning of what was dialed. So, the digits sent on the trunk are actually 442001#3001. The receiving site sees 44 first, which tells it to expect the calling party's extension (2001). When the host system saw the *6 it sent 2001 which the receiving site accepts as the calling party's extension number. The # tells the receiving site that no more digits will be sent for that Information Element. The 3001 is sent and the receiving system dials 3001 and displays Ext. 2001 as the calling party.

Programming Example 2

This example shows how CO type trunks can be used with analog networking. The trunks are programmed as Special DISA trunks on the destination node so that the network information can be dialed on the far end and accepted. The Special DISA account code must be sent prior to the analog networking information. Since the originating node must receive dial tone from the destination node before dialing the Special DISA account code, precede the Special DISA account code with a pause (the number and type of pauses depend on the trunk and the destination node DISA answer time). For example, if the destination node has a DISA answer time of 4 seconds, a Special DISA account code of 3333, and an accept caller's extension access code 49, the following modified digit string is required:

*2 3333 49 *6 # where:

- *2 is wait for dial tone (in originating node) - to wait for dial tone from the far end
- 3333 is a Special DISA account code (in destination node)
- 49 is accept callers extension (in destination node)
- *6 is insert callers extension number (in originating node)
- # is callers extension terminator (in destination node)

Operation

The SX-200 ICP system detects an incoming network information element on an incoming analog network call, and then displays the calling extension number on the called console or Mitel display telephone.

Table: Glossary of Analog Networking Terms

Term	Definition
COS	Class of Service
COR	Class of restriction
ARS	Automatic Route Select
ANI	Automatic Number Identification
Traveling Class Marks	A means of tagging the call with the caller's level of authorization (COS and COR), based on account codes, to be used by the private network in the handling of the call.
End Node	A network node that supplies network information elements to the network, but does not receive information elements from the network; (i.e., information flows from the end node out into the network.)
Intermediate Node	A network node linking at least two other network nodes. Information elements received by Node B from Node A, are supplied to Node C.
Hub	A node in the network that uses information elements from other nodes in the network in order to provide some network feature.
Information Elements	The following information elements are available: Caller's Extension Number, Caller's Dialed Account Code, and Primary Node-ID of the PBX.
Null Network Element	This is a network element's access code followed immediately by a terminator.
Marker	This is a special character to identify Information Elements that are programmed in CDE Form 22 (ARS Modified Digit Table); currently "*6", "*7", and "*8" are used.

Table: Information Elements for Analog Networking

Information Element	Feature Access Code	Marker	Comments
Caller's extension number	Accept Caller's Extension (destination node)	*6	The extension number passed in intermediate node or the extension number of the forwarding party or the extension number associated with the telephone is sent.
Caller's dialed account code	Verifiable Account Code (destination node)	*7	The account code the caller dialed is sent. If the trunk at the destination is a Special DISA trunk and this information element appears first in the digit string, the feature access code is not required.
PBX Node ID	Node ID (originating node)	*8	The Node ID of this PBX is sent. This marker usually precedes the caller's Extension marker, forming a compound extension number.

Attendant Abbreviated Dial Number Entry

Description

This feature allows the attendant to program system abbreviated dial numbers from the attendant console. Selected attendants have the option of making abbreviated dial numbers confidential. This restricts the viewing and changing of the number to only those attendants permitted to do so.

See Abbreviated Dial.

Conditions

The following conditions apply to this feature:

- Only one device (console or terminal) can be programming system abbreviated dial numbers at one time.
- While the console is using the abbreviated dial number entry feature, CDE Form 31 (System Abbreviated Dial Entry) cannot be accessed. Similarly, the console cannot use the abbreviated dial number entry feature while CDE Form 31 (System Abbreviated Dial Entry) is being accessed.
- The number programmed is not limited to external numbers - it can be any access code in the system.

Programming

To allow abbreviated dial programming, enable COS Option 111 (Attendant Abbreviated Dial Programming) for the console.

To allow the display of confidential abbreviated dial numbers, enable COS Option 110 (Attendant Abbreviated Dial Confidential Number Display) for the console. This option applies to both programming and dialing of the confidential abbreviated dial numbers.

The console has unrestricted use of all abbreviated dial numbers. Program an access code in CDE Form 02 (Feature Access Codes) for Feature 24 (Abbreviated Dial Access).

See Abbreviated Dial for further information.

Operation

To program an abbreviated dial number, perform the following procedure:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the ABBR DIAL softkey.
- Enter the desired index number.

  You can insert pauses in the numbers. Enter ∗9 for each 1-second. There are some conditions that apply to inserting pauses in speed calls.
- Press the ENTER softkey.
- Press the PRIVATE softkey (only if this is to be a confidential number).
- Enter the desired index number.
- Press the SET softkey.
- Press the EXIT softkey.

To view abbreviated dial numbers, enter the following:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the ABBR DIAL softkey.
- Enter the desired index number.
- Press the EXIT softkey.

Attendant Access (Dial 0)

Description

A feature access code (usually 0) is provided for reaching the attendant. The destination can change based on night/day service. The destination can be a device type other than a console or LDN. There is also a second class of dial 0. See Priority Dial 0 in this section.

The attendant can be reached by dialing:

- attendant access code (usually 0),
- console directory number
- attendant access code which is routed to an LDN key.

Conditions

The following conditions apply to this feature:

- The attendant access code can generally be used in the system wherever an attendant individual directory number can be programmed except in the call rerouting table; this can include call forwarding.

Programming

Assign an access code to Feature 11 (Extension General Attendant Access) in Form 02 (Feature Access Codes).

Assign a console directory number to Station Dial 0 Routing in CDE Form 19 (Call Rerouting Table).

Operation

Dial Attendant Access Code.

Attendant Advisory Message Setup

Description

There are eight default and seven programmable messages for use on Mitel telephones with LCD displays. The attendant can read a set's currently displayed message, or read through the available messages and choose one for display on the set; see Messaging - Advisory.

Conditions

None.

Programming

None.

Operation

To read all of the available messages, perform the following procedure:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the STATIONS softkey.
- Enter a SUPERSET display telephone extension number.
- Press the SET UP MSG softkey.

The messages will appear at the start of the second line on the console's LCD display. All of the available messages can be read by pressing the NEXT softkey. An OFF softkey is presented for the message currently displayed (if any) on the specified Mitel telephone. All other messages will result in an ON softkey being presented.

To set up or remove a message from a Mitel telephone, enter the same keys as above in the same order, and press the ON or OFF softkey as required.

Attendant Alarm Readout

Description

The attendant console can display the alarm logs presently active in the system. The attendant can cause a readout of the alarm messages one by one using the softkeys. The message indicates the fault and its location.

Conditions

For the attendant to access the alarms, the following conditions must apply:

- There must be no other console accessing alarms.
- There must be no console or maintenance terminal accessing maintenance or CDE function.
- Logs can be reviewed only once from the attendant console.
- The ALARM softkey indicator appears only if an alarm state currently exists.
- The alarm icon is not presented for minor alarms conditions.

Programming

Enable COS Option 102 (Attendant Display of System Alarms) for the console.

Operation

To obtain an alarm readout, press the following keys in sequence:

- FUNCTION key.
- ALARM softkey.
- MORE softkey.
- CANCEL softkey.

Attendant Automatic Overflow

Description

Attendant automatic overflow provides a recorded announcement to incoming calls that are not answered by the attendant or subattendant within a pre-defined time. This feature operates primarily during peak periods of incoming traffic. For more information on the operation and programming see the feature RAD Support.

Conditions

An answering machine should not be used. Because the connection to the recorded announcement device is listen only, the caller cannot leave a message.

Programming

- In the console's class of service, set COS Option 118 (Attendant Call Forward - No Answer Timer) for the time that the answering machine should wait before playing the recorded announcement.
- Enable COS Option 705 (Automatic Overflow from Attendant) in the caller's COS (trunk, extension, or Mitel telephone).
- Enter the extension of a recording device hunt group into CDE Form 19 (Call Rerouting Table) under "UCD/Attendant Recording Routing for This Tenant", under the console's tenant.
- Set System Option 51 (Final Ring Time-Out timer) to allow the attendant sufficient time to answer waiting calls. The default is one minute. Calls still waiting when the timer expires will be terminated.

Operation

When an incoming call is not answered within the programmed time, a recording is connected to the caller to advise the caller that there are many incoming calls and the call will be answered shortly. The actual message given is a pre-recorded message, recorded by the customer on a customer-provided recorded announcement device. The position of an incoming call being held in the calls waiting to be answered queue, is maintained while receiving the recorded message. The recorded message will be terminated as soon as an attendant becomes available. If the incoming call is still waiting to be answered after the complete message has been delivered, the incoming call is connected to Music-on-Hold from the connected tenant. The Final Ring Time-out (System Option 51) in CDE Form 4 continues to run until the console answers the call. It does not stop when the RAD answers the call.

Attendant Bell Off

Description

This feature allows the attendant to mute the console ringer. Incoming calls are indicated by a flashing Answer Key LED and LDN softkeys displayed on the console. When the console ringer is disabled, “BELL OFF” appears on the second line of the console LCD display.

Conditions

The following conditions apply to this feature:

- The system default state for this feature at power-up is “BELL ON”.
- The status of the bell is ignored by lockout alarms ringing at the console.
- There is no bell on the console if Attendant Tone Signaling has been enabled on the console.

Programming

Enable COS Option 100 (Attendant Bell Off) for the console.

Operation

To disable the console ringer, press the following keys:

- FUNCTION key.
- BELL OFF softkey.

To enable the console ringer, press the following keys:

- FUNCTION key.
- BELL ON softkey.

Attendant Busy Override

Description

This option allows the attendant who encounters a busy connection, to override the connection and enter the call; see Override (Intrude).

Conditions

The following conditions apply to this feature:

- This applies to override of trunk calls through the Attendant Direct Trunk Select feature. See Override (Intrude) for more conditions.
- You cannot override a call if a trunk or extension in the call has COS Option 238 (Override Security) enabled in its class of service.
- An Override warning tone is heard by both parties before voice contact is established.
- If Call Forward Busy is programmed on an extension, the calling party can override that extension when busy if the calling party has the Camp-on Priority Over Call Forward Busy feature enabled. In other cases the call is forwarded.

Programming

Enable COS Option 500 (Override) for the console.

Operation

Having reached a busy number:

- Press and hold down the console OVERRIDE softkey. All parties in the connection hear a warning tone and the attendant is connected to the call.
- Release the OVERRIDE softkey. The attendant is released from the call.

Note: If the call cannot be overridden, re-order tone is returned, and the console LCD displays “CANT”.

Attendant Callbacks - Busy/No Answer

Description

The attendant can set up a callback if the called destination is busy or does not answer. The attendant can also cancel all callbacks in the system.

See Expensive Route Warning for callbacks to ARS for less expensive routes.

See Callbacks for details on callbacks.

Conditions

The following conditions apply to this feature:

- See Callbacks.
- The attendant can only cancel callbacks from a system device if the Device Interconnection Control Table allows the device and the console to be connected (check CDE Form 30 - Device Interconnection Table).
- See Attendant Default Call Position for information on the call position where the CALLBACK softkey appears.

Programming

None.

Operation

Having reached a busy or non-answering number, press the CALLBACK softkey. The console rings as soon as the destination becomes available. When the attendant answers the ring, CALLBACK appears on the display. As soon as the attendant answers the callback, the called party rings.

Attendant Call Forward Setup and Cancel

Description

The Attendant Call Forward Setup and Cancel feature allows the attendant to set up, review and cancel call forwarding for any extension. The extension for which the attendant sets up forwarding need not have any of the call forwarding features in its COS. The attendant may also set up call forwarding from the extension to the attendant. The attendant can also cancel call forwarding for all extensions at once.

Conditions

The following conditions apply to this feature:

- The extension or console to which the calls are forwarded must not have Option 234 (Never a Forwardee) in its COS.
- Valid forward destinations are: hunt groups (not data), consoles, telephones, abbreviated dial numbers, Dial 0 access code, LDN, night bells, and ACD paths.

- The cancel of all forwarding by the console only applies to extensions for which device interconnection control checks pass between the console and the extension.
- When the console operator presses the Cancel All Call Forward key, a warning message will appear saying that all station and set call forwarding will be cancelled. The operator will then cancel or confirm the action.
- The console operator can disable all the call forward modes for an extension if the system option Feature Level is set to level 1 or greater.
- The TURN ALL OFF, TURN ON, and SAVE/OFF softkeys and the asterisk which shows that a forwarding type is on, only appear if the System Option Feature Level is set to level 1 or greater.

Programming

Enable COS Option 123 - Call Forward Setup and Cancel in the attendant COS, and if split call forwarding is required, enable COS Option 260 (Internal/External Split Call Forwarding) on the extension to be programmed.

Operation

To set up call forwarding, perform the following procedure:

- Press FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the STATIONS softkey.
- Dial the extension number.
- Press the CALL FWD softkey.
- If split forwarding is enabled, press the EXTERNAL or INTERNAL softkey. The display shows the status of ALWAYS, BUSY, and NO ANSWER.The External or Internal softkey displays the other half of the call forwarding.
- Press the ALWAYS **or** NO ANSWER **or** ON BUSY softkey.
- Enter the desired call forward destination extension number.
- Press the SAVE/ON or SAVE/OFF softkey. The SAVE/ON softkey saves and enables the call forward type. The SAVE/OFF softkey saves but disables the call forward type. The SAVE/OFF softkey only appears if the system option Feature Level is set to level 1 or greater.

To review call forwarding for an extension, press the following keys, in this sequence:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the STATIONS softkey.
- Dial the extension number.
- Press the CALL FWD softkey.
- If split forwarding is enabled, press the EXTERNAL or INTERNAL softkey .
- Press the ALWAYS **or** NO ANSWER **or** ON BUSY softkey.
- If split forwarding is enabled, press the EXTERNAL or INTERNAL softkey to toggle to the other source for call forwarding.

Note: A call forwarding that is enabled shows an asterisk beside it and a TURN OFF softkey on the display screen. A call forwarding that is disabled does not show an asterisk and does have a TURN ON softkey. The asterisk and the TURN ON softkey only appear if the Feature Level system option is enabled.

To cancel **one** mode of call forwarding for a single extension, press the following keys, in this sequence:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the STATIONS softkey.

- Dial the extension number.
- Press the CALL FWD softkey.
- Press the EXTERNAL or INTERNAL softkey (only when split forwarding enabled).
- Press the ALWAYS **or** NO ANSWER **or** ON BUSY softkey.
- Press the TURNOFF softkey.

To cancel **all** modes of call forwarding for a single extension, press the following keys, in this sequence:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the STATIONS softkey.
- Dial the extension number.
- Press the CALL FWD softkey.
- Press the EXTERNAL or INTERNAL softkey (only when split forwarding enabled).
- Press the TURN ALL OFF softkey.The TURN ALL OFF softkey only appears if the FOR System Option 102, Feature Level is set to level 1 or greater and if at least one of the call forwarding modes is on.

To cancel call forwarding at all stations:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the MORE . . . softkey.
- Press the CAN. ALL FWD softkey.
- Confirm the warning message that all station and set call forwarding will be cancelled.

Attendant Call Park and Page

See Park and Page.

Attendant Call Park - Specific Orbit

See Call Park System - Specific Orbit.

Attendant Call Park - System Orbit

See Call Park System - System Orbit.

Attendant Call Selection

Description

The attendant console has up to ten call selection positions that appear as softkeys when the console is receiving an incoming call. The system sets up some positions by default for certain call types. Other positions are user defined LDN keys.

Calls arriving at the console are queued on a first come first served basis and the answer LED indicator flashes. The LCD display also indicates the number of calls waiting.

The attendant call selection feature allows the attendant to answer calls in sequence or by call type. Pressing the answer key answers calls in the order that they arrived at the console regardless of call type.

Using a selection position softkey answers calls by call type regardless of the time they arrived at the console.

Conditions

None.

Programming

For setting up answering positions other than the defaults listed in ATTENDANT DEFAULT CALL POSITIONS, see Console LDN Keys.

Operation

To answer the first call in the attendant queue, press the ANSWER key - the tone ringer stops, the prompt associated with the call type is removed if there are no more calls of that type calling, the display shows the number of the calling trunk or extension and the attendant is connected to the calling party.

To answer a specific call type, press the softkey associated with the desired call type - the tone ringer stops, the ANSWER LED indicator indicates steadily, the display shows the number of the originating party, and the attendant is connected to the calling party.

Attendant Call Splitting and Swapping

Description

During the setting up of a call between two parties, the attendant may be required to speak to both parties, or to speak privately with either party. The attendant can do this by using the CONF, SOURCE, and DEST softkeys.

Conditions

The feature is subject to Attendant Conference conditions

Programming

The feature is subject to Attendant Conference programming

Operation

Establish a 3-party conference via the CONF softkey - the attendant may now speak to both parties.

Press either the SOURCE or DEST softkey to split the call and talk to the selected party privately. The attendant may alternate between the parties by pressing one of the two softkeys, as required to select the desired party.

Press the RELEASE key to disconnect the attendant from the conference, leaving the two parties connected.

Press the CANCEL key to drop the conference, disconnecting both parties from the attendant, and leaving the console in idle mode.

Attendant Calls Forwarded On No Answer

Description

Calls directed to the Listed Directory Number (LDN) of a console which are not answered within a predetermined time-out period are rerouted to a NIGHT 1 destination (if there is one). For CO trunks, the reroute is to the trunk's NIGHT 1 answer point. For DID trunks, the reroute is to the Attendant DID Access NIGHT 1 answer point for the trunk's tenant. For dial-in Tie trunks, the reroute is to the Attendant Dial-in Tie Access NIGHT 1 answer point for the trunk's tenant. For sets, consoles, and DISA trunks, the reroute is to the Dial 0 NIGHT 1 Answer Point for the set's or console's or trunk's tenant.

See Call Rerouting for how this fits in with recalls to the console.

Conditions

The following conditions apply to this feature:

- Rerouting does not occur unless the NIGHT 1 and day service answer points are different.
- No further rerouting is done for DID/TIE trunks after a DID/TIE BUSY or ALWAYS reroute, or after a DID/TIE trunk is routed to the Night Access Point for DID or TIE trunks.
- No reroute is done for calls directed to the LDN from a DND, Vacant or Illegal number intercept, or from a DID/TIE intercept routing.
- No reroute is done for a call recalling to the console (calling the default RECALL softkey).
- The LDN must be the prime answering point for the calls in order for rerouting to occur. No rerouting is done if the call is forwarded to the LDN.
- The feature applies after forwarding has occurred to an LDN.
- This feature is mutually exclusive with Attendant Automatic Overflow; also see Attendant Automatic Overflow.

Programming

Enable COS Option 107 (Attendant Automatic Call Forward - No Answer) for the console where the LDN is programmed. If the LDN is programmed on more than one console, enable COS Option 107 in the Class of Service of the console that has the lowest bay/slot/circuit location.

Set the time-out period via COS Option 118 (Attendant Call Forward - No Answer Timer) for the console where the LDN is programmed; the default time is 30 seconds.

Enter a NIGHT 1 routing point for the appropriate call types via CDE Form 19 (Call Rerouting Table).

Note: BRI circuits are considered DID trunks for the purposes of forwarding, and therefore use the Night 1 answer point in Form 19 for DID Attendant Access Night Points, not the Night 1 answer point in Form 45.

Operation

When the call arrives at the console, the timer for the Forward On No Answer is started. When the timeout occurs, the caller is routed to the appropriate answer point.

Attendant Conference

Description

This feature allows the attendant to enter into a conference with the destination party and the source party of a call. The attendant may also initiate a 3-party conference call. When the attendant is in a conference, a periodic warning beep is given to all internal parties if System Option 10 (Attendant Conference Beeps) is enabled.

Conditions

The following conditions apply to this feature:

- The console may be involved in a conference with a maximum of two other parties.
- A conference cannot be created involving another console.
- Device interconnection checks do not apply to the conference.

Programming

To enable attendant conference warning beeps, enable System Option 10 (Attendant Conference Beeps).

COS Option 120 (Attendant Conference Disable) must be DISABLED in the console's COS.

Operation

To enter into a conference with the source and destination parties, press the CONF softkey.

To initiate a conference, perform the following procedure:

- Dial the first party and establish a connection.
- Dial the second party and establish a connection.
- Press the CONF softkey. The three parties are now in a conference.

Attendant Console Key Function for DID Printing

Description

This feature allows the attendant to print the guest's DID number (using the Function key) on limited-width printers from DID Server Application.

Conditions

None.

Programming

Name must be entered in Form 9 (Desktop Device Assignment) to appear on the DID printout.

If a DID number is not assigned to the Room Extension, the "Direct" display shows "Not Available".

Operation

To print the DID Number:

- Press the FUNCTION key.
- Press the DID Print softkey.

This prints the Name of the Guest, the Guest Room number, followed by the DID Number (Direct). The printer output from the console can be output to a label or card printer to give the Guest his own personal DID number on a card or label. See below for an example:

Name: JOHN SMITH
<blank line>
Room: 2000
<blank line>
Direct: 6135432559

Attendant Console Display Language

Description

This feature allows the attendant to display the attendant console softkeys in English, French or Spanish.

Conditions

None.

Programming

None.

Operation

To change the softkey language:

- Press the FUNCTION key.
- Press the LANGUAGE softkey.
- Press the softkey of the desired language.

Attendant Console Handset and Headset Receiver Volume Control

Description

Users of the SUPERCONSOLE 1000 and 5540 IP Console can adjust the volume of the handset and the headset receivers.

Conditions

- The SUPERCONSOLE 1000 must have the Part Numbers 9189-000-300 or 9189-000-301.
- Volume adjustments for the handset receiver also affect the headset receiver.
- Volume adjustments can be made while a call is in progress or while the console is idle.

Programming

None.

Operation

To adjust the handset and headset receiver volume:

- During a conversation (using the handset or headset) or while the console is idle, press the Volume ∧ and Volume ∨ keys repeatedly to increase or decrease the receiver volume.

Caution: **Because continuous exposure to loud sounds can contribute to hearing loss, keep the volume at a moderate level.**

Attendant Console Last Call Retrieve

Description

This feature allows the user of the attendant console to retrieve a ringing call to the original dialled extension. When the operator of the console accidentally releases a call to the wrong extension number, the operator may hit the CANCEL hardkey to retrieve the call.

Conditions

The following condition applies to this feature:

- Retrieving the last call is only available to attendant console users.
- After the attendant console releases the call accidently, the retrieved call can only be retrieved if the caller is listening to ringback and is not in talk or idle state

Programming

None.

Operation

To retrieve the last ringing call that was released on an attendant console

- Press the CANCEL hardkey.

Attendant Console LCD Display

Description

The system continually displays the time of day on the right-hand portion of the status line of the attendant console LCD display. When the console is idle, the date (month, day, year) is also displayed. The displayed time is used by Message Waiting, Traffic Measurement, SMDR, Family 1 and Family 4 telephones, and other features. The console attendant can change the date and/or time (see Date and Time Setup).

The attendant console may have calls from outside trunks and extensions queued that are waiting to be answered. The total number of calls in the queue is displayed in the attendant console queue (Calls Waiting) area of the display located in the top right corner of the LCD display.

The attendant can put a party on softhold or hardhold that also has a call (one party or Conference) on softhold. This is called stacked hold.

When the attendant console establishes or answers a call, the display provides information about the call. The available items of call information are:

- Extension number and set name
- Tenant name
- Trunk name
- Trunk group name
- Trunk number
- ANI information
- COS and Class of Restriction (COR)
- COS name
- Call forward recall information
- On hold information (Extension Number or Conference) of the calling party is provided if that party has a call on softhold.

The system programmer can assign names to sets, classes of service, tenants, trunks and trunk groups; see Names.

Conditions

The following conditions apply to this feature:

- A name must be programmed for the set. Names are programmed in CDE Form 09 (Desktop Device Assignments). Users of Family 1 and Family 4 telephones can program their name at their set.
- The COS number and COR number appear only if there is no COS name programmed.
- Names associated with trunks appear only after the console answers a trunk call.
- If the attendant answers a call and that caller has a call (one party or Conference) on softhold, the attendant does not take over the caller's softhold.
- When a HOLD key is pressed to put someone on hold, only the party that is talking to the attendant will be put on hardhold. If there is a softhold party (either the SOURCE or DEST), this party will be connected to the attendant. The attendant must press the CANCEL or RELEASE key to hang up on this call.
- The tenant name appears only on rerouted TIE, DID, or Dial-In calls where the tenant name is programmed in the CDE Call Rerouting Table. If the tenant name is displayed, no other trunk information appears.
- The trunk group name appears only if there is no trunk name programmed.
- The trunk number appears only if there is no trunk name or trunk group name programmed.
- If there are no calls in the calls waiting queue, there is nothing in the CW area of the LCD display.
- The maximum number of calls waiting that can be displayed at the console is 99.

Note: The actual queue maximum is 200 calls waiting.

Programming

None.

Operation

When the attendant console establishes or answers a call, the display provides information described above about the call. For Attendant Calls Waiting operation refer to Attendant Call Selection and/or Attendant Hold Positions for operational details.

Attendant Console LDN Keys

Description

Each console has nine programmable listed directory number (LDN) positions. Each LDN position can be programmed as the answer point for a particular type of call. Each LDN key can be given a descriptive label, allowing the attendant to answer the call with an appropriate response.

LDNs can appear at more than one console in a system to allow calls to be presented to specified consoles simultaneously.

LDN's may also appear at subattendant positions. See feature description for Subattendant Console LDN Keys.

The attendant console can answer calls from an LDN by either using the ANSWER key or by selecting the LDN key directly. See Attendant Call Selection in this document.

Conditions

The following conditions apply to this feature:

- Any or all LDN keys and labels may be programmed to appear.

- Each console must have its own LDN numbers explicitly programmed in CDE Form 08 (Attendant LDN Assignments).
- There are several default call positions provided by the system for various call types; see Attendant Default Call Positions.
- The RECALL key (key #1) cannot be changed.
- Station and Mitel telephones and consoles cannot directly dial an LDN.
- Each LDN key is provided with a default label. For key F2, the label is 'INTERNAL'. For key F3 to key F0 it is 'LDN n', where n is from 1 to 8. An LDN programmed at the key can have a label that replaces the default label.
- It is recommended that key F2 not be assigned to an LDN key since some calls are directed to that key by the system; see Attendant Default Call Positions.
- If the Trunk Answer From Any Station (TAFAS) feature is used then is it recommended that key F0 not be programmed as an LDN or that calls not be routed to the LDN when TAFAS calls are also to be answered; see Attendant Default Call Positions.
- For multi-appearance LDN keys, the console (or subattendant position) with the lowest physical location number (bay #/ slot #/ circuit #) is always the owner of an LDN.

Programming

Assign access codes to the console's LDN positions via CDE Form 08 (Attendant LDN Assignments). The feature descriptions in this section identify which features can use console LDN keys as answer points.

Operation

The attendant may selectively answer any incoming call type by pressing the appropriate LDN softkey.

Attendant Console Lockout

Description

The attendant can enter an access code to restrict the capabilities of the attendant console. This prevents system tampering via the console during breaks, etc. When the console is locked out, the following restrictions take effect:

- no outgoing trunk calls can be made.
- there is no attendant function access.

The attendant console can still be used to initiate internal calls, and to answer incoming trunk calls.

Conditions

The attendant can lock out the console at any time as long as there is no source party connected.

While the console is locked out, you can still access Emergency 911 alarm information, but you cannot clear the 911 alarms.

Programming

Assign an access code to Feature 17 (Console Lockout Access Code).

Operation

To lock out the console:

- Enter the console Lockout access code. The display changes to "Console in Restricted Service".

To return the console to normal operation:

- Re-enter the code.

Attendant Console Macro Keys

Description

This feature allows users of the SUPERCONSOLE 1000 to program Macro keys. A macro is a series of keystrokes that you assign to a single key. Instead of repeating the keystrokes each time you want to perform a task, you press the macro key to execute all the keystrokes at once. Macro keys may be used for transferring calls to voice mail that have recalled to the console (recovering calls released to the wrong extension), or for one-button dialing of frequently called telephone numbers.

Conditions

The following conditions applies to this feature:

- This feature is available on the SUPERCONSOLE 1000 (Part Numbers 9189-000-300 and 9189-000-301).
- The blank key next to the Set Page key on the console can be programmed as a macro key.
- The Trunk Group and Set Page keys can be reprogrammed as macro keys. To return the functionality back, delete the macro key functionality.
- Resetting the console does not erase the programming for the macro keys.
- Upgrading firmware does erase the programming for the macro keys.
- While programming a macro key, you can answer a call by pressing the Answer key. After Release Idle, the initial screen appears.
- If you want to recover a call that you transferred to a wrong extension, you must press the macro key while the extension is still ringing. Recovery also depends on no other action being started, such as dialing or answering a call, following the transfer. The Cancel key can also be used to recover calls.

Programming

To program a macro or to clear a previously programmed macro

1. While holding down the Function key, press the hardkey (the blank key or Set Page or Trunk Grp key) that you want to program until the display shows the macro programming screen. See the list of special function softkeys for programming macros.
   The display shows the macro keystrokes (if any) currently programmed to the selected key, for example, FIRM KEY 1: [TONES ON] 7224609.
2. Enter the keystrokes you want to include in your macro.
   You can enter up to 15 keystrokes using the dial pad keys and the special function softkeys .
   Use the backwards arrow key to correct any mistakes.
   Tip: Write down the keystrokes in case you experience problems with the console and have to reprogram your macros.
3. To clear a previously programmed macro, press the [CLEAR] softkey.
   Clearing macros from keys formerly programmed as Trunk Group or Set Page keys restores their original function.
4. Press the SAVE/EXIT softkey to complete the programming or press the EXIT softkey to exit without saving the programming.

Special Function Softkeys for Programming Macros	
Softkey	**Action**
[DEST]	Dials the number displayed on the DST line when the macro key is pressed. (Used to program a macro in order to transfer calls to voice mail; see the next section for details.)
[LAST NUM]	Redials the last number dialed from the console when the macro key is pressed. (Used to program a macro in order to recover the most recently released call; see the next section for details.)
[TONES ON]	Sends subsequent digits as DTMF tones. (Required when programming a macro to transfer calls to voice mail; see the next section for details.)
[PAUSE]	Inserts a 1.5 second pause. When the pause ends, digits after the pause are dialed. You can create a longer pause by pressing the [INSERT PAUSE] softkey multiple times.
[FLASH HOOK]	Inserts a flash for further dialing on a trunk. For more information, see "Flashing on Trunks" in the SUPERCONSOLE 1000 User Guide.
[RELEASE]	Disconnects the console from the call in progress. (Required when programming a macro to transfer calls to voice mail; see the next section for details.)

To program a macro that transfers calls to a voice mailbox

1. Follow the first three steps in the procedure to program a macro.
2. Dial the voice mail hunt group number.
3. Press the [PAUSE] softkey to allow the voice mail system time to respond.
4. Press the [TONES ON] softkey.
5. Press * (or other key as required by the voice mail system).
6. Press the [DEST] softkey to dial the digits displayed on the DST line when you press the macro key.
7. Press the [RELEASE] softkey to complete the transfer and release the console from the call
8. Select another firmkey to program or press the SAVE/EXIT softkey to leave programming mode. Now, when a call you transferred to an unanswered extension recalls, you can press the macro key to transfer the caller to the extension's voice mailbox.

To program a macro that recovers calls that you transferred to the wrong extension

1. Follow the first step in the procedure to program a macro.
2. Enter the Directed Call Pickup access code.
3. Press the [LAST NUM] softkey.
4. Press the SAVE/EXIT softkey to complete the programming.

Operation

Press the key programmed with the macro that you want to execute.

Note: If you want to recover a call that you transferred to a wrong extension, you must press the macro key while the extension is still ringing. Recovery also depends on no other action being started, such as dialing or answering a call, following the transfer. The Cancel key can also be used to recover calls.

Attendant Console Set Paging - Directed, Group, or All Set

Description

The attendant may make a directed page, group page, or all set page by pressing the Set Page key on the console.

Conditions

The following condition applies to this feature:

- A page cannot be made to a set that has DND enabled.
- Mitel 5212, 5020, 5220, 5224 5312, and 5324 IP Phone, SUPERSET 4025, SUPERSET 4125, SUPERSET 410, and SUPERSET 420 telephone, and Family 1 telephone users can respond handsfree to a directed page. Refer to Handsfree Answerback to a Directed Page for details.
- The attendant cannot page SUPERSET 401+ telephones.
- Only one group page may be performed to a page group at any one time.
- A Group Page cannot be initiated if an All Set Page is active.
- The paged party can have COS option 685 "Key System - Can receive All Set Page" disabled to block receipt of an All Set Page.
- An All Set Page does not check for tenanting restrictions.
- The SUPERCONSOLE 1000 and 5540 IP Console supports a whisper page. See Whisper Announce.

Programming

Enable COS Option 684, Key System - Can Make All Set Page in the class of service of the attendant console.

Operation

Directed Page: Press the Set Page key, then dial the extension number, then make the page.

Group Page: Press the Set Page key, then press the Group Page softkey, then press the two-digit group number, then make the page.

All Set Page: Press the Set Page key, then press the All Set Page softkey, then make the page.

Attendant Date and Time Setup

See Date and Time Setup.

Attendant Default Call Positions

Description

Three incoming call indicators are provided by the system to identify calls to the console directory number. In addition to these call indicators are LDN keys on the console; see Console LDN Keys.

The three default positions are:

- F0 (NIGHT BELL) - calls ringing any night bell in the console's tenant group (see TAFAS).
- F1 (RECALL) - recalls of calls handled by the console, or for multiple console operation, by any console in the system (see Recall).
- F2 (INTERNAL) - calls directed to the console's internal directory number.

Conditions

The following conditions apply to this feature:

- The INTERNAL position uses whatever label is programmed at key F2. The label is by default INTERNAL but an LDN key with a different label may be programmed at the same key. This does not affect the direction of internal calls to this position - only the label used changes.
- Callbacks ringing the console appear at the INTERNAL position.
- If key F0 is programmed with an LDN, calls to the LDN take precedence over night bell calls. In this case, the key label shows the name programmed for the LDN key. Pressing the key answers calls to the LDN. Calls to the night bell appear at key F0 only after all LDN calls have been handled; the label changes to NIGHT BELL.
- Serial calls, being recalls, appear at the RECALL position.
- The RECALL position (key F1) cannot be changed and cannot have an LDN key programmed at the same key.

Programming

None.

Operation

None.

Attendant Destination (DEST) Key

Description

This feature allows the attendant to press a softkey (DEST) to speak to the destination party of a call, to SWAP between the destination and source parties or to SPLIT a conference call.

The destination party's extension number, COS, and COR are displayed on the second line of the console's LCD display and the source party is put on consultation hold.

See Attendant Call Splitting and Swapping.

Conditions

This softkey only appears when the attendant console is connected to a multi-party call and the source party can be put on consultation hold.

Programming

None.

Operation

Press the DEST softkey - the console is connected to the destination party and the source party is put on consultation hold.

Attendant Directed Call Pickup

Description

The attendant can perform a directed call pickup from the console. This will permit calls to be retrieved before the recall timer expires or if calls have been transferred to the wrong extension.

Conditions

None.

Programming

Assign an access code to Feature 09 (Directed Call Pickup).

Enable COS Option 218 (Directed Call Pickup).

Operation

- Dial the Directed Call Pickup code.
- Dial the extension number of the ringing telephone - the call is connected.

Attendant Direct Trunk Select

Description

The console can be used to directly access (seize) a trunk for maintenance or operational procedures.

Conditions

The following conditions apply to this feature:

- For viewing the status of and accessing a trunk, the attendant must be allowed to connect to the trunk. See Device Interconnection Control.
- For accessing the trunk, the trunk must not be in the process of being seized by another device in the system.
- SMDR applies to the call when a trunk is accessed. If there are no SMDR records available, the call continues without an SMDR record; see Station Message Detail Recording.
- Trunks cannot be accessed by “Attendant Function” if the console’s COS has COS Option 200 (Account Code, Forced Entry - External Calls) or COS Option 201 (Account Code, Forced Entry - Long Distance Calls) enabled.
- When an LS/GS trunk is used as a dictation trunk, the M/MM leads are used to make the trunk busy while the tape is removed or the unit is powered down. If a console tries to access this type of busy trunk, only the EXIT softkey appears (the OVERRIDE softkey and FORCE RLS softkey are not available) and the message “Trunk busy due to M/MM leads” is displayed.

- The trunk must be a member of a trunk group.

Programming

None.

Operation

To check the current status of a trunk, perform the following procedure:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the TRUNKS softkey.
- Enter the desired trunk number.
- Press the STATUS softkey.
- Press the EXIT softkey.

To select a trunk for attendant access only, enter the following softkeys:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the TRUNKS softkey.
- Enter the desired trunk number.
- Press the ATT ACCESS softkey.

To override a call on a busy trunk:

- Press the OVERRIDE softkey (see Conditions).

To force-release a non-idle trunk (a call in progress is dropped), enter the following softkeys:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the TRUNKS softkey.
- Enter the desired trunk number.
- Press the ATT ACCESS softkey.
- Press the FORCE RLS softkey (see Conditions).

To release a trunk from attendant access, press the RELEASE key.

Attendant DISA Code Setup

Description

This option allows the attendant to change the Direct Inward System Access (DISA) security code (Feature 19) in CDE Form 02 (Feature Access Codes) that a DISA caller must dial to access the system.

Conditions

The following conditions apply to this feature:

- The DISA code cannot conflict with the numbering plan.
- The DISA code is limited to a maximum of five digits.
- The DISA code cannot be displayed.
- An attendant cannot delete the DISA code; it may only be deleted via CDE Form 02 (Feature Access Codes). Refer to the Form 02 - Feature Access Codes section in Program CDE, for further details.

- While the DISA code (Feature 19) in CDE Form 02 (Feature Access Codes) is being setup from the Console, CDE Form 02 cannot be accessed from customer data entry. Similarly, while Feature 19 in CDE Form 02 (Feature Access Codes) is being accessed from customer data entry, the DISA code cannot be changed from the console.
- The DISA access code used with DID and Tie trunks must not lead with an asterisk *, if ANI/DNIS is to be collected.

Programming

Enable COS Option 103 (Attendant DISA Code Setup) for the console.

Operation

To change the DISA access code, perform the following procedure:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the MORE... softkey.
- Press the DISA CODE softkey.
- Enter a new DISA access code.
- Press the SET softkey.

Attendant Do Not Disturb (DND) Setup, Cancel or Override

Description

The attendant can set up or cancel Do Not Disturb (DND) for an extension.

When calling an extension with DND enabled, the attendant can override DND.

See Do Not Disturb.

Conditions

If the extension does not have COS Option 121 - Station Do Not Disturb enabled in its COS, the extension is not able to alter the DND setting; it can only be done through the console.

Programming

Enable COS Option 121 - Station Do Not Disturb in the console COS.

Operation

To set up Do Not Disturb on an extension, perform the following procedure:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the STATIONS softkey.
- Enter the extension number.
- Press the NO DISTB softkey.
- Press the EXIT softkey.

or

- Press the FUNCTION key.

- Press the GUEST ROOM softkey.
- Enter the extension/room number.
- Press the NO DISTB softkey.
- Press the EXIT softkey.

To cancel Do Not Disturb on an extension, press the same keys in the same order.

To set up or cancel Do Not Disturb while talking to an extension, toggle the NO DISTB softkey.

Attendant Emergency Call (911) Detection

See Emergency Calls (911) - Detection and Reporting.

Attendant Extension Busy-Out

Description

This feature allows the attendant to busy out any ONS extension (the extension is removed from service and cannot originate or receive any calls), and to remove the busy-out condition. The same operation is provided in maintenance. Refer to the RS-232 Maintenance Terminal section. If the attendant dials the number of a busied-out extension, the console displays the extension number and “BSY OUT” in the destination display and the attendant will receive reorder tone.

Conditions

The following conditions apply to this feature:

- If the extension is idle when the attendant sets the busy-out condition, the extension is busied-out immediately.
- If the extension is not idle when the attendant sets the busy-out condition, the extension is busied-out as soon as the extension becomes idle.
- The extension is treated as being busy with regard to forwarding. If the extension has “Call Forwarding - Busy” or “Call Forwarding - Always” set up, the forwarding occurs.
- If the extension is a member of a hunt group, all calls to the hunt group bypass the busied-out extension.
- A locked out extension cannot be busied out.

Programming

Enable COS Option 112 (Attendant Station Busy-Out).

Operation

To busy out an extension, perform the following procedure:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the STATIONS softkey.
- Enter the extension number.
- Press the BUSY OUT softkey.
- Press the SET softkey.

To remove the Busy-Out condition from an extension, perform the following procedure:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the STATIONS softkey.
- Enter the extension number.
- Press the BUSY OUT softkey.
- Press the CLEAR softkey.

Attendant Flash Over Trunk

Description

The attendant can flash on a trunk by pressing the FLASH softkey. A flash is sent out on the trunk, and dialing is restarted on the trunk.

Conditions

The following conditions apply to this feature:

- There must be only one party at the console.
- The attendant may flash on incoming and outgoing trunks.
- The trunk must be in a trunk group.
- The feature is not available when the console is locked out.
- The flash duration is based on the trunk hardware used. For trunks in digital bays, the duration is programmed using the Flash Duration Trunk Circuit Descriptor option. For all other trunks, the time is approximately 200 ms.
- The flash type is programmable for certain trunks. See Trunk Circuit Descriptor Options. The Flash Over Trunk option is ignored when an attendant does a flash on a trunk.
- If the trunk is the source party then the trunk becomes the destination party.
- Dialing continues until an inter-digit timeout or the maximum number of digits (26) have been dialed. Digits dialed on the console keypad are sent out on the trunk by the system.
- When dialing is restarted:
  - If COS Option 802 (Limited Wait For Dial Tone) is not enabled for the trunk then the system will wait indefinitely for dial tone to be detected. COS Option 805 (Trunk No Dial Tone Alarm) does not apply.
  - Tones are turned off at the console and must be turned on again if needed later; see Attendant Tone Signaling.
  - The trunk's SMDR is completed before dialing is restarted. SMDR applies to the dialing that is started on the trunk.

Programming

None.

Operation

While the attendant is connected to an outgoing trunk:

- Press the FLASH softkey.
- Dial as required.

Attendant Function Access

Description

By pressing the console FUNCTION key and the ATT FUNCTION softkey, the attendant can access the following attendant functions:

- ABBR DIAL
- ALARM (read alarms)
- APPLICATION (To access CDE or maintenance)
- BELL ON/OFF (if named)
- BUSY OUT
- CALL FORWARD
- CANCEL ALL CALLBACKS
- CANCEL ALL CALL FORWARDING
- DAY/NIGHT 1/NIGHT 2 switching
- DO NOT DISTURB SETUP AND CANCEL
- ESPAÑOL (Spanish language prompts and messages on console)
- FLEXIBLE NIGHT SERVICE
- FORCED TRUNK RELEASE
- FRANÇAIS (French language prompts and messages on console)
- GUEST ROOM functions (if enabled)
- MESSAGE WAITING SETUP AND CANCEL
- SEND MESSAGE
- SET DATE
- SET TIME
- SET UP MESSAGE
- STATIONS
- SYSTEM IDENTIFIER
- TRUNK STATUS/ACCESS/BUSY OUT.

For more information about attendant functions, refer to the individual feature descriptions in this section.

Conditions

Attendant functions are not available if the attendant console Lockout option (Feature 17) is invoked.

Programming

None.

Operation

Press the FUNCTION key - the attendant function softkeys appear on the console LCD display. One of the softkeys is MORE. Press this key for access to more functions. Press MORE again to return to the first set of functions.

Attendant Hold Positions

Description

The attendant may place an extension or trunk on hold in one of eight HOLD positions. There are four keys; HOLD1, HOLD2, and HOLD3 are for hold positions 1 through 3, HOLD4 is for hold positions 4 through 8. A call hold recall time of 10 to 240 seconds may be programmed (default is 30).

Conditions

The following conditions apply to this feature:

- A HOLD key LED is on when it has a call on hold.
- If music is available, the held party receives Music-on-Hold while on hold, even after the party starts alerting the console after a hold time-out. If no music is provided, the caller hears silence rather than ringback.
- When the call hold timeout occurs, the call alerts the console at that HOLD key - the call does not appear at any LDN position nor does it recall. The calls waiting display is updated.
- The console can selectively answer the held call by selecting individual HOLD keys.
- Non-busy extensions cannot be placed on hold.
- Conference calls cannot be placed on hold.
- If the source has a call on consultation hold, the call on hold cannot be retrieved.
- If the attendant with a call on hold is talking to a destination, and presses the HOLD key to retrieve the call, the destination party is placed on consultation hold and becomes the source. The held call is connected to the attendant and becomes the destination. The attendant can switch between parties by pressing the SRC and DEST keys on the console, or can connect the parties by pressing the RELEASE key.
- If the attendant is visually impaired and unable to see the HOLD key LED, enable COS 124 - Attendant Hold Position Security. When this class of service is enabled, an error beep sounds if the attendant attempts to put a call on hold by pressing a HOLD key that already has a party on hold.

Programming

Program COS Option 116 (Attendant-Timed Recall - Hold) in the console's COS to set the time-out period (default time is 30 seconds).

Enable COS Option 124 - Attendant Hold Position Security for consoles that require an error beep to sound when the attendant presses a HOLD key that already has a party on hold.

Operation

To put a call on hold at the console:

- Press the ANSWER key when call rings console.
- Press an idle HOLD key (1-4+); call is put on hold at this HOLD key.

To retrieve a call on hold at the console:

- Press the HOLD key to speak with the call on hold.

Note: If the call has been recalled by a call hold time-out, the HOLD key indicator flashes.

If the call is to be retrieved before a time-out, the attendant may press the HOLD key where the call is being held. By pressing the HOLD key, the call is transferred to the SOURCE, or to the DEST if there is a SOURCE already.

If HOLD key 4 is used, the user must next press one of the softkey hold slots (HOLD slots 4 through 8) for both holding and retrieving.

Attendant Implicit New Call

Description

When the attendant presses a key on the console dial pad, by default a new call is initiated. When the first key is pressed, an existing party is automatically placed on hold. At the completion of dialing, the attendant can transfer the call to the dialed destination by releasing from the call.

By pressing the TONES ON softkey (if available), this feature is temporarily disabled. See Attendant Tone Signaling.

Conditions

None.

Programming

None.

Operation

Press any dial pad key.

- Immediately, the current call (if one exists) is placed on hold, and a new call is initiated.

Attendant Individual Directory Number

Description

Each attendant console has a unique directory number identifying that console.

The directory number is in addition to the general attendant access number (usually 0) used to obtain the attendant or any LDN keys programmed at that console. A calling party has the choice of either dialing the general attendant access number, or dialing the directory number which is dedicated to a particular attendant position (useful when there is more than one attendant position).

Conditions

The following conditions apply to this feature:

- Calls to an individual directory number are not presented to other attendants (if any).
- Calls directed to the directory number appear at a default softkey position provided by the system; see Attendant Default Call Positions.

Programming

Program the directory number of each attendant position in CDE Form 07 (Console Assignments).

Operation

None.

Attendant Intercom Calls

See Intercom Calls.

Attendant Interposition Calling and Transfer

Description

In a multiple console environment, an attendant can call or transfer a call to any other attendant using the individual attendant directory number. The call is transferred in the same method as a call to an extension.

Conditions

The following condition applies to this feature:

- Since consoles cannot be put on hold, normal attendant operations (hold, swap, etc.) are not available when talking to another console.
- One console can call and talk to another by dialing the DN number but not by dialing 0.

Programming

Program the directory number of each attendant position in CDE Form 07 (Console Assignments).

Operation

When the call has been answered, dial the directory number of the attendant console to which the call is to be transferred.

While listening to ring back or when the attendant answers, press the RELEASE key to transfer the SOURCE caller to the called console.

Attendant Lockout Alarm

See Lockout Alarm.

Attendant Message Waiting Setup and Cancel

Description

The Attendant Message Waiting Setup and Cancel feature allows the attendant to inform extension users that there is a message waiting. The message waiting indication may take the form of:

- a message on a Mitel display telephone,
- a continuously flashing lamp on the extension (if equipped), or
- a distinctive ringing pattern repeated every 20 minutes. The pattern is three cycles of 3.5 ips ringing.

When the user returns and calls the attendant, the “MSW” indicator appears on the console display to indicate that there is a message waiting for that extension.

When Transparent Multi-Console Operation is used, a console may review or cancel a message waiting indication for any station; without this feature, only the console that set Message Waiting for a specific station can review or cancel it. See Attendant Transparent Multi-console Operation.

A message may also be setup and canceled from the front desk or PMS. See the Hotel/Motel Feature Package Description Package.

Use the Attendant Message Waiting Setup and Cancel feature if you know that the person you are trying to contact is out of the office. Use the Attendant Callbacks- Busy/No Answer feature if you know that the person you are trying to contact is somewhere in the office.

Conditions

The following conditions apply to this feature:

- If either message waiting COS Option is enabled, the extension rings every 20 minutes when idle, or until the message waiting is canceled. If the message waiting indication is given by a lamp, the lamp flashes (at 60 ipm).
- If the message waiting indication is given by ringing the extension, the first ring starts 10 seconds after the extension becomes idle.
- The message waiting lamp or display indicator on Mitel telephones is always lit for any message irrespective of COS options.
- If the extension has Do Not Disturb enabled, the ringing indication is not given.
- Any console within this tenant with COS Option 320 (Transparent Multi-Console Operation) enabled may cancel Message Waiting, instead of just the console that set it.

Programming

Enable COS option 232 (Message Waiting Setup - Lamp) and/or COS Option 231 (Message Waiting Setup - Bell) for each extension on which the console is to place a message.

Enable COS Option 320 (Transparent Multi-Console Operation) for each console that is to operate in Transparent Multi-Console Operation mode.

Operation

To set up Message Waiting on an extension, perform the following procedure:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the STATIONS softkey.
- Enter the extension number.
- Press the SEND MSG softkey.
- Press the EXIT softkey.

  or
- Press the FUNCTION key.
- Press the GUEST ROOM softkey.
- Enter the extension/room number softkey.
- Press the SEND MSG softkey.
- Press the EXIT softkey.

To cancel Message Waiting on an extension:

- Press the CANCEL MSG softkey.

Attendant Multi-New Call Tone

Description

If an attendant is actively engaged with an incoming call, the first call placed in the attendant call waiting queue signals the attendant with a single burst of tone. As long as there are one or more calls waiting in the queue, the attendant will continue to hear the single burst of tone at the programmed time interval. The presence of any calls waiting is also shown by the call waiting indication on the top line of the display.

Conditions

The following conditions apply to this feature:

- This feature is disabled if the attendant bell is turned off from the console. Refer to Attendant Bell Off.
- The frequency of the new call tone is controlled by COS Option 404 (Recording Failure to Hangup Timer).

Programming

Enable COS Option 106 (Attendant New Call Tone) for the console.

Enable COS Option 125 (Attendant Multi-New Call Tone).

Set the desired time on COS Option 404 (Recording Failure to Hangup Timer). The default for COS Option 404 is 30 seconds.

Operation

None.

Attendant New Call Ring

Description

If an attendant is already actively engaged with an incoming call, the first call placed in the attendant call waiting queue signals the attendant with a single burst of ringing. Subsequent calls do not alert the attendant when they are added to the queue. Their presence is shown by the call waiting indication on the top line of the display.

Conditions

The following conditions apply to this feature:

- This feature is disabled if the attendant bell is turned off from the console. Refer to Attendant Bell Off.
- This feature is disabled if TONES ON is on.

Programming

Enable COS Option 106 (Attendant New Call Tone) for the console.

Operation

None.

Attendant Night/Day Switching

Description

The attendant may select NIGHT1 service, NIGHT2 service, or DAY service via softkeys; also see Night Services.

Conditions

The following conditions apply to this feature:

- When NIGHT1 or NIGHT2 has been selected by an attendant console, the status is displayed on the right-hand side of the destination line of the display for all affected consoles.
- The console switches service for tenants controlled by that console; refer to CDE Form 06 (Tenant Night Switching Control) and to the Tenanting section.

Programming

Refer to Night Services and to Tenanting.

Operation

To switch to NIGHT1, NIGHT2, or DAY service, select the following keys:

- FUNCTION
- NIGHT 1 or NIGHT 2 or DAY SERVICE

Attendant Paging Access - Directed, Group, and All Set

Description

The attendant can access a paging zone or zones by using the PAGE key on the attendant console. Pressing the PAGE key connects the console handset directly to the zones of the paging equipment programmed for default access for the console. This overrides any extension announcement in progress. The attendant can alternatively access the paging circuit by dialing the associated access code followed by a digit 0 - 9 for the zone required (0 accesses all zones).

See also Attendant Paged Hold Access and Paging.

The attendant may also initiate pages to sets: all set, group, or directed.

- All Set Page - pages telephones (with or without PA paging)
- Group Page (and Meet Me Answer) - pages all the telephones in their paging group or in another paging group simultaneously via their telephone speakers (with or without PA paging).
- Directed Page - pages a specific telephone via its telephone speaker.

The paging party and the paged party receive a single burst of tone to indicate that a page is about to occur. When the paged set is busy or if all sets in the paged group are busy, a short burst of busy tone precedes the voice announcement.

For the Group Page with an overhead page, the attendant can use the GROUP & PA softkey. The overhead paging zone is defined with the Paging Default COS Option from the COS of the paging group members selected.

For the All Set Page with an overhead page, the attendant can use the ALL SET & PA softkey. The overhead paging zone is defined with the Paging Zones programmed in COS Options 303-311 from the COS of the paging party.

See also Paging - Telephones.

Conditions

The following conditions apply to this feature:

- If the attendant is to connect to specific zones, program an access code in Feature 13 (Paging Access to Specific Zones) in CDE Form 02 (Feature Access Codes).
- Any extension(s) using the paging zone(s) that the console is attempting to access is overridden and removed from the pager and given busy tone.
- The console bell is turned off while the PAGE key is held down.
- If System Option 03 (Single Paging Amplifier) is enabled then the attendant cannot override the current pager user(s).
- A console cannot override another console in a paging zone.
- For paging on all zones simultaneously, all zones must be either not in use or the console must be able to override the user of the zone(s).
- While the PAGE key is held down, the rest of the console keys are non-operational.
- When on a pager that has been accessed by a feature access code, tone signaling can be turned on using the TONES ON softkey. See Attendant Tone Signaling.
- Paging telephones with an overhead page requires Feature Level 3 or greater in System Option 102. Systems with Feature Levels 2 or less, will not have the GROUP & PA and ALL SET & PA softkeys.
- The GROUP & PA softkey on the attendant console pages the sets in the paging group and pages the overhead paging zones defined in COS Option 312, Paging Default, from the COS of the members of the paging group selected. The default zone can be one of 0 through to 9 with 0 enabling all zones. All the members of the paging group should be in the same COS group.
- The ALL SET & PA softkey on the attendant console pages all of the telephones plus the overhead paging zones defined in COS Options 303 to 311 from the COS belonging to the attendant console.

Programming

Enable one or more of the following COS Options for the console in CDE Form 03 (COS Define) as shown in the following table. The COS Options 303 to 311 also defines the overhead paging zones with the All Set Page.

COS Option Number	Description
303	Paging Zone 1 Access
304	Paging Zone 2 Access
305	Paging Zone 3 Access
306	Paging Zone 4 Access
307	Paging Zone 5 Access
308	Paging Zone 6 Access
309	Paging Zone 7 Access
310	Paging Zone 8 Access
311	Paging Zone 9 Access
312	Paging Default (0 to 9) (0 Gives All Enabled Zones)

For access to the default zone, assign an access code to Feature 12 (Paging Access to Default Zone) in CDE Form 02 (Feature Access Codes).

Assign an access code to Feature 13 (Paging Access to Specific Zones) in CDE Form 02 (Feature Access Codes) for access to zones other than the default zones.

Program Feature Level 3 or greater for System Option 102 in CDE Form 4. This feature level allows attendants to specify a paging group for a group page and enables the PA paging with a group page. Systems with Feature Levels 2 or less, will not have the GROUP & PA softkey and ALL SET & PA softkey for the attendant console.

For the telephones in the paging groups, enable COS Option 312 - Paging Default and specify the default paging zone for the paging group members. Ensure that all paging group members are in the same COS, to ensure that the PA zone is the same zone for all of the paging group members.

Operation

To connect to the default paging zone, hold down the PAGE key. The connection remains until the PAGE key is released.

To connect to a paging zone other than the default zone, dial the ‘Paging Access To Specific Zones’ access code, followed by the desired paging zone number (1-9).

While the console’s default page zone is in use, the PAGE key LED is lit on all consoles for which the same default zone applies.

If the paging zone(s) cannot be accessed, busy tone is returned.

To initiate a Direct, Group, or All Set Page from a Console:

- Press the Set Page key.
- For a Directed page, dial the extension number
  For a Group page, press the Group Page softkey, enter a 2-digit paging group number.
  For an All Set page, press the All Set Page softkey.
  For a Group page and a PA page, press the Group & PA softkey, and enter the paging group number.
  For an All Set page and a PA page, press the All Set & PA softkey.
- Broadcast the page message.

To initiate a Direct, Group, or All Set Page from a Console:

- Press the Set Page key.
- For a Directed page, dial the extension number
  For a Group page, press the Group Page softkey, enter a 2-digit paging group number.
  For an All Set page, press the All Set Page softkey.
  For a Group page and a PA page, press the Group & PA softkey, and enter the paging group number.
  For an All Set page and a PA page, press the All Set & PA softkey.
- Broadcast the page message.

Note: If the Set Page key on the console is programmed as a macro key, then dial the Direct Paging feature access code. The paging softkeys will appear and then you can press the Group Page softkey or the All Set Page softkey.

The Attendant can also page other sets on a remote controller:

To initiate a Directed Page from a Console to a set on another controller:

- Dial the Key System - Direct Paging feature access code.
- Dial the ARS digit(s) to connect to the other controller (if required).

- Dial the extension number.

To initiate a Group Page from a Console to a paging group on another controller:

- Dial the ARS digit(s) to connect to the remote controller.
- Dial 9
- Dial the two-digit paging group number.

To initiate an All Sets Page from a Console to all sets on another controller:

- Dial the ARS digit(s) to connect to the other controller.
- Dial the Key System - Direct Paging feature access code.
- Dial 0.

Attendant Paged Hold Access

Description

The attendant can put a party on hold and page for someone to pick up the call from the attendant hold position. When paging the called party, the attendant announces the digit string that must be dialed to pick up the call.

Also see Attendant Paging Access.

Conditions

The following conditions apply to this feature:

- Consoles, industry-standard telephones, Mitel telephones, DISA trunks, and TIE trunks can pickup the held calls.
- The party picking up the call must be able to connect to the held party; see Device Interconnection Control.
- The attendant cannot pick up a held call if it has a source party.
- An extension cannot pick up the held party if the extension has a consultation hold in progress and the held party has COS Option 233 (Never A Consultee) enabled.
- A station or Mitel telephone (with a consultation hold in progress) cannot pick up the held trunk if the station or Mitel telephone has COS Option 214, (Cannot dial a trunk after flashing) enabled.
- A station or Mitel telephone on a conference with a trunk on consultation hold cannot pick up the held trunk if the station or Mitel telephone has COS Option 215, (Cannot Dial a Trunk if Holding or in Conf With a Trunk) enabled.

Programming

Assign an access code to Feature 16 (Hold Pickup Access - Attendant Hold Slots).

Enable COS Option 225 (Hold Pickup Access - Attendant Hold Slots) in the COS of the device from which the pickup call is made.

Operation

If paging the default paging zone:

- Put the calling party on hold using one of the console HOLD slots.
  - When the attendant presses the PAGE key, the console displays the access code assigned to Feature 16 followed by two digits that identify the console, followed by n. The “n” digit represents

the hold slot number. The attendant should learn the Hold Pickup Access code; it is not displayed when specific zone paging is used.

- Page the second party, specifying the displayed number (the last number being the number of the hold slot)
  - When the second party dials the displayed number, the second and held parties are connected.
  - If the paged party does not call, the held party recalls to the attendant automatically; see Attendant Hold Positions.

If paging a zone other than the default zone:

- Put the calling party on hold using one of the console HOLD slots.
- Dial the ‘Paging Access To Specific Zones’ access code, followed by the desired zone.
- Page the second party, specifying the console’s Hold Pickup Access Code, followed by the number of the hold slot, e.g.; 677002 (where 677 is the Access Code, 00 is the Console, and 2 is the Hold Slot number).
  - The paged party dials the announced code. If the paged party does not call, the held party recalls the attendant automatically.

Attendant Phonebook

See Phonebook.

Attendant Serial Call

Description

The attendant serial call feature allows an incoming trunk call to be set as a serial call before being transferred by the attendant. After the call is finished, the serial call recalls the attendant. This allows a caller to speak to several individuals in the system without the need for transfers by the called extensions.

Conditions

The following conditions apply to this feature:

- Attendant Serial Call is available on all trunk calls for all trunk types.
- Serial call returns a trunk to the console that established the call under the following conditions:
  - The party, except a console, that is talking to a serial trunk hangs up.
  - ACD interflows a serial trunk to a DROP CALL interflow point.
  - A final ringback time out occurs for the serial trunk and it is not ringing a console, LDN or night bell.
- Transparent Multi-Console Operation has no effect on serial calls.
- The RING AGAIN softkey does not appear for a serial trunk call that recalls back to the console (however it recalls).
- A serial call that is released to a subattendant will not have its recall point changed to the subattendant telephone; see Subattendant.
- Enabling serial call clears any previous serial call setting by another console and any recall point set up to any other device; see Recall.
- Serial calls appear at the RECALL call position at the console; see Attendant Default Call Positions.

Programming

Enable COS Option 109 (Attendant Serial Call) for the console.

Operation

To establish a serial call:

- Answer an incoming trunk call.
- Press the SERIAL CALL key.
- Dial the required destination extension number.
- Press the RELEASE key.

To answer a serial recall:

- ANSWER lamp flashes and RECALL softkey appears.
- Press the ANSWER key - ANSWER lamp is lit; SER, SRC, and SERIAL CALL is displayed on console.
- The attendant is connected to the recalling trunk.

To cancel a serial recall:

- ANSWER lamp flashes and RECALL softkey appears.
- Press the ANSWER key - ANSWER lamp is lit; SER, SRC, and SERIAL CALL is displayed on the console.
- Press the SERIAL CALL key and the RELEASE key; SERIAL CALL goes out and the call is cleared.

Attendant Source Key

Description

This feature allows the attendant to press the SOURCE softkey to speak to the source party of a call, to swap between the source and destination parties or to split up a conference call.

The source party's extension number, COS, and COR are displayed on the first line of the console's LCD display and the destination party is put on consultation hold.

A party that is on consultation hold at the console does not hear music.

See Attendant Call Splitting And Swapping.

Conditions

This softkey only appears when the attendant console is connected to a multi-party call and the destination party can be put on Consultation Hold.

Programming

None.

Operation

Press the SOURCE softkey - the console is connected to the source party and the destination party is put on consultation hold.

Attendant Single Button Transfer to Voice mail

Description

The Single Button Transfer to Voice mail feature provides a voice mail key that transfers a caller to a user's voice mail. The voice mail key appears as a softkey on console.

The voice mail key functions under three different modes:

- transfer-recall mode—allows the attendant to transfer the call to the voice mailbox for an extension after receiving a recall
- direct mode (no recall)—allows the attendant to transfer the call directly to the voice mailbox for an extension
- consultation hold mode—allows the attendant to transfer the call to the voice mailbox for an extension while the call is on consultation hold

Conditions

The following conditions apply to this feature:

- Requires Feature Level 3 or greater in System Option 102.
- The voice mail softkey does not appear when COS Option 269, Record a Call: Start Recording Automatically, is enabled on the COS for the console.
- This feature is for embedded voice mail and DNIC voice mail. Only embedded voice mail and DNIC voice mail hunt groups are entered in CDE Form 19.
- If there is not a DN programmed in Voice mail Number for This Tenant in CDE Form 19 for the tenant group of the called party, the Voice mail softkey will not appear in the Transfer - Recall mode and the user will receive a reorder tone after dialing a valid DN in the Direct (Non-recall) mode.
- COS Option 269 (Record a Call - Start Recording Automatically) must be disabled in a console's Class of Service in order to get all available voice mail softkeys to display over IP trunks.
- A mailbox must have an extension (real or phantom) associated with with it in system programming; otherwise, attempts to transfer calls to it will fail.

Programming

In CDE Form 04, System Option and Timers, enable Feature Level 3 in System Option 102.

In CDE Form 03, COS Define, enable Option 275 (Enable Voicemail Single Key Access) in the attendant console’s COS.

In CDE Form 19, Call Rerouting Table, for Voice mail Number for This Tenant, enter the voice mail number for the Day, Night 1, and Night 2 service modes for the tenant group of the called extension number. This voice mail number is typically the master hunt group number for embedded or DNIC voice mail.

Operation

To transfer a caller to the destination party's voice mailbox after receiving a recall from the unanswered transfer

- Press the Voice mail key.

To transfer a caller directly to the destination party's voice mailbox

- Press the Voice mail key.
- Dial the required destination extension number.

To transfer a caller to the destination party's voice mailbox while on consultation hold

- Press the Voice mail key.

Attendant Timed Recall

Description

This feature automatically alerts the attendant when a call extended through the console or a call held at the console has not been answered within the preselected time.

Selectable recall times include:

COS Option	Name	Timer Range
115	Attendant-Timed Recall - No Answer	5 to 240 s (0 = disable)
116	Attendant-Timed Recall - Hold	10 to 240 s
117	Attendant-Timed Recall - Campon	5 to 240 s (0 = disable)

For full details of Recall, see Recall. Also see Attendant Transparent Multi-console Operation.

Conditions

The following conditions apply to this feature:

- A value of 0 for COS Option 115 and COS Option 117 disables the Recall; however, the final ring timeout applies.
- Recalls to the console are inoperative during Night Service unless the console is the night answer point.
- If COS Option 242 (Repeated Camp-On Beeps) is enabled, Attendant Timed Recall (Campon) is disabled.

Programming

Select the desired recall times for COS Options 115 (Attendant-Timed Recall - NO ANSWER), 116 (Attendant-Timed Recall - HOLD), and 117 (Attendant-Timed Recall - CAMPON), in the attendant console’s COS.

COS Options 115 (Attendant-Timed Recall - NO ANSWER), 116 (Attendant-Timed Recall - HOLD), and 117 (Attendant-Timed Recall - CAMPON) apply to attendant consoles only. They do not apply to sets.

Operation

See Recall.

Attendant Tone Signaling

Description

The attendant console usually does not transmit DTMF tones. Applications such as voice mail, however, may require the attendant to transmit tones. The attendant tone signaling feature allows the console to transmit DTMF tones during a call.

Conditions

The following conditions apply to this feature:

- The TONES ON/OFF softkey does not appear if the Attendant is receiving an Audible Lockout Alarm (if enabled).
- The tones remain on unless turned off or the attendant places a party on hold, or retrieves a party from hold, or goes idle.
- The key appears if the console is talking to a trunk, Mitel telephone or industry-standard telephone or the console is on the pager through dialing a pager access code (not through the PAGE key).
- Dialing on the keypad by the attendant usually starts a new call implicitly, and puts the current party on consultation hold as the source party. With TONES ON enabled, this feature is disabled and either tones must be turned off or the call must be put on hold at one of the attendant hold positions before another call can be started.
- New Call Tone is disabled while TONES ON is enabled.

Programming

Enable COS Option 119 (Attendant Tone Signaling) for the attendant console.

Operation

Do the following:

- During a call, press the TONES ON key.
- Send DTMF tones (by pressing dial pad keys).
- Press TONES OFF to terminate DTMF signaling.

Attendant Training Jacks

Description

Training jacks are provided on the attendant console for use by a supervisor or trainer who is training a new attendant. Each console is equipped with two attendant jacks; either jack may be used by the attendant, while the other provides a monitoring, supervisor, or training function.

Conditions

Removal of both headsets and/or handsets does not automatically switch the console into Night Service.

Programming

None.

Operation

Console or system operation does not change in any way.

Attendant Transfer To Campon

Description

This feature allows the attendant to connect calls to a busy extension, hunt group or trunk group for automatic completion when the called busy party becomes free. The attendant itself cannot camp on but can transfer calls into campon; see Campon. For details of recall from campon, see Recall.

Conditions

The following conditions apply to this feature:

- Calls that are not completed within the campon time-out will recall; see Recall.
- The transferred party must be able to connect to the busy party; see Device Interconnection Control.
- A transfer cannot be made to a locked-out extension.
- To permit the attendant console to transfer calls into campon, COS Option 266 (Campon Before Forward on Busy) must be disabled. This condition does not apply to subattendant consoles, which can transfer calls to campon even if COS Option 266 is enabled. See Campon Priority Over Call Forward Busy.
- To permit the attendant console to transfer calls to campon to a busy party, the busy party must not have call forward busy enabled in CDE Form 19 (Call Forward Busy Number For This Tenant option), or through a Superkey/FAC.

Programming

Specify a time-out period in COS Option 117 (Attendant-Timed Recall - CAMPON); (default time is 30 seconds). A value of 0 disables recall from campon to the attendant.

Enable COS Option 301 (Campon) for the console to transfer to internal devices; enable COS Option 237 (Outgoing Trunk Campon) for transfers to busy trunk groups.

Operation

To camp a call onto a busy number:

- Attempt a call to a busy extension.
- Press the RELEASE key.
  - this automatically camps on the calling party to the busy number. If the transfer is not allowed then a beep tone is heard, CANT is displayed on the console, and no transfer is done.

Attendant Transparent Multi-Console Operation

Description

The Attendant Transparent Multi-Console Operation feature allows some features to apply to a group of consoles within a tenant. Messages set by any console or telephone in this group can be read or canceled by any consoles within this group.

Conditions

The following conditions apply to this feature:

- Transparent Multi-Console Operation must be enabled for each participating console.
- When a Mitel 5020, 5220, 5224, 5324, or 5330 IP Phone, SUPERSET 4025, SUPERSET 4125, or SUPERSET 420 telephone, Family 1 or Family 3 telephone user presses the CALL softkey to call the console in response to a message received, if the message was left by a console with the Transparent Multi-Console Operation feature enabled, the call is turned into a normal Dial 0 call and routes to the Dial 0 point of the Mitel telephone.
- Recalls to a console can be answered from any other console that the console is allowed to connect with. Form 05 (Tenant Interconnection Table) specifies which tenant groups may be connected together.

Programming

Enable COS Option 320 (Transparent Multi-Console Operation) for each console within this tenant that is to be a member of the group.

Assign access codes to the Dial 0 entry in CDE Form 19 (Call Rerouting Table) for the tenant of the caller. To have the calls ring the attendant in the group, the access code must be an LDN on the consoles in the console group.

Operation

One attendant can read or cancel another attendant's message waiting indications.

Calls that have been extended by one attendant will recall at multiple attendant consoles.

Attendant Trunk Busy-Out

Description

The attendant may busy-out a trunk to prevent access to the trunk, and may remove the busy condition as required. If the Trunk Busy-Out Enable option is not selected, the attendant may still access individual trunks, but is unable to force them into a busy condition.

Conditions

The following conditions apply to this feature:

- As with station and Mitel telephones, if the trunk is not idle when the busy is attempted, the busy out will be pending and will be processed when the trunk becomes idle.
- The console must be able to connect to the trunk; see Device Interconnection Control.

Programming

Enable COS Option 114 (Attendant Trunk Busy-Out) for the console.

Operation

To busy out a trunk, perform the following procedure:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the TRUNKS softkey.
- Enter trunk number.
- Press the BUSY OUT softkey.
- Press the SET softkey.

To return a trunk to service, perform the following procedure:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the TRUNKS softkey.
- Enter trunk number.
- Press the BUSY OUT softkey.
- Press the CLEAR softkey.

Attendant Trunk Group Status Display

Description

This feature allows the attendant to check the status of the PBX trunk groups. An attendant can display the status of all 50 trunk groups, although a maximum of 20 trunk groups are shown in each screen display. A black square, under a trunk group number, indicates that all the trunks in that group are busy.

The attendant can display the trunk status whether the console is idle or busy. If this feature is activated while the console is idle, the display is refreshed approximately every 5 seconds.

Conditions

None.

Programming

None.

Operation

To display the trunk group status, press the following keys:

- Press the TRUNK GROUP key. The display shows the status of the first 19 trunk groups.
- Press the MORE softkey to display the status of the next 20 trunk groups.
- Press the MORE softkey to display the status of the remaining 11 trunk groups.
- Press the EXIT softkey.

Auto - Answer

Description

When the Auto-Answer feature is active, incoming calls ring briefly, then the set answers the call in Handsfree mode; see Handsfree Operation. After the caller hangs up, a short burst of tone is heard over the Mitel telephone's speaker and the set goes idle. Call origination is not affected.

This feature is available on Family 1, Family 3, Family 4, Family 5, Mitel 5207, SUPERSET 410, and SUPERSET 3DN telephones; it is also available on Family 2 telephones when a headset is attached.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- A directed page to a Family 1, Family 4, Family 5, or SUPERSET 410 telephone in auto-answer mode is handled like a normal directed page: the set rings briefly, then the set speaker turns on, but the set microphone remains off.
- There are no restrictions on the connection of Auto-Answer Mitel telephones and loop start CO and DISA trunks.
- Auto-Answer is not available on industry-standard telephones and Family 2 telephones.
- Auto-Answer is only available on a Family 3 telephone if it is operating in Headset Operation mode.

- When a call arrives at the set, the set does not warble. Instead the user hears a short tone through the speaker. The user hears another short burst of tone when the call is completed.
- If Auto-Answer is used with the headset feature then the tones are heard in the headset and not through the speaker.
- The feature prevents calls directed at the prime line of the Mitel telephone from ringing other appearances of the prime line because the call is automatically answered. If the call is not automatically answered (the prime appearance or the Mitel telephone is busy) then the other appearances ring if possible.
- The feature only operates for calls directed at the prime line of a Mitel telephone.
- The feature prevents any other line appearances on the set from causing the set to ring. Instead, the new call ring is used to notify the set user of new calls on other lines.
- Auto-Answer is ignored when callbacks, recalls and hold timeout recalls ring the Mitel telephone.
- In ACD, agents can be forced into Auto-Answer mode upon login; refer to the ACD TELEMARKETER Application Package.
- If Auto-Answer is activated at a set and if the prime line of the set is busy, calls to any other line appearances on the set will not be indicated with standard ringing. Instead, the calls will be indicated by New Call Ring (a single burst of ringing).
- Do not enable Auto-Answer on telephones that are programmed to receive a reminder or wake-up call. If Auto-Answer is enabled on the telephone, the reminder call will not be presented to the set.
- On a SUPERSET 4150 telephone, the Ring Adjust softkey is not available when the Auto Answer feature is enabled.
- If the telephone has a MUTE key, it can be used to turn off the handsfree microphone during a conversation.

Programming

Enable COS Option 600 (Telephone Auto-Answer) in the telephone's COS.

To provide access code activation of Auto-Answer, program a Feature 02 (Auto-Answer) access code in CDE Form 02 (Feature Access Codes).

To provide feature key activation of Auto-Answer at Mitel telephones, program an AUTO-ANSWER feature key (See Feature Keys.)

To enable Auto-Answer for a Mitel telephone that is equipped with a headset, enable COS Option 600 (Telephone - Auto-Answer) and COS Option 612 (Telephone - Headset Operation) in the telephone's COS.

Operation

Operation varies depending on the type of telephone as described below.

Family 4 Telephones:

To activate Auto-Answer:

- Press SUPERKEY.
- Press the NO softkey until AUTO ANSWER? appears in the display.
- Press the TURNON softkey. AUTO ANSWER ON appears briefly in the display. The display returns to showing the date and time.

To deactivate Auto-Answer:

- Press SUPERKEY.
- Press the NO softkey until AUTO ANSWER? appears in the display.

- Press the TURNOFF softkey. AUTO ANSWER OFF appears briefly in the display. The display returns to showing the date and time.

Note: You can also use the AUTO ANSWER feature key to turn Auto-Answer on and off.

Family 5 Telephones:

- Press the key you have programmed as **Auto Answer**. The key is illuminated and all incoming calls are answered in Handsfree mode.

Family 1 and Family 3 Telephones, and SUPERSET 3DN and SUPERSET 410 Telephones:

To activate Auto-Answer:

- Press the AUTO-ANSWER feature key. The adjacent LCD indicator darkens, or key LED illuminates, or,
- Dial the Auto-Answer feature access code, followed by the digit “1”. If a feature key has been programmed, the adjacent LCD indicator darkens or key LED extinguishes.

To deactivate Auto-Answer:

- Press the AUTO-ANSWER feature key. The adjacent LCD indicator clears, or key LED extinguishes, or,
- Dial the Auto-Answer feature access code, followed by the digit “2”. If a feature key has been programmed, the adjacent LCD indicator clears or key LED extinguishes.

Telephones with a MICROPHONE Key:

To disable the handsfree microphone during a call:

- Inform the caller, then press the MICROPHONE key (the LED turns off).

To re-enable the microphone:

- Press the MICROPHONE key or lift the handset (the LED turns on).

Telephones with a MUTE Key:

To disable the handsfree microphone during a call:

- Inform the caller, then press the MUTE key (the LED turns on).

To re-enable the handsfree microphone during a call:

- Press the MUTE key or lift the handset (the LED turns off).

Auto - Hold

Description

A Mitel telephone user automatically places a call on hold when another Line Select on the telephone is pressed. When this is not desirable, a COS option can be programmed which allows a call to be placed on hold only by pressing the HOLD key.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Auto-Hold is not available on Family 2 telephones or on industry-standard telephones.
- When this feature is disabled, calls can only be placed on hold by pressing the HOLD key.
- See Hold for conditions on putting a call on hold. If the hold is not allowed then the Line Select is ignored.
- When this feature is disabled, and another line is selected, the call on the current line is dropped.
- The auto-hold only occurs when a Line Select key is pressed while dialing or talking.
- If a Line Select key is pressed when dialing, if there is a consultation hold in progress then that party is automatically placed on hold.

Programming

Enable COS Option 601 (Telephone - Auto-Hold Disable) to disable the Auto-Hold feature.

Operation

Establish a call.

Press any Speedcall key or Line Select key; the call is placed on hold. If a Speedcall key is pressed, the system prompts for a line to use to start the Speedcall call.

Auto Latch Microphone

Description

Auto Latch Microphone allows a phone's handsfree microphone to automatically turn on or off when receiving a direct page or an Intercom call. The called party is alerted as follows:

- If the set has a MICROPHONE key, the Microphone LED will flash to indicate that the handsfree microphone is on.
- If the set has a MUTE key, the Mute LED will flash rapidly to indicate that the handsfree microphone is on.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

- This feature is available only with a Directed Page or Intercom Call, not a Group Page or All Set Page.
- See Intercom Call for interactions with the Auto Latch Microphone feature.
- If COS Option 278 (Intercom Mode) is enabled, then COS Option 683 (Key System - Direct Paging Handsfree Answerback) must be disabled to allow users to respond to handsfree intercom calls.

Programming

Disable COS Option 683, Key System - Direct Paging Handsfree Answerback, in Form 03, Class of Service Options, to allow paged phones to use this feature.

Operation

Family 1 Telephones:

To set the handsfree microphone to automatically turn on or off when receiving a page or Intercom call

- Press SUPERKEY.
- Press **More** until "Auto Latch Mic" appears.
- Press **AutoLatch Mic**.
- Press **TurnOn** or **TurnOff**.

Family 3, Family 4 and Mitel 5207 IP Telephones:

To set the handsfree microphone to automatically turn on or off when receiving a page or Intercom call

- Press SUPERKEY.
- Press **No** until "Auto Latch Mic?" appears.
- Press **TurnOn** or **TurnOff**.

Telephones with a MICROPHONE Key:

To disable the handsfree microphone during a call:

- Inform the caller, then press the MICROPHONE key (the LED turns off).

To re-enable the handsfree microphone:

- Press the MICROPHONE key or lift the handset (the LED turns on).

Telephones with a MUTE Key:

To disable the handsfree microphone during a call:

- Inform the caller, then press the MUTE key (the LED turns on).

To re-enable the handsfree microphone:

- Press the MUTE key or lift the handset (the LED turns off).

Family 5 Telephones:

- Press the key you have programmed as **Auto Latch Mic**. The LED turns on. Note: This feature is only available on Family 5 telephones programmed as subattendants.

Automated Attendant

Description

The automated attendant feature connects incoming calls from a DTMF telephone to a recording and a receiver. The recording directs the caller to dial one or more digits to be routed to a specific answering point, such as sales, service, parts, or general office. Once a digit has been dialed, the system can add prefix digits in front of the dialed digit to provide a valid extension number, a hunt group number, a system abbreviated dial number, or a feature access code, to route the call to a specific answering point. If a digit is not dialed (or cannot be dialed because the incoming telephone is not DTMF) the incoming caller is routed to a default answering point at the completion of the recording.

For details refer to the Automated Attendant Application Package.

Conditions

Refer to the Automated Attendant Application Package.

Automated Attendant is a purchased option.

Programming

The following System Options apply:

- 106 - Automated Attendant
- 59 - Receivers Reserved for Non-Auto-Attendant Use.

Refer to the Automated Attendant Application Package.

Operation

None.

Automatic Call Distribution (ACD)

Description

Automatic Call Distribution (ACD) is a method of distributing calls evenly among trained operators (Agents). Refer to the ACD TELEMARKETER Application Package, for more information.

Conditions

- Refer to the ACD TELEMARKETER Application Package.
- Automatic Call Distribution is a purchased option.
- A silent monitor (either a one-time monitor or a continuous monitor) is not permitted when an agent is on a two-party call with an embedded voice mail or DNIC voice mail set, and when COS option 267 and system option 97 are enabled.

Programming

The following System Options apply:

- 104 - Maximum ACD Agents
- 42 - ACD Silent Monitoring
- 43 - ACD Silent Monitoring Beeps
- 44 - ACD Reports.

The following COS Options apply:

- 650 - ACD Agent Template
- 651 - ACD Supervisor Template
- 652 - ACD Senior Supervisor Template
- 653 - ACD Agent Always Auto-Answer
- 654 - ACD Display Path Always
- 655 - ACD Allow Continuous Monitor of Agent
- 812 - Loop Start Trunk to ACD Path Connect.

The following Feature Access Codes apply:

- 44 - ACD Login/Logout
- 45 - ACD Silent Monitoring.

Refer to the ACD TELEMARKETER Application Package for further information.

Operation

Refer to the ACD TELEMARKETER Application Package.

Automatic Number Identification (ANI) on Outgoing Trunks

Description

This feature is a mechanism that allows the system to identify a calling party on an outgoing trunk. The identifying information consists of the calling party's extension number which is transmitted (tones or pulses) on the trunk, after the system has successfully dialed an external number on that trunk.

Conditions

The following conditions apply to this feature:

- The far-end must provide answer supervision and be capable of handling the ANI protocol.
- The extension numbers of station and Mitel telephones and consoles are sent.
- If the caller has forwarded to the ANI trunk then the forwarding extension's number is sent.

Programming

Enable COS Option 800 (ANI Applies) for the trunk.

Enable Circuit Descriptor option "Far End Gives Supervision" for the trunk.

The Circuit Descriptor option "DTMF" for the trunk must match the far end.

Operation

When a call is made on an ANI trunk, an answer supervision received on the ANI trunk is treated as a signal to send the calling party's extension number. During transmission of the calling extension number, the calling party is not connected to the speech path. This inhibits putting any noise (that could be mistaken for DTMF signaling) onto the line until all dialing is finished. Once the extension number has been sent, the calling party's audio will be cut through and no further signaling will occur.

Automatic Number Identification (ANI)/Dialed Number Identification Service (DNIS) on Incoming Trunks

Description

This feature allows the system to identify Automatic Number Identification (ANI) numbers and Dialed Number Identification Service (DNIS) numbers that are transmitted to the system on an incoming trunk. ANI provides the telephone number of the calling party; DNIS provides the telephone number dialed by the calling party.

The system can receive ANI and DNIS numbers on specific types of incoming non-DISA trunks. After receiving ANI and DNIS numbers for an incoming call, the system can then make the numbers available to other functions within the system. You can program the system to provide ANI and DNIS numbers to Mitel display telephones, to Mitel consoles, to SMDR printers, and to the Mitel Application Interface (MAI) platform package.

On a Mitel console, the ANI and DNIS digits for an incoming call are displayed after the call is answered. The ANI digits appear where the trunk information is normally displayed; the DNIS digits overwrite the COS and COR fields on the console display. When ANI digits are not received, the trunk information is shown instead; when DNIS digits are not received, the COS and COR fields are visible. By default, ANI and DNIS numbers are not displayed.

On Mitel display telephones, the display shows DNIS digits (the last 10 digits) while the call is unanswered. After the user answers the call, the ANI digits are displayed. Whenever ANI digits are not available for a call, the display shows the DNIS digits during both the unanswered and answered states. Whenever DNIS digits are not available, the display shows the ANI digits during both the unanswered and answered states. If neither are available, the display shows the normal trunk information. By default, ANI and DNIS numbers are not displayed.

If COS Option 613 (Display ANI Information Only) is enabled, the Mitel telephone displays the ANI digits during both the unanswered and answered states. In this case, if ANI digits are not provided, the display shows the normal trunk information.

You can program the system to record ANI and DNIS digits in the SMDR reports. The digits are recorded at the end of the trunk SMDR record. The DNIS column at the far right, shows the last 10 DNIS digits. The Dialed Digits column shows up to 17 DNIS digits. Both ANIS and DNIS numbers are recorded up to a maximum of 23 digits in length. By default, ANI and DNIS numbers are not recorded in SMDR reports.

Conditions

The following conditions apply to this feature:

- Only non-DISA trunks that are programmed with a T1 E&M, T1 E&M DISA, T1 TIE DISA, or T1 DID/TIE trunk circuit descriptor can receive ANI and DNIS digits. Ensure that the trunk's circuit descriptor has the Debounce Timer is set to 100 ms or greater.
- ISDN bays with the PRI card, Release 6.0 and greater, accept up to 17 DNIS digits.
- Incoming trunks must use DTMF signaling, and must be set to Wink Start.
- Only MCI and US Sprint standards are supported.
- If US Sprint standard is used, calls cannot be routed to extensions that have an * in their access number.
- The system waits a maximum of 300 ms for each ANI/DNIS digit. If it doesn't receive a digit within the 300 ms timeout period, the system routes the call through to the extension.
- The system uses the DNIS digits that are collected for an incoming call to route the trunk call through the system.
- The ANI and DNIS digit displays take priority over analog networking digits, trunk group names, trunk names and trunk numbers.
- If the system doesn't receive any ANI or DNIS digits for a call, then the console or set display shows the trunk information of the call instead.
- ANI and DNIS digit displays follow standard call forwarding rules (See Call Forwarding).
- If COS Option 504, SS420 Optional CLASS/ANI Display, is enabled, a SUPERSET 420 telephone will receive ANI/DNIS information about a call that is forwarded to it (rather than receiving information about the telephone that forwarded the call).
- In an ACD path environment, a call to an agent phone that comes through the path will display the path name during ringing if programmed, regardless of the status of option 613 (Display ANI only). To see ANI digits during ringing, the path name must be deleted.
- SUPERSET 4 telephones do not support ANI.
- On Mitel Consoles, tenant names and calling line identification displays take priority over ANI digits, and the following displays take priority over DNIS digits:
  - Intercept reasoning (FROM xxxx DND)
  - Forwarding information (FWD FROM xxxx).

Programming

Enable COS Option 811 (ANI/DNIS Trunk) and disable COS Option 801 (Incoming Trunk Call Rotary) for the trunk in Form 03.

In Form 13, assign the ANI/DNIS trunk with one of the following trunk circuit descriptors:

- T1 E&M

- T1 E&M DISA
- T1 DID/TIE
- T1 TIE/DISA

In the circuit descriptor options subform of Form 13, program the following options for the ANI/DNIS trunk:

- Set the “Incoming Start Type” option to “Wink”.
- Set the “Wink Timer” to the required standard (100-350 ms for MCI; 140-290 ms for US Sprint).
- Set the “Debounce Timer” to 100 ms or greater.
- Enable DTMF.

Define the ANI/DNIS trunk as a (non-dial-in) trunk in Form 14, or as a TIE or DID trunk (non-DISA) in Form 15.

To program the system to display ANI and DNIS digits on a Mitel display set or console, enable COS Option 502 (Display ANI/DNIS/CLASS Information) in the class of service for the set/console.

When COS Option 502 (Display ANI/DNIS/CLASS Information) is enabled, Mitel display telephones show the DNIS digits during the ringing state, and ANI digits during the answered state. To program a Mitel display telephone to show the ANI digits during both the unanswered and answered states, enable COS Option 613 (Display ANI Information Only) in the Class of Service for the set. By default, COS Option 613 (Display ANI Information Only) is disabled.

Enable COS Option 504, SS420 Optional CLASS/ANI Display, to display ANI/DNIS information about a call that is forwarded to a SUPERSET 420 telephone (rather than displaying information about the telephone that forwarded the call).

To program the system to display DNIS digits only, enable COS Options 502 and 229, and disable COS Option 613.

If you want ANI and DNIS digits recorded in the SMDR records, enable COS Option 806 (SMDR-Record Incoming Calls) and COS Option 814 (SMDR-Record ANI/DNIS/CLASS) in the class of service for the trunk.

Operation

None.

Automatic Route Selection (ARS)

Description

The Automatic Route Selection (ARS) feature package combines the concepts of standard ARS (selecting optimum call routes, inserting/deleting routing digits) and Toll Control (allowing/disallowing specific extensions the ability to make specific types of external/long distance calls).

Also provided are:

- a universal numbering plan capability,
- six time-of-day zones,
- three day zones for week days, and
- overlap outpulsing.

Conditions

Refer to the Automatic Route Selection and Toll Control section.

Programming

System Option 47 - ARS Unknown Digit Length Timeout applies. Refer to the Automatic Route Selection and Toll Control section, for further information.

Operation

Refer to the Automatic Route Selection and Toll Control section.

Background Music

Description

This feature permits the users to have background music played through the speaker of the telephone when idle. The Music-on-Hold source provides the music; see Music-on-Hold.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Background music is automatically temporarily suspended when a call is placed or received; it is restored after the Mitel telephone is returned to idle.
- Background music is not available if a headset is connected to the set.

Programming

Enable COS Option 602 (Telephone - Background Music) in the set's COS.

To activate and deactivate music on a Family 3, Family 4, Family 5, Mitel 5207 IP, SUPERSET 410, and SUPERSET 3DN telephones, program a MUSIC feature key. (See Feature Keys.)

Operation

Operation varies depending on the type of SUPERSET telephone:

Family 3, Family 4, Family 5, Mitel 5207, SUPERSET 410, and SUPERSET 3DN telephones:

To turn on background music:

- Press the MUSIC feature key. The adjacent LCD indicator darkens or key LED illuminates. Music is heard through the set's speaker. Adjust the set's speaker volume as desired. This adjustment does not affect the handsfree speaker volume.

To turn off background music:

- Press the MUSIC feature key. The adjacent LCD indicator, which was dark, clears or key LED extinguishes.

Family 1 Telephones:

To receive background music:

- Press the MUSIC ON softkey. Music is heard through the set's speaker. Adjust the set's speaker volume as desired. This adjustment does not affect the handsfree speaker volume.

To turn off background music:

- Press the MUSIC OFF softkey.

Broker's Call (Station Swap)

Description

Broker's Call allows the user to speak privately with two separate parties. When the user hangs up, the two parties are not connected. This feature applies to Family 2 telephones and industry-standard telephones.

Users that have this feature enabled can't transfer calls. However, a variant on Broker's Call is the Broker's Call with Transfer feature. The Broker's Call with Transfer feature enables the user to speak privately with two separate parties while allowing transfers; see Broker's Call with Transfer.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Enable COS Option 203 (Broker's Call) in the COS of industry-standard telephones, and Family 2 telephones that require the Broker's Call feature. Users of Family 1, Family 3, Family 4, Family 5, SUPERSET 410 and SUPERSET 3DN telephones perform Broker's Calls using a SWAP feature key or SWAP softkey or TRD softkey on the Symbol MiNET Wireless Phone. (see Swap-Trade Calls).
- Industry-standard and Family 2 telephones that have COS Option 203 (Broker's Call) enabled cannot set up conference calls or make call transfers.
- This feature prevents Family 2 or industry-standard telephones from putting a conference on consultation hold; reorder tone is returned if it is attempted.
- If the extension originating the Broker's Call hangs up with a party on hold, the held party recalls to the extension instead of being transferred to a third party. COS Option 403 (Trunk Recall Partial Inhibit) is an exception to this.
- An extension with the Broker's Call feature may access the Call Hold, Call Hold and Retrieve, and Paging features after flashing on a call.
- COS Option 203 (Broker's Call) and the following COS Options are mutually exclusive:

  302 (Flash-In Conference)

  252 (Broker's Call With Transfer)

  223 (Flash Disable)

  224 (Flash for Attendant).
- The following COS Options do not apply to an extension with Broker's Call enabled:

  214 (Cannot Dial a Trunk after Flashing)

  215 (Cannot Dial a Trunk if Holding or Conf With One).

Programming

Enable COS Option 203 (Broker's Call) for the extension.

Operation

Operation varies depending on the type of telephone as described below.

Industry-standard Telephones:

A normal 2-party call is established, and the extension user wishes to consult a third party:

- Flash the switchhook - dial tone is heard, and the second party is placed in consultation hold.
- Dial the number of the third party and establish a private 2-party conversation.
- To alternate between calls, flash the switchhook.

Mitel 5201 IP Phone

A normal 2-party call is established, and the extension user wishes to consult a third party:

- Press the **Trans/Conf** key - dial tone is heard, and the second party is placed in consultation hold.
- Dial the number of the third party and establish a private 2-party conversation.
- To alternate between calls, press the **Trans/Conf** key.

SUPERSET 4001 and SUPERSET 401+ Telephones:

A normal 2-party call is established, and the extension user wishes to consult a third party:

- Press the FLASH key - dial tone is heard, and the second party is placed in consultation hold.
- Dial the number of the third party and establish a private 2-party conversation.
- To alternate between calls, press the FLASH key.

Broker's Call With Transfer (Transfer With Privacy)

Description

The Broker's Call With Transfer (Transfer With Privacy) is similar to the Broker's Call feature but users are also able to transfer calls. A flash is interpreted as a Swap instead of an attempt to conference (see Broker's Call). A conference cannot be formed by an extension that has Broker's Call With Transfer enabled. However, unlike Broker's Call, when the extension goes on–hook to transfer an extension, the transfer is not prevented. As well, some consultation COS option checks are not done with the Broker's Call feature but they are done for Broker's Call With Transfer. Also see Broker's Call (Trade Calls) for general information about Broker's Calls.

This feature applies to Family 2 telephones and industry-standard telephones.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- COS Option 252 (Broker's Call With Transfer) (previously Transfer With Privacy) is mutually exclusive with the following options:
  - 302 (Flash-in Conference)
  - 224 (Flash for Attendant)
  - 223 (Flash Disable)

- 203 (Broker's Call).

- An extension with COS Option 233 (Never a Consultee) may not be consulted.
- COS Options 214 (Cannot Dial a Trunk After Flashing) and 215 (Cannot Dial a Trunk After Flashing If Holding or in Conference With One), do not apply to an extension with COS Option 252 (Broker's Call With Transfer) (previously Transfer With Privacy) enabled.

Programming

If desired for extensions enable COS Option 252 (Broker's Call With Transfer).

Operation

Industry-standard Telephones:

A normal 2-party call is established, and the extension user requires to consult a third party:

- Flash the switchhook - dial tone is heard, and the second party is placed in consultation hold.
- Dial the number of the third party and establish a private 2-party conversation.
- To alternate between calls, flash the switchhook.
- Hang up to connect both parties.

Family 2 Telephones:

A normal 2-party call is established, and the extension user requires to consult a third party:

- Press the FLASH key - dial tone is heard, and the second party is placed in consultation hold.
- Dial the number of the third party and establish a private 2-party conversation.
- To alternate between calls, press the FLASH key.
- Hang up to connect both parties.

Busy Lamp Field

Description

A Busy Lamp Field (BLF) on a Programmable Key Module (5410 PKM, 5425 PKM, 5412 PKM, 5448 PKM, PKM 48, PKM 12, or PKM) indicates the status (idle, busy, DND) of a line appearance for a device, such as, normal stations, SUPERSET or 5000 series prime lines, logical lines and trunks. Any Mitel telephone with line keys may be programmed to use BLF indicators.

The 5448 PKM provides 48 additional BLF indicators to a Mitel 5220, 5224 or 5324 IP Phone. The 5415 PKM provides 48 additional BLF indicators to a Mitel 5020 IP Phone. The PKM 48 provides 48 additional BLF indicators to a SUPERSET 4025, SUPERSET 4125, or SUPERSET 4150 telephone. You can connect up to two 5415 PKM, 5448 PKM or PKM 48 devices to either of the sets.

The 5412 PKM provides 12 additional BLF indicators to a Mitel 5220, 5224 or 5324 IP Phone. The 5410 PKM provides 12 additional BLF indicators to a Mitel 5020 IP Phone. The PKM 12 provides 12 additional BLF indicators to a SUPERSET 4025, SUPERSET 4125, or SUPERSET 4150 telephone. You can connect one 5410 PKM, 5412 PKM or PKM 12 devices to either of the sets.

The PKM provides 30 additional BLF indicators to SUPERSET 400-series telephones. You can connect up to three PKMs to a SUPERSET 400-series set. Refer to Programmable Key Module in this section for details.

Note: If you program an indicator on a PKM as a BLF, the key beside the BLF indicator is not assigned any function. You must program the key as a Direct Station Select key - see Direct Station Select in this document.

The BLF indicator of a trunk line appearance, indicates the state of the trunk (idle or not idle).

The BLF indicator of logical lines, indicates whether or not a call to the line will find the line busy. If the line is a multi-call line, the BLF indicator indicates idle until there are no free line appearances to ring. If the line is a key line, the BLF indicator indicates whether or not the line is occupied.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

Mitel 5220, 5224 or 5324 IP Phones use the 5448 PKM or 5412 PKM.

Mitel 5020 IP Phones use the 5410 PKM or 5415 PKM.

SUPERSET 4025, SUPERSET 4125, or SUPERSET 4150 telephones use the PKM 48 or PKM 12.

SUPERSET 400-series telephones use the PKM. You cannot connect a PKM to a SUPERSET 401 or SUPERSET 401+ telephone.

COS options 650, 651 and 652 in Form 3 MUST be 0.

Programming

To program BLF indications of line appearances on a programmable key module (PKM)

- Display CDE Form 09 (Desktop Device Assignments).
- Move the cursor to the bay/slot/circuit number of the telephone. An asterisk (*) appears to the left of the set type if programmable key modules are programmed for the set.
- Select the EXPAND PKM softkey. “ENTER PKM NUMBER (1-X):” appears in the command line (where X is 3 for a PKM, 1 for a PKM 12, 1 or 2 for a PKM 48).
- Type the address setting (1, 2, or 3 for the PKM; 1 for a PKM 12, 1 or 2 for a PKM 48) of the desired programmable key module in the command line.
- Select the ENTER softkey to display the Expand PKM Set Subform for the programmable key module. The Expand PKM Set Subform allows you to program the functions of the keys.
- Move the cursor to the desired key. The system displays the correct quantity of keys according to the set type. See DATASET 4122 PCB Major Components Figure for the PKM 48 numbering; see the Programmable Key Module General View Figure for the PKM numbering.
- Move the cursor to the TYPE column (use the TAB key if necessary).
- Select the BLF/DSS softkey (Type becomes “Busy Lamp”).
- Press TAB key three times to get to EXT NUM column (or four times to get to the TRK NUM column).
- Enter the number of the device to be monitored by the Line Appearance.
- Select the ENTER softkey.

Note: The keys are preceded by a 1, 2, or 3 to indicate which programmable key module is associated with the set (keys belonging to the first programmable key module are preceded by a “1”).

Operation

When answering an incoming call for one of the monitored stations, the attendant or SUPERSET or 5000 series user can scan the Busy Lamp Field on the PKM to determine the station’s current status.

Calculator

Description

Family 1 telephones provide a basic four function calculator using the telephone keypad, display, and softkeys. The calculator has 2 modes of operation - General and Programmer. Once the calculator feature is accessed, the # key may be used to switch between the two modes. The default mode is the General mode.

General Mode: In this mode of operation, the calculator allows for arithmetic operations. The telephone keypad is used as the numeric keypad; the * key is used as the decimal point key. Arithmetic operators (x, ^{3}, -, +/=), clear entry/clear (CE/CLR) and another decimal point key appear on the softkeys. The display is on the top line of the LCD display.

Programmer Mode: In this mode of operation, the calculator allows for integer arithmetic / logical operations in either decimal or hexadecimal. In this mode, each softkey has 2 functions - the active function is designated by the → pointer. To change to the second function, press the 2ND FUNCTION softkey. The display is on the top line of the LCD display. The second line of the display is the status register (nzvc - negative, zero, overflow, carry). When a bit in the status register is set, the corresponding letter is capitalized. The RADIX softkey switches between hexadecimal and decimal. The CLR softkey clears the displayed value (if there is no pending operator. Pressing the CLR softkey twice will always clear the status register. All operations are 32-bit. To enter hexadecimal characters A through F, enter the following: A=*1, B=*2, C=*3, D=*4, E=*5 and F=*6.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

This feature is available only on Family 1 telephones.

Programming

None.

Operation

The calculator feature operates as follows:

- Press SUPERKEY.
- Press CALCULATOR softkey.
- Press SUPERKEY when completed.

Call Forwarding

Description

Call Forward lets you redirect incoming calls to an alternate number. Call Forwarding is set up by the extension, by an attendant console, by a subattendant, by another extension for the extension (for example - I'm Here), or by tenant programming in CDE.

Call Forwarding destinations can be internal or external to the system. External destinations (Call Forwarding - External) are programmed by system abbreviated dial numbers, personal speedcall keys, or personal speedcalls.

The three most common types of call forwarding are

- Busy (the user is busy with a current call)
- No Answer (the user does not answer the call within a certain time period)
- Always (the user has no intention to answer any calls)

With the FOR System Option 102, Feature Level set to 1 or greater, each forwarding type may have one destination for incoming internal calls (within the SX-200 ICP) and another destination for incoming external calls (outside the SX-200 ICP). An incoming internal call may be directed to your voice mail and an incoming external call may be directed to your home.

Call Forward Types and Descriptions

Call Forward Type	Used to redirect ...
Busy (External & Internal Source)	incoming calls when your extension is busy
Busy (External Source)	incoming external calls when your extension is busy
Busy (Internal Source)	incoming internal calls when your extension is busy
No Answer (External & Internal Source)	incoming calls when you do not want to answer within a certain time period
No Answer (External Source)	incoming external calls when you do not want to answer within a certain time period
No Answer (Internal Source)	incoming internal calls when you do not want to answer within a certain time period
Always (External & Internal Source)	all calls when you have no intention of answering any calls
Always (External Source)	incoming external calls when you have no intention of answering any external calls
Always (Internal Source)	incoming internal calls when you have no intention of answering any internal calls
Busy/No Answer (External and Internal Source)	all calls when your telephone is busy or when you do not want to answer calls within a certain time period
Busy/No Answer (External Source)	incoming external calls when your telephone is busy or when you do not want to answer calls within a certain time period
Busy/No Answer (Internal Source)	incoming internal calls when your telephone is busy or when you do not want to answer calls within a certain time period

The following chart has links to the options that Call Forwarding provides.

Table: Call Forwarding Options
Call Forwarding - External (destination)
Call Forwarding - Internal/External Split (source)

Table: Call Forwarding Options
Call Forwarding - Forced Call Forward
Call Forwarding - Display Prime as Fowarder
Call Forwarding - Forward Call
Call Forwarding - I'm Here
Call Forwarding - Toggle Key
Call Forwarding - Forced Call Forward
Forward Campon

Feature Access Codes That Activate Call Forwarding Types

To activate call forwarding, the extension must dial the digits programmed for one of the following feature access codes.

- 03 - Call Forwarding - All Calls
  (split forwarding must be disabled)
- 04 - Call Forwarding - Internal Only
  (split forwarding must be enabled)
- 05 - Call Forwarding - External Only
  (split forwarding must be enabled)
- 06 - Call Forwarding - I’m Here
- 07 - Call Forwarding - Cancel I’m Here.

Condition Codes To Be Dialed Following Feature Access Codes 03 - 05

After dialing a feature access code, the user must dial the condition code to identify the condition under which the call is to be forwarded. This does not apply for Call Forward - I’m Here and Cancel I’m Here.

- 1 - Always (Unconditionally)
- 2 - Busy
- 3 - No Answer
- 4 - Busy or No Answer

Two examples of call forwarding from an extension are:

Internal calls are to be forwarded to 1701 when not answered	**External calls are to be forwarded to 1702 when busy or not answered**
Go off-hook and receive dial tone	Go off-hook and receive dial tone
Dial Call Forward Internal access code - 65	Dial Call Forward External access code - 62
Dial Condition code for No Answer - 3	Dial Condition code for Busy/No Answer - 4
Dial destination extension number - 1701	Dial destination extension number - 1702
Receive dial tone and hang up	Receive dial tone and hang up

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to the Call forwarding feature:

- With Family 1 telephones, the DELETE softkey that appears upon entering destinations for internal and external calls is dependent on the System Option 102, Feature Level option (level 1 or greater).
- With Family 4 telephones, the DEL softkey that appears in the ALWAYS, IF BUSY, and NO ANS screen is dependent on the System Option 102, Feature Level option (level 1 or greater).
- With Attendant Consoles, the TURN ALL OFF, TURN ON, SAVE/OFF, and asterisk (indicating on) that appears with Call Forwarding is dependent on the System Option 102, Feature Level option (level 1 or greater).
- Family 1, Family 3, and Family 4 telephones treat Busy/No Answer as two separate types: Busy and No Answer when using softkeys for programming.
- COS Option 260 (Internal/External Split Forwarding) must be disabled for non-split forwarding or enabled for split forwarding, according to the source of the call.
- With Split Forwarding enabled, an extension can have six forwarding destinations. Busy, No Answer, and Always can each have a destination for internal calls and one destination for external calls. Forwarding must be enabled in the set's COS.
- Call Forward - I'm Here only affects Call Forward - Always (Internal and External).
- Call Forward - I'm Here can only be canceled from the extension that set it up originally by using the Cancel I'm Here feature access code; it can also be canceled by the attendant or by the forwarded party by dialing the CLEAR ALL FEATURES access code. The forwarded party can re-program the forwarding via normal forwarding programming or via the CLEAR ALL FEATURES access code.
- An active Call Forward - Always takes priority over any other Call Forward type and the Always destination will apply. The number to which the calls are forwarded is the only party that may call the forwarding extension while Call Forwarding - Always is active. The extension may originate calls in the normal manner.
- Cancelling call forwarding destinations for internal calls using the feature access code 04, Call Forwarding - Internal Only disables all three call forward internal modes: Always, Busy and No Answer. An exception to this occurs when Call Forward - I'm Here sets up the call forwarding (internal and external destinations for Always is disabled and deleted).
- Cancelling call forwarding destinations for external calls using the feature access code 05, Call Forwarding - External Only disables all three call forward external modes: Always, Busy and No Answer. An exception to this occurs when Call Forward - I'm Here sets up the call forwarding (internal and external destinations for Always is disabled and deleted).
- Cancelling all call forwarding destinations using the feature access code 03, Call Forwarding - All Calls disables the internal and external destinations for Always, Busy and No Answer.
- Pressing the Forward All feature key when some call forwarding types are enabled disables all destinations programmed for all call forwarding types. Pressing the Forward All feature key when all the call forwarding types are disabled, enables all the destinations for all the call forwarding types that have been programmed.
- The user of Family 4 telephones in the talking mode can have the option to forward a camped-on call with the FORWARD softkey or answer the camped-on call with the TRADE softkey. The FORWARD and TRADE softkeys appear at the same time.
- Valid call forwarding destinations are: Dial 0 access code, night bell, LDN's, hunt groups (not data or modem pooling type), industry-standard telephones, SUPERSET telephones, Mitel IP telephones, consoles, ACD Paths, personal speedcall keys, system abbreviated dial numbers, or personal speedcalls.

- An extension is considered BUSY for Call Forwarding - Busy if the extension is not idle, has Do Not Disturb activated, or has its prime line appearance busy.
- COS Option 262 (Ignore Forward Busy with Free Appearance) allows Call Forward Busy to be ignored when there are any free multi-line appearances of the station or set. The next free multi-line appearance will be rung. This COS option applies only to multi-line appearances, not key line appearances or logical line appearances. COS option 262 will also cause call forward busy to be ignored on sets using a line other than prime and which do not have multi-line appearances of their prime elsewhere. This applies only to the busy portion of Call Forward Busy /No Answer. See COS Option 262 for exceptions to Call Forward Busy in the CDE section.
- Forwarding is operational for when the called party is a logical line if the line appears on only a single telephone in the system. The forwarding setting of the MITEL telephone where the line is programmed is used to forward calls.
- Forwarding can be set up to the Dial 0 access code. The destination is translated from the access code to the routing point based upon the current NIGHT/DAY service at the time of forwarding (giving NIGHT/DAY based forwarding). Priority Dial 0 can be used as well, with the feature being checked for on the forwarding extension.
- Forwarding does not apply if the calling extension is the party to which the call would be forwarded.
- Forwarding does not apply if the forwardee is an industry-standard telephone, Mitel telephone, or console and has COS Option 234 (Never a Forwardee) in its COS.
- An extension can set up forwarding to an LDN.
- Forwarding cannot be programmed from an extension to itself.
- Forwarding is ignored when callbacks are honored, except when Call Forwarding - No Answer is programmed and enabled.
- No forwarding is done if the caller is the attendant, and the forwarding destination is the Dial 0 access code, and the forwarding destination is the calling attendant itself or an LDN. This restriction on the Dial 0 access code applies for both the first and second hop of call forwarding.
- If an extension is a member of a hunt group and has forwarding enabled, the extension will still be rung if called via the hunt group (its immediate forwarding is ignored, except for Call Forward - No Answer). If it is called directly, its forwarding is honored.
- For Call Forwarding, all devices are treated as internal calls unless COS Option 709 (Follow External Call Forward) is enabled for the calling device. Therefore, trunks without COS 709 enabled are treated as "internal" devices by Call Forwarding.
- Calls will not be forwarded from an extension that has COS 200 (Account Code Forced Entry External Calls) activated, unless the forwarding destination is an ONS voice mail port.
- With multi-hop forwarding, calls may be forwarded twice (maximum) on any combination of conditions.
- When forwarding to a speedcall or abbreviated dial number, the forwarding always occurs regardless of the outcome of the call, unless to an illegal forwarding destination, in which case forwarding is ignored. For example, if toll control denies access to an external number or the forwarding is to an invalid number, no forwarding occurs. For forwarding to a personal speedcall key, the contents of the key are examined as per the validity of forwarding; if unsuitable, the forwarding is de-programmed and no forwarding occurs.
- Mitel telephone key or multi-call line appearances of extension numbers do not ring or provide visual indication of incoming calls if the extension has activated Forwarding - Always or Forwarding - Busy, if the caller is actually forwarded. See programming COS Option 262 for exceptions to Call Forward Busy.
- Forwarding is not done if the forward destination is busy (it is busy if: Do Not Disturb is activated, it is not idle, or its prime line appearance is not idle). An exception is for Subattendant Mitel telephones and enhanced answering positions. In these cases, the Mitel telephone is considered busy if there are no idle appearances of its prime line or multi-call appearances of itself; see Subattendant - Basic Function.

- When Call Forwarding - Always is activated, calls are always forwarded, even if the forward destination is busy.
- If the Mitel telephone being forwarded to has the Auto-Answer feature activated then the caller is answered with that feature; see Auto-Answer.
- Connection checking is done between the calling party and the party being forwarded to; see Device Interconnection Control.
- If forwarding is activated but for some reason the forwarding is not done, the call continues as if forwarding was not active. If the call forward -don't answer timer expires and no forwarding is performed, the recall no answer timer is started; see Recall.
- Calls forwarded to a speedcall or abbreviated dial number are treated as an external call by the system even if the entry is an internal destination.
- For Mitel display telephones and the console, the forwarding party's identity is displayed when the forwarded party rings the Mitel telephone or is answered at the console. In a multi-hop forwarding situation, the first hop forwarder is displayed.
- If a caller is a trunk or a party with a single trunk on soft hold, System Option 21 (Incoming to Outgoing Call Forward) is checked; otherwise, check the COS Option 208 (Call Forward - External) of the forwarder, console or extension is externally call forwarded, the forwarding continues only if System Option 21 (Incoming to Outgoing Call Forward) is enabled.
- Mitel telephone keys with a ring type of DELAY, or multi-call DELAY RING line appearances of extension numbers, will not ring if the extension has a DELAY RING timer active (controlled by COS option 263 Delay Ring Timer) which has a longer duration than the COS option 253 (Forwarding - Don't Answer timer). When both options are active, both timers begin at the same time. If the delay ring timer expires first, the line will ring. If the forwarding timer expires first, the delay timer is canceled and ringing will not occur.
- When an incoming DID call is made to a set or station with COS 264 (Half Fwd NA timer for DID call when VM msg on), and when that set or station has Voice mail messages waiting, then the Call Forward No Answer timer is shortened by half. This is a toll saving option, allowing users who are calling in to check their messages to hang up if the cfna timer is not shortened, knowing they have no messages to retrieve.
- When a Mitel display telephone receives a forwarded call, the display shows the number or name of the set that forwarded the call.
- If COS Option 258 (Display Prime Line as Forwarder) is disabled, the logical line will appear as the forwarder for all types of forwarding.
- Trunk calls forwarded by a set, subattendant, or console with COS Option 321 (Ignore Call Forward After Transfer) enabled will not follow the destination station's forwarding programming and will instead recall the forwarding station.
- If call forwarding is enabled at a set, you hear broken dial tone followed by regular dial tone when you go off-hook.
- For Call Forwarding - No Answer, a caller camps on to the forwarding destination if allowed to campon; see Campon for audio details.
- Call Forwarding - No Answer has no effect when a Mitel telephone answers the call using the Auto-Answer feature.
- Programming of tenant-based call forwarding via CDE requires requires Feature Level 3 or greater.
- Tenant-based call forwarding destinations do not appear on the telephone display.
- Tenant-based call forwarding has no affect on the Forward Call and I'm Here call forwarding features.
- COS Option 206 (Call Forwarding - Busy), COS Option 207 (Call Forwarding - No Answer), COS Option 208 (Call Forwarding - External), and COS Option 209 (Call Forwarding - Always) have no affect on the tenant-based call forwarding.
- Programming for tenant-based call forwarding will be ignored with Feature Levels 2 or less.

- Call forwarding of Intercom calls to phones with COS Option 278 (Intercom Mode) enabled is ignored when the forwarding destination is another extension. This is to prevent unintended (and potentially embarrassing) calls to extensions that have Handsfree Answerback to an Intercom enabled.

Programming

Assign access codes to the desired types of forwarding in CDE Form 02 (Feature Access Codes):

- Feature Access Code 03 (Call Forwarding - All Calls) *
- Feature Access Code 04 (Call Forwarding - Internal Only) **
- Feature Access Code 05 (Call Forwarding - External Only) **
- Feature Access Code 06 (Call Forwarding - I'm Here)
- Feature Access Code 07 (Call Forwarding - Cancel I'm Here)

  * Available only if COS Option 260 (Internal / External Split Call Forward) is disabled.

  ** Available only if COS Option 260 (Internal / External Split Call Forward) is enabled.

Programming is done in two steps:

- Select one of five types of Call Forwarding - internal, external, both (int + ext), I'm Here, and Cancel I'm Here.
- Dial the feature access code.
- Dial one of four Condition codes under which a call is forwarded (does not apply for Call Forward - I'm Here and Cancel I'm Here):
  1 - always, 2 - busy, 3 - no answer, 4 - busy/no answer.

Enable System Option 21 (Incoming to Outgoing Call Forward) to allow trunks to be call forwarded externally.

Enable one or more of the following COS options as required:

- 206 Call Forwarding - Busy
- 207 Call Forwarding - No Answer
- 208 Call Forwarding - External
- 209 Call Forwarding - Always
- 210 Call Forwarding Inhibit on Dial-In trunks
- 222 Call Forwarding - Inhibit on Hold Timeout
- 234 Never a Forwardee
- 253 Call Forwarding - No Answer Timer
- 258 Display Prime As Forwarder
- 260 Internal / External Split Call Forward
- 262 Ignore Forward Busy with Free Appearance
- 263 Delay Ring Timer (2 - 6 Rings)
- 264 Half Fwd NA timer for DID call when VM msg on
- 266 Camp-on before Forward on Busy
- 321 Ignore Call Forward After Transfer

Program feature keys on all the Mitel telephones. The feature keys are Forward Always, Forward Busy, Forward No Answer, Forward Busy/No Answer and Forward All. The SUPERSET 3DN and SUPERSET 4DN only support the Forward All feature key. See Feature Keys.

Enable the FOR System option 102, Feature Level to level 1 or greater to receive multiple call forwarding options.

For programming common call forwarding destinations for Busy and No Answer, use CDE Form 19 for the sets that are in the same tenant group. Tenant-based call forwarding requires Feature Level 3 or greater. In CDE Form 19, scroll to the following options: Call Forward Busy Number For This Tenant and Call Forward No Answer Number For This Tenant. Enter the DN for the call forwarding destinations for each of the call forwarding options (Busy and No Answer) for the tenant group. The destination number programmed in the DAY column will automatically appear in the N1 and N2 columns.

Operation

Operation varies depending upon the device type as described below.

Family 2 and Industry-standard Telephones:

Note: Mitel display telephones can also use this procedure.

To set up Call Forwarding (Busy, No Answer, Busy/No Answer, or Always)

- Lift the handset - wait for dial tone.
- If split forwarding is disabled, dial the access code for Call Forwarding - All Calls. If split forwarding is enabled, dial the access code for Call Forwarding - Internal Only or Call Forwarding - External Only.
- Dial the Condition code for Busy, No Answer, Busy/No Answer, or Always.
- Dial the number to which calls are to be forwarded.
- Dial tone returns.
- Hang up - the extension is available for normal use.

To cancel Forwarding (all types):

- Lift the handset - wait for dial tone.
- Dial the Call Forwarding - All Calls, Call Forwarding - Internal Only , or Call Forwarding - External Only feature access code.
- Hang up - the forwarding is canceled.

or

- Lift the handset - wait for dial tone.
- Dial the Clear All Features access code (see Clear All Features). This also cancels all forwarding at the telephone.

SUPERSET 3DN and SUPERSET 410 Telephones:

To program Call Forwarding - (Busy, No Answer, Busy/No Answer, or Always):

- Follow the same procedure as for industry-standard telephones; it is also possible to dial the codes and extension destination directly on the keypad and then press the SPEAKER ON/OFF key to go on-hook.

To activate or deactivate forwarding already programmed:

- Press the Forward Always, Forward Busy, Forward No Answer, Forward Busy/No Answer, or Forward All feature key, if programmed (the SUPERSET 3DN only supports the Forward All feature key. The LCD indicator is dark or key LED illuminated (activated) to indicate forwarding active, clear or key LED extinguished to indicate not active.

Family 3 and Family 4 Telephones (with split forwarding disabled):

To program Call Forwarding (Busy, No Answer, Busy/No Answer, or Always)

Note: Split forwarding is disabled.

- Press **Superkey**.
- Press **No** until CALL FORWARDING? appears in the display.
- Press **Yes**. The display shows the ALWAYS screen.

- If you want to program Always press **Program**, or **Change** and then **Program** (the Change softkey appears if there is a destination programmed). If you want to program another call forwarding type, press **Next** to get to the desired call forwarding type and press **Program** when the desired screen appears.
- Enter the destination number. If an invalid number is entered, the display shows INVALID. Use the ← softkey (the * key on a Mitel 5010 and 5215 IP Phones, and SUPERSET 4015 telephone) to back space and erase the entry. Enter the correct number (can be a speedcall key, if programmed).
- Press **Save** (down arrow on a Family 3 telephones). The type of forwarding, the forwarding destination and an * appear momentarily in the display. The asterisk indicates that the call forwarding is enabled. The display then returns to another forwarding type.

To deactivate Call Forwarding (Busy, No Answer, Busy/No Answer, or Always)

- Press the Forward All feature key if you want to deactivate all Call Forwarding, or press the Forward Always, Forward Busy, Forward No Answer, or Forward Busy/No Answer feature key to be more specific. The indicator beside the key turns off to indicate that call forwarding is deactivated.
  OR, if a feature key isn't programmed,
- Press **Superkey**.
- Press **No** until CALL FORWARDING? appears in the display.
- Press **Yes**.The display shows the ALWAYS screen.
- If you want to deactivate Always, press **Change** and then **TurnOff**. The asterisk disappears next to the destination to indicate that this call forwarding destination is deactivated.
- If you want to deactivate another type of forwarding, press **Next** until you reach the required display and press **Change** and then **TurnOff**.The asterisk disappears next to the destination to indicate that this call forwarding destination is deactivated.
- Press **Superkey** to return to normal display.

To activate Call Forwarding (already programmed):

- Press the Forward All, Forward Always, Forward Busy, Forward No Answer, or Forward Busy/No Answer feature key. The indicator beside the key turns on to indicate that Call Forwarding is activated. (Refer to Feature Keys for instructions on programming this feature key).
  or, if a feature key isn't programmed,
- Press **Superkey**.
- Press **No** until CALL FORWARDING? appears in the display.
- Press **Yes**. The display shows the ALWAYS screen.
- If you want to activate Always, press **Change** and then **TurnOn**.
- If you want to activate another call forwarding type press **Next** until the desired type appears and press **Change** and then **TurnOn**. An asterisk appears next to the destination to indicate that this call forwarding destination is activated.
- Press **Superkey** to return to normal display.

To display the current forwarding type and destination

Note: On a Family 3 telephone, press the dial pad * key instead of the YES softkey, and press the dial pad # key instead of the NO softkey.

- Press **Superkey**.
- Press **No** until CALL FORWARDING? appears in the display.
- Press **Yes**. The display shows the ALWAYS screen. The screen shows the destination if programmed and an asterisk if the forwarding mode is active.
- If you want to display the Busy or No Answer forwarding type, press **Next** until you reach the desired screen.
- Press **Superkey** to return to normal display.

To delete Call Forwarding

Note: Deleting a destination on a call forward type is dependant on the System Option 102, Feature Level option being set to level 1 or greater. Otherwise you may only de-activate the destination number. Family 3 telephones do not support the delete softkey.

- Press **Superkey**.
- Press **No** until CALL FORWARDING? appears in the display.
- Press **Yes**. The display shows the ALWAYS screen.
- If you want to delete Always, press **Del**.
- If you want to delete another call forwarding type press **Next** until the desired type appears and press **Del**.The Del softkey turns off the forwarding and deletes the destination number. The Del softkey appears only if the System Option 102, Feature Level option is set to level 1 or greater.

Family 3 and Family 4 Telephones (with split forwarding enabled):

Note: With split forwarding enabled, Call Forwarding Busy, No Answer and Always may have a destination for incoming external calls and a separate destination for incoming internal calls.

To program Call Forwarding (Busy, No Answer, Busy/No Answer, or Always):

- Press **Superkey**.
- Press **No** until CALL FORWARDING? appears in the display.
- Press **Yes**. The display shows the EXTERNAL CALLS screen with the Yes and No softkeys. Pressing the No softkey, leads to the INTERNAL CALLS screen. Pressing the No softkey again leads to the I'M HERE screen. Pressing the No softkey again returns to the EXTERNAL CALLS screen. The presence of the asterisk indicates that at least one forwarding mode is turned on.
- Press **Yes**.The display shows the ALWAYS screen.
- If you want to program Always press **Program**, or **Change** (the Change softkey appears if there is a destination programmed) and then **Program**. If you want to program another call forwarding type, press **Next** to get to the desired call forwarding type and press **Program** when the desired screen appears.
- Enter the destination number. If an invalid number is entered, the display shows INVALID. Use the ← softkey to back space and erase the entry. Enter the correct number (can be a speedcall key, if programmed).
- Press **Save** (down arrow on a Family 3 telephone). The type of forwarding, the forwarding destination and an * appear momentarily in the display. The asterisk indicates that the call forwarding is enabled. The display then returns to another forwarding type. Press **Next** to get to the desired call forwarding type.
- Repeat the last four steps to program call forwarding for internal calls.
- Press **Superkey** to return to the normal display.

To deactivate Call Forwarding (Busy, No Answer, Busy/No Answer, or Always)

- Press the Forward All feature key if you want to deactivate all Call Forwarding, or press the Forward Always, Forward Busy, Forward No Answer, or Forward Busy/No Answer feature key to be more specific. The indicator beside the key turns off to indicate that call forwarding is deactivated.
  OR, if a feature key isn't programmed,
- Press **Superkey**.
- Press **No** until CALL FORWARDING? appears in the display.
- Press **Yes**. The display shows the EXTERNAL CALLS screen with the Yes and No softkeys. Pressing the No softkey, leads to the INTERNAL CALLS screen. Pressing the No softkey again leads to the I'M HERE screen. Pressing the No softkey again returns to the EXTERNAL CALLS screen. The presence of the asterisk indicates that at least one forwarding mode is turned on.
- Press **Yes**.The display shows the ALWAYS screen.

- If you want to deactivate Always, press **Change** and then **TurnOff**. The asterisk disappears next to the destination to indicate that this call forwarding destination is deactivated.
- If you want to deactivate another type of forwarding, press **Next** until you reach the required display and press **Change** and then **TurnOff**. The asterisk disappears next to the destination to indicate that this call forwarding destination is deactivated.
- Press **Superkey** to return to normal display.

To activate Call Forwarding (already programmed):

- Press the Forward All, Forward Always, Forward Busy, Forward No Answer, or Forward Busy/No Answer feature key. The indicator beside the key turns on to indicate that Call Forwarding is activated. (Refer to Feature Keys for instructions on programming this feature key).
  or, if a feature key isn't programmed,
- Press **Superkey**.
- Press **No** until CALL FORWARDING? appears in the display.
- Press **Yes**. The display shows the EXTERNAL CALLS screen with the Yes and No softkeys. Pressing the No softkey, leads to the INTERNAL CALLS screen. Pressing the No softkey again leads to the I'M HERE screen. Pressing the No softkey again returns to the EXTERNAL CALLS screen. The presence of the asterisk indicates that at least one forwarding mode is turned on.
- Press **Yes**.The display shows the ALWAYS screen.
- If you want to activate Always, press **Change** and then **TurnOn**.
- If you want to activate another call forwarding type press **Next** until the desired type appears and press **Change** and then **TurnOn**. An asterisk appears next to the destination to indicate that this call forwarding destination is activated.
- Press **Superkey** to return to normal display.

To display the current forwarding type and destination

Note: On a Family 3 telephone, press the dial pad * key instead of the YES softkey, and press the dial pad # key instead of the NO softkey.

- Press **Superkey**.
- Press **No** until CALL FORWARDING? appears in the display.
- Press **Yes**. The display shows the EXTERNAL CALLS screen with the Yes and No softkeys. Pressing the No softkey, leads to the INTERNAL CALLS screen. Pressing the No softkey again leads to the I'M HERE screen. Pressing the No softkey again returns to the EXTERNAL CALLS screen. The presence of the asterisk indicates that at least one forwarding mode is turned on.
- Press **Yes** to view the desired type (INTERNAL CALLS?, EXTERNAL CALLS? or I'M HERE?).The first screen you see is the ALWAYS screen .
- Press **Next** in order to view the IF BUSY and NO ANS screen destinations. Pressing Next in the NO ANS screen leads you to the INTERNAL CALLS.
- Continue the previous two steps to view all the destinations.
- Press **Superkey** to return to normal display.

To delete Call Forwarding

Note: Deleting a destination on a call forward type is dependant on the System Option 102, Feature Level option. Otherwise you may only de-activate the destination number. Family 3 telephones do not support the delete softkey.

- Press **Superkey**.
- Press **No** until CALL FORWARDING? appears in the display.
- Press **Yes**. The display shows the EXTERNAL CALLS screen with the Yes and No softkeys. Pressing the No softkey, leads to the INTERNAL CALLS screen. Pressing the No softkey again leads to the I'M HERE screen. Pressing the No softkey again returns to the EXTERNAL CALLS screen. The presence of the asterisk indicates that at least one forwarding mode is turned on.

- Press **Yes**.The display shows the ALWAYS screen.
- If you want to delete Always, press **Del**.
- If you want to delete another call forwarding type press **Next** until the desired type appears and press **Del**.The Del softkey turns off the forwarding and deletes the destination number. The Del softkey appears only if the System Option 102, Feature Level option is set to level 1 or greater.
- Press **Superkey** to return to normal display.

Family 1 Telephones (with split forwarding disabled):

To program Call Forwarding (Busy, No Answer, Busy/No Answer, or Always)

- Press **Superkey**.
- Press **More** until you see the Forwarding softkey.
- Press **Forwarding**. The display shows the status of the forwarding types: ALWAYS, BUSY and NO-ANSWER and prompts you to select a type.
- Press the appropriate type of call forwarding softkey. The display requests the destination number.
- Enter the destination number. If an invalid number is entered, the display shows INVALID NUMBER. Use the ← softkey to back space and erase the entry. Enter the correct number (can be a speedcall key, if programmed) or press the Current No. softkey to restore the previously programmed destination, if available (only available when the current forward destination is anything but a speedcall).
- Press **Save/On** or **Save/Off**. The Save/On key saves and enables the programmed forwarding. The Save/Off softkey saves and disables the programmed forwarding.The display shows a confirmation screen and returns to the screen that appears after pressing the Forwarding softkey .The Save/Off softkey only appears if the System Option 102, Feature Level is set to level 1 or greater.

To deactivate Call Forwarding (already programmed)

- Press **Forwarding**. You may then scan the forwarding that is programmed by pressing the Previous and Next softkeys.
- Press **Turn FWD Off** to deactivate the stated call forwarding.

To activate Call Forwarding (already programmed)

- Press **Forwarding**. You may then scan the forwarding that is programmed by pressing the Previous and Next softkeys.
- Press **Turn FWD On** to activate the stated call forwarding.

To display the current forwarding type and destination:

- Press **Superkey**.
- Press **More** to access the Forwarding softkey.
- Press **Forwarding**. The display shows the status of the forwarding types: ALWAYS, BUSY and NO-ANSWER. Press Backup if you wish to return to the previous screen.
- Press the desired forwarding type softkey to obtain the programmed destination.
- Press **Superkey** to return to the normal display.

To delete Call Forwarding

Note: Deleting a destination on a call forward type is dependant on the System Option 102, Feature Level option being set to level 1 or greater. Otherwise you may only de-activate the destination number.

- Press **Superkey**.
- Press **More** until you see the Forwarding softkey.
- Press **Forwarding**. The display shows the status of the forwarding types: ALWAYS, BUSY and NO-ANSWER and prompts you to select a type.
- Press the appropriate type of call forwarding softkey.

- Press **Delete**. The Delete softkey turns off the forwarding and deletes the destination number. The Delete softkey only appears if the System Option 102, Feature Level option is set to level 1 or greater. The display then returns to the previous screen.

Family 1 Telephones: (with split forwarding enabled):

Note: With split forwarding enabled, Call Forwarding Busy, No Answer ,and Always can have a destination for incoming external calls and a separate destination for incoming internal calls.

To display the current forwarding type and destination

- Press **Superkey**.
- Press **More** until you see the Forwarding softkey.
- Press **Forwarding**. The display shows the status of the forwarding types: ALWAYS, BUSY and NO-ANSWER. Each of these forwarding types has a separate status for internal calls and external calls.
- Press the desired forwarding type to obtain the programmed destination.
- Press **Superkey** to return to normal display.

To program Call Forwarding (Busy, No Answer, Busy/No Answer, or Always)

Note: Split forwarding is enabled.

- Press **Superkey**.
- Press **More** until you see the Forwarding softkey.
- Press **Forwarding**. The display shows the status of the forwarding types: ALWAYS, BUSY and NO-ANSWER. Each of these forwarding types has a separate status for internal calls and external calls
- Press the appropriate type of call forwarding softkey. ON or OFF appears on the right hand corner of the display depending on the status of the forwarding type.
- Press **Internal** or **External** depending on the source type of call that you want to program. The display requests the destination number.
- Enter the destination number. If an invalid number is entered, the display shows INVALID NUMBER. Use the ← softkey to back space and erase the entry. Enter the correct number (can be a speedcall key, if programmed) or press the Current No. softkey to restore the previously programmed destination, if available (only available when the current forward destination is anything but a speedcall).
- Press **Save/On** or **Save/Off**. The Save/On softkey saves and enables the programmed forwarding. The Save/Off softkey saves and disables the programmed forwarding.The display shows a brief confirmation and then returns to the screen that appears after pressing the Forwarding softkey.

To deactivate Call Forwarding (already programmed)

- Press **Superkey**.
- Press **More** until you see the Forwarding softkey.
- Press **Forwarding**. You may then scan the forwarding that is programmed by pressing the Previous and Next softkeys.
- Press **Turn FWD Off** to deactivate the stated call forwarding.

To activate Call Forwarding (already programmed)

- Press **Superkey**.
- Press **More** until you see the Forwarding softkey.
- Press **Forwarding**. You may then scan the forwarding that is programmed by pressing the Previous and Next softkeys.
- Press **Turn FWD On** to activate the stated call forwarding.

To delete Call Forwarding

Note: Deleting a destination on a call forward type is dependant on the System Option 102, Feature Level option being set to level 1 or greater. Otherwise you may only de-activate the destination number.

- Press **Superkey**.
- Press **More** until you see the Forwarding softkey.
- Press **Forwarding**. The display shows the status of the forwarding types: ALWAYS, BUSY and NO-ANSWER. Each of these forwarding types has a separate status for internal calls and external calls.
- Press the appropriate type of call forwarding softkey. ON or OFF appears on the right hand corner of the display depending on the status of the forwarding type.
- Press **Internal** or **External** depending on the source type of call that you want to delete.
- Press **Delete**. The Delete softkey turns off the forwarding and deletes the destination number. The Delete softkey only appears if the System Option 102, Feature Level option is set to level 1 or greater. The display then returns to the previous screen.

Family 5 Telephones:

5330 Phones (split forwarding not required):

To program Call Forwarding (Busy, No Answer, Busy/No Answer, or Always)

- Press the **Settings** key.
- Press **Programmable Keys**.
- Press the key you want to program.
- If Applications are displayed, press **View Features**.
- Use the Page Navigation keys to move through the list of features.
- Select the desired forwarding option (Busy, No Answer, Busy/No Answer, or Always).
- Press **Save**.

5330 Phones (split forwarding required):

To program Call Forwarding (Busy, No Answer, Busy/No Answer, or Always)

- Press the key you have programmed as **Superkey**.
- Press **No** until CALL FORWARDING? appears in the display.
- Press **Yes**. The display shows the EXTERNAL CALLS screen with the Yes and No softkeys. Pressing the No softkey, leads to the INTERNAL CALLS screen. Pressing the No softkey again leads to the I'M HERE screen. Pressing the No softkey again returns to the EXTERNAL CALLS screen. The presence of the asterisk indicates that at least one forwarding mode is turned on.
- Press **Yes**.The display shows the ALWAYS screen.
- If you want to program Always press **Program**, or **Change** (the Change softkey appears if there is a destination programmed) and then **Program**. If you want to program another call forwarding type, press **Next** to get to the desired call forwarding type and press **Program** when the desired screen appears.
- Enter the destination number. If an invalid number is entered, the display shows INVALID. Use the ← softkey to back space and erase the entry. Enter the correct number (can be a speedcall key, if programmed).
- Press **Save**. The type of forwarding, the forwarding destination and an * appear momentarily in the display. The asterisk indicates that the call forwarding is enabled. The display then returns to another forwarding type. Press **Next** to get to the desired call forwarding type.
- Repeat the last four steps to program call forwarding for internal calls.
- Press **Superkey** to return to the normal display.

To deactivate Call Forwarding (already programmed):

- Press **Superkey**.
- Press **No** until CALL FORWARDING? appears in the display.
- Press **Yes**. The display shows the EXTERNAL CALLS screen with the Yes and No softkeys. Pressing the No softkey, leads to the INTERNAL CALLS screen. Pressing the No softkey again leads to the I'M

HERE screen. Pressing the No softkey again returns to the EXTERNAL CALLS screen. The presence of the asterisk indicates that at least one forwarding mode is turned on.

- Press **Yes**.The display shows the ALWAYS screen.
- If you want to delete Always, press **Del**.
- If you want to delete another call forwarding type press **Next** until the desired type appears and press **Del**.The Del softkey turns off the forwarding and deletes the destination number. The Del softkey appears only if the System Option 102, Feature Level option is set to level 1 or greater.
- Press **Superkey** to return to normal display.

5340 Phones (split forwarding not required):

To program Call Forwarding (Busy, No Answer, Busy/No Answer, or Always)

- Press the **Settings** key.
- Press **Programmable Keys**.
- Press the key you want to program.
- If Applications are displayed, press **View Features**.
- Use the Page Navigation keys to move through the list of features.
- Select the desired forwarding option (Busy, No Answer, Busy/No Answer, or Always).
- Press **Save**.

5340 Phones (split forwarding required):

To program Call Forwarding (Busy, No Answer, Busy/No Answer, or Always)

- Press the key you have programmed as **Superkey**.
- Press **More** until you see the Forwarding softkey.
- Press **Forwarding**. The display shows the status of the forwarding types: ALWAYS, BUSY and NO-ANSWER. Each of these forwarding types has a separate status for internal calls and external calls
- Press the appropriate type of call forwarding softkey. ON or OFF appears on the right hand corner of the display depending on the status of the forwarding type.
- Press **Internal** or **External** depending on the source type of call that you want to program. The display requests the destination number.
- Enter the destination number. If an invalid number is entered, the display shows INVALID NUMBER. Use the ← softkey to back space and erase the entry. Enter the correct number (can be a speedcall key, if programmed) or press the Current No. softkey to restore the previously programmed destination, if available (only available when the current forward destination is anything but a speedcall).
- Press **Save/On** or **Save/Off**. The Save/On softkey saves and enables the programmed forwarding. The Save/Off softkey saves and disables the programmed forwarding.The display shows a brief confirmation and then returns to the screen that appears after pressing the Forwarding softkey.

To deactivate Call Forwarding (already programmed)

- Press **Superkey**.
- Press **More** until you see the Forwarding softkey.
- Press **Forwarding**. You may then scan the forwarding that is programmed by pressing the Previous and Next softkeys.
- Press **Turn FWD Off** to deactivate the stated call forwarding.

To activate Call Forwarding (already programmed)

- Press **Superkey**.
- Press **More** until you see the Forwarding softkey.
- Press **Forwarding**. You may then scan the forwarding that is programmed by pressing the Previous and Next softkeys.
- Press **Turn FWD On** to activate the stated call forwarding.

Attendant Consoles

See Attendant - Call Forward Setup and Cancel

SubattendantTelephones

See Subattendant - Call Forward Setup and Cancel

Call Forwarding - Display Prime as Forwarder

Description

This feature displays either the forwarder's prime line or the logical line on the forwardee's set display. If COS Option 258 (Display Prime as Forwarder) is enabled, the prime line of the set that forwarded the call is displayed. If this COS is disabled, the logical line appears as the forwarder for all types of forwarding.

Conditions

The following conditions apply to this feature:

- COS Option 258 is only checked if the forwarder is a logical line.
- Calls to logical lines are only forwarded if the logical line appears only on one set.
- Calls forwarded from a single appearance direct trunk select key that has COS option 815 enabled will always display the prime as forwarder.
- This feature is only applicable when the forwardee's set is a display set.

Programming

Enable COS Option 258 (Display Prime as Forwarder) in the COS of the extension where the logical line appears.

Operation

None.

Call Forwarding - External

Description

This feature directs call forwarding to a personal speed call key or a system abbreviated dial number.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Call Forwarding - External requires a receiver for dialing. If one is not available then forwarding is ignored in Call Forward - No Answer. If it is done during a reroute, the caller is dropped or given reorder tone.
- An extension with COS Option 200 (Account Code, Forced Entry - External Calls) in its COS cannot set up the Call Forwarding - External feature.
- An extension must have COS Option 245 (Abbreviated Dialing Access) enabled, COS Option 208 (Call Forwarding External) enabled, and COS Option 200 (Account Code, Forced Entry - External

Calls) disabled in its COS to be able to program call forwarding to system abbreviated dial numbers. The option is not needed for callers which are forwarded to abbreviated dial numbers.

- Toll Control applies to the calling party for Call Forwarding - External to speed call keys or to personal speed calls.
- Toll Control is not done for forwarding to external numbers using the system abbreviated dial feature.
- No Toll Control checking is done for CO trunks which are externally call forwarded.
- A trunk or a party with a single party trunk on hold can only be forwarded externally if System Option 21 (Incoming to Outgoing Call Forward) is enabled.
- An extension cannot set up forwarding to an external number unless it has COS Option 208 (Call Forwarding - External) in its COS.
- Once a call is forwarded externally, no more forwarding hops occur.
- Campon to busy trunk groups after forwarding external is possible if the caller is allowed to campon to trunk groups; see Campon.
- When an attendant or SUPERSET 4DN telephone user calls an extension that is forwarded to an external number, the display shows the external number and indicates the extension which forwarded the call. If the speed call or abbreviated dial number is private, text indicating a private number appears on the display instead of the external number. The console can display the number if it has COS Option 110 (Attendant Abbr. Dial Confidential Number Display) enabled in its COS.
- Consoles and extensions can transfer a party on consultation hold as soon as forwarding to an external number starts, at any point in dialing the external number.
- The reference is to the stored speedcall key number or the personal or system abbreviated dial number. If this stored number is changed, then the forwarding destination changes also.
- External call forwarding will proceed if the trunk is: a DID or Tie trunk, COR restricted in CDE Form 26, and has COS Option 606 (Telephone - Enhanced Answering Position) enabled.

Programming

Enable COS Option 208 (Call Forwarding - External) in the calling party's COS (extension's COS or trunk's COS).

For external call forwarding involving two trunks, ensure that the incoming and outgoing trunk can be connected together. CDE Form 30 - Device Interconnection Table specifies which devices can be connected together.

Enable COS Option 245 (Abbreviated Dialing Access) for the extension to allow forwarding to system abbreviated dial numbers or to personal speed calls.

Disable COS Option 200 (Account Code, Forced - External Calls) in the extension's COS.

Enable System Option 21 (Incoming to Outgoing Call Forward) to allow trunks to be externally call forwarded.

Ensure Feature Access Code 03 (Call Forwarding - All Calls) has a feature access code programmed.

For feature key activation of call forwarding on a Family 3, Family 4, SUPERSET 410, or SUPERSET 3DN telephone, program a FORWARD ALL feature key. (See Feature Keys.)

Operation

Operation varies depending upon the device type as described below.

Family 2 and Industry-standard Telephones:

To set up Call Forwarding - External at an extension:

- Lift the handset - dial tone is heard.

- Dial the Call Forward - All Calls access code if the forwarding is non-split. Other access codes exist for internal/external split.
- Dial the condition code for the desired type of forwarding.
- Dial the abbreviated dial index number, which contains the external telephone number to which calls are to be forwarded. Dial tone is returned if the above codes are valid; reorder tone is returned if the codes are invalid.
- Replace the handset - Call Forwarding - External is now active.

To cancel Call Forwarding - External at an extension:

- Lift the handset - dial tone is returned.
- Dial the Call Forward - All Calls access code (assuming Split Call Forwarding is disabled).
- Replace the handset - Call Forwarding - External is now inactive.

OR

- Lift the handset - dial tone is returned.
- Dial the Clear All feature access codes - Dial tone is returned.
- Replace the handset.

SUPERSET 410 and SUPERSET 3DN Telephones:

To set up Forwarding - External:

- Follow the same procedure as for industry-standard telephones; it is also possible to dial the codes and extension destination directly on the keypad and then press the SPEAKER ON/OFF key to go on-hook.

To activate or deactivate forwarding already programmed:

- Press the FORWARD ALL feature key, if programmed. The LCD indicator is dark or key LED illuminated (activated) to indicate forwarding active, clear or key LED extinguished to indicate not active.

Family 3 , Family 4 and Family 5 Telephones:

Note: On a Family 3 telephone, press the dial pad * key instead of the YES softkey, and press the dial pad # key instead of the NO softkey.

To display the current forwarding type and external destination:

- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey. The display shows the current forwarding type and external destination, if programmed.
- Press SUPERKEY to return to normal display.

To activate forwarding already programmed:

- Press the FORWARD ALL feature key. The indicator beside the key turns on to indicate that call forwarding is activated. (Refer to feature keys for instructions on programming this feature key.)

OR, if a FORWARD ALL feature key isn't programmed,

- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey.
- Press the CHANGE softkey.
- Press the TURNON softkey. An asterisk appears next to the destination to indicate that call forwarding is activated.
- Press SUPERKEY to return to normal display.

To deactivate forwarding to an external destination:

- Press the FORWARD ALL feature key. The indicator beside the key turns off to indicate that call forwarding is deactivated.

OR, if a FORWARD ALL feature key isn't programmed,

- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey.
- Press the CHANGE softkey.
- Press the TURNOFF softkey. The asterisk next to the destination disappears.
- Press SUPERKEY to return to normal display.

To program the set to forward all calls to an external number:

- Press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey. The display shows the current forwarding type and destination, if programmed.
- Press the CHANGE softkey.
- Press the PROGRAM softkey.
- Press the NO softkey until the desired type of call forwarding appears in the display.
- Press the YES softkey.
- Enter the destination number, or press a SPEED CALL key on the Mitel telephone or associated PKM. If an invalid number is entered, the display shows INVALID NUMBER. Use the ← softkey to back space and erase the entry.
- Press the SAVE softkey (down arrow on a SUPERSET 4015 set). The type of forwarding, the forwarding destination, and an * appear momentarily in the display. The display then returns to showing the date and time. Call forwarding is now programmed and enabled.

Note: The asterisk indicates that call forwarding is enabled.

Family 1 Telephones:

To set up Call Forwarding - External:

- Press SUPERKEY.
- Find the FORWARDING softkey. (Use the MORE softkey).
- Press the FORWARDING softkey. The display shows the current forwarding type and destination, if programmed.
- Select the required type. The display requests the destination number.
- Enter the destination number, or press a SPEED CALL key on the Mitel telephone or associated PKM. If an invalid number is entered, the display shows INVALID NUMBER. Use the ← softkey to back space and erase the entry. Enter the correct number (can be a speedcall key, if programmed) or press the CURRENT NO. softkey to restore the previously programmed destination (if available).
- Press the SAVE/ON softkey. The display returns to the display you had when you pressed the Forwarding softkey.
- Press SUPERKEY to return to normal display.

To display the current forwarding type and destination:

- Press SUPERKEY.
- Find the FORWARDING softkey. (Use the MORE softkey).
- Press the FORWARDING softkey. The display shows the current forwarding type and destination, if programmed.

- Press SUPERKEY to return to normal display.

To deactivate Call Forwarding (already programmed)

- Press SUPERKEY.
- Press MORE until you see the Forwarding softkey.
- Press FORWARDING. You may then scan the forwarding that is programmed by pressing the Previous and Next softkeys.
- Press TURN FWD OFF to deactivate the stated call forwarding.

To activate Call Forwarding (already programmed)

- Press SUPERKEY.
- Press MORE until you see the Forwarding softkey.
- Press FORWARDING. You may then scan the forwarding that is programmed by pressing the Previous and Next softkeys.
- Press TURN FWD OFF to activate the stated call forwarding.

Family 5 Telephones:

Refer to the 5330/5340 IP Phone User Guide available at Mitel OnLine.

Call Forwarding - Forced Call Forward

Description

This feature allows a telephone user to force a dialled call to forward immediately rather than waiting for the ringing timeout.

One application of this feature is to leave a quick voice mail message for someone who you know is not at their desk.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply for this feature:

- This feature is available to Family 3, Family 4, and SUPERSET 4150, SUPERSET 410, and SUPERSET 430 telephones.
- Destinations for Call Forwarding - No Answer (Internal) or Call Forwarding - Busy (Internal) must be programmed and activated.
- The Forced Forward feature uses the Call Forwarding - No Answer (Internal) destination when the called telephone is not answering (i.e., ringback tone is present).
- The Forced Forward feature uses the Call Forwarding - Busy (Internal) destination if the called telephone is busy.
- This feature applies to individual call forwarding and tenant-based call forwarding.

Programming

For Mitel telephones without softkeys, program a Forward Call feature key. This feature key will not follow the tenant-based call forwarding.

To program this feature on a Symbol MiNET Wireless Phone

1. Access the Superkey menu (press the Function key, scroll to "Superkey", press the Send key)

2. Scroll to locate "Call Forwarding".
3. Press the left-hand softkey.
4. Do one of the following:
   - Press the Yes softkey to program Call Forwarding for external calls.
   - Press the No, and then press the Yes softkey to program Call Forwarding for internal calls.
5. Press the NXT softkey until the desired type of Call Forwarding appears.
6. Press the PRG softkey.
7. Enter the number to which you want your calls forwarded.
8. Press the SAV softkey.

Operation

To force forward a ringing call:

Family 5 Telephones

- Press the Forward softkey (5330) or the Forward Call softkey (5340)

Family 4 Telephones (except Symbol MiNET Wireless phone):

- Press the FwdMe softkey.

SUPERSET 4150 and SUPERSET 430 Telephones:

- Press the Forward Me softkey.

Family 3 and SUPERSET 410 Telephones:

- Press the Forward Call feature key (its LED is illuminated at this stage of a call).

Symbol MiNET Wireless Phone:

- Press the FWD softkey.

Call Forwarding - Forward Call

Description

This feature allows a Family 1, Family 3, Family 4, Family 5, and SUPERSET 410, or SUPERSET 3DN telephone user to force an incoming call to be forwarded to a pre-programmed forward destination. Forward Call uses the appropriate (Internal or External) Call Forward - Busy destination. If there is no (Internal or External) Call Forward - Busy destination programmed, Forward Call uses the Call Forward - No Answer destination.

Users can forward both ringing calls and camped-on calls.

Family1 telephone users can view the calling party identity on the LCD display, and decide if it is to be forwarded or not, rather than having the system forward it automatically.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- This feature is available to the Family 1, Family 3, Family 4, Family 5, SUPERSET 410, and SUPERSET 3DN telephones.
- This feature will apply to telephones which have forwarding programmed.

- Forward Call uses the appropriate (Internal or External) Call Forward - Busy destination. If there is no Internal or External Call Forward - Busy destination programmed, Forward Call uses the Call Forward - No Answer destination. The Internal or External Call Forward - Busy destinations or the Call Forward - No Answer destinations must be programmed and activated. If Forward - Always is programmed, the Forward - Always will take precedence over the Forward - Busy and Call Forward - No Answer.
- The conditions which apply to Call Forward - No Answer also apply to this feature.
- The feature applies only to prime line calls, and only to calls which can normally be forwarded (i.e., not callbacks or wake-ups, etc.).

  This feature applies to individual call forwarding and not tenant-based call forwarding.

Programming

The following programming is required:

- For Mitel Family 1, Family 3, Family 4 (except Symbol MiNET Wireless telephones), Family 5, SUPERSET 410, and SUPERSET 3DN telephones, program call forwarding as described in Call Forwarding.
- To activate this feature from a SUPERSET 4015 , SUPERSET 410, or SUPERSET 3DN telephone, you must program a FORWARD CALL feature key. (See Feature Keys.) This feature key will not follow the tenant-based call forwarding.

To program this feature on a Symbol MiNET Wireless Phone

1. Access the Superkey menu (press the Function key, scroll to "Superkey", press the Send key)
2. Scroll to locate "Call Forwarding".
3. Press the left-hand softkey.
4. Do one of the following:

   - Press the Yes softkey to program Call Forwarding for external calls.

   - Press the No, and then press the Yes softkey to program Call Forwarding for internal calls.
5. Press the NXT softkey until the desired type of Call Forwarding appears.
6. Press the PRG softkey.
7. Enter the number to which you want your calls forwarded.
8. Press the SAV softkey.

Operation

To forward a ringing call:

Family 4 and Family 5 Telephones (except Symbol MiNET Wireless phone):

- Press the FORWARD softkey.

Family 3, SUPERSET 410, and SUPERSET 3DN Telephones:

- Press the FORWARD CALL feature key.

Family 1 Telephones:

- Press the FORWARD CALL softkey.

Symbol MiNET Wireless Phone:

- Press the FWD softkey.

Call Forwarding - I'm Here

Description

This type of forwarding operates the same as Call Forwarding - Always, but it is activated from another extension. All calls (internal and external calls) are forwarded to the new location. The forwarded extension can originate calls in the normal manner.

When Call Forwarding - I'm Here is cancelled, all programming for Call Forwarding - Always on the normal extension will have to be reprogrammed because the Call Forwarding - Always will be completely deleted.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- COS Option 234 (Never a Forwardee) must be disabled for this extension.
- For a display telephone to receive the I'm Here message prompt, it must have COS Option 234 (Never a Forwardee) disabled and COS Option 209 (Call Forward - Always) enabled.
- This feature applies to individual call forwarding and not tenant-based call forwarding.

Programming

Enable COS Option 209 (Call Forwarding - Always) in the set's COS.

Program access codes for Feature Access Code 06 (Call Forwarding - I'm Here) and Feature Access Code 07 (Call Forwarding - Cancel I'm Here).

Operation

Operation varies depending upon the device types as described below.

Family 2 and Industry-standard Telephones:

To select Call Forwarding - I'm Here:

- At another extension, lift the handset - wait for dial tone.
- Dial the Call Forwarding - I'm Here access code.
- Dial your own extension number - dial tone returns.
- Hang up - the extension is available for normal use.

To cancel Call Forwarding - I'm Here from the telephone which set Call Forward - I'm Here:

- Lift the handset - wait for dial tone.
- Dial the Cancel I'm Here access code.
- Dial the extension whose forwarding is to be canceled; hang up - the forwarding is canceled for that extension.

OR

- Hang up - all I'm Here forwarding to this extension is canceled.

SUPERSET 410 and SUPERSET 3DN Telephones:

To select Call Forwarding - I'm Here:

- At another extension, dial the Call Forwarding - I'm Here access code.
- Dial your own extension number - dial tone returns.
- Press the SPEAKER ON/OFF key - the extension is available for normal use.

To cancel Call Forwarding - I'm Here from the telephone which set Call Forward - I'm Here:

- Dial the Cancel I'm Here access code.
- Dial the extension whose forwarding is to be canceled and press the SPEAKER ON/OFF key. The forwarding is canceled for that extension.

OR

- Press the SPEAKER ON/OFF key to cancel all I'm Here forwarding to this extension.

Family 3, Family 4 (except Symbol MiNET Wireless telephones) with split forwarding enabled:

Note: On Family 3 telephones, press the dial pad * key instead of the YES softkey, and press the dial pad # key instead of the NO softkey.

To set up and turn on Call Forwarding - I'm Here:

- At the destination Family 3 , Family 4 (except Symbol) telephone, press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey. The EXTERNAL CALLS? screen displays.
- Press the NO or NEXT softkey until I'M HERE? appears in the display.
- Press the YES softkey.
- Enter extension number from which calls are to be forwarded. If an invalid number is entered, the display shows INVALID. Use the ← softkey to back space and erase the entry.
- Press the SAVE softkey (the down arrow on a Family 3 telephone). The extension number and name from which your calls will be forwarded, appear momentarily in the display. The display then returns to showing the date and time. Call Forwarding is now programmed and enabled.

To cancel Call Forwarding - I'm Here from the telephone at the Call Forwarding - I'm Here destination:

- Lift the handset.
- Dial the Cancel I'm Here access code.
- Dial the extension whose forwarding is to be canceled and hang up. The forwarding is canceled for that extension.

To cancel Call Forwarding - I'm Here from the Family 3 telephone that was forwarded:

- Press SUPERKEY.
- Press # until Call Forwarding? appears in the display.
- Press * . EXTERNAL CALLS appears.
- Press * twice.
- Press #.
- Press # until INTERNAL CALLS? appears.
- Press * twice.
- Press # .

To cancel Call Forwarding - I'm Here from the Family 4 (except Symbol) telephone that was forwarded:

- Press SUPERKEY.
- Press the NO softkey until "Call Forwarding?" appears.
- Press the YES softkey. EXTERNAL CALLS? appears.
- Press the YES softkey. ALWAYS with the Extension Number appears.
- Press the DEL softkey.
- Press the NEXT softkey until INTERNAL CALLS ? appears.

- Press the YES softkey. ALWAYS with the Extension Number appears.
- Press the DEL softkey.

Family 5 Telephones:

5330 Phones

To forward calls from a remote station to your current station:

- Press the key that you have programmed as **Superkey**.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey.
- Press the NEXT softkey until "I AM HERE" appears and then press YES.
- Enter the extension of the remote station. If you make an error, use the ← softkey to backspace.
- Press **Save**.

To cancel "Call Forward - I Am Here" from the station that was forwarded:

- Press the key that you have programmed as **Superkey**.
- Press **No** until CALL FORWARDING? appears in the display.
- Press **Yes**. Programmed settings appear with an * beside the name.
- Press **Yes** to change.
- Press **TurnOff**.

5340 Phones

To forward calls from a remote station to your current station:

- Press the key that you have programmed as **Superkey**.
- Press the **More** softkey until **Forwarding** appears and then press **Forwarding**.
- Press **To Me**.
- Enter the extension of the remote station. If you make an error, use the ← softkey to backspace.
- Press **Save/On**.

To cancel "Call Forward - I Am Here" from the station that was forwarded:

- Press the **Forwarding** softkey.
- Press **Prev** or **Next** if necessary to navigate to the setting you want to change.
- Press **Fwd/Off**.

Family 3 or Family 4 (except Symbol MiNET Wireless telephones) with split forwarding disabled:

Note: On a Family 3 telephone, press the dial pad * key instead of the YES softkey, and press the dial pad # key instead of the NO softkey.

To set up and turn on Call Forwarding - I'm Here:

- At the destination Family 3, SUPERSET 4125, or SUPERSET 420 telephone, press SUPERKEY.
- Press the NO softkey until CALL FORWARDING? appears in the display.
- Press the YES softkey.
- Press the NEXT softkey until I'M HERE? appears in the display.
- Press the YES softkey.
- Enter extension number from which calls are to be forwarded. If an invalid number is entered, the display shows INVALID. Use the ← softkey to back space and erase the entry.
- Press the SAVE softkey (down arrow on a Family 3 telephone). The extension number and name from which your calls will be forwarded, appear momentarily in the display. The display then returns to showing the date and time. Call Forwarding is now programmed and enabled.

To cancel Call Forwarding - I'm Here from the telephone at the Call Forwarding - I'm Here destination:

- Lift the handset.

- Dial the Cancel I'm Here access code.
- Dial the extension whose forwarding is to be canceled and hang up. The forwarding is canceled for that extension.

To cancel Call Forwarding - I'm Here from the Family 3 telephone that was forwarded:

- Press SUPERKEY.
- Press # until CALL FORWARDING? appears in the display.
- Press *.
- Press * twice.
- Press # .

To cancel Call Forwarding - I'm Here from the Family 4 (except Symbol MiNET Wireless) telephone that was forwarded:

- Press SUPERKEY.
- Press the NO softkey until "Call Forwarding?" appears.
- Press the YES softkey.
- Press the DEL softkey.

Family 1 Telephones:

To set up and activate Call Forwarding - I'm Here, using softkeys:

- At the destination Family 1 telephone, press Superkey, then find and press the FORWARDING softkey.
- Press the TO ME softkey.
- Enter extension number from which calls are to be forwarded.
- Press SAVE/ON softkey.
- The set displays NOW FROM XXXXX ALWAYS.

To cancel Call Forwarding I'm Here from the set that the Call Forwarding I'm Here was sent to:

- Lift the hand set.
- Dial the Cancel I'm Here access code.
- Dial the extension whose forwarding is to be canceled and hang up. The forwarding is canceled for that extension.

To cancel Call Forwarding - I'm Here from the SUPERSET 4150 telephone that was forwarded:

- Press SUPERKEY.
- Press **More** until the **Forwarding** softkey appears.
- Press the **Forwarding** softkey.
- If EXTERNAL CALLS? appears do the following:
  - Press the **Always** softkey.
  - Press the **Internal** softkey.
  - Press the **Delete** softkey.
  - Press the **Always** softkey.
  - Press the **External** softkey.
  - Press the **Delete** softkey.
- If EXTERNAL CALLS? does not appear, do the following:
  - Press the **Always** softkey.
  - Press the **Delete** softkey.

Call Forwarding - Internal / External Split

Description

Calls may be forwarded to different destinations according to the source of the call - either an internal call from within the PBX or an external call from outside of the PBX.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- COS Option 260 (Internal / External Split Call Forward) must be enabled for the extension to allow forwarding according to the source of the call.
- For Call Forwarding, all devices are treated as internal calls unless COS Option 709 (Follow External Call Forward) is enabled for the calling device. Therefore, trunks without COS 709 enabled are treated as "internal" devices by Call Forwarding.
- A feature key which turns forwarding ON or OFF, turns both internal and external on or off together.
- To allow call forwarding to an external destination, COS Option 208 (Call Forwarding - External) must be enabled.
- When split forwarding is enabled, cancelling call forwarding destinations for internal calls using the feature access code 04, Call Forwarding - Internal Only disables all three call forward internal modes: Always, Busy and No Answer. An exception to this occurs when Call Forward - I'm Here sets up the call forwarding (internal and external destinations for Always are disabled and deleted).
- When split forwarding is enabled, cancelling call forwarding destinations for external calls using the feature access code 05, Call Forwarding - External Only disables all three call forward external modes: Always, Busy and No Answer. An exception to this occurs when Call Forward - I'm Here sets up the call forwarding (internal and external destinations for Always are disabled and deleted).
- When split forwarding is disabled, cancelling all call forwarding destinations using the feature access code 03, Call Forwarding - All Calls disables the internal and external modes for Always, Busy and No Answer.

Programming

Enable COS Option 260 (Internal / External Split Call Forward) for the extension to allow forwarding according to the source of the call.

Enable COS Option 709 (Follow External Call Forward) for devices to be treated as external calls.

Enable the desired call forwarding type in the extension’s COS.

Ensure Feature Access Code 04 (Call Forwarding - Internal Only) and Feature Access Code 05 (Call Forwarding - External Calls Only) have feature access codes programmed.

For feature key activation of call forwarding on a Family 3, Family 4 (except Symbol MiNET Wireless telephones), Family 5, SUPERSET 410, or SUPERSET 3DN telephone, program a FORWARD ALL feature key. (See Feature Keys.)

For the Symbol MiNET Wireless Phone, program the phone as specified in the Call Forwarding - Forward Call feature.

Operation

Operation varies depending upon the device type and the call forwarding type as described previously in Call Forwarding.

Call Forwarding - Toggle Keys

Description

This feature allows users of all Mitel telephones to direct their call forwarding (Always, Busy, No Answer, Busy/No Answer) temporarily to another destination with a touch of a button. You may toggle back and forth between one forwarding mode to another without having to reprogram the forwarding destinations.

This is very convenient for people who leave their desk phone and use a cordless phone for short periods of time. In this case the user could press a Forward Always button on his desk phone when he leaves his desk and takes his cordless. After returning, he would press the Forward Always button again to return the forwarding to its normal state.

The attendant can also cancel call forwarding for all extensions at once. The console operator can disable all the call forward modes for an extension if the system option Feature Level is set to level 1 or greater. See Attendant Call Forward Setup and Cancel.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

- The toggle key (feature key) for the forwarding type must be programmed. The keys are Forward Always, Forward Busy, Forward No Ans, and Forward Busy/NA . These feature keys (excluding Forward Always) are dependant on the FOR System Option 102, Feature Level being set to level 1 or greater.
- Pressing the Forward All feature key when some call forwarding types are enabled, disables all destinations programmed for all the call forwarding types. Pressing the Forward All feature key when all the call forwarding types are disabled, enables all the destinations for all the call forwarding types that have been programmed.
- The toggle keys for call forwarding can be used with PKM keys.
- Pressing the feature key for the forwarding type activates the call forwarding for that particular type of call forwarding, for internal and external calls if the destinations are programmed. If there is only an internal call destination programmed, then only the internal calls for that type of call forwarding will be activated, and vice versa.
- The LED of the feature key will be on solid when the call forwarding is enabled for either the external or internal calls of that particular type of call forwarding.The LED of the feature key will be off when the call forwarding of that particular type is disabled for internal and external calls.
- Pressing the feature key for the forwarding type when the LED is on solid disables the internal and external destinations for that particular forwarding type.

Programming

- Program a feature key for the forwarding type. The feature keys are Forward Always, Forward Busy, Forward No Ans, and Forward Busy/NA . See Feature Keys.
- Program the destinations for the call forwarding. See Call Forwarding.

Operation

- Press the programmed feature key: Forward Always, Forward Busy, Forward No Ans, or Forward Busy/NA .

Call Logging

Description

Call Logging provides the user with a log showing a list of all the calls to their telephone. The log includes all calls answered and not answered, as well as, calls forwarded to another destination (Call Forwarding - Always excluded). Call logging for missed calls or all calls can show internal calls, external calls, or both. The user can return a call by selecting the log of the call and pressing the Call softkey.

Feature Availability

Click here to see a list of the phones that can use this feature.

Note: The 5207 IP Phone does not support this feature.

Conditions

- In order to log a caller, the caller must be identifiable by at least an extension number or a public network number. Callers that do not have a return number; that is, callers identified in the CLASS message as unknown, out of area, or private are not logged. Callers that are only identified by trunk number are also not logged. Call logging of external calls require one of the following trunk types: T1, PRI, or LS/CLASS. Recommended from the Central Office are ANI (T1) or CLID (PRI and LS/CLASS).
- Logged calls do not include calls on logical/phantom lines.
- The system stores up to 2000 calls in call logging. Each user can store up to 20 calls in their log. COS Option 620, SUPERSET Telephone - Max Call Logs Allowed (0-20), determines the number of call logs that the user may have.
- Logged calls include calls to the prime line, and calls to a single appearance of the CO Line key or DTS key. Calls to multiple appearances of the CO Line key or DTS key can be logged by enabling COS Option 621 - Multiple CO/DTS in the COS of each set with an appearance.
- Calls to the prime line, CO line key, and DTS key appear together in the same list; the user is not able to differentiate the calls coming in on a prime line, CO line key or a DTS key. If the line key is programmed on another set, future calls will not be logged and the existing calls logged shall remain until they are deleted.
- Calls that are transferred or forwarded from a console or another set are logged. The log identifies the caller that was transferred or forwarded.

  Call Logging interacts with Forwarding and Transferring in the following manner:

  - Records a supervised transferred call (the transferring party stays on the line until the called party answers) and identifies the transferring party as the caller.
  - Records an unsupervised transfer call (the transferring party leaves the caller after initiating the transfer) and identifies the caller that was transferred as the caller.
  - Does not record forwarded calls on a set with Call Forward Always.
  - Records forwarded calls due to the set being busy or not answering. The log identifies the caller as Forwarded in the Missed Calls list.
  - Records an accepted forwarded call. The log identifies the actual caller, not the forwarding party.
- Calls to multiple appearances of a DTS/CO Line key are treated in the following manner (with COS Option 621 enabled):

  - If the call is answered, a single "answered call" record will be created on the set that answered the call.
  - If the trunk drops the call, a "missed call" record will be created on each set with a line appearance.
  - If the call is not answered and then forwarded via tenant-based call forwarding, a "missed call" record will be created on each set with a line appearance. If tenant-based call forwarding is not programmed, the call will be dropped and following ringback timeout and a "missed call" record will be created on each set.
- Family 1 , Family 4 (except SUPERSET 4DN and Symbol MiNET Wireless), and Family 5 telephones support call logging. Subattendants of these telephones also support call logging. If a user changes sets from one that supports call logging to one that does not, all existing logs are deleted.
- When there are any call logs (old or new) to be read, users of Family 1 telephones will see the Callers softkey and users of Family 4 and Family 5 telephones will see the LED of the Callers feature key lit.
- A user of an attendant console (5540 IP Console, SUPERCONSOLE 1000 and LCD consoles) cannot have logs themselves but may read how many logs a telephone has and may delete these logs. The user of an attendant console may also view how many system logs are in use (max. is 2000).
- If an attendant or a subattendant is accessing log data from a set, the user of the set will have no access to this data. The opposite also occurs.
- The displays on the attendant console and on the subattendant telephone show a message to indicate that the call logs are being accessed by someone else.
- If the user has reached their maximum number of call logs, and the system has not reached its capacity, the new call log will overwrite the oldest call log.
- If the system has reached its maximum number of call logs (2000 undeleted call logs), the system deletes the oldest call log in the system to make space for the new call log and also generates a maintenance log to indicate that call logs should be deleted.
- In Hotel/Motel, call logs are deleted on guest room telephones after the guest is checked out; maintenance logs are not generated.
- If an attendant or a subattendant deletes any logs from a set; a maintenance log is generated.
- Call Logging is dependant on the COS option 620, Telephone -Max Call Logs Allowed (0-20), and the FOR System Option 102, Feature Level. Sets with a value of 0 for COS option 620, will not have the Call Logging softkey. Reducing the number in COS option 620 does not delete call logs.
- A system reset does not affect the call logs, however; the logs will not be part of a database backup and restore.

Programming

- In Form 03:
  - Program COS Option 620, Telephone - Max Call Logs Allowed (0-20), in the COS of the telephones.
  - Enable COS Option 621, Multiple CO/DTS, in the COS of each telephone with a DTS or CO line key appearance (COS Option 620 must be a value greater than 0).
- In CDE Form 04, set System Option 102, Feature Level to level 1 or greater.
- Program the CALLERS feature key, in Form 9, Form 37, or Form 38. See Feature Keys.

Operation

Family 1 **(except SUPERSET 4DN) telephones:**

To program the logging of all calls or just missed calls to the call log

- Press **Superkey**.
- Press **Call Logging**.

- Press **Missed Calls** or **All Calls**.
- Press **Internal**, **External**, or **Both**.

To access the log of callers

- Press **Callers** to display the number of new calls and old calls.
- Press **New Calls** or **Old Calls**, to obtain the log record.

  The softkeys Next and Previous appear if more than one call log exists. The user has the option to press Next, Previous, Call, Delete, Delete All or Backup.

To call a caller recorded on the call log

- If the caller number is an internal call, press **Call**.
- If the caller number is an external call, dial 9 and then press **Call**. Telco enters the area code if necessary.
- If the caller number is external and long distance, dial 91 and then press **Call**. The user can use up to three digits prior to Call. Telco enters the area code if necessary.

To delete a call log record

- Press **Callers** to display the number of new calls and old calls.
- Access the call log, and choose **New Calls** or **Old Calls**.
- Press **Delete**.
- Press **Confirm** to validate your request.

  Pressing the Confirm softkey displays the next call log, if there are any remaining in that particular list (old or new). If all call logs are deleted in that particular list, but some remain in another list, the display shows how many call logs exist in the other list. If all the call logs are deleted in both lists (old and new), the set gives a message notification beep (provided that the message lamp is currently lit), and the display returns to the idle screen.

To delete all call log records

- Press **Callers** to display the number of new calls and old calls.
- Access the call log, and choose **Old Calls** or **New Calls**.
- Press **Delete All**.
- Press **Confirm** to validate your request.

  Pressing the Confirm softkey will display the other log list (old or new) if there are any call log records remaining. Repeat the procedure if necessary. If all the call logs are deleted in both lists (old and new), the set gives a message notification beep (provided that the message lamp is currently lit), and the display returns to the idle screen.

Family 4 (except Symbol MiNET Wireless) telephones

To program the logging of all calls or just missed calls to the call log

- Press **Superkey**.
- Press **No** until Call Logging appears.
- Press **Yes**.
- Press **Missed** or **All**.
- Press **Intl**, **Extl**, or **Both**.

To access the log of callers

- Press the feature key programmed as Callers.
- Press **New** or **Old** to obtain the log record.

  The user has the option to press Call, Delete, and Name or Number. To view the next or previous log, the user presses the up or down arrow.

To call a caller recorded on the call log

- If the caller number is an internal call, press **Call**.
- If the caller number is an external call, dial 9 and then press **Call**.
- If the caller number is external and long distance, dial 91 and then press **Call**.

To delete a call log record

- Press **Delete**.
- If you want to delete the current log, press **Yes**.

  Pressing the Yes softkey displays the next call log, if there are any remaining in that particular list (old or new). If all call logs are deleted in that particular list, but some remain in another list, the display shows how many call logs exist in the other list. If all the call logs are deleted in both lists (old and new), the display returns to the idle screen.
- If you want to delete all the logs, press **All** and press **Yes** to confirm.

Family 5 Telephones:

5330 Phones:

To program the logging of all calls or just missed calls to the call log

- Press the key you have programmed as **Superkey**.
- Press **No** until Call Logging appears.
- Press **Yes**.
- Press **Missed** or **All**.
- Press **Internal** to log calls from other extensions only, **External** to log outside calls only, or **Both**.

To access the log of callers

- Press the feature key you have programmed as Call History.
- Press **New** or **Old** to obtain the log record.

  The user has the option to press Call, Delete, and Name or Number. To view the next or previous log, the user presses the up or down arrow.

To call a caller recorded on the call log

- If the caller number is an internal call, press **Call**.
- If the caller number is an external call, dial 9 and then press **Call**.
- If the caller number is external and long distance, dial 91 and then press **Call**.

To delete a call log record

- Press **Delete**.
- If you want to delete the current log, press **Yes**.

  Pressing the Yes softkey displays the next call log, if there are any remaining in that particular list (old or new). If all call logs are deleted in that particular list, but some remain in another list, the display shows how many call logs exist in the other list. If all the call logs are deleted in both lists (old and new), the display returns to the idle screen.
- If you want to delete all the logs, press **Delete All** and press **Yes** to confirm.

5340 Phones

To program the logging of all calls or just missed calls to the call log

- Press the key that you have programmed as **Superkey**.
- Press **More** until **Call Logging** appears and then press **Call Logging**.
- Press **Missed Calls** or **All Calls**.
- Press **Internal** to log calls from other extensions only, **External** to log outside calls only, or **Both**.

To access the log of callers

- Press **Callers** to display the number of new calls and old calls.(Note: the Callers softkey is only present when call history exists.)
- Press **New Calls** or **Old Calls**, to obtain the log record.

  The softkeys Next and Previous appear if more than one call log exists. The user has the option to press Next, Previous, Call, Delete, Delete All or Backup.

To call a caller recorded on the call log

- If the caller number is an internal call, press **Call**.
- If the caller number is an external call, dial 9 and then press **Call**. Telco enters the area code if necessary.
- If the caller number is external and long distance, dial 91 and then press **Call**. The user can use up to three digits prior to Call. Telco enters the area code if necessary.

To delete a call log record

- Press **Callers** to display the number of new calls and old calls.
- Access the call log, and choose **New Calls** or **Old Calls**.
- Press **Delete**.
- Press **Confirm** to validate your request.

Attendant Consoles

To view the number of system logs in use

- Press **FUNCTION**.
- Press **ATT FUNCTION**.
- Press **CALL LOGS**.

To delete logs on a set

- Press **FUNCTION**.
- Press **ATT FUNCTION**.
- Press **STATIONS**.
- Enter the extension number of the set.
- Press **CALL LOGGING**.
- Press **DELETE ALL**, **DELETE NEW**, or **DELETE OLD**.

To disable Call Logging

- Press **FUNCTION**.
- Press **ATT FUNCTION**.
- Press **STATIONS**.
- Enter the extension number of the set.
- Press **CALL LOGGING**.
- Press **DISABLE LOGS** .

SUPERSET 430 and SUPERSET 4150 (Subattendants)

To view the number of system logs in use

- Press **Superkey**.
- Press **More** three times.
- Press **Call Logs**.

To delete logs on a set

- Press **Superkey**.

- Press **Stations**.
- Enter the extension number of the set.
- Press **Call Logging**.
- Press **Delete All**, **Delete New**, or **Delete Old**.

To disable Call Logging

- Press **Superkey**.
- Press **Stations**.
- Enter the extension number of the set.
- Press **Call Logging**.
- Press **Disable Logs**.

Call Monitor

Description

The Call Monitoring feature allows you to listen in on calls. During a call monitor, the system gives the monitoring set a one-way audio path preventing the sets included on the caller from hearing the monitoring set.

A similar feature is available as part of the ACD TELEMARKETER Applications Package. For more information, refer to Call Monitoring under ACD Supervisor and Senior Supervisor Sets.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- A Silent Monitoring access code must be assigned.
- System Option 102 (Feature Level) must be set to 6.
- System Option 42 (Silent Monitoring) must be enabled.
- COS Option 405 (Can Initiate Call Monitor) must be enabled to initiate call monitoring.
- COS Option 406 (Allow To Be Monitored) must be enabled to permit a set to be monitored.
- COS Option 655 (Continuous Call Monitor) must be enabled to permit a device to monitor a phone continuously.
- Call monitoring may be initiated by sets, consoles, POTs and DISA trunks.
- Sets and POTs (excluding RADs, VM ports and music source) may be monitored.
- Call Monitoring is not permitted on an extension that uses IP trunking and has COS option 229 (Voice mail Port) enabled.

Programming

- Assign an access code to Feature 45 (Silent Monitoring) in CDE Form 02 (Feature Access Codes).
- Enable System Option 42 (Silent Monitoring), and set System Option 102 (Feature Level) to level 6.
- In the COS of monitoring devices, enable COS 405 (Can Initiate Call Monitor). Optionallly, enable COS Option 655 (Continuous Call Monitor).
- In the COS of monitored devices, enable COS Option 406 (allow To Be Monitored).

Operation

The monitoring party initiates a call monitor by dialing the Silent Monitoring access code, followed by the extension number of the monitored set.

- If the monitored set is idle and COS option 655 is disabled in the monitoring set's class of service, the monitoring set will indicate that the monitored set is idle.
- If the monitored set is in a call and call monitoring begins, the monitoring set will display the extension number of the monitored set.
- A set can be monitored by only one party at a time. If an attempt is made to monitor a set that is already being monitored, the monitoring party will receive a busy tone and the normal busy display.
- An illegal user attempting to set up a monitor will cause an ACCESS DENIED message to appear on the set.
- An attempt to monitor someone who is not logged in will result in the monitoring set displaying that the number is invalid.
- Attempting to monitor someone who has Do Not Disturb activated and is idle will cause the monitoring set to display that the set has Do Not Disturb activated.

At any time while monitoring, the monitory party may enter the conversation by pressing the TRANS/CONF key.

If COS option 655 is disabled in the monitoring set's class of service, the monitoring party will be disconnected from monitoring when the monitored party places the caller on hold, transfers the call, or the call is terminated. The monitoring set will display "DISCONNECTED" and the system will give re-order tone.

If COS option 655 is enabled in the monitoring set's class of service, the monitoring party will not be disconnected regardless of the call state, unless the monitored party logs off. If the monitored party places the caller on hold, performs a transfer, or the call is terminated, the monitoring set will go into waiting state and display "WAITING".

The monitory party ends monitoring by pressing the CANCEL key.

Call Park from Single-line Sets

Description

This feature allows users of single-line telephones to put a call on hold (parked) and then replace the handset. The call may be retrieved at the extension at which the call was parked, or from any other extension in the system. If Music-on-Hold is available, the parked party hears music. The parking extension may not originate or receive new calls until the parked call is retrieved. It can only access paging equipment.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- If there is no music-on-hold, a caller placed on hold will hear nothing; that is, the line is silent.
- COS Options 401 (Call Park) and 403 (Trunk Recall Partial Inhibit) are mutually exclusive.
- The paging and call park feature access codes are the only access codes that the parking single-line telephone can dial after the park is done.
- The single-line telephone is considered busy by the Campon and Callback features until the parked party is retrieved.

- The parked call can only be picked up from another extension once the parking extension has gone on–hook.
- When the call is parked, the timer for COS Option 254, Call Hold Recall Timer is started.
- When the Call Hold Recall timer times out, the single-line telephone audibly starts to ring. Forwarding on the extension is ignored and the timer for COS Option 115 (Attendant Timed Recall - No Answer) is started. Music or silence continues for the parked party. When the Recall No Answer timer expires, recall handling is done; see Recall.

Programming

- Assign an access code to Feature 33 (Call Park).
- Enable COS Option 254 (Call Hold Recall Timer) for the extension, to set the Call Park recall time.
- Enable COS Option 401 (Call Park) for the extension that parks the call and for extensions using the Call Park access code to pick up the parked call.
- Enable COS Option 218 (Directed Call Pickup) for others to pick up the parked call.

Operation

To park a call:

- Flash the switchhook (standard telephone), press the FLASH key (SUPERSET 401+ or SUPERSET 4001 telephone), or press TRANS/CONF (Mitel 5201 IP Phones) - wait for dial tone.
- Dial the Call Park access code - wait for dial tone.
- Replace the handset, or access paging equipment; see Paging - PA.

To retrieve a parked call from the original extension:

- Lift the handset - connection is made.

To retrieve a parked call from another extension:

- Dial the Call Park or the Directed Call Pickup access code.
- Dial the number of the extension where the call was parked.
- The call is connected to the remote extension.

Call Park from Multi-line Sets

Description

This feature allows you to park a call from your prime line. After you park a call you can make or accept other calls from your prime line.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- You can only park one call at a time.
- You can only park a call from your prime line.
- You cannot park a conference call
- You cannot park a call if you are using an appearance of a prime line from another set.

- A parked caller hears music if music-on-hold is programmed. If there is no music-on-hold, the parked caller hears nothing; that is, the line is silent.
- After the call is parked, the timer for COS Option 254, Call Hold Recall Timer is started.
  - If the set that parked the call is programmed as a SUPERSET Telephone in Form 09, a recall to the set behaves the same as a recall from an external call on hold (see Hold Timeout).
  - A recall is identified by a single burst of tone at regular intervals (see Hold Reminder).
- If the Call Hold Recall timer times out, the telephone rings. Forwarding on the extension is ignored and the timer for COS Option 115 (Attendant Timed Recall - No Answer) is started. Music or silence continues for the parked party. When the Recall No Answer timer expires, recall handling is done; see Recall.
- You cannot use the Call Park Retrieve - Remote access code from a multi-line set to retrieve a call from another set while that set is on a call (in talk state). However, you can use the Directed Call Pickup access code to retrieve the parked call.

Programming

- Enable COS Option 211 (Call Hold and Retrieve Access) for the extension
- Enable COS Option 401 (Call Park) for the extension that parks the call and for extensions that will use the Call Park access code to pick up the parked call.
- Enable COS Option 254 (Call Hold Recall Timer) for the extension, to set the Call Park recall time.
- Enable COS Option 218 (Directed Call Pickup) for others to pick up the parked call.
- Program a CALL PARK feature key on the set (see Feature Keys).
- Ensure that the following feature access codes are programmed in Form 02:
  - Feature Access Code 33, Call Park
  - Feature Access Code 22, Call Hold Retrieve (Local)
  - Feature Access Code 23, Call Hold Retrieve (Remote).

Operation

To park a call:

Family 1, Family 3, Family 4, Mitel 5207 IP, and SUPERSET 410 telephones

- While on a call that is on your prime line, press the CALL PARK feature key.
  or
  Press the TRANS/CONF key and then enter the Call Park feature access code.
- The call is parked and you receive dial tone on your prime line.
- If a CALL PARK key is programmed on your set, the key's status indicator flashes to indicate that a call is parked.

Family 5 Telephones:

- Press the key you have programmed as Call Park.
  or
- Press TRANS/CONF, and then dial the appropriate feature access code. The display shows PARKED@ followed by the park destination and index number, if applicable.

Symbol MiNET Wireless Phone

1. Press the FCT key .
2. Use the scroll keys to locate "Trans/Conf."
3. Press the SEND key to select Trans/Conf.

4. Dial Call Park feature access code.
5. Hang up.

To retrieve a parked call from your set while your set is idle, or while you are off-hook and receiving dial tone

Family 1, Family 3, Family 4, Mitel 5207 IP, and SUPERSET 410 telephones

- Press the CALL PARK feature key
  or
- Dial the Call Hold Retrieve - Local feature access code.

Family 5 Telephones:

1. Dial the "Call Park Orbit Retrieve" feature access code
2. Dial the two-digit orbit number.

Symbol MiNET Wireless Phone:

1. Press the SEND key.
2. Dial the Call Hold Retrieve - Local feature access code.
3. Dial your extension number.

To retrieve a parked call from your set while you are on a call on your prime line

- Press the CALL PARK feature key.
  or
- Press the TRANS/CONF key and then enter the Call Park feature access code.
  You are connected to the parked call and the caller that was on your prime line is parked.

To retrieve a parked call from your set while you are on a call on your prime line, and you want both parties on the same line

- Press the TRANS/CONF key and then dial the Call Hold Retrieve - Local feature access code.
  The active call will be put on consultation hold and the held call is retrieved as the active call.

To retrieve a parked call from another set:

Family 1, Family 3, Family 4 Mitel 5207 IP, and SUPERSET 410 telephones

- At the other set, dial the Call Hold Retrieve - Remote feature access code.

  You can use any line to retrieve the call.
- Dial the prime line number (extension number) of the set that parked the call.

Note: If there is a call parked at the set and a call on hold at the set, dialing the Remote Hold Retrieve feature access code will retrieve the parked call.

Symbol MiNET Wireless Phone

1. Press the SEND key.
2. Dial the Call Hold Retrieve - Remote feature access code.
3. Dial the number of the extension that parked the call.

Call Park - Destination Phone

Description

Call Park - Destination Phone allows the called party to park an answered call on another phone. If the phone parking the call can perform set-to-set paging, then the system will automatically establish the page with the destination phone after the call is parked.

If a parked call is not retrieved after a specified length of time, a reminder occurs. This reminder type and duration is determined by COS Option 681, Key Set/Sub Att - Call Hold Notify Timer 0-600s. If the value for the option is set to 0, then the phone will ring. If the value is set from 1 to 600 seconds, a burst of tone will occur at regular intervals.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

- Available to all system-supported Mitel telephones, ONS phones, and the Superconsole 1000 or 5540 IP Console.
- Destination phones include the Family 3, Family 4, Family 5, SUPERSET 410, 430, and 4150 telephones, and the Mitel 5207 IP Phone.
- Only one parked call is permitted on a destination phone.
- Conference calls cannot be parked.

Programming

- In Form 04, System Option and Timers, enable Feature Level 5 in System Option 102.
- To program a Call Park reminder, enter a value from 0 - 600 seconds in COS Option 681, Key Set/Sub Att - Call Hold Notify Timer 0-600s, in Form 3, Class of Service Options.
- Enable COS Option 211, Call Hold and Retrieve Access, and COS Option 401, Call Park for destination phones.
- Program a Call Park key on the telephone.

Operation

Family 2, Family 3, Family 4, Family 5, SUPERSET 430 and 4150 telephones, and Mitel 5207 IP Phone:

To park an active call on another phone

- Press **Trans/Conf** to get dial tone.
- Enter the Call Park Remote FAC, and then dial the extension of the destination phone.

If the park attempt is unsuccessful, the user hears reorder tone, otherwise, the user hears dial tone, or busy tone or pages the destination phone. If the destination phone has a Call Park key, it will flash.

To retrieve a call parked on another phone after notified by a page

- Press the flashing **Call Park** key.

OR

- Dial the "Call Hold Retrieve" feature access code.

Call Park System - Specific Orbit

Description

Call Park System - Specific Orbit allows the called party to park a call in a specific orbit by entering a two-digit (01-25) orbit number or by pressing a feature key assigned to a specific orbit number.By using the feature key, and pressing it twice, the user can initiate either an All Sets Page or a PA page. The type of page initiated is determined by CDE programming.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

- For conditions on Call Park System - Specific Orbit, see Call Park System Orbit.
- Up to 25 feature keys on a set can be programmed for this feature. Each feature key on a set can only be programmed with one individual orbit number. System Option 102 must be enabled to Feature Level 6 to use these new feature keys.

Programming

- In Form 04, System Option and Timers, enable Feature Level 5 in System Option 102.
- In Form 02, program Feature Access Code 57, Call Park Orbit Retrieve.

Note: System Option 102 must be enabled to Feature Level 6 to use feature keys.

- To program Call Park System - Specific Orbit, disable System Option 127, Autoselect Park Orbits, in Form 4, System Options. This will not apply to feature keys programmed for Call Park to a specific orbit.
- Program an Orbit #*x* (with *x* representing the number of an orbit) key on the telephone to allow Call Park for a specific orbit. (See Feature Keys.).
- For park and page programming requirements, see Hold and Page.

Operation

Family 3, Family 4 (except Symbol MiNET Wireless), Family 5, SUPERSET 4150, and SUPERSET 430 telephones

To park a call in a specific orbit by entering a two-digit number

- Press the **System Park** key.
- Enter a two-digit Orbit Number (01-25).
- If the selected orbit already has a parked call, press the **Cancel** key, press the **System Park** key again, and then enter another orbit number.

SUPERCONSOLE 1000 or 5540 IP Attendant Console

To park a call in a specific orbit by entering a two-digit number

- Press the SYSTEM PARK softkey.
- Enter a two-digit Orbit Number (01-25)

If the selected orbit already has a parked call, press the ORBIT NUMBER softkey, and then enter another orbit number.

Using A Feature Key

- To park a call in a specific orbit, press the Orbit #*x* feature key corresponding to the orbit you wish to use. Once a call has been parked in a specific orbit, press the same Orbit #*x* feature key to page (see Paging - Telephones for more information on paging).
- To retrieve a call that has been parked in a specific orbit, press the Orbit #*x* feature key corresponding to the orbit in use.

Note: If a call is parked in a specific orbit (as indicated by the feature key's LED flashing), additional calls may not be parked to the same orbit.

Call Park - System Orbit

Description

This feature allows you to park up to 25 calls in a parking orbit from any line on your display set or console. When you park a call, the system assigns the call to a parking orbit, and displays the access code and the park orbit number. You can then page and inform the paged party where they can retrieve the call. After you park a call, the line that parked the call is free to make or accept other calls.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- You can park calls in orbit from Family 3, Family 4 (except Symbol MiNET Wireless), Family 5, SUPERSET 4150, and SUPERSET 430 telephones as well as the SUPERCONSOLE 1000.
- You can retrieve a call from orbit from any station, Mitel telephone, or attendant console.

Note: You cannot retrieve a parked call from orbit if device interconnection tables (CDE Form 5 and Form 30) prohibit the connection.

- The call is retrieved on the line that dials the Call Park Orbit Retrieve access code.
- The set must have a System Park key programmed to park a call in orbit.
- You can park up to 25 calls in the system orbit. The system assigns the parking orbit in a circular fashion, that is, if the last orbit assigned was 11, the next orbit will be 12, then 13 and so on up to 25 and then 01.
- You can park calls from any line on your display set.
- You cannot park a conference call or a call with a party on soft hold. Parking such a call will result in the user hearing a reorder tone, seeing a no access pop-up, and the call being put on consultation hold.
- If all the system orbits are in use, pressing the System Park key puts the calling party on consultation hold, and you receive a reorder tone.
- After you park a call in orbit, the display on your set shows the digits that must be dialled to retrieve the caller. The display shows the Call Park Retrieve access code followed by the park orbit number (for example: *402). Park orbit numbers are 01-25.
- On the SUPERSET 4150 telephone, SUPERSET 430 telephone and attendant console, the display shows the digital sequence (access code and park orbit number), the name of the caller (if available) follows. The phone number of the caller appears only if the name is not available.
- After a caller is parked, the call is treated as a call on hard hold. That means, a parked caller hears music if music-on-hold is programmed. If there is no music-on-hold, the parked caller hears nothing; that is, the line is silent.
- After the call is parked, the recall timer from the set or console parking the calls (Call Hold Recall Timer, COS Option 254 or Attendant-Timed Recall Hold Timer, COS Option 116) starts.
  - Once the timer has expired for the set, the system will place a recall to the prime line of the set that parked the call. The prime line will ring when the prime line is free and the display will look like a new call coming in. See Recall.
  - Once the timer has expired for the console, the system will place a recall to the Recall key. See Recall.
- When paging from a SUPERSET 430 or SUPERSET 4150 telephone (Directed, Group Page, All Set Page, or PA Page), the display shows the digital sequence (access code and park orbit number) of the last call parked in orbit on the second line of the display.

- When paging from a Family 4 telephone (Directed, Group Page, All Set Page, or PA Page), an Orbit softkey appears that allows the user to identify the digit sequence (access code and park orbit number) of the last call parked. After pressing the Orbit softkey, a Back softkey appears allowing the user to toggle back to the normal paging display.
- When paging, the identification of the call parked (the access code and park orbit number)
  - Stays as long as the page is in progress. When the page is converted to a normal talk state, the paging and parked party information on the display is overwritten.
  - Will not appear on the display if the call has been picked up or if the caller has hung up before the page is made.
  - Stays there even after the caller hangs up or after someone accepts the call.
- The display on an Attendant or Subattendant telephone will not identify the parked call while performing a PA page if there is a call held on a Hold key. The display identifies the held party.
- A user of a SUPERSET 430, SUPERSET 4150 telephone,or a console can repark a retrieved call using the Park Again softkey. The Park Again softkey on the SUPERSET 430 and SUPERSET 4150 telephone will be overridden by the Call Waiting softkey if the user camps on to the set subsequent to the repark. The Park Again softkey is also available to the console when the attendant answers the recall from the parked call or when the attendant retrieves the parked call.
- When a supervisor monitors an ACD agent in the talk state and the agent parks a call in orbit, the supervisor will loose the call, and the call to orbit shall proceed.
- An ONS CLASS set will not display the name of the caller after a Call Park System Orbit retrieve.

Programming

- In Form 03, program the following COS options features:

  In COS Option 254 (Call Hold Recall Timer) of the set's extension, set the Call Park recall time.

  In COS Option 116 (Attendant-Timed Recall Hold) of the console, set the Call Park recall time.
- In Form 02, program Feature Access Code 57, Call Park Orbit Retrieve.
- Program a SYSTEM PARK feature key for the SUPERSET telephones. See Feature Keys.

Operation

The operation differs for the Mitel telephones and the attendant consoles.

Family 3, Family 4 and Family 5 telephones

To park a call on system orbit:

- If the SYSTEM PARK key is lit, press the SYSTEM PARK key.
- Hang up or press NEW CALL to get a dial tone or press a line key to answer a call, unhold a call, or get a dial tone.

If all system orbits are occupied, pressing the SYSTEM PARK key puts the caller on consultation hold. You will receive a Reorder tone and your display indicates that all orbits are busy.

To return to the caller if all system orbits are occupied:

- Press the BACK TO HELD softkey or the CANCEL hardkey to return to the caller.

To retrieve a call parked on a system orbit from your set while your set is idle, or while you are off-hook and receiving dial tone

- Dial the Call Park Orbit Retrieve access code followed by the park orbit number.

SUPERSET 430 and SUPERSET 4150 telephones

To repark the call into the system orbit

- Press the PARK AGAIN softkey. The PARK AGAIN softkey only appears immediately after answering the call.

Note: With SUPERSET telephones, the CALL WAITING softkey overrides the PARK AGAIN softkey if a user camps on to the set shortly after. The PARK AGAIN softkey will override the RELEASE PRIVACY softkey if both operations are in use.

SUPERCONSOLE 1000 or 5540 IP Attendant Console

To park a call on system orbit

- Press the SYSTEM PARK key.
- Press SET PAGE to page another set, or press PAGE to PA Page, or press CANCEL or RELEASE to get back to the idle state.

If all system orbits are occupied, pressing the SYSTEM PARK key puts the caller on consultation hold. You will receive a Reorder tone and your display indicates that all orbits are busy.

To return to the caller if all system orbits are occupied

- Press CANCEL to return to the caller.

To retrieve a call parked on a system orbit from your console

- Dial the Call Park Orbit Retrieve access code followed by the park orbit number.

  The attendant can retrieve a parked call in orbit if the console is idle or when the attendant is on a call.

To repark the call into the system orbit

- Press the PARK AGAIN softkey.

  The PARK AGAIN softkey only appears immediately after answering the call.

Call Rerouting

Description

This feature provides flexibility for the routing of incoming calls, attendant access, call interception, and routing for various features. Different types of calls can be routed to different answering points in DAY, NIGHT 1, and NIGHT 2 service for each tenant. You program the call rerouting answering points in CDE Form 19 (Call Rerouting Table). Refer to Form 19 - Call Rerouting Table in the Program CDE CDE section, for a list of the call rerouting options.

Rerouted calls are processed differently than normal calls. The system considers rerouted calls to be important calls that must get through. Some features on the reroute point are ignored because of this.

Calls can be rerouted using a speed dial number. The Day, N1, and N2 fields are preceded by the letters sc, in CDE Form 14 and CDE Form 19, after speed dial is selected.This method of rerouting calls is not valid for Automatic Wake-up Routing, UCD/Attendant Recording, and UCD on HOLD Time out for this Tenant.

Also see Did/Dial-in/Tie Intercepts, Night Services, Trunk Support CO (LS/GS), and Uniform Call Distribution.

Conditions

The following conditions apply to this feature:

- Do Not Disturb is ignored on reroute points.
- Rerouted callers are automatically camped on to the reroute point regardless of COS options. Campon is not done for consoles.

- Valid access codes for reroute types, with the exceptions listed below, are: LDNs, consoles, industry-standard telephones, Mitel telephones, logical lines, hunt groups (all types), ACD paths and night bells.
- Console access codes are not valid rerouting points for the following entries in Form 19:

  DID always routing to this tenant.

  DID forward on busy/no answer.

  Dial-in Tie always routing to this tenant.

  Dial-in Tie forward on busy/no answer.
- LDN Access Codes are not valid rerouting points for Non-dial-in trunks alternate recall points.
- Only Recording Hunt Groups are valid for recording routing.
- CO trunks use rerouting when they originate, following the routing points programmed for each individual trunk; see Trunk Support CO (LS/GS).
- Forwarding is examined on a reroute point (regardless of the number of forwardings already done). See Call Forwarding.
- Whenever a DID trunk reroutes to the Night 1 point of its Attendant Access Night Point, it does not recall due to a no answer on that point. The same applies for Tie trunks and the Tie Attendant Access Night Point and for DISA trunks, and for internal extensions and the Dial 0 point.
- Device Interconnection checking is done between the rerouted party and the reroute point. For LDN keys as reroute points, the checking is done between the calling party and the console with the lowest Bay/Slot/Circuit PLID where the LDN key appears. If the connection is not allowed then the reroute point is ignored.
- Invalid or illegal call rerouting will not function if the originator has a party on soft hold, unless the originator is a ONS voice mail or a DNIC voice mail set.
- COS option 210 "Call Forwarding Inhibit on Dial-In Trunks" must be disabled to allow rerouting of incoming calls on dial-in trunks.

Programming

All Call Rerouting entries are programmed in CDE Form 19 (Call Rerouting Table). Enter a valid extension number or LDN access code in one of the DAY, N1 or N2 service columns as desired. Whether the tenant of the calling or called party is used depends upon the feature selected.

Also see Night Services.

Operation

See Attendant Night/Day Switching and Night/Day Switching for changing the night/day service.

Callback

Description

The Callback feature allows a user to be notified when a busy device becomes free or when a set has been used after a no answer condition was encountered.

Callback busy allows a user who has encountered a busy set, hunt group or trunk group to have the call completed when the required set, hunt group or trunk group becomes idle. The system continuously monitors the originating set or console and the required device. When the originating set or console is idle and the call can now be completed, the system calls the originating set or console; when that set or console answers, it calls the required device.

Callback No Answer allows a user, after dialing an extension which does not answer, to have the call completed later after the called party uses the telephone. The system continuously monitors the originating set or console and the called set. When the called set goes off-hook and then returns to idle, the callback is handled in the same way as Callback Busy.

Up to 100 Callback requests may be active within the system at any time; however, a maximum of only 25 ARS callbacks is permitted in these requests.

See Attendant Callback Busy/No Answer and Callback Busy/No Answer.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Use the Callback feature if you know that the person you are trying to contact is somewhere in the office. Use the Message - Call Me Back feature if you know that the person you are trying to contact is out of the office.
- If more than one callback request is active on any device, the requests are queued and serviced in order of placement.
- Duplicate callback requests supercede the original request.
- If the two parties involved in a callback hold a telephone conversation (not a conference) before the callback is honored, the callback is canceled automatically.
- Callbacks to a busy ARS digit string are NOT canceled if the same ARS digit string to which the callback was set is successfully dialed.
- A callback is canceled as soon as the telephone that originated it is rung, even if a conversation is not established.
- Internal callbacks outstanding for more than eight hours are canceled automatically; callbacks to a busy ARS digit string are canceled after 1 hour.
- If a callback is not answered by the originating set or console within 20 seconds, it is automatically canceled.
- If the called party becomes busy before the originating party answers the callback, the originating party hears busy tone when the callback is answered. The callback is purged.
- When a callback is placed to a hunt group, it is placed to the whole hunt group and not to any particular member.
- The scan for an idle hunt group member for the callback does not alter the next extension to hunt for in a circular hunt group.
- Callback - Busy must be activated within 10 seconds of receiving busy tone.
- Campons to extensions are honored before callbacks.
- Originate Only extensions do not have access to this feature.
- Call forwarding is ignored when a callback rings at the originating set.
- Callbacks ringing at a set cannot be picked up using CALL PICKUP.
- If a Mitel telephone originates a callback, the callback always rings the set's Prime Line. If the prime line has Key line appearances, the system places the key appearances into a simulated busy state to prevent them from ringing during the callback. If the prime line has Multi-line appearances, they remain unchanged.
- A callback only rings a Mitel or industry-standard telephone if the set and the prime line are available.
- Callbacks cannot be left on phones with COS Option 278 (Intercom Mode) enabled.

Programming

See Callback - Busy or Callback - No Answer.

Operation

See Callback - Busy or Callback - No Answer.

Callback - Busy

Description

The Callback - Busy feature allows a user who has encountered a busy telephone, hunt group, or trunk group to have the call completed when the required telephone, hunt group, or trunk group becomes idle.

See Callbacks for details on callbacks.

See Expensive Route Warning for callbacks to less expensive ARS routes.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

- Originate Only extensions do not have access to this feature.
- Callbacks cannot be left on phones with COS Option 278 (Intercom Mode) enabled.

Programming

Enable COS Option 300 (Automatic Callback) in the set's COS.

Assign an access code to Feature 20 (Callback - Busy).

For feature key activation of callback on Family 3, SUPERSET 410, and SUPERSET 3DN telephones, program a CALLBACK feature key. (See Feature Keys.)

If Callback - Busy is to be permitted on outgoing trunks, also enable COS Option 236 (Outgoing Trunk Callback) in the set's COS.

Operation

Operation varies depending on the type of telephone as described below.

Family 2 and Industry-standard Telephones:

The called extension, hunt group, or trunk group is busy. To set a callback:

- Dial the Callback - Busy access code within 10 seconds. Dial tone is returned. Your set is now ready for normal use.
- When the busy extension, hunt group, or trunk group becomes idle, your set rings.
- Lift the handset - ringback or audio for ARS dialing is heard.
- The required extension rings or the trunk is seized.

Family 3, SUPERSET 410, and SUPERSET 3DN Telephones:

The called extension, hunt group, or trunk group is busy. To set a callback:

- Press the CALLBACK feature key - dial tone is returned. Your set is available for normal use.
- When the busy extension, hunt group, or trunk group becomes idle, your set rings.
- Go off-hook - ringback or audio for ARS dialing is heard.

- The required extension rings or the trunk is seized.

Note: You can also follow the steps given for industry-standard telephones to set a Callback - Busy on SUPERSET 4015, SUPERSET 410, or SUPERSET 3DN telephones.

Family 4 and Family 5 Telephones:

The called extension, hunt group or trunk group is busy. To set a callback:

- Press the CALLBACK softkey - if you're off-hook, dial tone is returned; if you're using handsfree operation, the set goes idle. CALLBACK SET... appears briefly in the display. Your set is available for normal use.
- When the busy extension, hunt group, or trunk group becomes idle, your set rings. Your set's LCD indicates that a callback is calling.
- Go off-hook - ringback or audio for ARS dialing is heard.
- The required extension rings or the trunk is seized.

Family 1 Telephones:

The called extension or trunk group is busy. To set a callback:

- Press the CALL ME BACK softkey - if you're off-hook, dial tone is returned; if you're using handsfree operation, the set goes idle. CALLBACK SET... appears briefly in the display. Your set is available for normal use.
- When the busy extension, hunt group, or trunk group becomes idle, your set rings. Your set's LCD indicates that a callback is calling.
- Go off-hook - ringback or audio for ARS dialing is heard.
- The required extension rings or the trunk is seized.

Callback - No Answer

Description

Callback - No Answer allows a user, after dialing an extension that does not answer, to have the system complete the call after the called party returns and uses the telephone. The system continuously monitors the originating telephone and the called telephone. When the called telephone goes off–hook, the callback is handled in the same way as Callback - Busy.

See Callbacks for details on callbacks.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Callback - No Answer cannot be activated by access code while listening to ringback; see Operation.
- Callback - No Answer can be activated on extension numbers only.
- Originate Only extensions do not have access to this feature.
- Callbacks cannot be left on phones with COS Option 278 (Intercom Mode) enabled.

Programming

Enable COS Option 300 (Automatic Callback) in the set's COS.

Assign an access code to Feature 43 (Callback - No Answer).

For feature key activation of callback on a Family 3, SUPERSET 410, or SUPERSET 3DN telephone, program a CALLBACK feature key. (See Feature Keys.)

Operation

Operation varies depending upon the type of telephone as described below.

Family 2 and Industry-standard Telephones:

The called extension does not answer. To set a callback:

- Hang up.
- Go off–hook again.
- Dial the Callback - No Answer access code.
- Dial the unanswered set's extension number. Dial tone is returned. Your set is available for normal use.
- The next time the called (unanswered) set goes on–hook, your set rings.
- Lift the handset to call the extension.

Family 3, SUPERSET 410, and SUPERSET 3DN Telephones:

The called extension doesn't answer. To set a callback, follow the steps given for industry-standard telephones, or:

- Press the CALLBACK feature key - dial tone is returned. Your set is available for normal use.
- When the called (unanswered) set goes on–hook, your set rings.
- Go off–hook to call the extension.

Note: You can also follow the steps given for industry-standard telephones to set a Callback - Busy on SUPERSET 4015, SUPERSET 410, SUPERSET 3DN telephones.

Family 4 and Family 5 Telephones:

The called extension doesn't answer. To set a callback:

- Press the CALLBACK softkey - if you're off-hook, dial tone is returned; if you're using handsfree operation, the set goes idle. CALLBACK SET... appears briefly in the display. Your set is available for normal use.
- When the called (unanswered) set goes on–hook, your set rings. Your set's LCD indicates that a callback is calling.
- Go off-hook to call the extension.

Family 1 Telephones:

The called extension doesn't answer. To set a callback:

- Press the CALL ME BACK softkey - if you're off-hook, dial tone is returned; if you're using handsfree operation, the set goes idle. CALLBACK SET... appears briefly in the display. Your set is available for normal use.
- When the called (unanswered) set goes on–hook, your set rings. Your set's LCD indicates that a callback is calling.
- Go off–hook to call the extension.

Calling Party Number (CPN) Substitution

Description

The CPN substitution feature provides calling party identification for outgoing calls for purposes of network identification and call back. This feature is available for onboard digital trunk modules (Dual Link T1/E1 Dual Framer and T1/E1 Combo). CPN substitution can also be programmed on PRI cards and NSUs using IMAT (see Using the ISDN Maintenance and Administration Tool).

When a call is made using a route for which CPN is enabled, the system will attempt to replace the caller's extension number with a CPN. If none is found, the system will attempt to use a tenant-based default CPN. If both attempts fail, the call will be made with the original extension number.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- SX-200 ICP requires a Dual T1/E1 Framer module (MX/AX) or T1/E1 Combo module (CX/CXi/AX) with PRI.
- This feature applies to only embedded PRI, not to offboard PRI NSUs and PRI cards in Peripheral cabinets.
- This feature is not available with Direct Trunk Select (DTS).
- If an embedded PRI module is added to a system that already has offboard PRI NSU, Form 22 (ARS: Modified Digit Table) will require two entries: one for the embedded PRI trunks (with CPN enabled), and another for the PRI NSU trunks (with CPN disabled).

Programming

In Form 54, Calling Party Number, enter CPNs to replace extension numbers for outgoing calls. Use range programming to assign a single CPN to a block of extension numbers.

(Optional) In Form 19, Call Rerouting Table, program default CPNs for tenant groups.

In Form 22, ARS: Modified Digit Table, enable CPN for the entry that applies to embedded PRI (Dual Link T1/E1 Dual Framer and T1/E1 Combo modules), and disable CPN for the entry that applies to offboard PRI cards and PRI NSUs.

Operation

None.

Campon

Description

A device is able to indicate to a busy party that communication is desired, and to be connected when the party is free. Also, the user can make a continuing request for a trunk when the trunk group is busy, and be connected to a trunk when one becomes free.

When calling an extension, hunt group, or ARS, if the destination is busy, the caller usually receives a tone for a period and then camps on to the busy device. The tone given indicates whether or not campon is allowed during the period, or is done at the end of the period. For some calls, the period is skipped.

The busy called extension receives a tone alerting the party that there is a call waiting; see Campon Warning Tone.

An extension can consult the first waiting party (in hunt groups as well) using the SWAP CAMPON feature.

Campon may be initiated on a trunk group that has been programmed to give the expensive route warning; see Expensive Route Warning.

For recall from Campon, see Recall.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- On Family 1 telephones, the camped on party identification display takes precedence over the held party display.
- Calls camped on to a device are serviced in two groups, trunks and internal callers, with trunks being served first. Within each group calls are serviced on a first-come, first-served basis.
- When a party camps on to a busy trunk group, the DTMF receiver used to reach the trunk group is released. The campon to the trunk group is honored when a trunk group is free and a DTMF receiver is free.
- The attendant or an extension can transfer a call to a busy destination. The transferred party camps on immediately.
- The transferred party can camp on as long as the transferring party can camp on. COS checks are done on the transferring party and the transfer is disallowed if the transferring party cannot camp on.
- The transferred party hears connected music while camped on. If there is no music-on-hold, the party will hear nothing; that is, the line is silent.
- All devices except the console can camp on; consoles and extensions can transfer a call into campon. Exception: If COS Option 266, Campon Before Forward on Busy is enabled, then the attendant console cannot transfer incoming trunk calls into campon.
- If campon is allowed for a call, special busy tone is supplied for 10 seconds, after which the device is camped on automatically.
- If campon is not allowed for the call then busy tone is heard for 30 seconds and then the call is disconnected.
- DID and CO trunk calls that campon to busy devices receive ringback tone.
- Tie and DISA trunk calls that campon to busy devices receive busy tone (the same audio as internal callers).
- Campon is not supported over PRI/IP trunks programmed as Tie.
- If there is no busy intercept then a DID trunk immediately camps on to a busy device if the trunk is allowed to campon to the device.
- On busy tone timeout, a serial call is dropped, and does not recall.
- On Family 1 telephones an I WILL WAIT softkey is provided. On the Family 4 and Family 5 telephones a Wait softkey is provided. On Family 3, SUPERSET 410, and SUPERSET 3DN telephones, a CAMPON feature key can be programmed to activate campon. (See Feature Keys.)
- If a Mitel telephone is using the display or SuperKey feature while camped on to ARS, the campon is not honored when ARS becomes available; it is only honored after the telephone exits from the feature and ARS is available.
- Industry-standard telephones and Mitel telephones that camp on to a device without using a softkey hear busy tone while camped on.

- All trunk types and station and Mitel telephones can camp on.
- Calls can camp on to busy industry-standard telephones, Mitel telephones, logical lines, hunt groups, and trunk groups.
- Campon is done immediately for reroute points that are busy regardless of Campon COS options; see Call Rerouting.
- Campon is done for Automated Attendant hunt groups regardless of campon COS options; see Automated Attendant.
- Campon tones are not passed to lines which have COS Option 216 (Data Security) enabled in their COS.
- The campon warning tone applies to the party to which the call is being transferred.
- If COS Option 242 (Repeated Camp-On Beeps) is enabled, Attendant Timed Recall (Campon) is disabled.

Programming

Enable COS Option 301 (Campon) for the device to allow it to camp on.

If Campon is to be permitted on outgoing trunks, enable COS Option 237 (Outgoing Trunk Campon) for the device in addition to COS Option 301 (Campon).

Set the attendant campon recall timer through COS Option 117 (Attendant Timed Recall - Campon). See Attendant Timed Recall.

Operation

Operation varies depending upon the device as described below:

Family 2 and Industry-standard Telephones:

To camp on to a busy device:

- Dial the number - special busy tone is returned.
- After 10 seconds of special busy tone, campon is done and busy tone is returned. The called extension receives campon warning tone.
- When the called extension goes on-hook, the calling extension hears ringing tone and the called extension is rung.

  OR, if the camp on was performed on a trunk group.
- When the called trunk group becomes idle, the calling extension hears ringing tone the system dials the originally dialed digits.

Family 3, SUPERSET 410, and SUPERSET 3DN Telephones:

After the CAMPON feature key is programmed (see Feature Keys), use the following procedure.

To camp on:

- Dial the number - busy tone is returned.
- Press the CAMPON feature key. Remain off-hook.

Family 4 Telephones:

- While receiving the busy tone, press the WAIT softkey. Remain off-hook.

Family 1 Telephones:

- While receiving special busy tone, press the I WILL WAIT softkey. Remain off-hook.

To Transfer into Busy:

- Put a party on consultation hold.
- Dial the extension to transfer the call to. Special busy tone is heard.
- Hang up.

Family 5 Telephones:

- While receiving the busy tone, press the WAIT (5330) or the I'LL WAIT (5340) softkey. Remain off-hook.

Consoles:

See Attendant Transfer To Campon.

Campon Priority Over Call Forward Busy

Description

If a call to a set that is busy and has Call Forward - Busy or Busy/No Answer activated, the call is immediately forwarded. With the Campon Priority Over Call Forward Busy feature, the caller has the option of camping on to the busy set or allowing the forward to take place. This feature can be used for both calls on internal lines as well as trunk calls.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- System Option 102 feature level must be set to 6 for trunk calls.
- COS Option 301, Campon, must be enabled on the set or the incoming trunk that is performing the campon.
- COS Option 266, Campon Before Forward on Busy, must be enabled on the calling set for a set-to-set call.
- COS Option 266, Campon Before Forward on Busy, must be enabled on the set receiving the call for a trunk-to-set call. The trunk call will campon to the busy set for the duration of the COS Option 253, Call Forward - Don't Answer Timer in the called set's COS. If the called set does not answer the call, the call will be forwarded to the Call Forward busy destination when the Call Forward - Don't Answer timer expires on the trunk call.
- Limitations on trunk calls using this feature are the same as those for Campon.
- To permit attendant consoles to transfer incoming trunk calls into campon, COS Option 266, Campon Before Forward on Busy must be disabled. This condition does not apply to subattendant consoles, which can transfer calls to campon even if COS Option 266 is enabled. See Attendant Transfer to Campon.
- This feature is subject to the same conditions as Forward Campon.

Programming

- Enable COS Option 301 (Campon) on the set or the incoming trunk that does the campon (not the busy set).
- Enable COS Option 266, Campon Before Forward on Busy, for the calling set to allow it to be subject to camp on or forward
- For Mitel telephones without softkeys, program a Forward Call feature key.

Operation

Operation varies according to the set type as follows:

Family 4 and Family 5 Telephones:

- While receiving busy tone, press the Wait softkey. Remain off-hook.

Family 1 Telephones:

- While receiving busy tone, press the I Will Wait softkey. Remain off-hook.

Family 3, SUPERSET 410, or SUPERSET 3DN Telephones:

- While receiving Busy Tone, press the Forward Call feature key (its LED is illuminated at this stage of a call) and remain off-hook.

The trunk call will campon to the busy set for the duration of the COS Option 253, Call Forward - Don't Answer Timer.

Campon Warning Tone

Description

When a device camps on to an extension or hunt group, a warning tone is sent to the extension user over the current call. The warning tone can be programmed to repeat every 5 to 15 seconds.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- For internal calls from a set, the warning tone is a single burst of 440 Hz tone for 200 ms; for trunk calls and calls from consoles, the warning tone is a double burst of 440 Hz tone.
- The tone is not heard by other parties involved in the call with the busy extension, but a short silent period may be noticed.
- COS Options 242 (Repeated Camp-On Beeps) and 216 (Data Security) are mutually exclusive.
- COS Option 216 (Data Security) prevents the tone from being applied to particular extensions. This has no effect on the SWAP CAMPON feature.
- The tone is not heard on ONS phones that are connected to AMB, AOB, and ASU circuits if COS Option 510 (ONS Stations Support CLASS Visual Call Waiting) is disabled in the Class of Service of the phones.
- Repeated campon beeps applies only when trunks are camped on.
- Trunks camped onto extensions with Repeated Campon Beeps do not recall.
- For hunt groups, the tone is given to the first busy member of the hunt group that is not Do Not Disturb and that is logged in (UCD).
- The COS of this member is used to check if repeated campon beeps on a hunt group applies to the first party in the hunt group. The tone is applied irrespective of the line that is in use.
- If the extension selected from the hunt group for the tone has Data Security enabled then no tone is given. No other extension is selected.
- The tone is not given to members of recording hunt groups.

- The feature applies to Mitel telephones only if their prime line is a key line (in which case all parties on the line get the tone) or their prime line has no appearances and they are on their prime line appearance.
- The feature applies to logical lines only if the line is a key line or a single appearance multicall line. The first Mitel telephone where the line appears gets the tone. All parties on the key line get the tone.
- The warning tone is applied only if the party to get the tone is talking to another party or is held by another party.
- Music is removed while the campon tone is applied (digital Bays only).

Programming

Enable COS Option 242 (Repeated Camp-On Beeps) for the extension if trunk calls should continue to notify the extension. For hunt groups, this option should be enabled for the first extension in the hunt group.

Enable COS Option 216 (Data Security) for the extensions that should not receive the warning tone. For hunt groups, this option should be enabled for each extension in the group.

Set the cycle time for the repeated beeps via the trunk's COS Option 255, Repeated Camp-On Beeps Timer. The default setting is 10 seconds.

Operation

A call camps on and a tone is heard. For Mitel telephones, the SWAP CAMPON feature can be used to consult with the party that camped on. Alternatively, for any extension, finish the current call; hang up, and the camped on party rings the extension.

Centralized Attendant on Release Link Trunk

Description

The Centralized Attendant feature allows an attendant or Subattendant on a PBX to answer calls that arrive at another interconnected PBX. The attendant transfers the call back to the remote PBX using the same trunk. Once the call is transferred back, a Release Link Trunk (RLT) releases the tie line between the two PBXs.

The call arrives at the attendant via a dedicated RLT, which can be T1 E&M, T1 E&M DISA, E&M, or E&M DISA. When the attendant releases a call to its destination, the RLT is released.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

Both PBXs must be interconnected by at least one Release Link Trunk.

System option 38, Switch-hook Flash does not apply to Centralized Attendant.

The Disconnect timer must be set greater than the Flash timer.

No conference is permitted between both PBXs and the attendant while using the RLT.

A party on soft hold by an RLT will not ring back to the RLT if the attendant hangs up. It will route either to its Dial 0 point or to whatever tone the RLT was listening to at the time of hangup.

The RLT receiving the flash must not be programmed as a line key (for example, DTS or CO line). Flash Over Trunk (feature access code 46) and Double Flash Over Trunk (feature access 53) cannot be performed on a RLT. The following flash-related COS options do not apply to RLTs or to subattendants who are talking to one other party who is an RLT:

COS 203, Broker's Call
COS 205, Flash For Waiting Call
COS 212, Can Flash If Talking to an Incoming Trunk
COS 213, Can Flash If Talking to an Outgoing Trunk
COS 214, Cannot Dial a Trunk after Flashing
COS 215, Cannot Dial a Trunk if Holding or in Conference with One
COS 223, Flash Disable
COS 224, Flash for Attendant
COS 252, Broker's Call with Transfer
COS 302, Flash-in Conference

The SERIAL and FLASH softkeys are not available to the console when talking on an RLT.

A central office switch that hosts a voice mail service for PBX users, can receive the billed extension numbers for calling parties that have calls forwarded to their voice mail. This feature enhancement is dependant on the purchasable System Option 113 (Centralized Attendant/Voice mail) being enabled. A new special digit sequence *05 in CDE Form 22 enables this functionality.

Programming

In Form 04, System Options and Timers, set System Option 113, Centralized Attendant / Voice mail, to enabled (must be purchased).

In Form 01, System Configuration, program required T1 E&M trunks or E&M modules.

In Form 02, Feature Access Codes, for the local PBX (with the attendant) program feature 11, Extension General Attendant Access with an access code (usually 0). For the remote PBX (without the attendant), program Feature 11, Extension Attendant General Access also with an access code (usually 0).

In Form 13, Trunk Circuit Descriptors, program a new separate trunk circuit descriptor for the E&M trunk(s). Programming must be the same at each PBX for the trunks used for Centralized Attendant.

E&M Trunk Module:

- disconnect timer 900 (greater than flash timer)
- flash timer 300
- Release Link Trunk Yes
- E lead invert as required
- M lead invert as required
- DTMF Yes

T1 E&M trunk(s):

- disconnect timer 900 (greater than flash timer)
- flash timer 300
- Release Link Trunk Yes
- Incoming Start Type as required (must match at each end)
- Outgoing Start Type as required (must match at each end)
- DTMF Yes

In Form 31, System Abbreviated Dial Entry, program a unique index number which translates to the Centralized Attendant via ARS.

In Form 03, Class of Service Options, select a unique and separate COS for the E&M trunk(s), and program trunk COS and Tenant options, as required:

In Form 03, Class of Service Options, for each telephone that is to have Attendant access, set COS Option 239, Priority Dial 0, to Enabled

In Form 14, Non-Dial-In Trunks or Form 15, Dial-In Trunks, as required, program the Centralized Attendant trunks with the CDN from Form 31.

In Form 16, Trunk Groups, select a unique trunk group and program these centralized attendant trunks into that group. Set SMDR, Terminal or Circular, as required.

In Form 19, Call Rerouting Table, for Priority Dial 0 Routing, use the Speed Dial softkey to program Day, Night1, and Night2 destinations with a speed dial to the abbreviated bin number programmed in Form 31.

In Form 22, ARS Modified Digit Table, insert *05 (Insert Billed Extension) in the digit string to be sent to the voice mail system, if you want the identity of the last extension that forwards a call to go to the CO.

Operation

A call to the Centralized Attendant arrives on a RLT to be transferred to a destination.

If the console is the attendant, just dial the destination, a flash is sent out on the RLT (the calling party is put on soft hold) followed by the dialed digits. The attendant can press CANCEL to retrieve the call over the RLT or RELEASE to disconnect the RLT and complete the transfer.

At a subattendant telephone (Family 1 or Mitel 5020, 5220, 5224, or 5324 IP Phone), press TRANS/CONF to send a flash on the RLT; when dial tone is returned, it can dial the destination. The subattendant can press BACK TO HELD to retrieve the held party over the RLT or press RELEASE/HANG UP to disconnect the RLT and complete the transfer.

CENTREX Compatibility (Double Flash Over Trunk)

Description

This feature provides the ability to send a double switchhook flash out over a trunk. Flashing over a trunk enables a telephone on the system to use CENTREX features. Callers must follow the instructions specified by the local central office concerning when a double flash should be used.

A CENTREX caller can put a CENTREX extension on softhold and dial another extension (by sending out a Flash Over Trunk followed by the digits of the extension). If the caller hears busy or ringing, the caller can put the CENTREX extension on softhold again, and dial the Double Flash Over Trunk feature access code. This sends out two flashes on the trunk, with one second between the flashes. When the CENTREX service receives the double flash, it clears down the ringing/busy tone. The caller returns to a talking state with the trunk that was on softhold.

Using Abbreviated Dial or Speed Call

The access code for Flash Over Trunk, followed by digits which activate CENTREX features may be programmed into a speed call key, or a system abbreviated dial number. Any central office access code with an asterisk (*) in it must be entered as **. After inserting the centrex feature in the speed dial number, the speed dial must not contain 9 and the local or id number; otherwise, pressing the speed dial key results in an invalid number being displayed on that set. In summary, the abbreviated dial number must not contain a local or id number. See Abbreviated Dial in this document for information on special codes.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Only one SMDR record will be generated by a call using this feature.
- The trunk must be an 8-circuit LS/CLASS trunk, 6-circuit CO (LS/GS) trunk, a T1 LS/GS trunk, or an AMB CLASS.
- The trunk must be in a trunk group in order to flash over the trunk.
- This feature may be used by telephones only.
- This feature cannot be used with a PBX conference on consultation hold. It only works if there is one trunk on consultation hold.
- Extensions cannot send a flash signal over analog trunks.

Programming

Perform the following:

- To allow a CENTREX caller to flash over the trunk, enable COS Option 257 (Flash Over Trunk) in the COS of the telephone(s) permitted to flash over trunks.
- Assign a feature access code to Feature 53 (Double Flash Over Trunk) in Form 02 (Feature Access Codes).
- In the Select Options Subform of CDE Form 13 (Trunk Circuit Descriptors), set the Flash Over Trunk option to YES and set the Flash Type option to either LOOP FLSH or RING GND (if applicable), and set the Flash Timer to an appropriate value (if applicable).
  Note: The trunk must be in a trunk group.

Operation

Generally:

- While on a busy or ringing CENTREX trunk, do one of the following:
  - press the TRANS/CONF key
  - press the FLASH key (SUPERSET 4001 or SUPERSET 401+ telephones)
  - flash the switchhook (industry-standard telephone).
- Dial the Double Flash Over Trunk access code.

Or

- Press the DOUBLE FLASH key on multi-line (Family 3 , Family 4, and Family 5 (5330)) Mitel telephones

Example CENTREX Transfer (Access Code Method):

- Answer call from a CENTREX line.
- Press the TRANS/CONF key, press the FLASH key (SUPERSET 4001 or SUPERSET 401+ telephones), or flash the switchhook (industry-standard telephone). The CENTREX trunk is placed on softhold.
- Dial the Flash Over Trunk feature access code.
- Dial the CENTREX number of a second CENTREX extension.
- When the second CENTREX extension does not answer or is busy, press the TRANS/CONF key, press the FLASH key (SUPERSET 401+ telephones), or flash the switchhook (industry-standard telephone). Until the ARS unknown digit length timeout expires, the flashook is ignored and the TRANS/CONF key or FLASH key is not available.

- Dial the Double Flash Over Trunk feature access code. (The CO cancels the second call, clears the ringing/busy tone and reconnects the caller to the softheld CENTREX party. The system then returns the caller to the original softheld CENTREX trunk.)

Example CENTREX Transfer (Feature Key Method):

- Answer call from a CENTREX line.
- Press the SINGLE FLASH key on multi-line (Family 3 , Family 4 and Family 5) Mitel telephones.
- Dial the CENTREX number of a second CENTREX extension.
- When the second CENTREX extension does not answer or is busy, press the FLASH key on SUPERSET 410, SUPERSET 420, SUPERSET 430, SUPERSET 3DN, and SUPERSET 4DN telephones. (The CO cancels the second call, clears the ringing/busy tone and reconnects the caller to the softheld CENTREX party. The system then returns the caller to the original softheld CENTREX trunk.)

CENTREX Compatibility (Single Flash Over Trunk)

Description

This feature provides the ability to send a switchhook flash out over a trunk. Flashing over a trunk allows for the use of CENTREX features by telephones within the system. Callers must follow instructions specified by the local central office concerning which access codes to dial, and when to wait for dial tone. After sending the flash over the trunk, the system will wait for dial tone from the central office, or for the Limited Wait For Dial Tone timer.

Using Abbreviated Dial or Speed Call

The access code for Flash Over Trunk, followed by digits which activate CENTREX features may be programmed into a speed call key, or a system abbreviated dial number. Any central office access code with an asterisk (*) in it must be entered as **. After inserting the centrex feature in the speed dial number, the speed dial must not contain 9 and the local or id number; otherwise, pressing the speed dial key results in an invalid number being displayed on that set. In summary, the abbreviated dial number must not contain a local or id number. See Abbreviated Dial in this document for information on special codes.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Only one SMDR record will be generated by a call using this feature; also, digits dialed after the flash are not appended to the SMDR record.
- The trunk must be an 8-circuit LS/CLASS trunk, 6-circuit CO (LS/GS) trunk, a T1 LS/GS trunk, or an AMB CLASS.
- The trunk must be in a trunk group in order to flash over the trunk.
- Ensure that the DTMF option in CDE Form 13 - Trunk Circuit Descriptors (Select Options Subform) is set up to the option which is understood by the local central office. If this is not done, the local central office will not receive the digits dialed after the flash.
- If the user makes an error while dialing digits to the local central office, the user must wait for it to timeout and return to the previous state (there is no CANCEL function, and no back arrow key).
- This feature may be used by telephones only.

- The feature will not function if there is a conference on consultation hold. It will only function if there is one trunk on consultation hold.
- Extensions cannot send a flash signal over analog trunks.
- For Abbreviated Dial or Speed Call Key to function as described in the Description, COS Option 816 must be removed from the COS of the trunk permitted to flash.

Programming

Perform the following:

- To allow a CENTREX caller to flash over the trunk and go directly into dial state, enable COS Option 257 (Flash Over Trunk) in the COS of the telephone(s) permitted to flash over trunks.
- To allow a manual CENTREX call to go directly into talk state after a Flash Over Trunk is performed, enable COS Option 816 (CENTREX Flash Over Trunk) in the COS of the trunk permitted to flash. Note: The trunk must be in a trunk group.
- To allow an Abbreviated Dial or Speed Call CENTREX call to function as described in the Description, disable COS Option 816 in the COS of the trunk permitted to flash.
  Note: The trunk must be in a trunk group.
- Assign a feature access code to Feature 46 (Single Flash Over Trunk) in Form 02 (Feature Access Codes).
- In the Select Options Subform of CDE Form 13 (Trunk Circuit Descriptors), set the Flash Over Trunk option to YES, set the Flash Type option to either LOOP FLSH or RING GND (if applicable), and set the Flash Timer to an appropriate value (if applicable). Note: The trunk must be in a trunk group.

Operation

Generally:

- While talking on a trunk CENTREX trunk, do one of the following:
  - press the TRANS/CONF key
  - press the FLASH key (SUPERSET 4001 or SUPERSET 401+ telephones)
  - flash the switchhook (industry-standard telephone).
- Dial the Single Flash Over Trunk access code. Or press the DOUBLE FLASH key on multi-line (Family 3, Family 4 and Family 5 (5330)) Mitel telephones.
- Dial the applicable CENTREX codes, and follow the CENTREX instructions.

Example CENTREX Transfer:

- Answer call from CENTREX line.
- Press the TRANS/CONF key, the FLASH key (SUPERSET 4001 or SUPERSET 401+ telephones), or flash the switchhook (industry-standard telephone).
- Dial the Single Flash Over Trunk feature access code. Or press the SINGLE FLASH key on multi-line (Family 3 , Family 4, and Family 5) Mitel telephones.
- Dial the CENTREX number of the second external party.
- Announce the caller.
- Hang up. The caller is transferred within the CENTREX network and trunk is released.

Close the contact and a call is made to the Dial 0 Routing point or to the Priority Dial 0 Routing point.

CLASS for Analog Telephones

Description

CLASS (station side) for Analog Telephones allows the SX-200 ICP system to pass Calling Line ID digits and CLASS (Custom Local Area Signaling Services) name information through to analog stations (display sets), that support Caller ID functionality. The analog telephones show the calling party name and number during the ringing state and talking state. The CLASS message also activates/de-activates the message lamp.

This feature supports

- CLASS name
- Calling party ID (if available) to a ringing set
- Calling party ID (if available) to a set in talk state (campon)
- CLASS activation and de-activation of the message lamp

If a hotel operator chooses not to display the caller room number between guest room calls, the number can be blocked with COS Option 507, Station/Set: Allow My Number to be Displayed, set to disabled. The default is enabled.

If a hotel operator chooses to display the caller room number between guest room calls, the hotel operator may want to share this information with their guests and display a placard on the telephone saying "Be Aware That Caller ID Functionality Is In Service On These Phones".

To maintain privacy in a hotel/motel environment, the CLID information is cleared on checkout.

CLASS (station side) for Analog Telephones is a purchasable option.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- SX-200 Peripheral Cabinets require an ONS/CLASS Line card and the BCC III with a DSP module (single). **Note**: BCC III requires SX-200 ELx cabinet.
- The trunk side CLASS functionality requires a LS/CLASS Trunk card, T1 (ANI/DNIS), or ISDN PRI.
- The eight CLASS generators on the DSP module are assigned and released as they are required. The CLASS generators are only assigned while information transmits to the set.
- COS Option 502, Display ANI/DNIS/CLASS Information, must be enabled for the analog sets.
- COS Option 505, ONS Stations Support CLASS must be enabled.
- COS Option 507, Station/Set: Allow My Number to be Displayed controls whether or not the number of a station or set appears on another station or set on the same system. The blocking of the number works independently of who is the caller and who is the called party.

  If COS option 507 is disabled and Set A calls Set B, Set B will not see the number belonging to Set A. If Set B calls the front desk and requests to be connected to another guest and is transferred to set A, set B will again not see the number belonging to set A. The default for COS Option 507 is ENABLED (showing the internal number).
- COS Option 508, Station/Set: Show Internal Numbers on My Phone, overrides COS Option 507. COS Option 508 allows a station or set to always see the number of the other party. Administration telephones such as room service would use this COS option.
- The purchasable System Option 89, CLASS Functionality for ONS Sets must be enabled.
- The analog telephone must interface to the ONS/CLASS Line card to obtain the CLID information.
- The Calling ID of the originating outside caller is dependant on the programming of COS Option 273, Display Held Caller ID to ONS/CLASS, for the transferring party. The ONS CLASS set will display the caller ID of the outside original caller if the transferring party has COS Option 273 enabled (the default setting is disabled). If COS Option 273 is disabled for the transferring party or if the held party

is an internal telephone, the ONS CLASS set will show the caller ID of the transferee i.e. attendant, automated attendant, voice mail etc.
- Mitel telephones with a SIM2 do not support CLASS.
- A CLASS command clears stored calling/called numbers and names on ONS sets when a hotel-guest checkout occurs.
- Third-party telephones may not provide the expected functionality.
- Telephones with COS option 505 enabled, cannot take part in a Message Lamp Test. CLASS protocol controlling the message waiting lamp interferes with this.
- An ONS CLASS set will not display the name of the caller after a Call Park System Orbit retrieve.
- The timing co-ordination for the name and number on PRI trunks is controlled with COS Option 813, Delay ONS Ring - Wait for Network Name (0-6 s).

Programming

In Form 01, System Configuration, program the ONS/CLASS Line card as an ONS LINE CARD.

In Form 03, Class of Service Options, enable COS Option 505, ONS Stations Support CLASS and COS Option 502, Display ANI/DNIS/CLASS Information. Program COS Option 507 or COS Option 508 in order to block/or not to block the display of the internal number.

If you want analog telephones to display the CLID of the originating outside caller as opposed to the transferee for transferred calls, enable (default setting) COS Option 273, Display Held Caller ID to ONS/CLASS, for the transferee. Transferees include attendants, subattendants, automated attendants, Mitel Speech Server ports, and voice mail ports.

If you want the analog telephones to receive the CLID of the transferee, disable COS Option 273, Display Held Caller ID to ONS/CLASS, for the transferee.

In Form 04, System Options/Timers enable System Option 89, CLASS Functionality for ONS Sets.

Operation

None.

CLASS for Digital Sets

Description

CLASS (Custom Local Area Signaling Services) for Digital Sets allows the SX-200 ICP system to receive Calling Line ID digits or CLASS name on incoming CLASS trunks. The calling directory number or name is presented to Mitel display telephones, to Mitel consoles, to SMDR printers, and to the Mitel Application Interface (MAI) platform package. The CLASS feature is available with the LS/CLASS Trunk card in the rackmount cabinet.

On a Mitel console, the Calling Line ID digits or the name for an incoming call are displayed after the call is answered. A toggle softkey (CALLER NAME or CALLER NO.) allows the user to switch between CLASS name or calling number.

On the SUPERSET 4150 and the SUPERSET 430 telephone, both name and Calling Line ID digits of an incoming call are displayed.

On Family 4 and Family 5 telephones, the display shows the Calling Line ID digits or the name while the telephone is ringing, and a toggle softkey (CALLER NAME or CALLER NO.) allows the user to switch between CLASS name or calling number display while the call rings. After the call is answered, the CLASS name or calling number remains on the display. If the Calling Line ID digits or the name are not available, then the display shows the normal trunk information.

The central office sends CLASS modem tones, which the SX-200 ICP system receives on LS/CLASS trunks. This tone is then encoded, and the PCM representation of the tones is decoded by the LS/CLASS Trunk card for delivery to the telephone's display.

The system can be programmed to record the CLASS digits and name in the trunk SMDR reports. The CLASS digits, up to a maximum of 10, are presented in columns 90 through 99. The CLASS name, up to a maximum of 15 letters, is recorded in columns 114 to 128. Both fields are optional.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Once the call is answered on a Family 4 or Family 5 telephone, whichever CLASS display is currently selected (name or number) will be displayed for the remainder of the call, the CLASS toggle softkey will not be available.
- For the Mitel Family 4 or Family 5 telephones, the CLASS toggle softkey will only be available when both name and number information is available and when COS 502 is enabled for the set/console.
- The SUPERSET 4150 and the SUPERSET 430 telephone displays both name and number of an incoming call. There is no toggle switch.
- The trunk must be a CLASS trunk, connected to a LS/CLASS Module or LS/CLASS Trunk Card.
- Only Mitel telephones with displays receive CLASS Digits and Name.
- For CLASS functionality on ONS circuits, see CLASS for Analog Telephones.
- The Calling Line ID (CLASS) digit display takes priority over trunk group names, trunk names, and trunk numbers.
- The CLASS name takes priority over CLASS number whenever information about the trunk is to be displayed on the Mitel display telephone or console.
- If the system doesn't receive any CLASS digits for a call, then the console or set display shows the trunk information of the call instead.
- CLASS digit displays follow standard call forwarding rules (See Call Forwarding).

Programming

Mitel telephone programming

- To program the system to display Calling Line ID digits on a Mitel display set or console, enable COS Option 502 (Display ANI/DNIS/CLASS Information) in the class of service for the set/console.
- To program the system to display CLASS name before Calling Line ID digits on a Mitel display set or console, enable COS Option 503 (Display CLASS Name) in the class of service for the set/console. Note that COS Option 502 will also be required. When COS Option 503 is not enabled, the Calling Line ID digits will be displayed before the CLASS name.

Trunk programming

- To have Calling Line ID digits reported in the SMDR records, enable COS Option 806 (SMDR-Record Incoming Calls) and COS Option 814 (SMDR-Record ANI/DNIS/CLASS) in the class of service for the trunk.
- To have CLASS name reported in the SMDR records, enable COS Option 246 (SMDR-Extended Record) and COS Option 814 (SMDR-Record ANI/DNIS/CLASS) in the class of service for the trunk.
- In Form 13, assign the CLASS trunk with the circuit descriptor “4-CIRCUIT CLASS” if using a LS/CLASS module, or assign “8-CIRCUIT CLASS" if using a LS/CLASS Trunk card.
- In the circuit descriptor options subform of Form 13, program the option “CLASS Trunk”.

- Program the CLASS trunks as non-dial-in trunks in Form 14 or as dial-in trunks in Form 15.

Operation

Operation varies depending upon the device as described below.

SUPERSET 4150 and SUPERSET 430 Telephones:

While the set is ringing

- No operation is required because the SUPERSET 4150 and SUPERSET 430 telephones display the name and the number of the incoming call.

Family 4 and Family 5 Telephones::

While the set is ringing:

- Press the CLASS toggle (Caller No. or Caller Name) softkey to switch display between name and number.

When the call is answered, the whichever CLASS display that is currently selected (name or number) will be displayed for the remainder of the call, and the CLASS toggle softkey will not be available.

Consoles:

- Press ANSWER key.
  - CLASS name or Calling Line ID digits are displayed.
- Press the CLASS toggle (CALLER NO. or CALLER NAME) softkey to toggle the display.

Class of Restriction (COR)

Description

Fifty Class Of Restriction (COR) groups are available in the system to provide 50 different levels of outgoing call capabilities. Each extension, Mitel telephone, dataset, console or dial-in trunk is assigned a COR which defines the outgoing call capabilities for that device. All devices with the same COR have access to the same outgoing call capabilities. By using CORs, the amount of ARS programming is reduced.

The Class of Restriction allows the system to restrict which trunk can not be accessed by a user. For example, COR 01 could restrict users from accessing CO trunks (local and DDD), WATS, and TIE lines. COR 02 could restrict users from accessing WATS, tie lines, and DDD (but allow local calls). COR 03 could restrict users from accessing DDD and WATS (but allow local calls and tie line calls).

Note: All extensions belong to COR group 1 in the default database.

Conditions

The following conditions apply to this feature:

- CO trunks do not have a COR. When they access ARS, it is either using the COR of the transferring or forwarding party, or no COR at all, giving universal access.
- The COR may change temporarily for a caller through Verified Account Codes.
- The maximum dialed digits feature is based on the COR of the caller. Refer to the Automatic Route Selection and Toll Control section.

Programming

Assign a COR number to each extension, Mitel telephone, dataset, Dial-In Trunk and console via CDE Form 09 (Desktop Device Assignments), 07 (Console Assignments), 15 (Dial-In Trunks) and 12 (Data Assignment).

Refer to the Automatic Route Selection and Toll Control section, for additional information on COR programming.

Operation

None.

Class of Service (COS)

Description

Each extension, trunk, Mitel telephone, dataset, ACD position, or console is assigned a Class Of Service (COS) which defines the features available for that device. All devices with the same COS (which defines the COS Options) have access to the same features. Each device type should have its own COS. Fifty Classes Of Service are available in the system to provide 50 different levels of feature accessibility. Each COS can have a name associated with it.

In the default database there are seven preprogrammed Classes of Service: 1 = IP Sets, 2 = ONS, 3 = Sub Attendant, 4 = Console, 5 = LS Trunks, 6 = Voice Mail, 7 = IP Trunks. See Defaults by Form for COS definitions.

Conditions

The following conditions apply to this feature:

- Ringing patterns for CO Line keys is selectable. See Distinctive Ringing.
- Several COS options are mutually exclusive; these are identified with the description of each feature.
- The COS may change temporarily for a caller through Verified Account Codes (Traveling Class Marks).
- The COS used is sometimes that of the caller and sometimes that of the called party. Check the feature to determine whose COS is used.
- LDNs do not have a COS. The COS for an LDN is determined from the lowest Bay/Slot/Circuit PLID of the console where the LDN is programmed.
- Logical lines do not have a COS. The COS is determined from the first Mitel telephone on which it is programmed. See Logical Lines.

Programming

Assign the desired features to each COS via CDE Form 03 (COS Define).

Assign a COS to each console, extension, Dial-In Trunk, Non Dial-in Trunk, dataset, and ACD position via CDE Forms 07, 09, 12, 14, 15, 39 and 40.

Operation

None.

Clear All Features

Description

An extension user may cancel all Call Forwarding, Do Not Disturb, and Callbacks active at that extension.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

None.

Programming

Enable COS Option 221 (Clear All Features) for the extension.

Assign an Access Code to Feature 25 (Clear All Features).

Operation

To cancel all Call Forwarding, Callbacks, and Do Not Disturb:

- Lift the handset - dial tone is returned.
- Dial the Clear All Features feature access code.
- Dial tone is returned.
- Replace the handset.

CO Line Group Key

Description

This type of key allows a selection of an idle CO line from a CO line group. The key accesses a group of CO lines without having a dedicated appearance for each line on the set. Toll control is handled by ARS. The LCD or LED indicator corresponding to the key has no function.

Feature Availability

Click here to see a list of the phones that can use this feature..

Conditions

The following conditions apply to this feature:

- The CO line group key needs an available internal line to make the call. The CO line group key cannot be used while listening to external dial tone.
- If all the CO lines in the CO line group are busy when a user presses the CO line group key, the user hears busy tone. The user then has the option of camping on, or setting a callback (see Campon or Callback-Busy). If the user sets a callback, the callback will ring the Prime key when a CO line becomes available.
- Redial does not operate on a CO line group key.

Programming

The following programming steps are required:

- Enter trunks in CDE Form 14 (Non-Dial-In Trunks) for the CO line group.
- Assign those trunks to CO Line Groups in CDE Form 16 (Trunk Groups).
- Enter ARS leading digits into CDE Form 26 (ARS: Digit Strings) for the CO line groups.

- Enter the CO line group numbers into CDE Form 23 (ARS: Route Definition).
- Program the remaining ARS tables (CDE Forms 20 - 27).
- Enter the CO line group key and the associated ARS leading digit(s) into the Expand Set Subform of CDE Form 09 (Desktop Device Assignments).

Operation

To originate a call:

- If the origination line preference is programmed as a CO line group key, all the user has to do is to go off–hook to originate the call, providing there is a free internal line.
- Acquire internal dial tone (via the Prime key, or some other means).
- Press the CO line group key.
- Dial the required digits to complete the call.

CO Line Key

Description

This key type originates and answers calls to or from parties outside the system. The key accesses a specific trunk directly. A CO line key may be shared by up to 64 sets, but only one may access it at a time. One other party can join in on a call on the line if the CO line is non-private, or privacy is released. Dialing is checked against Key System Toll Control; if a call is not answered, it reroutes as programmed in CDE Form 19.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- A CO line key can appear on a maximum of 64 IP sets, or 32 IP and 32 DNIC sets.
- Ring variants can be programmed independently for each appearance.
- Calls made via a CO line key are subject to key system toll control restrictions.
- Auto answer will not operate on a CO line key.
- If another key is selected while dialing on a CO line, the CO line will be cleared down. If another key is selected while there is a call on the CO line, the CO line call will be put on hold.
- When camping on to a busy CO line, there is no campon warning tone given.
- When a callback is set on a busy CO line, the call will be returned to the Prime line key. When the callback is answered, the call will switch back to the CO line key - both must be idle for a successful callback.
- On SMDR records, the ARS leading digit field will be blank for outgoing CO Line calls.
- The back arrow, for deleting misdialed digits, is not available when dialing on the CO line key.
- Incoming calls will follow call forwarding programmed on the set only if COS Option 815 "DTS/CO Line Key Honors Forwarding" is enabled for the trunk and the CO Line key appears on only one set.
- The user can transfer calls on this line to other extensions. The behavior of the call on the extension receiving the transfer depends on whether the extension has an appearance of the line and whether Option 689 (DTS/CO Line Transfer Call Handling) is enabled in the extension's Class of Service. For more information, see the Conditions section in the Transfer feature description.

Programming

The following programming steps are required:

- Enter the trunk into CDE Form 14 (Non-Dial-In Trunks).
- If answer points are programmed for the trunk in Form 14, delete them. Deleting the answer points will allow calls to ring the CO line key, and allow the user to access the trunk using the CO line key.

Note: If answer points are programmed then the calls will be directed to the answer points based on the system service mode (Day, Night 1, or Night 2). Although the calls will be indicated on the set by the flashing line key appearance, the set user will not be able to answer the call or access the trunk.

- Enter a routing point for CO Line Routing Points on No Answer in Form 19 (Call Rerouting Table).
- Enter the trunk into a group in CDE Form 16 (Trunk Groups).
- Enter the CO line key into the Expand Set Subform of CDE Form 9 (Desktop Device Assignments).
- Enter digit strings into CDE Form 46 (Key System Toll Control) for analysis of dialed digits on the CO line. To add to restriction, enter a suitable COR.
- Enter COR group members into CDE Form 20 (ARS: COR Group Definition) to restrict users from dialing specific digit strings.
- Specify the maximum permitted number if dialed digits for each COR group in CDE Form 27 (ARS: Maximum Dialed Digits).

Operation

To originate a call:

- Press an idle CO line key

or

- Go off-hook (if CO line key is the preference -- see Line Preference).
- Dial outgoing digits; ARS codes are not required.

To answer a call:

- Press the ringing CO line key

or

- Go off-hook (if it is the first ringing line).

CO Line - Select Direct

Description

This feature allows a user to directly access a specific CO trunk which may or may not appear on the user’s telephone. This feature must be accessed through any internal line.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Calls made in this manner are subject to key system Toll Control restrictions.
- If the CO trunk is busy, the user is not permitted to barge in under any circumstances.
- If the CO trunk is busy, the user may set up campon or callback. If callback is set up, the call is returned to the prime line.

- It is possible to use speed call to access this feature: store the Direct CO Line access code and the three-digit trunk number.

Programming

The following programming steps are required:

- Enable COS Option 680 (Key System - Direct CO Access) in the telephone’s COS.
- Assign an access code to Feature Access Code 50 (Direct CO Line Select) in Form 02 (Feature Access Codes).
- See CO Line Key for information on toll control programming.

Operation

To access a CO trunk:

- Dial the Direct CO Access access code.
- Dial the 3-digit trunk number. (Trunk numbers are 1 thru 200 - if a user is accessing trunk number one, one, two or three digits can be dialed, e.g., 1, 01 or 001.)

Note: It is possible to use speed call to access this feature: store the Direct CO Line access code and the three-digit trunk number. See Personal Speed Call.

CO Line Type - Direct Access - Bypass Key System Toll Control

Description

This feature allows an extension seizing a CO trunk with a line key to bypass dial tone and Key System Toll Control. Instead, dial tone from the CO is immediately received. This allows users to hear stutter CO dial tone on their CENTREX lines, indicating the presence of voice mail messages. Users may then access their voice mail or other CENTREX features (see CENTREX Compatibility (Flash Over Trunk)) or they may dial an external destination number.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- The trunk must be a 6-circuit CO trunk, a 6-circuit DISA trunk, a 4-circuit CLASS trunk, a T1 (LS/GS, DID/TIE, or E&M) trunk, an E&M module or a T1 DISA (TIE, E&M, or CO).
- Personal Speed Call does not work on direct CO line access because no digits are analyzed, but Speed Call Key, Redial, SMDR record, and other CO Line Key functionality is unaffected by this option.
- To prevent all parties from being released from a conference call when a party on a CO trunk hangs up, enable Feature Level 4 or greater in System Option 102.

Programming

Perform the following:

- Set the "Direct Access on CO Line Keys: Bypass Key System Toll Control" option to "Yes" in the "Outgoing" parameter of the Options subform of the Trunk Circuit Descriptor form (CDE Form 13) for the CO Line Key’s trunk.

Operation

To originate a call:

- Press an idle CO line key.

or

- Go off-hook (if CO line key is the preference -- see Line Preference).
- Dial outgoing digits; neither ARS codes nor Key System Toll Control digits are required.

Conference

Description

This feature allows a set user to establish a conference of up to five parties (including the originating extension), without the assistance of the Attendant.

See Device Interconnection Control for information on controlling trunk conferencing, and Attendant Conference.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- If a party on a trunk line hangs up during a conference, the remaining two-party call is moved to any idle appearance of the user's prime line. If there is no such idle appearance, all parties are dropped. To prevent this from happening, enable Feature Level 4 or greater in System Option 102.
- The DSP resources installed in the system determines the maximum number of conferences that can take place at one time; see DSP Configuration Options.
- An attendant console cannot be a member of a set-initiated conference.
- Only one party may flash out of the conference at a time.
- An extension with COS Option 203 (Broker's Call) or 252 (Broker's Call With Transfer) enabled cannot set up a conference. The switchhook flash is interpreted as a SWAP and not as a conference attempt.
- A non-busy extension forms conferences automatically.
- The Override feature forms a conference call.
- If an industry-standard telephone attempts to conference with a trunk that gives answer supervision and the trunk has not given answer supervision yet then the trunk is dropped.
- Two conference calls cannot be conferenced together.
- All Mitel telephones (except a SUPERSET 401+ telephone) in a conference can put the conference on hold.
- Another method of creating conferences is through key line privacy; see Privacy Enable/Privacy Release.
- See Flash Control for controlling flashing on an extension.

Programming

Enable COS Option 302 (Flash-in Conference) for the extension. This allows an extension to create conferences of greater than three parties.

Operation

Operation depends upon the type of device as described below.

Industry-standard Telephones:

To establish a conference:

- Establish a 2-party call.
- Flash the switchhook.
- Transfer dial tone is returned (if programmed).
- Dial the number of the next conferee.
- When the conferee answers, flash the switchhook - a conference is established.

Any extension in the conference may add additional conferees to the conference by repeating the above steps.

Mitel 5201 IP Phone:

To establish a conference:

- While on a call, press the **Trans/Conf** key.
- Transfer dial tone is returned (if programmed).
- Dial the number of the desired extension or the telephone number of the desired external party.
- When the called party answers, press the **Trans/Conf** key to form the conference.

Any extension in the conference may add additional conferees to the conference by repeating the above steps.

SUPERSET 4001 and SUPERSET 401+ Telephones:

To establish a conference:

- While on a call, press the FLASH key.
- Transfer dial tone is returned (if programmed).
- Dial the number of the desired extension or the telephone number of the desired external party.
- When the called party answers, press the FLASH key to form the conference.

Any extension in the conference may add additional conferees to the conference by repeating the above steps.

Family 3, Family 4 , Family 5 (except Symbol MiNET Wireless), SUPERSET 410, and SUPERSET 3DN Telephones:

To establish a conference:

- While on a call, press the TRANS/CONF key.
- Transfer dial tone is returned (if programmed).
- Dial the number of the desired extension or the telephone number of the desired external party.
- When the called party answers, press the TRANS/CONF key to form the conference. If you receive busy tone, or if the called party doesn't answer, press the CANCEL key to be connected with the original party.

Any extension in the conference may add additional conferees to the conference by repeating the above steps.

Symbol MiNET Wireless Phone:

1. Press the FCT key.

2. Use the scroll keys to locate "Trans/Conf."
3. Press the SEND key to select Trans/Conf.
4. Dial the number of the next party.
5. Wait for an answer.
6. Press the FCT key and scroll to Trans/Conf.
7. Press the SEND key.
   Callers are now in a three-way call (conference).
8. Press the END key to leave a conference call.

Family 1 Telephones:

To establish a conference:

- While on a call, press the TRANS/CONF softkey.
- Transfer dial tone is returned (if programmed).
- Dial the number of the desired extension or the telephone number of the desired external party.
- When the called party answers, press the TRANS/CONF softkey to form the conference. If you receive busy tone, or if the called party doesn't answer, press the BACK TO HELD softkey. You'll be connected with the original party.

Any extension in the conference may add additional conferees to the conference by repeating the above steps.

Conflict Dialing

Description

The system can differentiate between conflicting extension numbers such as “52345” and “5234”. This implies that extensions can be programmed as 1-, 2-, 3-, 4-, or 5-digit numbers with the first digits being identical. The system selects the shorter extension number if the next digit is not dialed within a preselected time.

A conflict exists between two extension numbers if the first number is contained in the second number, starting with the first digit. For example, 1234 conflicts with 12345 but 1234 does not conflict with 123 (123 conflicts with 1234).

Conditions

The following conditions apply to this feature:

- First digit conflicts between the access codes assigned to Executive Busy Override and the Callback - Busy features, and other numbers within the numbering plan, are permitted.
- Feature access codes are not permitted to conflict with any other access codes in the system.
- Extension numbers may not conflict with feature access codes.
- ARS leading digit strings are not permitted to conflict with any other access codes in the system. Extension numbers may conflict with ARS leading digits.
- Conflict dialing applies to features that need to have access codes entered (call forwarding for example). This includes the programming features on the Mitel telephones.
- Normal system inter-digit timeout is 15 seconds (not ARS dialing). If a conflict exists, the Dialing Conflict Timer applies.

Programming

Select an appropriate time-out period for System Option 50 (Dialing Conflict Timer 2-10 s).

Operation

None.

Consoleless Operation

Description

The system may be operated without the use of an attendant console. Under these conditions all features associated with the console are not available. Mitel telephones may be used as subattendant positions. These may switch the system night service (see Night/Day Switching) and have enhanced call handling and recall capabilities (see Subattendant).

Conditions

Some attendant features are not available at subattendant positions.

Programming

No attendant consoles are programmed.

Operation

None.

Contact Monitor

Description

This feature allows a station line circuit to be used for monitoring an alarm contact. The contact to be monitored is connected across Tip and Ring of the circuit. When the contact closes, a call is originated by the station line circuit and the call is directed to its tenant's Dial 0 or Priority Dial 0 Routing Point. The system handles the call as a call reroute; see Call Rerouting, Attendant Access (Dial 0), and Priority Dial 0.

Conditions

The following conditions apply to this feature:

- Contact Monitor is available on OPS and ONS Line cards in peripheral cabinets. (The controller provides relay contacts that perform a similar function; see Install an Alarm Device for more information.)
- COS Options 400 (Contact Monitor) and 241 (Receive Only) are mutually exclusive.
- The contact signal is non–latching; if the contact opens, the Dial 0 call is terminated.
- If the station line circuit has COS 239 (Priority Dial 0) enabled then the Priority Dial 0 Routing point is used.
- If there is no routing point then the origination is still processed (the station line circuit is now busy) and the station set is given reorder tone.
- If the call is answered at a console, the console display indicates that the caller is a contact monitor.

Programming

Enable COS Option 400 (Contact Monitor) for the extension.

Operation

Close the contact and a call is made to the Dial 0 Routing point or to the Priority Dial 0 Routing point.

Customer Data Entry

Description

Customer Data Entry (CDE) is a full screen application, using softkey prompts and simple graphics, for entering and changing customer programming.

Enter customer data from a terminal via the RS-232 connector on the rear of the cabinet, from an Attendant Console (a screen of reduced height is used), or from a PC via a secure Telnet client or the SX-200 ICP Web Interface. The Mitel Telnet client version 1.0.0.1 or later is recommended.

Customer Data Entry can also be performed from a remote location, using a terminal connected to the PBX via modems (with a Null Modem Connector). Refer to the Using the Maintenance Terminal section for remote connection instructions.

Conditions

None.

Programming

Refer to the Program CDE section, for detailed programming information.

Operation

None.

Customer Data Entry - Default Data

Description

The system is preprogrammed with default Class-Of-Service (COS) Options and Class-Of-Restriction Options; if no alternates are programmed, the system defaults to the preprogrammed data. The Feature Access Codes can be entered during Customer Data Entry.

Conditions

None.

Programming

Refer to the Program CDE section.

Operation

Default data in the database includes:

- Device Interconnection Table
- T1 Link Descriptor
- Class of Service Timers
- System Timers

- Trunk Circuit Descriptors
- Data Terminal Equipment (DTE) Profile
- Data Circuit Descriptors

For the complete list of data, see Default Database Configuration.

Customer Data Entry - Range Programming

Description

This feature allows range programming for blocks of extensions and a range of trunks.

By entering a range of equipment numbers, one may assign extension numbers, COR, tenant, and COS to a selected block of equipment numbers. The start extension number and defaults for the other values are entered by the programmer. The extensions are assigned sequentially starting at the entered value, and the COS, tenant and COR are assigned to the entire group.

You can also simultaneously program a range of Dial-in or Non-Dial-in trunks.

Conditions

Refer to the Program CDE section.

Programming

Refer to the Program CDE section.

Operation

None.

Customer Data Print

Description

This feature provides a means of displaying the current state of programming of the PBX. Each or all of the CDE forms may be printed, one at a time, in a presentable format.

Conditions

The printer must be PBX compatible; refer to Directed Input/Output.

Programming

See Printer/Terminal Support for details on directing the output to the desired printer.

Operation

Refer to the Program CDE section for details.

Date and Time Setup

Description

Idle Mitel display telephones continuously show the time and date (day, month, year). The console always displays the time but the date displays only when the console is idle. The time and date are also used by Message Waiting, Traffic Measurement, SMDR, and other features. The time may be displayed in 12- or 24-hour format.

The system time and/or date can be changed from the console, a subattendant telephone, and the phones listed in the linked document under Feature Availability below.

Feature Availability

Click here to see a list of the phones that can use this feature..

Conditions

- A time/date change may cause some traffic measurements to be lost, and can also affect Automatic Call Distribution (ACD) Reports. Care must be taken when setting the COS option for this feature.
- Subattendant telephones require enhanced subattendant functionality (see Subattendant - Enhanced Functions) to use this feature.
- Requires Feature Level 6 in System Option 102.

Programming

Enable COS Option 122 (Setup Time/Date) in the Class of Service of the device requiring use of this feature.

Set System Option 102 (Feature Level) to level 6.

If 12-hour time display is required, no clock options are required. If 24-hour time display is required, enable System Option 01 (24-Hour Clock).

Operation

Attendant Console

To set time:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the SET TIME softkey.
- Enter the desired time.
- Press the SET or PM softkey.

To set date:

- Press the FUNCTION key.
- Press the ATT FUNCTION softkey.
- Press the SET DATE softkey.
- Enter the desired date.
- Press the SET softkey.

Subattendant

To set the time from a subattendant telephone:

- Press SUPERKEY.
- Press the MORE softkey three times.
- Press the TIME softkey.

- Enter the time in the correct format.
- Press the AM or PM softkeys (if necessary).
- Press the SAVE softkey.

To set the date:

- Press SUPERKEY.
- Press the MORE softkey three times.
- Press the DATE softkey.
- Enter the date in the indicated format.
- Press the SAVE softkey.

5020, 5220, 5224, and 5324 IP telephones

To set the time:

- Press SUPERKEY.
- Press the MORE repeatedly until TIME? appears on the display.
- Press the YES softkey.
- Enter the time in the correct format.
- Press the AM or PM softkeys (if necessary).
- Press the SAVE softkey.

To set the date:

- Press SUPERKEY.
- Press the MORE repeatedly until DATE? appears on the display.
- Press the YES softkey.
- Enter the date in the indicated format.
- Press the SAVE softkey.

Family 5 Telephones:

5330 Phones:

To set the time:

- Press the key you have programmed as **Superkey**.
- Press **No** until **Time**? appears and then press **Yes**.
- Enter the time in the correct format.
- Press the AM or PM softkeys (if necessary).
- Press **Save**.

To set the date:

- Press the key you have programmed as **Superkey**.
- Press **No** until **Date**? appears and then press **Yes**.
- Enter the date in the correct format.
- Press **Save**.

5340 Phones:

To set the time:

- Press the key you have programmed as **Superkey**.
- Press **More** until **Time** appears and then press **Time**.
- Enter the time in the correct format.
- Press the AM or PM softkeys (if necessary).
- Press **Save**.

To set the date:

- Press the key you have programmed as **Superkey**.
- Press **More** until **Date** appears and then press **Date**.
- Enter the date in the correct format.
- Press **Save**.

Daylight Savings Time Adjustment

Description

This feature allows you to program the system to automatically set the system clock

- ahead for daylight savings time, and
- back to standard time.

You program the month, day, and hour of the time changes, and the length of the time adjustment through CDE each year. The system automatically performs the time changes. A system reset is not required for a time change to take effect.

Conditions

The following conditions apply to this feature:

- By default, daylight savings time adjustments are disabled.
- The system does not automatically select the correct days each year for the time changes. Therefore, you must reprogram the dates of the changes each year.
- After the system makes a time adjustment, the parameter (either "Advance to Daylight Savings Time" or "Go Back to Standard Time") is disabled (that is, the month field is reset to 00).

Programming

To program the system to automatically make Daylight Savings Time Adjustments:

1. In CDE, enter Systems Options (Form 4).
2. Scroll down to the "Advance to Daylight Savings Time" parameter (Option 73). Enter the month, day, and hour of the time change in the STATUS field. Enter the hour (hh) in 24-hour format (00 to 23).

Note: If the month field is set to 00, this parameter is disabled.

3. Scroll down to the "Go Back to Standard Time" parameter (Option 76). Enter the month, day, and hour of the time change in the STATUS field. Enter the hour (hh) in 24-hour format (00 to 23).

Note: If the month field is set to 00, this parameter is disabled.

4. Scroll down to the "Daylight -Standard Time Difference" parameter (Option 79). Enter the amount of the time adjustment in minutes (30 to 240 minutes). The default is 60 minutes.
5. QUIT the form.

Example

You want to program the system to

- advance the system clock by 60 minutes on April 5, at 2:00 am, and
- set the system clock back by 60 minutes on October 25, at 2:00 am.

  In System Options (Form 04) program the parameters as follows:

Form 04 - System Options/System Timers		
System Options	**STATUS**	**OPTION NUM**
Advance to Daylight Savings Time (mm:dd:hh)	04:05:02	73
Go Back to Standard Time (mm:dd:hh)	10:25:02	76
Daylight - Standard Time Difference	60	79

Device Interconnection Control

Description

This feature provides a means of disallowing connection between devices of different types. The feature is primarily intended to control trunk interconnection but applies to other devices as well. This is intended to provide a method of meeting interconnection regulations imposed by various regulatory authorities.

The checks apply when a device calls another device, when a transfer (supervised or unsupervised) is attempted and when the console attempts to perform operations on devices. The sections in this section mention Device Interconnection checks where they apply.

See Flash Control for details on controlling the ability of extensions to flash when extensions are involved with trunk calls.

There are three aspects of the feature: Tenant Interconnection, the Device Interconnection Table, and COS Options.

Tenant Interconnection, CDE Form 05 (Tenanting Interconnection Table) is described in the Tenanting section. The check for tenant connection is done when the Device Interconnection Table check is done. Other aspects of interconnection are done only if the tenant connection check passes.

The Device Interconnection Table is a table that the installer can program to limit interconnection from one device type with another and with itself. This restriction can be applied in either direction if desired. The Device Interconnection Table alone applies when the two devices in question are not both trunks. When two trunks are to be connected together, the Device Interconnection Table restrictions apply. In this case, COS options may apply additional restrictions.

The COS options apply additional checks when trunks are to be connected together. The COS options do not override the Device Interconnection Table but add finer control to the table. The options apply when a third party (a console, extension or trunk) attempts to leave two or more trunks connected together by a supervised transfer. The COS of the third party is used for the checks.

The following rules apply for the COS option checks:

- CO to CO trunk checks COS Option 313 (CO Trunk To CO Trunk Connect)
- CO to Tie checks COS Option 314 (CO Trunk To Tie Trunk Connect)
- Tie to CO checks COS Option 314 (CO Trunk To Tie Trunk Connect) and 319 (Extension non-CO Trunk To Trunk Connect)
- CO to DID checks COS Options 315 (CO Trunk To DID Trunk Connect)
- DID to CO checks COS Options 315 (CO Trunk To DID Trunk Connect) and 319 (Extension Non-CO Trunk To Trunk Connect)
- Tie to Tie checks COS Option 316 (Tie Trunk To Tie Trunk Connect) and 319 (Extension Non-CO Trunk To Trunk Connect)

- DID to Tie, Tie to DID checks COS Option 317 (Tie Trunk To DID Trunk Connect) and 319 (Extension Non-CO Trunk To Trunk Connect)
- DID to DID checks COS Option 318 (DID Trunk To DID Trunk Connect)
- For DISA trunks, the underlying circuit descriptor trunk types are checked against the device interconnection COS Options.
- Incoming Tie trunks have no COS option checking.

Conditions

The following conditions apply to this feature:

- When a device hangs up from a three party call, the Device Interconnection feature applies between the remaining parties. The third party that is used in the COS option checks is the party that hung up. If the two remaining parties cannot connect, then the call is cleared down.
- When a device hangs up from a four or five party conference call, if there are only trunks left in the call, the Device Interconnection feature is applied between each trunk in the call (both ways). The third party that is used in the COS option checks is the party that hung up from the call. If at least one trunk can connect to one other trunk in the conference then the Device Interconnection check passes. If no trunks can connect to any other trunks in the call then the call is completely cleared down.
- If the console is not allowed connection to a device, it cannot exercise features affecting that device.
- Device Interconnection applies directly to trunks, stations, sets, data stations, modem pools and consoles.
- Connection is always allowed to night bells and LDNs.
- Device connection checking is done when an ACD position is directly dialed. The checks apply between the caller and the set where the position is logged in.
- No checking is done for ACD calls from an ACD path.
- No checking is done between callers and members of recording or auto attendant hunt groups.
- Checking is done with hunt groups using the tenant of the first programmed member of the hunt group.
- Checking is done with logical lines using the tenant of the Mitel telephone where the first appearance of the line resides.
- When a trunk is accessing ARS directly, or for trunks transferred into ARS before a trunk is seized, the COS used is that of the trunk itself.
- When an extension times out from reorder tone with a conference on hold, the above rules apply to the call that is left; it is as if the extension had hung up from the call.

Programming

Set the interconnection rules in CDE Form 30 (Device Interconnection Table) as required.

Enable the appropriate trunk connection COS options, see Description.

Operation

None.

Dial Tone Disable

Description

Assignment of this feature to a dial-in trunk suppresses dial tone on an incoming trunk call. If this feature is assigned to an extension, the extension does not receive dial tone whenever dialing is initiated.

Conditions

When applied to DISA trunks, the option suppresses the initial dial tone returned for dialing the DISA feature access code. The second dial tone, after a successful code is dialed, is not affected by the option.

Programming

Enable COS Option 701 (No Dial Tone) for the device.

Operation

None.

Dial Tone - Discriminating (Stutter Dial Tone)

Description

An extension having a feature enabled that prevents calls from ringing the extension hears a distinct dial tone (350/440 Hz, 400 ms on, 100 ms off for six cycles, then continuous tone) when going off-hook to make a call. These features include Do Not Disturb, Call Forwarding - Always, or Call Forwarding - I Am Here. The Messaging - Call Me Back feature also uses the discriminating dial tone or stutter dial tone to alert the station user of a message waiting when the station user goes offhook.

Conditions

The following conditions apply to this feature:

- The tone heard is the same one that is used for transfer dial tone.
- COS Option 701 (No Dial Tone) must be disabled.
- A station (an ONS or OPS telephone) with Discriminating Dial Tone uses the discriminating dial tone to indicate the presence of a message waiting at the station.

Programming

Enable COS Option 219 (Discriminating Dial Tone) for the extension.

Operation

None.

Dictation Trunks

Description

This feature indefinitely extends the dialing stage on a trunk to allow tone-to-pulse or pulse-to-tone conversion, based on trunk circuit descriptors, of all digits dialed during a trunk call. Without the feature, only DTMF signaling is possible from extensions so equipped.

Conditions

The following conditions apply to this feature:

- A DTMF receiver circuit is used for the duration of the call.
- Dictation trunks cannot be put on hold but callers can be transferred to dictation trunks.

- The dialing stage cannot be terminated on the trunk during the call.

Programming

Set the switches on the analog CO trunks cards to the appropriate setting; refer to the Installation Information section.

Enable appropriate options for the trunks via CDE Form 13 (Trunk Circuit Descriptors); also see Trunk Circuit Descriptor Options.

Operation

Dial the necessary digits to access the dictation trunk. The system keeps the extension in dialing mode on the trunk and converts all digits dialed to the appropriate format for the trunk selected (tone or pulse).

DID/Dial-in/Tie Intercepts

Description

This feature allows a customer to specify that all DID and Dial-in Tie Trunk calls directed to a busy extension (or one which does not answer within a selected time period) are rerouted to a call rerouting point. As well, the trunks can be programmed to be redirected immediately or to be redirected under certain error conditions. See Recall for how this fits in with general recall operation.

When a DID trunk dials in, the following conditions are tested in the following order:

- If calling an LDN and the DID Attendant Access Night point is defined, then the trunk is rerouted to that point.
- If calling a station, Mitel telephone, hunt group, or logical line and the DID Routing Point is defined, then the trunk is rerouted to that point.
- If calling a busy industry-standard telephone, Mitel telephone, or logical line and the extension has forwarding - busy/always programmed, then the trunk is forwarded.
- If calling a busy industry-standard telephone, Mitel telephone, logical line, or hunt group and the DID Recall Points on Busy is defined, then the trunk is rerouted to that point.
- If calling a station or Mitel telephone with Do Not Disturb enabled and a Do Not Disturb Intercept point is defined, then the trunk is rerouted to that point.
- If the station or Mitel telephone called does not answer and the extension has forwarding - no answer programmed, then the trunk is forwarded.
- If the industry-standard telephone, Mitel telephone, hunt group member, or logical line called does not answer and the DID no answer point is defined, then the trunk is rerouted to that point.

The above applies, except for DND intercept, for Tie trunks, but using the equivalent Tie routing points.

Conditions

The following conditions apply to this feature:

- If an illegal number is dialed or an illegal feature is attempted and the DID Intercept Routing point is defined, then the DID trunk is rerouted to that point. The tenant of the trunk is used. The same applies for Tie trunks and the Dial-in Tie Intercept point. If no point is specified, reorder tone is given.
- Calls routed to an LDN or to a console through the DID/Tie intercepts are identified as intercepted calls on the console display, when they are answered.
- Calls routed to an LDN through DID Intercept busy/no answer routing point, show the tenant name of the called party when answered at the console.

- If a vacant number is dialed and the DID Vacant Number Routing point is defined, then the DID trunk is rerouted to that point. The tenant of the trunk is used. The same applies for Tie trunks and the Dial-in Tie Vacant Number Routing point.
- The DID or Tie no answer point applies in recall situations if the trunk was ringing a party and has not been answered yet by any device in the system (excluding recordings). See Last Number Redial.
- The reroute for no answer does not occur if calling a console, LDN or Night Bell. It also does not occur if already ringing the reroute point.
- The reroute point on busy or no answer cannot be a console.
- The tenant of the called party is used to determine the rerouting point. When a logical line is called, the tenant of the first appearance of the line is used. When a hunt group is called, the tenant of the first member of the hunt group is used. When an LDN is called, the tenant of the console with the lowest Bay/Slot/Circuit PLID where the LDN appears is used.
- The answer time timing for reroute/no answer is taken from COS Option 115 (Attendant Timed Recall - No Answer) of the trunk used for the reroute on no answer. If the time value is zero, then no reroute is done.
- The application of busy and attendant night access points applies only when the trunk initially dials in. The no answer point applies when ringing any extension unless the trunk has been answered. This includes ringing an extension after a call forward no answer on the initial dialed destination.
- COS Option 210 (Call Forwarding Inhibit on Dial-In Trunks) for the called party is checked on the always, busy, and no answer routing points. If the COS option is enabled for the point, then the reroute is not done.
- Extensions may be restricted from receiving DID trunk calls directly from DID trunks, by selecting COS Option 226 (Inward Restriction - DID) for the extension.
- For DID and Tie calls, the routing for all calls is done once dialing is completed.
- The busy and no answer points do not apply when calling consoles, LDNs, or Night Bells.
- If the console is programmed as the DID/TIE Busy/No Answer recall point (in Form 19, Call Rerouting Table), DID/TIE calls that are redirected to the console will not go to Attendant Automatic Overflow.

Programming

Enter the desired rerouting points via CDE Form 19 (Call Rerouting Table) in the appropriate tenant. Refer to Form 19 - Call Rerouting Table in the Customer Data Entry section, for a list of the call rerouting options.

To disable rerouting when a particular station or Mitel telephone is called, enable COS Option 210 (Call Forwarding Inhibit on Dial-In Trunks) for the required extension(s).

See Inward Restriction (DID) for details on blocking DID trunk calls altogether to a particular extension.

Operation

None.

Direct-in Lines (DIL)

Description

This feature allows non-dial-in type trunks to ring specific answering points, rather than at the attendant console; this may vary with night service changes. An answering point may be one of the following in addition to a console or LDN:

- an ACD Path
- an extension number (industry-standard telephone, Mitel telephone, logical line)

- a hunt group access code
- a night bell access code.

See Recall for effects on recall.

Conditions

Refer to Trunk Operation - Non-dial-in CO.

Programming

In CDE Form 14 (Non-Dial-In Trunks), enter the desired answering points for each of the DAY, NIGHT1 and NIGHT2 modes of operation.

Operation

None.

Direct Inward Dialing Translation

Description

The DID Translation feature enables calls received by dial-in trunks to be routed based on Day, Night1 and Night2 service. Any incoming digit string can be translated into a new number and then routed to an answer point. There is no need for the incoming and translated numbers to correspond.

Conditions

The following conditions apply to this feature:

- If the Digit Translation Table (Form 55) is not programmed with answer point, then calls on DID trunks are routed normally. If a vacant number is dialed and the DID Vacant Number Routing point is defined, then the DID trunk is rerouted to that point.
- A DID Translation number can be routed to any valid extension or Abbreviated Dial number, but not to a Feature Access Code.
- The DID Translation feature is available only with Direct Inward Dial trunks.

Programming

In Form 02, Feature Access Codes, program Feature Access Code 67 (Digit Translation Table Access). The code is restricted to two digits or less.

In Form 15, Dial-In Trunks, select the dial-in trunk and program the following fields:

N - match the expected number of incoming digits

M - absorb leading digits (maximum 8 digits)

X - Feature Access Code 67 (use the code programmed in Form 02)

Note: If field X on Form 15 contains a number rather than Feature Access Code 67, the Digit Translation Table is not consulted. Instead, the number (up to two digits) is inserted ahead of the digit string before the call is routed to its destination.

In Form 55, Digit Translation Table, program the answer points (extensions or Abbreviated Dial numbers) where calls received by dial-in trunks will route based on Day Service, Night1 Service, and Night2 Service.

In Form 19, Call Rerouting Table, program the "DID Illegal # Intercept for this Tenant" intercept.

Operation

None

Direct Station Page/Busy Lamp Field (DSP/BLF)

Description

This feature allows you to direct a page with a DSS (Direct Station Select) softkey to a selected telephone. This feature functions the same way as a manually directed page except that the DSS key is pre-programmed with a destination. Setting the DSS key to NO inhibits key operation to the destination regardless of the BLF state.

The DSS key set to PAGE in Form 09 shall provide the following functions:

- If the DN is idle and not in Do Not Disturb (DND), the BLF lamp is off. Pressing the DSS key places a direct page to the DN.
- If the DN is with a call in the talk state, in DND, or in some transitional state, the BLF lamp is on solid. Pressing the DSS key attempts to page the DN. When the BLF lamp is on solid, you may perform an Off Hook Announce or Whisper Announce depending on the call state and the COS options of the DN.

Feature Availability

Click here to see a list of the phones that can use this feature..

Conditions

The following conditions apply to this feature:

- DSS keys on sets and on PKMs associated with a Mitel telephone or an attendant console support this feature. The DSS keys must be programmmed with the setting PAGE in Form 09.
- If System Option 35, DSS/BLF Call Pickup is enabled, pressing the DSS key set to PAGE
  - When the BLF lamp flashes slowly, you can answer the ringing call on the DN on the line selected (either manually or via line preference) providing COS 218, Directed Call Pickup is enabled.
  - When the BLF lamp flashes quickly, you can retrieve the held call on the DN on the line selected, providing COS 211 Call Hold and Retrieve Access is enabled.
- If you cannot complete a page, you will receive a Busy or a Re-order tone.
- When you are performing a Direct Station Page, your active call automatically goes on soft hold.
- When you use a DSS Page key for announcing or transferring a call, the call cannot be released unless the paged party either picks up the handset or hits the prime key signalling acceptance of the call. The previous actions are not necessary if the paged party has the Microphone key on prior to being paged. If the Microphone key is latched on, any call that is released will be 'Auto Answered' to a hot microphone. This could result in an embarrassing situation if the end user was not aware that the calling party can hear everything being said around the phone after a call is released.
- You cannot turn on Secretarial with the DSS Page key, only the DSS call key does this.
- The Direct Station Page can function as a Whisper Announce, or Off-hook Voice Announce depending on the DN's call state and COS options.
- Users of Family 1, Family 4, (except Symbol MiNET Wireless), Family 5, SUPERSET 410, and SUPERSET 3DN telephones can respond handsfree to a Direct Station page.
- COS options 650, 651 and 652 in Form 3 MUST be 0.

Programming

To program a DSS key for Direct Station Page/Busy Lamp Field:

- While in CDE Form 09 (Desktop Device Assignments), position the cursor on the desired device.
- Select the EXPAND SET softkey (Expand Set sub-form is displayed). If you are programming a PKM Host that is associated with a console, select the EXPAND PKM softkey.
- Move cursor to the desired key.
- Move cursor to the TYPE column (use TAB key if necessary).
- Select the BLF/DSS softkey (Type becomes “Busy Lamp”).
- Press the TAB key twice to get to the DSS column.
- Select the DSS PAGE softkey (PAGE appears in the DSS column).
- Press the TAB key to get to the EXT NUM column.
- Enter the extension number of the device to be monitored by the BLF appearance.
- Select the ENTER softkey.

Programmable Key Modules (PKM, PKM 12, or PKM 48)

- While in CDE Form 09 (Desktop Device Assignments) move the cursor to the bay/slot/circuit number of the Family 1, Family 4, (except SUPERSET 4DN and Symbol MiNET Wireless), Family 5 and SUPERSET 410 telephone. An asterisk (*) appears to the left of the set type if programmable key modules are programmed for the set.
- Select the EXPAND PKM softkey. “ENTER PKM NUMBER (1-X):” appears in the command line (where X is 3 for a PKM, 2 for a PKM 48, 1 for a PKM 12).
- Type the address setting (1, 2, or 3 for the PKM; 1 or 2 for the PKM 48; 1 for a PKM 12) of the desired programmable key module in the command line.
- Select the Enter softkey. The system displays the Expand PKM Set Subform for the PKM. The Expand PKM Set Subform allows you to program the functions of the 30, 12 or 48 keys available from the programmable key module.
- Move cursor to the desired key.
- Move cursor to the TYPE column (use TAB key if necessary).
- Select the BLF/DSS softkey (Type becomes “Busy Lamp”).
- Press the TAB key twice to get to the DSS column.
- Select the DSS_PAGE softkey (PAGE appears in the DSS column).
- Press the TAB key to get to the EXT NUM column.
- Enter the extension number of the set to be monitored by the BLF appearance.
- Select the ENTER softkey.

Note: The keys are preceded by a 1, 2, or 3 to indicate which programmable key module is associated with the set (for example, keys belonging to the first programmable key module are preceded by a “1”).

Operation

To direct page the BLF party:

- When the BLF lamp is off, press the DSS key.

Note: When the BLF lamp is on, the Direct Station Page can function as a Whisper Announce, or Off-hook Voice Announce depending on the DN's call state and COS options.

Direct Station Select (DSS) Key

Description

You use the Direct Station Select (DSS) keys on a multi-line Mitel telephone, or Programmable Key Module (PKM) in conjunction with the Busy Lamp Field (BLF) indicators located beside each DSS key. A

DSS key is used to call, and connect calls to a device. The BLF indicator corresponding to the DSS key indicates the busy status of the device.

The DSS key operates when the line appearance of the device is:

- idle - pressing the DSS key will initiate a call to the associated device.
- talking to another party that can be put on soft hold - pressing the DSS key will put the other party on consultation hold, and initiate a call to the associated device.
- listening to dial tone - pressing the DSS key will initiate a call to the associated device.

The DSS key is inoperable in all other states. For example, if the device is talking to one party with another party on soft hold, the DSS key will have no effect.

Secretarial Option

The BLF indicator associated with the DSS key always displays the status of the BLF party, thus limiting the need for supervised transfers. An Automatic Transfer option is available to users with the SECRETARIAL enabled on their DSS keys. To transfer a call, the DSS key is pressed - the call is transferred with an automatic release to the BLF party.

If the BLF party is idle, the transferred party rings it. If the BLF party is busy, the system camps on the transferred call. If the two parties are not allowed to be connected together because of device interconnection restrictions, the system prevents the automatic transfer; i.e., the same conditions apply to these transfers as to any others.

See also Busy Lamp Field in this document.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- DSS keys will not function on line appearances associated with trunks.
- DSS keys will not function when any feature associated with a template such as Hotel/Motel and Automatic Call Distribution are placed on the line appearance.
- System option 35, DSS/BLF Call Pickup is disabled.
- The DSS page key does not turn on SECRETARIAL., only the DSS call key does this.

Programming

Multi-line Mitel telephones

To program DSS keys:

- While in CDE Form 09 (Desktop Device Assignments), position the cursor on the desired device and select the EXPAND SET softkey (Expand Set sub-form is displayed).
- Move cursor to the desired key.
- Move cursor to the TYPE column (use TAB key if necessary).
- Select the BLF/DSS softkey (Type becomes “Busy Lamp”).
- Press the TAB key twice to get to the DSS column
  OR
  To program the key with Secretarial Option, press the TAB key once to get to the SEC column. Then press the SECRETARIAL softkey (YES appears in the SEC column). Press the TAB key again to get to the DSS column.

- Select the DSS CALL softkey (CALL appears in the DSS column) or the DSS PAGE softkey (PAGE appears in the DSS column).
- Press the TAB key to get to the EXT NUM column.
- Enter the extension number of the device to be monitored by the BLF appearance.
- Select the ENTER softkey.

Key Labels:

- DSS Key labels are derived from the NAME field of the extension in CDE Form 09 (main form) and prepended with "DSS". The Name field is not editable when programming a DSS/BLF key but can be modified in the main form. For example, to edit a DSS Key label on Extension 123, change the Name associated with Extension 123 in CDE Form 09 (main form). If the name field is blank, phones will display "DSS 123".
- Labels do not appear when an ACD template is applied.
- When upgrading to Release 4.0 UR1 or later from a prior release, DSS/BLF keys will not automatically assume the label in the NAME field. You must reprogram the key before the label will appear. Labels will be maintained in subsequent upgrades.

Programmable Key Modules

- Display CDE Form 09 (Desktop Device Assignments).
- Move the cursor to the bay/slot/circuit number of the Family 1, Family 4, (except SUPERSET 4DN and Symbol MiNET Wireless), Family 5 or SUPERSET 410 telephone. An asterisk (*) appears to the left of the set type if programmable key modules are programmed for the set.
- Select the EXPAND PKM softkey. “ENTER PKM NUMBER (1-X):” appears in the command line (where X is 3 for a PKM, 2 for a PKM 48, 1 for a PKM 12).
- Type the address setting (1, 2, or 3 for the PKM; 1 or 2 for the PKM 48; 1 for a PKM 12) of the desired programmable key module in the command line.
- Select the Enter softkey. The system displays the Expand PKM Set Subform for the PKM. The Expand PKM Set Subform allows you to program the functions of the 30, 12 or 48 available keys from the programmable key module.
- Move cursor to the desired key.
- Move cursor to the TYPE column (use TAB key if necessary).
- Select the BLF/DSS softkey (Type becomes “Busy Lamp”).
- Press the TAB key twice to get to the DSS column
  OR
  To program the key with Secretarial Option, press the TAB key once to get to the SEC column. Then press the SECRETARIAL softkey (YES appears in the SEC column). Press the TAB key again to get to the DSS column.
- Select the DSS CALL softkey (CALL appears in the DSS column).
- Press the TAB key to get to the EXT NUM column.
- Enter the extension number of the device to be monitored by the BLF appearance.
- Select the ENTER softkey.

Note: The keys are preceded by a 1, 2, or 3 to indicate which programmable key module is associated with the set (for example, keys belonging to the first programmable key module are preceded by a “1”).

To program a PKM host in Form 9 and to associate the PKM host to the attendant console, see the Programming under the Direct Station Select/Busy Lamp Field (DSS/BLF) Interface Unit feature. The DSS/BLF interface unit allows you to associate up to two PKM 48 devices with an attendant console. Each PKM 48 provides the attendant console user with 48 DSS/BLF keys.

The DSS/BLF Interface Unit uses a separate line connection to a DNIC port. You attach the PKM 48 to the DSS/BLF Interface Unit and associate the PKM 48 with the attendant console through Customer Data

Entry (CDE). You can attach up to two PKM 48 devices to the DSS/BLF Interface Unit, for a maximum of 96 DSS/BLF keys.

COS options 650, 651 and 652 in Form 3 MUST be 0.

Operation

To call the BLF party:

- Press the DSS key.

To transfer a call to a station (without Secretarial option):

- Press the DSS key.
- Press the RELEASE key/softkey.

To transfer a call to a station (Secretarial option):

- Press the DSS key.

NOTE: To delete a DSS/BLF key on the Mitel 5330 or 5340 IP Phones, use the Superkey. Using the Settings key results in the error message "This key can only be programmed by your administrator."

Direct Station Select/Busy Lamp Field (DSS/BLF) Call Pickup

Description

This feature allows you to pick up a held or ringing call from a selected directory number (DN) with the Direct Station Select (DSS) key on a multi-line Mitel telephone, or Programmable Key Module (PKM). The BLF indicator corresponding to the DSS key indicates the status of the DN (Directory Number) of your selected destination.The BLF lamp has two flash rates to indicate whether the call is a held or a ringing call.The BLF flashes quickly to indicate a held call, and flashes slowly to indicate a ringing call.

The NO setting for the DSS key prohibits retrieving calls, but allows the BLF to show the status of the DN.

The CALL setting and the PAGE setting for the DSS key provides the following call pickup functionality:

- If the DN is ringing with an initial call (not camped-on), the BLF lamp flashes slowly to indicate a ringing call. Pressing the DSS key retrieves the DN call on the line selected.
- If the DN is on hard hold, the BLF lamp flashes quickly. Pressing the DSS key retrieves the held call on the selected line of the set where the key was pressed.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- DSS keys on SUPERSET telephones (excluding Family 2 telephones) and DSS keys on PKMs associated with a telephone or console support DSS/BLF Call Pickup. The set associated with the PKM or PKM host must be programmed in CDE Form 9 to support DSS keys.
- System Option 35, DSS/BLF Call Pickup must be enabled in Form 4.
- New systems have Option 35, DSS/BLF Call Pickup enabled by default.
- Restrictions to retrieve a call from hard hold using DSS/BLF Call Pickup are the same as using the Call Hold Retrieve access code, i.e. you cannot retrieve a conference call or a call on a logical line.
- Restrictions to pickup a ringing call using DSS/BLF Call Pickup are the same as using the manually directed call pickup, i.e. you cannot retrieve a call on a logical line.

- Conditions for DSS/BLF Call Pickup include the conditions for Directed Call Pickup.
- This feature adheres to the tenant and device connections programmed in CDE Form 5, Tenant Interconnection Table and CDE Form 30, Device Interconnection Table.
- DSS keys will not function on line appearances associated with trunks.

Programming

In CDE Form 4, System Options, enable Option 35, DSS/BLF Call Pickup.

In CDE Form 3, Class of Service Options enable COS 211, Call Hold and Retrieve Access, and COS 218, Directed Call Pickup.

Program the DSS key with the option CALL using the procedures in Direct Station Select (DSS) Key.

Operation

To retrieve a ringing call from the DN, using the DSS key

- Press the DSS key when the BLF lamp shows a slow flash.

  Pressing the DSS key during this slow flash retrieves the ringing call on the selected line of your set, providing COS 211, Call Hold and Retrieve Access is enabled.

To retrieve a call that is on hard hold from the DN, using the DSS key

- Press the DSS key when the BLF lamp shows a fast flash.

  Pressing the DSS key during this fast flash retrieves the held call on the selected line of your set, providing COS 218, Directed Call Pickup is enabled.

Direct to ARS

Description

This option allows an industry-standard telephone to be directly routed to ARS without dialing, and for other devices to be routed after dialing a valid account code. The system dials up to five digits automatically for the extension.

Conditions

The following conditions apply to this feature:

- For industry-standard telephones, the feature only applies when the station initially starts a call. Once the initial trunk call has been established, normal operation resumes (for example, flashing is not affected).
- This feature applies to all devices that can access account codes, only after a valid account code is entered; see Account Codes.
- The Contact Monitor and Manual Line features cannot access the Direct To ARS feature.
- Industry-standard telephones cannot have the Forced Account Code feature enabled.
- There must be an existing ARS Leading Digit string programmed in Form 26, ARS digit strings that matches the Direct to ARS code.

Programming

In CDE Form 03, COS Define, enable COS Option 217 (Direct To ARS) for the extension.

In CDE Form 02, Feature Access Codes, enter the ARS leading digits string that is programmed in Form 26 (e.g. enter '9') as the feature access code for Direct to ARS (Feature 37).

To provide an internal dial tone so users can access their voice mail when Direct to ARS is enabled, do the following:

1. In CDE Form 16, ensure that a trunk group with no members is available.
2. In CDE Subform 26, ARS Digit Strings, enter an access code (for example, *59). Enter the route number that represents the trunk group from Step 1.
3. In CDE Form 23, ARS Route Definition, assign the route number that you entered in Step 2.

Operation

For industry-standard telephones: lift the handset and the system automatically dials the ARS access code.

For other devices: enter a valid account code and the system automatically dials the ARS access code.

Direct Trunk Select (DTS)

Description

This feature allows the user to directly access an outside trunk for both incoming and outgoing calls without the need of trunk access codes. The trunk is assigned to a line appearance of the telephone through system programming. Telephones having the Direct Trunk Select feature can be programmed for ring, delayed ring, or no ring.

Direct trunk select calls bypass the system's Automatic Route Selection feature and are therefore unaffected by COR (Toll Control). An account code can be entered while established in a call.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- A DTS line key can appear on a maximum of 64 IP sets, or 32 IP and 32 DNIC sets.
- To prevent all parties from being released from a conference call when a party on a trunk line hangs up, enable Feature Level 4 or greater in System Option 102.
- Direction and Ring variants can be programmed independently for each appearance. The Direction variant allows control of the type of trunk call, incoming only or both directions (incoming and outgoing).
- If a DTS trunk has answer points (DAY, NIGHT 1 or NIGHT 2) programmed in Form 14, they will be ignored. Instead, incoming calls on the DTS trunk will be routed to the telephone(s) with the DTS appearance.
- The trunk ignores Do Not Disturb on the Mitel telephone when it originates if there is only one appearance of the line.
- A DTS trunk must be in a trunk group in order to make outgoing calls.
- It can appear on several Mitel telephones.
- This feature is not available on Family 2 telephones.
- There can be only one appearance of any one Direct Trunk Select line on a Mitel telephone.
- The calls on the trunks bypass ARS but do use the SMDR feature if enabled.
- The user can transfer calls on this line to other extensions. The behavior of the call on the extension receiving the transfer depends on whether the extension has an appearance of the line and whether

Option 689 (DTS/CO Line Transfer Call Handling) is enabled in the extension's Class of Service. For more information, see the Conditions section in the Transfer feature description.

- When the far end hangs up on a direct trunk select line, all parties connected are cleared down, and the trunk occupies the line while it clears down.
- When the trunk hangs up, all parties in the call on the line are put idle.
- DTS trunks are unavailable to ARS, regardless of ARS programming.
- Only CO type trunks can be programmed in a direct trunk select configuration. Incoming-only trunks are not compatible.
- Incoming direct trunk select calls follow call forwarding programmed on the set only if COS Option 815 "DTS/CO Line Key Honors Forwarding" is enabled for the trunk and the DTS key appears on only one set.
- Do Not Disturb is overridden by an incoming direct trunk select call.
- If a DTS trunk is programmed as the preferred line on a set, when the system selects the line, the user may hear a broken dial tone followed by a normal dial tone.

Programming

Program trunks that are to be selected directly, instead of through ARS, in the Mitel Telephone Lines subform of CDE Form 09 (Desktop Device Assignments). These trunks must be programmed into trunk group(s).

Operation

To access Direct Trunk Select:

- Lift the handset. Dial tone is heard.
- Press a Direct Trunk Select line key. Central Office dial tone is returned.
- Dial the external number.

To answer an incoming Direct Trunk Select call:

- Lift the handset and press the Direct Trunk Select line key when the set rings and/or the line appearance flashes.

Disconnect Alarm

Description

This feature provides an alarm indication when a Mitel telephone is unplugged.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The alarm is not cleared by the Mitel telephone being plugged back in.

Programming

Enable COS Option 603 (Telephone - Disconnect Alarm) in the Mitel telephone's COS.

Operation

Unplug the Mitel telephone - an alarm is generated; see Attendant Alarm Readout for reading alarms.

Display Caller ID on Non-Prime Lines

Description

This feature allows Mitel display telephones to automatically display the caller ID from non-prime lines when the set is idle or in a talk state.

When available, the number and/or name of the calling party is displayed for internal calls and for incoming external calls over a CLASS trunk.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

- Family 1 telephones display the name and number if both are available.
- Family 3, Family 4 (except Symbol MiNET Wireless), and Family 5 telephones display the name OR number, if both are available, for an incoming trunk. Otherwise, the number is displayed. Users of these telephones can toggle between name and number for incoming trunks. Both name and number will appear for internal calls.
- The automatic display of caller ID for non-prime lines requires Feature Level 3 or greater, in System Option 102.
- This feature also supports the SUPERKEY access (pressing SUPERKEY and then pressing the line key to obtain caller ID).
- Sets with Feature Level 2 or less, only show the caller ID on non-prime lines through the use of the SUPERKEY and the line key.
- SUPERSET 4DN telephones do not automatically get caller ID for non-prime lines.
- Multi-call lines, Key lines, Direct Trunk Select lines, and CO lines support Display Caller ID.
- Caller ID is not provided for calls to LDN keys, Recall keys, and DSS line appearances (DSS/BLF keys).
- Prime line ringing takes absolute precedence. Therefore if the prime line rings, the caller ID for the prime line will always appear, even if the identifier is only a trunk number.
- If the set is idle and a line appearance rings, the caller ID for that line appearance will display.
- If more than one line appearance rings, the caller ID will reflect the identity of the line appearance that has the highest priority based on the key position, (the line that would normally be answered by going off hook). If other non-prime lines ring at the same time, this highest priority caller ID remains on the display when the line appearance stops ringing or goes busy.
- If the set is engaged in a simple talk state, the caller ID for an incoming call will replace the caller ID of the connected party. If the caller from the incoming call hangs up or is answered by another telephone, the display will revert to show the identity of the connected party. While the user is in a simple talk state, the display will update the caller ID with other incoming calls.
- The incoming caller ID does not appear on the display of a set in a talk state, if calls are camped-on or on soft hold.
- If the set is engaged in a simple talk state and there is an incoming call, the display will change but the softkeys will remain the same, that means, the small display set user will not be able to toggle between the name and number.

Programming

Form 04, System Options/System Timers - Enable Feature Level 3 in System Option 102.

Form 03, COS Define - To activate the Calling ID display, you must program the following COS options for the sets and the trunks. For the sets, enable COS option 509, Display Caller ID for non-prime lines, COS option 502, Display ANI/DNIS/CLASS Information, and COS option 503, Display CLASS Name. For the trunks, enable COS option 811, ANI/DNIS Trunk.

Operation

Displaying the caller ID automatically does not require a procedure.

Note: Small display sets require you to use the Name or Number softkeys to obtain the full identity for incoming trunk calls.

To **manually** display the caller ID, follow the operation described below for that particular set.

Family 3, Family 4 (except Symbol MiNET Wireless) and Family 5 Telephones:

You can display the calling name or number of a ringing line key on your set.

- Press SUPERKEY.
- Press a ringing line key. The calling name or number for the key is displayed.
- Press the toggle key labelled NAME or NUMBER to toggle between the name and number of the caller.
- Press SUPERKEY to exit.

You must exit by pressing SUPERKEY before you can answer a call by pressing its line key.

Family 1 Telephones:

You can display the calling name or number of a ringing line key on your set.

- Press SUPERKEY.
- Press a ringing line key. The calling name and/or number for the key is displayed.
- Press SUPERKEY to exit.

You must exit by pressing SUPERKEY before you can answer a call by pressing its line key.

Display Keys

Description

This feature allows users of Mitel display telephones (with the exception of SUPERSET 3DN telephones) to display the function of certain keys on their sets.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

None.

Programming

Refer to Feature Keys for instructions on how to program feature keys on Mitel display telephones.

Refer to Speedcall for instructions to program personal speedcall numbers into personal keys on Mitel display telephones.

Operation

Operation varies depending upon the type of set as described below.

Family 4 (except Symbol MiNET Wireless) Telephones:

You can display the line numbers, speedcall numbers, or features that are programmed into the personal keys on your set.

- Press SUPERKEY.
- Press a personal key. The line number, speedcall number or feature that is programmed for the key is displayed.

  If the key is not programmed for, the display shows UNUSED KEY. If the line key is programmed with a private speedcall number, PRIVATE NUMBER appears.

  For feature keys and speedcall keys, the CHANGE and CLEAR softkeys also appear. Press the CHANGE softkey if you want to reprogram the personal key. Press CLEAR if you want to clear the feature or speed call number from the personal key.
- Press SUPERKEY to exit.

With Family 3 , Family 4 (except Symbol MiNET Wireless) and Family 5 telephones, you can also display the functions of some of the hardkeys on the set:

- Press SUPERKEY.
- Press the desired hardkey. The function of the hardkey appears in the display.

  If you press the REDIAL hardkey, the number that is currently stored against it appears in the display. If no number is stored in the REDIAL key, NO NUMBER STORED appears.

  If you press the MESSAGE key, and you have don't have any messages waiting, NO MESSAGES appears in the display.
- Press SUPERKEY to exit.

Family 1 Telephones:

You can display the line numbers, speedcall numbers, or features that are programmed into the line keys/personal keys on your set.

- Press SUPERKEY.
- Press the MORE softkey until the DISPLAY KEYS softkeys appears.
- Press the DISPLAY KEYS softkey.
- Press a line (personal) key. The line number, speedcall number or feature that is programmed for the key is displayed.
  If nothing is programmed for the key, the display shows NO NUMBER SAVED. If the key is programmed with a private speedcall number, PRIVATE NUMBER appears.
- Press SUPERKEY to exit.

Distinctive Ring Tones

Description

This feature offers 16 distinctive ring tones that can be programmed for DTS, CO Line and keyline keys on phones (IP and DNIC) and PKMs.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

- Distinctive ring tones apply to DTS, CO line, and Keyline keys with ring variants of "immediate ring" or "delay ring".
- If a phone is ringing and another call arrives at the same time, the ring tone will not change.
- Ring tone may change if a caller hangs up on a phone that has more than one line ringing.
- Default ring tone is 5.

Programming

- In Form 04, System Option and Timers, enable Feature Level 5 in System Option 102.
- Enter a ring tone value (1 - 16) in sub-forms Expand Set or Expand PKM in Form 9, Desktop Device Assignment.

Operation

None.

Do Not Disturb

Description

Do Not Disturb prevents incoming calls from ringing the telephone. It also prevents incoming directed and group paging announcements from occurring over the set speaker. Outgoing calls are unaffected.

Callers to a telephone with Do Not Disturb active receive reorder tone; the message DO NOT DISTURB appears on display sets. The alternative is to program the system to reroute such calls to a predetermined answering point; see Call Rerouting.

Consoles and all Family telephones can override Do Not Disturb.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- If Do Not Disturb is activated, users hear broken dial tone followed by normal dial tone when they go off-hook.
- Either the set user or the attendant can set up or cancel Do Not Disturb.
- The phone can override Do Not Disturb if COS Option 500 (Override) is enabled in its COS, and COS Option 238 (Override Security) is disabled in the called extension's COS. For details on how to program and use override, see Override (Intrude).
- When overriding Do Not Disturb, the Auto-Answer feature is ignored.
- When overriding Do Not Disturb, if the called set becomes busy in the meantime, then the overriding device will attempt to intrude on the call; see Override (Intrude).
- A hunt group member that activates Do Not Disturb is treated as busy during hunting.
- While Do Not Disturb is active, forwarding works as if the extension were busy.
- If the set user activates a callback, the callback is honored, even if Do Not Disturb is active.
- When a Reminder expires, Do Not Disturb is ignored.
- Call Rerouting ignores Do Not Disturb.

- If Discriminating Dial Tone is enabled then discriminating dial tone is heard at an extension when it has Do Not Disturb activated; see Discriminating Dial Tone.
- DID trunks ignore DND if there is no Do Not Disturb routing point for the extension's tenant.
- If a call is placed on hold by the telephone with Do Not Disturb set, is not retrieved, and recalls when the hold times out, the telephone with Do Not Disturb set does not ring and does not see the recalling call.
- If there are multiple call line appearances of the Do Not Disturb station or Mitel telephone then the behavior of the feature changes. If the extension is dialed, busy tone is heard if there are no free appearances to ring and all Mitel telephones where the line appears have Do Not Disturb activated. If all Mitel telephones where the line appears have Do Not Disturb activated, then the caller gets Do Not Disturb Reorder Tone. If at least one appearance is free then the call to the extension proceeds as if the Do Not Disturb was not activated.
- Consoles are never directed to the Do Not Disturb reroute point.
- Do Not Disturb settings are not maintained through a system reset.
- The state of the Do Not Disturb key LED when the feature is active depends on the Feature Level. See the Programming section below for details.

Programming

To enable the Do Not Disturb feature, assign an access code to Feature 10 (Do Not Disturb) in Form 02. To permit a set user to activate Do Not Disturb, enable COS Option 220 (Do Not Disturb) in the set's COS.

For feature key activation of Do Not Disturb at a multi-line Mitel telephone, program a DO NOT DISTURB feature key. (See Feature Keys.)

To intercept calls dialed to a Do Not Disturb extension, program CDE Form19 (Call Rerouting Table) for the extension's tenant.

If the Feature Level is set to level 6 or higher (Form 04, System Option 102) the Do Not Disturb key flashes when the feature is active; otherwise, it lights steadily.

Operation

Operation varies depending upon the type of device as described below.

Symbol MiNET Wireless Phone:

To turn Do Not Disturb on:

1. Press the No softkey until "Do Not Disturb?" appears.
2. Press the Onsoftkey.

To turn Do Not Disturb off:

1. Press the Onsoftkey until "Do Not Disturb?" appears.
2. Press the Offsoftkey.

Family 2 and Industry-standard Telephones:

To activate Do Not Disturb:

- Dial the Do Not Disturb access code followed by the digit 1.
- Dial tone is heard.
- Hang up.

To remove Do Not Disturb from the extension:

- Dial the Do Not Disturb access code followed by the digit 2.

- Dial tone is returned.
- Hang up.

Multi-line Mitel Telephones:

To activate Do Not Disturb:

- Press the DO NOT DISTURB feature key. The adjacent LCD indicator darkens or key LED illuminates.

To remove Do Not Disturb from the extension:

- Press the DO NOT DISTURB feature key. The adjacent LCD indicator, which was dark, clears or key LED extinguishes.

Door Phone / Open Door

Description

This feature provides two-way communication between an extension and a door phone in an entryway. When a person requiring entry into a building presses the Door Phone button, the door phone rings the answer point programmed for the door phone. The person responding can press a key or dial a feature access code to open the door. The door locks again after five seconds.

The system supports up to four door phones, each connecting to an ONS circuit.

Note: the Door Phone does not have to be called to work - you can open the door at any time by pressing a key or dialing a feature access code.

Feature Availability

All phones can use the Door Phone Answer feature.

Conditions

- This feature replaces the Door Opener feature introduced in Release 1.0. Systems upgraded to Release 2.0 or later that were using the Door Opener feature will need to install door phone(s) and program them as described below.
- The five second time period before the door locks again is not programmable.
- The door can be unlocked from the designated answering point and from any other station by dialing the required feature access code or pressing the equivalent feature key.
- The door can be unlocked from a station in idle, dialing, or talk state. It cannot be unlocked when the station is ringing.
- A door phone placed on hold is considered in talk state and may be opened.
- The ONS door phone extension cannot be programmed in any other form—e.g., Form 10 (Pickup Groups) and Form 17 (Hunt Groups). Otherwise, the system will mimic a delete of the set (remove it from hunt groups/pickup groups if possible and generate an alarm if not possible). The set itself will not be deleted and all other information (besides pickup group and hunt group) will be retained.
- Door phones should not be assigned to plods associated with power or system fail transfer circuits as the phone would become the answering point for any incoming calls in the event of a failure.

Programming

Door Phone

- In Form 18, program a general-use relay as a Door Relay PLID and assign it an extension number.

- In Form 9, enter the Door Relay extension number from Form 18 into the ASSOC field of the ONS door phone. Assign the door phone a Name (e.g., Door Phone) to identify it on the door answerer's display.
- Program a call forwarding destination for the door phone. This can be done from a console or a subattendant (MX only). The forwarding destination can be a single extension or a hunt group of multiple extensions. For the CX/CXi—and as an alternative for the MX—assign the door phone to its own tenant in Form 19 and program Station Dial 0 Routing to direct calls from the door phone to an extension. The extension can be a keyline that appears on several phones. Note that call forwarding programmed on a phone overrides tenant routing. Thus, when a forwarding destination for the door phone is NOT programmed, the system will attempt to route the call to the "Station Dial 0 Routing" or "Priority Dial 0 Routing" number (if programmed) in the door phone's tenant.
- Program the door phone's COS option 115, Attendant-Timed Recall (NO ANS) timer to ring the forwarding destination the desired length of time.

Open Door

- Assign a feature access code to Feature 66, Open Door, in CDE Form 02 (Feature Access Codes).
- (Optional) Program an Open Door feature key on one or more extensions to allow the user to open the door. The key can be programmed to open a specific door by entering the corresponding Door Relay number (1- 4) or the door that the user has answered by entering 0.

Operation

The door opener can be operated by:

- pressing the "Open Door" feature key
- dialing the "Open Door" feature access code.
- pressing the Open Door softkey (available on SUPERSET 4DN, 430, 4025, 4125, and 4150 telephones only)
- pressing the MESSAGE key (not available on SUPERSET 4DN, 430, 4025, 4125 and 4150 telephones).

DSS/BLF Interface Unit

Description

The Direct Station Select/Busy Lamp Field (DSS/BLF) Interface Unit allows you to associate up to two PKM 48 devices with an attendant console. Each PKM 48 provides the attendant console user with 48 DSS/BLF keys.

The DSS/BLF Interface Unit uses a separate line connection to a DNIC port. You attach the PKM 48 to the DSS/BLF Interface Unit and associate the PKM 48 with the attendant console through Customer Data Entry (CDE). You can attach up to two PKM 48 devices to the DSS/BLF Interface Unit, for a maximum of 96 DSS/BLF keys.

Conditions

You can only connect PKM 48 devices to a DSS/BLF Interface Unit. SUPERSET 400-series PKMs are not supported.

The maximum cable length between the DSS/BLF Interface Unit and the first PKM is 5 m (16.4 ft.). The maximum PKM cable length between the first PKM and second PKM is 0.6 m (2 ft.). Only use the modular cables that are supplied with the DSS/BLF Interface Unit and PKM. See the DSS/BLF Console Interface Box Figure.

When a PKM 48 is connected to a DSS/BLF Interface Unit, you can only program the PKM 48 keys as DSS/BLF keys.

A SUPERCONSOLE 1000 Attendant Console can support either a printer port (see Printer/Terminal Support) or a DSS/BLF Interface Unit, but not both. You can't program submarket 2 of the console's Bay/Slot/Circuit location as a DSCONS in CDE Form 12 and also program the console as a PKM HOST (in Form 9).

If you connect a DSS/BLF Interface Unit to the system before you program the device type as PKM HOST (in CDE Form 09), the system will program the device type as either SS410 or SS4025. If the circuit has not been programmed before, the system programs it as a SS410. If the circuit has been programmed previously with a device type, it will be reset to SS4025. In either case, you must reprogram the DSS/BLF Interface Unit to device type PKM HOST.

COS options 650, 651 and 652 in Form 3 MUST be 0.

The PKM 48 must be associated with the attendant console through CDE.

You cannot place a call with a DSS key

- while using attendant functions
- while dialing
- during paging
- while overriding another party
- when a SOURCE and DEST are already established.

For installation instructions see DSS/BLF Interface Unit.

Programming

To program the DSS/BLF Interface Unit and PKM 48

1. In CDE Form 09, Desktop Device Assignments
   - assign the DSS/BLF Interface Unit with device type "PKM HOST"
   - enter the directory number of the associated attendant console in the "ASSOC" field.
2. In the Expand PKM Subform of Form 09
   - select the PKM 48 address setting (1 - 2)
   - program the PKM 48 keys as DSS keys or BLF keys.

Operation

See Busy Lamp Field and Direct Station Select (DSS) Key in the Features section.

DTMF-To-Rotary Dial Conversion

Description

This feature automatically converts DTMF tones from DTMF equipment to rotary dial outpulsing on outgoing trunks which have been programmed as rotary dial trunks. The DTMF digits also appear on the trunk, as early line split is not provided.

Conditions

None.

Programming

Specify the trunk as non-DTMF in CDE Form 13 (Trunk Circuit Descriptors).

Operation

None.

DTS/CO Line Transfer Call Handling

See Transfer.

Emergency Call Handling

Description

Emergency call handling ensures that a call to an emergency number is allowed for all stations except for those programmed as Hotlines and Receive Only (see exceptions under Conditions).This feature allows you to remove call blocking for emergency calls by designating digit strings (for example; 911 or 8888) as emergency calls in ARS programming.

Because you define the emergency call digits defined in ARS and Toll Control, emergency calls are not restricted to 911 calls. This permits emergency call handling in areas where 911 service is not available or where the emergency code is another set of digits.

Emergency calls can proceed from

- Guest rooms with outgoing call restrictions (System Options 57 and 58)
- Stations with forced account code access (COS Option 200)
- Consoles in restricted service (Attendant Console Lockout)
- Stations with SMDR Overwrite Buffer (COS 702) disabled
- Stations with SMDR buffers full due to printer failure
- Stations restricted with device interconnection and tenant interconnection.
  - Interconnection restrictions will proceed on trunks with CO line access, if you enable the trunk descriptor on the set with the feature Direct Access on a CO Line Keys - Bypass Key System Toll Control.

Emergency calls cannot proceed from

- Sets that are designated as Hot Lines (i.e. direct lines to taxi services)
- Sets that are designated as Receive Only.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- You must program the emergency digit strings in ARS: Digit Strings (Form 26) and/or Key System Toll Control (Form 46)
- You must program the number of digits that the emergency call requires in ARS: Maximum Dialed Digits (Form 27)
- Emergency calls will still be restricted by the Restricted COR group programmed in ARS: Digit Strings (Form 26)
- Emergency calls will override the COR in ARS Route Definition (Form 23)
- A set designated as a Hot line will not be able to originate an emergency call

- A set designated as a Receive Only (COS 241) will not be able to originate an emergency call
- When the option "Direct Access on a CO Line Key - Bypass Key System Toll Control" is enabled in a trunk's circuit descriptor, the interconnection restrictions will be obeyed. This means that if a CO Line appears on a set that does not have permission to use that line, the line will not function even for emergency calls. Because the CO line is accessed immediately, the PBX has no opportunity to analyze the digits and allow emergency calls.
- The Emergency designation in the ARS and Toll Control forms will not enable a 911 type alarm to the attendant console and will not enable the sending of CESID information to the network when these digits are dialed, unless the emergency digits that are sent to the CO are 911.

Programming

Complete the following programming:

- Designate the emergency digits as EMERGENCY in the Key System Toll Control (Form 46) and in Show Strings, subform of ARS: Digit Strings (Form 26). Form 46 and Form 26 are independent of each other. Form 46 is for CO lines.

  Program the emergency digits for a
  - CO Group key in the Show Strings, subform of ARS: Digit Strings (Form 26)
  - CO Line key in Key System Toll Control (Form 46).

Operation

Dial the emergency number.

Emergency Calls (911) - Detection and Reporting to Attendant Consoles

Description

This feature presents an alarm to the attendant when an extension user places a 911 call. The alarm displays the user's extension number and location, e.g., Extension 2056, Room 212. With this information, the attendant can provide assistance or direct emergency services (for example, police or ambulance personnel) to the extension where the call originated.

This feature generates an audible alarm at all the attendant consoles when a user places a 911 call from an extension, an attendant console, or a trunk. All consoles are notified of a 911 call regardless of their tenant group.

If the console is idle, the console display will identify that a 911 call has been made and will display the location of the originator of the 911 call, if you have specified the physical location in the Comments field (see programming) of each

- Extension in Form 9
- Attendant console in Form 7
- Non-dial-in trunk in Form 14
- Dial-in trunk in Form 15.

The system also generates maintenance logs for 911 calls:

- when the system detects a 911 call, it generates a log that identifies the extension number of the set that placed the 911 call (example).
- if an attendant clears a 911 alarm the system generates two logs. The first log identifies the attendant console that cleared the 911 alarm (example). The second log identifies the extension that placed the 911 call. (example).

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to the detection of 911 calls:

- The system detects the digit sequence 911 if it is sent to a trunk driver for outpulsing regardless of how the digit string arrived at the trunk driver (for example, the digits can be dialed manually, speed dialed, dialed on a DTS key, dialed on a CO Line key, or modified in ARS).
- The 911 alarms are detected with Consoles. The 911 alarms will not be detected if the user has programmed a Sub Attendant set with the options needed for an attendant console.
- The system detects the digit sequence 911 regardless of the trunk type.
- The system will NOT detect 911 calls if the 911 digit sequence is preceded by any other digits when it arrives at SMDR. However, you can use the *4 delimiter in Form 22, Modified Digits Table, to ensure that SMDR only receives 911. For example, to remove the leading digit 9 from 9911, program 1 as the Quantity To Delete and *49*4 as the Digits To Be Inserted. SMDR will receive 911 and the system will detect the 911 call. Note that the *4 delimiter does not impact the digits that are sent to the trunk for outpulsing; it only removes the digits that are sent to SMDR.

  You must use the ∗4 delimiter to not send digits to SMDR that precede the 911 digit string if ARS programming adds

  - a carrier code
  - an access code for network routing.
- If a user dials digits before dialing 911 and if these digits are sent to the trunk before the 911 sequence, the 911 call will not be detected by the system.
- Digits that are used by the PBX and that are not sent to the trunk driver will not prevent the system from detecting 911 calls. For example, if a user dials 9 to access an outgoing line followed by 911.The system needs to be programmed in the ARS tables to strip the digit 9 just like any other digit. (through Form 22) for the 911 call to take place. The leading digit 9 is programmable in Form 26 and can be any digit.
- If an attendant console is in Attendant Console Lockout state, you can still access 911 alarm information, but you cannot clear the 911 alarms.
- If the attendant console has COS 617, Immediate Off Hook Alarm and COS 616, Alarm Monitor Point enabled, then the attendant console will immediately receive an alarm when a lockout occurs on a set.
- The console can display a total of 32 alarms (lockout alarms and 911 alarms). For example, the system could report 30 lockout alarms and two 911 alarms. Any additional alarms could not be displayed on the console; however, maintenance logs would be generated for the additional alarms.
- Only one console can access and clear alarm information at a time. If an attendant at a console is reading alarm information all other attendants will see the following message when they press the FUNCTION key:

Lockout or 911 Alarm being displayed by <extension>

where **<extension>** is the extension number of the console that is displaying the alarm information.

- A one minute timeout applies to the alarm display on the console. After one minute,the system takes the console out of the 911 log and returns the console to the idle screen. At that point the console starts ringing again.
- In order to have the physical location of the 911 caller displayed on the console, you must have entered the physical location (see programming) for each

  - Extension in Form 9
  - Attendant console in Form 7
  - Non dial-in-trunk in Form 14

- Dial-in-trunk in Form 15.

- If the physical location is not programmed and if the caller is in a different tenant than the console, the caller's tenant group is displayed. The console cannot display both the physical location and the tenant information. The physical location takes precedence over the tenant information.
- The console stores the alarms in the order that they were received (first to last). The first time that you read the alarms, they are displayed in this order. However, if you exit alarm display mode and access the alarms again, you will start the list at the last displayed alarm.
- A 911 alarm will ring the console even if you set the Bell to off.
- 911 maintenance logs and 911 alarms are maintained after a system reset.
- This feature is not supported with a Centralized Attendant. If a 911 call is detected, ready to be delivered to a trunk, ONLY attendants ON THE SAME PBX as that trunk will receive the 911 reports/alarms.

  However Networking can be covered to SOME degree:

  - If you centralize both trunks and attendants on the same PBX in a network, the attendants will receive the reports/alarms regardless of where on the network they originate, because the 911 digits will ultimately be transmitted from the central PBX.
  - If the trunk used is on a different PBX than the originating caller, the report/alarm will be raised on the PBX that hosts the trunk, and may also be raised on the PBX that hosts the originating set, as long as the *4 delimiters are used to hide any digits that would precede 911 string going between the PBX's.

Programming

Complete the following programming

- In Form 3, COS Define, enable COS Option 102 (Attendant Display of System Alarms) to allow alarm information to be displayed on an attendant console.
- In Form 3, COS Define, enable COS Option 108, Attendant Audible Alarm, to allow audible notification of 911 and Lockout alarms on an attendant console.
- Enter the physical location of each
  - Extension in Form 9
  - Attendant console in Form 7
  - Non dial-in-trunk in Form 14
  - Dial-in-trunk in Form 15

  Enter the physical location in the Comments field by typing an "@" followed by the physical location of the set or trunk. Only text following the @ is displayed. A maximum of 14 characters will be displayed on the Console. For an example, see the Comments field in this Figure. Note that if you program the physical location, the tenant information will not be displayed.
- Program delimiters to strip off digits that precede the 911 digit string if ARS programming adds
  - an access code for network routing for Analog networking
  - a carrier code for Long Distance use, only if you have another carrier.

  In Form 22 "ARS :Modified Digit Table", enter ∗4 in the "Digits to be Inserted" field to start deleting digits and to stop deleting digits in SMDR.

Example:

In ARS, Form 22 is programmed to insert the digits 000 for outgoing calls through the an ISDN gateway. If a user dials 9911, assuming the first digit is programmed to be deleted, the digit string 000911 would be outpulsed to the trunk driver. In this scenario, the 911 call would go out on the trunk, but it would not be detected at the attendant console (see example).

To have the 911 call detected at the console, you must program the delimiter ∗4 before and after the ISDN gateway digits in the "Digits to be Inserted" field of Form 22 (see example).

Note that the ∗4 number sequence also stops and starts the modified digits from appearing in the SMDR records.

- If the Hotel/Motel Feature option is enabled, and if it is a requirement that 911 calls be made from all rooms, ensure that in Form 4, "System Options and Timers":
  - Option 32, "Outgoing Call Restriction" is disabled
  - Option 57, "Vacant /Reserved Room Default Call Restriction" is not set to INTERNAL
  - Option 58, "Occupied Room Default Call Restriction" is not set to INTERNAL.
- Program ARS so that the digit sequences 9911 and 911 both outpulse 911 on the trunk.

Operation

If a caller places a 911 call, the attendant console provides an audible alarm and the display shows

911 Call <Press Function Key for Details> ALARM

- Press FUNCTION

  If an attendant at another console is reading alarm information, you will not be able to access the alarm information. You will see the following:

  Lockout or 911 Alarm being displayed by <extension>

- Press SHOW 911

  The display shows the extension number that called 911. If the physical location of the set is programmed in Form 9, the location is also displayed. For example,

  1998-JUL-21 08:14:36 Extension 2056 called 911 <ROOM#212>

  If the physical location is not programmed and if the caller is in a different tenant than you, the caller's tenant group is also displayed. For example:

  1998-JUL-21 08:14:36 Extension 2056 called 911 <Tenant 1 - Sierra>

- Press CLEAR to delete the 911 alarm

  If other 911 alarms are present, the MORE softkey will be present.

- Press MORE to display the other 911 alarms
- Press EXIT to return to idle.

WARNING: Although the console alarm indicates the calling extension's location, the operator should verbally confirm the location of the emergency with the caller. For example, in a hotel environment, a guest may go to a different room to make the 911 call. In this case, the emergency is not occurring in same location as the calling extension. By confirming the location of the emergency, the operator can avoid directing the emergency assistance (for example, paramedics) to the wrong location.

Emergency Calls (911) - Detection and Reporting to Display Sets

Description

This feature presents an audible alarm and a visual signal to the display set when an extension user places a 911 call. The same audible alarm occurs when the extension user causes the handset to go locked out:

- The alarm (a lockout alarm) consists of a distinctive ringing cadence (approx. 0.6 seconds on and 0.6 seconds off) when the display set is on hook, idle, or ringing. A new call tone will alert the user if the display set is in the talk state. The volume of the alarm can be set with the COS Option 618, Alarm Audio Level for Sets.

- The visual signal consists of a flashing LED on the alarm key. Pressing the alarm key or Superkey provides the details of the alarm; the user's extension number and location, e.g., Extension 2056, Room 212. With this information, the user of the display set can provide assistance or direct emergency services (for example, police or ambulance personnel) to the extension where the call originated.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

- This feature is dependent on FOR Option 102 (Feature Level) and the COS Option 616 (Alarm Monitor Point) being enabled.
- The Alarm Key only appears on the Mitel telephone display if COS Option 616 is enabled.
- The alarm overrides and terminates an All Set Page or a Group Page if sets or consoles have COS Option 616 enabled.
- The alarm overrides Do Not Disturb.
- COS Option 618, Alarm Audio Level for Sets provides three options for the alarm level: HIGH, MEDIUM, and RINGER LEVEL.
- The LED on the alarm key will be on solid while someone is reading the alarms data. Only one set can access this data at one time. To prevent one set from monopolizing access, a two minute timer will regulate the time spent. If no keys are pressed within two minutes, the set will loose access to this data.
- The telephone display shows ALARM(S) ACTIVE on the idle display if there are one or more active alarms present. ALARM(S) ACTIVE on the display of Family 4 telephones (except the Symbol MiNET Wireless telephone) overrides the call information when the prime line rings. ALARM(S) ACTIVE on the display of Family 1 telephones (except the SUPERSET 4DN) appears on the second line of the display. On a Symbol MiNET Wireless Phone, the display shows ALARMS(S) ACTIVE.
- The display set shows whether the alarm is a 911 alarm or a lockout alarm. The SHOW LOCKOUT softkey and the SHOW 911 softkey only appears if there is an alarm of that type active. COS Option 617, Immediate Off Hook Alarm provides a lockout alarm as soon as the off-hook condition is detected. COS Option 227, Lockout Alarm Applies, provides a lockout alarm after the telephone has been off-hook for more than 45 seconds. If both of these options are enabled, the immediate alarm takes precedence.
- The alarms data receives the location information from the COMMENTS field in Form 09. If the first character is not an "@" there will be no location data and no LOCN softkey on the small display set.
- The NEXT softkey only appears if there is more than one alarm of the type (911 or Lockout) selected.
- If an alarm occurs while a set user is on a call with the handset, the user will see the alarm feature key flash and hear a single burst of tone from the speaker phone; the user on a call with the speaker phone only sees the alarm feature key flash.
- The set user will not hear the alarm if the alarm occurs at the same time as the set user is adjusting the volume or pitch on the telephone.
- Telephones and consoles return to a normal state if the system resets while the 911 or lockout alarms are active. A record will be recorded in the maintenance logs.

Programming

Complete the following programming:

- In Form 3, COS Define, enable COS Option 616 (Alarm Monitor Point)

- In Form 3, COS Define, enable COS Option 617, Immediate Off Hook Alarm, if you want immediate notification when the extension goes offhook.
- In Form 3, COS Define, program COS Option 618, Alarm Audio Level for Sets, to set the volume level for the audible alarm on the display set. You may choose HIGH, MEDIUM, or RINGER. The level for MEDIUM is 70% of the HIGH. RINGER is the same level as the normal ringing that the user has programmed for the set.
- In Form 4, System Options and Timers set the FOR Option 102, Feature Level to level 1 or greater to allow alarm information to be displayed on a display set.
- In Form 9 or with the Mitel display set, program a feature key, Alarm Key. See Feature Keys. The Alarm Key only appears on the Mitel telephone display if COS Option 616 is enabled.
- Enter the physical location of each
  - extension in Form 9
  - attendant console in Form 7
  - non-dial-in trunk in Form 14
  - dial-in trunk in Form 15

  Enter the physical location in the Comments field by typing an "@" followed by the physical location of the set or trunk. Only text following the @ is displayed. A maximum of 14 characters will be displayed on the Console. For an example, see the Comments field in this Figure. Note that if you program the physical location, the tenant information will not be displayed.
- Program delimiters to strip off digits that precede the 911 digit string if ARS programming adds
  - an access code for network routing for Analog networking
  - a carrier code for Long Distance use, only if you have another carrier.
- In Form 22 "ARS :Modified Digit Table", enter *4 in the "Digits to be Inserted" field to start deleting digits and to stop deleting digits in SMDR.

Example:

In ARS, Form 22 is programmed to insert the digits 000 for outgoing calls through an ISDN gateway. If a user dials 9911, assuming the first digit is programmed to be deleted, the digit string 000911 would be outpulsed to the trunk driver. In this scenario, the 911 call would go out on the trunk, but it would not be detected at the set.

To have the 911 call detected at the set you must program the delimiter *4 before and after the ISDN gateway digits in the "Digits to be Inserted" field of Form 22

Note that the *4 number sequence also stops and starts the modified digits from appearing in the SMDR records.

- If the Hotel/Motel Feature option is enabled, and if it is a requirement that 911 calls be made from all rooms, ensure that in Form 4, "System Options and Timers":
  - Option 32, "Outgoing Call Restriction" is disabled
  - Option 57, "Vacant /Reserved Room Default Call Restriction" is not set to INTERNAL
  - Option 58, "Occupied Room Default Call Restriction" is not set to INTERNAL.
- Program ARS so that the digit sequences 9911 and 911 both outpulse 911 on the trunk.

Operation

If a caller places a 911 call or causes their set to go locked out, the display set provides an audible alarm and a visual notification (display shows ALARM(S) ACTIVE and the alarm key flashes).

Family 1 telephones

- Press SUPERKEY or the ALARM feature key.

- Press the SHOW LOCKOUT softkey or the SHOW 911 softkey. These softkeys only appear if there is an alarm of that type active. The name and location of the extension shows on the display.
- Press the NEXT softkey if the softkey appears. The NEXT softkey only appears if there is more than one alarm.
- Press the DELETE ALARM softkey to clear the alarm.

Family 4 and Family 5 telephones

- Press SUPERKEY or the ALARM feature key.
- Press the LOCKOUT softkey or the 911 softkey. These softkeys only appear if there is an alarm of that type active.
- Press the NUM softkey or the LOCN softkey to obtain the information (extension number and location) of the extension that placed the alarm.
- Press the NEXT softkey if the softkey appears. The NEXT softkey only appears if there is more than one alarm.
- Press the DELETE softkey to clear the alarm.

Symbol MiNET Wireless Phone:

- Press SUPERKEY or the ALARM feature key.
- Press the LOCKO or the 911 softkey. These softkeys only appear if there is an alarm of that type active.
- Press the NUM softkey or the LOC softkey to obtain the information (extension number and location) of the extension that placed the alarm.
- Press the NXT softkey if the softkey appears. The NXT softkey only appears if there is more than one alarm.
- Press the DELET softkey to clear the alarm.

Emergency Calls (911) - Reporting by E-mail

Description

This feature sends an e-mail to as many as three addresses whenever a 911 is dialed from a station in the system. The recipients can use the alarm information to provide assistance or to direct emergency services (for example, police or ambulance personnel) to the extension where the call originated.

The following is an example 911 Notification e-mail:

From: E911@SX200.xyz.com
Sent: Friday, May 09, 2003 10:03 AM
Subject: E911 Caller: R.Smith, 3702 4th Fl Stat A3

Location: 350 Legget Dr, Kanata, ON
6 story building, across from big parking lot

The "E911 Caller" portion of the message is from the Comments field in Form 9 and only appears in the e-mail message if the comment begins with "@". The language of the message is the DEFAULT LANGUAGE programmed in Form 49, Voice Mail Options.

The Location portion comes from the Comments entered in Form 52.

The call is also recorded in the Maintenance log.

Conditions

- Requires SMTP e-mail service that is capable of forwarding emails.
- Failed delivery of SX-200 ICP generated emails registers a message in the pstlogA.txt or /db/pstlogB.txt logs files.

Programming

- In CDE Form 04, System Options/System Timers,
  - enable Option 126, Email Messaging
  - for Option 81, Enter offset from GMT (+/-hh:mm), enter the difference in hours and minutes between the time zone that the SX-200 ICP is in and Greenwich Mean Time (GMT). The offset, which appears in the timestamp on the 911 Notification e-mail, is important for recipients to know in case the e-mail originated from another time zone.

    Note: Manual time changes (as opposed to automatic changes that the system makes to adjust for daylight savings) will require a manual update to the GMT offset.
- In Form 47, Subform 01, System IP, enter the system hostname. The system hostname must be a valid domain host name of 49 or fewer characters that has been registered in your DNS, or has been listed in your SMTP e-mail server's Hosts file.
- In Form 49, enter the IP address of the SMTP server.
- In Form 09, enter the physical location of each extension in the Comments field by typing an "@" followed by the physical location of the extension.
- In Form 52, enter the e-mail addresses of the parties to receive the notification along with three lines of COMMENTS for the building location.

Operation

None

Emergency Calls (911) - Detection and Reporting to ONS (CLASS) Sets

Description

This feature presents a distinctive-ringing alarm on ONS (CLASS) sets and displays a caller's location when

- An extension user places a 911 call, or
- An extension goes off-hook for an extended period.

The alarm cannot be cleared from an ONS set. It must be cleared from an attendant console or display set. The user can use the alarm information to provide assistance or to direct emergency services (for example, police or ambulance personnel) to the extension where the call originated.

The alarm consists of a distinctive ringing cadence of 0.4 seconds on, 0.4 seconds off, 0.4 seconds on, followed by 3 seconds off. If the user is on the phone when an alarm is generated, the user hears a camp-on tone.

The visual signal consists of

- The display characters "911: " or "LCK: " (for an extended off-hook condition) followed by
  - the first 14 characters of location information entered for that set in the Comments field of CDE Form 09, Desktop Device Assignments. The comments must begin with an @ character to be displayed as location information.
  - the Caller ID name, if no location information is entered in the Comments field of CDE Form 09
- The Caller ID extension number

Conditions

The following conditions apply to this feature:

- Requires Feature Level 4 or greater in System Option 102.
- The ONS sets that receive notification must have a display and must support Caller ID. It is also recommended that the sets support call logs so that users can determine where a call originated if it was not answered.
- Only one extension number per tenant group can be programmed for notification. The extension number must be an ONS Ring Group master number. All ONS phones in that Ring Group must be associated with that master number.
- The master number extension used for ONS notification must not have a set connected to it and the extension must only be used for notification purposes.
- Any additional calls directed to the master number during the ringing state are not queued for ONS notification. These additional calls cannot be used to notify ONS sets.

Programming

- In CDE Form 04, System Options/System Timers, enable Feature Level 4 or greater in System Option 102.
- In CDE Form 19, Call Rerouting Table, select the ONS Notification Number for 911 Calls and Lockouts option and specify the master number in the Day Service, Night1 Service and Night2 Service fields.
- In CDE Form 09, Desktop Device Assignments, enter the location information for the extension or enter the Caller ID name in the Comments field. The location information must begin with an @ character to be displayed as location information.
- Program all ONS phones that are to receive notification as members of an ONS Ring Group.
- It is recommended that all ONS phones be programmed with COS Option 809, Standard Ring Applies, in CDE Form 03, COS Define.

Operation

If an extension user places a 911 call or if an extension goes off-hook for an extended period, all ONS phones that are programmed to receive notification will

- Present distinctive ringing
- Display the caller's location or name, plus the extension number.

To respond to a distinctive ringing alarm on an ONS (CLASS) set

1. Answer the call.

   You hear silence.
2. View the display to determine the source of the call.

To respond to a camp-on tone alarm on an ONS (CLASS) set

1. Flash the switch hook.

   You hear a dial tone.
2. View the display to determine the source of the call.

Note: Alarms must be cleared from an attendant console or display set.

Emergency Calls (911) - Reporting to PSAP

Description

This feature allows you to assign a Customer Emergency Services Identification (CESID) number to an extension. For each 911 call, the SX-200 system transmits the caller's CESID number to the network. The 911 Selection Router in the network then routes the call to the Public Safety Answering Point (PSAP).

The PSAP stores information, such as name, address, postal code, general location, and so forth, for telephone users in an Automatic Line Information (ALI) database. The PSAP uses the CESID number that you have assigned to an extension to locate the user's information in the ALI database.

Conditions

The following conditions apply to this feature

- The CESID functionality requires the PRI card or the ISDN PRI Application Software Release 6.0 or greater software.
- The CESID number is provided by the public network administrator.
- The CESID number can be a maximum of 10 digits in length (0-9).
- The CESID number is only sent to the network for a 911 call.
- The CESID number is the only information sent to PSAP from the PBX.
- An AIM device uses the CESID number that is assigned to the host DNIC set.
- A single CESID number may be used for more than one extension that is physically located together.
- CESID transmission requires an ISDN trunk. Therefore, you must dedicate ISDN trunks for 911 calls. If you do not dedicate an ISDN trunk, the 911 call will not go through if all ISDN trunks are busy.
- The SETUP message, in the Calling Party Number information element of the ISDN protocol contains the CESID.
- CAMA (Centralized Automatic Message Accounting) equipment does not support CESID transmission.
- You must provide the information data to the ALI database administrator initially, and for all future additions, moves, and changes associated with the CESID number. For example, if you move a user you must inform the administrator of the new location.
- The CESID number is sent to the network only with a 911 call. The CESID number is not dependant upon the Emergency label in ARS. The Emergency designation in ARS will not cause an alarm.
- The *03 number sequence must be programmed in the Digits To Be Inserted Field in Form 22 (ARS: Modified Digit Tablef) for all emergency 911 entry numbers, to allow the CESID to pass through.
- Form 9 and Form 7 store the name, extension number, CESID number, and the location for each extension number. The location is in the COMMENTS field with "@" as its first character. The information in the COMMENTS field appears on the attendant console and the SUPERSET telephones (Mitel 5020, 5220, 5224, and 5324 IP Phones, and SUPERSET 4125, SUPERSET 4025, and SUPERSET 4150 telephones).
- The name, extension number, location, and the CESID of the extension numbers can be printed from Form 32 (CDE Data Print) by using the entries in Forms 09 (Desktop Device Assignments), or 07 (Console Assignments). The printout could then be provided to the ALI database administrator for input into the ALI database.

Programming

To assign a CESID number to an extension on a station or a SUPERSET telephone

- In Form 9, select the SHOW CESID softkey. The CESID field appears.
- Enter the CESID number for the extension in the CESID field. The number can be a maximum of 10 digits (0-9).

To assign a CESID number to an extension on a console

- In Form 7, enter the CESID number in the CESID field.

Notes:

1. To allow the CESID to pass through, make sure that the *03 number sequence is programmed in the Digits To Be Inserted Field in Form 22 (ARS: Modified Digit Table) for all emergency 911 entry numbers.
2. To allow the attendant console or subattendant to detect the 911 call, *4 delimiters must be programmed in Form 22, before and after th e *03.

Operation

Dial 911.

Enhanced Guest Room

See Subattendant - Guest Room Enhanced

Expensive Route Warning

Description

A trunk route can be programmed to give an expensive route warning tone (three short tones). On Mitel display telephones, and on the console, the LCD displays a message. The user can continue with the call or hang up and try again later when a less expensive route may be available. SUPERSET 420, SUPERSET 430, and SUPERSET 4DN telephones provide the additional options of camping on to wait for a less expensive route or placing a callback on a less expensive route.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- If Callback is activated, the callback is to the least cost route.
- If the caller waits (camps on), the caller waits for the least cost route.
- Campon and callback to an expensive route are available to Mitel telephones that have CAMPON and CALLBACK feature keys programmed.
- Warning tone is given to all devices (including incoming tandem trunks).

Programming

Program Warning Tone (WT) in CDE Form 24 (ARS Route Lists). Refer to the Automatic Route Selection and Toll Control section, for designating expensive routes.

Enable COS Option 301 (Campon) and COS Option 237 (Outgoing Trunk Campon) for the extension to allow campon for less expensive routes.

Enable COS Option 300 (Automatic Callback) and COS Option 236 (Outgoing Trunk Callback) in the COS for the extension to allow callbacks to less expensive routes.

Operation

A trunk or extension attempts an external call and ARS cannot find a route without the expensive route designation. The warning tone is provided; if no action is taken by the caller, the call proceeds after a 5 second delay.

For Family 1 , Family 4 and Family 5 telephones, see Callback - Busy for the operation of the CALLBACK softkey and see Campon for the operation of the softkeys CAMPON, WAIT and I WILL WAIT.

Fax Tone Detection

Description

Fax tone detection allows incoming fax calls arriving via the Automated Attendant to be automatically routed to the Fax Destination

Conditions

System Option 106, Automated Attendant, must be enabled before System Option 99, Fax Tone Detection can be enabled.

System Option 99, Fax Tone Detection, must be disabled before System Option 106, Automated Attendant can be disabled.

RAD message length in the Automated Attendant should not exceed 20 seconds.

In CDE, a Fax destination can be programmed without being enabled. If the Fax Tone Detection option is enabled, but there is no Fax destination, fax tone detection is not performed.

Programming

Enable System Option 106, Automated Attendant

Enable System Option 99, Fax Tone Detection

Adjust System Option 59, Receivers Reserved for Non-Auto-Attendant. Auto-Attendant groups cannot be programmed until the option is changed from the default UNKNOWN to a number from 1 to 99.

In the Auto Attendant Hunt Group Options Subform of Form 17, program Fax Destination with the extension number of the fax machine.

Operation

When the Automated Attendant determines that the incoming call is a fax call, it routes the call to the programmed Fax Destination.

Feature Keys

Description

The programmable line keys on multi-line Mitel telephones, IP Phones and Programmable Key Modules that are commonly used for speedcall and line appearances, may also be used for feature activation; the user simply presses a feature key.

The feature keys available on Family 3, SUPERSET 410, and SUPERSET 3DN telephones include:

- Speedcall
- *Forward Always (not available on SUPERSET 3DN)
- *Forward Busy (not available on SUPERSET 3DN)
- *Forward No Ans (not available on SUPERSET 3DN)
- *Forward Busy/No Ans (not available on SUPERSET 3DN)
- *Forward All

- Account Code
- *Do Not Disturb
- Open Door
- *Auto Answer
- *Music
- Direct Page
- PA Paging
- Call Park/Remote Retrieve (not available on SUPERSET 3DN)
- Call Park System - Specific Orbits
- Call Park and Page
- Call Pickup
- Campon (I Will Wait)
- Callback
- Swap (Trade Calls)
- Phonebook
- Privacy Release
- Override (Intrude)
- Night Answer
- Night/Day Switching
- Headset Mode
- Handset Mute (not available on SUPERSET 410 telephone)
- Forward Call
- ** Single Flash
- ** Double Flash
- Release
- Respond (5207 only)
- VM Prompts
- Group Listen
- Mailbox Key
- System Park (only available on Mitel 5010 and 5215 IP Phones, SUPERSET 4015 telephone)
- Record a Call (only available on Mitel 5010 and 5215 IP Phones, SUPERSET 4015 telephone)
- *Voice mail (not available on SUPERSET 3DN)

For the feature keys above marked with an asterisk (*), an indication is given on the adjacent LCD display when the feature is active. For the feature keys above marked with two asterisks (**), the adjacent LCD or LED indicators are inoperative. For the remaining feature keys listed, an indication on the adjacent LCD or LED indicator indicates when the feature becomes available to the user. The LCD indicator for Direct Page is not used.

Fewer Feature Keys are available on the Family 1 and Family 4 telephones because most features are provided via softkeys.

The Feature Keys available on a Family 4 telephone (except Symbol MiNET Wireless) include:

- Speedcall
- Forward Always
- Forward Busy
- Forward No Ans
- Forward Busy/No Ans

- Forward All
- Call Park/Remote Retrieve
- Call Park System - Specific Orbits
- Account Code
- Do Not Disturb
- Auto Answer
- Music
- Direct Page
- PA Pager Access
- Privacy Release
- Intrude (Override)
- Night Answer
- Night/Day Switching
- ** Single Flash
- ** Double Flash
- Release
- LinePrivacy
- Call Pickup
- Call / Attn
- Data Disc
- Callback
- Forward Call
- Headset Mode
- Handset Mute (not available on SUPERSET 420 telephones)
- Group Listen
- Mailbox Key
- VM Prompts
- System Park.
- Callers
- Record a Call (not available on SUPERSET 420 telephones)
- Alarm
- Voice mail
- Open Door

For the feature keys listed above and marked with two asterisks (**), the adjacent LCD or LED indicators are inoperative.

The Feature Keys available on Family 1 telephones include:

- Speedcall
- Forward Always (not available on SUPERSET 4DN)
- Forward Busy (not available on SUPERSET 4DN)
- Forward No Ans (not available on SUPERSET 4DN)
- Forward Busy/No Ans (not available on SUPERSET 4DN)
- Forward All
- Do Not Disturb
- Auto Answer
- Call Park/Remote Retrieve (not available on SUPERSET 4DN telephones)

- Direct Page
- ** Single Flash
- ** Double Flash
- Headset Mode
- Handset Mute (not available on SUPERSET 430 or SUPERSET 4DN telephones)
- Group Listen.
- VM Prompts (not available on SUPERSET 4DN telephones)
- System Park (not available on SUPERSET 4DN telephones)
- Callers (not available on SUPERSET 4DN telephones)
- Record a Call (not available on SUPERSET 430 or SUPERSET 4DN telephones)
- Alarm (not available on SUPERSET 4DN telephones)
- Call Block (not available on SUPERSET 4DN telephones)
- Voice mail (not available on SUPERSET 4DN telephones).
- Open Door

For each of the keys listed above and not marked with two asterisks, an indication on the adjacent LCD or LED indicator is provided when the feature becomes active. For the feature keys listed above and marked with two asterisks (**), the adjacent LCD or LED indicators are inoperative.

The Feature Keys available on Family 5 telephones include:

- Account Code Verified
- Alarm
- Auto Answer
- Call Block
- Call Park
- Call Pickup
- Callback
- Call History
- Campon
- Day/Night
- Direct Page
- Do Not Disturb
- Double Flash
- Forward all
- Forward Always
- Forward Busy
- Forward Busy/No Answer
- Forward Call
- Forward No Answer
- Group Listen
- Guest Room
- Handset Mute
- Headset
- Music
- Night Answer
- Open Door
- Orbit
- Override
- Paging
- Park PA
- Park Page
- Park Page Group
- Park Page Set

- Phonebook
- Privacy Release
- Record Call
- Release
- Respond
- Single Flash
- Speed Call
- Superkey
- Swap
- System Park
- Voice Mail
- Voice Mail Prompts
- Wake Up Alarm

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- The keys are effective when the feature is applicable. For example, the Override feature key will not be effective if there are no calls in progress.
- The Single Flash and Double Flash feature keys operate only if COS 257 "Flash Over Trunk" is enabled in the set’s Class of Service Options Assignment form and "Flash over trunk" is enabled in the trunk’s circuit descriptor. These keys apply only to two-way calls between trunks in trunk groups and keysets that do not have any other calls on hold.
- The Auto-Answer, Forward All, and Do Not Disturb feature keys are effective at all times.
- The feature keys do not preclude the use of feature access codes.
- Line/personal keys on SUPERSET 410 telephones that are not assigned to a line default to being Speed Call keys. However, if you program a personal key as another type of feature key, for example as a DO NOT DISTURB key, you can’t program a speed call number into the key until you reprogram it as a PERSONAL SPEED CALL key.

Programming

Assign an access code to Feature Access Code 47 (Program Feature Key) in CDE Form 02.

Feature keys may be programmed in 2 ways:

From CDE

- Enter the Expand Set Subform of CDE Form 09 (Desktop Device Assignments).
- In the TYPE field, select feature key via the FEATURE softkey.
- Press the softkey corresponding to the desired feature.

From the Telephone

SUPERSET 3DN telephones:

- Dial the Program Feature Key access code.
- Press the desired line key.
- Dial the feature code for the desired feature key (see following table).

- Press the PROGRAM key.
- Press the SPEAKER key.

SUPERSET 410 telephones:

- Dial the Program Feature Key access code.
- Press an unused personal key.
- Dial the feature code for the desired feature key (see following table).
- Press the SPEAKER key.

Family 3 telephones:

To program a Feature Key that is not selectable via SUPERKEY:

- Dial the Program Feature Key access code.
- Press an unused personal key.
- Dial the feature code for the desired feature key (see following table).
- Press the CANCEL.

To determine what feature, if any, is programmed for a personal key::

- Press SUPERKEY.
- Press the personal key or REDIAL key. The function of the key is displayed.
- Press SUPERKEY to exit.

Family 4 telephones (except Symbol MiNET Wireless):

- Press SUPERKEY.
- Press the NO softkey until PERSONAL KEYS? appears in the display.
- Press the YES softkey.
- Press a personal key that isn't assigned to a line.
- Press the CHANGE softkey. SPEED CALL? appears.
- Press the YES softkey to program the key as a speed call key
  or
  Press the NO softkey. Another feature key option is displayed.
- Press the NO softkey until the desired feature key option is displayed.
- When the desired option is displayed, press the YES softkey to program the personal key.
- Press SUPERKEY to return to the date and time display.

To determine what feature, if any, is programmed for a personal key:

- Press SUPERKEY.
- Press the personal key. The function of the key is displayed.
- Press SUPERKEY to exit.

Family 5 Telephones:

To program a phone feature using the Settings key or Settings Application:

Note: Some features require you to program and use a Superkey feature key.

1. Press **Settings**
2. Press **Programmable Keys**.
3. Press the key you want to program.
4. If Applications are displayed, press the **View Features** softkey. (Conversely, if you want to program an application to this key, and features are displayed, press the View Applications softkey.)
5. Use the Page Navigation keys to move through the features list.

6. Select the desired feature or fixed function key from the list. A default label is automatically assigned to the key.
7. Enter the required information in the fields provided, if necessary.
8. Press **Save**.
9. Press **Close**.

To program a feature key to be the Superkey:

1. Press the **Settings** key.
2. Press **Programmable Keys**.
3. Press the key you wish to program.
4. If Applications are displayed, press **View Features**.
5. Use the Page Navigation keys to move through the features list.
6. Select **Superkey**.
7. Press **Save**.

To program feature keys using the Superkey you have programmed:

5330 Phones:

1. Press the key that you have programmed as Superkey.
2. Press No until Personal Keys? appears and then press Yes.
3. Press the key you want to program.
4. Press Change.
5. Press No until the key type you want to program appears, and then press Yes.
6. Press Save.

5340 Phones:

1.Press the key that you have programmed as Superkey

2.Press the **More** key until the **Feature Key** option appears, if necessary

3.Press **Feature Key**.

4.Press the key you want to program.

5.Press **Change**.

6.Select the feature to program. (Press More to scroll, if necessary.)

7.Press **Save**.

Family 1 telephones:

- Press SUPERKEY.
- Press the MORE softkey.
- Press the FEATURE KEY softkey.
- Press the desired line key.
- Press the CHANGE softkey.
- Press the softkey of the desired feature.
- Press SUPERKEY.

Feature Code	Feature Key
00	Speedcall*
01	Forward All
02	Account Code

Feature Code	Feature Key
03	Do Not Disturb
04	Auto Answer
05	Music
06	Direct Page
07	PA Paging
08	Pickup
09	Campon (I Will Wait)
10	Callback
11	Swap (Trade Calls)
12	Privacy Release
13	Override (Intrude)
14	Night Answer
15	Forward Call
18	Release
19	Single Flash
20	Double Flash
21	Headset Mode
22	Handset Mute
23	Call Park
24	System Park
25	Forward Always
26	Forward Busy
27	Forward No Answer
28	Forward Busy/No Answer

* After dialing 00, dial the number to be stored.

Operation

Rather than dialing the feature access code for the desired feature, simply press the feature key. Refer to the specific feature for further details.

Flash - Calibrated

Description

Telephone users access many system features using a switch hook flash from industry-standard telephones. Calibrated Flash allows the system to consistently create the proper flash time thus preventing confusion between flash and hang up attempts. On rotary dial sets, the user sends a calibrated flash by dialing the digit “1”. On DTMF sets equipped with a flash key, the user presses this key to send a switch hook flash to the system.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- DTMF sets equipped with a flash button must have a calibrated flash time of 50 to 140 ms.
- Rotary dial users can perform a calibrated flash, except when dialing a number. (The system interprets the digit 1 as part of the dialed number.)
- For ONS and OPS line cards, flash timers have no effect on the calibrated flash detection circuits.

Programming

In CDE Form 04 (System Options/System Timers):

- Enable System Option 37 (Calibrated Flash).
- Set System Option 52 (Minimum Flash Timer) to 200 ms if a Calibrated Flash line card is used.
- Enable System Option 38 (Switch-hook Flash).

In the CDE Form 03 (COS Define):

- Disable COS Option 223 (Flash Disable) to allow a flash to be processed.

Operation

In an established call:

- From a rotary set, dial the digit 1 and proceed as appropriate for the feature.
- From a DTMF set, press the calibrated flash button and proceed as appropriate for the feature.

Flash Control

Description

This set of options limits the use of consultation hold (switch–hook flash) under certain conditions when an extension is in a call with a trunk or attempts to establish a call with a trunk.

Flash On Incoming Trunk

This feature allows extension users to place an incoming trunk on consultation hold. This enables the trunk call to be transferred, held, or added to a conference. The option does not apply when the extension is talking to a DISA trunk that has dialed into the system.

Flash on Outgoing Trunk

This option is the same as the previous option but it applies to outgoing trunks.

Cannot Dial a Trunk After Flashing

This option prohibits the extension user from accessing a trunk, through dialing or picking up a trunk on hold at another extension, while a consultation hold is in progress. The option does not apply to industry-standard telephones with COS Option 203 (Broker's Call) or COS Option 252 (Broker's Call With Transfer) in their COS or when picking up trunk calls that are ringing at another extension.

Cannot Dial a Trunk If Holding or in Conference with a Trunk

This option prevents devices from dialing a trunk call or picking up a trunk from another extension while another trunk is in a call (conference or two party) on consultation hold. This option does not apply to industry-standard telephones with COS Option 203 (Broker's Call) or COS Option 252 (Broker's Call With Transfer) in their COS.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- The options are disabled if the extension has COS Option 223 (Flash Disable) enabled in its COS.
- The options are disabled for industry-standard telephones if System Option 38 (Switch–hook Flash) is disabled.

Programming

Enable any or all of the following COS Options for the extension:

212 (Can Flash if Talking To An Incoming Trunk)
213 (Can Flash if Talking To An Outgoing Trunk)
214 (Cannot Dial a Trunk After Flashing)
215 (Cannot Dial a Trunk if Holding or in Conference with a Trunk)

Operation

None.

Flash Disable

Description

An extension may be inhibited from using all services requiring the use of the switch hook flash. For Mitel telephones, this prevents the extension from putting a call on consultation hold.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- COS Options 223 (Flash Disable) is mutually exclusive with:

COS Option 224 (Flash for Attendant)
COS Option 203 (Broker's Call)
COS Option 252 (Broker's Call With Transfer)
COS Option 302 (Flash-In Conference).

Programming

Enable COS Option 223 (Flash Disable) for the extension.

Enable System Option 38 (Switch–hook Flash) to allow flashing for all industry-standard telephones in the system.

Operation

None.

Flash For Dial 0 (Attendant)

Description

An extension can be set to ring the Dial 0 Routing Point (usually the Attendant) automatically if a transfer is attempted while in an established call.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- The extension cannot access any other feature requiring a switch hook flash; such as Broker's Call, or Transfer/Conference, Call Park, or Call Hold.
- The conditions for consultation hold must be met before the call can proceed; see Transfer.
- The extension calls the DIAL 0 or PRIORITY DIAL 0 routing, based on the extension's tenant, COS, and the current NIGHT/DAY service. If there is no point specified for the given conditions then no call is made and reorder tone is supplied.
- This only applies if there is not a call on Consultation Hold.

Programming

Enable COS Option 224 (Flash for Attendant) for the extension.

Operation

While on an established call, flash the switch–hook or press the **Transfer** softkey. The extension rings the Dial 0 (Attendant) Routing Point; the other party is placed on consultation hold.

Flash For Waiting Call

Description

COS Option 205 (Flash for Waiting Call) allows a user to place a call on consultation hold and connect to a waiting call by performing a switchhook flash.

If COS Option 252 (Broker's Call with Transfer) is also enabled, a user can toggle between the two calls using the switchhook. The user is unable to form conferences if COS Option 252 is enabled.

See Campon Warning Tone to allow the telephone to receive an audible notification of waiting calls.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- This feature is for standard telephones. It is not intended for use on Mitel telephones.
- Other features that are also activated with a switchhook flash affect the operation of this feature.

  For example, if COS Option 302 (Flash in Conference) is also enabled, when the station user performs a flash to answer a camped on call, the call is conferenced in with the current call. The user can add each new call that camps on to the conference by performing another flash.
- This feature takes precedence over any other feature which uses the switchhook flash.
- The feature *Flash Disable* will not prevent this feature from functioning.
- When this feature is enabled, the features Flash For Attendant, Broker's Call, and Broker's Call With Transfer will operate normally, provided that there are no calls waiting.
- The use of this feature in conjunction with other features which use the switchhook flash may be complicated. The system will provide an indication that there is a call waiting (campon warning tone); however, the system does not provide an indication when the call is no longer waiting.
- The first waiting caller is connected; see Campon for details on the ordering of waiting calls.

Programming

The following steps are required:

- Enable System Option 38 - Switch-hook Flash.
- Enable COS Option 205 - Flash For Waiting Call in the telephone's COS.
- COS Option 252 - Broker's Call with Transfer in the telephone's COS (recommended).
- Enable campon for calling devices - see programming under *Campon.*

Operation

Upon hearing campon warning tone, flash the switchhook. The current call is placed on consultation hold, and the waiting call is connected.

If COS Option 252 - Broker's Call with Transfer is also enabled:

While on a call you hear campon tone.

- Flash the switchhook.

  Your current call is placed on consultation hold and you are connected to the camped on call.
- Flash the switchhook to toggle between the two calls.

To transfer one of the parties:

- Inform the party that you will perform a transfer.
- Flash the switchhook to connect to the other party and ask this party to hang up.

  After this party has hung up, you hear dial tone.
- Dial the number of the desired station and hang up.

The call is transferred.

Flash Timing

Description

The flash timer is a systemwide programmable item; its value applies to all industry-standard telephones in the system.

Minimum Flash Timer (10 ms units) 20 - 50 units 200 to 500 ms
Maximum Flash Timer (10 ms units) 20 - 150 units 200 to 1500 ms

Conditions

None.

Programming

Set the desired times in Form 04 (System Options/System Timers) - System Options 52 - Minimum Flash Timer and 53 - Maximum Flash Timer.

Operation

None.

Forward Campon

Description

Calls that camp on to a multi-line Mitel telephone can be selectively forwarded to the telephone's call forwarding destination. When a party camps on to a busy telephone that has call forwarding programmed (it may be active or inactive) or that belongs to a tenant that has Call Forward Busy programmed in Form 19, the person at the busy telephone can press a softkey or the FORWARD CALL feature key to forward that waiting party to the call forwarding destination. This feature can be used for both calls on internal lines as well as trunk calls.

See Call Forwarding for selecting a call forwarding destination. See Campon and Swap Campon also.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- The softkey on Family 4 and Family 5 telephones is only displayed for the first 10 seconds after a new caller camps on. After that time the softkey is removed.
- The feature key on Mitel Family 3, SUPERSET 410, and SUPERSET 3DN telephones is only displayed for the first 10 seconds after a new caller camps on. After that time the feature key is removed.
- The feature forwards the first waiting caller. See Campon for information on the ordering of waiting callers.
- The feature is only available if the forwarding destination is idle; see Call Forwarding.
- The user of a small display (Family 4) telephone in the talking mode can have the option to forward a camped-on call with the FORWARD softkey or answer the camped-on call with the TRADE softkey. The FORWARD and TRADE softkeys appear at the same time. On the Symbol MiNET Wireless Phone, the Forward key appears as "FWD" and the Trade key appears as "TRD".

Programming

Program a forwarding destination for the Mitel telephone.

For Family 3, SUPERSET 410, and SUPERSET 3DN telephones, program a FORWARD CALL feature key.

Operation

Operation varies depending upon the type of device as described below.

Family 3, SUPERSET 410, and SUPERSET 3DN Telephones:

- A new caller camps on to the telephone while you are talking to another party.
- Press the FORWARD CALL feature key. The waiting caller is forwarded to the forward destination.

Family 4 (except Symbol MiNET Wireless) and Family 5 Telephones:

- A new caller camps on to the telephone while the telephone is talking to another party.
- Press the FORWARD softkey. The waiting caller is forwarded to the forward destination.

Family 1 Telephones:

- A new caller camps on to the telephone while you are talking to another party.
- Press the CALL WAITING softkey.
- Press the FWD WAITING softkey. The waiting caller is forwarded to the forward destination.

Symbol MiNET Wireless Phone:

- A new caller camps on to the telephone while the telephone is talking to another party.
- Press the FWD softkey. The waiting caller is forwarded to the forward destination.

Group Listening

Description

Group listening allows a user to carry on a conversation using the handset while allowing others nearby to listen to the far end voice over the telephone's speaker. The set microphone is disabled in group listening mode.

Feature Availability

Click here to see a list of the phones that can use this feature.

Condition

Group listening is available on Mitel 5212, 5020, 5220, 5224, 5312, and 5324 IP Phones, and SUPERSET 4125, SUPERSET 4025 and SUPERSET 4150 sets only.

The telephone must be off hook with the voice path connected to the handset (it may be part of a conference call).

An offhook voice announce cannot be in progress.

While in group listening mode, the set volume control adjusts both the speaker volume and the handset volume together.

If the SPEAKER key is pressed while in group listen mode, the call becomes a normal two-way speaker call.

Pressing the MICROPHONE or MUTE key has no effect while in Group Listen mode.

If the handset is replaced while in group listen mode, the call becomes a normal two-way speaker call.

The Group Listen key LED is on while Group Listen mode is activated, and off when not in Group Listen mode.

Selecting a call handling key (Trade, Trans/Conf) will activate the function selected and cancel Group Listen mode.

Programming

Program Group Listen feature access key for sets that are to have access to this feature (CDE Form 09 and associated subforms).

Operation

To initiate group listening while in a handset or headset call, press the GROUP LISTEN key. The far end is now heard in the speaker.

While in Group Listen mode:

- To make the call a normal two-way speaker call, press the SPEAKER key and then hang up, or simply hang up.
- To return to a normal handset or headset call, press the GROUP LISTEN key.
- To adjust the speaker volume, press the Volume ∧ and Volume ∨ keys.

Guest Room Enhanced

See Subattendant - Guest Room Enhanced

Handset Mute

Description

This feature allows you to mute the handset microphone during an off-hook conversation. You disable or enable the handset microphone during a conversation by pressing the HANDSET MUTE key (toggle action).

Pressing the HANDSET MUTE key also allows you to toggle the audio between the handset and the headset microphone, while you are in a talk state with a headset connected. The LED for the HANDSET MUTE key is on when the headset microphone is on; off when the handset microphone is on. Only one microphone is on at any one time. Only Mitel 5020, 5220, 5224, and 5324 IP Phones, and SUPERSET 4025 (Rev level G.1 or greater), SUPERSET 4125 (Rev level C.1 or greater) and SUPERSET 4150 (Rev level G.1 or greater) telephones support this toggle action between the handset and headset operation while in the talk state.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply:

- Group Listening must not be activated.

- A feature key must be programmed as a HANDSET MUTE key.
- Pressing the MICROPHONE or MUTE key has no effect while handset mute is activated.
- If you press the SPEAKER key while the handset is muted, the call will change to a two-way handsfree call.
- Selecting a call handling key (Trade, Trans/Conf) will activate the function selected and cancel the handset mute.

Programming

Program the HANDSET MUTE key as a key type in Form 09.

You can program a HANDSET MUTE key from the SUPERKEY.

Operation

Family 3, Family 4 (except Symbol MiNET Wireless), Family 5 and SUPERSET 4150 Telephones:

- Press the HANDSET MUTE feature key to disable the handset microphone. The key LED is on while the handset microphone is muted.

  Your caller cannot hear you speak.
- Press the HANDSET MUTE feature key to enable the handset microphone. The key LED is off while the microphone is enabled.

  Your caller can hear you speak.

Handsfree Announce

Description

When paging to a set that is in handsfree conversation, the paging set receives a burst of busy tone. The paged set can press a “RESPOND” key to answer the page (the paging party has no indication until the paged party speaks).

Feature Availability

Click here to see a list of the phones that can use this feature.

Condition

Paged set cannot be in Do Not Disturb (DND) mode.

Paged set cannot be part of a conference call.

COS 501, Override Announce, must be enabled.

The handsfree call must be in a stable condition without a party on consultation hold.

Paged set cannot be operating in Group Listen mode.

Programming

Program the Direct Paging feature access code (48) in Form 02, or program a Direct Paging feature key in the Expand Set Subform of CDE Form 09 (Desktop Device Assignments).

Program a RESPOND feature key to 5207 telephones in the Expand Set Subform of CDE Form 09 (Desktop Device Assignments).

Operation

Paging Party

- press the Direct Page key
- dial the paged party number
- hear a burst of busy tone
- listen for the paged party to speak
- speak.

Paged Party

While the paged party is talking in handsfree mode, the incoming page will be preceded by a burst of ringback tone in the speaker and the display indicates the number and/or name of the paging party.

- press and hold the RESPOND softkey (or feature key on 5207 telephones) to talk with the pager. Release the RESPOND softkey (or feature key on 5207 telephones) to drop the paging connection and return to the original party.

If the paged party goes off-hook before pressing RESPOND, the original call is switched to the handset and the display remains the same.

Handsfree Answerback to a Directed Page

Description

This feature allows users to respond handsfree to a directed page. In order to respond handsfree, the user must turn on the set’s microphone in advance. Then, if a directed page is broadcast over the user's set, the set microphone is activated allowing the user to speak handsfree to the calling party.

The called party is alerted to the page by a short tone. Also, depending on the set type, the MICROPHONE key LED will flash or the MUTE key LED will flash rapidly.

Feature Availability

Click here to see a list of the phones that can use this feature.

Note: The SUPERSET 3DN and SUPERSET 4DN Auto-Answer for Directed Page Calls feature provides a similar type of functionality for SUPERSET 3DN and SUPERSET 4DN telephones.

Conditions

The following conditions apply to this feature:

- For DNIC phones, the COS Option 683 (Key System - Direct Paging Handsfree Answerback) must be enabled so that the user can turn their microphone on (i.e., latch the microphone) in advance by pressing the microphone key while the set is idle. For IP Phones, a user can choose to have the set's microphone automatically turned on or off when the user receives a set-to-set page without pre-latching the microphone; see Auto Latch Microphone. This feature and COS Option 683 are mutually exclusive.
- During a handsfree answerback call, pressing the MICROPHONE or MUTE key toggles the set microphone on and off.

- The telephone can be put in handsfree answerback mode while the telephone is idle (i.e., if you answer a directed page call by pressing the MICROPHONE or MUTE key, when you go back on hook the set will not be in handsfree answerback mode).
- When you use a DSS Page key for announcing or transferring a call, the call cannot be released unless the paged party either picks up the handset or hits the prime key signalling acceptance of the call. The previous actions are not necessary if the paged party has the MICROPHONE key on prior to being paged. If the MICROPHONE key is latched on, any call that is released will be 'Auto Answered' to a hot microphone. This could result in an embarrassing situation if the end user was not aware that the calling party can hear everything being said around the phone after a call is released.
- The user can switch from handsfree answerback operation to the handset by lifting the receiver, or to the headset by pressing the SPEAKER key or the prime line key.
- During a handsfree answerback call, users have access to all the features that are available during a normal call.
- A directed page cannot be made to a set in DND.
- A directed page cannot be made to a SUPERSET 401+ telephone.
- Call forwarding of a direct page to phones with COS Option 278 (Intercom Mode) enabled is ignored when the forwarding destination is another extension. This is to prevent unintended (and potentially inappropriate) calls to extensions that have Handsfree Answerback to a Directed Page enabled.
- This feature is available only with a Directed Page, not a Group Page or All Set Page.

Programming

Enable COS Option 683 (Key System - Direct Paging Handsfree Answerback) in the set’s class of service. By default, this COS Option is disabled.

Operation

Telephones with a MICROPHONE Key:

To set your telephone for handsfree answerback:

- While your set is idle, press the MICROPHONE key. The Microphone lamp on the set turns on. You are now able to answer any directed paging calls to your set using handsfree answerback.

To answer a directed page handsfree:

- When you hear a short tone from the set speaker, listen for the caller. (The Microphone lamp flashes to indicate that the handsfree microphone has turned on.)
- Speak towards the set in a normal tone of voice.
- Press the SPEAKER key or CANCEL key to end the call.

To disable the handsfree microphone during a directed paging call:

- Inform the caller, then press the MICROPHONE key (the LED turns off).

To re-enable the handsfree microphone during a call:

- Press the MICROPHONE key or lift the handset (the LED turns on).

Note: If you receive a directed page call while your set Microphone lamp is off, you can answer the page call by pressing the MICROPHONE key. However, when you hang up, the Microphone lamp will turn off and you will not be in handsfree answerback mode.

Telephones with a MUTE Key:

To set your telephone for handsfree answerback:

- While your set is idle, press the MUTE key. The Mute lamp on the set flashes slowly to indicate that the set is ready to activate the handsfree microphone when a directed page is received. Press the MUTE key a second time to turn the Mute lamp off and cause the handsfree microphone to be muted when a directed page is received. Press the MUTE key again to re-enable the handsfree microphone for directed paging calls.

To answer a directed page handsfree:

- When you hear a short tone from the set speaker, listen for the caller. (The Mute lamp will flash rapidly to indicate that the handsfree microphone has turned on.)
- Speak towards the set in a normal tone of voice.
- Lift the handset to disable the handsfree microphone and use the handset to continue the conversation.
- Press the SPEAKER key or CANCEL key to end the call.

To disable the handsfree microphone during a directed paging call:

- Inform the caller, then press the MUTE key (the LED turns on).

To re-enable the handsfree microphone during a call:

- Press the MUTE key or lift the handset (the LED turns off).

Note: When the call is terminated the set will return to the state it was in prior to receiving the directed page.

Handsfree Answerback to an Intercom Call

Description

This feature auto-answers Intercom calls and activates the called extension's microphone, allowing the user to respond handsfree. To respond handsfree, the user must turn on the set’s microphone in advance, either by pressing the MICROPHONE or MUTE key or by using the Auto Latch Microphone feature.

The called party is alerted to the page by a short tone. Also, depending on the set type, the MICROPHONE key LED will flash or the MUTE key LED will flash rapidly.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

- If the called phone is busy, then one of the following will occur, if programmed:
  - Off-hook Voice Announce
  - Whisper Announce
  - Handsfree Announce
  - Busy Tone, if none of the above three options are programmed.
- Call forwarding of a intercom calls to phones with COS Option 278 (Intercom Mode) enabled is ignored when the forwarding destination is another extension. This is to prevent unintended (and potentially inappropriate) calls to extensions that have Handsfree Answerback to an Intercom Call enabled.

Programming

In CDE Form 03, enable Option 278 (Intercom Mode) and disable Option 683 (Key System - Direct Paging Handsfree Answerback) to allow called phones use this feature.

Operation

Family 2, Family 3, Family 4, Family 5, SUPERSET 430 and 4150 telephones, and Mitel 5207 IP Phone:

See Auto Latch Microphone.

Handsfree Operation

Description

This feature enables users to have a telephone conversation without lifting the handset.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- If the set has a MICROPHONE key, it can be used to turn off the handsfree microphone during a conversation.
- If the set has a MUTE key, it can be used to turn off the handsfree microphone during a conversation.
- Calls can be changed from handsfree to the handset by lifting the handset from the set. The call can be changed back to handsfree by pressing the SPEAKER key and then putting the handset back into the handset cradle.
- Handsfree cannot be used while the Mitel telephone is using the pager.
- Calls can be ended by using the SPEAKER or CANCEL key or the hangup prompt.
- COS Option 604 (Telephone - Automatic Outgoing Line) affects handsfree dialing; see Line Selection.
- Line Preference affects handsfree dialing.

Programming

None.

Operation

Telephones with a MICROPHONE Key:

To initiate handsfree dialing:

- Press a dial pad key, a SPEEDCALL key, a line select key, or the SPEAKER key; the system selects a line to dial on.

To disable the handsfree microphone during a call:

- Inform the caller, then press the MICROPHONE key (the LED turns off).

To re-enable the microphone:

- Press the MICROPHONE key or lift the handset (the LED turns on).

Telephones with a MUTE Key:

To initiate handsfree dialing:

- Press a dial pad key, a SPEEDCALL key, a line select key, or the SPEAKER key; the system selects a line to dial on.

To disable the handsfree microphone during a call:

- Inform the caller, then press the MUTE key (the LED turns on).

To re-enable the microphone:

- Press the MUTE key or lift the handset (the LED turns off).

Handset Receiver Volume Control

Description

Mitel 5000 series IP Phone and Mitel telephone users can adjust the volume of the set's handset receiver. The handset receiver volume is independent of the set speaker (see Speaker Volume Control).

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

None.

Programming

None.

Operation

To adjust the handset receiver volume:

- During a conversation using the handset, press the VOL ∧ and VOL ∨ keys repeatedly to increase or decrease the handset receiver volume receiver volume.

Headset Operation

Description

Mitel telephones can be used with a headset. The SUPERSET 4000 series phones and the 5000 series IP phones (except the 5207) have a dedicated headset port that allows users to have their handset and a headset plugged in at the same time. Only a Mitel-approved unamplified headset can be used in this position. See the FRUs section for a list of approved headset.

The headset user answers incoming calls by pressing the line select key or the SPEAKER ON/OFF key and hangs up by pressing the CANCEL key or HANGUP softkey.

The user can combine headset operation with the auto-answer feature for complete handsfree operation. Auto-Answer is available to all Mitel telephones (excluding Mitel 5201 IP Phones, SUPERSET 401, and SUPERSET 4001 telephones) when the telephone is in headset mode.

Standard Hard Hold allows a headset (or handset) user to place a call on hard hold while the Automatic Line Selection feature is enabled. (COS Option 604, Telephone - Automatic Outgoing Line.)

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Mitel 5207 IP Phones DO NOT support the direct connection of any headset. Use of a headset with this phone requires a Plantronics M12 Vista Universal Amplifier. The amplifier plugs into the handset port on the underside of the phone and allows the user to adjust the volume level. Follow all installation and safety instructions in the Plantronics M12 documentation. Plantronics models P51, P81, P101N and Supra NC headsets are the ONLY headsets approved by Mitel for use with the 5207 IP Phone through the Plantronics M12 Vista. For optimum audio quality, set the M12 volume control to a low setting (1 to 3). Higher settings will degrade the audio quality.

  CAUTION: Using the 5207 IP Phone and a headset without a Plantronics M12 Vista Universal Amplifier may cause hearing damage.

- Users of Mitel 5020, 5220, 5224, and 5324 IP Phones, SUPERSET 4025 (Rev level G.1 or greater), SUPERSET 4125 (Rev level C.1 or greater) and SUPERSET 4150 (Rev level G.1 or greater) telephones can answer a call with the handset while the telephone is in headset mode.
- Users in talk state with a headset on a Mitel 5020, 5220, 5224, and 5324 IP Phones, SUPERSET 4025 (Rev level G.1 or greater), SUPERSET 4125 (Rev level C.1 or greater) and SUPERSET 4150 (Rev level G.1 or greater) telephones can press the Handset Mute key to toggle the audio between the headset and the handset microphones. The LED for the HANDSET MUTE key is on when the headset microphone is on; off when the handset microphone is on. Only one microphone is on at any one time. See Handset Mute.
- If the headset is unplugged (from the headset jack of a Family 3, Family 4, Family 5, or SUPERSET 4150 telephone or from the quick disconnect plug) and the handset is in the cradle, the system puts the telephone in a handsfree mode with all microphones muted. This action prevents the listener from hearing unwanted comments if your headset disconnects. The Family 3 telephone user goes offhook to talk again. The Family 4, Family 5, or SUPERSET 4150 telephone users go offhook or press the Microphone key to talk again, and may press the Speaker key to hang up.
- The system detects the connection of the headset to the dedicated headset jack of Mitel Family 3, Family 4, Family 5, and SUPERSET 4150 telephones and automatically puts that telephone in headset mode. This includes the reconnection of the quick disconnect attachment.

CAUTION: Do not enable COS option 612 (Telephone - Headset Operation) or program Headset Mode keys for SUPERSET telephone users that connect their headset to the dedicated headset jack of these telephones. If the headset mode is on when the headset is unplugged, the system will direct the audio to the handset when the user answers a call in Handsfree.

- If a Headset Mode key is programmed, the LED will be on solid when the unamplified headset is connected to the dedicated headset jack. The user will not be able to use the Headset Mode key to toggle out of the headset mode. Disconnecting the headset cord from the set disables the headset mode.
- Customers that connect their amplified headset in the handset jack can still use COS option 612 and the Headset Mode key.
- Mitel telephone compatible headsets must be supplied by the user.
- Headsets used with SUPERSET 400 series telephones must be externally powered (amplified).
- SUPERSET 4000 series telephones use a specific, unamplified headset when inserted into the headset jack on the base of the set. When the headset is not in use, the headset must be disconnected by the quick disconnect.
- The handset microphone is disconnected when a headset is plugged into the headset jack. Separate the quick-disconnect connector or unplug the headset from the dedicated headset jack to restore handset operation.
- The handset microphone is muted when a headset plugged into the headset jack disconnects.
- Amplified headsets used with SUPERSET 4000 series telephones must be plugged in to the telephone handset jack (the handset must be in the cradle).

- Agents in ACD using a Family 3 telephone must operate in headset mode to use the Auto Answer feature.
- To allow headset calls to be placed on hard hold, enable COS Option 282 (Standard Hard Hold) and disable COS Option 690 (Hold and Page).

Programming

Enable COS 612 (SUPERSET - Headset Operation) only for SUPERSET 4000 series telephones that use amplified headsets that plug into the handset jack.

Program a Handset Mute feature key if you want to toggle between the handset and headset microphones. See Feature Keys.

To program Headset Hard Hold:

- Enable COS Option 604 (Telephone - Automatic Outgoing Line).
- Enable COS Option 282 (Standard Hard Hold)

Operation

A call is ringing at the set:

- Press the appropriate line select key, or,
- Press the SPEAKER ON/OFF or HEADSET key, or,
- With Auto-Answer, the call is answered automatically.

To terminate the call:

- Press the SPEAKER ON/OFF, CANCEL, or the HANGUP key.

To toggle between headset and handset while in talk state with the headset connected:

- Press the lit Handset Mute feature key to switch from the headset microphone to the handset microphone.
- Press the unlit Handset Mute feature key to switch from the handset microphone to the headset microphone.

The LED of the Handset Mute feature key is ON when the headset microphone is on and the handset microphone is off.

The LED of the Handset Mute feature key is OFF when the headset microphone is off and the handset microphone is on.

Headset Mode Feature Key

Description

This feature allows you to turn an amplified headset on and off with a feature key. The amplified headset connects to the handset jack of a Mitel telephone.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply:

- COS Option 612 (Telephone - Headset Operation) must be disabled. If this option is enabled the set is in Headset mode, irrespective of the operation of the Headset Mode key.
- To change from headset operation to handset operation, the user must turn off Headset mode AND unplug the headset connector from the telephone (or separate the quick-disconnect connector on the headset cord).
- The handset microphone is disabled whenever the headset is connected to the telephone.
- An ACD Keys template can include a Headset Mode key.
- You can program a Headset Mode key from the SUPERKEY.
- A Headset Mode key will not function for a Mitel telephone that has an unamplified headset connected to the dedicated headset jack. The LED will be on solid when the headset is connected but the user will not be able to use the Headset Mode key to toggle out of the headset mode.
- Family 2 telephones do not support this feature.

Programming

Disable COS Option 612 (Telephone - Headset Operation) in the set's COS.

Enable Feature Access Code 56 (Headset Mode On/Off).

Program a Headset Mode feature key from the SUPERKEY on the Mitel telephone or use CDE Form 09.

Operation

To use a headset:

- Plug the headset into the telephone.

  On SUPERSET 400 series and SUPERSET 4000 series telephones, an amplified (externally powered) headset plugs into the handset jack.
- Press the Headset Mode feature key. The associated LED turns on.

To answer a call that is ringing at the set:

- Press the appropriate line select key.
- Press the Line key or Speaker key once to answer in Headset mode,
  or
  Press the SPEAKER ON/OFF key twice to answer in handsfree mode.

Note: If the Auto-Answer feature is activated, a call is answered automatically through the headset.

To terminate the call:

- Press SPEAKER ON/OFF, CANCEL, or the HANGUP softkey,
  or
  Put the headset on-hook.

To restore handset operation:

- Press the Headset mode feature key. The associated LED turns off.

or

- Unplug the headset from the telephone or separate the quick-disconnect connector on the headset cord.

Headset Operation (Amplified Headset)

Description

You can equip Mitel telephones with amplified (externally powered) headsets.You answer incoming calls by pressing the line select key or the SPEAKER ON/OFF key and hang up by pressing the CANCEL key or HANGUP softkey. You can combine headset operation with the auto-answer feature for complete handsfree operation.

Conditions

The following conditions apply to amplified headset operation:

- Sets that have COS Option 612 (Telephone - Headset Operation) enabled can only be operated with headsets.
- You must supply an amplified headset that is compatible with the Mitel telephone.
- Amplified headsets must be plugged in to the telephone handset jack and the handset must remain in the cradle.
- The handset must remain in the cradle in order to activate the amplified headset after a system reset.
- Headset Operation is not supported on the Mitel 5201 IP and SUPERSET 4001 telephone.
- Headset Operation on the Mitel 5207 IP Phone requires the installation of a Plantronics M12 Vista Universal Amplifier. See Headset Operation for further details.

Programming

Enable COS Option 612 (Telephone - Headset Operation) in the set's COS; or disable COS Option 612 and program a Headset Mode feature key (see Headset Mode feature).

Operation

To answer a call that is ringing at the set:

- Press the appropriate line select key,
  or,
  Press the SPEAKER ON/OFF key

Note: With Auto-Answer enabled, the call is answered automatically.

To terminate the call:

- Press SPEAKER ON/OFF, CANCEL, or the HANGUP softkey.

Headset with In-line Switch Operation

Description

Mitel 5000 series IP Phones and SUPERSET 4000 series telephones (excluding the Mitel 5201 IP, 5207 IP and SUPERSET 4001 telephone) can be equipped with headsets that have an in-line switch. You can use this in-line switch to

- answer an incoming call
- terminate a call
- mute the headset microphone.

You can combine headset operation with the auto-answer feature for complete handsfree operation.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- The Mitel 5201, SUPERSET 4001 telephone does not support headset operation.
- Sets with Headset Operation enabled must be operated only with headsets.
- SUPERSET 4000 telephone compatible headsets must be supplied by the user.
- SUPERSET 4000 series telephones (excluding use a specific, unamplified headset that connects to the headset jack on the base of the set. The handset must remain in the cradle.
- The handset microphone is disabled when a headset is plugged into the headset jack on the base of the set. To restore handset operation, turn off headset mode and separate the quick-disconnect connector on the headset cable or unplug the headset from the headset jack.
- If the headset is connected and ready for use, background music is not available at the set.

Programming

To program headset mode through CDE, enable COS Option 612 (Telephone - Headset Operation) in the set's COS.
OR
To allow a user to turn headset mode on and off from the set, disable COS Option 612 and program a Headset Mode feature key (see Headset Mode feature).

Operation

To answer a call that is ringing at the set:

- Flash the headset in-line switch.

Note: With Auto-Answer enabled, the call is answered automatically.

To terminate the call:

- Flash the headset in-line switch.

To mute the headset microphone:

- Press and hold the headset in-line switch.

Hold

Description

This feature enables the telephone user to place the current call on hold, then replace the handset or use the telephone for other calls. While the call is held, the user can select all features normally available on the telephone. The held call can be retrieved at the telephone that placed it on hold or at another telephone.

Do not confuse the call hold feature described here with the temporary consultation hold that occurs during a call transfer; see Transfer.

See Add Held And Auto-hold for set hold features.

See Standard Hard Hold for headset hold features.

See Attendant Hold Positions for attendant hold features.

See Subattendant Hold Positions for subattendant hold features.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- When the hold is done, the Mitel telephone user enters the select line feature to choose another line; see Line Selection.
- Connected music is heard if it is available and continues even if the held party starts recalling to the Mitel telephone. If music is not programmed, silence is provided until the hold timeout period, when ringback is provided.
- All Mitel telephones in a conference call can put the conference call on hold.
- The HOLD button is only available when talking to another party or when in a conference call. A hold can be done implicitly at other times through AUTO-HOLD.
- For industry-standard telephones, COS Option 211 (Call Hold and Retrieve Access) must be enabled in the set's COS.
- At industry-standard telephones, the Call Hold feature does not operate if COS Option 223 (Flash Disable) or COS Option 224 (Flash For Attendant) is enabled in the set's COS.
- An industry-standard telephone can have only one extension or trunk on hold at a time. It cannot put a conference call on hold.
- When an industry-standard telephone puts a call on hold it is free to make and receive other calls.
- When the user of an industry-standard telephone that has COS 203 (Broker's Call) enabled attempts to place a second call on hold, the call currently on hold is swapped with the call being placed on hold (i.e., when the second call is placed on hold, the user is reconnected to the first call that was placed on hold).
- When an industry-standard telephone puts a call on hold, the call is not occupying the station's key line if it has one programmed. A Mitel telephone with an appearance of the line can put a call on hold on the industry-standard telephone's key line as well.
- The Call Hold access code does not apply to MMe. telephones that have a HOLD key.
- If a call is on hold at a Mitel telephone's prime line, other calls can be made only if the set has other line appearances programmed.
- There is no time limit on holding conference calls on a line.
- There is no time limit on holding calls on a personal outgoing line.

Hold Retrieve

- Calls can be retrieved remotely when calls are held on industry-standard telephone key lines or Mitel telephone prime line appearances (key line appearances only).
- Calls cannot be retrieved from logical lines, private trunk and private outgoing lines or from multi-call line appearances.
- A call held on a key line appearance of a Mitel telephone can be retrieved by pressing the line key of the held call. Calls held on key line appearances on a Mitel telephone can also be retrieved from an industry-standard telephone.
- A held call cannot be retrieved if the retrieving party has a consultation hold in progress and the held party is an industry-standard telephone.
- A held call cannot be retrieved if the retrieving party has a consultation hold in progress and the held party is a Mitel telephone with COS Option 233 (Never a Consultee) enabled.
- A held call cannot be retrieved if COS Option 215 (Cannot Dial A Trunk If Holding A Conference With A Trunk) is enabled and the retrieving extension has a trunk in the call on consultation hold.
- When a two-party call on a key, private trunk line or direct trunk select line is put on hold, other Mitel telephones where the line appears can select the held line to connect to the held party.
- Conference calls on hold cannot be retrieved from other sets. They can be taken from hold through other appearances of the line only if the line was put on hold at those appearances as well.

- When a call is retrieved from an industry-standard telephone, the first check is for a call held by the telephone itself. If no call is being held, and if the industry-standard telephone has a key line appearance, then the line is checked for a held call.
- The activity of the holding set does not affect the ability of other devices to retrieve held calls.
- Industry-standard telephones, Mitel telephones, and TIE trunks can retrieve remotely-held calls.
- Calls placed on Hold (red HOLD button) at a set cannot be retrieved by the console.

Hold Timeout

- When the held party starts ringing the holding set, the ringing is done as if the held party had just called the holding line. That is, all appearances of the line ring, with delay ring etc. operational.
- If Auto-Answer is activated on the holding Mitel telephone then it is ignored when the held party rings the set.
- If call forward - no answer is activated on the holding set, the forwarding is done unless COS Option 222 (Call Forwarding Inhibit on Hold Timeout) is in the holding set's COS. Other types of call forwarding are ignored.
- Once the held party starts ringing the holding set, the Call Hold Retrieve feature access code no longer picks up the held party because it is ringing the holding party. The Directed Call Pickup feature can then be used.
- Recall no answer is operational when the held party rings the holding set.
- If a party held by an industry-standard telephone attempts to ring the holding telephone, while the holding telephone is busy, then a recall is performed. See Recall.
- The Do Not Disturb feature is ignored when a held party rings an industry-standard telephone.
- The party is held on the line for the time specified in COS Option 254, Call Hold Recall Timer in the holding extension's COS. After that time, the held party starts ringing the line it is held on.

A call cannot be put on hold:

- if an attendant is in the call
- if there is a party in the call with the non-busy extension feature enabled
- if a party in the call has the call on consultation hold
- if there is an override in progress and the set is not the party being overridden
- if the set is a Family 1 telephone using the call waiting feature
- if the call is on consultation hold
- if the trunk in a two party call is still dialing digits out.

Programming

Assign an access code to Feature 21 (Call Hold) for industry-standard telephones to put a call on hold.

Assign an access code to Feature 22 (Call Hold Retrieve - Local) to allow industry-standard telephone users to retrieve calls on hold at their sets.

Assign an access code to Feature 23 (Call Hold Retrieve - Remote) to allow devices to retrieve calls at other extensions.

Enable COS Option 211 (Call Hold and Retrieve Access) for the station set to put a call on hold, and for all devices to retrieve a call.

Set COS Option 254, Call Hold Recall Timer in the holding extension's COS to the desired Call Hold recall time (1 to 10 minutes).

Enable COS Option 222 (Call Forwarding Inhibit On Hold Timeout) in the extension's COS, if forwarding is not desired on a hold time-out.

Operation

Operation varies depending upon the type of device as described below.

Industry-Standard Telephones:

To place a call on hold:

- Flash the switch–hook.
  Transfer dial tone is returned.
- Dial tone is returned.

The caller is held and hears music, if provided. The extension may make or receive calls or access features in the normal manner.

To retrieve the call locally:

- Obtain dial tone.
- Dial the Call Hold Retrieve - Local access code.
- The set is reconnected to the held call.

To retrieve the call remotely (from another extension):

- Obtain dial tone.
- Dial the Call Hold Retrieve - Remote access code.
- Dial the number of the extension holding the call.
- The call is connected to the remote extension.

Family 2 Telephones:

To place a call on hold:

- Inform the caller, then press the red HOLD key.
- Dial tone is returned.

The caller is held and hears music, if provided. The extension may make or receive calls or access features in the normal manner.

To retrieve the call locally:

- Press the red HOLD key.
- The set is reconnected to the held call.

To retrieve the call remotely (from another extension):

- Dial the Call Hold Retrieve - Remote access code.
- Dial the number of the extension holding the call.
- The call is connected to the remote extension.

Multi-line SUPERSET Telephones, and Mitel 5020, 5220, 5224, and 5324 IP Phones:

To place a call on hold:

- Inform the caller, then press the red HOLD key. The call is held and the caller hears music, if provided.
- The line status display associated with the call on hold flashes as a reminder.
- The set may select another line to make calls or to access features in the normal manner, or hang up.

To retrieve the call at a phone that has an appearance of the held line:

- Press the line select key associated with the call on hold. The call is connected to the remote set.

The set user can add a call on hold to another line to form a conference or to move an established call from one line to another; see Add Held.

To retrieve the call at a phone that does not have an appearance of the held line:

- Obtain dial tone.
- Dial the Call Hold Retrieve - Remote access code.
- Dial the number of the extension holding the call.
- The call is connected to the phone.

Mitel 5010, 5212, 5215, and 5312 IP Phones:

To place a call on Hold:

- Press HOLD.

To retrieve a call from Hold:

- Press the flashing line key.

To retrieve a call from Hold at another station:

- Press the flashing line key.

-or-

- Dial the Call Hold Retrieve (Remote) access code and the number of the station that placed the call on Hold.

Family 5 Telephones:

To place a call on hold:

- Inform the caller, then press the red HOLD key. The call is held and the caller hears music, if provided.
- The line status display associated with the call on hold flashes as a reminder.
- The set may select another line to make calls or to access features in the normal manner, or hang up.

To retrieve the call at a phone that has an appearance of the held line:

- Press the line select key associated with the call on hold. The call is connected to the remote set.

To retrieve the call at a phone that does not have an appearance of the held line:

- Obtain dial tone.
- Dial the Call Hold Retrieve - Remote access code.
- Dial the number of the extension holding the call.
- The call is connected to the phone.

Symbol MiNET Wireless Phone:

To place a call on hold:

1. Press the FCT key.
2. Use the scroll keys to locate "Hold".
3. Press the SEND key to select "Hold".

To take a call off of hold:

1. Press the FCT key.
2. Use the scroll keys to locate "Off Hold."
3. Press the SEND key to select "Off Hold".

Hold and Page

Description

The Hold and Page feature provides users with the option of placing a Page after putting a call on hold. The type of paging used will be:

- If the extension belongs to a Page Group, the system will try to page all members of this group as well as making a PA Page if that option has been enabled.
- If the extension does not belong to a page group, an All Set Page will be carried out provided that:
  - The holder has All Set Page enabled and there are sets with Can Receive All Set Page enabled.
  - In addition to the All Set Page, a PA page will be attempted if the extension has All Set Page Includes Overhead Paging enabled.

After a call has been placed on hold, the user must press the Hold key a second time within a defined time period for the system to search for a free internal line and initiate a page. The type of page initiated will differ depending on the following:

- If the user belongs to a paging group, all members of the group will be paged. If COS Option 686 (Group Page includes Overhead Page) is enabled, the page will also be directed to a public address system.
- If the user does not belong to a page group, an All Set Page will be initiated provided the user has COS Option 684 (Key System – Can Make All Set Page) enabled and sets with COS Option 685 (Key System - Can Receive All Set Page) enabled are available. Along with the All Set Page, a PA page will be initiated if the user has COS Option 687 (All Set Page Includes Overhead Paging) enabled.

For Subattendants using Hold Position keys, the user must press the same Hold Position key within the time period to activate the page.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- System Option 102 in Form 4 (Feature Level) must be set to 6.
- At least one other internal line (pog, multi, key) must be programmed and available to use.
- Existing restrictions for placing a call on Hold apply.
- Existing restrictions for Paging (Group Page, All Set Page, and PA page) apply.
- The system timer will be cancelled if the user retrieves the call from hold before the time period ends.
- Paging will not take place if the user presses the hold key after the held party hangs up.
- Paging will not take place if the user presses the hold key after the system timer expires.
- Paging will not take place unless there is a free internal line available for multi-line sets. (See Line Selection for a definition of a free internal line.)
- Hold and Page applies to subattendant sets when a call is put on hold via the Hold Position key.
- Hold and Page does not apply to the Auto-Hold feature when a user is on a call on one line and presses another line.
- The Hold feature will remain unchanged if COS Option 690 is disabled.
- Hold and Page is not available on SpectraLink NetLink Wireless telephones.

Programming

Set System Option 102 (Feature Level) to 6.

Enable COS Option 690 (Hold and Page) in the extension's COS.

Set System Option 136, Dual Function Key Timer to the desired Hold and Page time (1 to 5 seconds) for sets placing calls on hold. The default value of this timer is 2 seconds.

Operation

Operation varies depending upon whether the red hold key is being used or a subattendant hold position key.

Red Hold Key

To place a call on hold and page:

- Inform the caller, then press the red HOLD key.
- Press the red hold key again. The system will try to find a free internal line to page.

To place a call on hold and not page:

- Inform the caller, then press the red HOLD key.
- Dial tone is returned.
- Press any key except for the red Hold key to use the normal Hold feature.

Hold Position Key

While a call is in progress, a subattendant pressing a Hold Position key will place the call on hold on that Hold Position key as follows:

- If the call placed on hold was from an internal line, the system will automatically return dial tone to the subattendant on that internal line.
- If the call that is put on hold was using an external line, the external line will also be in use and the subattendant's telephone will display SELECT LINE.

To place a call on hold and page:

- Inform the caller, then press the a Hold Position key.
- Press the same Hold Position key again.

Note: If the call held on the Hold Position key was from an external line, the system will try to find a free internal line to page. If the call held on the Hold Position key was from an internal line, the system will use the same line to page.

Hold Reminder

Description

This feature reminds a user that there is a call on hold at the set. The user hears a single burst of tone at regular intervals until the call is retrieved from hold.

You can program the length of time that the system waits before providing the first reminder tone, as well as the time interval between the reminder tones.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- A held party will never recall.

See Hold for more details on holding calls.

- A reminder tone is supported if COS 681, (Key Set/Sub Att - Call Hold Notify Timer) is disabled.

Programming

Complete the following programming:

- Program the length of time that the system should wait before providing the first reminder tone. Enter the desired time (1 to 10 minutes) for COS Option 254 (Call Hold Recall Timer) in the telephone's COS.
- Set the time interval between the reminder tones by entering the desired time interval (0 to 600 seconds) for COS Option 681 (Key Set/Sub Att - Call Hold Notify Timer) in the telephone's COS. A reminder tone will not exist if this COS option is set to zero.

Operation

To retrieve a call after hearing the hold reminder:

- Press the flashing held party's key.

Holiday Messages

Description

Mitel telephones can display a holiday message at Christmas and New Years.

Every minute, the holiday messages alternate with the usual time and date message that appear on the Mitel telephone display.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The Christmas message is displayed from December 23 to December 27 and the New Year's message from January 1 to January 4. The holiday message will over write the time and date display on the telephone.

Programming

Enable System Option 20 (Holiday Messages) to show holiday messages.

Operation

The regular message and the holiday message alternate every minute.

Hot Line

Description

Individual telephones can be programmed through CDE as hot lines. When a hot line telephone goes off-hook, the system automatically dials a pre-programmed number (internal or external). The number is specified as the call forwarding destination for that extension.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- The extension must be programmed as a Manual Line.
- Manual Line and Automatic Callback are mutually exclusive.
- The extension must have access to Abbreviated Dial, and External Call Forwarding (if it is to dial an external number).
- The type of forwarding programmed has no effect on the feature.
- If the extension is called, then the forwarding feature still works as usual.
- The hot line call is a reroute to the forward number; see Call Rerouting.
- Valid hot line destinations include all valid call forwarding destinations.
- A DTMF receiver circuit is needed for the hot line call regardless of the type of the destination.

Programming

Enable COS Option 228 (Manual Line) for the extension.

The attendant must program a “Call Forward - Always” destination number for that extension; see Attendant Call Forward Setup And Cancel.

If programming the destination number from the extension then enable one of the call forwarding COS options in the extension’s class of service.

Operation

Lift handset; the extension is rerouted to the forwarding destination automatically.

Hotel/Motel (Lodging)

Description

For information on this feature, refer to the Hotel/Motel Feature Package Description section.

Conditions

Hotel/Motel is not compatible with the Property Management System (PMS).

Programming

The following System Options apply:

- 02 - Message Lamp Test Enable
- 04 - Message Waiting and Message Register Clear Print
- 09 - Attendant Call Block
- 11 - Automatic Wakeup

- 12 - Auto Wakeup Alarm
- 13 - Auto Wakeup Print
- 14 - Auto Wakeup Music
- 23 - Message Register Count Additional Supervisions
- 24 - Message Register Audit
- 25 - Message Register Zero After Audit
- 27 - Room Status Audit
- 32 - Outgoing Call Restriction (mutually exclusive with 33)
- 33 - Room Status (mutually exclusive with 32 and 108)
- 34 - Auto Room Status Conversion / Auto Wakeup Print
- 40 - Message Register Follows Talker
- 49 - Pseudo Answer Supervision Timer
- 56 - Auto Room Status Conversion / Auto Wakeup Print Timer
- 57 - Vacant/Reserved Room Default Call Restriction
- 58 - Occupied Room Default Call Restriction
- 107 - Lodging (Hotel/Motel) (mutually exclusive with 108)
- 108 - Property Management System (mutually exclusive with 33 & 107)

The following COS Options apply:

- 101 - Attendant O/G Restriction / Room Status Setup
- 105 - Attendant Guest Room Key
- 113 - Attendant Call Block Key
- 202 - Alarm Call
- 204 - Call Block Applies
- 220 - Do Not Disturb
- 230 -Message Register Overflow Alarm
- 231 - Message Waiting Setup - Bell (mutually exclusive with 232)
- 232 - Message Waiting Setup - Lamp (mutually exclusive with 231)
- 244 - Room Status Applies
- 322 - Confirm Wakeup by Offhook
- 608 - SUPERSET Telephone - Room Status Display
- 610 - SUPERSET Telephone - Guest Room Template
- 703 - Message Register Applies

The following Feature Access Codes apply:

- 35 - Maid In Room
- 36 - SUPERSET Telephone Room Status Display
- 40 - SUPERSET Telephone Maid In Room Status Display

Refer to the Hotel/Motel Feature Package Description section, for further information.

Operation

Refer to the Hotel/Motel Feature Package Description section.

Hunt Groups

Description

Hunt groups, or master number hunting, allows a collection of devices to share a common access code. A caller can be routed to or dial the access code (the master hunt number of the hunt group), and have the call completed to an available extension in that hunt group. Extensions within a hunt group may still be accessed directly by dialing the extension number.

A hunt group is busy if all members of the hunt group are busy.

If all devices in a hunt group are busy and there is an overflow point programmed, the call is forwarded to the overflow point (if it is available). If there is no overflow point programmed or if it is busy, then the caller finds the hunt group busy. See Campon for information on campon to busy hunt groups.

If the caller camps on then the first extension in the hunt group list without Do Not Disturb activated receives a campon beep and Swap Campon capability for the first party camped on to the hunt group; see Swap Campon.

See Trunk Groups for details on trunk groups.

Some special types of hunt groups include:

- Recording hunt group; see RAD Support.
- UCD Agent hunt group; see Uniform Call Distribution.
- Automated Attendant hunt group; see Automated Attendant.
- Voice mail hunt group; see Voice mail - ONS Port.
- Recall Appearance Back to Originating Set (RABTOS) hunt group; see Mitel Application Interface (MAI).
- Ring Groups; see Ring Groups.

Two types of hunting are provided by the system, circular and terminal:

- Circular hunting starts at the extension after the last extension in the hunt group to which a call was completed (the extension rung), and hunts overall extensions in the hunt group in the sequence programmed. Hunting stops at the first idle extension found.
- Terminal hunting starts at the first extension in the hunt group and terminates at the first idle extension found. Hunting takes place in the order in which the extensions were programmed into the hunt group.

Conditions

The following conditions apply to this feature:

- An extension must be programmed before programming it into a hunt group.
- A maximum of 99 hunt groups may be programmed.
- A hunt group may have a maximum of 50 members.
- The overflow is only checked once, when the call is initially made to the hunt group.
- An extension cannot be assigned to more than one hunt group.
- An extension in a hunt group is passed in the hunt if:

  it is busy
  Do Not Disturb is set
  it is busied-out
  it is locked out
  it is a SUPERSET telephone with a busy prime line.

- The following overflow points are allowed: hunt groups, sets, extensions, datasets, consoles, LDNs, Night Bells, and ACD paths.
- A hunt group can have only one overflow point; however, a device can be the overflow point for more than one hunt group.
- Overflow is not permitted if the overflow point has forwarding active.
- Do not delete programming for a hunt group overflow point. A hunt group should always have a valid overflow point.
- When a hunt group overflows to another hunt group, overflow to the second hunt group's overflow point is not permitted.
- All tenant and COS checks involving a hunt group use the first programmed member of the hunt group.
- Recall never involves hunt groups (unless a reroute through CALL REROUTING) - recall can only occur to the individual hunt group members.
- If there is no hold timeout point and recordings are given, then for loop start trunks that do not provide release supervision, the system does not disconnect the trunk. It is important to provide a routing point for these unanswered calls to ensure that the loop start trunk is either answered, or timed out for no answer and then released.

Programming

Program all extensions via CDE Form 09 (Desktop Device Assignments). Program datasets via CDE Form 12 (Data Assignment).

Program the hunt group via CDE Form 17 (Hunt Groups).

Enter the desired extension numbers of sets, datasets.
Enter the hunt group access code.
Enter the appropriate options for the hunt group.
Assign the appropriate hunt group type.

Operation

None.

I Hold You Hold

Description

I Hold You Hold allows users of shared lines to identify who put a call on hold. The Hold LED of the phone on which the call was first held flashes green (I Hold). The Hold LED flashes red for users with a line appearance of that held line (You Hold).

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

None.

Programming

None.

Operation

None.

Illegal Access Intercept

Description

Calls to restricted access codes or extension numbers can be routed to a given answering point for completion. This point can be an LDN position on the attendant console (see Console LDN Keys) or any valid reroute point. Illegal number intercept points can be programmed to be different or the same for DAY, NIGHT1, and NIGHT2 operation.

Conditions

The following conditions apply to this feature:

- If the required programming is not done, such calls receive reorder tone.
- Only sets, DISA trunks, and CO trunks are routed to the answering point.
- See DID/Dial-in/Tie Intercepts for illegal number handling for DID and Tie trunks.
- If the call is routed to a console, the call is shown as an intercept call at the console.

Programming

To cause all calls to restricted numbers to be routed to a specific answering point, access CDE Form 19 (Call Rerouting Table) and enter the desired answering point access code, into the appropriate column for the “Station Illegal Number Routing For This Tenant” Call Type.

Operation

None.

Inhibit Trunk Ring-Me-Back During Dialing

Description

This feature inhibits the operation of a particular instance of the Station Transfer Security feature. If an industry-standard telephone is dialing and goes on–hook while a trunk is on consultation hold, that trunk does not ringback the station and is instead dropped. This prevents trunk lock-ups when the flash on the trunk was intended as a hang-up and the station user did not expect a trunk to be on consultation hold.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- COS Options 401 (Call Park) and 403 (Trunk Recall Partial Inhibit) are mutually exclusive.
- This only operates for industry-standard telephones.
- This only operates for single trunks on consultation hold. Non-trunks and conference calls ring back the industry-standard telephone.
- Serial trunks are not dropped; they recall back to the console.

- If the station has called another party that has external call forwarding enabled and hangs up during the dialing of the external number, the held trunk is transferred to the external number and is not dropped.

Programming

Enable COS Option 403 (Trunk Recall Partial Inhibit) for the extension.

Operation

Establish a trunk call.

Flash. Dial tone is returned.

Hang up. Trunk is dropped.

Intercept to Recorded Announcement

Description

Incoming trunk calls can be intercepted to groups of recording devices after dialing vacant numbers, reaching busy extensions, obtaining no answer, or as required.

Conditions

None.

Programming

See RAD Support.

Enter the extension of the recording device hunt group into CDE Form 19 (Call Rerouting Table) for the appropriate routing.

Operation

None.

Intercom Calls

Description

An Intercom call is a call between two extensions. Users of phones with paging capabilities can switch between ringing and paging while calling a set that supports paging.

A page can be one- or two-way, depending on whether the option to automatically latch the microphone is enabled or disabled on the called extension.

An Intercom called placed by dialing the party's extension number (as opposed to using the Directed Page feature) can also be one- or two-way. For more information see Handsfree Answerback to an Intercom Call.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

- Phones with COS Option 278 (Intercom Mode) enabled always get paged when called from a device with paging capabilities.
- Call forwarding of Intercom calls to phones with COS Option 278 (Intercom Mode) enabled is ignored when the forwarding destination is another extension. This is to prevent unintended (and potentially inappropriate) calls to extensions that have Handsfree Answerback to an Intercom enabled.
- Applies to all system-supported Mitel telephones (IP and non-IP) that can receive a set-to-set page.
- Phones can only block pages through the Superkey when Intercom Mode is disabled.
- When a caller with a call on softhold pages a phone, the caller can transfer the held call to the paged party if the paged party is not already on an active call. A supervised transfer occurs when the page is answered and the caller transfers the call. If the page is not answered, the page will terminate and the transfer will be unsupervised.
- If the paged phone is busy, then one of the following options will occur, if programmed:
  - Offhook Announce
  - Whisper Announce
  - Handsfree Announce
  - Busy Tone, if none of the above three options are programmed.
- Callbacks cannot be left on phones with COS Option 278 (Intercom Mode) enabled.
- If the telephone has a MUTE or MICROPHONE key, it can be used to turn off the handsfree microphone during a conversation.
- Sets can switch between ringing and paging, and the console can switch from ringing to paging.

Programming

- In CDE Form 04, System Option and Timers, enable Feature Level 5 in System Option 102
- To always page phones, enable COS Option 278 (Intercom Mode) in Form 03.

Operation

Family 2, Family 3, Family 4, Family 5, SUPERSET 430 and 4150 telephones, and Mitel 5207 IP Phone:

To switch between a ringing intercom call and a paged intercom call

- Make a call.
- Do one of the following:
  - Press **Direct Page**.
  - Enter the Key System - Direct Paging feature access code.

    Repeat to toggle between paging and ringing.

Superconsole 1000 Attendant Console or 5540 IP Console:

To switch from a ringing intercom call to a paged intercom call

- Make a call.
- Press **Set Page**.

Handsfree Microphone on Telephones with a MICROPHONE Key:

To disable the handsfree microphone during a call:

- Inform the caller, then press the MICROPHONE key (the LED turns off).

To re-enable the microphone:

- Press the MICROPHONE key or lift the handset (the LED turns on).

Handsfree Microphone on Telephones with a MUTE Key:

To disable the handsfree microphone during a call:

- Inform the caller, then press the MUTE key (the LED turns on).

To re-enable the handsfree microphone during a call:

- Press the MUTE key or lift the handset (the LED turns off).

Internal Number Block

Description

Internal Number Block provides set/station number confidentiality between calls. This feature blocks the number of a station/set from appearing on another station or set on the same system.

In a hotel environment, a hotel operator can choose whether or not the room number of a guest will appear on another guest's telephone.

Conditions

- Applies to display sets from the SUPERSET 4000 and the SUPERSET 400 telephone series. Also applies to ONS/CLASS stations. SUPERSET 4DN telephones, SUPERSET 3DN telephones, attendant consoles, and sets programmed as subattendants do not support this feature, that is, their numbers are not blocked and they will always show internal numbers.
- This feature is dependant on System Option 102, Feature Level 2 or greater.
- For ONS/CLASS stations, the purchasable System Option 89, CLASS Functionality for ONS Sets must be enabled.
- For ONS/CLASS stations, COS Option 505, ONS Stations Support CLASS must be enabled.
- COS Option 507, Station/Set: Allow My Number to be Displayed controls whether or not the number of a station or set appears on another station or set on the same system. The blocking of the number works independently of who is the caller and who is the called party. The default for COS Option 507 is ENABLED (showing the internal number).
- COS Option 508, Station/Set: Show Internal Numbers on My Phone, overrides COS Option 507. COS Option 508 allows a station or set to always see the number of the other party. Administration telephones such as room service would use this COS option. COS Option 508 should be enabled on all DNIC voice mail ports.
- Call Logging honors the blocking with COS Option 507. If COS 507 is disabled, the call logging will show only the name, not the number, and the Call softkey will not appear.

Programming

In CDE Form 04, set System Option 102 to feature level 2 or greater. For ONS/CLASS stations, enable System Option 89, CLASS Functionality for ONS Sets.

In Form 03, Class of Service Options, enable COS Option 505, ONS Stations Support CLASS for the ONS/CLASS stations. Enable COS Option 508 on all DNIC voice mail ports. Program COS Option 507 or COS Option 508 in order to block/or not to block the display of the internal number.

Operation

None.

Inward Restriction (DID)

Description

An extension may be restricted to not receive calls directly from DID trunk calls.

Conditions

None.

Programming

Enable COS Option 226 (Inward Restriction - DID) for the extension.

Operation

None.

Language Change

Description

This feature allows Mitel display telephones to display text and softkey prompts in a different language.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The language for the Mitel display telephones is selected by a softkey. The language chosen stays with the Digital Line Card port if the telephone itself is changed.

Programming

None.

Operation

Operation varies depending upon the type of set as described below.

Note: On a Mitel 5020, 5220, 5224, 5324 IP Phone, or SUPERSET 4015 set, press the dial pad * key instead of the YES softkey, and press the dial pad # key instead of the NO softkey.

Mitel 5020, 5220, 5224, and 5324 IP Phones, and SUPERSET 4015, SUPERSET 4025, SUPERSET 4125, and SUPERSET 420 Telephones:

- Press SUPERKEY.
- Press the NO softkey until LANGUAGE? appears in the display.
- Press the YES softkey.
- Press the CHANGE softkey.
- Select the desired language.
- Press SUPERKEY.

Family 1 Telephones:

- Press the SUPERKEY.
- Find and press the LANGUAGE softkey.
- Choose the desired language.

Family 5 Telephones:

- Press the key you have programmed as Superkey.
- Scroll through the Superkey menu until the Language option appears.
- Press Language.
- Select the desired language.

Last Number Redial

Description

This feature allows the attendant console and telephone users to redial the last manually-dialed internal or external number with a single key operation. Single line telephones may use the Last Number Redial feature using an access code.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature on attendant consoles:

- The redial number is changed when an internal destination is successfully dialed or a trunk group is dialed and found busy or an external number is successfully dialed.
- The REDIAL softkey does not appear until a destination has been dialed and the number can be changed.
- The number is not updated after a call is made using the Abbreviated Dial feature.
- The REDIAL softkey can be used while calls are ringing the console.
- If a call is directed to an LDN programmed at the same key as the REDIAL key (softkey 9), the LDN label replaces the REDIAL softkey and the Redial feature is not available.
- The REDIAL key does not appear when the console is locked out; see Attendant Console Lockout.
- The redial number is lost after a system reset.

The following conditions apply to this feature on telephones:

- The Redial feature is not available when a consultation hold is in progress.
- The redial number is changed when an internal destination is successfully dialed or a trunk group is dialed and found busy or an external number is successfully dialed.
- A redial is permitted when the conditions mentioned immediately above are met. On display telephones, the REDIAL softkey is presented.
- The number is not updated when dialing on direct trunk select keys, private trunk lines, or on CO line group keys.
- The number is not updated after an external call is made using a speedcall or a system abbreviated dial.
- The redial number is lost after a system reset.
- When the REDIAL softkey on Mitel display telephones is pressed, the rules for initiating dialing and selecting a line apply as if the user had selected a speedcall key; see Line Selection.

- The redial number contains only the digits dialed until ARS dialing is completed, and the two parties are talking without further ARS processing. Digits dialed during the established call are not stored in the redial number.
- The Forced Account Codes feature disables Last Number Redial.
- Hotel/Motel internal only call restrictions applied to this extension disables Last Number Redial.

Programming

Enable System Option 29 (Telephone Last Number Redial).

Assign an access code to Feature 30 (Last Number Redial) for use with single line telephones.

Operation

Operation varies depending upon the device as described below.

Family 3, SUPERSET 4025, SUPERSET 4125, SUPERSET 410, SUPERSET 420, and SUPERSET 3DN Telephones, and Mitel 5020, 5220, 5224, and 5324 IP Phones:

- Press the REDIAL key.

Users of Mitel 5020, 5220, 5224 and 5324 IP Phones, and SUPERSET 4125, SUPERSET 4025 and SUPERSET 420 telephones can display the telephone number that is stored in the REDIAL key.

- Press the SUPERKEY and then press the REDIAL key.
  The redial number appears in the display. If the number contains more than 16 digits, the MORE softkey will be present.
- Press the MORE softkey to display the remaining digits.
- Press the SUPERKEY to exit.

Family 5 Telephones:

- Press the Redial key.

Attendant Console, and Family 1 Telephones:

- Press the REDIAL softkey.

Industry-standard and Family 2 Telephones:

- Lift the handset - dial tone is returned.
- Dial the Last Number Redial feature access code.

Symbol MiNET Wireless Phone

Note: Feature access codes are programmed at the SX-200 and are unique to each customer environment. Contact the System Administrator for specific feature access codes.

To redial the last number manually dialed:

1. Press the FCT key .
2. Press the scroll key to locate "Redial."
3. Press the SEND key to redial the number.

To redial a saved number:

1. Press the SEND key.
2. Enter "Last Number Saved" feature access code, 30.
   The name/number being dialed is displayed.

Last Party Receives Dial Tone

Description

This feature allows the last party left on a call, after the other party(s) hang up, to receive dial tone and be able to dial. Normally, this party would receive silence and after 30 seconds be locked out; see Line Lockout.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- The feature applies only to station and Mitel telephones.
- The feature has no effect on Mitel telephones which have the handset in the cradle, have the auto answer feature set, or are on a line that cannot be used for originating calls.
- The feature has no effect on station sets that are members of recording hunt groups.
- This feature does not work on telephones with either COS Option 228 (Manual Line) or COS Option 241 (Receive Only) enabled in their class of service.

Programming

Enable System Option 22 (Last Party Clear - Dial Tone).

Operation

Establish a call. All of the parties in the call except one hang up. The remaining extension hears dial tone and can dial a new call.

LCD Display

Description

The Mitel 5010, 5207, 5212, 5215, 5020, 5220, 5224, 5312, and 5324 IP telephones, SUPERSET 4015, SUPERSET 4025, SUPERSET 4125, SUPERSET 4150, SUPERSET 420, SUPERSET 430, SUPERSET 3DN, and SUPERSET 4DN telephones are equipped with a liquid crystal display (LCD). The SUPERSET 4125 telephone comes with a backlit LCD and the SUPERSET 4025 telephone has a variant with the backlit LCD. The display (not to be mistaken for the LCD line appearance indicators) indicates the date and time of day, along with softkey names for the set’s softkeys. A redial number is displayed, if applicable. Also, when a SUPERSET display telephone accesses a trunk and establishes a call, the duration of the call is displayed. For other information displayed, details can be found under the appropriate feature heading.

Conditions

The following conditions apply to this feature:

- This feature is only available on Mitel display telephones.
- The duration shown is the time that all parties in the PBX spend talking on the trunk call. If a trunk is transferred from extension to extension, the time is not restarted.

- The duration display does not indicate the complete time that the trunk has been answered where recordings are involved. The recording answers the trunk but the call duration counting does not start until the trunk is answered by an extension.
- The duration display indicates the time up to a maximum of ten hours (0 thru 9:59). After ten hours, the duration display resets to zero then continues to count for another ten hours.
- If Analog Networking is enabled, the extension number of the party calling from the other node is displayed.
- If the trunk has a name, the trunk name is displayed. If the trunk has no name programmed, the group name of the trunk is displayed. For Mitel display telephones, the digits dialed are displayed instead of the name for outgoing trunk calls.

Programming

None.

Operation

None.

Line Lockout

Description

The system locks out an extension if the extension goes off–hook and does not dial digits or go back on–hook for a length of time. Lockout also occurs if the extension does not hang up at the end of a call. In the locked-out state, the extension cannot originate or receive calls, and appears busy to potential callers.

See Lockout Alarm for an alarm generated from line lockout.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Dial tone time-out is 15 seconds, with an additional 30 seconds of reorder tone before lockout is applied to the extension.
- Calls cannot be transferred to locked out extensions.
- Locked out extensions are not hunted in hunt groups.

Programming

None.

Operation

To remove the extension from the locked-out state, go back on-hook.

Line Preference

Description

Allows the system to automatically select which line is used when the set goes off-hook to originate a call. One of the following may be selected:

- Prime line key
- Personal outgoing (O/G) key
- Multicall line key
- CO line group key
- CO line key
- DTS line key
- Private line key
- Manual (user must press a line key to originate a call).

The lines are divided into two groups. The prime, personal, and multicall lines belong to the internal group, while the CO, DTS, and private lines belong to the external line group. Additionally, the CO line group enables the selection of the first available internal outgoing line, and manual disables the feature for the set.

When the user initiates a call by going off–hook or dialing, the system checks for an available line appearance. The system scans the preferred line first. If the preferred line is idle, it is selected. If it is not idle, then other lines from the same group are scanned, starting with the first key on the set and proceeding upwards until an idle line is found.

Examples:

1. If the preferred line is a personal line, the system will scan it first. If it is not idle, the system will scan other internal lines (the prime line, other personal lines, and single appearances of key and multicall lines). If an idle internal line cannot be found, the user is prompted to select a line.
2. If the preferred line is a CO line group, the system will scan the internal lines (the prime and personal lines, and single appearances of key and multicall lines), starting with the first key on the set. If an idle internal line cannot be found, the user is prompted to select a line.
3. If the preferred line is a DTS line, the system will scan it first. If it is not idle, the system will scan other external lines (CO and private lines, and other DTS lines). If an idle external line cannot be found, the user is prompted to select a line.

The system only scans the preferred line and other lines from the same group. The system will not scan external lines if the preferred line is internal; nor will it scan internal lines, including the prime line, if the preferred line is external.

The user may override the line preference by pressing another line key prior to going off-hook for a call origination.

An alternative method of automatic line selection is available. Enable COS Option 604 (Telephone - Automatic Outgoing Line) and select MANUAL as the line preference; the system will then check for available appearances of single-line internal lines. For details, see Line Selection.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- This feature has no effect on the answering of calls.
- This feature is mutually exclusive with Line Selection; also see Line Selection.
- If this feature is used with DSS/BLF, Key System Direct Paging, or Speak@Ease, the system will select the first available internal line (starting with the prime line), even if the preferred line is an external line.

- To ensure that ARS rules are applied consistently, do not intermix CO and DTS lines on a set.
- When the system automatically selects a DTS line, users may hear a broken dial tone followed by a normal dial tone.
- A preferred line cannot be programmed on a PKM; however, if a preferred line is programmed on a set with an attached PKM, then the system will scan for available lines on both the set and the PKM.

Programming

To select or change the line preference:

From within the Expand Set Subform of CDE Form 09 (Desktop Device Assignments), press the LINE PREF softkey, and choose the desired line type.

Operation

None.

Line Privacy

Description

This feature ensures that calls on key lines, direct trunk select lines, private trunk lines and CO line keys are private. Line privacy is assigned to extensions and trunks. If an extension is assigned with line privacy, calls made from that extension are private. Other users cannot enter the call by selecting another appearance of the line. If a trunk is assigned with line privacy, a user cannot enter a call on that trunk by selecting another line appearance of the trunk.

A Mitel telephone user can override Line Privacy using Privacy Release to allow other sets to join a conversation. Refer to Privacy Enable/Privacy Release for details.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- You control line privacy by enabling or disabling COS Option 240 (Line Privacy) in the COS of an extension or trunk. By default, COS Option 240 (Line Privacy) is enabled in the COS of extensions and trunks.
- You can only disable line privacy on key lines. Line privacy cannot be disabled on key lines of logical lines, direct trunk select lines, private trunk lines and CO lines.
- Multi-line Mitel telephones can use the Privacy Release feature to disable privacy during a call.
- A maximum of five parties can be on the line at a time (including the industry-standard telephone, Mitel telephone, or trunk).
- Direct line selection by the user is the only way to join a call in progress on a line. The system does not choose an occupied line automatically; see Line Selection.

Programming

To disable line privacy, disable COS Option 240 (Line Privacy) in the COS of the industry-standard telephone, Mitel telephone, or trunk.

Operation

The two modes of operation are described below.

Line Privacy Enabled

When the line is in use, attempts by other Mitel telephones with appearances to access the line are ignored. If it is a key line of an industry-standard telephone, and the telephone goes off–hook then it receives busy tone.

Line Privacy Disabled

When the line is in use, any other appearance that accesses the line joins the conversation, preceded by the override warning tone. For key lines of industry-standard telephones this includes the station when it goes off–hook.

Line Selection

Description

The Mitel telephones are equipped to have many line appearances programmed on them. When the user of the telephone initiates dialing, the system selects a line for dialing if programmed to do this. When the set is ringing and the user goes off–hook, the system selects the line to answer. The user can also select a specific line to place or answer a call.

When the user goes off–hook to dial, the system checks for an available internal line appearance. The system looks at the prime line first. If it is idle then it is selected. Otherwise the rest of the lines on the set are scanned for the first personal outgoing line that is idle, or a single appearance key or multicall line that is idle and is not an in-only line. If such a line is found then it is selected for dialing. Direct trunk select, CO line and private trunk lines are never automatically selected and key and multicall lines that have more than one appearance (in the system) are never selected. If a line is selected then the user is put dialing on that line. If there are no available lines then the user is prompted to select a line to dial on.

When the user initiates dialing by pressing a digit or a speedcall key, the same line selection process is done as with going off–hook. When the system or the user selects a line, the system then automatically initiates speedcall dialing for the SPEEDCALL key initially pressed.

When the user answers a call by going off–hook, the system scans the set from the prime line up the set, to find a line that is actually ringing the set. The system selects the first line found and answers the call on that line.

If this feature is not selected, handsfree dialing on the keypad to initiate a call from idle is prevented. Also, when a speedcall key is pressed, the set will wait for a line select key before dialing. The set will also wait for a line key selection after (a) the SPEAKER key is pressed, or (b) when the set goes off-hook.

This process of automatic line selection is controlled using COS Option 604 (Telephone - Automatic Outgoing Line). If the option is enabled, the system automatically selects a line if one is available. If the option disabled, the user must always manually select a line when originating a call. Note that COS Option 604 does not apply to Mitel 5201, SUPERSET 4001 or SUPERSET 401+ telephones, which provide dial tone regardless of whether COS Option 604 is enabled.

COS Option 604 can be overridden by selecting a preferred line in the Expand Set subform of CDE Form 09. This alternate process of automatic line selection takes priority regardless whether COS Option 604 is enabled or disabled. For details, see Line Preference.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- The line selection mode is entered when a line is put on hold on a Mitel telephone; see Hold.
- If the user presses a speedcall key and has to select a line to use, if a direct trunk select or private trunk line is selected then the speedcall key is ignored.
- Line selection after pressing the REDIAL softkey is treated the same as pressing a speedcall key.
- This feature is mutually exclusive with Line Preference; also see Line Preference.
- If COS Option 604 is disabled for a Symbol Wireless Phone, on-hook dialing is prevented regardless of the Line Preference setting. This guards against accidental calls.
- To prevent the system from selecting a new line when a headset user places a call on hard hold (red button hold), enable COS Option 282 (Headset Hard Hold), and disable COS Option 690 (Hold and Page).

Programming

To enable automatic line selection:

Enable COS Option 604 (Telephone - Automatic Outgoing Line) for Mitel telephones to have the system select a line for outgoing calls.

Note: Ensure that LINE PREF is set to MANUAL in the Expand Set Subform of CDE Form 09.

Operation

None.

Line Types and Appearances

Description

Mitel telephones are equipped with keys that can be used as line select keys to provide additional lines to the telephones, appearances of other telephones, and direct access to trunk lines.

Mitel telephones have two components then - the telephone and the lines on the telephone. The telephone and the lines are not always busy at the same time. The system sometimes has to distinguish between the Mitel telephone and the lines on the telephone. One or more lines (except the prime line) may be in use, but the telephone itself is still idle and available for a call.

Lines have root devices, which can be telephones, Mitel telephones, or logical lines.

Every Mitel telephone must have a Prime Line. The system identifies a set by the set’s prime line extension number.

- A prime line can be a multicall or key line depending upon the type of the next programmed appearance of the prime line. If an appearance of a prime line is programmed at other sets as a multicall line, then the prime line is a multicall line. If an appearance of a prime line is programmed at other sets as a key line, then the prime line is a key line.
- The prime line is always a both way and immediate ring line.

Line types:

- Personal Outgoing Line
- Key Line
- Multicall Line
- Direct Trunk Select (DTS) Line

- Private Line.

Other line types are described under subattendant and set feature headings.

Line Appearance Variants

1. Direction: Both Way, Incoming Only, Outgoing Only. - A line appearance can be restricted to use for originating calls only, incoming calls only, or both.

Note: The outgoing only direction for a line appearance on a set is ONLY available if the line is programmed for no ring; if programmed for delayed ring or immediate ring then the line must be either incoming only or both way.

2. Ring: No Ring, Delayed Ring, Immediate Ring. - This allows new calls to an appearance to cause a Mitel telephone to ring immediately, ring after a delay or not ring at all.

Note: The duration of the delay time is determined by the class of service of the root device (trunk, Mitel telephone, industry-standard telephone) of the line appearance. This is determined by COS Option 263, Delay Ring Timer. For logical lines, the first Mitel telephone where the line is programmed is used.

3. Secretarial: Non-Secretarial, secretarial. - This allows special interaction with the Do Not Disturb feature; see Secretarial Line.

Line Ringing

For key line types, there can be only one caller calling the line at a time. When the caller calls the line, all of the appearances indicate a ringing line and the Mitel telephones where the appearances appear may start to ring if the Mitel telephones are idle.

Each line on a Mitel telephone that rings can cause the Mitel telephone to ring if the set is not in use and as long as the set does not have the Auto-Answer feature activated. If the Mitel telephone is in use and off–hook or the Auto-Answer feature is enabled, then each line that starts to ring causes the set to warble briefly if the set is off–hook. This new call ring can be limited with COS Option 611 (Telephone - Limited New Call Ring) so that only the first ringing line on a telephone provides the short ring.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Any line of any line type can have a maximum of 64 appearances (64 IP sets, or 32 IP and 32 DNIC sets), including the prime line of a set.
- An extension or trunk can be the root of only one line type.
- See Call Forwarding for details on lines and forwarding.
- For some features, the programmed sequence of appearance is important; the Review feature in CDE allows the sequence to be viewed.
- The new call ring is not applied while the party to get the ring is dialing.

Note: If a line is a phantom line in an ISDN bay, key line appearances of that line will not ring.

Programming

Specify the delay ring time in COS Option 263, Delay Ring Timer.

For Mitel telephones with heavy call traffic on several lines on the set, enable COS Option 611 (Telephone - Limited New Call Ring) to limit the new call ring given to the set.

Operation

The following table defines the meaning of the LED or LCD associated with a line key.

Flash Rate	Meaning
Off	Line is idle
On	Line is busy at this telephone (or at another telephone)
Slow Flash	Incoming Call
Fast Flash	Call on Hold at this telephone (or at another telephone)

Line Type: Personal O/G Line

Description

This line is an outgoing-only line that allows the user to make an outgoing call without making the prime line busy to incoming calls.

Conditions

The following conditions apply to this feature:

- It has no extension number associated with it.
- This feature is dependant on COS 262, Ignore Forward Busy with Free Appearance. COS 262 controls the availability of the prime line. If COS 262 is disabled the forwarding call will go to voice mail as if the phone is busy. If COS 262 is enabled, the forwarding call will ring the prime line and follow the Call Forward - No Answer route.

Programming

From within the Expand Set Subform of CDE Form 09 (Desktop Device Assignments), move the cursor to the TYPE field. Press the PERSONAL O/G softkey.

Operation

None.

Line Type: Key Line

Description

This is an appearance of an extension number that can be an industry-standard telephone's line or a Mitel telephone's Prime Line or a logical line. If the line is in use at one set, the other appearances of the line are busy and unavailable for separate calls. See the Line Privacy feature description in this section for joining established calls on key lines.

Conditions

The following conditions apply to this feature:

- Direction and ring variants can be programmed independently for each appearance.

- It can appear on several Mitel telephones.
- There can be only one appearance of any one key line on a Mitel telephone.
- A key line key can appear on a maximum of 64 IP sets, or 32 IP and 32 DNIC sets.
- If a line is a phantom line with a programmed PLID, the line can be called. Telephones with key lines can be called whether the line card is plugged in or not.

Programming

From within the Expand Set Subform of CDE Form 09 (Desktop Device Assignments), move the cursor to the TYPE field. Press the KEY LINE softkey.

Operation

None.

Line Type: Multicall Line

Description

A multicall line is an appearance of one extension number on two or more telephones. This extension number can be for another Mitel telephone or an industry-standard telephone. When one appearance of a multicall line is in use, the other appearances are still available to make or answer calls.

- Call direction, ring and secretarial variants can be programmed independently for each appearance.
- There can be up to 64 appearances of any one multicall line on a given Mitel telephone.

Conditions

The following conditions apply to this feature:

- Direction, ring and secretarial variants can be programmed independently for each appearance.
- It can appear on several Mitel telephones.
- There can be up to 64 appearances of any one multicall line on a given Mitel telephone (64 IP sets, or 32 IP and 32 DNIC sets).
- The secretarial operation feature description in this section only applies to multicall lines.
- When a multicall line is called, there can be as many callers ringing the line as there are appearances that are free to be rung. When a caller rings the line, the first appearance of the called line is rung on all Mitel telephones where the line appears. The next caller of the line makes the next appearance on all Mitel telephones (the next level or tier of appearances) ring. On a single Mitel telephone, the appearances of a multicall line are rung in order from lowest key number to highest key number.
- If a line is a phantom line with a programmed PLID, the line can be called. Telephones with multi-lines can be called whether the line card is plugged in or not.
- If the multicall line appearance is on the prime line, or on a non-prime line and COS option 509, Display Caller ID for non-prime lines is enabled, the CLID is presented automatically on the set display. If the multicall line appearance is on a non-prime line and COS option 509 is disabled, the CLID can be displayed by pressing Superkey and then the ringing line. Note: COS option 509 requires Feature Level 3 in System Option 102.

Programming

From within the Expand Set Subform of CDE Form 09 (Desktop Device Assignments), move the cursor to the TYPE field. Press the MULTI-CALL softkey.

Enable COS Option 262 (Ignore Forward Busy with Free Appearance) in the class of service form of the set to have Call Forward Busy ignored when there are any free multi-line appearances of the set. Note that this COS option applies only to multi-line appearances, not key line appearances or logical line appearances.

Operation

None.

Line Type: DTS Line

Description

A Direct Trunk Select (DTS) Line operates like a key line, but it directly accesses a specified CO trunk. It can be used for incoming and outgoing calls.

- The DTS trunk can appear on 64 Mitel telephones (64 IP sets, or 32 IP and 32 DNIC sets).
- There can be only one appearance of any one direct trunk select line on a Mitel telephone.
- Direction and ring variants can be programmed independently for each appearance. The direction variant allows control of the type of trunk call, incoming or outgoing or both.
- The user can transfer calls on this line to other extensions.
- DTS outgoing calls bypass ARS.
- DTS calls use the SMDR feature if enabled.

For further information, see Direct Trunk Select in this document.

Conditions

See Direct Trunk Select.

Programming

From within the Expand Set Subform of CDE Form 09 (Desktop Device Assignments), move the cursor to the TYPE field. Press the DIR TRK ACC softkey. Note that the trunk must be programmed into a trunk group.

Operation

None.

Line Type: Private Line

Description

Like a DTS Line, a private line accesses a specified dedicated CO trunk directly. However, the user can only transfer established calls on this line to other Mitel telephones that have an appearance of the line, using Privacy Release (see Privacy Enable/Privacy Release).

Conditions

The same conditions apply as for direct trunk select lines.

Programming

From within the Expand Set Subform of CDE Form 09 (Desktop Device Assignments), move the cursor to the TYPE field. Press the PRIVATE TRK softkey.

Operation

None.

Lockout Alarm

Description

The system locks out any set that remains off-hook and not connected to another set or trunk for more than 45 seconds. The lockout alarm feature:

- generates an audible alarm through the console
- activates the alarm relays
- displays the location of the locked out device.

When a set is locked out, if lockout alarm is enabled, all consoles warble with a long-short-long cadence. This cadence overrides other cadences that might be active. The attendant can display the time and date the lockout alarm occurred, the extension number of the device, and a message stating that the device has been off-hook too long.

Note: Lockout alarms can also be detected and reported to display sets. In this environment, the lockout alarm functions as a 911 alarm. See Emergency Calls (911) - Detection and Reporting to Display Sets.

Conditions

The following conditions apply to this feature:

- Stations or sets must have COS Option 227 (Lockout Alarm Applies) enabled for the lockout alarm to report to the attendant console.
- Stations or sets can have COS Option 227 or COS Option 617 (Immediate Off Hook Alarm) enabled for the lockout alarm to report to the display set.
- The attendant console requires COS Option 102 (Attendant Display of System Alarms) for the visual alarm and/or COS Option 108 (Attendant Audible Alarm) for the audible alarm, in order to receive the lockout alarm from the stations or sets.
- Family 1 and Family 4 telephones (except the SUPERSET 4DN and Symbol MiNET Wireless) with COS 616, Alarm Monitor enabled will receive a visual and audible alarm for lockout alarms. The lockout alarm functions as a 911 alarm. See Emergency Calls (911) - Detection and Reporting to Display Sets. These sets will report the lockout alarms if COS Option 617 or COS Option 227 is enabled.
- A lockout alarm occurs 45 seconds after the station or set goes off-hook.
- The alarm totals are updated when a lockout alarm occurs.
- There can be 32 lockout alarms active at any one time; further lockout alarms are only recorded in the maintenance logs.
- The lockout alarm cadence still rings if COS Option 100 (Attendant Bell Off) is enabled.
- The lockout alarm feature operates with COS Option 701 (No Dial Tone) enabled.
- When a console is in restricted service, the lockout alarm cadence rings but the user is not able to read the alarm or turn the warbling off. The restricted access code must be re-dialed before the user can read the alarm.
- When no DTMF receivers are available, the 45 second time period before the lockout alarm occurs is increased by the time required to wait for an available receiver.

- Lockout alarm information is displayed on the console regardless of the state of the maintenance terminal.
- If an attendant console is in Attendant Console Lockout state, you can still access lockout alarm information, but you cannot clear the alarms.
- The console can display a total of 32 alarms (lockout alarms and 911 alarms). For example, the system could report 30 lockout alarms and two 911 alarms. Any additional alarms could not be displayed on the console; however, maintenance logs would be generated for the additional alarms.
- Only one console can access and clear alarm information at a time. If an attendant at a console is reading alarm information all other attendants will see the following message when they press the FUNCTION key:

Lockout or 911 Alarm being displayed by <extension>

where **<extension>** is the extension number of the console that is displaying the alarm information.

- A one minute timeout applies to the alarm display on the console. After one minute, the alarm information is cleared and the console returns to the idle state.
- The console stores the alarms in the order that they were received (first to last). The first time that you read the alarms, they are displayed in this order. However, if you exit alarm display mode and access the alarms again, you will start the list at the last displayed alarm.

Programming

Enable COS Option 227 (Lockout Alarm Applies) in the class of service of the extensions for which the feature applies.

Enable COS Option 102 (Attendant Display of System Alarms) for those consoles which can read and cancel the lockout alarms.

Enable COS Option 108 (Attendant Audible Alarm) to cause the console to audibly ring for a lockout alarm.

Operation

If an extension enters lockout state, the attendant console provides an audible alarm and the display shows

LOCKOUT <Press Function Key for Details> ALARM

- Press FUNCTION

  If an attendant at another console is reading alarm information, you will not be able to access the alarm information. You will see the following:

Lockout or 911 Alarm being displayed by <extension>

- Press SHOW LOCKOUT

  The display shows the extension number that in LOCKOUT.
- Press CLEAR to delete the lockout alarm.

  If other lockout alarms are present, the MORE softkey will be present.
- Press MORE to display the next lockout alarm.
- Press EXIT to return to idle.

Logical Lines

Description

A logical line is a line on a Mitel telephone that is not an appearance of any station or other telephone. Each logical line has its own extension number. A logical line can appear on up to 64 multi-line Mitel telephones. Logical line extension numbers can be used in many places where station or Mitel telephone lines can be programmed (Call Rerouting, etc).

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Logical lines are either key or multicall type lines. Logical lines are not available to single-line sets.
- The Mitel telephone where the first appearance of the logical line resides is used for COS, COR, and tenant checks.
- Logical lines may be viewed by using CDE Form 09 (Desktop Device Assignments) review procedures.
- Logical lines are created implicitly when a vacant extension number is programmed in for a line key.

Programming

Refer to CDE Form 09 (Desktop Device Assignments).

Maintenance

Description

The SX-200 ICP system provides extensive maintenance coverage. All types of peripheral hardware are tested automatically by the system. Maintenance users may also test individual circuits on demand. The system also maintains a fault / event history log (maintenance log).

Conditions

None.

Programming

None

Operation

Refer to the General Maintenance Information section.

Mailbox Key

Description

A Mailbox Key is the line key on a telephone or programmable key module (PKM) that has been programmed with the extension number of a device with an associated mailbox. Subscribers can use Mailbox Keys to receive notification of new messages, and to access their voice mailboxes to listen to messages.

Typically, the administrator will program a number of Mailbox Keys on a single telephone or PKM. This enables multiple subscribers to manage their voice mail from one location. For example, although a number of teachers may share a single telephone in a school staff room, each teacher can have a personal Mailbox Key on the telephone. When a teacher receives a message, their Mailbox Key LED will flash.

A Mailbox Key provides the following functions:

- If the mailbox contains any new messages, the Mailbox Key lamp will flash.
- If the mailbox is empty or contains saved messages, the Mailbox Key lamp will be off.
- A caller can be transferred to a mailbox by pressing the Trans/Conf key followed by the Mailbox Key.
- The Mailbox Key can be used to call voice mail even if the mailbox does not contain any new messages. The MESSAGE key can be used for the same purpose. This functionality is available with embedded voice mail systems.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- The Mailbox Key feature is available for embedded voice mail systems only. It cannot be used with third-party (OEM) voice mail systems, or with centralized voice mail systems.
- Mailbox Key functionality is part of Feature Level 6, a purchasable FOR option.
- Line keys on Mitel telephones and on PKMs associated with a telephone or console support the Mailbox Key feature.
- The extension number assigned to a Mailbox Key must be a telephone programmed in Form 9. It cannot be a logical line.
- Mailbox Keys programmed on unassociated BLFs or HOSTs (blank ASSOC field on Form 9) can only be used to indicate new messages. They cannot be used to call voice mail.
- To call a mailbox using the Mailbox Key, a free internal line is required. A telephone will attempt to use its prime line first and then scan for the next available line. A console will attempt to use either its SRC or DST. If no lines are available, the call cannot be made.
- Pressing a Mailbox Key for an extension that does not have a mailbox associated to it will result in the general mailbox greeting being played.
- Mailbox Keys for the same extension can be programmed on multiple devices, but the mailbox for that extension can only be accessed by one caller at a time.
- A single Mailbox Key can be programmed for telephones that use Phone Twinning. The key is associated with the mailbox for the primary extension.

Programming

In CDE Form 07 (Console Assignment) or Form 09 (Desktop Device Assignment), select a device and press EXPAND SET or EXPAND PKM. In the subform, select a Speed Dial key and press MAILBOX. Enter the extension number in the EXT NUM field.

Note: You must enable COS option 231 or 232 in the COS of the ONS (real or phantom) for successful mailbox key operation.

Operation

To access a mailbox using the Mailbox Key:

- Press the Mailbox Key to be connected to voice mail. When prompted, enter your mailbox passcode.

To connect a caller to a mailbox using the Mailbox Key:

- While engaged in a call, press the Trans/Conf key followed by the Mailbox Key.

Mailbox Lockout

Description

After three failed attempts to log in to a passcode-protected mailbox, the system locks it. Attempts that span consecutive voice mail sessions count as failures. For example, entering an invalid passcode twice, hanging up (which ends the session), and then trying and failing again results in a lockout. So does entering "*" after two invalid attempts, and then trying again by entering the mailbox number and passcode.

After a mailbox is locked out, the system responds to each succeeding log-in attempt with a voice prompt or display. The response varies depending on the following:

- the type of mailbox (extension or administrator)
- whether the passcode entered was valid or invalid

Mailbox Type	Passcode Entered	Response
Extension	Valid	"I'm sorry, your mailbox has been locked for security reasons. Please contact your system administrator." (See Note below)
Administrator	Valid	"I'm sorry, your mailbox has been locked for security reasons."
Extension or Administrator	Invalid	"The passcode is invalid, please try again."

Note: The complete voice prompt is only offered in the English language. In other languages, only the "Please contact your system administrator" portion of the prompt is offered.

Mailbox Lockout affects login access only. Other functionality, for example, the ability to leave a message in a mailbox, remains available even though the mailbox is locked.

Conditions

None

Programming

- In CDE Form 49 (Voice Mail Options), enable Mailbox Lockout.

Note: Mailbox Lockout is a system-wide option. By default, it is disabled.

Operation

To unlock an extension mailbox, update its passcode using the Administrator's Mailbox TUI or CDE Form 50 (Mailboxes).

To unlock the administrator's mailbox:

1. Contact Mitel Product Support to obtain a special code.
2. Access the Administrator's Mailbox TUI from an internal DTMF telephone.

3. Enter the special code when prompted for a password.

Note: You can also reset the administrator's password using the SET PASSWORD softkey on CDE Form 50 (Mailboxes).

Manual Line (Dial 0 Hotline)

Description

When a manual line extension goes offhook it is routed directly to the extension’s dial 0 routing point. The extension can still receive calls.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- The extension does not receive dial tone, but does receive ringback tone.
- If there is no Dial 0 routing point for the current night service then reorder tone is given to the extension.
- Priority Dial 0 applies.
- The call is made as a reroute to the Dial 0 point.
- COS Option 228 (Manual Line) and COS Option 300 (Automatic Callback) are mutually exclusive.
- Call Forwarding programming takes priority over Manual Line programming. A manual line extension that has been programmed with Call Forwarding will call the forwarding destination when the handset is lifted instead of the dial 0 point.

Programming

Enable COS Option 228 (Manual Line) for the extension.

Enter appropriate answer point (e.g., an Attendant LDN) in Station Dial 0 Routing in CDE Form 19 (Call Rerouting Table) for the extension’s tenant.

Operation

To originate a call, lift the handset; the dial 0 point rings.

Messaging - Advisory

Description

This feature allows users with a Mitel display telephone to display a short message on display sets and consoles that call their set. On all sets except the 4150 and 53xx, the message replaces the time and date display on the sets where it is activated; see Attendant Setup Of Set Advisory Messages. Family 1 telephone users can also program messages to be displayed. The system provides the following system-wide messages:

Message Number	Default Message
01	IN A MEETING

Message Number	Default Message
02	OUT OF TOWN
03	ON VACATION
04	OUT ON A CALL
05	OUT TO LUNCH
06	GONE FOR DAY
07	GONE HOME
08	IN TOMORROW
09 through 15	(BLANK)

Optionally, set users can be permitted to change these messages.

Note: Any change to a message is applied system-wide; the system has only one set of 15 messages.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- This feature applies only to Mitel display telephones.
- A message currently in use cannot be altered, except for **Family 5** phones where the Force Off softkey can be used to replace the current message. (The softkey appears when you press the Create Message button when the current message is activated and in use by another user.)
- The message is not displayed by the telephone that set it when it is in Night Service; however, another telephone (or console) calling that telephone receives the message on its display.
- This feature is not supported with embedded voice mail.

Programming

To permit set users to create or change messages, enable COS Option 605 (Telephone - Message Program) in the set's COS.

Operation

Operation varies depending upon the type of set as described below.

Family 3, and Family 4 telephones (except the Symbol MiNET Wireless telephone):

Note: On a Family 3 telephone, press the dial pad * key instead of the YES softkey, and press the dial pad # key instead of the NO softkey.

To activate an advisory message:

- Press SUPERKEY.
- Press the NO softkey until ADVISORY? appears in the display.
- Press the YES softkey. Message #1, IN A MEETING, appears in the display.

- Press the TURNON softkey to select the IN A MEETING message
  or,
  Press the NEXT softkey until the desired message is displayed. You can also display a specific advisory message by dialing its 2-digit number (see preceding table).
- When the desired message is displayed, press the TURNON softkey. The selected message is displayed on your set.

To deactivate an advisory message:

- Press SUPERKEY.
- Press the NO softkey until ADVISORY? appears in the display.
- Press the YES softkey. The message that is currently programmed at your set appears in the display.
- Press the TURNOFF softkey. The message is turned off and your set displays the date and time.

Family 1 Telephones:

To activate an advisory message:

- Press the MESSAGING softkey.
- Press the ADVISORY softkey.
- If displayed message is inappropriate, press the NEXT MSG softkey repeatedly to cycle through the repertoire of messages. If the message number is known, press the SHOW MSG NO. softkey, dial the message number (01 through 15) and press the ENTER softkey.
- Press the TURN MSG ON softkey. The selected message is now displayed on the LCD. Any Mitel display telephone, or console calling the set receives the message on its display.

To deactivate an advisory message:

- Press the MESSAGING softkey.
- Press the ADVISORY softkey.
- Press the TURN MSG OFF softkey.

To program an advisory message:

- Press the MESSAGING softkey.
- Press the ADVISORY softkey.
- Press the SHOW MSG NO. softkey or the Next softkey. Dial an unused message number, unless an existing message is to be changed.
- Press the CREATE MSG softkey. Dial in the message as follows:

Letters are displayed on the LCD as they are dialed on the telephone keypad. The first press of any of these keys displays the first letter that appears on its key cap. The second press of the same key changes the display to the second letter and so on. When all the letters associated with a key have been displayed, the number is displayed. Further presses cycle through the letters again. When the desired letter is displayed, enter it by pressing the → softkey or by entering the next letter, if it is on a different key. (The → key is also used to enter spaces). Follow the same procedure to find and enter the other letters in the name.

Telephone keypad key caps 1, *, 0, and # are not marked with all the characters they can enter. They are as follows:

Key	Characters
1	! ? % 1
*	' - *
0	@ & $ 0
#	., / #

When the message is complete, press the SAVE softkey. The message is now saved.

Family 5 Telephones:

To activate an advisory message:

5330 Phones:

- Press the key you have programmed as Superkey.
- Press the No softkey until ADVISORY appears and then press Yes.
- Press the Next softkey repeatedly to cycle through the repertoire of messages.
- Press the TurnOn softkey. The selected message is now displayed on the LCD. Any Mitel display telephone, or console calling the set receives the message on its display.

5340 Phones:

- Press the Messaging softkey.
- Press the Advisory softkey.
- Press Next Msg until the desired message appears.
- Press Turn Msg On.

To deactivate an advisory message:

5330 Phones:

- Press the key you have programmed as Superkey.
- Press the No softkey until ADVISORY appears and then press Yes.
- Press the TurnOff softkey.

5340 Phones:

- Press the Messaging softkey.
- Press the Advisory softkey.The set displays the current advisory message.
- Press the Turn Msg Off softkey.

To program an advisory message:

- Press the MESSAGING softkey.
- Press the ADVISORY softkey.
- Press the SHOW MSG NO. softkey or the Next softkey. Dial an unused message number, unless an existing message is to be changed.
- Press the CREATE MSG softkey. Dial in the message as follows:

Letters are displayed on the LCD as they are dialed on the telephone keypad. The first press of any of these keys displays the first letter that appears on its key cap. The second press of the same key changes the display to the second letter and so on. When all the letters associated with a key have been displayed, the number is displayed. Further presses cycle through the letters again. When the desired letter is displayed, enter it by pressing the → softkey or by entering the next letter, if it is on a different key. (The → key is also used to enter spaces). Follow the same procedure to find and enter the other letters in the name.

Messaging - Call Me Back

Description

A set user calling a busy or unanswered set can leave a message for the party to return the call. The Message Waiting indication can be:

- A flashing lamp on the set, flashing at 0.5 seconds on, 3.5 seconds off (if equipped),
- Ringing with a distinctive ringing pattern at the set

- A discriminating (stutter) dial tone when the station user goes offhook.

The Message Waiting indication continues until the set user reads the message. Messages can be read at any time (i.e., when the set is idle or during a call).

On Mitel display telephones, the display provides the time of the call, and the caller's extension number and name (if programmed).

Optionally, the system can be programmed to record each occurrence of Message Waiting on the system printer. See Hotel/Motel.

Messages can be left at industry-standard and Mitel telephones.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Use the Message - Call Me Back feature if you know that the person you are trying to contact is out of the office. Use the Callback feature if you know that the person you are trying to contact is somewhere in the office.
- Only one message exists between any two parties in the system at anytime. Leaving a message from telephone A to telephone B cancels the message left from telephone B to telephone A.
- The message waiting indication is not active while the set is in use.
- ONS/CLASS lines do not support high-voltage message-waiting lamps.
- When sending a message using a feature access code, if the destination set cannot receive messages, reorder tone is returned.
- When sending a message using headset mode or handsfree using the send message feature, the set is in new call mode waiting for digits to be dialed or for the call to be canceled or released. After the interdigit timer times out, reorder tone is returned and after timeout, the set is locked out.
- Messages cannot be left for members of a recording group or an Automated Attendant recording group.
- A message is canceled automatically if the sender and receiver have a telephone conversation before the message is read or the receiver leaves a message for the sender. Note that this does not apply to messages from the console. These must be canceled at the Console.
- The system cancels the message automatically after 24 hours if System Option 7 (Cancel 24-hour Message Waiting) is enabled; otherwise, it is not canceled by the system.
- 750 messages can be active at a time.
- If COS Option 231 - Message Waiting Setup - Bell is enabled in the industry-standard or Mitel telephone's COS:
  - Message Waiting indication is three cycles of 3.5 impulse per second (ips) ringing. This is in addition to the visual indicators on Mitel telephones.
  - The set receives ringing:
    1) 10 seconds after the message is sent if the set was idle, or in do not disturb, or idle in auto answer mode, at the moment when the message was sent,
    2) each time it returns to idle (10 seconds after),
    3) and every 20 minutes while the set is idle until the message is read, canceled or acted upon.

  Note: Message Waiting - Bell is <u>not</u> supported on ASUs.

- Only stations (ONS and OPS) support the discriminating (stutter) dial tone when a message is waiting. This dial tone alerts the user of a message waiting when the user goes offhook. The station must have the COS 219, Discriminating Dial Tone enabled.
- An analog set connected to a SIM2 does not support Messaging Call Me Back. These sets should have COS Option 231, Message Waiting Setup - Bell disabled.
- The Messaging – Call Me Back feature will function across analog networking on a remote extension only if the extension leaving the call me back message (the local extension) is assigned to a hunt group in CDE Form 17.

Programming

To allow industry-standard telephones in a COS to receive messages, enable COS Option 231, Message Waiting Setup - Bell or COS Option 232, Message Waiting Setup - Lamp if the industry-standard telephone is equipped with a message waiting lamp.

Note: These two COS Options are mutually exclusive. Neither of them is required for messaging on Mitel telephones.

- An analog telephone connected to a SIM2 does not support this feature and should have COS Option 231 disabled.

To allow ONS and OPS stations to receive a discriminating (stutter) dial tone when a message is waiting, enable COS 219, Discriminating Dial Tone.

To allow users of industry-standard telephones to send messages, enable COS Option 259 (Message Sending) in their COS. Enable COS Option 259 (Message Sending) in the COS of Mitel telephones to allow users to send messages using the MESSAGE key.

Note: COS Option 259 enables and disables both methods of sending messages.

Industry-standard telephones need access codes to send or answer messages. In CDE Form 02 (Feature Access Codes), assign access codes to Feature 41 (Send Message) and Feature 42 (Call Message Sender of Oldest Message).

Enable COS Option 231 (Message Waiting Setup - Bell) to allow SUPERSET 3DN, SUPERSET 4DN, SUPERSET 410, SUPERSET 420, SUPERSET 430, and SUPERSET 4000 series telephones to warble when a message is sent.

In CDE form 04 (System Options/System Timers), enable System Option 07 (Cancel 24-hour Message Waiting) if messages are to be canceled by the system.

For Mitel telephones with a display, enable COS Option 605, Telephone - Message Program.

Operation

Industry-standard Telephones:

To send a message signal to a busy or unanswered set:

- Dial the Send Message access code.
- Dial 1 to activate the message waiting signal.
- Dial the extension number.
- Dial tone is returned. Re-order tone is returned if the other telephone cannot receive messages.

To cancel a message signal previously sent:

- Dial the Send Message access code.
- Dial 2 to cancel the message waiting signal.
- Dial the extension number.
- Dial tone is returned.

To answer a message:

- Go off–hook.
- Dial the Call Message Sender of Oldest Message access code.
- The message sender (voice mail auto-attendant or set) is called. Re-order tone is returned if there are no messages waiting.

To clear the oldest message at your set:

- Go off-hook.
- Dial the Send Message access code.
- Dial 3 to clear the message.
- Dial tone is returned.

Mitel 5020, 5220, 5224 and 5324 IP Phones, and Family 2, SUPERSET 4015, SUPERSET 410, and SUPERSET 3DN Telephones:

To send a message waiting signal:

- While receiving busy tone or ringback, press the MESSAGE key if the Message LED is on steady.
- Dial tone is returned, or,
- Follow the procedure given for industry-standard telephones. This is convenient when leaving a message without calling the message recipient.

The message LED flashes when an idle telephone has a message waiting for it.

To answer a message waiting signal:

- Lift the handset and press the MESSAGE key (Family 2 telephones)
  or,
  Press the MESSAGE key.
- The set automatically places a callback to the device (voice mail auto-attendant or set) that left the oldest message.

To cancel a message waiting signal at your set:

- Press the MESSAGE key. You automatically ring the extension of the person that left the message signal.
- If the extension is busy or there is no answer, press the MESSAGE key again to cancel the message waiting signal at your set, and send a message signal back to the station that sent the original signal.

Mitel 5020, 5220, 5224, 5324, 5330, and 5340 IP Phones, and SUPERSET 4025, SUPERSET 4125, and SUPERSET 420 Telephones:

To send a message waiting signal:

- Press the MESSAGE key if you receive busy or no answer when you call a station (your message lamp must be on steady). MESSAGE SENT appears briefly in the display and the call is terminated, or,
  Follow the procedure given for industry-standard telephones. This is convenient when leaving a message without calling the message recipient.

The message LED flashes whenever the telephone is idle and has a message waiting for it.

To respond immediately to a message waiting signal at your set:

- Go off-hook and press the MESSAGE key.
- The set automatically places a callback to the device (voice mail auto-attendant or set) that left the oldest message.

To view information about the message before responding:

- Remain onhook and press the MESSAGE key (your Message Lamp is flashing while the set is idle).

  The extension number of the caller, and the time that the message signal was sent appears in the display.
  or
  The name of the caller appears in the display. Press the MORE softkey to display the extension number of the caller, and the time that the message signal was sent.
- Press the CALL softkey to ring the extension (voice mail or set). This softkey appears only when the telephone can dial the call (e.g., idle or listening to dial tone).

  or,

  Press the ERASE softkey to delete the message signal.
- Press SUPERKEY to return to the date and time display.
  1. This softkey will not appear on telephones that have no free internal lines.

SUPERSET 4150 and SUPERSET 430 Telephones:

To send a message:

- While receiving busy tone or ringback, press the LEAVE A MSG softkey. This softkey appears only if the other telephone is able to receive messages.
- The top line of the display briefly shows a confirmation and the call is terminated, or,
- Follow the procedure given for industry-standard telephones.

The message LED flashes whenever the telephone is idle and has a message waiting for it. Also, the second line of the LCD shows < MSG > on the right, if there are any messages waiting.

To respond immediately to a message waiting signal at your set while your set is idle or while you are receiving a dial tone

- Press the MESSAGING softkey or MESSAGE key.

Note: The MESSAGING softkey or MESSAGE key is not available while you have a party on consultation hold. The Messaging softkey will appear on the lower middle softkey position when the user goes off hook or presses the Speaker/Prime line key from idle.

- The set automatically places a callback to the device (voice mail auto-attendant or set) that left the oldest message.

To read the messages before responding, or if you are engaged in a call:

- Press the SUPERKEY.
- Press the MESSAGING softkey.
- Press the READ MSG softkey. The LCD second line shows the extension number and the time of the call.
- Press the CALL softkey (this softkey will not appear on telephones which have no free internal lines) to call the device (voice mail auto-attendant or set) that left the message. This softkey appears only when you can dial the call or,
- Press the ERASE softkey to cancel the message.

If there are more messages, the NEXT MSG softkey appears. To read:

- Press the NEXT MSG softkey; the next message is displayed. Follow the procedure above for each message.
- When there are no more messages, press the SUPERKEY to exit.

To view messages sent:

SUPERSET 4150, and SUPERSET 430 telephones also allow you to view the messages that you have sent. To view messages that you have sent:

- Press MESSAGING softkey.
- Press MSGS I SENT softkey.
  - For each message, the set displays the name, extension number, time and date that the message was sent.
  - A softkey also appears that allows you to cancel the message.

SUPERSET 4DN Telephones:

To send a message:

- While receiving busy tone or ringback, press the LEAVE A MSG softkey. This softkey appears only if the other telephone is able to receive messages.
- The top line of the display briefly shows a confirmation and the call is terminated, or,
- Follow the procedure given for industry-standard telephones.

The message LED flashes whenever the telephone is idle and has a message waiting for it.

To read the messages:

- If engaged in a call, press the SUPERKEY.
- Press the MESSAGING softkey.
- Press the READ MSG softkey. The LCD second line shows the extension number and the time of the call.
- Press the CALL softkey to call the party that left the message. This softkey appears only when the telephone can dial the call (e.g., idle or listening to dial tone), or,
- Press the ERASE softkey to cancel the message.

If there are more messages, the NEXT MSG softkey appears. To read:

- Press the NEXT MSG softkey; the next message is displayed. Follow the procedure above for each message.
- When there are no more messages, press the SUPERKEY to exit.

To view messages sent:

SUPERSET 4DN telephones also allow you to view the messages that you have sent. To view messages that you have sent:

- Press MESSAGING softkey.
- Press MSGS I SENT softkey.
  - For each message, the set displays the name, extension number, time and date that the message was sent.
  - A softkey also appears that allows you to cancel the message.

Symbol MiNET Wireless Phone:

To leave a callback message for another extension user:

- Select CLBK, and then press the SEND key .

Meter Pulse Collection

Description

Meter pulses are often used to calculate the cost of outgoing trunk calls thus allowing the call to be charged back to the originator. The system can be set up to detect and collect certain types of meter pulses sent to a trunk circuit during outgoing calls; these are then recorded in the trunk's SMDR reports. Types of meter pulses which can be detected by the system without additional hardware include:

- Tip-Ring reversals
- XT lead signaling (Analog CO Trunk)
- M&MM lead signaling (Digital LS/GS Trunk).

Refer to the Installation Information section, for location of correct leads and proper interface to the sending equipment.

Other types of meter pulses common in the telephone industry include 50Hz, 12 kHz, and 16 kHz type pulses. Detection of these types requires the addition of an external interface which converts these pulses to a ground signal which is then applied to the XT Lead for the analog CO Trunks, or to the M or MM lead for the Digital LS/GS trunks (for Digital LS/GS trunks, -48 V dc must be applied to the other lead so that when the ground is applied to the M or MM lead, current flows through the circuit and gets detected as a pulse).

This feature is associated with the Message Registration feature. See Hotel/Motel Features for additional information.

Conditions

The following conditions apply to this feature:

- The system can only detect and collect the types of meter pulses identified above.
- The trunk must provide answer supervision. This is counted as the first meter pulse.
- Meter pulses are not recorded for ACD agents if System Option 44 (ACD Reports) is enabled.
- The system can record a maximum of 65535 pulses.
- Pulses are always recorded regardless of what is happening to the trunk (hold, talking etc.).

Programming

In the COS for the trunk, enable COS Option 247 (SMDR - Record Meter Pulses) in CDE Form 3 (COS Define).

Enable System Option 23 (Message Register Count Additional Supervisions) in CDE Form 4 (System Options/System Timers).

Enable Option (Far-end Gives Answer Supervision) in CDE Form 13 (Trunk Circuit Descriptors).

Operation

As meter pulses are received, they are collected by the system and reported in the trunk's SMDR record.

Mitel 5010 IP Phone

Description

The Mitel 5010 IP Phone (see figure) is a multi-line telephone that connects to a 10/100BaseT Ethernet network. Features of the telephone include

- Twenty-character alpha-numeric liquid crystal display (LCD)
- Seven line keys, each with a built-in line status indicator
- Six fixed-function keys: SuperKey, Cancel, Hold, Redial, Transfer/Conference, and Message
- Automatic selection of ringing line
- Key selection of non-prime line
- Handset and ringer volume controls (Up Arrow and Down Arrow)
- Ringer pitch control

- Message Waiting lamp
- Dual Ethernet port to provide connectivity to the LAN for both your telephone and computer
- Dedicated Headset port.

Conditions

- Support for 5010 IP Phones is available only with the SX-200 ICP MX migration package, not with new installations.
- Each IP Phone requires a license. Licenses are purchasable and enabled in Form 04, System Options/System Timers.
- System Option 131 in Form 04 and COS Option 280 in Form 03 are required to activate the phone's second Ethernet port.

Programming

Program 5010 IP Phones in CDE Form 09 (Desktop Device Assignments); enable the required options in Form 04 (System Options/System Timers), and select their COS Options in CDE Form 03 (COS Define).

Operation

Refer to the specific feature.

Mitel 5020 IP Phone

Description

The Mitel 5020 IP Phone (see figure) is a multi-line telephone that connects to a 10/100BaseT Ethernet network. Features of the telephone include

- Twenty-character alpha-numeric LCD with contrast control
- Three softkeys for feature access
- Fourteen line keys, each with a built-in line status indicator
- Eight fixed-function keys: SuperKey, Cancel, Hold, Redial, Transfer/Conference, Message, Microphone, and Speaker
- Automatic selection of prime line
- Key selection of non-prime line
- Handsfree operation (half-duplex)
- Handset, speaker, and ringer volume controls (Up Arrow and Down Arrow)
- Ringer pitch control
- Message waiting lamp
- Dual Ethernet port to provide connectivity to the LAN for both your telephone and computer
- Dedicated Headset port.

Conditions

- Support for 5020 IP Phones is available only with the SX-200 ICP MX migration package, not with new installations.
- Each IP Phone requires a license. Licenses are purchasable and enabled in Form 04, System Options/System Timers.
- System Option 131 in Form 04 and COS Option 280 in Form 03 is required to activate the phone's second Ethernet port.

Programming

Program 5020 IP Phones in CDE Form 09 (Desktop Device Assignments); enable the required options in Form 04 (System Options/System Timers), and select their COS Options in CDE Form 03 (COS Define).

Operation

Refer to the specific feature.

Mitel 5201 IP Phone

Description

The Mitel 5201 IP Phone (see figure) is a low-cost, single port entry-level IP telephone that connects to a 10/100BaseT Ethernet network. Features of the telephone include:

- Three fixed-function keys: Hold, Message, and Transfer/Conference
- Handset and Ringer Volume Control
- Message Waiting Lamp
- Wall-mounting

Conditions

Each IP Phone requires a license. Licenses are purchasable and enabled in Form 04, System Options/System Timers.

Programming

Program 5201 IP Phones in CDE Form 09 (Desktop Device Assignments); and select their COS options in Form 03 (COS Define).

Operation

Refer to the specific feature.

Mitel 5207 IP Phone

Description

The Mitel 5207 IP Phone (see figure) is a multi-line telephone that connects to a 10/100BaseT Ethernet network. Features of the telephone include

- Alpha-numeric liquid crystal display (LCD) with contrast control
- Fourteen line keys, each with a built-in line status indicator
- Ten fixed-function keys: SuperKey, Cancel, Hold, Redial, Transfer/Conference, Message, Microphone, Speaker, Up Arrow, and Down Arrow
- Automatic selection of prime line
- Key selection of non-prime line
- Hands-free speakerphone operation (half duplex)
- Handset, speaker, and ringer volume controls
- Ringer pitch control
- Message waiting lamp.

Conditions

- Each IP Phone requires a license. Licenses are purchasable and enabled in Form 04, System Options/System Timers.
- The Mitel 5207 IP Phone handles New Call Ring alerts in a unique manner. When a 5207 is busy and a new call attempts to ring the set, the Message Waiting Lamp will flash and a short ring burst will be heard through the headset or handset (not the speaker). The alert can be limited with COS Option 611 (Telephone - Limited New Call Ring) so that only the first ringing line provides the short ring burst.

Programming

Program 5207 IP Phones in CDE Form 09 (Desktop Device Assignments) and select their COS Options in CDE Form 03 (COS Define).

Operation

Refer to the specific feature.

Mitel 5212 IP Phone (Dual Mode)

Description

The dual mode Mitel 5212 IP Phone (see figure) is a +full-feature, dual port telephone that provides voice communication over an IP network.

With Teleworker Solution Release 3.0 and higher, the dual mode 5212 IP Phone can be operated remotely in MiNET mode (i.e. ICP-supported). For information on how to configure the 5212 IP Phone (Dual Mode) to remote mode after installation, refer to the Remote IP Phones Configuration Guide for the Teleworker Solution available at www.mitel.com (see Supporting Documentation).

Features of the telephone include

- Alpha-numeric LCD
- Twelve line keys, each with a built-in line status indicator
- Eight fixed-function keys: SuperKey, Cancel, Hold, Redial, Transfer/Conference, Message, Speaker, Mute
- Automatic selection of prime line or ringing line (the bottom line key)
- Key selection of non-prime line
- Ringer pitch control
- Hands-free speakerphone operation (half duplex)
- Handset, speaker, and ringer volume controls (Up Arrow and Down Arrow)
- Message Waiting lamp
- Ring Indicator lamp
- Headset port
- Dual Ethernet port to provide connectivity to the LAN for both your telephone and computer
- Adjustable tilt mechanism
- Powered from an Ethernet connection compliant with 802.3af.

Conditions

- Each IP Phone requires a license. Licenses are purchasable and enabled in Form 04, System Options/System Timers.

- System Option 131 in Form 04 and COS Option 280 in Form 03 are required to activate the phone's second Ethernet port.

Programming

SX-200 ICP

Program the phones in CDE Form 09 (Desktop Device Assignments) as device type 5212d; enable the required options in Form 04 (System Options/System Timers), and select their COS Options in CDE Form 03 (COS Define).

Telephones

The dual mode 5212 IP Phone defaults to MiNET mode when first installed and becomes operational when registered with the SX-200 ICP. For instructions on registering phones, see Register the IP Phones.

Operation

Refer to the specific feature.

Mitel 5215 IP Phone

Description

The Mitel 5215 IP Phone (see figure) is a multi-line telephone that connects to a 10/100BaseT Ethernet network. Features of the telephone include

- Alpha-numeric LCD
- Seven line keys, each with a built-in line status indicator
- Eight fixed-function keys: SuperKey, Cancel, Hold, Redial, Transfer/Conference, Message, Microphone
- Automatic selection of prime line or ringing line (the bottom left line key)
- Key selection of non-prime line
- Handset and ringer volume control
- Ringer pitch control
- Hands-free speakerphone operation (half duplex)
- Message Waiting lamp
- Headset port
- Dual Ethernet port to provide connectivity to the LAN for both your telephone and computer
- Adjustable tilt mechanism.

Conditions

- Each IP Phone requires a license. Licenses are purchasable and enabled in Form 04, System Options/System Timers.
- System Option 131 in Form 04 and COS Option 280 in Form 03 is required to activate the phone's second Ethernet port.

Programming

Program 5215 IP Phones in CDE Form 09 (Desktop Device Assignments); enable the required options in Form 04 (System Options/System Timers), and select their COS Options in CDE Form 03 (COS Define).

Operation

Refer to the specific feature.

Mitel 5220 IP Phone

Description

The Mitel 5020 IP Phone (see figure) is a multi-line telephone that connects to a 10/100BaseT Ethernet network. Features of the telephone include

- Alpha-numeric LCD with contrast control
- Three softkeys for feature access
- Fourteen line keys, each with a built-in line status indicator
- Ten fixed-function keys: SuperKey, Cancel, Hold, Redial, Transfer/Conference, Message, Microphone, Speaker, Up Arrow, and Down Arrow
- Automatic selection of prime line
- Key selection of non-prime line
- Handsfree speakerphone operation (half duplex)
- Handset, speaker, and ringer volume controls
- Ringer pitch control
- Message Waiting lamp
- Dedicated Headset port
- Dual Ethernet port to provide connectivity to the LAN for both your telephone and computer
- Adjustable tilt mechanism.

Conditions

- Each IP Phone requires a license. Licenses are purchasable and enabled in Form 04, System Options/System Timers.
- System Option 131 in Form 04 and COS Option 280 in Form 03 is required to activate the phone's second Ethernet port.

Programming

Program 5220 IP Phones in CDE Form 09 (Desktop Device Assignments); enable the required options in Form 04 (System Options/System Timers), and select their COS Options in CDE Form 03 (COS Define).

Operation

Refer to the specific feature.

Mitel 5215 and 5220 IP Phones (Dual Mode)

Description

The dual mode Mitel 5215 and 5220 IP Phones can operate either as a standalone device connected to a SIP (session initiation protocol) service provider or as part of a Mitel ICP system using the MiNet protocol.

Except for dual mode operation, the phones are identical to the 5215 and 5220 IP Phones described above.

With Teleworker Solution Release 3.0, the dual mode 5215 and 5220 IP Phones can be operated remotely in MiNET mode (i.e. ICP-supported). For information on how to configure the 5215 and 5220 IP Phones (Dual Mode) to remote mode after installation, refer to the Remote IP Phones Configuration Guide for the Teleworker Solution available at www.mitel.com (see Supporting Documentation).

Conditions

- Each IP Phone requires a license. Licenses are purchasable and enabled in Form 04, System Options/System Timers.
- System Option 131 in Form 04 and COS Option 280 in Form 03 are required to activate the phone's second Ethernet port.

Programming

SX-200 ICP

Program the phones in CDE Form 09 (Desktop Device Assignments) as device types 5215d and 5220d; enable the required options in Form 04 (System Options/System Timers), and select their COS Options in CDE Form 03 (COS Define).

Telephones

The dual mode 5215/5220 IP Phones default to MiNET mode when first installed and become operational when registered with the SX-200 ICP. For instructions on registering phones, see Register the IP Phones.

To program a 5215/5220 IP Phone (Dual Mode) for SIP mode operation:

1. Press both Arrow keys simultaneously while powering up the phone. NETWORK SETTINGS? appears.
2. Press Yes. MODIFY SETTINGS? appears.
3. Press Yes, and then press NoChange until NETWORK TYPE: <current type> appears.
4. To change the network type, press Change.
5. Follow the prompts to select and accept the network type (LAN, xDSL (forPPPoE) or Cable), and to store the changes and reboot the phone.

   Note: For xDSL, an Internet Service Provider user name and password are required.
6. When the date and time appear, press and hold * and 7 simultaneously while powering up the phone.When the date and time reappear, the phone is in SIP mode.

To return the phone to MiNET mode:

- Press and hold * and 6 simultaneously while powering up the phone. When the date and time appear, the phone is in MiNET mode.

For additional SIP configuration information (e.g. setting the phone mode, performing upgrades, restoring factory defaults), refer to the 5215/5220 IP Phone SIP User Guide available at www.mitel.com (see Supporting Documentation).

Operation

Refer to the 5215/5220 IP Phone SIP User Guide available on the Mitel Customer Documentation website at http://edocs.mitel.com.

Mitel 5224 IP Phone (Dual Mode)

Description

The dual mode Mitel 5224 IP Phone (see figure) can operate either as a standalone device connected to a SIP (session initiation protocol) service provider or as part of a Mitel ICP system using the MiNet protocol.

With Teleworker Solution Release 3.0, the dual mode 5224 IP Phone can be operated remotely in MiNET mode (i.e. ICP-supported). For information on how to configure the 5224 IP Phone (Dual Mode) to remote mode after installation, refer to the Remote IP Phones Configuration Guide for the Teleworker Solution available at www.mitel.com (see Supporting Documentation).

Features of the telephone include

- Alpha-numeric LCD
- Three softkeys for feature access
- Twenty-four line keys, each with a built-in line status indicator
- Eight fixed-function keys: SuperKey, Cancel, Hold, Redial, Transfer/Conference, Message, Speaker, Mute
- Automatic selection of prime line or ringing line (the bottom right line key)
- Key selection of non-prime line
- Ringer pitch control
- Hands-free speakerphone operation (full duplex)
- Handset, speaker, and ringer volume controls (Up Arrow and Down Arrow)
- Message Waiting lamp
- Ring Indicator lamp
- Headset port
- Dual Ethernet port to provide connectivity to the LAN for both your telephone and computer
- Adjustable tilt mechanism
- Powered from an adapter or power brick, or an Ethernet connection compliant with 802.3af
- Supports a single Line Interface Module, 5310 IP Conference Module, or 5412 PKM, or up to two 5448 PKMs.

Note: The set provides power to all modules except the 5310 Conference Module, which requires an external source.

Conditions

- Each IP Phone requires a license. Licenses are purchasable and enabled in Form 04, System Options/System Timers.
- System Option 131 in Form 04 and COS Option 280 in Form 03 are required to activate the phone's second Ethernet port.

Programming

SX-200 ICP

Program the phones in CDE Form 09 (Desktop Device Assignments) as device type 5224d; enable the required options in Form 04 (System Options/System Timers), and select their COS Options in CDE Form 03 (COS Define).

Telephones

The dual mode 5224 IP Phone defaults to MiNET mode when first installed and becomes operational when registered with the SX-200 ICP. For instructions on registering phones, see Register the IP Phones.

To program a 5224 IP Phone (Dual Mode) for SIP mode operation:

1. Press both Arrow keys simultaneously while powering up the phone. NETWORK SETTINGS? appears.
2. Press Yes. MODIFY SETTINGS? appears.
3. Press Yes, and then press NoChange until NETWORK TYPE: <current type> appears.
4. To change the network type, press Change.
5. Follow the prompts to select and accept the network type (LAN, xDSL (forPPPoE) or Cable), and to store the changes and reboot the phone.

   Note: For xDSL, an Internet Service Provider user name and password are required.
6. When the date and time appear, press and hold * and 7 simultaneously while powering up the phone.When the date and time reappear, the phone is in SIP mode.

To return the phone to MiNET mode:

- Press and hold * and 6 simultaneously while powering up the phone. When the date and time appear, the phone is in MiNET mode.

For additional SIP configuration information (e.g. setting the phone mode, performing upgrades, restoring factory defaults), refer to the 5224 IP Phone SIP User Guide available at www.mitel.com (see Supporting Documentation).

Operation

Refer to the specific feature.

Mitel 5304 IP Phone (Dual Mode)

Description

The Mitel 5304 IP Phone (see figure) is a two-line, dual port telephone that provides voice communication over an IP network.

Features of the telephone include

- 2-line x 20-character white, backlit, graphics display with contrast control and auto-dimming
- Eight programmable multi-function keys (for speed dialing, line appearances, feature access)
- Two Lines with LED indicators
- Dual port (10 / 100 Mb Switched Ethernet)
- Calling Line ID Support
- Volume / Contrast Up/Down keys
- Six fixed-function keys (programmable through CDE): Superkey, Cancel, Hold, Redial, Transfer/Conference, and Message
- Speaker for ringing/paging
- Speed calling
- Call forward
- Call hold (Place / Retrieve)
- Call transfer
- Conference call setup
- Page Send / Receive

- Voice mail access
- Last number redial
- Hearing Aid Compatible (HAC) handset
- Wall mountable (optional)
- Multiple powering options (802.3af compliant)
- Designed for power conservation: reduces power consumption for overall energy savings
- Small footprint (4" x 7.5" or 10cm x 20cm)

Conditions

- Each IP Phone requires a license. Licenses are purchasable and enabled in Form 04, System Options/System Timers.
- System Option 131 in Form 04 and COS Option 280 in Form 03 are required to activate the phone's second Ethernet port.
- **NOTE**: If you are upgrading to Release 4.0 or later and plan to use the Mitel 53xx IP phones, you need a 512MB internal flash or a hard drive. If you do not plan to use the 53xx phones, you can still load the software using a 256 MB flash. For more information, refer to Upgrading to the Latest Release.

Programming

Program the phones in CDE Form 09 (Desktop Device Assignments) as device type 5304; enable the required options in Form 04 (System Options/System Timers), and select their COS Options in CDE Form 03 (COS Define).

Operation

Refer to the *5304 User Guide for the SX-200 ICP* available at Mitel OnLine.

Mitel 5312 IP Phone (Dual Mode)

Description

The dual mode Mitel 5312 IP Phone (see figure) is a full-feature, dual port telephone that provides voice communication over an IP network.

With Teleworker Solution Release 4.1 and higher, the dual mode 5312 IP Phone can be operated remotely in MiNET mode (i.e. ICP-supported). For information on how to configure the 5312 IP Phone (Dual Mode) to remote mode after installation, refer to the Remote IP Phones Configuration Guide for the Teleworker Solution available at www.mitel.com (see Supporting Documentation).

Features of the telephone include

- Alpha-numeric LCD
- Twelve programmable keys, each with a built-in line status indicator
- Eight fixed-function keys: SuperKey, Cancel, Hold, Redial, Transfer/Conference, Message, Speaker, Mute
- Automatic selection of prime line or ringing line (the bottom line key)
- Key selection of non-prime line
- Ringer pitch control
- Handsfree speakerphone operation (full duplex)
- Handset, speaker, and ringer volume controls (Up Arrow and Down Arrow)

- Message Waiting lamp
- Ring Indicator lamp
- Headset port
- Dual Ethernet port to provide connectivity to the LAN for both your telephone and computer
- Adjustable tilt mechanism
- Powered from an Ethernet connection compliant with 802.3af.

Conditions

- Each IP Phone requires a license. Licenses are purchasable and enabled in Form 04, System Options/System Timers.
- System Option 131 in Form 04 and COS Option 280 in Form 03 are required to activate the phone's second Ethernet port.
- **NOTE**: If you are upgrading to Release 4.0 or later and plan to use the Mitel 53xx IP phones, you need a 512MB internal flash or a hard drive. If you do not plan to use the 53xx phones, you can still load the software using a 256 MB flash. For more information, refer to Upgrading to the Latest Release.

Programming

Program the phones in CDE Form 09 (Desktop Device Assignments) as device type 5312; enable the required options in Form 04 (System Options/System Timers), and select their COS Options in CDE Form 03 (COS Define).

Operation

Refer to the *5312/5324 User Guide for the SX-200 ICP* available at Mitel OnLine.

Mitel 5324 IP Phone (Dual Mode)

Description

The dual mode Mitel 5324 IP Phone (see figure) is a full-feature, dual port telephone that provides voice communication over an IP network.

With Teleworker Solution Release 4.1 and higher, the dual mode 5324 IP Phone can be operated remotely in MiNET mode (i.e. ICP-supported). For information on how to configure the 5324 IP Phone (Dual Mode) to remote mode after installation, refer to the Remote IP Phones Configuration Guide for the Teleworker Solution available at www.mitel.com (see Supporting Documentation).

Features of the telephone include

- Alpha-numeric LCD
- Three softkeys for feature access
- Twenty-four programmable keys, each with a built-in line status indicator
- Eight fixed-function keys: SuperKey, Cancel, Hold, Redial, Transfer/Conference, Message, Speaker, Mute
- Automatic selection of prime line or ringing line (the bottom right line key)
- Key selection of non-prime line
- Ringer pitch control
- Hands-free speakerphone operation (full duplex)
- Handset, speaker, and ringer volume controls (Up Arrow and Down Arrow)

- Message Waiting lamp
- Ring Indicator lamp
- Headset port
- Dual Ethernet port to provide connectivity to the LAN for both your telephone and computer
- Adjustable tilt mechanism
- Powered from an adapter or power brick, or an Ethernet connection compliant with 802.3af
- Supports a single Line Interface Module, or 5412 PKM, or up to two 5448 PKMs.
- Supports a Conference Module with Power over Ethernet (PoE)

Conditions

- Each IP Phone requires a license. Licenses are purchasable and enabled in Form 04, System Options/System Timers.
- System Option 131 in Form 04 and COS Option 280 in Form 03 are required to activate the phone's second Ethernet port.
- **NOTE**: If you are upgrading to Release 4.0 or later and plan to use the Mitel 53xx IP phones, you need a 512MB internal flash or a hard drive. If you do not plan to use the 53xx phones, you can still load the software using a 256 MB flash. For more information, refer to Upgrading to the Latest Release.

Programming

Program the phones in CDE Form 09 (Desktop Device Assignments) as device type 5324 or 5324 S/ATT for sub-attendants; enable the required options in Form 04 (System Options/System Timers), and select their COS Options in CDE Form 03 (COS Define).

Operation

Refer to the *5312/5324 User Guide for the SX-200 ICP* available at Mitel OnLine.

Mitel 5330 IP Phone

Description

The dual mode Mitel 5330 IP Phone (see figure) can operate either as a standalone device connected to a SIP (session initiation protocol) service provider or as part of a Mitel ICP system using the MiNet protocol.

With Teleworker Solution Release 4.1 and higher, the 5330 IP Phone can be operated remotely in MiNET mode (i.e. ICP-supported). For information about configuring the 5330 IP Phone to remote mode after installation, refer to the Remote IP Phones Configuration Guide for the Teleworker Solution available at www.mitel.com (see Supporting Documentation).

Features of the telephone include

- Alpha-numeric LCD
- Three softkeys for feature access (top three keys)
- 24 self-labeling, programmable, multifunction keys (12 keys on each of two pages)
- Ten fixed-function keys: Settings, Cancel, Hold, Redial, Transfer/Conference, Message, Volume Up/Down, Speaker, Mute
- Automatic selection of prime line or ringing line (the bottom left line key)
- Key selection of non-prime line
- Ringer pitch control

- Hands-free speakerphone operation (full duplex)
- Handset, speaker, and ringer volume controls (Up Arrow and Down Arrow)
- Message Waiting lamp
- Ring Indicator lamp
- Headset port
- Dual Ethernet port to provide connectivity to the LAN for both your telephone and computer
- Adjustable tilt mechanism
- Powered from a power brick or an Ethernet connection compliant with 802.3af
- Supports 5310 IP Conference Module
- Supports Line Interface Module
- Supports Cordless Module and Devices
- New 5310 sets have backlit display (Rev C or higher)

Note: The set provides power to all modules except the 5310 Conference Module, which requires an external source.

Note: If you replace an existing 5330/40 IP phone with a Rev C or higher version using the same Directory Number, the initial display will not have optimum settings. Follow the instructions in the 5330/5340 User Guide to set the brightness to the desired level.

Conditions

- Each IP Phone requires a license. Licenses are purchasable and enabled in Form 04, System Options/System Timers.
- System Option 131 in Form 04 and COS Option 280 in Form 03 are required to activate the phone's second Ethernet port.
- **NOTE**: If you are upgrading to Release 4.0 or later and plan to use the Mitel 5330/5340 IP phones, you need a 512MB internal flash or a hard drive. If you do not plan to use the 5330/5340 phones, you can still load the software using a 256 MB flash. For more information, refer to Upgrading to the Latest Release.

Programming

SX-200 ICP

Program the phones in CDE Form 09 (Desktop Device Assignments) as device type 5330 or 5330 S/ATT for Subattendants; enable the required options in Form 04 (System Options/System Timers), and select their COS Options in CDE Form 03 (COS Define).

Telephones

The 5330 IP Phone defaults to MiNET mode when first installed and becomes operational when registered with the SX-200 ICP. For instructions on registering phones, see Register the IP Phones.

To program a 5330 IP Phone (Dual Mode) for SIP mode operation, refer to the 5330/5340 Installation Guide, available at Mitel OnLine.

Operation

Refer to the *5330/5340 User Guide for the SX-200 ICP* available at Mitel OnLine.

Mitel 5340 IP Phone

Description

The Mitel 5340 IP Phone (see figure) can operate either as a standalone device connected to a SIP (session initiation protocol) service provider or as part of a Mitel ICP system using the MiNet protocol.

With Teleworker Solution Release 4.1 and higher, the 5340 IP Phone can be operated remotely in MiNET mode (i.e. ICP-supported). For information on how to configure the 5340 IP Phone to remote mode after installation, refer to the *Remote IP Phones Configuration Guide for the Teleworker Solution* available at www.mitel.com (see Supporting Documentation).

The 5340 phone can be programmed to provide enhanced Hotel/Motel subattendant features like displaying/changing room status and setting wake-up calls. For more information about using the 5340 as a subattendant, refer to the *Subattendant User Guide for the Mitel 5340 IP Phone* available at Mitel OnLine.

Features of the telephone include

- Alpha-numeric LCD
- Six softkeys for feature access)
- 48 self-labeling, programmable, multifunction keys (16 keys on each of three pages)
- Ten fixed-function keys: Settings, Cancel, Hold, Redial, Transfer/Conference, Message, Volume Up/Down, Speaker, Mute
- Automatic selection of prime line or ringing line (the bottom right line key)
- Key selection of non-prime line
- Ringer pitch control
- Hands-free speakerphone operation (full duplex)
- Handset, speaker, and ringer volume controls (Up Arrow and Down Arrow)
- Message Waiting lamp
- Ring Indicator lamp
- Headset port
- Dual Ethernet port to provide connectivity to the LAN for both your telephone and computer
- Adjustable tilt mechanism
- Powered from a power brick or an Ethernet connection compliant with 802.3af
- Supports 5310 IP Conference Module (see Note)
- Supports Line Interface Module
- Supports Cordless Module and Devices

Note: The set provides power to all modules except the 5310 Conference Module, which requires an external source.

Conditions

- Each IP Phone requires a license. Licenses are purchasable and enabled in Form 04, System Options/System Timers.
- System Option 131 in Form 04 and COS Option 280 in Form 03 are required to activate the phone's second Ethernet port.
- **NOTE**: If you are upgrading to Release 4.0 or later and plan to use the Mitel 5330/5340 IP phones, you need a 512MB internal flash or a hard drive. If you do not plan to use the 5330/5340 phones, you can still load the software using a 256 MB flash. For more information, refer to Upgrading to the Latest Release.

Programming

SX-200 ICP

Program the phones in CDE Form 09 (Desktop Device Assignments) as device type 5340, or 5340 S/ATT for subattendants; enable the required options in Form 04 (System Options/System Timers), and select their COS Options in CDE Form 03 (COS Define).

Telephones

The 5340 IP Phone defaults to MiNET mode when first installed and becomes operational when registered with the SX-200 ICP. For instructions on registering phones, see Register the IP Phones.

To program a 5340 IP Phone (Dual Mode) for SIP mode operation, refer to the 5330/5340 Installation Guide, available at Mitel OnLine.

Operation

Refer to the *5330/5340 User Guide for the SX-200 ICP* available at Mitel OnLine.

Mitel Application Interface (MiTAI)

Description

The Mitel Telephony Application Interface (MiTAI) software package allows Mitel computer-based applications to access the system features. These applications run on external host computers, and are connected via IP sockets to the MiTAI server on the SX-200 ICP.

This purchased option is enabled in Form 4.

Conditions

- MiTAI Version 11.2 or later are the only versions supported.

Programming

The following steps are required:

- Enable System Option 105 (Mitel Application Interface) in Form 04 (System Options and Timers).
- You must program hunt groups that are used with MiTAI applications as Recall Appearance Back to Originating Set (RABTOS) type hunt groups. For example, if you want to program several Call Center Manager (CCM) Attendants into a hunt group, you must specify the hunt group as a RABTOS hunt group in Form 17 (Hunt Groups).
- In Form 09 (Desktop Device Assignments), program a desktop phone (5020 device) and a soft phone (YAPRO device) by entering the following for extensions 106 and 107:

BAY	SLT	CCT	TEN	EXTN	COS	COR	TYPE	PAGE (main columns may be left blank)
1	01	07	1	106	1	1	5020	01
1	01	08	1	107	1	1	YAPRO	01

- In Form 11 (Data Circuit Descriptors), program a data circuit descriptor for the MAI port.
- In Form 12 (Data Assignment), program PLID x/13/19 as a SOCKET data device type where "x" is the IP bay. (The IP bay is bay 1 by default; it becomes bay 8 when an SX-200 EL/ML database is installed to migrate the system to an SX-200 ICP). Enter the circuit descriptor number (CDN) programmed in Form 11, and enter any unused extension number.
- In Form 34 (Directed I/O), enter the extension number of the SOCKET, select MAI as the printout method and AUTOPRINT as the printout type.

Operation

Operation is dependent upon the application.

Mitel Cordless Module and Accessories

Description

The cordless handset and cordless headset provide 5330/5340 IP Phone users with the capability to move around within their own office or adjacent offices while using the telephone. Both cordless devices connect to your IP telephone through the cordless module, which attaches to the back of the phone. The cordless headset rests and recharges in a headset cradle that attaches to the side of the telephone. The cordless handset recharges in the handset cradle. The Cordless Devices Application provides access to the configuration settings and information screens that apply to the cordless module and accessories.

Conditions

Since only one module type can be attached to the phone at any given time, you can not use a Programmable Key Module (PKM) when the Cordless Module is attached.

Operation

For more information about the Mitel Cordless Module, see the 5330/5340 User Guide for SX-200 ICP available at Mitel OnLine.

Mitel Gigabit Ethernet Stand

Description

The Mitel Gigabit Ethernet (GigE) stand (see figure) is an accessory for 5200/5300 series IP Phones. The stand enables the phone to operate in a 10/100/1000 Mbps Ethernet LAN environment and allows unconstrained Internet bandwidth from the network to desktops.

Additionally, a new Mitel Gigabit Ethernet (GigE) stand (V2) (see figure) is available for 5300 series IP Phones. It is also compatible with the 5212 and 5224 IP Phones.

Conditions

The GigE stand attaches to the base of Mitel 5212, Dual Mode 5215, 5224, 5312, 5324, Dual Mode 5220, 5330 and 5340 IP Phones and requires SX-200 ICP Release 4.0 software.

The GigE stand (V2) attaches to the base of the Mitel 5312, 5324, 5330 and 5340 IP phones and requires SX-200 ICP Release 5.0 UR2 software.

Programming

- In CDE Form 03, enable COS Option 280 (PC (2nd) Port on IP Phone).

Operation

The GigE stand provides a 3-port 10/100/1000 Mbps Layer 2 switch that allow a Mitel phone and a GigE-equipped PC to connect to a GigE LAN.

The three ports are:

1. a GigE LAN port (to connect to the Gigabit switch)

2. a GigE PC port that allows a GigE-equipped PC to connect to the LAN via the stand

3. a 10/100 Mbps Ethernet connection from the stand to the attached phone. The GigE stand supports the IEEE 802.3af POE standards, eliminating the need for a separate power supply for the phone.

Mitel Managed VPN

Managed VPN is a subscription service that makes it easy for businesses of all sizes to reap the benefits of a managed IPSEC Virtual Private Network. The service allows resellers to create, edit or take down site-to-site VPNs between any two or more sites equipped with a server running the Managed VPN software. All VPN configuration is performed via the Mitel Applications Management Center (AMC) through a simple point-and-click web interface. Depending on customer requirements, VPNs can be set to operate in standard mode (fully meshed) or in hub-and-spoke configuration. The service supports both static and dynamic IP addresses.

To ensure quality of service (QoS) for voice calls, Managed VPN provides configurable settings for traffic control and shaping. After being identified and tagged, packets can be classified, queued, and scheduled before they exit an interface, and bandwidth restrictions can be enforced. These features enable voice traffic to be prioritized ahead of data traffic to ensure quality of service for voice calls.

Note: The Traffic Shaping feature applies only to the upload link of the Internet connection.

For more information on setting up and managing VPN, refer to the Mitel Managed VPN Administrator's Guide on the Mitel Edocs Web site at http://edocs.mitel.com.

Managed VPN requires the following at each site:

- server installed with Managed Application Server (MAS) software and the Managed VPN blade
- sufficient internet bandwidth

Conditions

- The Managed Application Server self-study course qualifies technicians to implement a Mitel Managed VPN for voice-only use. Implementation in a converged voice and data network with traffic shaping requires SX-200 ICP Advanced certification.

Programming

Managed Application Server (MAS)

For information on how to install MAS software on a PC or an Applications Processor Card, see the MAS Installation and Administration Guide available via Mitel Online at www.mitel.com.

Managed VPN

For information on how to set up a VPN tunnel with traffic shaping, see the Managed VPN Administrator Guide available via Mitel Online at www.mitel.com.

Operation

All operations are identical to a 5220, 5224 or 5324 IP Phone located at the corporate site.

Mitel Teleworker Solution

Description

The Mitel Teleworker Solution connects a remote office to the corporate voice network to provide full access to voice mail, conferencing and all the other features of the office phone system.

It requires the following:

Corporate Site	Remote Site
• server installed with Managed Application Server (MAS) software and the Teleworker Gateway blade • static IP address • sufficient internet bandwidth (approximately 50 kbps is required per teleworker if G.729 compression is enabled)	

- 5212 IP Phone, 5215 IP Phone, 5020 IP Phone, 5224 IP Phone, 5220 IP Phone, 5312 IP Phone, or 5324 IP Phone with phone load 5.22 or later

• DSL/cable router with Network Address Translation (NAT) and local DHCP • Broadband connectivity (static IP address is not required)

Note: The Managed Application Server (MAS) must be on the same VLAN (subnet) as the SX-200 ICP.

The Teleworker Solution can be completely configured at the corporate site using a 5020, 5212, 5215, 5220, 5224, 5312, 5324, 5330, 5340, or Navigator IP phone. Once configured, the phone can then be taken off-site and plugged into any broadband Internet connection. When the phone is powered up, it automatically establishes a connection with the Teleworker Gateway and is registered as a standard extension on the SX-200 ICP. The phone can also be returned to normal mode (non-teleworker) with the touch of a button.

Conditions

- Requires Teleworker Solution Version 3.0 or higher
- Each Teleworker phone requires an IP license.
- The MAS self-study course qualifies technicians to implement a Mitel Teleworker for voice-only use. Implementation in a converged voice and data network with traffic shaping requires SX-200 ICP Advanced certification.

Programming

SX-200 ICP

- Form 09, Desktop Device Assignments — Assign set type, extension number, COS, COR, and name (optional) for each bay/slot/circuit that will host a Teleworker phone.
- Phones with low bandwidth connections should have Option 688 (IP Set Requires Compression) enabled in their Class of Service

Teleworker Phone

Register the phone by entering the following information:

- the IP Address of the Teleworker server
- the Netmask or Subnet Mask for the office network (only required if giving the phone a static IP address).
- the Default Gateway IP Address for the office network.

See the Teleworker Remote Phone Configuration Guide available at www.mitel.com.

Managed Application Server (MAS)

See the Manged Application Server documentation available via Mitel Online at www.mitel.com.

Operation

All operations are identical to the IP Phone located at the corporate site.

Mitel Wireless LAN Phone Stand

Description

The Mitel Wireless LAN (WLAN) stand (see figure) is a network accessory for 5200/5300 series IP phones that provides an 802.11 b/g WiFi LAN connection in place of a wired Ethernet LAN and enables phones to operate in a wireless LAN (WLAN) environment.

The WLAN Stand and its attached IP Phone operate in two modes:

1. **Client Mode** where it interfaces with a standard Wireless LAN Client that supports WMM QoS and WPA/WPA2 for security, and as a Client device, connects to the LAN via the Access Points (Mitel or 3rd Party).

2. **Access Point Mode** where it acts as a LAN connected Wireless Access Point for up to 6 of the Mitel dual mode IP Phones equipped with the WLAN Stand and allows wireless LAN Access for wireless data devices, or wired PCs attached to the IP Phones themselves. The WLAN Stand is built into a 35 degree phone base stand that snaps onto a 5200 or 5300 Series IP Phone so that it can act as a replacement for the stand that ship with the phone.

Conditions

- The WLAN stand attaches to the base of Mitel 5212, 5312, Dual Mode 5215, 5224, 5324, Dual Mode 5220, 5330 and 5340 IP Phones and requires SX-200 ICP Release 4.0 software.
- An IP Phone with a WLAN Stand configured as an Access Point CANNOT have a PC plugged into it because the PC Port on the IP Phone is being used to connect the WLAN AP to the Network.
- Release 1.0 of the WLAN Stand does not allow a wired PC to be connected to the IP Phone with a WLAN Stand in Client Mode that is communicating with other Vendors' (3rd Party) APs. Only IP Phones with WLAN Stands (clients) talking to the IP Phone with the WLAN Stand (AP Mode) will support wired PC connections in the 2nd port on the Phone. This limitation will be fixed in a Firmware Upgrade (Version 2) of the Wireless LAN Stand that will be available to be downloaded from MOL at no charge.
- The WLAN Stand does NOT support PoE. A new Power Adapter is shipped with every WLAN Stand. The WLAN Stand Power Adapter also powers the IP Phone. It requires a country-specific C7 Power Cord to be purchased separately.
- 802.1x authentication is NOT supported on the WLAN Stan, nor any voice or data devices used behind the WLAN Stand.

Programming

- In CDE Form 03, enable COS Option 280 (PC (2nd) Port on IP Phone) for both Access Point and Client sets.

Operation

The Wireless LAN Stand provides wireless voice communication over an Internet Protocol (IP) network. Because the Wireless LAN Stand connects to your wireless LAN, you can place and receive phone calls from anywhere in your wireless environment. The Wireless LAN Stand attached to the IP phone can act as either a wireless client or an access point.

As an **Access Point**, the Wireless LAN Stand can connect a wired LAN to one or many wireless devices. Wireless clients communicate with other wireless clients via the access point. When configured as an access point the WLAN Stand bridges the wireless network to the wired network which allows wireless clients to access computers and clients on the wired network. The Wireless LAN Stand conforms to the 802.11b and 802.11g standards for wireless LANs. Access points, which serve as communication bridges between the wired LAN and the wireless LAN and also serve as central repeaters so WLAN clients can communicate with each other.

Access Point Capacity: Access points are a resource or link that is shared by all of the clients and any given AP can only offer a finite amount of bandwidth and processing capability. The Mitel WLAN Stand can support a maximum of six IP phones and six PCs, plus the phone attached to the AP WLAN Stand. Allowing more clients than the allowable maximum to access the AP will cause degradation in performance and voice quality.

Wireless Client: The Wireless LAN Stand and its attached IP phone comprise a wireless client that is associated with an access point for connection to the intranet or Internet. The Wireless LAN Stand acts as a bridge between local Ethernet (to the phone) and the WLAN and passes bidirectional VoIP phone and data traffic. Wireless LAN Stands configured as clients can have a PC connected to the PC port on the IP Phone to provide access to the office network.

Note: To achieve optimum voice quality, an access point's operating range should be limited to a 30 meter (100 feet) radius. This will vary depending on how many voice and data clients are to be connected and how much other 2.4GHz activity is in the area.

Note: For optimum performance, it is recommended that only 802.11g devices be connected, as even one 802.11b device on the wireless network will degrade the performance of the entire Network.

The Wireless LAN Stand is configured through a web browser on a PC that is connected to the WLAN Stand. For complete Configuration instructions, please refer to the Wireless LAN Stand Engineering and Configuration Guidelines available at Mitel OnLine.

Mitel Your Assistant

Description

Mitel Your Assistant is a PC-based application that lets users control their Mitel phone from their computer. With Your Assistant's intuitive interface, users can make and receive calls, manage conferences, time and annotate calls, monitor the presence availability of other users, and participate in secure chat sessions with other users. Your Assistant also provides a calling interface when dialing from a third-party Personal Information Manager (PIM) such as Microsoft Outlook, plus integration with MSN Messenger. By purchasing an additional license and enabling the Softphone module, users can make and receive calls using an embedded IP-based software phone.

A "Lite" version of Your Assistant is available in addition to the full-featured version. Major features supported by the two versions are listed in the following table:

Table: Your Assistant Feature List

Feature	YA	YA Lite
YA Softphone module	•	
Management software on separate server	•	

Table: Your Assistant Feature List

Feature	YA	YA Lite
Licensing through Mitel Applications Management Center (AMC)	•	
Support for 9 languages	•	•
Screen pop-ups on calls	•	•
Screen pop-ups on calls with ability to forward or send to voice mail	•	
In-call ability to transfer, conference, hold, and hang up	•	•
Drag-and-drop conference calls (5 parties)	•	•
Call forward profiles	•	
Personal directory	•	•
LDAP directory synchronization with support for Active Directory accounts	•	
Call history - local	•	•
Favorites menu	•	•
Call timer and annotation tools	•	•
Presence indicator - BLF	•	
Presence indicator - computer	•	
Advisory Message	•	
PIM integration (Outlook, Lotus Notes, ACT, Goldmine)	•	
Dial from PIM (Outlook only)	•	•
MSN integration	•	
Secure instant messaging with file transfer	•	
Ability to work offline	•	
Web window		•

Feature Availability

Mitel Your Assistant can be integrated with the following devices:

- SUPERSET 4015, 4025, and 4150 Phones
- Mitel 5207, 5010, 5212, 5215, 5020, 5220, 5224, 5304, 5312, 5324, 5330, and 5340 IP Phones
- Click here to see a list of the features supported by YA Softphone.

Supported PIMs include:

- Act! 2000 (Best Software)
- Lotus Notes R5 and R6 (IBM)
- Outlook 97 or greater (Microsoft)
- GoldMine (FrontRange)

Note: The maximum number of Your Assistant integrated desktops is equal to the number of IP phones supported on the controller (100 on the CX/CXi and 248 on the MX/AX).

Conditions

- The purchasable System Option 105, Mitel Application Interface must be enabled.
- Your Assistant requires MiSN Universal SDK 1.0.0.3 or higher
- The SX-200 ICP Controller supports release 3.0.7 of Mitel Your Assistant

Licensing/programming requirements (not applicable to Your Assistant Lite):

- One Your Assistant license per user.
- One voice mailbox per user (for the desk phone).
- One Mitel 5020 license per user (for the YA Softphone).
- Two extension numbers per user (one for the YA Softphone and one for the desk phone).

Programming

SX-200 ICP

The following steps are required:

- Enable System Option 105 (Mitel Application Interface) in Form 04, System Options/System Timers.
- Specify a circuit for the MiTAI connection to the SX-200 ICP in Form 12, Data Assignment. Use any unused extension number, and enter 1/13/19/SOCKET for BAY/SLT/CCT/TYPE.
- Specify the MiTAI connection to use the extension number in Form 34, Directed IO.
- Program the YA Softphone as a 5020 IP Phone, 5220 IP Phone, 5224 IP Phone, or 5324 IP Phone in Form 09, Desktop Device Assignments.

Notes:

1. After the YA Softphone connects to the Mitel Messaging Server (Speak@Ease), it will appear in Form 09 as a 5220 IP Phone regardless of how it was originally programmed. This is a display issue and does not cause a problem.
2. The YA Softphone supports only one line appearance and does not have feature keys.
3. The YA Softphone does not support paging.
4. If Your Assistant users are deleted from the system, the corresponding directory entries and voice mailboxes must also be deleted.

Your Assistant Server and Clients

For information on programming the Your Assistant server and clients, see the Your Assistant Administration Guide which is provided in PDF format on the Mitel Edocs website (http://edocs.mitel.com).

Operation

For information about using the Your Assistant client application, refer to the Your Assistant User Guide which is provided in PDF format on the Mitel Edocs website (http://edocs.mitel.com).

Mobile Extension

Description

Mitel Mobile Extension® is a software solution that lets users twin their desk phone with an internal or external PSTN-connected phone (a cell phone, for example). Calls arriving at the desk phone ring the cell phone simultaneously, until one or the other is answered, or if unanswered, are forwarded to voice mail. Administrators configure system settings using an administrative web interface. Users program their personal settings using either a web interface or a Telephony User Interface (TUI).

About Calling Line ID (CLID)

The CLID feature provides Mobile Extension users the ability to see who is calling them by displaying PSTN telephone numbers, and in some cases, caller name. CLID operation is dependent on your region, trunking interface, and CO switch type. The following restrictions apply to Mobile Extension CLID:

- Calling Line ID (and CPN substitution) are supported on the following trunks only:
  - PRI/BRI
  - DASS2
  - DPNSS
- CLID feature requires Mobile Extension Release 1.6 or higher.
- Maximum CLID length is 15 characters. In the case of internal calls, the first name is truncated to an initial and a period;
  last name is truncated as required.
- For proper CLID operation, calls must be identified as internal or external. For example, a call from a remote ICP is classified as External. In this case, Mobile Extension uses the calling line ID received (that is, CPN substitution does not occur). For local calls, CPN substitution is performed.
- The CLID feature is not supported if the User's mobile device is a local extension.
- Your region or CO may block CLID functionality.

Programming

1. In CDE Form 4 (System Options/System Timers), enable MiTAI (Option 105).
2. In CDE Form 4 (System Options/System Timers), enable Mobile Extension - CLID (Option 30) for CLID functionality.
3. In CDE Form 12 (Data Assignment), specify a circuit for the MiTAI connection to the SX-200 ICP. Use any unused extension number, and enter 1/13/19/SOCKET for BAY/SLT/CCT/TYPE.
4. In CDE Form 34 (Directed IO), specify the MiTAI connection to use the extension number.
5. Program a Class of Service (COS) for each of the following:

For these extensions:	**Set the following COS options:**
Desktop Monitors	

- Default settings for IP sets with options 683, 684, 685 disabled
  AND
- Enable COS Option 282 - Standard Hard Hold

Outgoing Lines and TUI Lines	• Default settings for IP sets with options 683, 684, 685 disabled.
Desktop Extensions	

- Disable COS Option 240 – Line Privacy
- Set COS Option 253 Call Forward – Don't Answer Timer value to 5 or 6 rings for both desk phone and twinned phone. The longer-than-usual timer provides the user of a twinned cell phone more time to pick up the call before it is routed to voice mail.
- Disable COS Option 681 – Call Hold Notify Timer and set to zero. (Call hold timer settings can interfere with call hold recall to the twin device.)

6. In CDE Form 9 (Desktop Device Assignments),
   - Program a dedicated Desktop Monitor for each twinned extension. A Desktop Monitor is a virtual extension (set type: 5220) that monitors the twinned extension for incoming calls. (A virtual or "phantom" extension is one that is programmed in the system but has no physical device associated with it. A virtual extension is programmed by leaving the MAC address field blank.) Assign the Desktop Monitor COS to each Desktop Monitor.
   - Program the Desktop Monitor virtual extension with a Key System type (Keyline) line appearance of its twin extension.
   - Program the Line Pref parameter to Manual.
   - In the Desktop Monitor Line Appearance subform, set DIR to In/Out and RING to Immed.
   - Program one or more virtual 5220 extensions as Outgoing Lines. The Outgoing Lines dial out to the twin phones and must be on a controller that has access to the PSTN. The number of Outgoing Lines required is typically a third the number of twinned extensions. Assign the Outgoing Line COS to each Outgoing Line.
   - Program one or more virtual 5220 extensions as TUI Lines. Users call the TUI Lines to enable or disable twinning on their extension or to change personal settings such as password and mobile phone number. Assign the TUI COS to each TUI line.
7. In CDE Form 17 (Hunt Groups), assign the TUI Line extensions to a hunt group.
8. If your outbound trunks are anything other than PRI or IP, in CDE Form 13 (Trunk Circuit Descriptors), enable "Far End Gives Answer Supervision".

Moving Stations and Mitel Telephones

Description

This feature allows extensions to be moved easily from one circuit to another. Previous programming for the extension, such as name, COR, COS, etc. is preserved and moved with the extension.

Conditions

The device must be idle when the move is attempted and the new location must have nothing programmed.

Programming

Refer to the Program CDE section, for full details of programming.

Operation

Refer to the Program CDE section, for full details of operation.

Multi-Attendant Positions

Description

The system can handle multiple attendant consoles with unique hold slots for each attendant. Incoming trunk calls can be programmed to appear at all consoles, or specific console(s). Similarly, extension Dial 0 calls, Priority Dial 0 calls, Intercept to Attendant calls, etc., can be programmed to appear at all consoles, or at a specific console(s).

Any console in a particular tenant group can switch that tenant group to Night Service or to Day Service; see Attendant Night/Day Switching.

See Tenanting, Recall, and Console LDN Keys; see also Attendant Transparent Multi-console Operation.

Conditions

The following conditions apply to this feature:

- When a call appears at more than one console, the first console to answer is connected to the call; other consoles stop being alerted for this call.
- There is a maximum of eleven consoles per system.

Programming

Assign consoles via CDE Form 07 (Console Assignments).

Operation

All operations are identical for all attendant consoles in the same COS and tenant group.

Music-on-Hold (MOH)

Description

A customer-provided system music source can be connected to the SX-200 ICP via the Music-On-Hold connector on the back panel, through a Music-on-Hold/Pager module on the Universal Card or through a DNIC Music-on-Hold/Pager (DMP) unit. This music source can then be used for Campon, Hold, UCD, ACD, and other features. Music can also be obtained through the Music from an ONS Source feature.

See Campon and Hold.

Refer to the Installation Information section, for wiring details.

Each tenant of the system can also have its own Music-on-Hold source through a DMP unit that is connected to a DNIC Port. Each DMP unit is accessed by a directory number programmed in its tenant group. Refer to the Peripheral Devices section for installation instructions.

Conditions

The following conditions apply to music-on-hold provided through the Music-On-Hold connector on back panel of the SX-200 ICP:

- Any music source (radio, CD player, etc.) with a mini (1/8" - 3.5 mm) phono plug may be connected to the Music-On-Hold connector. Use the device's volume control to adjust volume levels as required.
- Input signals should be in the range of 10 to 100 mVrms. Any DC voltage applied to the input shall be less than 50 VDC.
- Powering down the SX-200 ICP redirects the MOH source to the paging output. To stop the music from being heard over the pager, power down both the MOH source and paging amplifier before powering down the SX-200 ICP.

The following conditions apply to music-on-hold provided through a Music-on-Hold/Pager module:

- Use of the Music-on-Hold feature may under applicable copyright or other provincial, local, state and/or federal rules, regulations and/or statutes require that you obtain a license from the local performing rights society or copyright owner, before you can provide music on hold to telephone users. Contact your music supplier for more information.
- On the Universal Card, the music source should be between 50 and 500 mV rms.
- Up to 25 different music sources can be programmed.
- Input to the system is 600 ohms ac transformer coupled. A dc voltage should not be applied to this input.
- To display a MUSIC prompt, a music source must be connected to the PBX.

The following conditions apply to music-on-hold provided through a DMP unit:

- On the DMP unit, the music source should be between 10 and 100 mV rms.
- If no music source is programmed in Form 19, Call Rerouting Table, the system Music-on-Hold source programmed in Form 18, Miscellaneous System Ports, will be used.
- Each Bay that connects via a Control Dual FIM Carrier requires one dedicated channel for each music source programmed in Form 19, Call Rerouting Table.
- Each Bay that connects via a Control Triple FIM Carrier requires two dedicated channels for each music source programmed in Form 19, Call Rerouting Table.
- When a caller from within a tenant group puts a call on hold, music will be provided from that tenant's music source as programmed in Form 19 for the time of day, or from the system music source if outside the times programmed in Form 19.
- If neither a tenant or a system music is available, the call that is put on hold will receive music from the music source of its tenant group.
- A call to a DMP will receive reorder tone.
- Calls incorrectly routed or rerouted to a DMP directory number will not be routed.

Programming

The default database contains the necessary programming for Music-on-Hold. If using a different database, complete the following programming:

- In Form 18 (Miscellaneous System Ports), assign Music-on-Hold to PLID n/13/29/01, where "n" is the IP bay number (default 1).

For a DMP unit, program the following:

- Program a Digital Line Card or DNIC Module in Form 1.

- Program the DNIC Music/Pager unit with device type “DMP Music Source For This Tenant” in Form 9.
- Program each Music-on-Hold source in Form 09 (Desktop Device Assignments) with a directory number.
- Program each Music-on-Hold source in Form 19, Call Rerouting Table, including required Day, Night1, and Night2 service.

Operation

With the DMP unit, the Music-on-Hold source is selected first by Tenant, second by the system, according to programming of the telephone that put the call on hold, and third by the tenant of the telephone that is on hold.

Music from an ONS Source

This feature allows you to use music from ONS sources as a cost-saving alternative to music from DMP sources.

The ONS music source is connected in the same manner as an alternate music source for an ACD Path.

Conditions

The following conditions apply to this feature:

- It requires Feature Level 4 or greater enabled in System Option 102.
- If the music source is programmed in Form 19, Call Rerouting Table, each Bay that connects via a Control Dual FIM Carrier requires one dedicated channel for each programmed music source. Each Bay that connects via a Control Triple FIM Carrier requires two dedicated channels for each music source.

Programming

WARNING: The music source must be connected before it is programmed. If it is not connected when it is programmed, the device must be disconnnected and reconnected, the card must be removed and reinstalled, the bay must be reset and the main must be reset.

- In CDE Form 04, System Option and Timers, enable Feature Level 4 in System Option 102.
- In CDE Form 09, Desktop Device Assignments, program the music source as an ONS port.
- To set up the music source as a listen-only conference, define an ONS port using the Music Source Following option in CDE Form 41, ACD Path.
- To set up the music source as a system-wide music source, program a Music-on-Hold source in CDE Form 18, Miscellaneous System Ports. A new channel is allocated to each Bay.
- To set up music sources for tenants, specify the Music Source for this Tenant call rerouting option in CDE Form 19, Call Rerouting Table. A new channel is allocated to each Bay.
- After a programmed ONS source is physically connected, its status appears as "altms" in a maintenance report.

Operation

This music source can be used for Campon, Hold, UCD, ACD, and other features.

Names

Description

The system programmer can assign names to extensions, classes of service, tenants, trunks, trunk groups, ACD paths, ACD positions, ACD agent groups, and hunt groups. Mitel display telephone users can program their name from their set.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Trunk, trunk group, ACD path, ACD agent group, tenant, and class of service names can be up to 8 characters long.
- Extension and ACD position names can be up to 10 characters long; hunt group names can be up to 12 characters long.
- Hunt groups must have an access code programmed before a hunt group name can be added.
- The programmer cannot enter names from the attendant console because it has no alphabetic keyboard.
- Tenant and COS names appear only on consoles. Set and trunk names appear on consoles, and Mitel display telephones.
- Mixed case is only allowed for hunt groups.
- Programming the name of a Mitel telephone in CDE overwrites the name programmed at the Mitel telephone. Also, programming the name at the Mitel telephone overwrites the name in CDE.
- The characters \, ~, |, {,} are invalid except for hunt group names.
- Trunk groups cannot include single or double quotation marks.
- Hunt group names cannot start with a digit (0 through 9), # or * or have a blank or a - in them.
- Analog Networking information may replace trunk or trunk group names on displays; see Analog Networking.

Programming

Refer to the following forms in the Program CDE section, for further information:

Name Type	**CDE Form**
ACD Path Names	Form 41 (ACD Paths)
ACD Position Names	Form 39 (ACD Agent Groups) and Form 40 (ACD Supervisors)
Class of Service Names	Form 03 (Class of Service Define)
Hunt Group Names	Form 17 (Hunt Groups)
Set Names	Form 09 (Desktop Device Assignments)
Trunk Names	Form 15 (Dial-In Trunks) or CDE Form 14 (Non-Dial-In Trunks)
Trunk Group Names	Form 16 (Trunk Groups)

Name Type	CDE Form
Tenant Names	Form 19 (Call Rerouting Table)

Operation

Family 4 Telephones:

To program a new name or change an existing name:

- Press SUPERKEY.
- Press the NO softkey until NAME? appears in the display.
- Press the YES softkey
  - If a name isn't currently programmed, ENTER NAME appears in the display.
  - If a name is programmed, the name is displayed. Press the CHANGE softkey. ENTER NAME appears in the display.

Enter the name using the telephone keypad. Above each key are printed its associated letters, e.g., the "2" key has the letters "abc" above it. To choose the first letter, press the key once; to choose the second letter, press the key twice, etc. When the last associated letter appears, a subsequent press displays the key number. Further key presses cycle through the letters again. When the desired letter is displayed, press the → softkey to enter it. To correct an erroneous entry, use the ← softkey.

Telephone keys 1, *, 0 and # are not marked with all the characters they can enter. They are as follows:

Key	Characters
1	! ? % 1
*	' - *
0	@ & $ 0
#	. , / #

- When the name is complete, press the SAVE softkey.

To clear an existing name:

- Press SUPERKEY.
- Press the NO softkey until NAME? appears in the display.
- Press the YES softkey. The name currently programmed for the set is displayed.
- Press the ERASE softkey (DEL key on the Symbol MiNET Wireless Phone). NAME ERASED appears briefly in the display and then shows the date and time.

Family 1 Telephones:

To program a name:

- Press the SUPERKEY.
- Press the YOUR NAME softkey. The display changes to ENTER NAME:

Letters are displayed on the LCD as they are dialed on the telephone keypad. The first press of any of these keys displays the first letter that appears on its key cap. The second press of the same key changes the display to the second letter and so on. When all the letters associated with a key have been displayed, the number is displayed. Further presses cycle through the letters again. When the desired letter is displayed, enter it by pressing the → softkey or by entering the next letter, if it is on a different key. (The → key is also used to enter spaces). Follow the same procedure to find and enter the other letters in the name. If an error is made, press the ← softkey to back up and change a letter.

Telephone keys 1, *, 0 and # are not marked with all the characters they can enter. They are as follows:

Key	Characters
1	! ? % 1
*	' - *
0	@ & $ 0
#	., / #

- When the name is complete, press the SAVE softkey.

Family 5 Telephones:

To program a new name or change an existing name:

5330 Phones:

- Press the key you have programmed as Superkey.
- Press No until Name appears.
- Press the Yes softkey
  - If a name isn't currently programmed, ENTER NAME appears in the display.
  - If a name is programmed, the name is displayed. Press the CHANGE softkey. ENTER NAME appears in the display.

Using the onscreen keyboard, press the appropriate letters to spell the name. Use the page navigation keys to access more letter/number choices, and then press Insert after each entry.

- Press Save when the name is complete.

5340 Phones:

- Press the key you have programmed as Superkey.
- Press More until Name appears, and then press Name
- Using the onscreen keyboard, press the appropriate letters to spell the name. Use the page navigation keys to access more letter/number choices, and then press Insert after each entry.

Never a Consultee

Description

This feature protects an extension from being dialed or retrieved by extensions that have a consultation hold in progress.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- The feature is checked when a caller is retrieved from being held by an extension or by the console.

Programming

Enable COS Option 233 (Never a Consultee) for the extension.

Operation

An extension establishes a call. The extension flashes, putting the call on consultation hold, and dials a third party that has the feature enabled. The call to the third party is blocked and the calling extension receives reorder tone.

Never a Forwardee

Description

This feature prevents an extension or console from having any calls forwarded to it by another extension. Extensions are prevented from setting up forwarding to extensions or consoles with the feature enabled.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Calls directed to the extension via hunting or speedcalls are not affected by the selection of this feature.
- If an extension attempts to forward a call to an extension with this feature enabled (the feature was enabled after the forwarding was setup), the forwarding is ignored.

Programming

Enable COS Option 234 (Never a Forwardee) for the extension or console.

Operation

None.

New Call Ring

Description

When a Mitel telephone is busy, and a new call attempts to ring the set, a single burst of ringing will be heard through the set speaker and the Message Waiting lamp will flash.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- The new call ring alert is not available if the Mitel telephone is using the speaker in handsfree mode.
- On Mitel 5207 IP Phones, the new call ring alert is heard through the handset or headset, not the speaker.

Programming

(Optional) Enable COS Option 611 (Telephone - Limited New Call Ring) so that only the first ringing line provides a new call ring alert. This option is useful for sets and subattendants that have a high amount of traffic on one or more line appearances or [i]LDN indicators.

Operation

None.

NI3 Calling Name Delivery

Description

The NI3 Calling Name Delivery feature allows the called party to see the name of the caller on the display screen of the telephone on incoming calls. The NI3 protocol also allows a link between calling name and calling number for outgoing calls. The caller can set the calling party number presentation indicator to "Allowed", and the calling name stored at the Central Office will be displayed with the calling number. The presentation of calling number and calling name can be allowed or prohibited through IMAT programming.

Conditions

The following conditions apply to this feature:

- The Central Office must support NI3 Calling Name Delivery according to specification NI99 SR4619.
- Depending on the Central Office operation, the calling name may appear when the telephone is in the ringing state or in the answered state.
- NI3 Calling Name Delivery supports names with a maximum of 15 characters.

Programming

To enable Calling Name Delivery and extended SMDR display of calling name, enable the following COS Options in the Class of Service Options window:

- Option 502 (Display ANI/DNIS/CLASS Information) in Telephone COS
- Option 503 (Display CLASS Name) in Telephone COS
- Option 811 (ANI/DNIS Trunk) in Trunk COS
- Option 246 (SMDR - Extended Record) in Trunk COS.

Note: When Option 502 is enabled, Mitel telephones display Automatic Number Identification (ANI) digits (calling number) during the ringing state and Dialed Number Identification Service (DNIS) digits (called number) during the answered state. To program the Mitel to display ANI digits during both states, enable Option 613.

To enable outgoing calling name and calling number presentation, program the CLIR option in the IMAT Outgoing Call Characteristics screen to "Allow". To inhibit calling name and calling number presentation, program the IMAT CLIR option to "Restrict". If the CLIR option is set to "Null", call by call programming will be in effect.

Note: The programming of CPN Delivery in the IMAT Incoming Call Characteristics screen has no effect on NI3 Calling Name Delivery.

Operation

See CLASS for Digital Sets and Automatic Number Identification (ANI)/Dialed Number Identification Service (DNIS) on Incoming Trunks.

Night Bells

Description

This feature allows incoming and internal calls to be directed to common alerting devices. The call can be answered from the attendant console or from an extension with TAFAS Access; see Trunk Answer From Any Station (TAFAS).

The system provides a contact closure which operates the alerting device.

Night Bells are activated by relays in any of the following locations:

- Relay Port connector on SX-200 ICP controller
- ONS port on SX-200 ICP controller
- on the DNIC Music-on-Hold/Pager Unit

Refer to the Install a Night Bell for relay pinouts, PLIDs, and other installation details.

The extension number assigned to the night bell can be used as an answer point or alternate answer point for most features in the system.

Night bells can be integrated with the paging system on the DNIC Music-on-Hold/Pager Unit. A warble tone is available on the paging system when a night bell is active. The night bell waits for an idle pager.

Conditions

The following conditions apply to this feature:

- Up to 25 night bells can be programmed.
- See Attendant Default Call Positions for a description of the console NIGHTBELL softkey.
- More than one caller can ring a night bell at a time - it is always available to be called and no caller will find it busy.
- Extensions can dial a night bell directly.
- The night bell itself has no tenant. Tenant checks are done using the tenant of the night bell’s caller.
- Standard ringing cadence only is used for night bells.
- Calls can be transferred to night bells.
- Recall features do not operate when ringing a night bell.

Programming

- In Form 18 (Miscellaneous System Ports), assign a generic relay PLID and extension number to the night bell. The default is 1/13/29/01, extension 340 on the MX; and 1/13/30/04 extension 340 on the CX/CXi. For complete relay pinouts and PLID assignments for the MX, see Relay Connector Pinouts; and for the CX/CXi, see Analog Main Board Pinouts and Analog Option Board Pinouts.
- In Form 14 (Non-Dial in Trunks), assign the night bell extension as night answer point.
- If the DNIC Music-on-Hold/Pager Unit is being used, program the following as well:
  - Program a Digital Line Card or DNIC Module in Form 1.
  - Program the DNIC device type as “DMP” (DNIC Music/Pager) in Form 9.
  - Program a pager number in Form 18 with each night bell.

Operation

See Trunk Answer From Any Station (TAFAS).

Night/Day Switching

Description

Extension users can put the system (or particular tenant group or groups) into DAY service or one of two different night service modes, NIGHT1 or NIGHT2. In Night Service, Mitel display telephones display NIGHT 1 SERVICE or NIGHT 2 SERVICE as appropriate. Also see Night Services, Tenanting, and Attendant Night/Day Service Switching.

Note: Night/Day Switching is a purchaseable option. Upon purchase and activation, System Option 102 is set to 6.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Use of a Feature Key (in addition to existing Superkey functionality) to enable or disable this feature is available on multi-line Family 1, Family 3, Family 4, and Family 5 telephones only.
- Tenanting restrictions can be applied; see the Tenanting feature description or refer to the main Tenanting section.

Programming

- Enable COS Option 609 (Telephone - Night Service Switching) in the set's class of service to allow users to toggle from DAY to NIGHT1 and to NIGHT2.
- Enable COS Option 691 (Telephone - Day/Night1 Service Switching) in the set's class of service to allow users to toggle only from DAY to NIGHT1.

Note: Only one of these COS Options may be enabled at a time per telephone as they are mutually exclusive.

- To provide feature key activation of Night/Day Switching, program a DAY/NIGHT feature key on the telephone. (See Feature Keys.)

Operation

Operation varies depending upon the type of telephone and the method of activation as described below.

Note: If COS Option 691 has been enabled, NIGHT2 will not be displayed as one of the available options.

Feature Key LED States

- If the set is in Day Service mode, the LED will be off.
- If the set is in Night1 Service mode, the LED will flash slowly (green for sets that support a dual-colour LED, red if otherwise).
- If the set is in Night2 Service mode, the LED will quickly flash red.

Note: The Feature Key LED status follows the tenant setting programmed in CDE Form 06 (Tenant Night Switching Control) and is not dependent on COS Options 609 and 691.

Family 4 Telephones (except Symbol MiNET Wireless) using SuperKey:

- Press SUPERKEY.
- Press the NO softkey until NIGHT SERVICE? appears in the display.
- Press the YES softkey. The current service mode is displayed:
  DAY SERVICE, NIGHT 1, or NIGHT 2.

- Press the CHANGE softkey.
- If the mode shown is the one desired, press the YES softkey. If not, press the NO softkey and the other alternative is displayed; then press the YES softkey. To leave the system in its current mode, press SUPERKEY.
- Press SUPERKEY.

Family 3 Telephones using SuperKey:

- Press SUPERKEY.
- Press # for NO until NIGHT SERVICE? appears in the display.
- Press * for YES. The current service mode is displayed:
  DAY SERVICE, NIGHT 1, or NIGHT 2.
- Press * for CHANGE.
- If the mode shown is the one desired, press *. If not, press the # and the other alternative is displayed; then press the *. To leave the system in its current mode, press SUPERKEY.
- Press SUPERKEY.

Family 1 Telephones using SuperKey:

- Press the SuperKey.
- Press the More softkey until the Night Answer softkey appears.
- Press the Night Answer softkey. The top display line indicates which mode is currently active (e.g., CURRENTLY DAY SERVICE). Softkeys appear for the two alternatives.
- Press the softkey for the desired mode; or, to leave the system in the current mode, press the Backup softkey or the SuperKey.

Family 5 Telephones:

5330 Phones:

- Press the key you have programmed as Superkey.
- Press No until Night Service? appears in the display.
- Press Yes. The current service mode is displayed:
  DAY SERVICE, NIGHT 1, or NIGHT 2.
- Press the Change softkey.
- If the mode shown is the one desired, press the Yes softkey. If not, press the No softkey and the other alternative is displayed; then press the Yes softkey. To leave the system in its current mode, press Superkey.

5340 Phones:

- Press the key you have programmed as SuperKey.
- Press the More softkey until the Night Answer softkey appears.
- Press the Night Answer softkey. The top display line indicates which mode is currently active (e.g., CURRENTLY DAY SERVICE). Softkeys appear for the two alternatives.
- Press the softkey for the desired mode; or, to leave the system in the current mode, press the Backup softkey or the SuperKey.

Symbol MiNET Wireless Phone:

To turn night service on:

1. Press the No softkey until "Night Service?" appears.
2. Press the Yes softkey.
3. Press the CHG softkey.
4. Do one of the following:
   - Press the Yes softkey to select Night Service #1.

- Press the No softkey to select Night Service #2.

Using A Feature Key:

- Press the DAY/NIGHT feature key to toggle between the different options available.

Night Services

Description

The system has three different service modes: DAY, NIGHT1, or NIGHT2. When the system or tenant group is in night service mode, incoming trunk calls and calls to the attendant may be rerouted to specified extensions or be caused to activate common alerting devices (Night Bells).

Conditions

The following conditions apply to this feature:

- This feature is available on a per tenant basis.
- Some features, such as Alternate Recall Point, Attendant Call Forward- No Answer, and Call Rerouting, operate differently under NIGHT1 and NIGHT2.

Programming

CDE Form 14 (Non-Dial-In Trunks) and CDE Form 19 (Call Rerouting Table) programming is directly affected by Night Services.

Operation

See Attendant Night/Day Switching, Night/Day Switching, Call Rerouting, Trunk Operation - Non-dial-in CO, Attendant Access (Dial 0).

Night Services Flexibility

Description

This option allows the attendant to change the night service assignment of non-dial-in trunks. The system allows full flexibility of trunk assignment.

Conditions

The following conditions apply to this feature:

- The console must be able to connect to the trunk (see Device Interconnection Control).
- CDE is prevented from accessing Form 14 (Non-dial-In Trunks) when the console is updating the night point of a trunk. Similarly, the console is prevented from changing the night service while CDE is accessing Form 14.
- The rules for the assigned access codes are the same as in CDE.

Programming

Enable COS Option 104 (Attendant Flexible Night Service Setup) for the Console.

Operation

Change the night service assignment of non-dial-in trunks from the console, as follows:

- Press FUNCTION.
- Press ATT. FUNCTION.
- Press MORE.
- Press FLEX NIGHT.

The display prompts for trunk number.

- Enter trunk number.
- Press either NIGHT1 or NIGHT2.
- Enter new destination.
- Press SET.

The trunk is now routed to the new destination when the system is in Night 1 or Night 2 (as programmed above).

Node Identification

Description

The node identification feature works with the analog networking feature to provide consistent dialing of extension numbers throughout a network of IP PBXs.

For any extension, the node identification digits plus the extension number uniquely identifies the extension from all others on the network. The extension can be reached by dialing the same string of digits from any node in the network.

For the use of the node identification code in analog networking, see Analog Networking in this section.

Conditions

The following conditions apply to this feature:

- The node identification code must match the ARS leading digits specified in the other nodes in the network to reach that node.
- Dial tone is not returned after the node identification code is dialed.
- Any device may access the node identification code.
- Within any node, it is not necessary to dial the Node Identification digits to access extensions within that node.

Programming

In CDE Form 02 (Feature Access Codes) assign an access code to Feature 34 (Node ID).

Operation

Dial the Node Identification code and then the extension number.

Non-Busy Extensions

Description

An extension with the non-busy extension feature enabled never appears busy to the system. If a new call is directed at a non busy extension that is already in a call, the system automatically overrides the existing call. After a warning tone, the new caller joins the conversation.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- The Override Security and Data Security features are ignored for the parties in the established call.
- The feature does not work for members of recording hunt groups.
- The feature is ignored when the console calls the set.
- The same conditions for override and key lines apply.
- This feature conflicts with any feature requiring a switch–hook flash.
- The non-busy extension can be in conversation with no more than four other parties. Additional callers receive busy tone.
- Only calls dialed directly to the set override; Recalls, Call Reroutes, Callbacks, ignore the feature.
- A call transferred to a non-busy extension does not override a conversation in progress.

The override is disallowed under the following conditions:

  - if the caller has a consultation hold in progress and any party in the call has a consultation hold in progress.
  - if a console is in the call already.
  - if the established call is a three party call that is on consultation hold.
  - if one of the parties in the call has the call on hold.
  - if the set is overriding another extension or had called another non-busy extension.

- The feature is not available to CO trunks when they originate.
- If the override is not permitted, then normal busy destination processing results.
- No connection checks are done when the caller is added into the call.
- Once on a call with a non-busy extension, regardless of how the call was established, no party in the call except the non-busy extension can flash out of the call or put it on hold.

Programming

Enable COS Option 243 (Non-Busy Extension) for the extension.

Operation

A new incoming call joins in to an established call automatically (with a burst of warning tone).

Numbering Plan Flexibility (Conflict Dialing)

Description

The numbering plan used within the system is completely flexible. The user may select any combination of 1-, 2-, 3-, 4- and 5-digit numbers.

Also see Conflict Dialing.

Conditions

Leading digits of extension numbers must not match feature access codes except codes 20 (Call Back Busy), and 31 (Executive Busy Override).

Programming

Assign the required extension numbers.

Operation

None.

Off-Hook Alarm to Display Sets

Description

This feature provides an immediate audible alarm and a visual indication to a display set when a set user goes offhook.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The conditions, programming, and operations for this feature are similar to Emergency Calls (911) - Detection and Reporting to Display Sets.

Any device that has COS Option 617, Immediate Off Hook Alarm enabled, will generate a lockout alarm as soon as the device goes offhook. Both the attendant console and the display set will be notified of the lockout alarm if the attendant console has COS Option 102 and/or COS Option 108 enabled and the display set has COS Option 616 enabled.

COS Option 617, Immediate Off Hook Alarm provides a lockout alarm as soon as the off-hook condition is detected. COS Option 227, Lockout Alarm Applies, provides a lockout alarm after the telephone has been off-hook for more than 45 seconds. If both of these options are enabled, the immediate alarm takes precedence.

Programming

An alarm key can be programmed via the Superkey.

- With Family 3 and Family 4 telephones (except the Symbol MiNET Wireless telephone), ALARM appears under PERSONAL KEYS, after SYSTEM PARK .
- With SUPERSET 4150 and SUPERSET 430 telephones, ALARM appears under FEATURE KEY, after SYSTEM PARK.

- With Family 5 telephones, Alarm appears under the Programmable Keys menu of the Settings key.

Operation

None.

Off-Hook Voice Announce

Description

The paging party can page over the speaker of the paged party if the paged party is engaged in a handset conversation.

Off-hook Voice Announce also occurs when the paging party uses a DSS key (set to Page in Form 09) to page a destination number that is in a talk state, in Do Not Disturb, or in some transitional state. See DSP/BLF Direct Paging.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

Paged sets can be Family 4 and Family 5 telephones (excluding the Symbol MiNET Wireless telephone) only.

Paged set cannot be operating in Group Listen mode.

Paged set cannot be in Do Not Disturb (DND) mode.

Paged set cannot be in a conference call.

Programming

Enable COS 615, Off Hook Voice Announce, in CDE Form 3 for the paging party.

Operation

Paging Party

To make a direct page:

- Press the Direct Page key.
- Dial the paged party number.
- Hear a burst of ringback tone.
- Speak.

Paged Party (Telephones with a MICROPHONE Key)

To respond to a direct page while off-hook:

- The display indicates the number and/or name of the paging party.
- The paged party hears the pager on the handsfree speaker while the paged party has a handset call.
- If the handsfree microphone is not latched prior to the off-hook voice announce call, the paged party can respond to the paging party by pressing the MICROPHONE key.

To disable the handsfree microphone during a call:

- Inform the caller, then press the MICROPHONE key (the LED turns off).

To re-enable the handsfree microphone during a call:

- Press the MICROPHONE key or lift the handset (the LED turns on).

Paged Party (Telephones with a MUTE Key)

To respond to a direct page while off-hook:

- The display indicates the number and/or name of the paging party.
- If the handsfree microphone is not latched prior to the off-hook voice announce call:
  - the Mute LED will turn on to indicate that a page is being received; the handsfree microphone will be off
  - the paged party can respond to the paging party by pressing the MUTE key; the Mute LED will turn off and the handsfree microphone will turn on.
- If the handsfree microphone is latched prior to the off-hook voice announce call:
  - the Mute LED will flash rapidly to indicate that a page is being received and that the handsfree microphone is on.

To disable the handsfree microphone during a call:

- Inform the caller, then press the MUTE key (the LED turns on).

To re-enable the handsfree microphone during a call:

- Press the MUTE key or lift the handset (the LED turns off).

Off-Premise Extension (OPS)

Description

Industry-standard telephones not in the immediate vicinity of the PBX can be directly connected to the PBX without the use of special trunks, using the 6-circuit OPS Line Card (Off-Premises). Refer to the Engineering Information section.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- If the extension is located more than 2 kilometers away, it may be necessary to add a compromise balance network into the circuit. COS Option 402 (Long Loop - Off-Premise Extensions Only) does this.
- An adjustment is made to the Loss/Gain settings for the line using COS Option 402 (Long Loop - Off-Premise Extensions Only).
- There is no equivalent feature for Mitel telephones.
- COS Option 402 (Long Loop - Off-Premises Extension Only) applies to OPS Line Card circuits only.

Programming

Program the OPS Line Card in CDE Form 01 (System Configuration).

If necessary, enable COS Option 402 (Long Loop - Off-Premise Extensions Only) for the extension.

Operation

None.

ONS Positive Disconnect

Description

ONS Positive Disconnect forces an ONS device to go on-hook when the far-end disconnects. A momentary electrical break on the ONS device forces the disconnection. Some FAX machines,answering machines, and key systems require a momentary electrical break to recognize that the other party has terminated the call.

Conditions

The following conditions apply to this feature:

- Requires the ONS/CLASS Line Card and Feature Level 2 enabled. (This feature is also supported on the ONS ports located on the Analog Main Board in the SX-200 ICP controller.)
- COS Option 506, ONS Positive Disconnect enables this feature and applies the disconnect time (the momentary electrical break). The break period is programmable from 1 to 5 seconds in 1 second steps. Setting the break period to 0, disables this feature.

Programming

In CDE Form 04, System Options and Timers, program level 2 in System Option 102, Feature Level.

In CDE Form 03, Class of Service Options, apply a numeral (1-5) to the status of COS Option 506, ONS Positive Disconnect. The numeric value represents the time in seconds for the electrical break. A value of 0 represents no electrical break; no positive disconnect.

Operation

None.

ONS Ring Groups

Description

The ONS Ring Groups feature provides the ability of multiple ONS telephones to ring when a master telephone number is called. Each ONS telephone in the ring group has its own extension number, therefore allowing the ONS telephone user to accept calls intended for that telephone plus answer calls directed to the master telephone number.

The ONS Ring Group has a master (a primary DNIC or ONS telephone) associated with multiple ONS telephones. The master can be a logical phone, that means, that the telephone may or may not be physically plugged into a card. But, the master telephone must be programmed to a port on a DNIC card or an ONS CLASS Line card that is physically plugged in.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Requires Feature Level 3 or greater in System Option 102. Systems with Feature Level 2 or less will not activate COS Option 274, ONS Ring Group Member.
- Distribute the ONS ring groups among the bays so you do not exceed the maximum of 32 ONS telephones ringing simultaneously per bay.
- All telephones in the ring group (master included) must have COS Option 274, ONS Ring Group Member enabled.
- An ONS Ring Group supports up to 8 ONS ring group members. The master can be a DNIC or an ONS circuit. If the master circuit is equipped with a telephone, the master will also ring.
- The master (primary) telephone can not be programmed as a Mitel 5201, SUPERSET 401 or SUPERSET 4001 telephone. The master telephone can also not have any key line appearances or have a multi-call line appearance as its preferred line.
- The functionality offered to secondary telephones in Guest Suites, such as Integrated Forwarding and Messaging, Integrated SMDR, Single Check-in and Check-out, Group wake-up and DND displays on the Attendant Console and Front Dest Terminal, are not supported with ONS Ring Groups.
- COS Option 274, ONS Ring Group Member, is mutually exclusive with COS Option 272, Guest Suite Extension, and COS Option 607, Telephone - Associated Modem Line. That means, you can only enable one of these three options at any time. Mitel recommends that you do NOT program a primary number with some ONS telephones as Guest Suite Extensions and some as ONS ring group members. If a primary number has some ONS telephones with COS 272, Guest Suite Extension, enabled and some with COS 274 enabled, the system will begin the allowable maximum count of 9 with the telephones that have COS 272 first.

Programming

In CDE Form 04, enable Feature Level 3 or greater in System Option 102.

For the Primary (Master) Telephone - ONS or DNIC

In CDE Form 09, program the primary (ONS or DNIC) telephone. Make the prime line the preferred line. Ensure that the primary telephone has no key-line appearance in the system and that the primary telephone does not have a multi-call line as its preferred line. Note that the primary telephone cannot be programmed as a SUPSERSET 401 telephone or a Mitel 5201 or SUPERSET 4001 telephone.

In CDE Form 03, enable Option 274, ONS Ring Group Member (ensure that COS Option 607, Telephone - Associated Modem Line, or COS Option 272 Guest Suite Extension is not enabled). Recommendations are that all sets in the group have the same COS, COR, and Tenant Group.

For ONS Ring Group Members (excluding the Master)

In CDE Form 09, program the ONS telephones for the ring group. Program the Primary DN of the Master in their ASSOC field.

In CDE Form 03, enable Option 274, ONS Ring Group Member (ensure that COS Option 607, Telephone - Associated Modem Line, or COS Option 272 Guest Suite Extension is not enabled). Recommendations are that all sets in the suite have the same COS, COR, and Tenant Group.

Operation

None.

Originate Only Extensions

Description

This feature allows an extension or dataset to originate calls. The extension can only receive calls that are forwarded from another extension. The system treats calls dialed to originate only extensions as illegal numbers.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- An extension with this COS option may receive calls via Call Forwarding (unless COS Option 234 (Never a Forwardee) is selected in its COS).
- Only calls directly dialed to an Originate Only extension are invalid.
- COS Option 235 (Originate Only) is mutually exclusive with COS Option 300 (Automatic Callback), COS Option 241 (Receive Only), and COS Option 243 (Non-busy Extension).

Programming

Enable COS Option 235 (Originate Only) for the extension.

Operation

None.

Overlap Outpulsing

Description

Overlap outpulsing occurs when the PBX begins dialing on a trunk before the user has dialed all digits in the destination's telephone number. By default, the ARS package outpulses digits as soon as the trunk seizure is acknowledged. This provides a shorter total dialing time, especially on non-DTMF trunks. This feature can be turned off, forcing the ARS package to collect all dialed digits before outpulsing the resulting digit string on the outgoing trunk. Refer to the Automatic Route Selection and Toll Control section for more information.

Conditions

None.

Programming

Enable System Option 26 (No Overlap Outpulsing) to disable the feature.

Operation

None.

Override (Intrude)

Description

This feature allows a user who encounters a busy extension to enter the conversation. Before override voice contact is established, the overriding party and both parties in the original conversation receive a warning tone. The tone is repeated at regular intervals while the overriding party is connected to the existing call. Mitel display telephones display the name and/or extension number of the overriding party.

For override of a telephone with Do Not Disturb set; see Do Not Disturb.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- If there is a call-me-back message outstanding between the overriding party and the overridden party, it is canceled. This is not done if the console overrides.
- The overriding party cannot manipulate the original connection in any way.
- If the overridden extension puts the call on consultation hold or goes on–hook, the intruding extension is dropped and receives reorder tone. Display telephones will read DISCONNECTED.
- If Mitel telephones involved in a conversation have multicall appearances, an override only occurs if all appearances are busy. Otherwise, the calling extension receives ringing and the idle multicall appearance rings if programmed to do so.
- The warning tone is given to all internal parties in the call being overridden.
- If the overridden call collapses to a two party call with just the overriding and the overridden party, then the override condition is cleared and a normal two-party call continues.
- If a key line is overridden, if the root extension of the line is in the call on the line then that telephone is overridden. If the extension is not in the call on the line, then the first “I-busy” Mitel telephone using one of the key line appearances is overridden.
- If a logical key line is overridden then the first “I-busy” Mitel telephone using one of the line appearances is overridden.
- An extension can only be overridden when talking to a party or in a conference call.
- If the caller is prevented from overriding when the override is attempted, the caller receives reorder tone.
- If the extension called becomes idle while the caller is listening to busy tone, when the caller attempts to override, an override is not done. Instead, a new call attempt is made to the extension as if the extension was dialed again.

The user cannot override:

- a telephone on hold,
- an industry-standard telephone, Mitel telephone, DISA trunk, or TIE trunk, with COS Option 216, Data Security or COS Option 238, Override Security in its COS,
- an extension with a call on consultation hold and the extension does not have the non-busy extension feature enabled (see Non-busy Extensions),
- a consultee,
- a busy hunt group,
- a logical multicall line,
- an industry-standard telephone key line that is holding a party (but is not holding the party on the key line),

- a conference call on consultation hold,
- a 4-party call with one party flashed out of the conference,
- a call that has another party in the call with the Non-busy Extension feature enabled,
- a call with the attendant in it,
- a call with five parties in it,
- an extension that is overriding another extension,
- a member of a recording hunt group.

Programming

To allow extensions in a COS to override, enable COS Option 500 (Override) in that COS.

Industry-standard telephones, Family 2, SUPERSET 410, and SUPERSET 3DN telephones need an access code to perform Override. Assign an access code to Feature 31 (Executive Busy Override). This access code must be a single digit.

To provide feature key activation of Override from Family 3 , Family 4 , and Family 5 telephones (except the Symbol MiNET Wireless telephone) program an OVERRIDE (INTRUDE) feature key. (See Feature Keys.)

Operation

Operation varies depending upon the type of set as described below.

Mitel 5201 IP Phones, and Industry-standard, SUPERSET 4001, and SUPERSET 401+ Telephones:

- While receiving busy tone, dial the Executive Busy Override access code. After the warning tone, you are connected to the call.

Family 3, Family 4, (except Symbol MiNET Wireless), SUPERSET 410, and SUPERSET 3DN Telephones:

- While receiving busy tone, press the INTRUDE feature key. After the warning tone, you are connected to the call.

Family 1 Telephones:

- While receiving busy tone, press the INTRUDE softkey; this softkey appears only when override is permitted.
- After a warning tone, you are connected to the call.

Family 5 Telephones:

- While receiving busy tone, press the Override softkey; this softkey appears only when override is permitted.
- After a warning tone, you are connected to the call.

Override Security

Description

This option provides an extension, DISA trunk, or Dial-in Tie trunk with security against Override (Intrude); see Attendant Busy Override and Override (Intrude).

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- The feature is ignored by the non-busy extension feature.
- The feature also provides an extension with security against having Do Not Disturb overridden; see Do Not Disturb.
- If COS Option 238 is enabled for a paged party busy on a call, no handsfree answer, off-hook voice announce, or whisper announce can be performed (pager receives busy tone).

Programming

Enable COS Option 238 (Override Security) for the extension or trunk.

Operation

None.

Paged Party Ring Tone

Description

Paged Party Ring Tone provides a paging tone instead of a burst of ringback tone to announce a page.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

- Paging tone applies to the pager and the paged party for Set-to-Set Pages, Group Pages, and All Set Pages.

Programming

Enable System Option 130, Paging Tone, in Form 4, System Options.

Operation

None.

Paging - PA

Description

An extension, Tie trunk, or DISA trunk can be permitted to access the paging equipment by dialing the required access code. Access may be restricted to any of the nine zones depending upon the access code dialed. If an extension tries to access busy paging equipment, busy tone is returned.

Paging equipment may be connected to an SX-200 ICP via the Paging port on the back of the unit, a Mitel 5485 IP Paging Unit, a Music-on-Hold/Paging module on the Universal Card or via a DNIC Music-on-Hold/Pager Unit (DMP). Up to nine paging zones, with separate or simultaneous access, can be provided.

Also see Attendant Paging Access.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- A short burst of tone from the handset warns the user when paging equipment is successfully accessed.
- Campon or Callback may not be activated on busy paging equipment.
- Paging amplifiers and loudspeakers are customer-provided equipment.
- The attendant can override any extension using paging equipment to make an announcement. The extension receives busy tone when it is removed from the pager.
- If System Option 03 (Single Paging Amplifier) is enabled, only one user at a time can access all paging in the system and the attendant cannot override the parties on the pager.
- Accessing the pager operates a relay on the pager module which may be used for controlling an external paging amplifier.
- Mitel telephone users cannot access paging while in handsfree mode.
- If access to all zones is attempted and at least 1 zone is busy then no paging is done and busy tone is returned.
- Access is to a single zone or simultaneously to all zones programmed in the device's COS.
- The PAGE softkey is not available to Family 1 telephones with the HEADSET OPERATION feature unless there is a consultation hold in progress.
- For other Mitel telephones in HEADSET OPERATION, the user must first enter dialing state (by pressing a line key, and getting dial tone) before the PA PAGE feature key will be presented.
- The 5485 IP Paging Unit can be accessed from stations.
- Access to the 5485 IP Pager cannot be restricted by Class of Service.

Programming

For the Mitel 5485 IP Paging Unit

- In Form 09, Desktop Desktop Device Assignments
  - Program the unit as a 5220 IP Phone and enter the unit's MAC address.
  - Assign the paging unit to its own paging group. Otherwise, the unit will be included when an extension user in the group does an All Set Page. The Default database has all phones in Group 1.

  Note: Ensure that no feature keys are programmed in the Expand Set and Expand PKM Subforms; otherwise the paging unit will fail to register. To avoid registration problems altogether, assign the unit to an IP PLID that has not been programmed.
- Program the Directed Page feature access code (48) in Form 2, or program a Direct Paging feature key in the Expand Set Subform for Form 09 on phones to allow them access to the paging unit.
- Power the paging unit on and off to initiate registration.

For the Paging port on the controller, the Music-on-Hold/Paging Module or DMP

- Program a paging device in CDE Form 18 (Miscellaneous System Ports).
- If the DNIC Music-on-Hold/Pager Unit is being used, program the unit as follows:
  - Program a Digital Line Card or DNIC Module in Form 1.
  - Program the DNIC device type as “DMP” (DNIC Music/Pager) in Form 9.
- Enable System Option 03, Single Paging Amplifier, if the PBX has only one paging amplifier output.

- Enable one or more of the following COS Options for the extension in CDE Form 3 (COS Options) as shown in the following table.

COS Option Number	Description
303	Paging Zone 1 Access
304	Paging Zone 2 Access
305	Paging Zone 3 Access
306	Paging Zone 4 Access
307	Paging Zone 5 Access
308	Paging Zone 6 Access
309	Paging Zone 7 Access
310	Paging Zone 8 Access
311	Paging Zone 9 Access
312	Paging Default (0 to 9) (0 Gives All Enabled Zones)

- For access to the default zone, assign an access code to Feature 12 (Paging Access to Default Zone) in CDE Form 02 (Feature Access Codes).
- Assign an access code to Feature 13 (Paging Access to Specific Zones) for access to zones other than the default zones.
- To provide convenient access to this feature from a Family 3, Family 4, SUPERSET 410 or SUPERSET 3DN telephone, program a PA PAGE feature key (see Feature Keys).

Operation

For the 5485 IP Paging Unit

To page using the 5485 IP Paging Unit, use the Directed Page feature.

For the Paging port on the controller, the Music-on-Hold/Paging Module or DMP

Family 1 and Industry-standard Telephones:

- Lift the handset.
- Dial the appropriate paging access code.
- If access code is for specific zone paging, dial the zone number (0 through 9).
- A tone is returned. Make announcement.
- Hang up.

Family 3, Family 4, SUPERSET 410, and SUPERSET 3DN Telephones:

Note: If a PA PAGE feature key is not programmed for the set, follow the Industry-standard telephone operation.

- Go off-hook.
- Press and hold down the PA PAGE feature key for default zone access.
- Wait for a short burst of tone.
- Make the required announcement.
- Hang up.

Family 1 Telephones:

- Go off-hook.
- Press and hold down the PAGER softkey for default zone access.
- Wait for a short burst of tone.
- Make the required announcement.
- Hang up.

Family 5 Telephones:

- Go off-hook.
- Press and hold down the Page softkey or the key you have programmed as Page.
- Wait for a short burst of tone.
- Make the required announcement.
- Hang up.

Symbol MiNET Wireless Phone:

1. Press the SEND key.
2. Do one of the following:
   - To page the default zone, dial 12.
   - To page a specific zone, dial 13 followed by the zone number (0-9).
3. Make the announcement.

Note: The Symbol MiNET Wireless Phone cannot receive pages.

Paging - Telephones

Description

Sets, stations, and the console may initiate pages to sets.

There are three different types of paging (for PA Paging - based on a Pager and paging zones, see Paging - PA in this document).

1. Directed Page - based on the telephone speaker or a Mitel 5485 IP Paging Unit.
2. Group Page (and Meet Me Answer) - based on the telephone speaker (with or without PA paging).
3. All Set Page - pages telephones (with or without PA paging).

The paging party and the paged party receive a single burst of tone to indicate that a page is about to occur. When the paged set is busy or if all sets in the paged group are busy, a short burst of busy tone precedes the voice announcement. Users who do not want to receive pages while on a call can have COS Option 238 (Override Security) enabled in their Class of Service.

It is possible to page across trunks to sets on another controller. To facilitate this, it is necessary to enable a number of COS options on the incoming trunks of controllers that will receive the pages. For details, see Programming.

Directed Page: Allows a party to page a specific telephone via its telephone speaker. The connection is one-way audio, and is terminated when the paging party hangs up. Another party attempting to call a set that is being paged in this manner will receive busy tone. The paged party can answer the page as if it were a normal incoming call to the Prime key. A Family 1, and Family 4 telephone user has the option of receiving or not receiving direct pages by enabling or disabling the functionality in the Superkey menu (provided COS Option 683 is disabled). Mitel 5201 IP Phones can only make direct pages to other phones; it cannot receive them.

Group Page: Allows a telephone to page all the telephones in their paging group or in another paging group simultaneously via their telephone speakers. The connection(s) are one-way audio to each telephone in the page group, and are terminated when the paging party hangs up. A telephone being paged in this manner may originate and receive calls; when this occurs, the paging on that telephone is terminated.

If all the telephones in a paging group become busy during a user's group paging announcement, the user hears two beeps. The user receives the two beeps as soon as the last idle telephone in the paging group becomes busy. The two beeps indicate to the user that nobody can hear the remainder of his or her announcement.

You can program up to 50 paging groups.

Attendant console and telephone users can combine the Group Page with an overhead page. Attendant console users can use the GROUP & PA softkey. Telephone users can perform the function with COS Option 686, Group Page Includes Overhead Paging enabled. The overhead paging zone is defined with the Paging Default COS Option from the COS of the paging group members selected.

Meet Me Answer: Allows a party to respond to a page. It does not apply to directed page calls. For a group page, the paging and paged parties must both be in the same group. If a party is involved in a call but hears the page from another telephone, they may put the current call on hold and respond to the page. The paged party must respond to the page within 15 minutes of the termination of the page; after this, the system cancels the page. A paged party should not try to respond after another page has been made to another party. If the paging party has not hung up, a Meet Me Answer response will be connected immediately as a normal 2-way conversation. If the paging party has hung up, the response will be treated like a normal telephone to telephone call. Once connected, the call is treated like a normal call, with all other existing features accessible. A paged party that responds to a page responds to the last page received.

All Set Page: Allows a party to page all telephones simultaneously via their telephone speakers. The connection(s) are one-way audio to each telephone, and are terminated when the paging party hangs up. A telephone being paged in this manner may originate and receive calls - when this occurs, the paging on that telephone is terminated. Telephones that make the All Set Page must have COS option 684 (Can Make All Set Page) enabled. Telephones that receive the All Set Page must have COS Option 685 (Can Receive All Set Page) enabled.

Attendant console and telephone users can combine the All Set Page with an overhead page. Attendant console users can use the ALL SET & PA softkey. Telephone users can perform the function with COS Option 687, All Set Page Includes Overhead Paging enabled. The overhead paging zone is defined with the Paging Zones programmed in COS Options 303-311 from the COS of the paging party.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Direct Paging is enabled for all of the telephones by default. Users of Family 1 and Family 4 telephones can disable Direct Paging by programming the set using the Superkey. Other telephone users must accept the Direct Paging feature.
- A directed page, group page, or all set page cannot be made to a set that has DND enabled.
- The 911 or lockout alarm overrides and terminates an All Set Page or a Group Page if sets or consoles have COS Option 616 (Alarm Monitor Point) enabled.
- A set user can initiate a group page for the paging group that the user is in, or for another paging group. Initiating a group page for a paging group other than its own, requires the system to have Feature Level 3 or greater in System Option 102, plus the user must dial the two digit paging group

number prior to the # key to specify the different paging group. If the two digit paging group number is the same as a two digit extension number, the system will interpret the page as a directed page to the extension.

- Paging telephones with an overhead page requires Feature Level 3 or greater in System Option 102. Systems with Feature Levels 2 or less, will not have the GROUP & PA and ALL SET & PA softkeys and COS Options 686 and 687 will be disabled.
- The GROUP & PA softkey on the attendant console pages the sets in the paging group and pages the overhead paging zones defined in COS Option 312, Paging Default, from the COS of the members of the paging group selected. The default zone can be one of 0 through to 9 with 0 enabling all zones. All the members of the paging group should be in the same COS group.
- The ALL SET & PA softkey on the attendant console pages all of the telephones plus the overhead paging zones defined in COS Options 303 to 311 from the COS belonging to the attendant console.
- The overhead paging zone (with a Group Page) from a telephone is defined with the COS Option 312, Paging Default (0-9), from the COS of the members of the paging group selected. All the members of the paging group should be in the same COS group.
- The overhead paging zone (with All Sets Page) from a telephone is defined with COS Options 303 to 311 from the COS belonging to the paging telephone.
- Only one group page may be performed to a page group at any one time.
- If a telephone was listening to background music, a group page will cause the music to be turned off. The music will be turned on again when the paging is terminated.
- COS Option 600 (Telephone - Auto-Answer) enables Family 1, Family 4, and SUPERSET 410 telephone users to respond handsfree to a directed page that is received while the set is idle.
- If the set has Auto Answer enabled or the Microphone is latched on, the page call is automatically answered (two way) if the set receives a directed page while the set is idle (including a page from the console).
- A Group Page cannot be initiated if an All Set Page is active.
- An All Set Page does not check for tenanting restrictions.
- Telephones that make the All Set Page must have COS option 684 (Can Make All Set Page) enabled. Telephones that receive the All Set Page must have COS Option 685 (Can Receive All Set Page) enabled.
- Enabling COS Option 238 (Override Security) prevents the if you don't want to get a page over the speakerphone when off-hook
- Dialing the Group Page - Meet Me Answer access code responds to the telephone that originated the last page, either Group Page or All Set Page.
- Family 3, Family 4, and SUPERSET 410 telephone users can respond handsfree to a directed page. Refer to Handsfree Answerback to a Directed Page for details.
- Mitel 5201 IP Phone can only make direct pages to other phones; it cannot receive them.
- SUPERSET 3DN and 4DN telephones can make but not receive an All Set Page.
- SUPERSET 3DN or SUPERSET 4DN telephone users can also respond handsfree to a directed page. Refer to Auto-Answer for Directed Page Calls for details.
- If the telephone has CO Line Preference set, dialing any digits will automatically select the CO line. To page from these sets, first select the prime line key, then dial the paging feature access code.

Programming

For group page and directed page:

- Program the Direct Paging feature access code (48) in Form 2, or program a Direct Paging feature key in the Expand Set Subform of CDE Form 09 (Desktop Device Assignments).

  For a direct page using a DSS key see DSP/BLF.

- Program a page group number in the PAGE field in CDE Form 9 (Desktop Device Assignments) for group page.
- Program Feature Level 3 or greater for System Option 102 in CDE Form 4. This feature level allows sets to specify a paging group for a group page and enables the PA paging with a group page. Systems with Feature Levels 2 or less, will not have the GROUP & PA softkey for the attendant consoles and COS Option 686 will be disabled.
- Assign COS Option 686, Group Page Includes Overhead Paging, in CDE Form 3 for the telephones that will be performing an overhead page with a group page.
- For the telephones in the paging groups, enable COS Option 312 - Paging Default and specify the default paging zone for the group members. Make the PA zone the same zone for all of the paging group members by assigning the members to the same COS.
- Program the SUPERSET telephone to accept or not to accept direct pages:

  For SUPERSET 430 and SUPERSET 4150 telephones, press Superkey and press the MORE softkey until the DIRECT PAGE softkey appears. Press the appropriate softkey (Disallow or Allow).

  For Family 4 and Family 5 (5330) telephones, press Superkey and press NO until CAN BE PAGED? appears. Press the appropriate softkey (TurnOn or TurnOff).

  For Family 5 (5340) telephones, press Superkey and press More until Direct Page appears. Press the appropriate softkey (TurnOn or TurnOff).
- For users who do not want to receive pages while they are on a call, enable COS Option 238 (Override Security) enabled in their Class of Service.
- For Group Page across trunks, assign COS Option 686 (Group Page Includes Overhead Page) to the incoming trunks on the controller(s) that will receive the pages. This allows a set on one controller to perform a combined PA Page and Group Page on another controller.

For All Set Page:

- Program a page group number in the PAGE field in CDE Form 09 (Desktop Device Assignments) for group page.
- Assign COS Option 684 (Key System - Can make All Set Page) in CDE Form 3 for each set that is to be allowed to perform an All Set Page (default is DISABLED).
- Assign COS Option 685 (Key System - Can receive All Set Page) in CDE Form 3 for each set that is to be allowed to receive an All Set Page (default is DISABLED).
- For the telephones that will be performing a PA Page with an All Set Page, enable COS Option 687, All Set Page Includes Overhead Paging.
- For telephones and attendant consoles performing a PA Page with an All Set Page, program COS Options 303 to 311 to define the overhead paging zones with the All Set Page.
- Program Feature Level 3 or greater for System Option 102 in CDE Form 4 to obtain the All Set Page with a PA Page functionality. Systems with Feature Levels 2 or less, will not have the ALL SET & PA softkey for the attendant console and COS Option 687 will be disabled.
- To enable All Set Page across trunks, program the following on the controller(s) that will receive pages:
  - Assign COS Option 684 (Key System - Can Make All Set Page) to the incoming trunks.
  - Assign COS Option 685 (Key System - Can Receive All Set Page) to each set allowed to receive an All Set Page.
  - Assign COS Option 687 (All Set Page Includes Overhead Page) to the incoming trunks. This allows a set on one controller to perform a combined PA Page and All Set Page on another controller.

For Meet Me Answer:

- Assign an access code to Feature Access Code 49 (Key System - Group Page - Meet Me Answer) in Form 2 (Feature Access Codes).

Operation

To Initiate a Directed Page:

Family 2 Telephones and stations:

- Lift the handset.
- Dial the Key System - Direct Paging feature access code.
- Dial the extension number.

Note: If the page is across trunks, you may be required to enter ARS digits before dialing the extension number.

- Broadcast the page message.

Family 1, Family 3, Family 4 and SUPERSET 410 Telephones:

- Dial the Key System - Direct Paging feature access code, or press the DIRECT PAGE feature key.
- Dial the extension number, or press the DSS key of the party to be paged.

Note: If the page is across trunks, you may be required to enter ARS digits before dialing the extension number.

- Broadcast the page message.

Family 5 Telephones:

- Lift the handset.
- Press the Direct Paging feature key you have programmed or dial the appropriate feature access code.
- Dial the extension number.
- Speak to the dialed party after the tone.

To Respond to a Directed Page

Mitel 5010 and 5215 IP Phones, and SUPERSET 4001 and SUPERSET 4015 Telephones:

- Lift the handset.

Family 4 and SUPERSET 410 Telephones:

- Lift the handset, press the flashing line key, or press the SPEAKER key.

Family 5 Telephones:

- Lift the handset or press the MUTE key to answer in Handsfree mode.

To initiate a Group Page:

Family 2 Telephones and stations:

- Lift the handset.
- Dial the Key System - Direct Paging feature access code.
- To page your own paging group, dial #.
  To page another paging group on your controller, dial the two-digit paging group number, and then dial #.
  To page another paging group on a different controller, dial the ARS digit(s) to connect to the remote controller, dial the Key System - Direct Paging feature access code, dial 9, and then dial the two-digit paging group number.
- Broadcast the page message.

Family 1 (except the SUPERSET 4DN), Family 3, Family 4, Family 5 and **SUPERSET 410 Telephones:**

- Dial the Key System - Direct Paging feature access code, or press the DIRECT PAGE feature key.

- To page your own paging group, dial #.
  To page another paging group on your controller, dial the two-digit paging group number, and then dial #.
  To page another paging group on a different controller, dial the ARS digit(s) to connect to the remote controller, dial the Key System - Direct Paging feature access code, dial 9, and then dial the two-digit paging group number.
- Broadcast the page message.

To answer a Group Page:
- Lift the handset.
- Dial the Key System - Group Page - Meet Me Answer access code.

Note: 5201 IP Phone users do not receive pages.

To initiate an All Set Page:

Family 2 Telephones and stations:

To page all sets on your own controller:
- Lift the handset.
- Dial the Key System - Direct Paging feature access code.
- Dial *.
- Broadcast the page message.

To page all sets on another controller:
- Lift the handset.
- Dial the ARS digit(s) to connect to the other controller.
- Dial the Key System - Direct Paging feature access code.
- Dial 0.
- Broadcast the page message.

Family 3, Family 4, Family 5, and SUPERSET 410 telephones:

To page all sets on your own controller:

- Lift the handset.
- Dial the Key System - Direct Paging feature access code, or press the DIRECT PAGE feature key.
- Dial *.
- Broadcast the page message.

To page all sets on another controller:
- Lift the handset.
- Dial the ARS digit(s) to connect to the other controller.
- Dial the Key System - Direct Paging feature access code, or press the DIRECT PAGE feature key.
- Dial 0.
- Broadcast the page message.

To answer an All Set Page:
- Lift the handset.
- Dial the Key System - Group Page Meet Me Answer access code.

To initiate a Direct, Group, or All Set Page from a Console:

- Press the Set Page key.
- For a Directed page, dial the extension number
  For a Group page, press the Group Page softkey, enter a 2-digit paging group number.
  For an All Set page, press the All Set Page softkey.
  For a Group page and a PA page, press the Group & PA softkey, and enter the paging group number.
  For an All Set page and a PA page, press the All Set & PA softkey.
- Broadcast the page message.

Note: If the Set Page key on the console is programmed as a macro key, then dial the Direct Paging feature access code. The paging softkeys will appear and then you can press the Group Page softkey or the All Set Page softkeyPaging - PA and Telephones.

The Attendant can also page other sets on a remote controller:

To initiate a Directed Page from a Console to a set on another controller:

- Dial the Key System - Direct Paging feature access code.
- Dial the ARS digit(s) to connect to the other controller (if required).
- Dial the extension number.

To initiate a Group Page from a Console to a paging group on another controller:

- Dial the ARS digit(s) to connect to the remote controller.
- Dial 9
- Dial the two-digit paging group number.

To initiate an All Sets Page from a Console to all sets on another controller:

- Dial the ARS digit(s) to connect to the other controller.
- Dial the Key System - Direct Paging feature access code.
- Dial 0.

Park and Page

Description

With Park and Page, telephone users can park a call and initiate a page all in one step. Park and Page options include

- Park and Page Set: After a call is parked, the system performs an All Set Page. The system also performs a PA Page if COS Option 687 is enabled.
- Park and Page Group: After a call is parked, the system pages the user's Paging Group. The system also performs a PA Page if COS Option 686 is enabled.
- Park and Page: After a call is parked, the called party must enter the extension or Page Group number.
- Park and PA Page: After a call is parked, the called party must enter a Paging Zone number (0-9).

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

- Use of this feature requires Feature Level 5 in Form 04, System Option and Timers, enable Feature Level 5 in System Option 102.

- For conditions on Page Groups and Page Group/All Set Page to include PA Page, see Paging - Telephones.
- For conditions on Call System Orbits, see Call Park System Orbit.

Programming

The four Park and Page features can be programmed through CDE Forms or from the phone.

CDE Forms - Phones

Program any or all of the four Park and Page features to keys on phones using Form 9, Desktop Device Assignment.

Assign access codes to the Park and Page features in Form 2, Feature Access Codes.

CDE Forms - Attendant Consoles

Enable System Option 129 in Form 4 to replace the SYSTEM PARK softkey with the PARK & PAGE softkey.

SUPERKEY - Family 1 telephones:

To program any or all of the four Park and Page options from the phone

- Press SUPERKEY.
- Press **More** until "Feature Keys" appears, and then press an unused line key.
- Press **Change**.
- Press **More** until the desired option appears (i.e. Park & Pg Sets, Park & Pg Grp, Park & Page, Park & PA Page).

SUPERKEY - Family 3, Family 4, and SUPERSET 410 telephones:

To program any or all of the four Park and Page options from the phone

- Press SUPERKEY.
- Press an unused line key, and then press **Change**.
- Press **No** until the desired option appears (i.e. Park & Page Sets, Park & Page Grp, Park & Page, Park & PA Page).
- Press **Yes** to select the desired option.

Family 5 Telephones:

To program any or all of the four Park and Page options from the phone:

5330 Phones:

- Press the key you have programmed as **Superkey**.
- Press **No** until **Personal Keys?** appears and then press **Yes**.
- Use the page navigation arrows to move to the appropriate Park and Page feature and the press the corresponding key to select.

5340 Phones

- Press the key you have programmed as **Superkey**.
- Press More until Feature Keys appears and then press **Feature Keys**.
- Use the page navigation arrows to move to the appropriate Park and Page feature and then press the corresponding key to select.

Operation

Family 1, (except SUPERSET 4DN), Family 4, Family 5, and SUPSERSET 410 telephones:

To park a call and page using a programmed feature key:

- Answer or make a call.
- Press the desired paging key (i.e. Park & Page Sets, Park & Page Grp, Park & Page, Park & PA Page).

To "Park and Page" using feature access codes:

- To page an extension number, press the Trans/Conf key, enter the Call Park and Page - Telephone FAC, and then enter the extension.
- To page all phones and the PA, press the Trans/Conf key, enter the Call Park and Page - Telephone FAC, and then press *****. Note that COS Option 687, All Set Page Includes Overhead Paging, must be first enabled.
- To page all phones in a group and the PA, press the Trans/Conf key, enter the Call Park and Page - Telephone FAC, and then press **#**. Note that COS Option 686, Group Page Includes Overhead Paging, must first be enabled.
- To page all phones in a specified group and the PA, press the Trans/Conf key, enter the Call Park and Page - Telephone FAC, and then enter the two-digit Page Group number and #. Note that COS Option 686 must first be enabled.
- To park a call in orbit and perform a PA Page to any or all nine Paging Zones, press the Trans/Conf key, enter the Call Park and Page - PA FAC, and then enter the Paging Zone number (0 for all zones; 1-9 for specific zones).

Note: To Park and Page call using speed call keys, press TRANS/CONF to get dial tone before pressing a speed call key programmed with the Park and Page access codes plus the digits for the required paging option -- for example, an extension number to page a single phone or * to page all phones plus PA paging.

Superconsole 1000 Attendant Console or 5540 IP Console:

To park a call and page:

- Press **Park & Page**.
- Do one of the following:
  - Enter the extension number to page.
  - Select one of the displayed paging options: **All PA Zones**, **Group Page**, **All Set Page**, **PA Zone Num**, **Group & PA**, **All Set & PA**.
- For zone paging, enter the two-digit PA Zone number.

Parallel Connection of Industry-standard Telephones (Party Line)

Description

A maximum of three industry-standard telephones equipped with bells can be connected (hard-wired) together on one ONS line.

Conditions

The following conditions apply to this feature:

- When one industry-standard telephone is in use, if any other of the industry-standard telephones goes off–hook, it then joins in the call (without proper conferencing facilities).
- All of the industry-standard telephones ring when the extension number is called.
- If the telephones are equipped with Message Waiting lamps, further restrictions may apply.

Programming

None.

Operation

None.

PC (2nd) Port

Description

If your phone is equipped with a second port, you can plug a PC into it.

Programming

- Enable System Option 131 (PC (2nd) Port on IP Phone).
- Enable COS Option 280 (PC (2nd) Port on IP Phone) for the required extension.

Personal Speed Call

Description

This feature allows the user to program and access up to six personal speed call numbers. The telephone user enters the numbers at the telephone - they may then be accessed via an access code, followed by an index number. These personal speed call numbers may only be accessed from the telephone on which they were entered.

Users can program pauses in speed call numbers. Typically, pauses are inserted in calls to devices (such as, voice mail systems, FAX machines, pagers, and long distance carriers). The pause allows the call to be answered before the system outpulses further digits to the device. Note that, in addition to this feature, users also have access to the Abbreviated Dial feature - see Abbreviated Dial.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Personal speed call numbers may be dialed at any time on any line, provided that they are the first digits dialed, and that an ACD agent or supervisor is not currently logged in at the set.
- You can store up to six numbers on any Mitel telephone.
- The maximum length of a stored number is 25 digits.

Programming

Assign feature access codes 51 (Key System - Store Personal Speed Call) and code 52 (Key System - Retrieve Personal Speed Call) in Form 02 (Feature Access Codes).

Note: The Retrieve Personal Speed Call access code must begin with an asterisk (*).

Operation

To store a personal speed call number:

- Lift the handset.
- Dial the Store Personal Speed Call access code.
- Dial the desired index number, 1 to 6.
- Dial the number to be stored.

  You can insert pauses in the number. Press the red HOLD key or the Insert Pause softkey to insert a 1-second pause.
- Hang up.

To dial a personal speed call number:

- Dial the Retrieve Personal Speed Call access code.
- Dial the index number for the desired stored number.

Symbol MiNET Wireless Phone:

To store a Personal Speed Call number:

1. Press the SEND key.
2, Enter "Personal Speed Call" feature access code, 51.
3. Enter an index number 1, 2 or 3 (this is the location on the Speed Call list).

Note: The index number does not appear on the screen.

4. Enter the number to be stored.
5. Press the END key to save.

To dial a stored Personal Speed Call number:

1. Press the SEND key.
2. Enter "Personal Speed Call" feature access code, 52.
3. Enter the location of the stored number (a location/index number 1, 2, or 3).

Note: The index number does not appear on the screen.

Phonebook

Phonebook provides access to the voice mail directory which allows callers to reach an extension by entering the user's first or last name rather than their extension or mailbox number. The voice mail system can be configured to search either on first or last names (but not both at the same time).

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

- Phonebook cannot be used if Speak@Ease is enabled and vice versa.
- Phonebook is a purchasable option.
- Only extensions and consoles that have a voice mailbox can be dialed using Phonebook.
- When using Phonebook to transfer a call, the user, after entering the destination party's name and hearing the prompt "you are being transferred to <name>" must immediately hang up. Waiting too long will cause the transfer to fail and the user to reconnect to the party on softhold.

Programming

- Select System Option 128 (Phonebook) in Form 4.
- Program the voice mail hunt group number to the "Phonebook Number For This Tenant" field in Form 19.
- Program a Phonebook feature key in Form 9 on phones that do not have softkeys.
- Assign an access code to Feature 65 in Form 02 to allow DNIC-based single line phones to access the Phonebook. All types of phones can also use the access code.
- Ensure that a NAME is entered for each mailbox owner in Form 50. Only mailboxes with names are listed in the Phonebook.
- In Form 49, set the Directory Voice Prompt option to search the Phonebook by first or last name.

Note: Phonebook searches will only work if the first and last name entered in Form 50 are in the order specified by the Directory Voice Prompt option in Form 49. For example, if the option is set to LAST NAME, then the name "Tom Jones' should be entered as Jones Tom in Form 50. However, you may want to enter the name in Form 9 as first name, last name (Tom Jones) because it looks better on the called party's display. And at the same time, you may want Phonebook searching by last name. In this case, set "Directory Voice Prompt" in Form 49 to LAST NAME, enter "Tom Jones" as the device name in Form 9 Tom Jones, and then name the mailbox in Form 50 "Jones Tom."

Operation

To access the Phonebook from a set or console:

Access to the Phonebook from a set is by softkey (Family 4 and Family 5 telephones only), feature key, or feature access code. Access from the console is by the key to the right of the TRUNK GROUP key. Pressing the key or dialing the access code places a call to the voice mail system which responds by prompting the caller to enter the name or extension number of the person they wish to reach. Additional prompts guide the caller to enter the person's first or last name as determined by the "Directory Voice Prompt" option in Form 49.

To access the Phonebook from the Auto Attendant:

Note: The Phonebook feature is accessed by dialing 9 from the Auto Attendant. To prevent confusion between dialing a mailbox that starts with 9 and dialing 9 for the Phonebook, an inter-digit timer of 4 seconds is started when the first digit entered is 9. If no other digits are entered during these few seconds, the user is transferred to the Phonebook.

1. Access the Auto Attendant.
2. Dial 9 and wait a few seconds.
3. At the prompt, enter the name using the keypad (press 2 for A, B or C, press 3 for D, E or F, etc.). You can enter up to 6 letters.
4. If a unique match is found, the Auto Attendant transfers you to the user.
   If more than one user name start with the letters entered, the Auto Attendant lists the number of matches, then reads each match's name or number followed by the keypad number to enter to be transferred to that person.

Phone Twinning

Description

Phone twinning provides concurrent ringing and message waiting indication on as many as five phones. A typical use of twinning is for users who have both a wireline (desk) phone and a Symbol MiNET Wireless phone can use twinning to receive calls and message waiting indications on either device. Twinning may be used for a number of different types of phones and also benefits teleworkers who have a phone at the corporate office and a remote IP phone at home. Users can turn off/on the ringer on the phone that they are not using as well as enable/disable the SUPERKEY to prevent tampering with the phone's programming.

Note: Phone Twinning is not compatible with Automatic Call Distribution (ACD) or Guest Suite Services.

Feature Availability

Click here to see a list of the phones that can use this feature.

Note: The Superset 4001 set can be used as a Secondary phone but can not be used as a Primary.

Conditions

Phone Twinning is subject to the same conditions as Suite Services with the following exceptions or additions:

- Phone Twinning requires Feature Level 3 or greater in System Option 102.
- Phone Twinning does not require System Option 84, Multiple Guest Suite Phones.
- SUPERKEY functionality is available on primary and secondary phones (Suite Services disables SUPERKEY on primary and secondary DNIC phones.)
- Call forwarding cannot be programmed on a secondary phone via SUPERKEY.
- Call forwarding for the primary phone can be programmed from the secondary phone by using feature access codes.
- Twin Phone COS Option 276 and the Guest Suite Extension COS Option 272 are incompatible and must not be both enabled in the same Class of Service.
- The Superset 4001 set can be used as a Secondary phone but can not be used as a Primary.

Programming

Program the Disable Twin Phone access code (61) in Form 02.

For BOTH the Primary Telephone and Secondary Telephone(s)

In CDE Form 03, enable Option 276, Twin Phone, and then assign the COS to the primary and secondary telephone(s)

For the Primary Telephone

In CDE Form 09, program the primary telephone. Ensure that the primary telephone has no key-line appearance in the system.

For DNIC Secondary Telephones

In CDE Form 09, Expand Set Subform, assign a multicall line key to the primary DN and make the multiline key the preferred line (LINE PREF softkey) for incoming and outgoing calls. This is required to light the message waiting lamp on the secondary phones.

For ONS Secondary Telephones

In CDE Form 09, program the Primary DN (i.e., the DN of the Symbol phone) in the ASSOC field.

Operation

To enable/disable the ringer and SUPERKEY on the primary or secondary phone:

1. Lift the handset.
2. Dial the Disable Twin Phone access code.
3. Dial 1 to disable or 2 to enable.
4. Dial the directory number of the phone whose ringer/SUPERKEY you want to disable or enable.

Pickup Groups

Description

Extensions can be programmed into pickup groups, permitting users to answer calls on any other extension within their particular group; see Pickup - Local and Directed.

Members go off hook with their telephone to identify the ringing extension number within their pickup group, hear a dial tone, and then press a Pickup softkey or Pickup feature key if they choose to answer the call. Large display SUPERSET telephones also identify the name and/or number of the caller. Small display SUPERSET telephones present a Caller softkey (to toggle back and forth to the name or number of the caller) and a Called softkey (to return to the display that identifies the ringing set).

Conditions

The following conditions apply to this feature:

- A maximum of 50 Pickup groups are permitted per system, with a maximum of 50 extensions permitted per group.
- Calls are picked up in the group in order of the extensions in the pickup group. The search for a ringing extension starts with the first extension in the group and ends with the last in the group.
- An extension can only be in one pickup group at a time.
- The displaying of information (ringing extension and caller ID) when a member goes off hook at the time of a ringing extension within a pickup group, requires Feature Level 3 or greater in System Option 102.
- When a pickup group member goes off hook with a display set while the telephone of another member of the same group is ringing, dial tone is returned and a Pickup softkey is presented. Therefore, the member that goes off hook can either make a call or pickup the ringing call.
- Members going off hook with SUPERSET 430 and SUPERSET 4150 telephones will see the name and/or number of the caller on the display, if another extension is ringing. Mitel 5020, 5220, 5224, and 5324 IP Phones, and SUPERSET 420, SUPERSET 4025, and SUPERSET 4125 telephones present a Caller softkey that enables the user to toggle between the name and number of the caller.
- The display information regarding the ringing extension and the caller ID remains on the set until the set user performs a function with the set. If the call is answered by another set or if the caller hangs up, the information will remain on the display. This action could result in the user answering a different call than what is displayed.
- If the sets of two or more pickup group members ring at the same time, the caller ID for the first member will display and the user may answer a different call then what is displayed.

Programming

Assign the desired extensions to the appropriate pickup groups via CDE Form 10 (Pickup Groups).

Assign Feature Level 3 or greater to System Option 102, in CDE Form 04 (System Options and Timers).

Operation

See Pickup - Local And Directed.

Pickup - Local and Directed

Description

A telephone can be assigned to a pickup group, and can answer any ringing telephone within that group. This is Local Pickup.

Directed Call Pickup allows an extension user to answer any ringing telephone within the system.

See Pickup Groups.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Local Pickup operates only for extensions within the same pickup group.
- For ringing Mitel telephones, the ringing line is scanned for from the prime line up the Mitel telephone until a line that can be picked up is found.
- For Directed Call Pickup, the scan is done on the extension specified and not on the line appearances of that extension (unless the Mitel telephone has multicall line appearances of itself).
- For pickup groups, the scan for a ringing extension always starts with the first pickup group member (terminal scanning) each time a pickup is attempted. A ringing extension is determined to be ringing based on the scan on the extension for a ringing line that can be picked up.
- Reorder tone is returned if there is no call to pick up (or it is already picked up).
- The displaying of the caller ID on a ringing extension within a Pickup group is dependant on Feature Level 3 for System Option 102. See Pickup Groups.
- The following call types cannot be picked up:
  - callbacks,
  - wakeup/reminders,
  - calls to members of recording hunt groups,
  - calls ringing back an extension (Station Transfer Security),
  - Direct Trunk Select or Private Trunk lines,
  - silently ringing line appearances or delay ring line appearances that have not begun to ring,
  - console calls when a consultation hold is in progress,
  - callers that cannot connect to the party picking up the call (see Device Interconnection Control),
  - the party picking up the call and the ringing extension cannot connect,
  - the Mitel telephone is Auto-Answering the call.

Programming

Program Pickup Groups in CDE Form 10 (Pickup Groups).

Assign an access code in Form 02 to Feature 08 (Dial Call Pickup).

Assign an access code in Form 02 to Feature 09 (Directed Call Pickup).

To provide convenient access to this feature from a Family 3, SUPERSET 410, or SUPERSET 3DN telephone, program a PICKUP feature key (see Feature Keys).

To permit sets in a COS to pick up calls outside their Pickup Group, enable, in Form 03, COS Option 218 (Directed Call Pickup) in that COS.

Operation

Local Pickup - Family 2 Telephones:

When a telephone in the same pickup group rings:

- Lift the handset - dial tone is returned.
- Dial the Call Pickup code - the call is connected.

Local Pickup - Family 3, SUPERSET 410, and SUPERSET 3DN Telephones:

When a telephone in the same pickup group rings:

- Lift the handset - dial tone returned.
- Press the PICKUP feature key - the call is connected.

Local Pickup - Family 4 and Family 5 Telephones:

When a telephone in the same pickup group rings

- Lift the handset - dial tone returned.

  The display will show the extension that is ringing and a Caller softkey will appear if the name or number of the caller is known. The user can identify the caller by pressing the Caller softkey. After pressing the Caller softkey, the softkey changes to Called. Pressing Called returns the user to the display that identifies the ringing extension.
- Press the PICKUP softkey - the call is connected.

Local Pickup - Symbol MiNET Wireless Phone:

To answer a call that is ringing at another extension in your Pickup Group:

1. Press the SEND key.
2. Dial 08.

To answer a call that is ringing at a extension not in your Pickup Group:

1. Press the SEND key.
2. Dial 09.
3. Dial the number of the ringing extension.

Local Pickup - Family 1 Telephones:

When a telephone in the same pickup group rings:

- Lift the handset - dial tone returned.

  The display will show the extension that is ringing and will identify the caller if the information is available.
- Press the PICKUP CALL softkey - the call is connected.

Directed Pickup - All Telephone Types

When a telephone outside the pickup group rings:

- Go off–hook - dial tone is returned.
- Dial the Directed Call Pickup code.
- Dial the extension number of the ringing telephone - the call is connected.

PRI Support

Description

You must purchase ISDN services from the ISDN service provider. In addition, you must purchase software options from Mitel that are related to the desired ISDN services. The activation of these software options permit the ISDN service provider to interface with the matching option on the PRI card or module.

PRI Card in Peripheral Cabinet

The PRI card provides the SX-200 ICP with two links of Primary Rate Access (PRA) to the ISDN service provider. This PRI connection supports the simultaneous transmission of voice. Each channel is capable of transmitting voice calls at 64 kbps. With a Release 6.0 PRI card, the PRI connection from PRI card to PRI card supports voice and data on the same card. By interfacing the SX-200 switch with the ISDN service provider, it allows SX-200 users to access ISDN services, such as Direct Dial In (DDI), Calling Line Identification (CLID), and Call By Call Service Selection (CBC).

Embedded PRI in Controller

T1/E1 modules can be installed in a controller to provide direct PRA to the ISDN service provider. The Dual Link T1/E1 Framer module is available for the AX and MX and has two links, each of which can be programmed as either a T1 or PRI interface. The T1/E1 Combo module is available for the AX and CX/CXi and has a single link that can be programmed as either a T1 or PRI interface. Each link provides 23 channels. The MX supports a maximum of 92 channels (2 modules x 2 links x 23 channels), the CX supports a maximum of 23 channels (1 module x 1 link x 23 channels), whereas AX supports a maximum of 46 channels (1 module x 2 links x 23 channels).

Note: E1 is not supported on the SX-200 ICP.

PRI cards and Embedded PRI support the following ISDN network services:

- **Calling Line Identification (CLID) -** The ISDN network presents the telephone number of the calling party to the user's telephone.
- **Calling Party Number (CPN)** - This number substitutes the calling station number on outgoing calls for purposes of network identification and call back.
- **Calling Name ID (CNID)** - This ID is the incoming call name delivery per NA DMS100 and DMS250 custom specification or National ISDN-3 (NI3).This Calling Name feature is an option that must be enabled at the Central Office. The SX-200 only supports the inbound calling name.
- **Calling Line Identification Presentation (CLIP**) - The Calling Party Number can be provided to the ISDN Network for outgoing calls or provided to the PRI card from the ISDN Network for incoming calls. This information is passed onto the system and can be used for database applications such as screen pops and for inclusion in SMDR records.
- **Calling Line Identification Restriction (CLIR) -** This feature allows users to prevent their telephone number and their calling name ID from being presented to the called party.
- **Partial PRI Links -** The SX-200 PRI card will support COs that provide this feature.
- **Direct Dial-In (DDI) -** DDI is an ISDN option that allows direct access to a line behind a PBX through a unique directory number. This allows the dialed digits of an incoming ISDN call to be presented to the PBX. DDI also allows an external caller to call a specific extension without going through the attendant. All ISDN trunks are treated as Dial-In trunks; the CO always sends digits to the PBX.
- **Call By Call Service Selection (CBC) -** (PRI card and NSU only) This feature allows telephone users to select the ISDN network services that they wish to use on a per call basis.
- **DID Calling Party Number Forwarding -** Outgoing CPN delivers the calling party's DID number to the Network when the call has been identified as a call from a device with an associated DID number instead of delivering the main directory number associated with the PBX.
- **Equal Access to Interexchange Carriers -** The PBX provides a carrier access code which identifies to the Central Office which Interexchange Carrier is to receive the call. The PBX outpulses a digit string which includes a carrier access code, followed by an identification number, followed by the called number.
- **Min/Max** - (PRI card and NSU only) This feature allows a customer to control incoming and outgoing call traffic. Minimums are assigned to ensure that a particular type of call (such as INWATS) always

has a set number of lines available. Maximums are assigned to limit certain types of calls, i.e., OUTWATS. This ensures that resources are not used up by a single type of call. Different Min/Max databases can be created for different times of the day or for special occasions such as telethons or infomercials.

- **Auto Min/Max -** (PRI card and NSU only) This feature provides user programmable time-of-day automatic control of Min/Max parameters.
- **NFAS** (Non-Facility Associated signaling) - (PRI card and NSU only) NFAS allows you to use a single D-channel to handle the signaling requirements for a group of PRI links that all use the same Protocol. This feature eliminates the need to purchase a D-channel for each link. NFAS is mainly for North America.
- **D-Channel Backup -** (PRI card and NSU only) D-channel Backup provides an alternate D-channel for calls related to NFAS. If the active D-channel fails, the system switches to the backup D-channel to support call processing. This functionality is mainly for North America. NFAS is required in order to program D-channel Backup.
- **QSIG -** (PRI card and NSU only) QSIG is a protocol that allows you to connect PBXs together to form a private network. Because QSIG is an international standard, the private network can include PBXs from different vendors. QSIG supports incoming calls and the incoming Calling Name function for the SX-200 T1 card.
- **Remote LAN Access** - This feature provides LAN access to the wide area network (WAN) for both incoming and outgoing calls through LAN servers (routers or bridges).

Note: Embedded PRI does not support Min/Max, NFAS, D-Channel Backup, QSIG, or Call By Call Service Selection (CBC). These services require a PRI card or NSU.

The PRI card interfaces the following SX-200 peripheral devices with the ISDN network:

- Attendant consoles
- Mitel telephones
- Industry-standard telephones.

Conditions

The following conditions apply to this feature:

- In a main control cabinet, the PRI card fits into slots 10 or 11 and does not require a FIM II or CIM on the PRI card. The dip switch (S1 switch) settings on the PRI card must be set to 1 closed and 2 open.

Note: Only one PRI card can be installed in a control cabinet. This restriction allows the ISDN bay to be within the power budget. Two PRI cards can be installed in a peripheral cabinet.

- In a peripheral cabinet, the PRI card fits into slots 10 or 11 and requires a FIM II or CIM on the PRI card to connect to the FIM or CIM in the main control cabinet. The dip switch (S1 switch) settings on the PRI card must be set to 1 and 2 closed.
- The system identifies the PRI card as a peripheral bay, even though you install the PRI card in a cabinet already defined as a bay.
- The PRI card is not included in the high power card count since it is regarded as its own peripheral bay.
- In a main control cabinet, the PRI card in slot 10 is Bay 2; a PRI card in slot 11 is Bay 4. A PRI card can never be in the IP bay (Bay 1).
- In a peripheral cabinet, the PRI card will have the bay number that corresponds to the numbered position of the FIM termination at the main control cabinet. Therefore, a PRI card can be a Bay 2, 3, 4, 5, 6, or 7 depending on the FIM link that is being used on the Control FIM.
- The PRI Card supports up to 2 T1 links. Only one dual T1 module is allowed on the PRI card.
- Installation of PRI cards is Slot 10 or 11 does not restrict the peripheral interface cards in slots five and six.

- Purchasable FOR options enable the following in the PRI card(s): QSIG (option 86), NFAS (option 91), D -Channel Backup (option 92), Remote LAN Access (option 93), Min/Max (option 94), Auto Min/Max (option 95). Option 91 to 95 are downloaded to each PRI card installed. These FOR options serve all the PRI cards in the system. If you change any of these options, you must reset the PRI card to enable them, and you may need to download a new IMAT database on to the PRI card(s).

Note: Min/Max is required to program Auto Min/Max and NFAS is required to program D-Channel Backup.

- You must purchase the FOR option, Number of Links (option 96). This option enables the number of T1 links on the PRI card, NSU, BCC III, and embedded T1/E1 combo module.

Notes:

1 If you purchase a number of PRI card T1 links (System Option 96) less than the number of PRI card T1 links that you install and program, the system will only allow the number of links purchased in System Option 96. Mitel cannot reliably predict which PRI card T1 links the system will use after a reset. That means, if 6 PRI card links are installed and programmed, and 4 PRI card links are purchased, Mitel cannot predict which 4 of the 6 will come up after a system reset.

2 If you purchase a number of PRI card T1 links (System Option 96) greater than the number of PRI card T1 links that you install and program, the PRI card T1 links programmed will remain consistent.

- The SX-200 system supports the Bearer Capabilities: Speech and 3.1 kHz audio.
- The system uses CLID and DDI digits that are received from the ISDN Node the same way as it uses ANI and DNIS digits that are received from a peripheral node -- the only exception being that COS Option 801 (Incoming Trunk Call Rotary) must be enabled for ISDN trunks, and disabled for non-ISDN trunks (CLID digits are used the same as ANI digits; DDI digits are used the same as DNIS digits). Refer to Automatic Number Identification (ANI)/Dialed Number Identification Service (DNIS) on Incoming Trunks for details.
- The ISDN T1 trunks must have COS Option 801 (Incoming Trunk Call Rotary) enabled in order to receive CLID or DDI digits.
- Incoming ISDN calls to the SX-200 ICP are presented to the switch as T1 voice calls.
- Incoming ISDN calls can only connect to voice devices and modems. Calls will be rejected if the type of call is incompatible with the called device type.
- When a CLID call is transferred across the Advanced Analog Network (AAN), the CLID number is not sent.
- A DISA (Direct Inward System Access) number for the modem connected to the PRI card can provide remote access security.
- COS option 800, ANI Applies, should be disabled.
- The timing co-ordination of the name and number for ONS CLASS sets on PRI trunks is controlled with COS Option 813, Delay ONS Ring - Wait for Network Name (0-6 s).

Programming

Form 53, Bay Location Assignment - Assign bay numbers to the CIM or FIM ports that connect PRI cards to the controller, or to the MMC sites that contain T1/E1 modules.

Note: Reset the system after updating Form 53.

Form 1, System Configuration - Program the node type as ISDN NODE (PRI card) or ONB T1 NODE (T1/E1 module). Program the trunk card as T1 TRUNK (PRI card) or T1 ISDN (T1/E1 module) for the slots specified in the following table.

Card or Module	Location	Slot
PRI Card	Peripheral Cabinet	6 and 8
T1/E1 Combo (AX/CX/CXi)	MMC1	2

Card or Module	Location	Slot
MMC2	6	
Dual T1/E1 Framer (AX/MX)	MMC1	2 and 4
MMC2	6 and 8	

Notes:

1 The CX/CXi can have one T1/E1 Combo module installed in MMC1 or MMC2. The MX can have up to two Dual T1/E1 Framer modules installed in MMC1 and MMC2. The AX can have either T1/E1 Dual Framer or T1/E1 combo installed in MMC 1.

2. One link of the Dual T1/E1 Framer can be programmed for PRI (T1 ISDN) and the other for T1 (T1 Trunk).

Form 03, Class of Service

ISDN Trunks Class of Service.

- Enable ANI/DNIS Trunk COS option 811 to display CLID (the caller's number) and DDI (the number the caller dialed) on incoming trunks.
- Enable Incoming Trunk Call Rotary COS option 801.
- Enable Incoming Trunk No Dial Tone COS option 701.
- Enable Limited Wait for Dial Tone COS Option 802.
- Enable SMDR - Record ANI/DNIS COS Option 814 to allow CLID digits to appear in SMDR reports for Incoming calls.
- Enable SMDR - Extended Record, COS Option 246, to allow the calling name to appear in SMDR reports for Incoming calls.
- Enable SMDR - Overwrite Buffer Option 702 if SMDR - Record Incoming Calls Option 806 is Enabled.

Note: Ensure that COS Option 800 (ANI Applies) is NOT enabled in the class of service of the ISDN trunks.

Set & Station Class of Service

- Enable Display ANI/DNIS Information COS option 502 and Display ANI Information Only COS option 613 to allow display of CLID digits on display telephones.
- Enable COS option 503, Enable Calling Name Display, to allow the display of the calling name.
- Enable SMDR - Overwrite Buffer COS option 702.
- DISABLE Outgoing Trunk Callback COS option 236.
- DISABLE Outgoing Trunk Campon COS option 237.

Form 04, System Options/System Timers - Assign a one to five second value to System Option 48, Limited Wait for Dial Tone if ISDN trunks are present. Purchase and program the correct Number of Links (option 96), and, if required, purchase and enable Remote LAN Access (option 93).

PRI card and NSU only: If required, enable QSIG (option 86), NFAS (option 91), D-Channel Backup (option 92), Min/Max (option 94) and Auto Min/Max (option 95). These are purchasable FOR options that require a password from Mitel.

Notes:

1. Options 86, 91, 92, 94, and 95 are available only with PRI cards, not embedded PRI.
2. For option 92 to be changed to ENABLED, option 91 must already be ENABLED. Similarly, if both are ENABLED, option 92 must be DISABLED first.

3. For option 95 to be changed to ENABLED, option 94 must already be ENABLED. Similarly, if both are ENABLED, option 95 must be DISABLED first.
4. A system reset may be required after some options are programmed. For example, changing the number of links in option 96 requires a reset.

Form 13, Trunk Circuit Descriptors - Assign the ISDN trunk with a T1 trunk circuit descriptor. To receive CLID or DDI digits, the ISDN trunk must be a non-DISA trunk that is programmed with a T1 E&M, or T1 DID/TIE trunk circuit descriptor.

In the options subform for Form 13, program the following options:

- If you are connecting to a DMS100 switch at the central office, set QSIG Supplementary Services to "Yes" to enable the Calling Name feature.
- Set the DTMF option to No.
- Set the Debounce Timer to 100 milliseconds.
- Set the Incoming Start Type to WINK.
- Set the Outgoing Start Type to WINK.

Note: For QSIG features, see the programming section in QSIG.

Form 15, Dial-In Trunks - Define the incoming ISDN trunk as a dial-in trunk. Enter the trunk circuit descriptor programmed in Form 13 and and enter the following for each trunk:

- COS number – see previous requirements
- COR number
- Tenant number – unless site has tenanting set to 1
- N – number of incoming DID digits – usually 4
- M – number of digits to absorb – usually 0
- X – digits to be inserted (up to 2 digits) or Feature Access Code 67 (Digit Translation Table Access) from Form 02
- CDN – Circuit Descriptor Number
- Trunk Number – Enter a trunk number (1-200)

Note: Normally, PRI trunks are programmed as Dial-In Trunks. To program a Non-Dial-in configuration, use Form 14.

Form 16, Trunk Groups - Assign the ISDN trunks to a trunk group:

- Locate an empty Trunk Group and add the PRI trunks (trunk numbers programmed above) to the Trunk Group. (Program them in reverse order to prevent contention with incoming calls from the CO.)
- Add a name to the Trunk Group for future reference.
- Set Group Type to Terminal (recommended) or Circular.
- Enable SMDR (outgoing only) if required.

Form 42, T1 Link Descriptor - Select a T1 Link Descriptor for the ISDN trunks in Form 42. In the Link Descriptor Options Subform, enter the following values for the T1 Link Descriptor:

Link Descriptor Option	Value
Alarm debounce timer (300-3200)	2500
Line Coding (AMI, AMI&ZCS, B8ZS)	B8ZS
Line Build Out (0, -7.5, -15, -22.5 dB)	0 DB
Line Length (max. 132, 265, 398, 533, 655)	0 - 132
Framing (D4 or ESF)	ESF (embedded PRI) D4 (NSU & PRI card)

Link Descriptor Option	Value
Slip rate - maintenance limit (0-9000)/24 hrs	255
Slip rate - service limit (0-9000)/24 hrs	7000
Slip rate - network sync limit (0-9000)/24 hrs	7
BER - maintenance limit (10**-n, n = (3,4,5,6))/hr	4
BER - service limit (10**-n, n = (3,4,5,6))/hr	3
Framing losses - maintenance limit (0-9000)/24 hrs	225
Framing losses - service limit (0-9000)/24 hrs	9000
RTS timer - service limit exceeded (1-255 min)	30
RTS timer - net slip limit exceeded (1-255 min)	30
RTS timer - after alarm (0-300 sec)	10
Embedded PRI only	
Embedded PRI Protocol	DMS100
Embedded PRI Protocol Variant	None
Embedded PRI Network/User	User
Embedded PRI Unknown Numbering Plan	Disabled
Embedded PRI Bearer Capability Voice	Speech
Embedded PRI CLIR Voice	Allow
Embedded PRI Invert D Channel	No

Form 43, T1 Link Assignment - Assign the ISDN trunk T1 Link Descriptor to the slot(s) specified in Form 1.

Form 44, T1 Network Sync - This form determines the order in which the T1 links are used as the network synchronization source. Program the T1 Network Synchronization form using the T1 Trunk PLIDs in order of most preferred to least preferred.

Note: If the SX-200 ICP has T1 and ISDN programmed , the 1st clock source must be the ISDN.

ARS Programming

Form 22, Modified Digit Table

DID Calling Party Number to the network interacts with the current system networking functionality and ignores programmed node-id's in the PRI card or T1/E1 module. The Node ID Information Element (“*8”) for Analog Networking, if programmed in the Modified Digit Table, will be ignored by the PRI card or T1/E1 module. If *6 is programmed in the digit modification table the DID calling extension number is sent to the PRI card or T1/E1 module and can be sent to the network as the calling party. The extension must belong to a block of DID numbers purchased from the Network provider.

The following table shows how Analog Networking and Call-by-Call information will co-exist in the Modified Digit Table:

SX-200 ICP Modified Digit Table			
Entry	**Quantity to Delete**	**Digits to be Inserted**	**Comments**
01	1	*4000*6*4	ISDN CxC and DID

In the “Digits to be Inserted” field, the definition of characters is:

*4 No SMDR for further modified digits

000

Call-by-Call digits, (speech, default O/G, clid)

Note: If Call-by-Call digits are enabled through IMAT programming they are not required in the Modified Digit Table.

*6 Send caller id (DID to network)

*4 Start SMDR again for further modified digits

Complete ARS Programming

You must program ARS to allow route access to the ISDN options and services that are available through the PRI card. For installations with Embedded PRI (T1/E1 modules), programming can be done in CDE. For installations with PRI cards, CDE programming must match the service selection programming that is entered through the ISDN Maintenance Administration Tool (IMAT) computer.

You can program the system to provide network services on a fixed basis or on a per-call basis. To set up access to network services on a fixed basis create trunk groups for each combination of services that are required and define a separate ARS route for each trunk group. Then, for PRI card installations, define the services for the trunk groups at the PRI card bay through the IMAT program. Refer to IMAT online Help for instructions on how to program ISDN features on a per-call basis.

To set up access on a per-call basis, program different ARS routes for each combination of required services (bearer service, CBC service, and CLIR service) using the leading digits as a code. This code consists of up to three leading digits that are inserted before the dialed number. The code determines the route that the call takes through the switch, and which services the PRI card provides for the call.

PRI Card Programming for ARS

If you haven’t done so already, you must now complete Site Options programming, PRI LINK Characteristics, and Trunk programming from the ISDN PC computer using IMAT. Refer to IMAT online Help for instructions on connecting the ISDN PC computer and programming the PRI card.

The programming data that you enter into the PRI card must match the programming data that you entered into the SX-200 ICP. If it does not match, network services may not operate properly:

- If the Automatic Route Selection (ARS) programming does not match, Call by Call (CBC) services will not operate as planned.
- If the Trunk Descriptor programming does not match, the system will use the parameters that you programmed on the PRI card bay.
- If a trunk is programmed on the SX-200 ICP, but not on the PRI card, the network services will not be available.

- If a trunk is programmed on the PRI card, but not on the SX-200 ICP, the network services will not be available.

Embedded PRI Programming for ARS

If Calling Party Number (CPN) substitution is programmed, make sure that Form 22, ARS: Modified Digit Table, contains an entry with CPN enabled for embedded PRI. Entries for offboard PRI cards and NSUs should have CPN disabled.

Operation

Users are not required to perform any special actions to access the ISDN network through the PRI card or T1/E1 module.

Printer/Terminal Support

Description

This feature allows the routing of printouts to the system printer port, to any data port, an SX-200 ICP IP socket, or to the printer port on the SUPERCONSOLE 1000 Attendant Console. All printer ports are RS-232C interface. Printout types include:

- Traffic measurement
- SMDR (Trunk, Data, ACD)
- CDE
- Hotel/Motel system printouts
- PMS interface port
- ACD reports
- Maintenance logs.

Customer programming printouts may be directed to any or all of seven user-defined printers. A maximum of six DNIC-based printer ports can be defined; the remaining port is the system printer port.

CDE forms may be printed individually or collectively.

Certain CDE forms require the user to specify which sub-forms are to be printed (in a “from - to” format).

Conditions

The following conditions apply to this feature:

- Printouts may be sent to more than one printer at a time. The time required to print is determined by the slowest printer.
- Not more than five printouts may be directed to one printer.
- Some printouts can be guaranteed. If there is a printer failure, the information is preserved and printed when the printer is ready again.
- If the printer runs out of paper and can send flow control information, the system suspends printing until the printer is ready again.
- Printers must have an RS-232C interface. A dataset is required to connect the printer to a data port in the SX-200 ICP. To connect to an IP socket, an RS232-to-IP serial port converter is required between the printer and a port on the Layer 2 switch.
- If the printer, dataset, or RS232-to-IP serial port converter is powered down, the system waits until the device is ready again.
- A functioning printer must always be connected to the port assigned for SMDR printouts. If the printer fails or is disconnected, outgoing trunk calls are disabled as soon as the internal storage buffer is full

if ‘SMDR - Overwrite Buffer’ (COS Option 702) is disabled in the COS for the trunk. The system’s internal SMDR record buffer holds 200 SMDR records; this provides sufficient time for printer maintenance.

- The PMS printer port must be dedicated to PMS only; it cannot be shared with any other application.
- A SUPERCONSOLE 1000 Attendant Console can support either a printer port or a DSS/BLF Interface Unit, but not both. You can't program submarket 2 of the console’s Bay/Slot/Circuit location as a DSCONS in CDE Form 12 and also program the console as a PKM HOST (in Form 9).
- The SUPERCONSOLE 1000 Attendant Console printer port limitations are described in the Peripherals Devices section.
- The 5540 IP Console doesn't support an RS-232 serial printer port. Reports can be printed from the 5540 IP Console on a system printer port, to any data port, or on an SX-200 ICP IP socket.

Programming

For a printer connected via a dataset, define an appropriate Data Circuit Descriptor in CDE Form 11 (Data Circuit Descriptor). Program the dataset in Form 12 (Data Assignment) using the new Data.

For a printer connected via an IP socket, define an appropriate Data Circuit Descriptor in CDE Form 11 (Data Circuit Descriptor). Program the IP socket in Form 12 (Data Assignment). Program the RS232-to-IP serial port converter with the following settings:

Table: RS232-to-IP Serial Port Converter Settings

Setting	Value
Protocol	Transparent TCP (tunnel)
Local Port	0
Port speed	As required
Connection control	DTR/DSR
Remote IP	SX-200 RTC IP Address
Remote Port	As programmed in CDE Form 12
Terminators	Not required

Complete programming of CDE Form 34 (Directed I/O); specify the printer ports available, and then direct each printout to its associated printer.

After assigning the SUPERCONSOLE 1000 Attendant Console in CDE Form 07 (Console Assignments), program submarket 2 of the console’s Bay/Slot/Circuit location as a DSCONS in CDE Form 12 (Data Assignment), if the console port is to be a printer port.

Note: A printer connected to a printer port must have parameters set up as shown in the RS-232 Maintenance Terminal section.

Operation

The system checks CDE Form 34 (Directed I/O) whenever it generates a printout, to determine to which printer port it should route the printout.

Priority Dial 0

Description

Priority Dial 0 is a second class of Dial 0 call, with its own separate DAY/NIGHT routing points. This feature can be used to provide an alternate Dial 0 routing for extensions in the system.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Priority Dial 0 calls can be routed to the same type of answering points as Dial 0 calls.
- Wherever Dial 0 routing is used in the system, Priority Dial 0 is checked.
- Priority Dial 0 calls may be routed to the same point as Dial 0 calls or they may be routed to an LDN key which distinguishes Dial 0 and Priority Dial 0 calls.

Programming

Enable COS Option 239 (Priority Dial 0) for the required extension.

Program an Access Code (usually 0) in CDE Form 02 (Feature Access Codes) for Feature 11 (Extension General Attendant Access).

Operation

Dial the access code.

Privacy Enable/Privacy Release

Description

A multi-line Mitel telephone may have line appearances of key, direct trunk select, CO line and private trunk lines that are shared with other sets. When privacy is enabled, while a conversation is in progress on a line, other sets with an appearance of that line are denied access. See Line Privacy. The user of the line can, however, use the Privacy Release feature to allow the other sets to join the conversation.

The user may press a Release Privacy softkey or a Privacy Release feature key for each call that the user wants other parties to join.

If the user wants their privacy released as a default, a COS option, Privacy Released at Start of Call, may be enabled. To obtain privacy, the user then has to press a Make Private softkey or a Privacy Release feature key.

If privacy is released, and a new party joins a multi-party conference call with a key line appearance, the system will provide a warning tone to all of the parties alerting them of this occurrence. If customers do not want this warning tone, you can disable this functionality with COS Option 126, "Apply Key Line Conference Warning Tone". The default setting is ENABLED.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Line Privacy is dependant on COS Option 240, Line Privacy.
- Privacy is effective only against other appearances of the line; it has no effect on override.
- Privacy Release is available only on lines that have key line appearances and trunk lines.
- ONS, Family 1, SUPERSET 3DN, and SUPERSET 4DN telephones do not support COS Option 271, Privacy Released at Start of Call.
- COS Option 271, Privacy Released at Start of Call, provides a release of the privacy as a default. This COS option requires FOR Option 102, Feature Level (level 1 or greater) and COS Option 240, Line Privacy being enabled.
- If COS Option 271 is enabled, the privacy remains released until someone makes the line private; no change occurs when people join in or leave a call. A tone is presented to indicate that someone is listening. If the user requires no tone, as in ACD applications, the user should use silent monitor to listen in on agent's calls.
- If COS Option 271 is disabled, the line is made private automatically every time someone enters or leaves the line. To release privacy, the user must press the Privacy Release key each time someone enters or leaves the line.
- COS Option 271 and COS Option 240 apply to the owner of the line used, not to the set that uses the line; the owner of a CO line is a trunk not a set. The line owner of a phantom line with key appearances is the set that had the phantom line first programmed on. This information may be found in the Expand Set subform of Form 09, under REVIEW.
- COS Option 271 enabled on your set, that is having privacy released as a default, prohibits you from putting your caller on hard hold. In order to put your caller on hard hold, press the Privacy Release key that would make your set private and then press the red hold key.
- COS Option 271 enabled on your set, prohibits you and a listener on the key line appearance from putting another party into the conversation. The TRANS/CONF key (the soft hold) and the Feature key will not work.
- COS Option 126 "Apply Key Line Conference Warning Tone" applies to the owner of the conference (with privacy-released). The owner refers to the set that has the key line number as the prime line or the first set that was programmed with a key line appearance with this number. Depending on whether the option is enabled or disabled; all parties in the conference call with this line will, or will not, hear the warning tone when another party joins in with the key line appearance. This COS Option requires Feature Level 3 or greater in System Option 102. If the Feature Level is 2 or less, a warning tone will be given regardless of the programming of COS Option 126.

Programming

To allow Privacy Release operation at a Mitel Family 3, Family 4, SUPERSET 410, or SUPERSET 3DN telephone, program a PRIVACY RELEASE feature key (see Feature Keys).

To allow Privacy Release as a default, enable COS Option 271 (Privacy Released at Start of Call) and COS Option 240 (Line Privacy) in the trunk COS for CO line keys and in the keyline COS for key lines. Also, set the FOR System Option 102, Feature Level to level 1 or greater.

To allow the enabling/disabling of line privacy, enable COS Option 240, Line Privacy, on the line owner.

To disable a warning tone for key line conferences, disable COS Option 126, "Apply Key Line Conference Warning Tone" for the owner of the conference (the set that has the key line number as the prime line or the first set that was programmed with a key line appearance with this number).The default setting is enabled. Disabling the warning tone is dependant on Feature Level 3 or greater in System Option 102.

Operation

Family 3, Family 4, Family 5, SUPERSET 410, and SUPERSET 3DN Telephones (without COS 271 enabled)

- During an established call, press the PRIVACY RELEASE feature key to toggle between privacy on and off. The LCD or LED shows the status of privacy on the line as follows:
  - LCD clear or LED extinguished indicates Private.
  - LCD dark or LED illuminated indicates Not Private.
- If the LCD is dark or if the LED illuminated, another telephone user with an appearance of the same line can now enter the conversation by pressing the appropriate line select key.

Family 3, Family 4, Family 5, SUPERSET 410, and SUPERSET 3DN Telephones (with COS 271 enabled)

Having COS 271 enabled puts the start of the call in the privacy release mode.

To enable privacy

- During an established call, press the PRIVACY RELEASE feature key. Another telephone user with an appearance of the same line can NOT enter the conversation by pressing the appropriate Line Select key.

To return to the privacy release mode

- Press the PRIVACY RELEASE feature key.

Family 1 Telephones (without COS 271 enabled)

- During an established call, press the REL PRIVACY softkey. Another telephone user with an appearance of the same line can now enter the conversation by pressing the appropriate Line Select key.
- Re-establish privacy by pressing the MAKE PRIVATE softkey.

Family 1 Telephones (with COS 271 enabled)

Having COS 271 enabled puts the start of the call in the privacy release mode.

To enable privacy

- During an established call, press the MAKE PRIVATE softkey. Another telephone user with an appearance of the same line can NOT enter the conversation by pressing the appropriate Line Select key.

Note: A telephone user can enter the conversation if they use the Intrude feature key (COS Option 500, Override, enabled).

To return to the privacy release mode

- Press the REL PRIVACY softkey.

Programmable Key Module (PKM)

A Programmable Key Module provides a set with additional personal keys. There are seven models: the Mitel 5410 PKM, the Mitel 5415 PKM, the Mitel 5412 PKM, the Mitel 5448 PKM, the PKM 48, the PKM 12, and the PKM. The 5415 PKM, 5448 PKM and PKM 48 provide 48 extra keys, the 5410 PKM, 5412 PKM and PKM 12 provide 12 extra keys, and the PKM provides 30. See the following sections:

- PKM 48 and PKM 12 for SUPERSET 4025, SUPERSET 4125, and SUPERSET 4150 telephones .
- 5448 PKM and 5412 PKM for Mitel 5220, 5224 and 5324 IP Phones and the 5540 IP Console. Unless otherwise stated, this information also applies to the 5410 PKM and 5415 PKM for the Mitel 5020 IP Phone.
- Direct Station Select/Busy Lamp Field (DSS/BLF) Interface Unit for attendant consoles associating up to two PKM 48 devices via the interface unit.

- PKM 48 for the SUPERCONSOLE 1000 (Part numbers 9189-000-300 and 9189-000-301) for attendant consoles directly supporting up to two PKM 48 devices.
- PKM for SUPERSET 410, SUPERSET 420, and SUPERSET 430 telephones.

PKM 48 and PKM 12 for SUPERSET 4025, SUPERSET 4125 and SUPERSET 4150 Telephones

Description

The PKM 48 and PKM 12 provide additional personal keys for SUPERSET 4025, SUPERSET 4125, and SUPERSET 4150 telephones. The PKM 48 provides 48 extra keys, and the PKM 12 provides 12. The SUPERSET telephone attached to the PKM 45 or to the PKM 12 requires a SUPERSET interface module in its base to interface to the programmable key module. The additional personal keys can be programmed with the following functions:

- speed call keys
- feature keys
- key line appearances
- personal outgoing line keys
- key system appearances
- multi-call line appearances
- CO line keys
- busy lamp field/direct trunk select keys.

The PKM 48 has four vertical rows of keys. The PKM 12 has one vertical row. Beside each key is a Line Status Display that indicates the status of the key.

The PKM 48 and the PKM 12 are the only programmable key modules qualified by Mitel for connection to SUPERSET 4000 series telephones.

You can connect one PKM 12 or up to two PKM 48 devices to a set.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply:

- A PKM 48 and a PKM 12 cannot be used with SUPERSET 400-series telephones, the Mitel 5201 IP Phone, SUPERSET 4001 telephones, Mitel 5010, 5212 and 5215 IP Phones, SUPERSET 4015 telephones.
- The SUPERSET 4025, SUPERSET 4125, and SUPERSET 4150 telephone requires a SIM1 or SIM2 in its base to interface to the PKM 48 or PKM 12.
- The SUPERSET 4025, SUPERSET 4125, or SUPERSET 4150 telephone should have the latest firmware upgrade available from the system maintenance firmware upgrade command.
- The PKM 48 and the PKM 12 for the SUPERSET 4025 telephone derives its power from the SIM, and the SUPERSET 4150 and SUPERSET 4125, from the set. If the power adapter to the SUPERSET 4150 is plugged into the SIM you will loose full duplex functionality.
- COS options 650, 651 and 652 in Form 3 MUST be 0.

Programming

To program a telephone to support a PKM 48 or PKM 12:

- Display CDE Form 09 (Desktop Device Assignments).

- Move the cursor to the bay/slot/circuit number of the SUPERSET 4000-series telephone.
- Select the EXPAND PKM softkey. “ENTER PKM NUMBER (1-2):” appears in the command line.
- Type the address setting (1 or 2) of the programmable key module to be programmed.
- Select the ENTER softkey to display the Expand PKM Set Subform, which allows you to program the functions of the 12 keys on the PKM 12 or the 48 keys on the PKM 48.

Caution: CDE does not differentiate between a PKM 12 and a PKM 48. The programming for both modules is exactly the same. Be sure not to program keys 13 or greater for a PKM 12.

To program the PKM 48 or PKM 12 keys:

- Move the cursor to the desired key. See PKM 48 for the key numbering for a PKM 48. See PKM 12 for the key numbering for a PKM 12.
- Move the cursor to the TYPE column (use the TAB key if necessary).
- Select the key type using the softkeys on the Expand PKM Set Subform.

  Key numbers are preceded by 1 or 2. The 1 or 2 indicates the order in which you associated the PKM with the set (the keys belonging to the first PKM 48 that you associated with the set will be preceded by a “1”).

  To program the PKM keys on a SUPERSET telephone refer to Expand PKM Set Subform for Form 09 in the Customer Data Entry section for definitions of the available key types.
- Users can also program any PKM 48 or PKM 12 personal key that isn’t a line appearance as a feature key from their sets (refer to Feature Keys for instructions).

When deleting a set that has a PKM 48 or PKM 12:

- You cannot delete a set from CDE Form 09 (Desktop Device Assignments) until you delete all of the set’s PKM keys. For each PKM assigned to the set, you must access the Expand PKM Set Subform and delete the PKM keys.

To delete a PKM 48 or PKM 12:

- Delete all of the keys from Form 09, Expand PKM subform.
- Press the DELETE PKM softkey.

Operation

For instructions on using the features associated with PKM 48 or PKM 12 personal keys, refer to the specific feature.

5448 PKM and 5412 PKM for Mitel 5220, 5224 and 5324 IP Phones

Note: Unless otherwise stated, this information also applies to the 5410 PKM and 5415 PKM for the Mitel 5020 IP Phone.

Description

The Mitel 5448 PKM and Mitel 5412 PKM provide additional personal keys for the Mitel 5220, 5224 or 5324 IP Phones. The 5448 PKM provides 48 extra keys, and the 5412 PKM provides 12 additional keys. The 5220, 5224 or 5324 IP Phone attached to the 5448 PKM or 5412 PKM requires a Mitel 5422 PKM Interface Module (IM) in its base to interface to the programmable key module. The additional personal keys can be programmed with the following functions:

- speed call keys
- feature keys
- key line appearances
- personal outgoing line keys

- key system appearances
- multi-call line appearances
- CO line keys
- busy lamp field/direct trunk select keys.
- CO line group key
- private trunk key
- direct trunk (DTS) key

The 5448 PKM has four vertical rows of keys. The 5412 PKM has one vertical row. Beside each key is a Line Status Display that indicates the status of the key.

The 5448 PKM and 5412 PKM are the only programmable key modules qualified by Mitel for connection to 5220, 5224 and 5324 IP Phones.

You can connect one 5412 PKM or a maximum of two 5448 PKM devices to a 5220, 5224 or 5324 IP Phone.

Note: The Mitel 5410 PKM and Mitel 5415 PKM provides additional personal keys for the Mitel 5020 IP Phone. The 5410 PKM provides 12 additional keys, and the 5415 provides 48 additional keys. The 5020 IP Phone attached to the 5415 PKM or 5410 PKM requires a Mitel 5421 PKM Interface Module (IM) in its base to interface to the programmable key module.

Conditions

The following conditions apply:

- A 5448 PKM and 5412 PKM cannot be used with SUPERSET 400-series telephones, SUPERSET 4001, SUPERSET 4015, the Mitel 5201 IP Phone, or Mitel 5010, 5212 and 5215 IP Phones.
- A 5410 PKM and 5415 PKM cannot be used with SUPERSET 400-series telephones, SUPERSET 4001, SUPERSET 4015 telephones, the Mitel 5201 IP Phone, or Mitel 5010, 5212 and 5215 IP Phones.
- The Mitel 5220, 5224, and 5324 IP Phone requires a 5422 PKM IM in its base to interface with the 5448 PKM or 5412 PKM.
- The Mitel 5020 IP Phone requires a 5421 PKM IM in its base to interface to the 5410 PKM or 5415 PKM.
- The Mitel 5020, 5220, 5224, and 5324 IP Phones should have the latest firmware upgrade available from the system maintenance firmware upgrade command.
- The 5412 PKM and the 5448 PKM for the Mitel 5220, 5224 or 5324 IP Phone derives its power from the 5422 PKM IM.
- The 5410 PKM and 5415 PKM for the Mitel 5020 IP Phone derives its power from the 5421 PKM IM.

Programming

To program the phone to support a 5448 PKM or 5412 PKM:

- Display CDE Form 09 (Desktop Device Assignments).
- Move the cursor to the bay/slot/circuit number of the 5220, 5224 or 5324 IP Phone.
- Select the EXPAND PKM softkey. “ENTER PKM NUMBER (1-2):” appears in the command line.
- Type the address setting (1 or 2) of the programmable key module to be programmed.
- Select the ENTER softkey to display the Expand PKM Set Subform, which allows you to program the functions of the 12 keys on the 5412 PKM or the 48 keys on the 5448 PKM.

Caution: CDE does not differentiate between a 5448 PKM and 5412 PKM. The programming for both modules is exactly the same. Be sure not to program keys 13 or greater for a 5412 PKM.

To program the 5448 PKM or 5412 PKM keys:

- Move the cursor to the desired key.
- Move the cursor to the TYPE column (use the TAB key if necessary).
- Select the key type using the softkeys on the Expand PKM Set Subform.

  Key numbers are preceded by 1 or 2. The 1 or 2 indicates the order in which you associated the PKM with the set (the keys belonging to the first 5448 PKM that you associated with the set will be preceded by a “1”).

  To program the PKM keys on a 5220, 5224 or 5324 IP Phone refer to Expand PKM Set Subform for Form 09 in the Customer Data Entry section for definitions of the available key types.
- Users can also program any 5412 PKM and 5448 PKM personal key that isn’t a line appearance as a feature key from their sets (refer to Feature Keys for instructions).

When deleting a set that has a 5448 PKM and 5412 PKM:

- You cannot delete a set from CDE Form 09 (Desktop Device Assignments) until you delete all of the set’s PKM keys. For each PKM assigned to the set, you must access the Expand PKM Set Subform and delete the PKM keys.

To delete a 5412 PKM and 5448 PKM:

- Delete all of the keys from Form 09, Expand PKM subform.
- Press the DELETE PKM softkey.

Operation

For instructions on using the features associated with 5410 PKM, 5415 PKM, 5448 PKM and 5412 PKM personal keys, refer to the specific feature.

PKM 48 for the SUPERCONSOLE 1000 (Direct Connection) and the 5540 IP Console

Description

The PKM 48 provides the SUPERCONSOLE 1000 (5448 PKM for the 5540 IP Console) with 48 additional keys. The SUPERCONSOLE 1000 with part numbers 9189-000-300 and 9189-000-301 can directly connect up to two PKM 48 devices. The keys on the PKM 48 (5448 PKM for the 5540 IP Console) can only be used as Direct Station Select (DSS)/Busy Lamp Field (BLF) keys.

Conditions

The following conditions apply:

- The SUPERCONSOLE 1000 with part numbers 9189-000-300 and 9189-000-301 can directly connect up to two PKM 48 devices. The older models require a Direct Station Select/Busy Lamp Field (DSS/BLF) Interface Module to associate up to two PKM 48 devices with the attendant console.
- The 5540 IP Console supports up to two 5448 PKM's, both of which are powered by the 5540 IP Console's PoE connection.
- Only DSS/BLF key types may be programmed on the PKM 48 and 5448 PKM.
- Direct connection requires the purchasable FOR System Option 102, Feature Level (level 1 or greater).
- The SUPERCONSOLE 1000 with part numbers 9189-000-300 and 9189-000-301supports the connection of either PKM 48 devices or a printer, both are not supported.

Programming

To program an attendant console to directly support a PKM 48:

- Display CDE Form 07 (Console Assignments)
- Move the cursor to the bay/slot/circuit number of the attendant console.
- Select the EXPAND PKM softkey. “ENTER PKM NUMBER (1-2):” appears in the command line.
- Type the address setting (1 or 2) of the PKM 48 to be programmed.
- Select the ENTER softkey.
- Move the cursor to the desired key. The system displays the correct quantity of keys. See this Figure for the PKM 48 numbering.
- Move the cursor to the TYPE column (use the TAB key if necessary).
- Select the BLF/DSS softkey (Type becomes “Busy Lamp”).
- Press TAB key three times to get to EXT NUM column (or four times to get to the TRK NUM column).
- Enter the number of the device to be monitored by the Line Appearance.
- Select the ENTER softkey.

Operation

See Direct Station Select (DSS) keys.

When the line/extension is

- Idle, the indicator is Off
- Busy, the indicator is On
- Ringing, the indicator is a Slow Flash
- On Hold, the indicator is a Fast Flash

PKM for SUPERSET 400 Series Telephones

Description

A Programmable Key Module (PKM) provides SUPERSET 410, SUPERSET 420, and SUPERSET 430 telephones with 30 additional personal keys. You can program these personal keys as

- speed call keys
- feature keys
- key line appearances
- personal outgoing line keys
- key system appearances
- multicall line appearances
- co line keys
- busy lamp field/direct trunk select keys.

The keys are arranged in two vertical rows on the module (see PKM for the key numbering). Beside each key is a Line Status Display that indicates the status of the key. The flash rates for the Line Status Displays on the PKM are identical to those on the SUPERSET 410, SUPERSET 420, and SUPERSET 430 telephones.

Up to three PKM modules connect directly to the set.

Each SUPERSET 410, SUPERSET 420, and SUPERSET 430 telephone has one MILINK port. Each PKM has two MILINK ports. The first PKM connects to the MILINK port located on the base of the telephone. The other two PKMs are connected in series (i.e., daisy-chained together) to the first. For installation instructions refer to the Peripheral Devices section.

SUPERSET 410, SUPERSET 420, and SUPERSET 430 telephone users can program any PKM personal key that isn't assigned to a line appearance. Users can program personal keys as feature keys from their set. Refer to Feature Keys for instructions.

If you print Form 09 (Desktop Device Assignments) from Form 32 (CDE Data Print), any PKM key line appearances are printed following the associated set key line appearances. If you have more than one PKM programmed with key line appearances, the keys are printed in the order of the address settings. If a PKM is not programmed with any key line appearances, "UNPROGRAMMED PKM" is printed following the PKM's address setting.

Conditions

The following conditions apply:

- A SUPERSET 410 with three PKMs provides up to 96 key line appearances. A SUPERSET 420 or SUPERSET 430 with three PKMs programmed provides up to 102 key line appearances.
- The PKM has rocker dip switches located on the base of the set. You must set these switches to identify the address (1, 2, or 3) of the PKM. Each PKM must have a different address. For information on how to set the PKM addresses, refer to the Peripheral Devices section.
- A Programmable Key Module (PKM) cannot be used with SUPERSET 4000-series telephones.

Programming

To program a telephone to support a PKM:

- Display CDE Form 09 (Desktop Device Assignments)..
- Move the cursor to the bay/slot/circuit number of the SUPERSET 410, SUPERSET 420, or SUPERSET 430 telephone. An asterisk (*) appears to the left of the set type (410, 420, 430) if PKMs are programmed for the set.
- Select the EXPAND PKM softkey. "ENTER PKM NUMBER (1-3):" appears in the command line.
- Type the address setting (1, 2, or 3) of the next PKM that you can add.
- Select the ENTER softkey to display the Expand PKM Set Subform. The Expand PKM Set Subform allows you to program the functions of the 30 available PKM keys.

To program the PKM keys:

- Move the cursor to the desired key (see PKM for the key numbering).
- Move the cursor to the TYPE column (use the TAB key if necessary).
- Select the key type using the softkeys on the Expand PKM Set Subform.

  Key numbers are preceded by 1, 2 or 3. The 1, 2 or 3 indicates the order in which you associated the PKM with the set (e.g., the keys belonging to the first PKM that you associated with the set will be preceded by a "1"). The 1, 2 or 3 is not related to the PKM address settings.

  To program the PKM keys of a SUPERSET telephone refer to Expand PKM Set Subform for Form 09 in the Customer Data Entry section for definitions of the available key types.
- Users can also program any PKM personal key that isn't a line appearance as a feature key from their sets (refer to Feature Keys for instructions).

When deleting a set that has PKMs:

- You cannot delete a set from CDE Form 09 (Desktop Device Assignments). until you delete all of the set's PKM keys. For each PKM assigned to the set, you must access the Expand PKM Set Subform and delete the PKM keys.

To delete a PKM:

- Delete all of the keys from Form 09, Expand PKM subform.

- Press the DELETE PKM softkey.

Operation

For instructions on using the features associated with PKM personal keys, refer to the specific feature.

QSIG

Description

QSIG Basic Call

QSIG is a powerful signaling system used to connect networking equipment, such as PBXs, in corporate telecommunications networks.

Because QSIG is an international standard, it allows a private network to include networking equipment from different vendors. Multi-vendor commitment to QSIG standards means that corporate networks can operate internationally with confidence in the interoperability of PINX equipment from many suppliers. (In QSIG networks, private integrated services network exchange (PINX) is the generic term used to describe various types of corporate networking equipment, such as PBXs, multiplexors, and Centrex systems.) See figure for a typical QSIG network.

QSIG has been designed for compatibility with ISDN networks and business applications developed for public ISDN. ISDN bearer and teleservices used by multimedia operations can be extended over the QSIG network.

QSIG networks support any number of nodes and all types of network topology (meshed, star, main-and-satellite), so existing networks can be upgraded to QSIG regardless of their network configuration.

In the SX-200, the QSIG protocol resides on the SX-200 PRI card or NSU and runs over T1 circuits. The PRI card or NSU provides end node functionality and the calling name supplementary service in a 2-node topology. In the 2-node configuration, one system is designated as the master node and the other is the slave node. The master node represents the ISDN network side, and the slave node represents the user side.

Note: With the Multiple Variants feature (a standard feature in ISDN Release 6), one link on a PRI card can run QSIG and the other link can run an ISDN protocol.

The SX-200 ICP supports end node functionality in the network; in other words, the SX-200 ICP can only be connected to one other node in the network. The centralized attendant and the voice mail services must be on a node other than the SX-200 ICP.

QSIG Calling Name and Number

The QSIG Calling Name and Number feature allows the called party to see the name and number of the caller on the telephone display screen during the ringing state of incoming calls, when the called party is busy on the telephone, and when the called party answers a PRI call. The feature is available on all programmed QSIG links.

QSIG Call Diversion (Call Forwarding)

Incoming calls are diverted to another destination as defined by the user when the service is activated.

The SX-200 ICP QSIG Call Diversion as an originating node or as an end node; not as a tandem node.

- The SX-200 ICP the Call Forward Rerouting mode as an originating, terminating, or forwarding PBX.
- The SX-200 ICPs the Call Forward Switching mode as an originating or terminating PBX, but not as the forwarding PBX.

Several types of call forwarding (diversion) services are supported:

- Call Forward Busy (CFB) - incoming calls to a busy extension are diverted.
- Call Forward No Reply (CFNR) - incoming calls are diverted if not answered within a predefined period of time. Also known as Call Forward - No Answer.
- Call Forward Unconditional (CFU) - all incoming calls are diverted. Also known as Call Forward - Always.

QSIG Message Waiting Indication

With QSIG Message Waiting Indication, the SX-200 ICP accepts standard QSIG MW activate/deactivate messages from a remote PBX.

The centralized attendant and the voice mail services must be on a node other than the SX-200 ICP. Note that the remote PBXs must be able to understand our access code based control of message waiting lamps, and the voice mail system must provide a translation into QSIG messages.

QSIG Call Transfer

Users can connect two calls of the same basic service, together as a new call. The calls are transferred with Call Transfer by Join; not Call Transfer by Rerouting. The SX-200 ICP can be the originating node, the transferring node, or the terminating node for the call transfer. Calls are transferred across the network and transferred calls from the network are accepted.

QSIG Call Offer (Campon)

Users can offer calls to those at a busy destination. The busy user receives indication of a call offer, while the calling party receives indication that a call offer has been invoked. The called user has the choice of clearing the current call and being re-rung, putting the current party on hold and accepting the offered call, or ignoring the offered call. The SX-200 ICP uses the Without Path Retention mode only; not the With Path Retention mode. The SX-200 ICP will only originate a call offer without path retention.

QSIG Path Replacement (Partial)

The SX-200 ICP does not initiate path replacement but does support path replacement when initiated by another PBX. Active calls, connected through the QSIG network, can be replaced with new connections which are more efficient or cost effective. Therefore, unnecessary loops are eliminated with call forwarding and call transfers.

This feature only works if a desktop device on one PBX calls across a link that supports QSIG Supplementary Services to another SX-200 ICP and is transferred or forwarded back to the original PBX. If a call comes in to the original PBX, then goes across the QSIG link to another PBX and is transferred or forwarded back to the original PBX, Path Replacement does not occur.

QSIG Feature List

The table below displays the QSIG features for a SX-200 ICP ONLY network.

If the SX-200 ICP is behind an SX-2000 system, the QSIG features supported on the SX-200 ICP are the same except for the following features:

- Path Replacement is Partial
- Message Waiting is Yes (supporting incoming calls).

QSIG Features in a SX-200 ICP ONLY Network		
Standard	**Feature**	**Supported on the SX-200 ICP**
ETS 300 012 (Ed 1)	Layer 1	Yes
ETS 300 402 - 1&2	Layer 2	Yes
ISO 11574, 11572	Audio Speech	Yes
ISO 11571	Numbering Plan	Yes
ISO 11582	Generic SS Platform (GF)	Yes
ISO 14136	Calling Line Identification Presentation (CLIP)	Yes
ISO 14136	Connected Line Identification Presentation (COLP)	Yes
ISO 14136	CLIP/COLP Restriction (CLIR)	No
ISO 13864, 13868	Calling Name Identification Presentation (CNIP)	Yes
ISO 13864, 13868	Connected Name Identification Presentation (CONP)	Yes
ISO 13864, 13868	CNIP/CONP Restriction (CNIR)	No
ISO 13872, 13873	Call Forwarding Unconditional (CFU)	Yes (Call Rerouting)
ISO 13872, 13873	Call Forwarding Busy (CFB)	Yes (Call Rerouting)
ISO 13872, 13873	Call Forwarding No Reply (CFNR)	Yes (Call Rerouting)
ISO 13865, 13869	Call Transfer (CT)	Yes
ISO 13863, 13874	Path Replacement (PR)	No
ISO 13866, 13870	Call Completion to Busy Subscriber (CCBS)	No
ISO 13866, 13870	Call Completion on No Reply (CCNR)	No
ISO 14841, 14843	Call Offer (CO)	Yes (without path retention)
ISO 14844, 14842	Do Not Disturb Override (DNDO)	No
ISO 14844, 14842	Do Not Disturb (DND)	No
ISO 14846	Call Intrusion (CI)	No
ISO 15505, 15506	Message Waiting (MWI)	No
ISO 15055, 15056	Transit Count (TC)	No

Conditions

The following conditions apply to this feature:

- The QSIG FOR option and ISDN upgrades must be ordered from Mitel.
- The QSIG (and analog network) calls are considered external.
- The trunks supporting QSIG features (i.e., Diversion, Message Waiting Indication, Call Transfer, Call Offer, and Path Replacement) must be programmed in CDE Form 13, Trunk Circuit Descriptors, with

the QSIG Supplementary Services option. The standard feature programming for Call Forwarding, Transfer, Campon, and Messaging - Call Me Back still apply.

- The SX-200 ICP can only be connected to one other node in the network. The centralized attendant and the voice mail services must be on a node other than the SX-200 ICP.
- All PBXs in the network using Call Forwarding Rerouting for Call Diversion must program their ARS routing digits consistently.
- When a PBX transfers an unsupervised call over the network, the transferring PBX must start the No Answer timer. If the call is not answered within that time period, the PBX will drop the call and recall the party that initiated the transfer.
- Retrieving voice mail messages with the QSIG message waiting supplementary feature left by the remote PBX is dependant on having COS Option 265, Voice mail System Speed Dial Index, programmed on the first QSIG trunk on the first PRI link. This programming is required for users to retrieve voice mail messages that are left by the remote PBX.
- COS Option 613, Display ANI Only, also applies to QSIG calls. When COS Options 502, Display ANI/DNIS/CLASS Information and COS Option 613 are enabled, only the ANI information will show if it is provided, otherwise the trunk name, trunk group name, or trunk number is displayed.
- TIE trunks do not support QSIG Call Offer. Attempting to call offer or campon over a TIE trunk returns a busy tone. See Campon.
- The PRI interface designated as the QSIG master must be configured and programmed as a master interface:
  - set the NT/LT jumper for that interface to LT (see Installing the PRI Card)
  - in IMAT, program the link as a master interface (see Programming)
- The PRI interface designated as the QSIG slave must be configured and programmed as a slave interface:
  - set the NT/LT jumper for that interface to NT (see Installing the PRI Card)
  - in IMAT, ensure that the link is not programmed as a master interface (see Programming)
- QSIG Calling Name supports names with a maximum of 15 characters.
- QSIG trunks do **NOT** support # and * in the dialing string.

Programming

Enable Option 86 (PRI Card: QSIG) in the System Options form.

To enable QSIG Calling Name and extended SMDR display of Calling Name, enable the following COS Options in the Class of Service Options form:

- Option 502 (Display ANI/DNIS/CLASS Information) in Telephone COS
- Option 503 (Display CLASS Name) in Telephone COS
- Option 811 (ANI/DNIS Trunk) in Trunk COS
- Option 246 (SMDR - Extended Record) in Trunk COS.

In Form 22, Modified Digit Table, enter *04 in the ARS route that is used for QSIG calls. Enable CPN for the modified digit entry that applies to embedded PRI (Dual Link T1/E1 Dual Framer and T1/E1 Combo modules), and disable CPN for the entry that applies to offboard PRI cards and PRI NSUs.

To provide the other QSIG features besides Basic Call and Calling Name, in Form 13, Trunk Circuit Descriptors, select the trunk (T1 E&M, T1 DID/TIE, or T1 LS/GS) that requires these extra QSIG features, select Sel. Option, and program YES for the QSIG Supplementary Services option.

To support QSIG Message Waiting Indication:

- In Form 04, System Options and Timers, set System Option 113, Centralized Attendant / Voice mail, to enabled (must be purchased).

- In Form 31, System Abbreviated Dial Entry, program the index number and the digit string in order to retrieve the voice mail messages.
- In the Class of Service for the first QSIG trunk on the first PRI link, enter the index number programmed in CDE Form 31 into COS Option 265, Voice mail System Speed Dial Index.

Select QSIG in the IMAT Site Options screen.

Select Network-side Interface/Qsig master for the link in the IMAT PRI Link Characteristics screen if this system is to be the QSIG master.

Operation

For QSIG Calling Name, see CLASS for Digital Sets.

RAD Support

Description

Recorded Announcement Devices (RAD) are supported in the system as recording hunt groups. These special hunt groups have features and restrictions on them that allow efficient use of the recording resources. Recording hunt groups are used in ACD, UCD, Hotel/Motel Wakeup, Automatic Attendant Overflow and Automated Attendant.

For ACD, Attendant Automatic Overflow, and Automated Attendant, more than one caller at a time (a listen only conference) can listen to a recording in the recording hunt group. For UCD and Hotel/Motel Wakeup, only one caller at a time can listen to a recording in the recording hunt group.

For some applications, there is support for various kinds of recording devices and support for various recording device failures. These features only apply when the members of the group are accessed through the group - they do not operate when an individual member of the recording hunt group is called directly by dialing its individual extension number.

When a recording is playing for ACD, Automatic Attendant Overflow, and Automated Attendant, the Recording Message Length Timer (programmed on the recording group) applies. When the timer expires and the recording is still playing, the system terminates the recording. For some devices this acts as a safety timer for recording devices that fail to terminate their recording. For tape based recording devices, this allows the system to not play the dead time at the end of the tape.

When an error occurs, a maintenance log is created and the failed recording device is put into the Do Not Disturb state. There are two error conditions handled for members of recording groups:

- When a RAD does not answer a call within 20 seconds, the ringing is canceled and the recording is put into Do Not Disturb.
- When the system ends the recording, due to the message length timer or the caller hanging up, then the Recording Failure To Hangup Timer (in the recording device’s COS) is started (not for UCD or Hotel/Motel Wakeup recordings). If the recording fails to hang up during this time then the recording is put into Do Not Disturb.

See Attendant Automatic Overflow, Automated Attendant, Uniform Call Distribution (UCD) and Reminder. Also see Automatic Call Distribution and refer to the ACD TELEMARKETER Application Package section. For information about the RAD features of the embedded voice mail application, click here.

Conditions

The following conditions apply to this feature:

- Hunt group overflow does not apply to recording hunt groups.
- Industry-standard telephones only can be members of recording hunt groups.
- There is no limit to the number of simultaneous listeners to a recording.
- Recalls never occur to recordings that are not directly dialed.
- Trunks (all types) are answered before listening to a recording.

The following restrictions apply to members of recording hunt groups:

- they cannot have line appearances of them programmed on any Mitel telephone in the system
- they cannot originate calls (regardless of any options in the system)
- a flash is always ignored
- they cannot have call me back messages left at them
- they cannot be overridden
- campon warning tone is never applied
- they cannot be an alternate music source port in ACD
- calls ringing them cannot be picked up using the Directed Pickup feature
- the Last Party Clear Dial Tone feature does not operate for recording devices
- standard ringing is always applied when ringing.

Programming

For recording hunt groups used in Automatic Call Distribution, Attendant Automatic Overflow, and Automated Attendant, select a time for COS Option 404 (Recording Failure to Hangup Timer) in the COS of the recording hunt group members.

Since some RADs won't answer calls that use discriminating ringing (non-standard ringing), ensure that COS Option 809 (Standard Ring Applies) is enabled in the COS of the incoming trunks.

See CDE Form 17 (Hunt Groups) for details on programming the hunt group itself.

Operation

Not applicable.

Recall

Description

The recall feature ensures that calls do not remain unanswered or on hold for an extended period. Any call that has been extended by a console, or an external call that has been extended by an extension to another party, recalls the console or extension if the call is not answered or remains on hold at the end of a timeout period.

The recall feature also works for outgoing external calls. When a trunk is seized, the calling party becomes the recall point. If the trunk is transferred somewhere in the system, recall is by default to the party that made the call.

Setting Up the Recall Point to consoles

When a console answers a call, regardless of where the call came from, that caller's recall point is always set to be that console. The call recalls back to the console (at the Recall call position) unless some other recall feature has been added.

Performing an unsupervised transfer of a call to another party in the system does not change the recall point of the transferred party. For a supervised transfer, the recall point of the transferred party (if it is an internal party) is cleared, which prevents any recall back to the console. If the console is transferring a non-serial trunk to a subattendant extension, the recall point is set to be the subattendant.

A console also becomes a trunk's recall point if the console ever becomes alone in a conversation with the trunk. This could occur under three circumstances:

- an extension does a supervised transfer of a trunk to a console, and hangs up while the trunk is still softheld.
- an extension does a supervised transfer of a trunk to a console, the three are conferenced by the console, the transferring extension then hangs up.
- a console transfers a trunk to another console, and hangs up while the trunk is softheld.

Setting Up the Recall Point to extensions

Extensions only affect the recall point of trunk calls when the trunk is answered (this includes answering calls via features such as Auto-Answer, directed pickup or TAFAS). The recall point is only set to the answering extension if the trunk call is not a serial trunk and the trunk has not had a recall point set up already. If the extension is a subattendant, and the trunk is not a serial call trunk, then the trunk recall point is changed even if it is already set-up.

When any party performs a Hold Pickup Access (see Attendant Paged Hold Access) from an Attendant Hold Slot (Feature Access Code Number 16) it becomes the new recall point for the picked up party, provided that the picked up party was not a serial call trunk.

The recall feature works for outgoing external calls as well. When a trunk is seized, the recall point is set to be the calling party. If the trunk is transferred somewhere in the system, recall is by default back to the party that made the call unless the recall point is altered.

When A Recall Is Done

Recall is done when ringing a device, camping on to a device, or being held by a device.

When ringing another device in the system and forwarding is done, then the Recall No Answer Timer is started for the device and when the timer expires, a recall is attempted.

The time after which the recall occurs is determined by COS Option 254, Call Hold Recall Timer (PBX telephone). The time may be set from 0 to 10 minutes. The recall timer is dependant on COS Option 254 set to an appropriate non-zero time and COS Option 681 (Key Set/Sub Att. - Call Hold Notify Timer) set to "0".

If calling an LDN and the Attendant Calls Forwarded On No Answer feature is active then the timer for that feature is started instead; see Attendant Calls Forwarded On No Answer.

When camping on to another device, the attendant campon recall timer is started. When the timer expires, a recall is attempted.

When a call is held by an industry-standard telephone after the call hold Recall timer timeout, (COS Option 254), a recall is attempted if the industry-standard telephone is not idle.

Recall Processing

When a recall is attempted, the following decisions are made:

- if ringing a console, LDN, or Night Bell, then no recall at all is done.
- if the caller is a CO or DISA trunk and the Non Dial-in Alternate Recall Point is defined and the trunk is not already calling that point then a reroute is done to that point; see Alternate Trunk Recall.
- if a DID or Tie trunk is ringing a party, then DID/Tie Trunk Routing On No Answer is checked. This is only done if the recall point is not set up for the caller; see DID/Dial-in/Tie Intercepts.
- if the recall point has not been set up, then the Final Ringback Timer is started if the caller is recalling from ringing. Otherwise recall is attempted again after 10 seconds.

- if the recall point is set up and is not busy, then that point is called.
- if the recall point is set up and is busy, and if recalling from ringing, then the Recall No Answer Timer is started again. Otherwise, recall is attempted again after 10 seconds.

Conditions

The following conditions apply to this feature:

- Ringing an extension from a hold timeout for Mitel telephones or industry-standard telephones that are idle or have Do Not Disturb activated is handled as a recall after a ring no-answer timeout.
- Unlike a reroute point, a busy recall point does not have Recalls camp onto it. If the recall point has Do Not Disturb activated then no recalls to the point are done.
- A Mitel telephone with the Subattendant - Basic Function feature has less stringent tests for being available to handle a recall; see Subattendant.
- The serial call feature is similar to the basic recall in that it sets up the recall point for the serial trunk; see Attendant Serial Call.
- Recall occurs from campon after the Attendant-Timed Recall (CAMPON) time. The timer value is taken from the caller's recall point's COS. If no recall point has been setup then the timer value is taken from the caller's COS. If the timer value is 0 in the caller's COS then the final ringback timer is started and no recall is done.
- For a recall from campon to the console or an LDN key, the recalling party is not removed from campon. If the busy destination becomes available before the recall is answered at the console then the recall stops and the recalling party rings the now available party.
- For recall from ringing, the Attendant-Timed Recall (NO ANSWER) timer value is taken from the caller's recall point's COS. If no recall point has been set up then the timer value is taken from the caller's COS. If the timer value is 0 in the caller's COS then the Final Ringback Timer is started and no recall is done.
- When a recall timer is needed for the recall point, if the recall point is an LDN key then the COS of the console with the lowest bay/slot/circuit PLID where the LDN is programmed is used.
- No recall is done when a device is receiving busy tone and it does not camp on to the busy device.
- No recall of any kind is done for Direct Trunk Select and private trunks that are ringing in.
- Listening to recordings has no effect on recall for Uniform Call Distribution, Automated Attendant, Attendant Automatic Overflow, and Automatic Call Distribution callers.
- When a reroute is performed for UCD Busy Agent timeout, the recall point for the waiting caller is cleared.
- Whenever an internal caller talks to another device, the recall point of the internal caller is cleared.
- Enabling repeated campon beeps for a trunk prevents recall from campon; see Campon Warning Tone.
- The Auto-Answer feature is ignored for recalls directly back to a Mitel telephone.
- See Attendant Hold Positions and Subattendant Hold Positions and Hold for details on Hold Timeout handling.
- A console never recalls to any other device.
- A call recalls to extensions and consoles through a recall point. Recall to other device types must use features available through the Call Rerouting table.
- When a Mitel telephone answers a call on a non-prime line, the recall point for the caller, if it is altered, is set to the answering Mitel telephone and not to the line that was answered.
- Recalls to a console recall call position are directed to the Recall call position of a group of consoles if the Transparent Multi-Console Operation feature is used.
- Subattendant Recalls are not affected by Transparent Multi-Console Operation feature.
- If COS Option 242 (Repeated Camp-On Beeps) is enabled, Attendant Timed Recall (Campon) is disabled.

- COS Option 681(Key Set/Sub Att. - Call Hold Notify Timer) must be set to zero.

Programming

Select values for COS Options 117 (Attendant-Timed Recall - CAMPON), 115 (Attendant-Timed Recall (NO ANSWER) and System Option 51 (Final Ring Timeout) for the recall point, console, or extension, or in the caller's COS (see Conditions to determine which COS is used); see Attendant-timed Recall.

Program the desired time for the Recall timer in COS Option 254, Call Hold Recall Timer (PBX telephone). The recall timer is dependant on COS Option 254 set to an appropriate non-zero time and COS Option 681(Key Set/Sub Att. - Call Hold Notify Timer) set to zero.

Operation

None.

Receive Only Extensions

Description

An industry-standard telephone with this class of service (COS) option, can receive calls but cannot originate calls. The industry-standard telephone may, however, originate calls and select features specified in its COS after having received a call, and placed the call on hold by flashing.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- If the station tries to originate a call, the attempted call is ignored.
- If used in conjunction with the Flash Disable feature, ALL types of call origination are blocked.
- See Never A Forwardee and Callbacks.
- COS Options 241 (Receive Only) and 400 (Contact Monitor) are mutually exclusive.

Programming

Enable COS Option 241 (Receive Only) for the extension.

Operation

None.

Record a Call

Description

Record a Call is a feature of the embedded voice mail application. It allows you to record both ends of a two-party conversation (internal or external call) in progress at your set. The recorded conversation is stored in a voice mail mailbox. You have the option of using the ports that store your voice mail messages or you can set up voice mail ports exclusively for recorded messages. Constraints driving this decision are usually based on the available resources (DSP card type and Business 1, 2 or Hospitality options) within the SX-200 ICP. Optionally, you can forward recorded conversations to e-mail. Record a Call is a purchasable option.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Record a Call should only be used in conjunction with the laws of the jurisdiction where the call is placed from and/or the laws of the jurisdiction of the place being called. Mitel is not liable for misuse of this feature in a manner that does not conform with the applicable law; for example: laws involving wiretapping, eavesdropping, electronic surveillance, call recording etc. Dealers should warn the customer in writing that they are responsible to use this feature in accordance with the law and that in many jurisdictions both parties must be made aware that the call is being recorded in order to comply.
- Record a Call may be used for internal and external calls.
- Conference calls can not be recorded with the Record a Call feature.
- This feature is not compatible with a centralized voice mail system. You can only record a call if your voice mail is on the same system as your extension.
- No tones will be given to either party to indicate that the call is being recorded.
- Phones that support Record a Call can use a Record a Call feature key to initiate a recording session. The consoles can access their own mailbox and access messages using the Tones On feature. Messages can be sent or forwarded manually to that mailbox.
- The SUPERSET 4150 and the Mitel 5340 IP telephone user can use a Record a Call softkey to initiate a recording session. These two telephones are the only sets that have this softkey and the only sets that show "Recorder Unavailable" on the display when all the recording ports are busy.
- All of the Mitel sets can use a programmed Record a Call feature key to activate Record a Call and to access the recording device for voice mail messages and call recordings. The LED on the Record a Call feature key will be on solid while the recording is on; flashing while the recording is on pause; off when the recorder is unavailable. The recorded message will not turn the message waiting lamp on the set of the user.
- Each Record a Call session uses a conference bridge. The DSP resources installed in the system determines the maximum number of conferences that can take place at one time; see DSP Configuration Options.
- Record a Call messages will follow the e-mail path programmed against voice mail ports for the Voice Mail to e-mail feature. This will allow users to delete their messages on Voice Mail while keeping their messages in e-mail.
- The SX-200 ICP DSP modules will increase the number of regular conference call circuits for Bay 1 (IP extensions).

	With Default Dual (2) DSP	With an additional Dual (2) DSP = 4 DSP	With an additional Quad (4) DSP = 6 DSP	Default Dual replaced with Quad + another Quad = 8 DSP
Business Option 1 (CX/CXi)	3 conf x 3 parties	10 conf x 3 parties	10 conf x 3 parties	NA
Hospitality (AX)	N/A	10 conf x 3 parties (Base configuration)	10 conf x 3 parties (with an additional Dual)	10 conf x 3 parties
Business Option 1	3 conf x 3 parties	8 conf x 3 parties	12 conf x 3 parties	12 conf x 3 parties

	With Default Dual (2) DSP	With an additional Dual (2) DSP = 4 DSP	With an additional Quad (4) DSP = 6 DSP	Default Dual replaced with Quad + another Quad = 8 DSP
(MX)				
Business Option 2 (MX only)	8 conf x 3 parties	12 conf x 3 parties	18 conf x 3 parties	21 conf x 3 parties
Hospitality Option (MX only)	8 conf x 3 parties	8 conf x 3 parties	12 conf x 3 parties	12 conf x 3 parties

- Maximum number of simultaneous Record a Call calls is 12.
- Conversations are recorded manually, if the Class of Service for the set initiating the recording has Record a Call in Voice mail (option 268) enabled and the Record a Call: Start Recording Automatically (option 269) disabled.

  Note: Each Record a Call will record at a rate of approximately 8000 bytes per second. Record a Call users must perform proper mailbox maintenance and delete unimportant recordings on a consistent basis so that the Voice Mail system is not filled with messages. The user can also use the CANCEL key to abandon and erase the recording.
- Conversations are recorded automatically if the Class of Service of the IP Phone or extension has Record a Call (option 268) enabled and the Record a Call: Start Recording Automatically (option 269) enabled.

  Notes:

  1. COS Option 269, Record a Call: Start Recording Automatically, enables the automatic recording of external calls, incoming and outgoing. Option 269 only applies to external calls. The recording of internal calls must be started manually.
  2. If calls are automatically saved on hangup or manually saved on a consistent basis, the mailbox will fill quickly. Each Record a Call will record at a rate of 8000 bytes per second. Therefore, Automatic Record a Call must be restricted to certain phones in a controlled environment. These users must perform proper mailbox maintenance and delete unimportant recordings on a consistent and daily basis. The user can also use the CANCEL key to abandon and erase the recording
- To save a conversation automatically when the user hangs up or places the call on Transfer or Hold, the Class of Service of the set initiating the recording must have Record a Call: Save Recording on Hang up, option 270 enabled. The constraints in Note 2 above also apply.
- While the recording device is connected or on pause, the telephone user that has initiated the recording can
  - Pause, Resume, Save, or Erase the conversation. Mitel 5010, 5212, 5215, and 5312 IP Phones, and SUPERSET 4015 telephone do not have softkeys, therefore these telephone type users will not be able to pause and resume the recording.
  - Switch between the handset and handsfree.
  - Hang up (the voice mail port disconnects).
  - Place the call on Transfer or transfer a call using a DSS/BLF key (the voice mail port disconnects).
  - Place the call on Hold (the voice mail port disconnects).
    A new recording will start automatically if you retrieve the Hold on your set.

Note: Campon, Whisper Announce or Off-hook Voice Announce will not function in order to restrict intrusion. The COS option 266 , Camp-on before Forward on Busy, will be ignored for the set of the caller, if the caller calls a busy extension that is in a recording session.

- While the recording device is connected or on pause, the recorded party on an internal set or station can
  - Switch between the handset and handsfree
  - Hang-up. The display of the set or console initiating the recording shows that the call has hung up and allows the person to pause and resume the recording (allows the user to make additional comments to the recording).
  - Put the call on hard (red button) hold, either directly or by selecting another line. The display of the set initiating the recording shows that the call is on hold and allows the person to pause and resume the recording.

Note: If the recorded party is on the same system, the party that is being recorded cannot also invoke a recording, of the same conversation, once the other party has done so. If the recorded party is an external party (on another PBX), the person can invoke a recording.

- While the recording device is connected or on pause, the console user can perform the following actions relating to the call:
  - Pause, Resume, Save, or Erase the conversation
  - Switch between the handset and handsfree
  - Release (the voice mail port disconnects)
  - Place the call on Consultation Call with direct dial or with a DSS/BLF key (the voice mail port disconnects)
  - Place the call on Hold (the voice mail port disconnects)
    A new recording will start automatically if you retrieve the Hold on your set.
- An ACD agent may record a two-party conversation while being monitored by a supervisor, but this action will cause the monitoring party to be dropped.
- If an ACD supervisor is monitoring an ACD agent, the ACD supervisor cannot use Record a Call.
- The voice mail system saves the recording if the system disconnects before the voice mail receives a SAVE or ERASE command.
- The Record a Call VM ports must be in the same COS where COS Option 229 (Voice Mail Port) is enabled.
- In Form 17 (Hunt Group under Options), the user determines how many VM ports of the hunt can be used for Record a Call. For example, if the system only has resources for 4 VM ports, the user should not create a separate group for Record a Call because it will reduce the overall number of VM ports used for VM. In this scenario, the user should also not set the Maximum Ports allowed option to four since all four VM ports could get used for Record a Call instead of Voice Mail.
- With the Suite Services feature, Record a Call can be used by any DNIC telephone in the suite but all of the recordings will be in one mailbox (the primary DN).
- The Speed Dial softkey is not provided in CDE when you are programming Record a Call.

Note: **Record a Call is not intended for use in critical applications such as E911 call centers**.

Programming

- **System Options:** (Form 04); enable the purchasable System Option, Record a Call, option 87.
- **Feature Access Codes:** (Form 02); choose an access code for Send Message (feature number 41 - *41 by default) if you are sharing the Record a Call ports with voice mail messages.
- **COS Options:**(Form 03)

	COS of Voice Mail Port being used for Record a Call	COS of Sets using Record a Call	COS of Attendants using Record a Call
Attendant Tone Signalling (option 119)	not required	not required	enable

	COS of Voice Mail Port being used for Record a Call	COS of Sets using Record a Call	COS of Attendants using Record a Call
Voice mail Port (option 229)	enable	not required	not required
Record a Call in Voice mail (option 268)	enable	enable	enable
Record a Call: Start Recording Automatically (option 269)	not required	option, see conditions above	optional, see conditions above
Record a Call: Save Recording on Hang up (option 270)	not required	optional, see conditions above	optional, see conditions above

- **Hunt Groups:** (Form 17) under OPTIONS; enter the maximum number of ports allowed for Record a Call with the Record a Call: Maximum Ports Allowed <1-20> option. See conditions above.
- **Call Rerouting Table:** (Form 19); enter the Record a Call Voicemail Destination For This Tenant. This can be the separate VM group utilized strictly for Record a Call or the VM hunt group number if you wish to share the voice mail system for voice mail and record a call.
- **Mailboxes:** (Form 50); program a mailbox for each extension that requires use of Record a Call.

  (Optional) To allow Record a Call messages to be forwarded to e-mail,
  - Press SHOW MORE
  - Press EMAIL
  - Press SHOW MORE
  - Select a mailbox and enter the e-mail address for Record a Call messages. If no address is entered, Record a Call messages are sent to the e-mail address for voice messages (if programmed).
- On the Mitel 5010, 5212, 5215, 5020, 5220 and 5224 IP Phones, and SUPERSET 4015, SUPERSET 4025, SUPERSET 4125, SUPERSET 4150 telephones, program a feature key for Record a Call. See Feature Keys.

  Refer to the installation guide provided with the Mitel Express Manager Card for information on programming it.

Operation

Mitel 5010, 5212, 5215, 5020, 5220, 5224, 5330, and 5340 IP Phones, SUPERSET 4025, SUPERSET 4125, and SUPERSET 4150 telephones, 5540 IP Console, SUPERCONSOLE 1000

To record a call manually

- While you are in a two-party conversation, press the Record Call feature key (A SUPERSET 4150 telephone user or console user can press the Record a Call softkey).

  The LED of the feature key on the telephone will be on solid while you are recording.

  If the recording device is not available, press the Cancel softkey or CANCEL to return to the two party talk state.

  Pressing the Pause and Resume softkeys lets you have control over the voice mail to pause and resume the recording. The LED of the feature key on the telephone will flash while the recording is on pause.

To discard the recording, disconnect the voice mail and return to the two party call state (COS Option 270 disabled)

- Press the Stop and Erase, Erase softkey, or CANCEL.

To stop the recording, save the recording in your mailbox, and return to the two party call state

- Press the Stop & Save, or Save softkey.

To listen to a recording

- With the telephone, lift the handset, dial the number to access your voice mail box, and follow the prompts to retrieve the recording.
- With the console, dial the number to access your voice mail box, and follow the prompts to retrieve the recording.

Family 3 Telephones

To record a call manually

- While you are in a two-party conversation, press the Record Call feature key. The LED goes on solid when recording.

To stop and save a recording

- Press the Record Call feature key. The LED goes off when not recording.

To stop and erase a recording

- Press the Cancel key.

To listen to a recording

- Lift the handset, dial the voice mail number to access your voice mail box, and follow the prompts to retrieve the recording.

Reminder

Description

Extension users can program their telephones to provide a ringing reminder at a particular time within a 24-hour period. This feature acts as a reminder for appointments, meetings, etc.

Display sets behave differently depending on the COS Option 322, Confirm Wakeup by Off-hook. If this option is disabled, Family 1 , Family 4 and Family 5 telephones receive one ring every minute and the display flashes, REMINDER EXPIRED and Confirm. If this option is enabled, the display set will ring three times.

Non-display sets receive five rings (because they don't have a display).

Two types of reminders exist:

- Single reminder. This feature is supported with System Option 102, Feature Level disabled.
- Multiple reminders (up to three timers within a 24-hour period) with the option of repeating daily.This feature is supported with System Option 102, Feature Level enabled to level 1 or greater.

Hotel/Motel guests may set up wakeup times (reminders) from the room telephone if the Lodging system or a Property Management System is used.

Hotel/Motel staff using Family 1 and Family 5 telephones programmed as a subattendants may set up wakeup alarm calls for their guests.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

- Multiple reminders require FOR Option 102, Feature Level 1 or greater. Without it, the Reminder feature only offers one reminder (a single reminder). TIMER 1 is the one reminder.
- SUPERSET 4DN telephones do not support multiple reminders. Only one reminder exists, TIMER 1.
- Do Not Disturb has no effect on the Reminder feature.
- Do not enable Auto-Answer on telephones that are to receive a reminder or wake-up call. If Auto-Answer is enabled on the telephone, the reminder call will not be presented to the set.
- A reminder will occur each day at the programmed time until it is acknowledged.
- A daily reminder repeats every day at the set time until the reminder time is canceled.
- If COS Option 322, Confirm Wakeup by Offhook, is enabled, the set will ring until the handset goes offhook (or the ringing timeout expires).
- Reminder settings are not lost with system upgrades.
- An asterisk beside an active time on a small display set (Mitel 5020, 5220 and 5224 IP Phones, and SUPERSET 4025, SUPERSET 4125 and SUPERSET 420 telephones) indicates a timer that repeats daily at the same time until canceled.

Programming

Enable COS Option 202 (Alarm Call) in the extension's COS.

Enable COS Option 322, Confirm Wakeup by Offhook if the user wants to acknowledge the wakeup alarm by going offhook on the ringing set.

Set the FOR Option 102, Feature Level, in CDE Form 04 to level 1 or greater.

To allow industry-standard telephones, Family 2, Family 3, SUPERSET 410, and SUPERSET 3DN telephones to use this feature, enable System Option 11 (Automatic Wakeup), and assign an access code to Feature 32 (Automatic Wakeup).

Operation

Operation varies depending upon the type of set as described below.

Family 2, Family 3, SUPERSET 410, SUPERSET 3DN and Industry-Standard Telephones:

To set or modify a single reminder using the access code:

Note: System Option 102, Feature Level is disabled.

- Obtain dial tone.
- Dial the Automatic Wakeup code.
- Dial in the reminder time. The time is specified by dialing the hour and minutes in 24-hour time format.
- Reorder tone is returned for an invalid time; dial tone is returned for a valid time.

To set multiple reminders (up to three) using the access code:

- Obtain dial tone.
- Dial the Automatic Wakeup code (feature access code 32).
- Dial the pound character and the number of the timer (#1, #2, or #3). If you wish to repeat this timer daily, dial the asterisk (*).
- Dial in the reminder time. The time is specified by dialing the hour and minutes in 24-hour time format.
- Reorder tone is returned for an invalid time; dial tone is returned for a valid time.

To cancel a single reminder (or Timer 1):

- Obtain dial tone.
- Dial the Automatic Wakeup code, followed by four 9's.
- Replace the handset. The reminder is canceled.

To cancel multiple reminders (Timer 1, Timer 2 or Timer 3):

- Obtain dial tone.
- Dial the Automatic Wakeup code.
- Dial the pound character followed by the timer number (#1, #2, or #3).
- Dial four 9's.
- Replace the handset. The reminder is canceled.

To acknowledge the reminder

- You hear ringing.
- Go offhook to acknowledge the reminder.

Family 4 Telephones :

To set up a single reminder:

Note: This procedure reflects the 12-hour clock format programmed with Option 1 (Clock Format) in CDE Form 04. If 24-hour format is set, then the AM and PM are not included. System Option 102, Feature Level is disabled.

- Press **Superkey**.
- Press **No** until TIMED REMINDER? appears in the display.
- Press **Yes**. TIMER 1 with HH:MM appears.
- Dial the desired time in a 12-hour format (e.g., 01:45). In case of error, use the ← key.
- Specify AM or PM by pressing **AM/PM**. The top AM or PM reading on the display is the reading that will be programmed.
- Press **Save**. REMINDER SAVED appears briefly in the display.

To set up multiple reminders (up to three timers)

Note: Programming multiple reminders is dependent on the FOR Option, Feature Level, being set to level 1 or greater. The following procedure reflects the 12-hour clock format programmed with Option 1 (Clock Format) in CDE Form 04. If 24-hour format is set, then the AM and PM are not included.

- Press **Superkey**.
- Press **No** until TIMED REMINDER? appears in the display.
- Press **Yes**. TIMER 1 with HH:MM appears.
- Dial the desired time in a 12-hour format (e.g., 01:45). In case of error, use the ← key.
- Specify AM or PM by pressing **AM/PM**.
- Press **Save**. REPEAT DAILY? appears with Yes and No softkeys.
- Press **Yes** or **No** depending on your requirement. REMINDER SAVED appears briefly in the display.
- Repeat the first three steps.TIMER 1 displays.
- Press **Next** . TIMER 2 appears.
- Continue to program TIMER 2 in the same manner as TIMER 1.
- Repeat the first three steps again until TIMER 1 appears.
- Press **Next** twice to display TIMER 3. Continue to program TIMER 3 in the same manner as TIMER 1.
- Press **Superkey** to exit.

To display the current reminder settings:

- Press **Superkey**.
- Press **No** until TIMED REMINDER? appears in the display.

- Press **Yes**. The reminder time for TIMER 1 appears (an asterisk beside the time indicates that the timer will be repeated daily, HH:MM indicates that the timer has not been programmed).
- Press **Next** to display the time for TIMER 2.
- Press **Next** again to display the time for TIMER 3.
- Press **Superkey** to return to the normal display.

To cancel a reminder before it occurs:

- Press **Superkey**.
- Press **No** until TIMED REMINDER? appears in the display.
- Press **Yes**. The TIMER 1 display appears. If you choose to cancel a reminder for another timer, press **Next** till you see the required timer.
- Press **Del**. REMINDER CANCEL appears briefly in the display.
- Press **Superkey** to return to the normal display.

To change a reminder before it occurs:

- Press **Superkey**.
- Press **No** until TIMED REMINDER? appears in the display.
- Press **Yes**. The TIMER 1 display appears. If you choose to change a reminder for another timer, press **Next** till you see the required timer.
- Press **Change**. ENTER TIME HH:M M appears.
- Dial the desired time in a 12-hour format (e.g. 01:45). In case of error, use the ← key.
- Specify AM or PM by pressing **AM/PM**.
- Press **Save**. REPEAT DAILY? appears with Yes and No softkeys (the REPEAT DAILY? screen only appears if the FOR option Feature Level is set to level 1 or greater).
- Press **Yes** or **No** depending on your requirement. REMINDER SAVED appears briefly in the display.

To acknowledge a reminder (COS Option 322, Confirm Wakeup by Offhook, is disabled on the ringing set):

- You hear a short burst of ringing. REMINDER EXPIRED is shown in the display.
- Press **Confirm** to acknowledge the alarm. ACKNOWLEDGED appears briefly in the display.

To acknowledge a reminder (COS Option 322, Confirm Wakeup by Offhook, is enabled on the ringing set):

- You hear ringing. REMINDER EXPIRED is shown in the display.
- Go offhook to acknowledge the alarm.

Family 5 Telephones:

5330 IP Phones

See Family 4 instructions.

5340 IP Phones:

Symbol MiNET Wireless Phone:

To program a Timed Reminder:

1. Press the No softkey until "Timed Reminder" appears.
2. Press the Yes softkey.
3. Enter the time in in HH:MM format (e.g, 02:00 for 2 o'clock).
4. Press the right softkey to select AM or PM.
5. Press the SAV softkey.

"Repeat Daily?" appears.

6. Press the Yes or No softkey.
7. To program additional timers, press the NXT softkey and then repeat the above steps.
8. Press the SAV softkey.
9. Press the END key.

To view, change, and/or cancel a pending Reminder:

1. Press the No softkey until Timed Reminder appears.
2. Press the Yes softkey. Perform the following where applicable:
   - To change the Reminder, press the CHG softkey, enter the new time, and press the SAV softkey.
   - To cancel the Reminder, press the DEL softkey.
   - To exit without canceling the Reminder, press the END key.
   - To acknowledge a Reminder when your phone rings once press the CFM softkey.

SUPERSET 4150 and SUPERSET 430 Telephones:

To set up a single reminder:

Note: System Option 102, Feature Level is disabled.

- Press **Superkey**.
- Press **Reminder**.
- The display showing TIMER 1 appears. Dial the desired time in a 12-hour format (e.g. 01:45). In case of error, use the ← key.
- Specify AM or PM by pressing **AM/PM**.
- Press **Save**.

To set up multiple reminders (up to 3 timers):

Note: Programming multiple reminders is dependent on the FOR Option, Feature Level, being set to level 1 or greater. The following procedure reflects the 12-hour clock format programmed with Option 1 (Clock Format) in CDE Form 04. If 24-hour format is set, then the AM and PM are not included.

- Press **Superkey**.
- Press **Reminder**. The display showing TIMER 1 appears. Timer 2, Timer 3, Backup, and Same Time softkeys appear. The Backup softkey allows you to return to the previous screen.The Same Time softkey appears if you wish to reset the timer with the previous time.
- Dial the desired time in a 12-hour format (e.g. 01:45). In case of error, use the ← key. AM and ONCE appear after the time is entered. These are default settings.
- Press **PM** if required.
- Press **Daily** if you wish to have the timer repeated daily (the Daily softkey only appears if the FOR option Feature Level is set to level 1 or greater). DAILY appears after pressing the Daily softkey.
- Press **Save**.
- Repeat the first two steps.
- Press **Timer 2** to access the TIMER 2 display.
- Program the reminder time for TIMER 2 using the same procedure for TIMER 1.
- Repeat the first two steps again to access the TIMER 3 display.
- Program the reminder time for TIMER 3 using the same procedure for TIMER 1.

To acknowledge a reminder:

- You hear one short burst of ringing and REMINDER EXPIRED is shown in the display.
- Press the flashing **Acknowledge** softkey to acknowledge the alarm. ACKNOWLEDGED appears briefly in the display.

To display the reminder settings:

- Press **Superkey**.
- Press **Reminder**. The reminder setting for TIMER 1 is displayed. DAILY-AUTO appears if the reminder repeats on a daily basis. ONCE-AUTO appears if the reminder is set for a one time occurrence.
- Press **Timer 2** or **Timer 3** to display the reminder setting on the respective timers.
- Press **Superkey** to return to normal display.

To cancel a reminder before it occurs:

- Press **Superkey**.
- Press **Reminder**. The display showing the time setting for TIMER 1 appears. Pressing **Timer 2** or **Timer 3** displays the respective time setting for the chosen timer.
- Press **Cancel**.

To change a reminder:

- Press **Superkey**.
- Press **Reminder**. The display showing the time setting for TIMER 1 appears. Pressing **Timer 2** or **Timer 3** displays the respective time setting for the chosen timer.
- Dial the desired time in a 12-hour format (e.g. 01:45). In case of error, use the ← key. AM and ONCE appear after the time is entered. These are default settings.
- Press **PM** if required. The default is AM.
- Press **Daily** if you wish to have the timer repeated daily. DAILY appears after pressing the Daily softkey.
- Press **Save**.

SUPERSET 4DN Telephones:

To set up a reminder:

- Press **Superkey**.
- Press **Reminder**. The display showing TIMER 1 appears.
- Dial the desired time in a 12-hour format (e.g. 01:45). In case of error, use the ← key.
- Specify AM or PM by pressing **AM/PM**.
- Press **Save**.

To acknowledge an alarm that has occurred:

- You hear one short burst of ringing and a REMINDER EXPIRED is shown in the display.
- Press the flashing **Acknowledge** softkey to acknowledge the alarm. ACKNOWLEDGED appears briefly in the display.

To display the current reminder setting:

- Press **Superkey**.
- Press **Reminder**; the reminder time for TIMER 1 is displayed.
- Press **Superkey** to return to normal display.

To cancel a reminder before it occurs:

- Press **Superkey**.
- Press **Reminder**. The reminder time for TIMER 1 is displayed.
- Press **Cancel**.

5020, 5220 and 5224 IP telephones

To remotely set up Do Not Disturb on an extension from the subattendant position set:

- Press **Superkey**.
- Press the **TIMED REMINDER?** softkey. TIMER 1 with HH:MM appears.
- Dial the desired time in a 12-hour format (e.g. 01:45). In case of error, use the ← key.
- Specify AM or PM by pressing **AM/PM**.
- Press **Save**. PERSONAL? appears with Yes and No softkeys.
- Press **Yes** for a regular reminder **No** for a reminder that is repeated daily. REPEAT DAILY? appears when No is selected.
- Press **Yes**.
- Press **Save**.
- Press **Superkey** to exit.

5330 and 5340 IP telephones

Refer to the 5330/5340 IP Phone User Guide available at Mitel OnLine.

Remote LAN Access (Network-side Interface)

Description

Remote LAN Access provides access to the wide area network (WAN) for LAN servers, such as routers and bridges, using PRI Cards or embedded PRI modules (Dual T1/E1 Framers or T1/E1 Combos). The feature allows LAN access to the ISDN network for both incoming and outgoing voice and data calls.

The Remote LAN Access feature uses two PRI interfaces in the PBX to route incoming and outgoing calls. (See figure) Outgoing LAN calls are routed through the LAN server to the PRI interface and then the PBX. The PRI interface connected to the LAN is identified as a network-side interface. The outgoing calls are then routed through a second PRI interface to the WAN, or public switched telephone network (PSTN).

Incoming calls from the WAN, are routed through a PRI interface to the PBX, and then back through another PRI interface to the LAN server and the LAN behind it. The PRI interface connected to the WAN is identified as a user-side interface. Routing the calls through the PBX allows the use of PBX channel management, ARS routing, and SMDR.

Each link on a PRI card or embedded PRI module can be defined as either a user-side or network-side interface. With the Multiple Variant feature (a standard ISDN Release 6 feature), each link can also be assigned a different protocol. The supported protocols for Remote LAN Access are: DMS-100, DMS-250, 4ESS, and Bellcore NI2.

Conditions

The following conditions apply to this feature:

- The Remote LAN Access feature and ISDN Release 6 upgrades must be ordered from Mitel.
- Two PRI interfaces are needed to implement the Remote LAN Access feature. (The two interfaces can be on the same PRI card or different cards.) Similarly, two trunks are needed: one for the LAN interface, and one for the WAN interface.
- The PRI interface connected to the LAN server must be configured and programmed as a network-side interface:
  - set the NT/LT jumper for that interface to LT (see Installing the PRI Card)
  - in IMAT (PRI card) or CDE (embedded PRI), program the link as a network-side interface (see Programming)
- Any PRI interface involved in the Remote LAN Access topology should be upgraded to ISDN Release 6. LAN data calls should only be directed to PRI interfaces running ISDN Release 6 software.

Programming

Enable Option 93 (PRI Card: Remote LAN Access) in the System Options form.

PRI card

Select QSIG in the IMAT Site Options screen.

Select Network-side Interface/Qsig master for the link connected to the LAN in the IMAT PRI Link Characteristics screen.

Embedded PRI

Select Network as the Embedded PRI Network/User Side option in the T1 Link Descriptor Options form.

Operation

None.

Resale Package

Description

The Resale Package is a method of offering the system's Automatic Route Selection (ARS) "Least Cost Routing" facilities to external users requiring low cost Long Distance calling, much like the offerings of other Common Carriers.

DISA trunks are installed for external access to the system. The external user dials up one of the DISA trunks, enters a verified account code, and dials the desired external number. The Direct to ARS feature can be used to route the caller directly to ARS.

This feature is actually a specialized application of the Automatic Route Selection, Toll Control, and Verified Account Code features.

Conditions

The following conditions apply to this feature:

- Verified Account Codes can be activated and/or deactivated for problem accounts.
- See Automatic Route Selection and Verified Account Codes.

Programming

Program DISA trunks in CDE Forms 01 (System Configuration), 13 (Trunk Circuit Descriptors), and 15 (Dial-In Trunks) - see Trunk Operation - Direct Inward System Access (DISA).

Program verified account codes in CDE Form 33 (Account Code Entry); also see Verified Account Codes (Special DISA).

Operation

Dial into system via DISA trunk.

Enter a verified account code.

Make call as required.

Ring Groups

Description

This feature provides the ability to ring all members of a group simultaneously. Consoles and night bells can be ring group members in addition to stations and sets. Ring Groups can also:

- ring one member of a group at a time.
- ring all members if the group is set to ring one at a time and no one answers
- allow a group member to appear in more than one ring group (to a maximum of 25)

Conditions

The following conditions apply to this feature:

- The Ring Groups feature requires the use of the ASU II for proper ring function.
- Ring group members can not be members of other types of hunt groups.
- Stations/sets of ring group members can not appear as keyline/MCL anywhere in the system.
- Users can not place callbacks or camp on to a ring group.
- Maximum number of ring groups per system is 25 (maximum of 50 members per group, or 32 non-IP members)
- Ring group members should avoid the following features:
  - ONS Ring Groups
  - Twinning
  - Guest Suites
  - ACD

The following features are ignored for members of a ring group for calls routed to the ring group pilot number:

- Auto Answer
- Call forwarding (tenant- or set-based)
- Intercom Mode

Programming

Program ring groups in CDE Form 17(Hunt Groups) using Group Type "Ring Group".

Operation

Press the RING ALL softkey to ring all free group members simultaneously. If all members are busy, the caller is queued to this group. If the ring timer expires and the call is not answered, the system rings the overflow point.

Press RING ONE to ring one group member at a time (in the order programmed in Form 17). If the ring timer expires and the call is not answered, the system will ring the next member. If all members are busy, the caller is queued to this group. If the ring timer expires and the end of the group is reached, the "Ring All Members" option (Yes/No) determines whether all members are called (Yes) or the system rings the overflow point (No).

Ringer Control

Description

You can adjust the ringer in a Mitel telephone or attendant console.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

SUPERSET 3DN, SUPERSET 4DN, and attendant console users can adjust the ringer volume. Other Mitel telephones can can adjust the ringer volume and the ringer pitch.

Programming

Assign an access code to Feature 27 (Tone Demonstration).

Operation

To adjust ringer volume:

Family 2, SUPERSET 410, and SUPERSET 3DN Telephones:

- While the set is ringing, press the VOL ↑ and VOL ↓ keys to increase and decrease the ringer volume.

Family 3 and SUPERSET 420 Telephones:

Note: On a Mitel 5010, 5212 or 5215 IP Phone, or SUPERSET 4015 telephone, press the dial pad * key instead of the YES softkey, and press the dial pad # key instead of the NO softkey.

- Press SUPERKEY.
- Press the NO softkey until RINGER ADJUST? appears in the display.
- Press the YES softkey. ADJUST PITCH? appears in the display.
- Press the NO softkey.
- ADJUST VOLUME? appears in the display.
- Press the YES softkey.
- Press the VOL ↑ and VOL ↓ keys repeatedly to increase and decrease the ringer volume.
- Press SUPERKEY to exit.

Family 1, and Family 4 Telephones (except SUPERSET 420):

- Press SUPERKEY.
- Press the MORE softkey.
- Press the RING ADJUST softkey.
- Press the RINGER VOL softkey.
- Press the VOL ↑ and VOL ↓ keys repeatedly to increase and decrease the ringer volume.
- Press SUPERKEY to exit.

To adjust ringer pitch:

Family 2, SUPERSET 410, and SUPERSET 3DN Telephones:

- Lift the handset - dial tone is returned.
- Dial the Tone Demonstration Feature access code.
- Dial 33 - the code for Ringer Pitch Adjust (see Tone Demonstration).
- Press the VOL ↑ and VOL ↓ keys repeatedly to increase and decrease the pitch.
- Replace the handset.

Family 3 and SUPERSET 420 Telephones:

- Press SUPERKEY.
- Press the NO softkey until RINGER ADJUST? appears in the display.
- Press the YES softkey. ADJUST PITCH? appears in the display.
- Press the YES softkey.
- Press the VOL ↑ and VOL ↓ keys repeatedly to increase and decrease the pitch.
- Press SUPERKEY to exit.

Family 1 Telephones:

- Press SUPERKEY.
- Press the MORE softkey.
- Press the RING ADJUST softkey.
- Press the RINGER PITCH softkey.
- Press the VOL ↑ and VOL ↓ keys repeatedly to increase and decrease the pitch.
- Press SUPERKEY.

Family 5 Telephones:

To adjust ringer volume while the phone is idle

5330 Phones:

1. Press the key that you have programmed as Superkey.
2. Press the **No** softkey until **Ringer Adjust?** appears.
3. Press **Ring Adjust**.
4. Press **No**.
5. Press **Ringer Vol**.
6. Press Vol ↑ or Vol ↓ to adjust.
7. Press **Superkey**.

5340 Phones:

1. Press the key that you have programmed as Superkey.
2. Press More until the **Rng Adjust** option appears.
3. Press **Ringer Vo**l.
4. Press Vol ↑ or Vol ↓ to adjust.
5. Press **Superkey**.

Attendant Console:

- While the console is ringing, press the VOL ↑ and VOL ↓ keys to increase and decrease the ringer volume.

Ringing - Discriminating

Description

This feature provides two different ringing cadences to allow a user to distinguish between internal calls (standard ringing) and incoming trunk, attendant, or subattendant calls (discriminating ringing). The system can also be programmed to provide discriminating ringing for all calls.

Standard ringing	1 seconds on, 3 seconds off
Discriminating ringing (1)	400 msec on, 200 msec off, 400 msec on, 3 seconds off.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Discriminating Ringing does not apply to Night Bells or Recorded Announcement Devices.
- The feature applies to both the prime and additional lines on a Mitel telephone.
- In an unsupervised transfer situation, the ringing cadence of the called party is not changed after the transfer is done.

Programming

To provide discriminating ringing on incoming trunk and attendant calls:

Enable System Option 17 (Discriminating Ringing).
Disable System Option 18 (Discriminating Ringing Always).

To provide discriminating ringing for all calls:

Enable System Option 18 (Discriminating Ringing Always).

If System Options 17 and 18 are not selected, standard ringing is provided for all calls.

To provide only standard ringing for specific trunks or consoles calling (while still providing discriminating ringing to others), enable COS Option 809 (Standard Ring Applies) for the specific trunks or consoles.

To provide only standard ringing for specific stations (regardless of the ringing cadence determined by the trunk/console caller), enable COS Option 809 (Standard Ring Applies) for the specific station. This option is useful for fax machines or modems that require a standard ringing cadence.

Operation

None.

Ringing Plan

Description

The SX-200 ICP provides the North American ringing plan, used with the tone plan and rotary dial pulse-to-digit conversion features to enable the system to be used in the North American marketplace. The ringing plan is stored in the database on the CMOS. Refer to the Engineering Information section, for ringing plan information.

Conditions

None.

Programming

None.

Operation

None.

Ringing Time-Out (Final Ringback)

Description

A call to an extension can ring for 1 to 30 minutes before the call is dropped (default ringing time is 1 minute).

Conditions

This is only done when forwarding and all recall possibilities have been exhausted; see Call Forwarding and Recall.

Programming

Enable System Option 51 (Final Ring Timeout) and set it to the desired time-out period.

Operation

None.

Rotary Dial Digit Translation

Description

You can program the system to provide one of four Rotary Dial Digit Translation Plans for rotary telephones. The default Rotary Dial Digit Translation Plan is Plan 0.

Table: Rotary Dial Digit Translation Plans

Digit	Plan 0	Plan 1	Plan 2	Plan 3
1	1 pulse	2 pulses	9 pulses	10 pulses
2	2 pulses	3 pulses	8 pulses	9 pulses
3	3 pulses	4 pulses	7 pulses	8 pulses
4	4 pulses	5 pulses	6 pulses	7 pulses
5	5 pulses	6 pulses	5 pulses	6 pulses
6	6 pulses	7 pulses	4 pulses	5 pulses
7	7 pulses	8 pulses	3 pulses	4 pulses
8	8 pulses	9 pulses	2 pulses	3 pulses
9	9 pulses	10 pulses	1 pulse	2 pulses
0	10 pulses	1 pulse	10 pulses	1 pulse

Conditions

Only one translation plan may be in effect at a time.

Programming

Select the desired translation plan via System Option 46 (Rotary Dial Digit Translation Plan).

Operation

None.

Satellite PBX

Description

An SX-200 ICP system can be installed as a satellite PBX. In this configuration, the system has no direct connection to the serving central office for incoming traffic. The satellite PBX has no directory number and receives all its incoming calls over tie trunks from another PBX. Enabling the satellite PBX system option automatically adjusts any required settings for the loss and level plan.

Conditions

Some gain settings for the loss and level plan must be adjusted before the PBX can operate as a satellite PBX. Once the Satellite PBX System Option is enabled, these gain adjustments are done automatically.

Programming

Enable System Option 31 (Satellite PBX).

Refer to the Engineering Information section for loss and level plans.

Operation

None.

Secretarial Line

Description

A Mitel telephone programmed with a secretarial Multicall line appearance of another extension can override the Do Not Disturb feature on the second set. The second telephone has Do Not Disturb active and the first telephone answers the calls. The secretary can override the second telephone's Do Not Disturb at any time by making the call on one of the Multicall line appearances of the second telephone's prime line. If it is important to contact the second telephone, the first telephone can ring the second telephone, despite the Do Not Disturb feature.

See Line Types and Appearances for information on multicall appearances.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- The calling Mitel telephone has to make the call on the line appearance for the feature to operate.
- See Do Not Disturb for the interaction of Do Not Disturb with multicall line appearances.

- Multi-line Mitel telephones can be secretarial sets; the other telephone can be of any type.
- Mitel Family 2 telephones cannot be secretarial sets.

Programming

Program the secretarial set with at least one multicall appearance, secretarial variant, of the other extension, in the secretarial set's subform of CDE Form 09 (Desktop Device Assignments).

Operation

The other extension has Do Not Disturb enabled and there is an incoming call. At the secretarial set:

- The multicall appearance rings.
- Answer the call; the call needs to be transferred to the other extension.
- Press the TRANS/CONF key and dial the other extension's number.
- The other extension rings. Consult privately and/or hang up to transfer the call.

Secure Hot Swap

Description

Secure hot swapping enables a user to exchange programming between any IP Phone, DNIC set or station that is enabled to support the feature and the user's own PIN-protected device.

To initiate a secure hot swap, the user enters a feature access code on the "originating" device, followed by the Directory Number (DN) and PIN associated with the user's "destination" device. Most settings are exchanged, including key programming, tenanting information, COS, COR, call routes, and device-specific programming such a hunt groups and pickup groups. The only settings not exchanged are those associated with the physical device itself such as its MAC address, CESID, and Comments. To unswap a pair of devices, the user re-enters the hot swap feature access code from either device. No PIN is required.

Secure hot swapping allows a number of users to share a single physical telephone. Each user is assigned a virtual or "phantom" device, with personalized programming, which they can hot swap onto the shared telephone when the need arises. This functionality is suited to open environments such as factories and sales offices.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Hot swapping can be initiated on any physical device (IP Phone, DNIC set, or station). The feature cannot be initiated on a phantom device.
- Hot Swapping is not supported by Your Assistant softphones, subattendants, or special devices (voicemail ports, door phones, MOH devices, DMPs).
- Hot swapping is restricted to a pair of devices. While the two devices are swapped, they cannot be swapped again. They must be unswapped first.
- Hot swapping is restricted to devices of the same type. A 5220 IP Phone can hot swap with another 5220 IP Phone, but a 5220 IP Phone cannot hot swap with a 5215 IP Phone.
- All devices that use hot swapping, including phantom devices, must have COS Option 692 (Secure Hot Swapping) enabled.

- A hot swap PIN must be programmed on the extension of every destination device. Originating devices do not require a PIN. The PIN must changed from the default, which is all zeroes (0000).
- For DNIC and ONS implementations, the destination devices must be programmed on line cards (Digital or ONS) that are physically installed in the system.
- Before a device can be hot swapped, it must be idle and logged out of ACD.
- Hot swapping has no effect on the CESID (Customer Emergency Services ID) programmed for the device. For example, if someone makes a call from a device that has been hot swapped, the system sends the CESID for the device itself, not for the device that was hot swapped onto it. Hot Swapping also has no effect on any Comments programmed for the device.
- Do not replace a device that is currently swapped. Unswap it first.

Programming

In CDE Form 03, COS Define, enable COS Option 692 (Secure Hot Swapping) in the device's COS.

In CDE Form 02 (Feature Access Codes), assign an access code to Feature 68 (Secure Hot swap/unswap).

To assign a PIN to the extension of a destination device:

- Display CDE Form 09 (Desktop Device Assignments).
- Locate the desired extension.
- Cycle through the display options by repeatedly pressing the number 9 softkey. When the SHOW PIN softkey appears, select it to display the PIN NUMBER field.
- Enter the four-digit PIN number for the extension in the PIN NUMBER field.

Note: The PIN should be different from the extension number, and must not be all zeroes (0000).

Operation

To swap two devices

- From the originating (physical) device, dial the Secure Hot Swap access code.
- Dial the Directory Number for the destination device.

  Wait for the dial tone.
- Dial the PIN for the destination device.

  Wait for the devices to reset. After initialization is complete, the devices will be swapped and available for use.

To unswap two devices

- From either (physical) device, dial the Secure Hot Swap access code.

To unswap two devices using CDE

- Display CDE Form 09 (Desktop Device Assignments).
- Locate the extension for either device.
- Cycle through the display options by repeatedly pressing the number 9 softkey. When the SHOW PIN softkey appears, select it to display the PIN NUMBER field.

  An asterisk (*) displays in the PIN field to indicate that the device is swapped.
- Repeatedly press the **MORE** softkey. When the UNSWAP softkey appears, select it to unswap the two devices.

  When the form is reloaded, the asterisk is removed from the PIN NUMBER field to indicate that the device is no longer swapped.

Single Button Transfer to Voice mail

Description

The Single Button Transfer to Voice mail feature provides a voice mail key that transfers a caller to a user's voice mail. The voice mail key can be programmed as a feature key, and in some cases can appear as a softkey.

The voice mail key functions under three different modes:

- transfer-recall mode
- direct mode (no recall)
- consultation hold mode

You can use a voice mail key after a transfer and a recall, to route a caller to the destination party's voice mailbox. For example, if "A" receives a call, transfers this call to "B", "B" does not answer the call, and "A" receives a recall, "A" can press the Voice mail key to route the caller to the voice mailbox of B so the caller can leave "B" a message.

You can use a voice mail key to route a caller directly to the destination party's voice mailbox when you know that the destination party is not currently available. For example, if "A" receives a call for "B" and knows that "B" is out of the office, "A" can press the voice mail key, at the dial tone enter the DN for "B", and the caller can leave "B" a message.

You can use a voice mail key to route a caller to the destination party's voice mailbox while the call is on consultation hold. For example, if "A" receives a call for "B", "A" can press the TRANS/CONF key then enter the DN for "B". If "B" decides not to accept the call, "A" can press the voice mail key and transfer the caller to the voice mailbox for "B".

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Requires Feature Level 3 or greater, in System Option 102. If the Feature Level is 2 or less, the voice mail softkey will not appear, and the voice mail feature key (hardkey) along with the LED indicator will not function.
- The SUPERSET 430 and SUPERSET 4150 telephones support the voice mail softkey. If the voice mail softkey fails to appear, you must program a feature key to take its place. The voice mail softkey will not appear for the following reasons:
  - COS Option 269 (Record a Call: Start Recording Automatically) is enabled on the COS for the set, or,
  - COS Option 806 (SMDR - Record Incoming Calls) is enabled on the incoming trunk, or,
  - SMDR is enabled on the trunk group in CDE Form 16.
- This feature is for embedded voice mail and DNIC voice mail. Only embedded voice mail and DNIC voice mail hunt groups are entered in CDE Form 19.
- If there is not a DN programmed in Voice mail Number for This Tenant in CDE Form 19 for the tenant group of the called party, the Voice mail softkey will not appear in the Transfer - Recall mode and the user will receive a reorder tone after dialing a valid DN in the Direct (Non-recall) mode.
- When a call is on a keyline with privacy released, the Voice mail feature key lights when entering talk state.
- COS Option 269 (Record a Call - Start Recording Automatically) must be disabled in an extension's Class of Service in order to get all available voice mail softkeys to display over IP trunks.

- A mailbox must have an extension (real or phantom) associated with it in system programming; otherwise, attempts to transfer calls to it will fail.

Programming

In CDE Form 04, System Option and Timers, enable Feature Level 3 in System Option 102.

In CDE Form 03, COS Define, enable Option 275 (Enable Voicemail Single Key Access) in the class of service of the extension.

In CDE Form 19, Call Rerouting Table, for Voice mail Number for This Tenant, enter the voice mail number for the Day, Night 1, and Night 2 service modes for the tenant group of the called extension number. This voice mail number is typically the master hunt group number for embedded and DNIC voice mail.

In CDE Form 09, Desktop Device Assignments, or with the SUPERKEY on the telephone, program the voice mail feature key. See Feature Keys. Note that users of SUPERSET 430 and SUPERSET 4150 telephones can use the voice mail softkey, provided that COS 806 is disabled. If COS 806 is enabled, then a voice mail feature key must be programmed.

Operation

To transfer a caller to the destination party's voice mailbox after receiving a recall from the unanswered transfer

- Press the Voice mail key.

To transfer a caller directly to the destination party's voice mailbox

- Press the Voice mail key, and at the dial tone dial the DN belonging to the destination party.

To transfer a caller to the destination party's voice mailbox while the caller is on consultation hold

Note: To place a caller on consultation hold, see Transfer.

- Press the Voice mail key.

Softkey Support for Voice Mail

Description

This feature provides softkey support for voice mail features on the following systems:

- SX-200 ICP embedded voice mail.
- Mitel Express Messenger
- NuPoint Messenger™ systems that use DNIC integration
- Mitel 3300 Integrated Communications Platform (ICP) used as a standalone voice mail system.

Instead of dialing single-digit codes to select features (for example, dialing 7 to Play Message), users can press a softkey.

You can program a VM PROMPTS feature key that toggles the voice mail prompts off or on. This feature key allows users, while logged into their voice mailbox, to turn off the voice mail softkeys whenever they want to activate a feature with a call handling softkey (for example, to make a conference call). If the VM PROMPTS feature key indicator is on, the set displays the voice mail softkeys; if the indicator is off, the set displays call handling keys.

Conditions

The following conditions apply to this feature:

- Voice mail softkeys are only supported on DNIC-based, NuPoint Messenger systems (Release 6.1B software or later) and the SX-200 ICP embedded voice mail application.
- Voice mail softkeys are supported if the mailbox has an FCOS that supports all the softkey features (FCOS 10 is recommended).
- Only Family 1 , Family 4 and Family 5 telephones support voice mail softkeys.
- The SX-200 ICP will provide voice mail softkey support over the IP network when connected to a Mitel 3300 ICP or another SX-200 ICP where the centralized voice mail is located.
- NuPoint Messenger systems support voice mail softkey prompts in English, French, and Spanish. The SX-200 ICP embedded voice mail application supports voice mail softkey prompts in English, French, and Spanish. See Language for instructions on how to change the set language.

Programming

Enable Option 97 "Support Softkey Access to Voice mail" in Form 4 (System Options). A Mitel Options Password is required to enable this option.

Enable COS 267 "Softkey Support for Voice mail" in the class of service of the set.

Set System Option 69, DTMF ON time, in Form 4, System Options, to a minimum of 90 milliseconds (value 9).

Program a VM PROMPTS feature key, if desired, on the set (jump to instructions). Users can program a VM PROMPTS feature key on their set by using the Superkey:

- Mitel 5020, 5220 and 5224, and SUPERSET 4025, SUPERSET 4125 and SUPERSET 420 instructions
- SUPERSET 4150 and SUPERSET 430 instructions

If you have centralized voice mail, disable COS Option Record a Call: Start Recording Automatically to ensure that softkeys can be displayed.

Operation

Refer to the NuPoint Messenger User Guides or the SX-200 ICP Voice Mail User Guides for instructions.

Speak@Ease (Mitel Speech Server) Support

Description

The SX-200 ICP supports Mitel Speech Server, a PC based, speech recognition application. The users of the Speech Server use the Speak@Ease softkey on the display telephone to access the application.

Speech Server allows telephone users to place a call to a specific person or department within a company based on spoken commands.

This application interfaces to the PBX via a number of DNIC ports. The DNIC ports are configured as a hunt group that users can access if they want to use the service.

Speech Server may be accessed directly or indirectly.

With direct access, the system automatically dials the hunt group access code programmed in the Call Rerouting Table, with the following actions:

- Going offhook from the idle state
- Pressing the speaker key from the idle state
- Pressing the in-line headset switch from the idle state

- Pressing Trans/Conf to get dial tone.

With indirect access, the rerouting to this application only takes place after the user presses the Speak@Ease softkey.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

- Speak@Ease Support is a purchasable FOR Option. The System Option 85, Speak@Ease Integration must be enabled.
- Speak@Ease and Phonebook features are mutually exclusive. You cannot enable both simultaneously.
- Family 1 and Family 2 telephones support the Speak@Ease softkey (indirect access)
- All Mitel telephones and POTS (Plain Old Telephone Service) telephones support the direct access to Mitel Speech Server, with the exception of SUPERSET 3DN, and SUPERSET 4DN telephones.
- A Speak@Ease softkey appears in the idle state, when the user presses the Speaker key, or when Trans/Conf is used to obtain dial tone. For the softkey to appear, the hunt group access code, the extension number, or the speed dial number associated with the hunt group of the DNIC ports connecting to the Mitel Speech Server must be programmed in CDE Form 19, Call Rerouting Table.
- Pressing the Speak@Ease softkey after pressing trans/conf when the tenant is on a different PBX will transfer you even though you have someone on softhold. If the tenant is on the same PBX, the transfer will not occur.
- The DNIC ports connecting to the Speech Server must be programmed as a SUPERSET 430 subattendant so they receive the calling party name and number in the display message.
- Direct access to the Speech Server is dependant on COS Option 619, Direct Speak@Ease Access being enabled in CDE Form 03.
- In CDE Form 19, the hunt group access code, the extension member, or the speed dial number associated with the hunt group of the DNIC ports connecting to the Speech Server must be programmed.
- The Speak@Ease softkey only appears if the set is configured to select an internal line. The softkey will not appear if the user has to manually select a line after going off hook or if a trunk is automatically selected as a result of going off hook.
- A voice mail port that has COS option 229, Voice Mail Port enabled will not receive a Speak@Ease softkey.
- The softkey text will always be Speak@Ease regardless of the language selected.
- In Form 19, after programming the Day column, the N1 and N2 are automatically programmed if they are left blank. These values can be programmed individually by tabbing over to the desired column.
- COS Option 619, Direct Speak@Ease Access, does not interfere with selecting a line by pressing a line key while on hook.
- When Trans/Conf is pressed, dial tone will always be presented even though COS Option 619 Direct Speak@Ease Access is enabled and the voice dial number is programmed in CDE Form 19.

Programming

In CDE Form 09, Desktop Device Assignment assign the DNIC ports connecting to the Mitel Speech Server as a SUPERSET 430 Subattendant.

In CDE Form 03, enable COS Option 619, Direct Speak@Ease Access if the telephone user requires direct access. For indirect access, the user uses the Speak@Ease softkey on a Mitel 5020, 5220 and

5224 IP Phone, and SUPERSET 420, SUPERSET 430, SUPERSET 4025, SUPERSET 4125, or SUPERSET 4150 telephone.

In CDE Form 04, enable Option 85, Speak@Ease Integration.

In CDE Form 17, program the DNIC ports in a separate hunt group.

In CDE Form 19, Call Rerouting Table, scroll to "Speak@Ease Number for the Tenant" and enter the hunt group access code of the DNIC ports connecting to the Mitel Speech Server.

Operation

For Indirect Access to Speak@Ease, press the Speak@Ease softkey (the SPEZ softkey on the Symbol MiNET Wireless Phone) and say the name of the person you wish to speak to.

For Direct Access to Speak@Ease go offhook, or press the Speaker key, or press the in-line headset switch, or press Trans/Conf and say the name of the person you wish to speak to.

Speaker Volume Control

Description

SUPERSET users may control the volume of the set’s speaker.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

None.

Programming

None.

Operation

To control speaker volume:

Family 3 Telephones:

- While set is in on-hook dialing and listening to ringback, press the VOL ↑ and VOL ↓ keys repeatedly to increase or decrease the speaker volume.

Family 4, Family 5, SUPERSET 410, SUPERSET 3DN, and SUPERSET 4DN Telephones:

- While set is idle, or during a handsfree conversation, press the VOL ↑ and VOL ↓ keys repeatedly to increase or decrease the speaker volume.

SpectraLink Wireless Telephones

Description

The SX-200 ICP supports SpectraLink NetLink e340, NetLink h340, and NetLink i640 Wireless Telephones. Small, lightweight and supporting a complete range of call control features, the SpectraLink telephones communicate using 802.11b and appear to the SX-200 ICP as a wired, multi-line display with the same fixed feature keys as a Mitel 5020/5220 IP, namely: SuperKey, Trans/Conf, Redial, Message, Hold, Cancel, Speaker, and Microphone. (See below for the key mapping from the 5020/5220 phones to the SpectraLink phones.)

The Netlink e340 is designed for use in general office environments. The NetLink h340 is constructed of durable plastics to meet the needs of healthcare workplaces. The Netlink i640, which offers push-to-talk capability, is suited for applications that require a more durable and functional device, such as healthcare, retail, and industrial environments.

NetLink phones support Wired Equivalent Privacy (WEP), an 802.11 security protocol that provides security similar to that of wired Ethernet LAN. The handsets can use either 40-bit or 128-bit encryption.

In addition to the handsets and battery chargers, a complete SpectraLink NetLink wireless solution consists of the following components:

- SpectraLink-compatible access points (APs)
- SpectraLink Voice Priority Server (SVP) — provides voice packet prioritization for maximum voice quality on a shared wireless voice and data network.)

A list of compatible APs along with installation and setup documents for the SVP server and NetLink telephones are available on the Mitel Documentation website at http://edocs.mitel.com.

Additional information regarding SpectraLink Wireless Phones can be found in the SX-200-ICP Engineering Guidelines which is provided in PDF format on the Mitel Edocs website (http://edocs.mitel.com).

Conditions

- Requires SX-200 ICP Release 2.1 or later software and one IP device license per Netlink phone.
- A wireless LAN must be properly configured and operational through the use of 802.11b wireless access points.
- The SX-200 ICP must be connected to the network and fully operational.
- The push-to-talk feature of the NetLink i640 Wireless telephone uses IP multicast addresses. This requires that multicasting be enabled on the subnet used for the NetLink Wireless Telephones and SVP Server. Routers are typically configured with filters to prevent multicast traffic from flowing outside of specific domains. Where possible, the wireless LAN can be placed on a separate VLAN or subnet to reduce the effects of broadcast and multicast traffic from devices in other network segments.

Programming

NetLink telephones

Configure the telephones to associate with the wireless LAN and to download the required software from the TFTP server in the SX-200 ICP. For instructions, see the License Management section in the Setup and Administration guide for NetLink e340/h340/i640 Wireless Telephones. The guide is available on the Mitel Customer Documentation website.

Note: Software for the SpectraLink telephones is provided on the SX-200 ICP system software CD.

SX-200 ICP

- In CDE Form 47, IP Networking, DHCP Options, Common Options subform, program the following options:
  - 3, Default Gateway IP address (default is 192.168.1.1)
  - 66, SpectraLink firmware TFTP server IP address (default is 192.168.1.2)
  - 129, SX-200 ICP (RTC) IP address (default is 192.168.1.2)
  - 130, DHCP Server Identifier (default is MITEL IP PHONE)

- 151, SVP voice prioritization server IP address.

- Register the telephones with the SX-200 ICP either in CDE Form 09 (Desktop Device Assignments) or from the telephones by entering the IP Set PIN Registration code and the desired directory number.
- Assign a Class of Service to each telephone in Form 09.
- (Optional) Associate the SpectraLink Wireless phone with the user's desk phone. See Phone Twinning for programming instructions.
- (Optional) Assign features and line appearances to keys on the telephone. The following table shows how the keys on the 5020/5220 IP phones are mapped to key sequences on the SpectraLink Wireless Telephones:

5050/5220 Key	**Feature**	**SpectraLink Telephone Key Sequence**
Trans/Conf	Transfer or Conference	FCN + 1
Cancel	Cancel	FCN + 2
Message	Voicemail	FCN + 3
Redial	Redial	FCN + 4
Superkey	Superkey	FCN + 5
Feature Key 1	Line Appearance (Prime Line key)	
Programmable Feature Key 2-14. Programmable to either feature keys or line appearances Figure: Mitel 5020 and 5220 IP Telephone Key Numbers		

If a feature is programmed, the feature name appears on the Feature menu of the SpectraLink telephone with a shortcut key in the order of assignment to the feature key.

Note: There may be more features programmed than there are shortcut keys. In this situation, the feature may be activated by using the Up, Down, Select buttons on the side of the SpectraLink telephone.	FCN + 7 8 9 * 0 #
If a Line Appearance is programmed, the line appears on the Line Appearance list, with a line number the corresponds to the order of assignment in CDE.	

Line + 2
3
4
5
6
7
8
9

Note: Line appearances programmed to key positions three

and above fail to display unless the trunk name assigned in Form 14 is "Line n", where "n" is the trunk number.		
Softkey 1	Programmable	Softkey A
Softkey 2	Programmable	Softkey B
Softkey 3	Programmable	Softkey C
Hold	Hold	Softkey D

Operation

Refer to the specific feature in the Program Features section of this document. See also the SpectraLink NetLink User Guides on the Mitel Customer Documentation website at http://edocs.mitel.com/UG/Index.html.

Speed Call Key

Description

This feature allows a Mitel telephone user to save frequently dialed telephone numbers and to access these numbers by pressing a single key. Only unassigned Line Select keys can be used to save Speed Call numbers.

Access codes for features such as Directed Call Pickup, Remote Call Hold Retrieve, and Call Forwarding may be programmed into Speed Call numbers. Forwarding to a Speed Call number can be programmed; see Call Forwarding.

You can also use a speedcall key to send DTMF digits during a call. Typically, you would use this feature to access a voice mail box or long distance application that requires you to enter an account code or password. For example, after accessing your voice mail system, you could press a private speedcall key that is programmed with your password.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- The quantity of speedcall numbers available to a Mitel telephone user is dependent on the number of lines programmed to appear at the set. Lines which are not programmed default to Speed Call keys.
- Speed Call numbers can be made private so that they are not displayed when reviewed or used (similar to private abbreviated dial numbers).
- Only unassigned Line Select keys may be programmed with speedcall numbers.
- Except for in the *3 and ** codes described below, the asterisk (*) may not appear in speedcall numbers.
- *3 and ** are ignored in the digit string if ARS leading digits were dialed manually before the Speed Call was selected.
- The same set of speedcall numbers can be programmed for multiple extensions by using key templates; see Guest Room Key Programming in Form 37 in the Program CDE section.

- COS Option 610 - (Telephone - Guest Room Template) and Speed Call are mutually exclusive on a SUPERSET 3DN or SUPERSET 4DN telephone.
- You can use a speedcall key while dialing a call if the manually dialed digits are ARS leading digits. The Speed Call digits will be appended to the ARS digit string; otherwise, the manually dialed digits will be replaced by the Speed Call digits.
- If ARS leading digits appear in the speedcall string, a *3QQ is interpreted as insert manual digits.
- When Speed Calls are used in ARS, there is a minimum display update of 5 seconds to allow the user to view the number.
- See Line Selection for operation when the prime line is not free for the call and another line has not been selected.
- Private numbers are not displayed in SMDR reports.
- COS Option 807 - (SMDR - Display Private Speed Call) must be enabled to display private numbers in SMDR reports.
- A Speed Call number may contain an abbreviated dial number.
- Personal Speed Call numbers cannot be saved if a Copy Database or Verify Database operation is in progress.
- Personal Speed Call numbers are not available to logged in ACD agents or supervisors. Once logged in, the set reflects the information programmed for that Agent/Supervisor in Form 38.
- To send DTMF digits during a call, the set must be in a stable talk state.
- If COS Option 601, Telephone - Auto Hold Disable, is enabled, pressing a Speed Call key while in talking state, does not drop the existing call.
- Refer to *Abbreviated Dial* for a list of the conditions that apply to pauses in speed calls.

Programming

All programming is done at the telephone.

Operation

Operation varies depending upon the type of set as described below.

Note: The following codes can be inserted into a stored number:

*3QQ - wait for QQ manually dialed digits (QQ can range from 01 to 14)

e.g., the general number for external directory assistance is 9+1+(area code)+555-1212; the area code is to be dialed manually. The number to be stored would be 91*3035551212.

** - dial an asterisk

SUPERSET 3DN Telephones:

To enter or change a Speed Call number:

- With the handset on–hook, press the PROGRAM key.
- Press an unused speedcall key on the telephone.
- Dial the number to be stored or press the REDIAL key to save the last number dialed. If no entry is made, the current entry is cleared.
- Press the PROGRAM key; the Speed Call number is now saved.

Note: A feature key can be programmed in this case - see Feature Keys.

To dial a call using Speed Call:

- Press the programmed speedcall key with the telephone on–hook; if Immediate Line Select is enabled, or,
- Pick up the handset or press the SPEAKER ON/OFF key, then press the Speed Call key.

To set up a Speed Call key as a feature access code:

- With the handset on–hook press the PROGRAM key.
- Press an unassigned Speed Call key.
- Dial the feature access code and the extension number of the applicable set. To correct an entry, press the CANCEL feature key, then press the PROGRAM key, and restart the sequence.
- Press the PROGRAM key.

Family 2 Telephones:

To program or change a Speed Call number:

- With the handset on–hook, press the Speed Call programming button (on a SUPERSET 401+ set, remove the faceplate to expose the button).
- Press the desired speedcall key.
- Dial the number to be stored including the outside access code and area code, if necessary. Up to 40 digits may be saved. To insert a 1-second pause press the red HOLD key.
- Press the Speed Call programming button; the Speed Call number is now saved.

To dial a call using a Speed Call key:

- Pick up the handset, then press the desired speedcall key.

Family 3 Telephones:

To program, change or clear a Speed Call number:

- With the handset on–hook, press SUPERKEY.
- Press # until Personal Keys? appears.
- Press *.
- Press a personal Speed Call key that is not a line key.
- Press *.
- To enter a new number, dial the number. To insert a 1-second pause press the red HOLD key.
  or,
  To enter the last number dialed, press REDIAL.
- Press SUPERKEY; the Speed Call number is now saved.

To dial a call using a Speed Call key:

- Press the programmed Speed Call key with the telephone on–hook (if Immediate Line Select is enabled), or,
- Pick up the handset or press the SPEAKER key, then press the Speed Call key.
- With the handset on–hook, press SUPERKEY.

SUPERSET 410 Telephones:

To program, change or clear a Speed Call number:

- With the handset on–hook, press SUPERKEY.
- Press a personal Speed Call key. (For instructions on how to program a personal key as a Speed Call key, refer to Feature Keys.)
- Dial the number to be stored (to insert a 1-second pause press the red HOLD key)
  OR
  Press the REDIAL key to save the last number dialed.

  If no entry is made, the current entry is cleared.
- Press SUPERKEY; the Speed Call number is now saved.

To dial a call using a Speed Call key:

- Press the programmed speedcall key with the telephone on–hook; if Immediate Line Select is enabled, or,
- Pick up the handset or press the SPEAKER key, then press the Speed Call key.

To set up a Speed Call key as a feature access code:

- With the handset on–hook, press SUPERKEY.
- Press a personal Speed Call key. (For instructions on how to program a personal key as a Speed Call key, refer to *Feature Keys*.)
- Dial 00.
- Dial the feature access code and the extension number of the applicable set.
- Press SUPERKEY.

Family 4 Telephones (except the Symbol MiNET Wireless telephone):

To program, change, or clear a Speed Call number:

- Press SUPERKEY.
- Press the NO softkey until PERSONAL KEYS? appears in the display.
- Press a personal key that isn’t assigned to a line.
- Press the CHANGE softkey. SPEED CALL? appears in the display.
- Press the YES softkey.
- Dial the number to be stored (to insert a 1-second pause press the red HOLD key)
  OR
  Press the REDIAL key to save the last number dialed.

  If no entry is made, the current entry is cleared.
- Press the SAVE softkey.
- Press SUPERKEY; the Speed Call number is now saved.

To dial a call using a Speed Call key:

- Press the programmed Speed Call key with the telephone on–hook; if Immediate Line Select is enabled, or,
- Pick up the handset or press the SPEAKER key, then press the Speed Call key.

Family 5 Telephones:

5330 Phones:

To program, change, or clear a Speed Call number:

1. Press the key that you have programmed as Superkey.
2. Press **No** until **Personal Keys?** appears and then press **Yes**.
3. Press the key you want to program.
4. Press **Change**.
5. At the Speed Call? prompt, press **Yes**.
6. Enter the speed call number using the dial pad.
7. If you want the number to be private, press **Priv**.
8. Press **Save**. Key programming is saved with the default "Speed Call" label.
9. Press Superkey to exit.

To dial a call using a Speed Call key:

- Press the programmed Speed Call key with the telephone on–hook; if Immediate Line Select is enabled, or,
- Pick up the handset or press the SPEAKER key, then press the Speed Call key.

5340 Phones

1. Press the key that you have programmed as Superkey.
2. Press the **More** key until **Feature Key** appears.
3. Press **Feature Key**.
4. Press the key you want to program.
5. Press **Change**.
6. Press **Speed Call**.
7. Enter the speed call number using the dial pad.
8. If you want the number to be private, press **Priv**.
9. Press **Save**. Key programming is saved with the default "Speed Call" label.
10. Press Superkey to exit.

Family 1 Telephones:

To check saved numbers:

- Press the SUPERKEY.
- Find and press the DISPLAY KEYS softkey (use the MORE softkey).
- Press the required line appearance key. The Speed Call number currently saved is displayed on the LCD.

To program a Speed Call number:

- Press the SUPERKEY.
- Press the MORE softkey until the FEATURE KEY softkey appears.
- Press the FEATURE KEY softkey.
- Press an available personal key.
- Press the CHANGE softkey.
- Press the SPEED CALL softkey.
- Dial the number to be stored (to insert a 1-second pause press the Insert Pause softkey)
  OR
  To save the redial number, press the REDIAL softkey.

  If no entry is made, the current entry is cleared.
- If desired, press the MAKE PRIVATE softkey to prevent the DISPLAY KEYS function from revealing the number.
- Check the entry on the LCD. If correct, press the SAVE softkey. To correct an error before saving the entry, use the ← softkey to backspace and clear the incorrect entry, or press the BACKUP softkey to cancel the entire entry. Then re-enter the correct digits.

To dial a call using a Speed Call key:

- Press the programmed Speed Call key with the telephone on–hook; if Immediate Line Select is enabled.
  OR
  Pick up the handset or press the SPEAKER key, then press the Speed Call key.

Split

Description

Split allows a Mitel telephone user, engaged in a conference call, to split the call between the conferees. Once active, swapping can take place between the calls, or the conference can be reestablished.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- A 3-party conference must be active.
- Any one of the three parties may split the call.
- The party splitting the call must press the CONF feature key to re–establish 3-way conversation.
- There cannot be a console in the call.
- There can be no parties in the call holding the call.
- The call cannot be on consultation hold.
- There cannot be an extension with the Non-Busy Extension feature enabled.
- There cannot be a Mitel telephone in the call using the Add Held feature at the time.
- There can be no industry-standard telephone in the call with the Broker's Call feature enabled.

Programming

None.

Operation

Family 2 and Family 4 Telephones:

- A 3-party conference is established.
- Press the SPLIT softkey. The LCD indicates which party is connected. The other party is on hold.
- To converse with the other party press the TRADE CALLS softkey.
- To re-establish the conference, press the CONFERENCE softkey.

Symbol MiNET Wireless Phone:

- Select Split and then press the SEND key.

Standard Hard Hold

Description

Standard Hard Hold allows a headset (or handset) user to place a call on hard hold while the Automatic Line Selection feature is enabled. (COS Option 604, Telephone - Automatic Outgoing Line.) If COS Option 282 (Standard Hard Hold) is disabled (the default), the system selects a new line each time headset calls are placed on hard hold.

Conditions

- When the active call is a headset or handsfree call, pressing the red HOLD button puts the set into an idle state.

- When the active call is a handset call, pressing the red HOLD button puts the set in "waiting for line" state with SELECT LINE displayed.

Programming

- Enable COS Option 282, (Standard Hard Hold).
- Disable COS Option 690 (Hold and Page).
- In CDE Form 04, select feature level 6 in System Option 102.

Station Message Detail Recording (SMDR)

Description

Station Message Detail Recording (SMDR) allows data to be collected for each outgoing and incoming trunk call. This data can be output to a printer or a data recording device for subsequent processing.

Refer to the Station Message Detail Recording section.

Conditions

When pauses are entered into a speed dial, the digits that follow are not recorded by SMDR (these are DTMF digits for passwords, access codes, account codes, etc. that must be kept private).

Programming

The following system options apply:

- 06 - Analog Networking SMDR
- 08 - Five Digit SMDR
- 28 - SMDR - Indicate Long Calls
- 39 - Data SMDR - Indicate Long Calls
- 44 - ACD Reports
- 49 - Pseudo Answer Supervision Timer.

The following COS options apply:

- 246 - SMDR - Extended Record
- 247 - SMDR - Record Meter Pulses
- 700 - SMDR - Does Not Apply
- 702 - SMDR - Overwrite Buffer
- 803 - SMDR - Drop Calls < n Digits
- 804 - SMDR - Drop Incomplete Outgoing Calls
- 806 - SMDR - Record Incoming Calls
- 807 - SMDR - Display Private Speedcall
- 814 - SMDR - Record ANI/DNIS/CLASS
- 906 - Data SMDR - Does Not Apply
- 907 - Data SMDR - Extended Record

- 908 - Data SMDR - Overwrite Buffer

Ensure outgoing SMDR is enabled for the trunk groups in CDE Form 16 (Trunk Groups).

Refer to the Station Message Detail Recording section for details.

Operation

None.

Subattendant - Basic Function

Description

This feature provides a Mitel telephone with enhanced recall and call queuing capabilities, allowing the set to be used as a subattendant position.

Any calls that are handled by the subattendant will recall the subattendant instead of the attendant. Recalls to the subattendant ring the set's prime line.

A Mitel telephone is normally considered to be busy when the set and/or the prime line appearances are busy.

The Night/Day Switching feature can be used to allow the user to select DAY, NIGHT1, or NIGHT2 service for the system.

A subattendant position telephone can also be used as the alternate trunk recall point - see Trunk Recall.

Note: This feature is not related to, and does not interact with Subattendant - Enhanced Function features.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- The set to be used as an enhanced answering position should be programmed in its own COS.
- Line appearance checking is done for re-routes, forwarding, and recalls.
- There can be as many calls ringing at the enhanced answering position set as there are multicall appearances of its prime line on the telephone.
- A message set up on a telephone by the subattendant is automatically canceled when that telephone calls, and is answered by the subattendant that set up the message.
- An enhanced subattendant (COS Option 606) appears as an external device when calling a destination that has Call Forwarding Internal/External Split enabled.

Programming

Enable COS Option 606 (Telephone - Enhanced Answering Position) in the class of service form of the subattendant telephone.

Enable COS Option 262 (Ignore Forward Busy with Free Appearance) in the class of service form of the subattendant telephone to have Call Forward Busy ignored when there are any free multi-line appearances of the set. Note that this COS option applies only to multi-line appearances, not key line appearances or logical line appearances.

Program the unused line appearance keys of the ENHANCED answering position telephone as a multicall appearance of its prime line. Incoming calls that are waiting to be answered queue up on these keys. Note that this paragraph applies to an enhanced telephone with subattendant functionality and does not apply to a telephone programmed as a Subattendant in CDE.

Enable COS Option 262 (Ignore Forward Busy with Free Appearance) in the set's COS to have call forward busy ignored when there are any free multi-line appearances of the set.

Operation

None.

Subattendant - Enhanced Functions

Description

The enhanced subattendant functions allow SUPERSET 4150, SUPERSET 430, and SUPERSET 4DN telephones, and Mitel 5020, 5220, 5220 Dual Mode, 5224, 5324, and 5340 IP telephones to be used as a subattendant position for multi-tenanting, consoleless operations (but can be used in conjunction with a console) where call clearing of twenty five calls per hour is considered to be busy. The functions of the telephone remain unchanged; the Enhanced Subattendant functions are an addition.

SUPERSET telephones and the Mitel 5340 IP Phone support the full range of enhanced subattendant functions, while Mitel 5020, 5220,, 5224 and 5324 IP Phones support a limited range. The supported functions are listed below:

Table: Enhanced Subattendant Functions

Function	SUPERSET 4150/430/4DN	Mitel 5020/5220/5224/ 5324	Mitel 5340
Abbreviated Dial Programming	X	X	X
Advisory Message Setup	X		X
Call Blocking	X	X	X
Call Forward Setup and Cancel	X		X
Call Logging	X		X
Call Waiting Indication	X	X	X
Centralized Answering Position (RLT)	X	X	X
Date and Time Setup	X	X	X
Hold Positions	X		X
Listed Directory Number (LDN) Keys	X	X	X
New Call Ring	X	X	X
Paged Hold Access	X		X
Recall	X	X	X
Reminder	X	X	X

Table: Enhanced Subattendant Functions			
Function	**SUPERSET 4150/430/4DN**	**Mitel 5020/5220/5224/ 5324**	**Mitel 5340**
Station Do Not Disturb Setup and Cancel	**X**	**X**	**X**
Guest Room Enhanced			**X**

Paged Hold Access Discriminating ringing is available for calls from the subattendant position (See Ringing - Discriminating).

Note: The Subattendant - Enhanced Functions feature is not related to, and does not interact with, the Subattendant - Basic Function feature.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

The subattendant can program an extension call forwarding or do not disturb function without enabling the extension call forwarding COS options.

Programming

Enable COS Option 125 (Attendant Multi-New Call Tone) to provide an audible alert for calls waiting when the attendant is engaged in a call.

Set the time interval for audible alerts in COS Option 404 (Recording Failure to Hangup Timer). The default for COS Option 404 is 30 seconds.

In CDE Form 09 (Desktop Device Assignments):

- Press the DEVICE TYPE softkey to set or change the device type. Use the appropriate subattendant (S/ATT) softkey to set/change it to subattendant. The subattendant device type must be set before any other fields. If the device is already programmed, it must be deleted and re-entered.
- Press the EXPAND SET softkey to get to the Expand Set Subform - (subattendant keys are defined here) - refer to the specific subattendant feature for further information.

Operation

Refer to the specific subattendant feature.

Subattendant - Abbreviated Dial Programming

Description

The Subattendant - Abbreviated Dial Programming feature allows the subattendant to program system abbreviated dial numbers from the subattendant telephone. The subattendant has the option of making abbreviated dial numbers confidential.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply:

- This feature is available only to SUPERSET telephones and Mitel IP Phones that have been programmed with enhanced subattendant functionality (see Subattendant - Enhanced Functions).
- Only one device can program system abbreviated dial numbers at a time.

Programming

The following steps are required:

- In Form 3, enable COS Option 111 (Attendant Abbreviated Dial Programming) in the subattendant COS.
- For the display of confidential abbreviated dial numbers, enable COS Option 110 (Attendant Abbreviated Dial Confidential Number Display).

Operation

4150, 430 and 4DN telephones

To program an abbreviated dial number:

- Press the SUPERKEY.
- Press the MORE softkey.
- Press the FEATURE KEY softkey.
- Press the SYSTEM softkey.
- Enter an Abbreviated Dial index number.
- Press the ENTER softkey.
- Enter the new number (see Abbreviated Dial for details on special character entry).
- Press the MAKE PRIVATE softkey (if desired).
- Press the SAVE softkey.
- Press the SUPERKEY.

5020, 5220, 5224, 5324, 5330, and 5340 IP telephones

To program an abbreviated dial number:

- Press SUPERKEY.
- Press NO repeatedly until ABBREVIATED DIAL? appears on the display.
- Press the YES softkey.
- Enter an Abbreviated Dial index number. Press the ← softkey to erase incorrect digits.
- Press the ENTER softkey.
- Program the abbreviated dial (speed call) number:
  - Press the CHANGE softkey to change an existing number or enter a new number.
  - Press the DEL softkey to delete an existing number.
  - Press the PRIV softkey to make a number confidential.

  See Abbreviated Dial for details on special character entry.
- Press the SAVE softkey.
- Press the SUPERKEY.

Subattendant - Advisory Message Setup

Description

There are eight default advisory messages and seven programmable advisory messages that the subattendant may setup on behalf of another extension. The subattendant can read another extension's currently displayed advisory message, or read through the available advisory messages and choose one for display on the set, or program one for display.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply:

- This feature is available only to to SUPERSET telephones and Mitel IP Phones that have been programmed with enhanced subattendant functionality (see Subattendant - Enhanced Functions).

Programming

Enable COS Option 150 - Sub attendant Station Setup Advisory Message in the subattendant COS.

Operation

To remotely set up an advisory message on a SUPERSET telephone; from the subattendant position:

- Press SUPERKEY.
- Press the STATIONS softkey.
- Enter the extension number of the applicable SUPERSET telephone.
- Press the ADVISORY MSG softkey.
- Find the desired message - use the NEXT MSG (PREVIOUS MSG) and/or SHOW MSG NO. softkeys to move through the list of existing messages. Stop at the desired message.
- Press the TURN MSG ON softkey, or
- Press the CREATE MSG softkey to create a custom message.
- Press the TURN MSG ON softkey.

To cancel an advisory message on a SUPERSET telephone:

- Press SUPERKEY.
- Press the STATIONS softkey.
- Enter the extension number of the applicable SUPERSET telephone.
- Press the ADVISORY MSG softkey.
  - The currently displayed message will be displayed
- Press the TURN MSG OFF softkey.

Mitel 5340 IP Phone:

To set up an Advisory Message:

1. Press the key you have programmed as Superkey.
2. Press the **Stations** softkey.
3. Dial extension number.
4. Press the **Advisory Msg** softkey
5. Find the desired message using the Next Msg, Previous Msg and/or Show Msg No. softkeys.

6. Press the T**urn Msg On** softkey.

To cancel an Advisory Message:

1. Press the key you have programmed as Superkey.
2. Press the **Stations** softkey.
3. Dial the extension number.
4. Press the **Advisory Msg** softkey. The currently displayed message appears on your display.
5. Press the **Turn Msg Off** softkey

Subattendant - Call Blocking

See Call Blocking.

Subattendant - Call Forward Setup and Cancel

Description

The Subattendant - Call Forward Setup and Cancel feature allows the subattendant to setup, review, and disable call forwarding for any extension. The extension for which the subattendant sets up forwarding need not have any of the call forwarding features in its COS. The subattendant may also set up call forwarding from the extension to the subattendant.

All forwarding types can be setup or canceled in this function, whether or not forwarding types have been previously defined for either the subattendant or the affected extension.

Standard call forwarding routes all calls to the same destination. Split Call Forwarding is also available. Split Call Forwarding allows the subattendant to route internal and external calls to different destinations (see Call Forwarding - Internal/External Split).

Each call forwarding mode (Always, Busy and No Answer) can have a separate destination for internal calls and external calls if the FOR System Option 102, Feature Level is set to level 1 or greater.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply:

- For general conditions see Call Forwarding.
- This feature is available only to to SUPERSET telephones and Mitel IP Phones that have been programmed with enhanced subattendant functionality (see Subattendant - Enhanced Functions).
- The subattendant will not see a TO ME softkey or a DELETE softkey when programming Call Forwarding on behalf of another extension.

Programming

The following programming is required:

- Enable COS Option 123 (Attendant Call Forward Setup and Cancel) in the subattendant COS.
- If Split Call Forwarding is desired, enable COS Option 260 (Internal/External Split Call Forward) for the extension.

- Program FOR System Option 102, Feature Level, to level 1 or greater, for internal and external destinations for Call Forwarding - Always, Busy and No Answer.

Operation

To set up call forwarding for an extension with split call forwarding from the subattendant telephone :

- Press **Superkey**.
- Press **Stations**.
- Enter the extension number of the extension being setup.
- Press **Forwarding**. The display shows the status of the forwarding types: ALWAYS, BUSY and NO ANSWER. Each of these forwarding types has a separate status for internal calls and external calls.
- Press **Always**, **Busy**, or **No Answer**. ON or OFF appears on the right hand corner of the display depending on the status of the forwarding type.
- Press **Internal** or **External** depending on the type of calls you want to program. The display requests the destination number.
- Enter the destination number. If an invalid number is entered, the display shows INVALID NUMBER. Use the ← softkey to back space and erase the entry. Enter the correct number or press Current No. to restore the previously programmed destination, if available (only available when the current forward destination is anything but a speedcall).
- Press **Save/On**. The Save/On key saves and enables the programmed forwarding. The Save/Off softkey saves and disables the programmed forwarding.The display returns to the screen that appears after pressing the Forwarding softkey.

To set up call forwarding for an extension from the subattendant telephone without split call forwarding:

- Press **Superkey**.
- Press **Stations**.
- Enter the extension number of the extension being setup.
- Press **Forwarding**. The display shows the status of the forwarding types: ALWAYS, BUSY and NO ANSWER and prompts you to select a type.
- Press **Always**, **Busy**, or **No Answer**. The display requests the destination number.
- Enter the destination number. If an invalid number is entered, the display shows INVALID NUMBER. Use the ← softkey to back space and erase the entry. Enter the correct number (can be a speedcall key, if programmed) or press the Current No. softkey to restore the previously programmed destination, if available (only available when the current forward destination is anything but a speedcall).
- Press **Save/On** or **Save/Off**. The Save/On key saves and enables the programmed forwarding. The Save/Off softkey saves and disables the programmed forwarding.The display returns to the screen that appears after pressing the Forwarding softkey.

To disable call forwarding for an extension from the subattendant telephone:

- Press **Superkey**.
- Press **Stations**.
- Enter the extension number of the extension having forwarding canceled.
- Press **Forwarding**.
- Press **Always**, **Busy**, or **No Answer**. If split is enabled on the extension, press **Internal** or **External**.
- Press **Current No**.
- Press **Save/Off** to disable each mode independently.
- Press **Superkey** to exit.

Subattendant - Call Logging

See Call Logging.

Subattendant - Calls Waiting Indication

Description

The Calls Waiting indicator appears in all call processing states and is displayed in the subattendant Calls Waiting area in the upper right corner of the display, directly below the area where the Forwarding flag appears. The information displayed varies by phone type. Mitel 4150, 430 and 4Dn telephones display C/W followed by the number of calls in the queue, up to a maximum of 99 calls. Mitel 5020, 5220, 5224 and 5324 IP telephones display the number of calls in the queue, up to a maximum of 9 calls. If more than 9 calls are waiting, a flashing + sign is displayed.

The Call Waiting flag takes precedence over the Message Waiting and Mic On flags when clashes occur.

The subattendant may have calls from outside trunks and extensions queued that are waiting to be answered. The number shown by the Calls Waiting indicator, is the total number of calls in the queue. This includes only calls ringing LDN's (or the RECALL key) that appear on the subattendant telephone and any calls ringing the night bell.

Each new call ringing the subattendant position increments the indicator. The indicator decrements each time a caller hangs up.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- This feature is available only to SUPERSET telephones and Mitel IP Phones that have been programmed with enhanced subattendant functionality (see Subattendant - Enhanced Functions).
- If there are no calls in the Calls Waiting queue, there is nothing in the CW area of the LCD display and either the Msg Waiting or Mic On flag may occur.
- The maximum number of calls waiting that can be displayed at the subattendant is 99 (the actual queue maximum is 200).
- Only night bell callers allowed to connect to the subattendant as specified in Form 5, Tenant Interconnection, will be counted on the Subattendant C/W.

Programming

None.

Operation

None.

Note: The ring type is displayed by 4150, 430 and 4DN telephones when the RECALL key is selected. Mitel IP Phones do not display the ring type.

Subattendant - Centralized Answering Position on RLT

See Centralized Attendant on Release Link Trunk

Subattendant - Date and Time Setup

See Date and Time Setup.

Subattendant - Guest Room Enhanced

Description

The Guest Room feature key provides the 5340 IP telephone with the same Hotel/Motel functionality as the Superset 4150 DNIC telephone, including displaying/changing room status and setting multiple wake-up calls.

Conditions

- This feature key applies only to 5340 subattendant sets.
- Programming is supported on keys 02 to 15 only.

Programming

To program the Guest Room feature key from the 5340 telephone:

1. Press the **Settings** hard key.
2. Press **Programmable Keys**.
3. Press the key you want to program. NOTE: Even though the phone may have more than 16 keys, programming is supported on keys 02 to 15 only.
4. Use the Page Navigation keys to move through the features list and select **Guest Room**.
5. Click **Save**.

To program the Guest Room feature key from the CDE:

- In CDE Form 3, enable COS Option 105 (Guest Room Key) for the 5340 subattendant.
- In CDE Form 9, program Guest Room using the FEATURE key after selecting EXPAND SET or EXPAND PKM.
- In CDE Form 37, program Guest Room using FEATURE key.
- In CDE Form 38, program Guest Room using NON-ACD KEYS.

For more information about using the 5340 IP telephone as an enhanced subattendant, refer to the *Subattendant User Guide for the 5340 IP Phone* available at www.mitel.com.

Operation

The Guest Room feature key LED is illuminated when the the 5340 IP phone is in a Guest Room session.

Subattendant - Hold Positions

Description

This feature provides the subattendant with up to six hold position keys. When enabled, the hold position keys permit the subattendant to answer other Listed Directory Numbers (LDN) or the Prime line without having to release current calls on the subattendants telephone first. It is recommended that at least one hold position be programmed, otherwise the subattendant will have to release incoming calls on the prime line of the subattendant telephone before being able to answer other incoming LDN or Prime line calls.

The subattendant can transfer a current incoming call to one of the hold positions by selecting the corresponding hold position key. The call is then placed in the hold position, releasing the prime line for the subattendant to receive another incoming call.

The indicator for the hold position to which the call has been transferred flashes, indicating to the subattendant that a call is currently occupying that hold position. The ADD HELD softkey appears when a call is transferred to a hold position.

Should all programmed hold positions at a subattendant position become occupied, incoming calls on the subattendant prime line may be placed on hold by selecting the red HOLD key. The red HOLD key places the incoming call on a hard hold on the subattendant prime line, but since the line is occupied, subsequent LDN calls and Recalls cannot be answered until the call on hard hold is released. The subattendant may select a held party (via the HOLD key) while dialing on the Prime Line.

Recall

COS Option 116 - Attendant-Timed Recall - Hold timer applies to calls placed in subattendant hold positions. If this timer times-out for a call being held in one of the hold positions, the indicator associated with the hold position flashes and a new call ring is given, indicating to the subattendant that the hold position is recalling (at this point the Call Hold Notify timer is started).

COS Option 681 - Key Set/Sub Att - Call Hold Notify Timer also applies to calls placed in subattendant hold positions. This timer starts after the expiration of the Timer Recall Hold timer. Every time this timer expires, a new call ring is given, and the timer is restarted. This will continue until the call is retrieved from hold. If this is not programmed there will be no notifications and the caller will remain on hold until retrieved. This should no be confused with COS Option 254 - Call Hold Recall Timer, which is used in conjunction with hard held lines on the subattendant set, rather than the hold positions.

Add Held Prompt

An ADD HELD prompt softkey is provided to permit the subattendant to pickup calls currently in hold positions and add them to an existing conversation on one of the set’s active lines to create a 3-party conference.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- This feature is available only to SUPERSET telephones and Mitel IP Phones that have been programmed with enhanced subattendant functionality (see Subattendant - Enhanced Functions).
- Each subattendant position can be programmed to provide up to six hold positions.
- Hold keys must be programmed on the set, not on a PKM, PKM 48, or PKM 12.
- The Mitel 5340 IP phone has its programmable keys arranged on three pages that are accesssed using page navigation keys. If you program LDN, RECALL, or HOLD_POSITION keys to a key on page 2 or 3, and then return to page 1 (the default), you may not be able to see any visual notifications. We recommend that these features be programmed on keys 2 - 8 on page 1 of your 5340 IP Phone.

Programming

To program hold positions:

- Enter the Expand Set Subform of CDE Form 09 (Desktop Device Assignments), and program one or more keys as hold positions via the HOLD POS softkey.
  - Note that the given telephone must be a subattendant device type to be offered the HOLD POS softkey.

Related programming:

- Set an appropriate value for COS Option 116 - Attendant-Timed Recall (Hold) timer in the subattendant COS.
- Set an appropriate value for COS Option 681 - Key Set/Sub Att - Call Hold Notify Timer in the subattendant COS.

Operation

To place a call on hold using a subattendant hold position key:

- Press a free hold position key.

  The associated line appearance indicator flashes, and the ADD HELD softkey is enabled.

To retrieve call from subattendant hold position key:

- Press the desired hold position key.

  The associated line appearance indicator changes from a flashing state to an inactive (off) state, while the prime line appearance indicator indicates an active (on) line condition. The held call is transferred to prime line.

To add a held call to current conversation:

- In Off-hook state select the ADD HELD softkey, or while in handsfree mode, press the ADD HELD softkey after choosing a line.
- Press the desired hold position key.

  The held call is transferred to current conversation and associated line appearance indicator changes from a flashing state to an inactive (off) state.

Subattendant - LDN Keys

Description

Each subattendant can have up to six keys programmed as Listed Directory Number (LDN) keys. The LDN keys appear on the subattendant telephone line keys. The LDNs may be programmed to appear on other subattendant telephones or attendant consoles to permit greater call handling flexibility. When this occurs, the COS and tenant of the subattendant LDN is taken from the subattendant or console with the lowest bay, slot and circuit on which the LDN is programmed.

The LDN keys and the RECALL key (see Subattendant - Recall) act as call queueing indicators. Unlike line keys; they cannot be selected to dial on and conversations cannot be held on them. When a subattendant LDN call is answered, the call is automatically connected to the prime line of the subattendant telephone.

Each LDN position can be programmed as the answer point for a trunk or reroute destination for a particular type of call. To ensure that the prime line is free to answer any LDN calls, the subattendant prime line cannot be programmed to appear on other devices.

The subattendant can answer an LDN call three ways:

  - by going offhook; the longest waiting LDN call is then automatically connected to the subattendant’s prime line, or

- by pressing the SPEAKER key; the longest waiting LDN call is then automatically connected to the subattendant prime line, or
- by selecting the LDN key directly.

LDN keys with “ring type” set to NO RING are not automatically selected when the subattendant goes offhook (or selects the SPEAKER key), and therefore must be answered manually by selecting the No Ring LDN key.

Once answered by the subattendant, an LDN call is treated as though it were a regular call received on a SUPERSET telephone or Mitel IP Phone, with the exception of serial calls.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- This feature is available only to SUPERSET telephones and Mitel IP Phones that have been programmed with enhanced subattendant functionality (see Subattendant - Enhanced Functions).
- Subattendant LDNs and recall calls will override the subattendant DND settings and will ring the DND subattendant telephone on the LDN or RECALL key.
- Station and SUPERSET telephones and consoles cannot directly dial an LDN.
- There is a maximum of six LDN keys per subattendant position.
- There is a maximum of 32 appearances per LDN.
- LDN keys must be programmed on the set, not on a PKM, PKM 48, or PKM 12.
- Automatic Number Identification (ANI) displays once a ringing LDN is answered.
- For multi-appearance LDN keys, the console (or subattendant position) with the lowest physical location number (bay #/ slot #/ circuit #) is always the owner of an LDN.

Programming

To program LDN keys, enter the Expand Set Subform of CDE Form 09 (Desktop Device Assignments), and:

- Program one or more keys as LDN Keys via the LDN softkey.
- Select a ring type for each LDN.
- Enter an access code in the EXT NUM field for each LDN - this may be a unique number, or an existing LDN number.
- Enter a label for each LDN. This text will display on the subattendant set. If no label is entered, the set displays the extension number.

Note: If a subattendant desires to always automatically clear down the current call when another line, LDN, recall or hold position is selected, enable COS Option 601 (Telephone - Auto Hold Disable) in the subattendant COS.

Note: LDN keys can only be programmed on Page 1 keys of the 5340 IP Phone. You cannot change the label from the phone -- it must be changed via CDE Form 09 subform.

Note: 5340 phones may display a truncated label if Label information in Form 09 is too long for the phone display.

Operation

The subattendant can selectively answer any incoming call type by pressing the appropriate LDN softkey, or calls may be answered on a first-come first-served basis by using the SPEAKER key (handsfree operation) or by going offhook.

Note: The ring type is displayed by 4150, 430 and 4DN telephones when an LDN key is selected. Mitel IP Phones do not display the ring type.

New Call Ring

See New Call Ring.

Subattendant - Paged Hold Access

Description

The subattendant can place an incoming call on hold, page for the called party, then inform the called party of the digits to dial, the called party can then pick up the incoming call directly from the subattendant hold position. When the subattendant accesses a PA Pager with a call on hold position, the Hold Pickup access code is displayed along with the subattendant identifier code. The subattendant would then instruct the paged party to call those digits followed by the hold position number to pick up their call. Also see Paging - PA in this document.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- This feature is available only to SUPERSET telephones and Mitel IP Phones that have been programmed with enhanced subattendant functionality (see Subattendant - Enhanced Functions).
- Consoles, subattendants, industry-standard telephones, SUPERSET telephones, DISA trunks and TIE trunks can pickup the held calls.
- The party picking up the call must be able to connect to the held party; see Device Interconnection Control.
- An extension cannot pick up the held party if the extension has a consultation hold in progress and the held party has COS Option 233 (Never A Consultee) enabled.
- A station or SUPERSET telephone (with a consultation hold in progress) cannot pick up the held trunk if the station or SUPERSET telephone has COS Option 214, (Cannot dial a trunk after flashing) enabled.
- A station or SUPERSET telephone in a conference with a trunk on consultation hold cannot pick up the held trunk if the station or SUPERSET telephone has COS Option 215, (Cannot Dial a Trunk if Holding or in Conf With a Trunk) enabled.

Programming

The following programming is required:

- Assign an access code to Feature 16 (Hold Pickup Access - Attendant Hold Slots).
- Enable COS Option 225 (Hold Pickup - Attendant Page Access) in the COS of the device from which the pickup call is made.

Operation

If paging the default paging zone:

- Place the incoming calling party on hold by using one of the subattendant hold positions.

- Press the PAGE softkey - when the subattendant accesses the PA Paging function, the subattendant telephone display indicates:
  - the feature access code, assigned to Feature 16,
  - followed by two digits (13 to 37) that identify the subattendant,
  - followed by the hold position number.
- Page the desired party, specifying the displayed number (the last number being the hold position number); e.g., 677132 (where 677 is the feature access code, 13 is the subattendant identifier, and 2 is the hold position number).

When the paged party dials the subattendant Hold Pickup Access code number, the paged and held parties are connected.

Note: If the paged party does not call, the held party recalls the subattendant automatically after the hold recall timeout; see Subattendant - Hold Positions in this document.

If paging a zone other than the default zone:

- Place the incoming calling party on hold by using one of the subattendant hold positions.
- Dial the ’Paging Access To Specific Zones’ access code, followed by the desired zone.
- Page the desired party, specifying the Subattendant Hold Pickup access code and the subattendant Identifier, followed by the hold position number.

When the paged party dials the Subattendant Hold Pickup access code number, the paged and held parties are connected.

Note: If the paged party does not call, the held party recalls the subattendant automatically after the hold recall timeout; see Subattendant - Hold Positions in this document.

Subattendant - Recall

Description

The recall feature ensures that calls do not remain unanswered or on hold for an unlimited period of time. Any call that has been extended by a subattendant, recalls the subattendant position if the call is not answered or remains on hold at the end of a timeout period.

The LDN keys (see Subattendant - LDN Keys) and the RECALL key act as call queueing indicators. Unlike line keys; they cannot be selected to dial on and conversations cannot be held on them. When a subattendant recall is answered, the call is automatically connected to the prime line of the subattendant telephone.

To ensure that the prime line is free to answer any recall calls, the subattendant prime line cannot be programmed to appear on other devices. To avoid recall calls tieing up the prime line of the subattendant it is important to program the RECALL key. Recall calls to the subattendant will then be queued on the RECALL key.

The subattendant can answer a recall call three ways:

- by going offhook; the longest waiting recall call is then automatically connected to the subattendant prime line, or
- by selecting the SPEAKER key; the longest waiting recall call is then automatically connected to the subattendant prime line, or
- by selecting the RECALL key directly.

For further information on the recall feature, refer to Recall in this document.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- This feature is available only to SUPERSET telephones and Mitel IP Phones that have been programmed with enhanced subattendant functionality (see Subattendant - Enhanced Functions).
- Subattendant recall calls will override the Subattendant DND settings and will ring the DND subattendant telephone on the RECALL key.
- Station and SUPERSET telephones and consoles cannot directly dial a RECALL key.
- There is a maximum of one RECALL key per subattendant position.
- Also see Conditions under Recall.
- The Mitel 5330 IP phone has its programmable keys arranged on three pages that are accesssed using page navigation keys. If you program LDN, RECALL, or HOLD_POSITION keys to a key on page 2 or 3, and then return to page 1 (the default), you may not be able to see any visual notifications. We recommend that these features be programmed on keys 2 - 8 on page 1 of your 5330 IP Phone.

Programming

To program a Recall key, enter the Expand Set Subform of Form 09 (Desktop Device Assignments) and:

- Program a RECALL key via the RECALL softkey.

Also see Programming, under Recall in this document.

Operation

When a call recalls to the subattendant position, press the RECALL key, go off-hook, or press the SPEAKER ON/OFF key (handsfree mode).

Note: The ring type is displayed by 4150, 430 and 4DN telephones when the RECALL key is selected. Mitel IP Phones do not display the ring type.

Subattendant - Reminder

See Reminder.

Subattendant - Station DND Setup

Description

The subattendant can set up or cancel Do Not Disturb (DND) for an extension by selecting the DO NOT DISTURB softkey under the subattendant stations function.

Selection of the DO NOT DISTURB softkey causes the extension information on the subattendant display to be updated to indicate that the extension status has been changed to Do Not Disturb if the station was not already Do Not Disturb or if the device was already in the Do Not Disturb state when the softkey was selected, the Do Not Disturb indicator is turned off. The corresponding set will also have its DND indicator updated. See Do Not Disturb in this document.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

This feature is available only to SUPERSET telephones and Mitel IP Phones telephones that have been programmed with enhanced subattendant functionality (see Subattendant - Enhanced Functions).

Programming

Enable System Option 11 - Automatic Wakeup.

Enable COS Option 121 - Attendant Station Do Not Disturb in the subattendant COS.

Operation

4150, 430, 4DN, and Mitel 5340 IP telephones

To remotely set up Do Not Disturb on an extension from the subattendant position set:

- Press SUPERKEY.
- Press the STATIONS softkey.
- Enter the extension number.
- Press the DO NOT DISTURB softkey.
  - In the top right hand portion of the display, the extension's current state indication changes to DND.
- Press SUPERKEY to exit.

To cancel Do Not Disturb on an extension:

- Press SUPERKEY.
- Press the STATIONS softkey.
- Enter the extension number.
- Press the DO NOT DISTURB softkey.
- Press SUPERKEY to exit.
  - In the top right hand portion of the display, the DND indication changes to the current state of the extension.

5020, 5220, 5224 and 5324 IP telephones

To remotely set up Do Not Disturb on an extension from the subattendant position set:

- Press SUPERKEY.
- Press the STATIONS? softkey.
- Press the YES softkey.
- Enter the extension number. Press the ← softkey to erase incorrect digits.
- Press the ENTER softkey.
- Press the TurnOn softkey to enable DO NOT DISTURB.
- Press SUPERKEY to exit.

To cancel Do Not Disturb on an extension:

- Press SUPERKEY.
- Press the STATIONS? softkey.
- Press the YES softkey.
- Enter the extension number. Press the ← softkey to erase incorrect digits.
- Press the ENTER softkey.
- Press the TurnOff softkey to disable DO NOT DISTURB.

- Press SUPERKEY to exit.

SUPERSET 3DN and SUPERSET 4DN Auto-Answer For Directed Page Calls

Description

This feature enables users of SUPERSET 3DN and SUPERSET 4DN telephones to respond handsfree to a directed page. If a directed page is broadcast over the user's set, the set microphone is activated automatically allowing the user to speak handsfree to the calling party.

The Auto-Answer for Directed Page Calls feature is only available on SUPERSET 3DN and SUPERSET 4DN telephones. Note that the Handsfree Answerback feature provides a similar type of functionality for Mitel 5020, 5220 and 5224 IP Phones, and SUPERSET 410, SUPERSET 420, SUPERSET 430, SUPERSET 4025, SUPERSET 4125, and SUPERSET 4150 telephones.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Support for SUPERSET 3DN and SUPERSET 4DN telephones is a purchasable option (Form 04, Option 98). The option is available only with the SX-200 ICP MX migration package, not with new installations.
- COS Option 682 (Key System - Directed Page - Internal DN Sets) enables and disables this feature.
- COS Option 682 (Key System - Directed Page - Internal DN Sets) and COS Option 600 (Telephone - Auto-Answer) cannot be enabled at the same time.
- If this feature is used in combination with the headset feature then calling tones are heard through the headset and not through the speaker.
- A Directed Page cannot be made to a set that has DND enabled.
- If Auto Answer for Directed Page Calls is activated at a set and if the prime line of the set is busy, calls to any other line appearances on the set will not be indicated with standard ringing. Instead, the calls will be indicated by New Call Ring (a single burst of ringing).

Programming

To program this feature, enable COS Option 682 (Key System - Directed Page - Internal DN Sets) in the COS of the telephone.

To provide feature key activation of auto-answer in response to directed page calls at SUPERSET 3DN, or SUPERSET 4DN telephones, program an AUTO-ANSWER feature key. (See Feature Keys.)

Operation

SUPERSET 3DN and SUPERSET 4DN Telephones:

To activate auto-answer for directed page calls:

- Press the AUTO-ANSWER feature key. The adjacent LCD indicator darkens, or,
- Dial the auto-answer feature access code, followed by the digit "1". If a feature key has been programmed, the adjacent LCD indicator darkens.

To deactivate auto-answer for directed page calls:

- Press the AUTO-ANSWER feature key. The adjacent LCD indicator clears, or,

- Dial the auto-answer feature access code, followed by the digit “2”. If a feature key has been programmed, the adjacent LCD indicator clears.

SUPERSET 3DN and SUPERSET 4DN Option

Description

Support for SUPERSET 3DN and SUPERSET 4DN telephones is a purchasable option. To use these sets, you must enable COS Option 98 "Support 3DN, 4DN and 400 series Set Types" in Form 4, System Options/System Timers.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

- Support for SUPERSET 3DN and SUPERSET 4DN telephones is a purchasable option (Form 04, Option 98). The option is available only with the SX-200 ICP MX migration package, not with new installations.
- SUPERSET 3DN and SUPERSET 4DN telephones can not belong to a paging group.
- If you enable COS Option 98, "Support 3DN, 4DN and 400 series Set Types", sub-attendants will support 6 [ii]LDNs and 6 Hold positions. See Subattendant - LDN Keys and Subattendant - Hold Positions for more information.

Programming

Enable COS Option 98, "COS Option 98 "Support 3DN and 4DN Set Types" in Form 4 (System Options/System Timers).

SUPERSET 3DN

Description

The SUPERSET 3DN telephone has 12 keys with LCD indicators available as Speed Call keys, Feature Access keys or Line Appearances. The lowest key must be the Prime Line appearance. In addition to these 12 keys, there is one red key (hold) and 9 fixed function keys.

Conditions

Support for SUPERSET 3DN and SUPERSET 4DN telephones is a purchasable option (Form 04, Option 98). The option is available only with the SX-200 ICP MX migration package, not with new installations. See SUPERSET 3DN and SUPERSET 4DN Option.

Programming

Ensure COS Option 98, "COS Option 98 "Support 3DN, 4DN and 400 series Set Types" is enabled in Form 4, System Options/System Timers.

Program SUPERSET 3DN telephones in Expand Set Subform of CDE Form 09; select their COS Options in CDE Form 03 (COS Define).

Operation

Refer to the specific feature.

SUPERSET 4DN

Description

The SUPERSET 4DN telephone can have up to 11 speedcall numbers and one line appearance (Prime Line). The set can alternately have up to 12 line appearances (including Prime Line) and no speedcall numbers. In addition to these 12 keys, there is one red key (hold), the **SuperKey**, four feature keys and six softkeys. The SUPERSET 4DN telephone incorporates a liquid crystal display (LCD) for line status indication, user prompting and displays such as message waiting, time and date.

Conditions

Support for SUPERSET 3DN and SUPERSET 4DN telephones is a purchasable option (Form 04, Option 98). The option is available only with the SX-200 ICP MX migration package, not with new installations. See SUPERSET 3DN and SUPERSET 4DN Option.

Programming

Ensure COS Option 98, "Support 3DN, 4DN and 400 series Set Types" is enabled in Form 4, System Options/System Timers.

Program SUPERSET 4DN telephones in Expand Set Subform of CDE Form 09 (Desktop Device Assignments); select their COS Options in CDE Form 03 (COS Define).

Operation

Refer to the specific feature.

SUPERSET 401+

Description

The SUPERSET 401+ telephone is a single-line digital telephone that employs DNIC line transmission technology. It provides a **Flash** key, a **Message** key with associated indicator, a red **Hold** key, and keys for adjusting the ringer and handset receiver volume. The set also has keys designed for use as Speed call keys or feature access key.

Conditions

Support for SUPERSET 400 telephones is a purchasable option (Form 04, Option 98). The option is available only with the SX-200 ICP MX migration package, not with new installations.

Programming

Program SUPERSET 401+ telephones in CDE Form 09 (Desktop Device Assignments); select their COS Options in CDE Form 03 (COS Define) and System Options and System Timers in CDE Form 04.

Operation

Refer to the specific feature.

SUPERSET 410

Description

The SUPERSET 410 telephone is a multi-line digital telephone that employs DNIC line transmission technology. The SUPERSET 410 telephone has 6 programmable keys, each with an associated LCD indicator. These keys can be programmed as speedcall keys, feature access keys or line appearances. The lowest key must be the prime line appearance. In addition, there is a **Message** key and a **Microphone** key, a red **Hold** key, and 7 other fixed function keys. The **Message** key and **Microphone** key both have an associated indicator lamp.

You can use up to three Programmable Key Modules (PKMs) with this set. The PKMs are daisy-chained together and connect to a MILINK port on the base of the set.

Conditions

Support for SUPERSET 400 telephones is a purchasable option (Form 04, Option 98). The option is available only with the SX-200 ICP MX migration package, not with new installations.

Programming

Program SUPERSET 410 telephones in CDE Form 09 (Desktop Device Assignments); in Form 04 (System Options/System Timers), and select their COS Options in CDE Form 03 (COS Define).

Operation

Refer to the specific feature.

SUPERSET 420

Description

The SUPERSET 420 telephone is a multi-line digital telephone that employs DNIC line transmission technology. The SUPERSET 420 telephone has 12 programmable keys with associated LCD indicators. These keys can be programmed as speedcall keys, feature access keys or line appearances. The lowest key must be the prime line appearance.

Directly below the 12 programmable keys is a 2 x16 alphanumeric display and three softkeys. The three softkeys allow set users to select command prompts that appear in the display. The set also provides a **Message** key and **Microphone** key with indicator lamps, and 8 fixed function keys without indicator lamps.

You can use up to three Programmable Key Modules (PKMs) with this set. The PKMs are daisy-chained together and connect to a MILINK port on the base of the set.

Conditions

Support for SUPERSET 400 telephones is a purchasable option (Form 04, Option 98). The option is available only with the SX-200 ICP MX migration package, not with new installations.

Programming

Program SUPERSET 420 telephones in CDE Form 09 (Desktop Device Assignments); CDE Form 11 (Data Circuit Descriptors , in Form 04 (System Options/System Timers), and select their COS Options in CDE Form 03 (COS Define).

Operation

Refer to the specific feature.

SUPERSET 430

Description

The SUPERSET 430 telephone is a multi-line digital telephone that employs DNIC line transmission technology. The SUPERSET 430 telephone has 12 programmable keys with associated LCD indicators. These keys can be programmed as speedcall keys or line appearances.

Directly below the 12 programmable keys is a 4 x 40 alphanumeric display and six softkeys. The softkeys allow set users to select command prompts that appear in the display. The set also provides a **Message** key and **Microphone** key with indicator lamps, and five fixed function keys without indicator lamps.

You can use up to three Programmable Key Modules (PKMs) with this set. The PKMs are daisy-chained together and connect to a MILINK port on the base of the set.

Conditions

Support for SUPERSET 400 telephones is a purchasable option (Form 04, Option 98). The option is available only with the SX-200 ICP MX migration package, not with new installations.

Programming

Program SUPERSET 430 telephones in CDE Form 09 (Desktop Device Assignments); in Form 04 (System Options/System Timers), and select their COS Options in CDE Form 03 (COS Define).

Operation

Refer to the specific feature.

SUPERSET 4001

Description

The SUPERSET 4001 telephone is a single-line, digital telephone with:

- seven Speed Call keys
- six fixed-function keys: Program, Hold, Flash, Message, Up Arrow, and Down Arrow
- handset and ringer volume control
- ringer pitch control
- message waiting lamp.

Conditions

Support for SUPERSET 4000 telephones is a purchasable option (Form 04, Option 112). The option is available only with the SX-200 ICP MX migration package, not with new installations.

Programming

Program SUPERSET 4001 telephones in CDE Form 09 (Desktop Device Assignments); in Form 04 (System Options/System Timers), and select their COS Options in CDE Form 03 (COS Define).

Operation

Refer to the specific feature.

SUPERSET 4015

Description

The SUPERSET 4015 telephone is a multi-line digital telephone with:

- alpha-numeric liquid crystal display (LCD)
- seven line keys, each with a built-in line status indicator
- eight fixed-function keys: SuperKey, Cancel, Hold, Redial, Transfer/Conference, Message, Up Arrow, and Down Arrow
- automatic selection of prime line or ringing line
- key selection of non-prime line
- handset and ringer volume control
- ringer pitch control
- message waiting lamp.

Conditions

Support for SUPERSET 4000 telephones is a purchasable option (Form 04, Option 112). The option is available only with the SX-200 ICP MX migration package (not with new installations).

Programming

Program SUPERSET 4015 telephones in CDE Form 09 (Desktop Device Assignments); in Form 04 (System Options/System Timers), and select their COS Options in CDE Form 03 (COS Define).

Operation

Refer to the specific feature.

For softkey operations, press the indicated dial pad * or # key.

SUPERSET 4025

Description

The SUPERSET 4025 telephone is a multi-line digital telephone with:

- alpha-numeric liquid crystal display (LCD) with contrast control
- three softkeys for feature access
- fourteen line keys, each with a built-in line status indicator
- ten fixed-function keys: SuperKey, Cancel, Hold, Redial, Transfer/Conference, Message, Microphone, Speaker, Up Arrow, and Down Arrow
- automatic selection of prime line
- key selection of non-prime line
- handsfree operation
- handset, speaker, and ringer volume controls
- ringer pitch control

- message waiting lamp.

Conditions

Support for SUPERSET 4000 telephones is a purchasable option (Form 04, Option 112). The option is available only with the SX-200 ICP MX migration package, not with new installations.

Programming

Program SUPERSET 4025 telephones in CDE Form 09 (Desktop Device Assignments); and assign the options in Form 04 (System Options/System Timers), in Form 12 (Data Assignment) and select the COS Options in CDE Form 03 (COS Define).

Operation

Refer to the specific feature.

SUPERSET 4125

Description

The SUPERSET 4125 is a multiline, digital telephone with

- Alpha-numeric, 2 line by 20 character, backlit, liquid crystal display (LCD) with contrast control
- RS-232 connector
- Three softkeys for feature access
- Fourteen line keys, each with a built-in line status indicator
- Ten fixed-function keys: SuperKey, Cancel, Hold, Redial, Transfer/Conference, Message, Microphone, Speaker, Up Arrow, and Down Arrow
- Automatic selection of prime line
- Key selection of non-prime line
- Handsfree operation
- Handset, speaker, and ringer volume controls
- Ringer pitch control
- Message waiting lamp.

Conditions

Support for SUPERSET 4000 telephones is a purchasable option (Form 04, Option 112). The option is available only with the SX-200 ICP MX migration package, not with new installations.

Programming

Program SUPERSET 4125 telephones in CDE Form 09 (Desktop Device Assignments); and assign the options in Form 04 (System Options/System Timers), in Form 12 (Data Assignment) and select the COS Options in CDE Form 03 (COS Define).

Operation

Refer to the specific feature.

SUPERSET 4150

Description

The SUPERSET 4150 telephone is a multi-line digital telephone with:

- alpha-numeric liquid crystal display (LCD) with contrast control
- six touch-sensitive softkey areas for feature access on the LCD
- fourteen line keys, each with a built-in line status indicator
- six fixed-function keys: SuperKey, Hold, Microphone, Speaker, Up Arrow, and Down Arrow
- automatic selection of prime line
- key selection of non-prime line
- handsfree operation
- handset, speaker, and ringer volume control
- ringer pitch control
- message waiting lamp.

Conditions

Support for SUPERSET 4000 telephones is a purchasable option (Form 04, Option 112). The option is available only with the SX-200 ICP MX migration package, not with new installations.

Programming

Program SUPERSET 4150 telephones in CDE Form 09 (Desktop Device Assignments); and in Form 04 (System Options/System Timers). Select their COS Options in CDE Form 03 (COS Define).

Operation

Refer to the specific feature.

For softkey operations, press the indicated touch-sensitive area of the display.

To clean the display without activating any softkeys, press SuperKey, find and select CLEAN LCD, and then clean the screen. To return to regular telephone operation, press SuperKey.

SUPERSET Interface Modules (SIM1 and SIM2)

Description

A SUPERSET interface module allows the connection of one PKM 12 or up to two PKM 48 devices to a SUPERSET 4025, SUPERSET 4125, or SUPERSET 4150 telephone.There are two models:

- SIM1 can connect to one PKM 12, or one or two PKM 48 devices..
- SIM2 can connect to one PKM 12, or one or two PKM 48 devices plus an analog device (telephone, modem, FAX machine) .

Conditions

The following conditions apply to this feature:

- You can only use a SUPERSET interface module (SIM1 or SIM2) with a SUPERSET 4025, SUPERSET 4125, and SUPERSET 4150 telephones.

- The SIM2 does not support visual indication of ringing at an analog device (for example 2500 set with a message waiting lamp).
- The SIM2 does not support visual message waiting indication at an analog device (for example 2500 set with a message waiting lamp).
- All SUPERSET 4025, SUPERSET 4125 and SUPERSET 4150's that are being prepared for the interface module functionality, should be at the latest firmware (system) revision as verified in the firmware command in maintenance.
- The power adapter for the SUPERSET 4025 must be plugged into the interface module.
- The power adapter for the SUPERSET 4125 must be plugged into the telephone.
- The power adapter for the SUPERSET 4150 must be plugged into the telephone. If this is not the case, you will loose full duplex functionality.

Programming

A PKM 12 or a PKM 48 connected to the SUPERSET interface module must be programmed in CDE Form 09 with the associated telephone.

A SIM2 analog port is programmed in CDE Form 12, Data Assignment. The port type is AIM.

Operation

Refer to Operation for the connected device.

Swap (Trade Calls)

Description

This feature permits a SUPERSET telephone user to alternate conversation between two calls. Call Swap places one call on hold while conversation continues with the other call. This feature is similar to the Broker's Call feature available on industry-standard telephones.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Swap operates only on Mitel telephones.
- There must be a party in conversation and a party on consultation hold for Swap to operate.
- A swap cannot be done when talking to the console or a non-busy extension.
- The SWAP/TRADE CALLS feature key on a Mitel 5020, 5220, 5224 and 5324 IP Phone, the Symbol MiNET Wireless Phone, and SUPERSET 4015, SUPERSET 410, or SUPERSET 3DN telephone, performs a Swap Campon if a camped on caller is present and may be swapped in. Only one of the Swap or Swap Campon functions is available at the telephone at one time; see Swap Campon.
- If there is another party in the call with a consultation hold then the Swap cannot be done.

Programming

For feature key activation of Swap from a Mitel 5020, 5220, 5224 and 5324 IP Phone, and SUPERSET 4015, SUPERSET 410, or SUPERSET 3DN telephone, program a SWAP/TRADE CALLS feature key (see Feature Keys).

Operation

Operation varies depending upon the type of set as follows:

SUPERSET 4001 and SUPERSET 401+ Telephones:

There is a party on hold and a party in conversation (hold key has been pressed and a new party called); proceed as follows:

- Press the HOLD key repeatedly to alternately place one call on hold and connect to the other call.

Mitel 5201 IP Phone

To use Swap on the 5201 IP Phone, Broker's Call must first be enabled.

- Press the TRANS/CONF key repeatedly to alternately place one call on hold and connect to another call.

Mitel 5010, 5212, 5215 and 5312 IP Phones, and SUPERSET 4015, SUPERSET 410, and SUPERSET 3DN Telephones:

There is a party on consultation hold and a party in conversation (e.g., the TRANS/CONF key has been pressed and a new party called); proceed as follows:

- Press the SWAP/TRADE CALLS feature key repeatedly to alternately place one call on hold and connect to the other call.

Family 1 and Family 4 Telephones (except the Symbol MiNET Wireless telephone):

There is a party on consultation hold and a party in conversation (e.g., TRANS/CONF has been pressed and a new party called, or a conference has been split). Proceed as follows:

- Press the SPLIT softkey to break up the three party conference. The party in conversation is held, the held party is connected to the set.
- Press the TRADE CALLS softkey repeatedly to alternately place one call on hold and connect to the other call.

Symbol MiNET Wireless Phone:

- Select TRD and then press Send.

Swap Campon

Description

This feature allows the user of a telephone to place the current call on hold and converse with a camped-on party. The telephone user can alternate between the two calls as required, form a 3-party conference, or release the telephone from the call, leaving the other two parties connected. This also applies to members of hunt groups. The first extension in the hunt group that does not have Do Not Disturb activated and is logged in (UCD agent hunt groups) is able to swap in the first waiting caller on the hunt group.

Normally the SUPERSET 3DN and SUPERSET 410 telephone Swap operation is a “swap calls” operation; but it becomes a “swap campon” operation when there is a party camped on, and a Swap Campon is possible.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- For Family 3, SUPERSET 410 and SUPERSET 3DN telephones, this feature is accessed by a SWAP/TRADE CALLS feature key.
- This feature is accessed by a switch-hook flash for industry-standard telephones and Family 2 telephones.
- The feature is only available when talking on a call.
- The user of a small display set (Mitel 5020, 5220, 5224 and 5324 IP Phones, and SUPERSET 420, SUPERSET 4025, and SUPERSET 4125 telephones) in the talking mode can have the option to forward a camped-on call with the FORWARD softkey or answer the camped-on call with the TRADE softkey.
- When a call camps on to a key line then all SUPERSET telephones using an appearance of the line are able to use the feature.
- The caller that is swapped in is the first waiting caller; see Campon.
- The feature is unavailable when:

  - a consultation hold is already in progress,

  - overriding a call or another party in the call is being overridden,

  - a console is in the current call,

  - another party has the current call on consultation hold,

  - a party has the call on hold,

  - the call is ringing back a party in the call (Station Transfer Security),

  - a Non-busy Extension is in the call,

  - the current call is a conference call with a CO line in the call.
- For industry-standard telephones, flashing for a waiting call takes priority over the flash features *Flash for Attendant*, and *Flash for Dial Tone*.
- For industry-standard telephone, COS Option 223 Flash Disable is ignored for this feature.

Programming

For industry-standard telephones and SUPERSET 4001 or SUPERSET 401+ telephones, enable COS Option 205 (flash for waiting call) and enable System Option 38 (switch-hook flash).

For feature key activation of Swap from a Mitel 5010, 5212, 5215 and 5312 IP Phone, or SUPERSET 4015, SUPERSET 410, or SUPERSET 3DN telephone, program a SWAP/TRADE CALLS feature key (see Feature Keys).

Operation

Industry-standard, SUPERSET 4001, and SUPERSET 401+ Telephones:

A call is in progress at the extension; a second caller camps on;

- Flash the switch-hook or press the FLASH key (SUPERSET 4001 or SUPERSET 401+ telephones). The current call is put on hold. The telephone is connected to the second caller.
- To return to the original conversation, flash switch-hook or press the FLASH key if the telephone is a broker.
- To form a conference, flash the switch-hook or press the FLASH key if the telephone is not a broker.
- To exit from the 3-party conference, while leaving the two other parties connected, hangup.

Mitel 5010, 5212 and 5215 IP Phones, and SUPERSET 4015, SUPERSET 410, and SUPERSET 3DN Telephones:

A call is in progress at the extension; a second caller camps on;

- Press the SWAP/TRADE CALLS feature key. The current call is put on hold: the telephone is connected to the second caller.
- To return to the original conversation, press the SWAP/TRADE CALLS feature key.
- To form a conference, press the TRANS/CONF key.
- To exit from the 3-party conference, while leaving the two other parties connected, hangup.

Family 4 Telephones (except the Symbol MiNET Wireless telephone):

A call is in progress at the set; a second caller camps on; the TRADE softkey appears:

- Press the TRADE softkey. The current call is put on hold. The telephone is connected to the second caller.
- To return to the original conversation, press the TRADE softkey again,
  OR
  Press the TRANS/CONF key to form a conference. To exit from the conference, while leaving the two other parties connected, press the CANCEL key.

Family 1 Telephones:

A call is in progress at the set; a second caller camps on; the CALL WAITING softkey appears:

- Press the CALL WAITING softkey. At the next prompt, press the TRADE CALLS softkey. The current call is put on hold; the telephone is connected to the second caller.
- To return to the original conversation, press the TRADE CALLS softkey,
  OR
  To form a conference, press the CONFERENCE softkey. To exit from the conference call, while leaving the two other parties connected, press the RELEASE ME softkey.

Symbol MiNET Wireless Phone:

- Select TRD and then press Send.

Symbol MiNET Wireless Phone

Description

Symbol® MiNET Wireless Phone enables work mobility with all the functionality and features users have come to expect from wired voice networks. Small, lightweight and supporting a complete range of call control features, the Symbol MiNET Wireless phone operates over Symbol Technologies' Spectrum 24 Wireless data networks (802.11b) using Voice-over-IP (VoIP). Features of the phone include:

- Three line by twelve character backlit display
- Headset port
- Programmable settings for Ring Type, Ring Tone, key Volume, Backlight, and Contrast Control Vibrating, visual and audio ring signals
- Message Waiting Indicator
- Access to the complete range of telephony features available on SX-200 ICP including speed dials, transfer/conference, call waiting, call forwarding, call hold, Do Not Disturb, and ACD agent functionality.

The Symbol MiNET Wireless Phone can be "twinned" with wireline (desk) phone to provide concurrent ringing and message waiting indication on both devices.

Conditions

Symbol Wireless phones are supported on the SX-200 ICP MX only.

Programming

Program Symbol MiNET Wireless phones that are to operate in shared mode (a mode of operation that requires that user log into the phone by entering a DN and PIN number) in CDE Form 09 (Desktop Device Assignments). Select their COS Options in CDE Form 03 (COS Define).

If desired, associate the Symbol MiNET Wireless phone with the user's desk phone. See Phone Twinning for programming instructions.

Operation

Refer to the specific feature.

System Fail Transfer (SFT)

Description

In the event of a major alarm condition on the system, the SX-200 ICP controller can switch two phones on ONS circuits at bay/slot/circuit locations 1/13/3 and 1/13/4 to Central Office (CO) trunks at locations 1/13/7 and 1/13/8 respectively. When normal system operation is restored, calls on the transfer circuits remain in effect until they are terminated.

System Fail Transfer is also called Power Fail Transfer (PFT).

Generally, the circumstances which cause a system fail transfer are:

- The link between the main controller and the equipment bay stops functioning (the system is cut over into SFT mode).
- On power-up, the system bay fails to initialize properly (the system is cut over into SFT mode). SFT follows a Critical Alarm.
- Commercial power failure with no backup power source (UPS) for the SX-200 ICP.
- Bay Power Supply failure.

Feature Availabilty

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- If a transfer takes place, any existing calls on the transferred trunks are dropped.
- If trunks are rotary dial only, DTMF sets may not be used for dialing.
- MITEL telephones and consoles cannot be System Fail Transfer extensions.

Programming

None

Operation

None

System Identifier

Description

A unique 1- to 3-digit identifier may be assigned to the system. It appears on traffic measurement and SMDR reports to identify the system when central polling equipment is used for traffic measurement, trunk SMDR, ACD, DATA SMDR, ACD SMDR and analog networking.

Conditions

- The system identifier is programmed from the attendant console.

Programming

To program the System Identifier from the SUPERCONSOLE 1000 or the 5540 IP Console:

1. Press the Function key.
2. Press the [ATT FUNCTION] softkey.
3. Press the [SYSTEM IDENT] softkey.
4. Enter a new three-digit number if required.
5. Press the [SET] softkey to confirm the new number.

Operation

None.

Tandem Operation

Description

The SX-200 ICP support two PABXs connected in tandem using tie trunks to connect the two systems together. See Analog Networking, Satellite PBX and Resale Package.

Conditions

The following conditions apply to this feature:

- Appropriate automatic route selection (ARS) must be provided.
- When a supervision is received from an outgoing trunk, if the calling party on the trunk is an incoming DID or Tie trunk then the supervision is passed on to the incoming DID or Tie trunk (the incoming trunk is given another answer supervision).
- Device interconnection must be set up appropriately to ensure that valid trunk connections are allowed. Device interconnection must be checked carefully when using loop start CO or DISA trunks to make outbound calls; see Device Interconnection Control.

Programming

None.

Operation

None.

TeleMatrix 3000IP Phone

Description

The dual-mode TeleMatrix 3000IP Phone (see Figure) is a multi-line telephone that can operate either as a standalone device connected to a SIP (Session Initiation Protocol) service provider or as part of a Mitel ICP system using the MiNet protocol. The TeleMatrix 3000IP Phone is sold and supported by TeleMatrix, Inc.

Features of the telephone include

- Twelve keys, each with a built-in status indicator
- Seven fixed-function keys: Cancel, Hold, Redial, Transfer/Conference, Message, Speaker, Mute
- Automatic selection of prime line or ringing line (the bottom line key)
- Key selection of non-prime line
- Handset and ringer volume control
- Hands-free speakerphone operation (half duplex)
- Handset, speaker, and ringer volume controls (Up Arrow and Down Arrow)
- Message Waiting lamp
- Dual Ethernet port to provide connectivity to the LAN for both your telephone and computer
- Adjustable tilt mechanism
- The Message key can be used to call an embedded voice mail system, regardless whether any messages are waiting in the user's voice mailbox. This functionality can be extended to a centralized voice mail system by programming the Voice Mail Number For This Tenant option on Form 19 (Call Rerouting Table).
- Powered from an Ethernet connection compliant with 802.3af.

Conditions

- Each IP Phone requires a license. Licenses are purchasable and enabled in Form 04, System Options/System Timers.
- System Option 131 in Form 04 and COS Option 280 in Form 03 are required to activate the phone's second Ethernet port.
- The TeleMatrix 3000IP Phone does not support the connection of headsets, Line Interface Modules (LIMs), SUPERSET Interface Modules (SIMs), or Programmable Key Module (PKMs).

Programming

SX-200 ICP

Program the phones in CDE Form 09 (Desktop Device Assignments) as device type TMX3K; enable the required options in Form 04 (System Options/System Timers), and select their COS Options in CDE Form 03 (COS Define).

Telephones

The dual mode TeleMatrix 3000IP Phone defaults to MiNET mode when first installed and becomes operational when registered with the SX-200 ICP. For instructions on registering phones, see Register the IP Phones.

To program a TeleMatrix 3000IP Phone for SIP mode operation, refer to the technical documentation available at www.telematrixusa.com.

Operation

Refer to the specific feature in the Program Features section of this document.

Teleworker Solution

Description

The Mitel Teleworker Solution connects a remote office to the corporate voice network to provide full access to voice mail, conferencing and all the other features of the office phone system.

It requires the following:

Corporate Site	Remote Site
• server installed with Managed Application Server software and Teleworker software • static IP address • sufficient internet bandwidth (approximately 50 kbps is required per teleworker if G.729 compression is enabled)	

- 5212 IP Phone, 5215 IP Phone, 5020 IP Phone, 5224 IP Phone, or 5220 IP Phone with phone load 5.22 or later

• DSL/cable router with Network Address Translation (NAT) and local DHCP • Broadband connectivity (static IP address is not required)

Note: The MAS server must be on the same VLAN (subnet) as the SX-200 ICP.

The Teleworker Solution can be completely configured at the corporate site using a compatible Mitel IP phone. Once configured, the phone can then be taken off-site and plugged into any broadband Internet connection. When the phone is powered up, it automatically establishes a connection with the Teleworker Gateway and is registered as a standard extension on the SX-200 ICP. The phone can also be returned to normal (non-teleworker) mode with the touch of a button.

Conditions

- Requires Teleworker Solution Version 3.0 or higher
- Each teleworker phone requires an IP license.

• The MAS self-study course qualifies technicians to implement a Mitel Teleworker for voice-only use. Implementation in a converged voice and data network with traffic shaping requires SX-200 ICP Advanced certification.

Programming

For installation and programming information, refer to the Managed Application Server Installation and Maintenance Guide and the Teleworker Solution Blade Guide, both available at Mitel OnLine.

Tenanting

Description

Economy of scale makes sharing SX-200 ICP services practical. Using the tenanting features, up to 25 small businesses, or departments of a larger business, can share the services of a SX-200 ICP system. Logically, the SX-200 ICP is divided into 25 separate PBXs, each providing its tenant with customized features and services.

Consoles, night bells, CO trunks, and dial-in trunks can either be shared between tenants or allocated individually to each tenant. Switching to night service can be done centrally, or by an individual tenant. Calls through the PBX can be blocked, so tenants can only call each other on CO trunks.

Unanswered or after-hours calls can be answered by a “landlord” console. Tenants can gain additional flexibility by using Family 1 telephones as subattendant positions. The main console position could be handled by a Family 1 telephone, using line keys to receive tenant recalls.

Data calls may be assigned to separate tenants, as required, to control access to data devices.

Refer to the Tenanting section.

Conditions

The following conditions apply to this feature:

- A maximum of 25 Tenant groups may be programmed.
- If tenanting is used to segregate departments, CDE Form 06 (Tenant Night Switching Assignment) must be programmed accordingly.
- Tenants may have names associated with them.

Programming

Assign consoles to the desired tenant groups via CDE Form 07 (Console Assignments).

Assign extensions to the desired tenant groups via CDE Form 09 (Desktop Device Assignments).

Assign trunks to the desired tenant groups via CDE Forms 14 (Non-Dial-In Trunks) and 15 (Dial-In Trunks).

Assign a tenant number to ACD Paths via Form 41 (ACD Path Form).

Assign the desired tenant name, day and night service numbers in CDE Form19 (Call Rerouting Table).

Program CDE Form 06 (Tenant Night Switching Control) if tenanting is used to segregate departments.

Operation

None.

Toll Control

Description

The toll control feature forms part of the automatic route selection (ARS) feature. It allows the system to restrict external calls placed by designated groups of extensions. This may mean denying all outside calls, denying calls to specific locations, denying calls over expensive routes, or any combination of these.

See Automatic Route Selection (ARS) and Class of Restriction (COR) in this section; note that this is not key system toll control.

Conditions

Refer to the Automatic Route Selection and Toll Control section.

Programming

Refer to the Automatic Route Selection and Toll Control section.

Operation

None.

Tone Demonstration

Description

The tone demonstration feature familiarizes users with the tones the system generates. This feature also allows Mitel telephone users to adjust ringer volume and pitch.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Refer to Ringer Control for instructions on how to adjust ringer volume and pitch.
- If no new tone is selected after a while then the tone demonstration feature is canceled and regular dial tone to allow normal dialing is returned to the extension.

Programming

Assign an access code to Feature 27 (Tone Demonstration).

Operation

Dial the Tone Demonstration feature access code plus the code for the tone to be played. The codes, the tones, and the main uses of each tone in the system are as follows:

- Codes 1 through 10, 16, 22 and 25 are unassigned

11 - Dial Tone

This tone is heard when dialing can start, for internal callers as well as for external callers. ARS dial tone, interrupted dial tone and transfer dial tone use this basic tone.

12 - Transfer Dial Tone

This tone is heard when dialing can start and the extension has a party on consultation hold; see Transfer Dial Tone. The tone is the same as interrupted dial tone.

13 - Busy Tone

This tone is heard when:

- a busy device is reached and campon to the device is not allowed (device type, COS options, etc.)
- during campon to a busy device
- devices are busy when a feature is accessed such as paging.

14 - Special Busy Tone

This tone is heard by the calling party when a busy device is reached and campon to the device is allowed. Only some types of external callers receive this tone; see Campon.

15 - Standard Ringback Tone

This tone is heard by the calling party when a device is being rung. This is supplied to internal and external callers.

17 - Reorder Tone

This tone is heard when an illegal operation is attempted or when an invalid number is dialed (or no number is dialed at all); see Illegal Access Intercept and Vacant Number Intercept also. This is the same tone as busy tone; however, the cadence is different.

18 - Conference Tone

This tone can be heard when the attendant is in a conference; see Attendant Conference.

19 - Call Waiting (Campon) Tone

This tone is heard by a busy extension when another extension camps onto the busy extension; see Campon Warning Tone.

20 - Intrusion (Override) Tone

This tone is heard by a device overriding a call; see Override (Intrude).

21 - Interrupted Dial Tone

If call forwarding or Do Not Disturb is activated at a set, a user hears interrupted dial tone followed by regular dial tone when the user goes off-hook. Interrupted dial tone is regular dial tone broken by short intervals of silence.

23 - Waiting Tone (Silence or Music-on-Hold)

This is the tone supplied to devices on hold or consultation hold and to devices camped on to busy devices, see Music-on-Hold and Campon.

24 - Paging Tone

This tone is heard when the pager is successfully accessed; see Paging.

26 - Trunk Camp-on Double Beep Tone

This tone is heard by an extension when a trunk camps onto an extension: see Campon Warning Tone.

27 - ARS Warning Tone

This tone is heard when an expensive route is accessed in ARS; see Expensive Route Warning.

28 - ARS Dial Tone

This tone is heard if dial tone is supplied from ARS during an outbound call. Refer to the Automatic Route Selection and Toll Control section.

29 - Override Warning Tone

This tone is heard by parties in a call while a conversation is being overridden; see Override (Intrude).

30 - Privacy Release Tone

This tone is heard when an extension enters an existing call on a key, direct trunk select, or private trunk line; see Privacy Enable/Privacy Release.

31 - Auto-Answer Call End Tone

This tone is heard when a SUPERSET telephone that has the auto-answer feature enabled ends a call; see Auto-Answer.

32 - Attendant Error Tone

This tone is heard when the attendant attempts an illegal operation.

33 - Ringer Pitch Adjust

Dialing this code starts the SUPERSET telephone ringing, and enables the ringer pitch adjust mode for the volume keys on the telephone.

Tone Plans

Description

Numerous tones can be generated when dialing telephone numbers. To accommodate these, the system loads the required country-specific tone plan from the Flash card.

Refer to Call Progress Tones, in the Engineering Information section, for details of tone plans.

Conditions

None.

Programming

Tone plans are selected via System Option 60 (Tone Plan) in CDE Form 04 (System Options and Timers).

Operation

None.

Traffic Measurement

Description

Traffic measurements can be made on SX-200 ICP systems, and the results printed through a printer port (see Printer / Terminal Support Directed I/O*)*. The types of measurements made include the following:

- DTMF receiver activity
- DTRX calls
- CLASS activity
- System activity
- Pseudo receiver activity
- Console activity
- Feature activity
- Guest room activity

- Hunt group activity
- Line and trunk activity
- PCM channel activity
- Trunk activity
- Trunk group activity.

Information is accumulated during a user-programmed time period, and is then available for output. Programming is done from the maintenance terminal or from the attendant console.

Refer to the Traffic Measurement section.

Conditions

None.

Programming

Refer to the Traffic Measurement section.

Operation

None.

Transfer

Description

This feature allows a telephone user, on an established call, to put the call on consultation hold, dial a third party, and transfer the second party to the third party. The transfer can be done before the third party answers, after the third party answers, or if the third party is busy (Transfer Into Campon).

Mitel telephone users can consult privately with each party using the Swap feature (see Swap) and then form a conference or perform a transfer.

The user of an industry-standard telephone can consult privately with the third party only once prior to transferring or conferencing.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- When an extension with a consultation hold in progress calls the console and the console answers, the console has the caller as the DEST party and the party held by the caller as the SOURCE party. The calling party no longer has the third party on consultation hold - it is on hold by the console.
- When a party is transferred to a third party while the third party is ringing, the display of the third party is updated to show the identity of the new caller.
- When pressing the DSS page button with a caller on soft hold, upon hang-up, the caller on soft hold is placed in a handsfree talk state with the destination. This may cause confusion and the caller's private conversation to be heard.
- Mitel display telephones provide visual indications on the display that a consultation hold is in progress. Family 1 telephones identify the call on consultation hold.

- For transfers to campon, see Campon.
- COS Options 212 (Can Flash If Talking to an Incoming Trunk) and 213 (Can Flash If Talking to an Outgoing Trunk) must be enabled in the class of service of the transferring extension to allow trunk call transfers.
- Device Interconnection--as specified in the device's COS (Options 313 through 319) and in Form 30 for trunk-to-trunk connections--checking is done between the party being transferred and the party being transferred to. An exception is when a direct trunk select trunk line is being transferred.
- See Transfer Security (Recall) for situations where the transferring party accidentally hangs-up or attempts an illegal transfer.
- Private trunk lines cannot be transferred.
- Conferences can be put on consultation hold.
- Single parties on consultation hold receive connected music.
- Calls may not be transferred to the paging circuit or to a conference.
- COS Options 223 (Flash Disable), 400 (Contact Monitor), 228 (Manual Line), 224 (Flash for Attendant) prevent an extension from putting a party on consultation hold.
- COS Option 214 (Cannot Dial a Trunk After Flashing) and COS Option 215 (Cannot Dial a Trunk if Holding or Conf with One) prevent trunks from being accessed as the third party under certain circumstances.
- An extension with COS Option 233 (Never a Consultee) may not be consulted.
- See Transfer Dial Tone for special dial tone on transfer.
- A call cannot be put on consultation hold if:
  - the party is an attendant,
  - the call has an incoming trunk in it and the Flash on Incoming Trunk feature is enabled (the same applies for outgoing trunks),
  - the call is on a private trunk line,
  - there is a party in the call using the ADD HELD feature enabled,
  - a party in the call is holding the call,
  - an extension in the call has the Non-Busy Extension feature,
  - the call is ringing back one of the parties in the call (Station Transfer Security),
  - another party in the call is being overridden,
  - the extension is overriding a party in the call,
  - there are five parties in the call,
  - the call is already on consultation hold,
  - the call is a conference and the extension has the Broker’s Call or Transfer With Privacy feature,
  - the call is a conference and the Flash In Conference option is disabled,
  - the called party in a member of a voice mail hunt group.
- When using Phonebook to transfer a call, the user, after entering the destination party's name and hearing the prompt "you are being transferred to <name>" must immediately hang up. Waiting too long will cause the transfer to fail and the user to reconnect to the party on softhold.
- COS Option 689 (DTS/CO Line Transfer Call Handling) allows incoming calls on a DTS/CO line to stay on the DTS/CO line key when transferred unsupervised between extensions that have an appearance of the line. Without Option 689, the call will use both the DTS/CO line key and the prime line, key line, or multicall line—whichever was used to answer the call at the extension receiving the transfer. For supervised transfers, the call remains on the prime line; it moves to the DTS/CO line key only when it is placed on hold using the red Hold key. **Note:** This feature requires Feature Level 5 (Form 4, System Option 102).

Programming

See Flash Control for restrictions on consultation hold.

See Conference for programming to allow conferences of more than three parties.

Operation

Industry-standard Telephones:

On an established call:

- Flash the switch-hook - dial tone is returned; the current party (second party) is on consultation hold and hears music, if provided.
- Dial the number of the third party.
- If desired, wait for the third party to answer (if the number is busy or does not answer, flash the switch-hook to return to the held call).
- If desired, consult privately with the third party.

Family 1 Telephones:

On an established call:

- Press the TRANS/CONF softkey - transfer dial tone is returned. The current party (second party) is on consultation hold and hears music, if provided.
- Dial the number of the third party.
- If desired, wait for the third party to answer. (If the number is busy or does not answer, press the BACK TO HELD softkey to return to the held party. If another call is desired without returning to the held party (e.g., wrong number), use the NEW CALL softkey).
- If desired, consult privately with this party.
- If desired, switch between parties by pressing the TRADE CALLS softkey. The conversation with each party is private.
- To connect both parties: hang up, or press the RELEASE ME softkey.

Family 2 Telephones:

On an established call:

- Press the FLASH key - transfer dial tone is returned, the current party (second party) is on consultation hold and hears music, if provided.
- Dial the number of the third party.
- Replace the handset,
  or,
  If desired, wait for the third party to answer and provide an introduction. If you receive busy tone, or if the called party doesn't answer, hang up. You will be recalled by the original party.

Family 3 , Family 4, and Family 5 Telephones (except Symbol MiNET Wireless Telephone):

On an established call:

- Press the TRANS/CONF key - transfer dial tone is returned, the current party (second party) is on Consultation Hold and hears music, if provided.
- Dial the number of the third party.
- Replace the handset,
  or,
  If desired, wait for the third party to answer and provide an introduction (if the number is busy or does not answer, press the CANCEL key or the TRANS/CONF key to return to the held call).
- If desired, switch between parties by pressing the SWAP feature key. The conversation with each party is private. Note that the SWAP key is not available on the Mitel 5201 IP Phone.

- To connect both parties: hang up, or press the RELEASE softkey (if provided).

Symbol MiNET Wireless Phone:

To transfer an active call:

1. Press the FCT key.
2. Use the scroll keys to locate "Trans/Conf."
3. Press the SEND key to select Trans/Conf.
4. Dial the number of the third party.
5. Do one of the following:
   - To complete the Transfer, press the END key.
   - To announce the Transfer, wait for an answer, consult and press the END key.
   - To swap between the incoming call and the called user during the transfer, press the left-hand softkey (TRD).

- To cancel the Transfer, press the FCT key, use the scroll keys to locate Cancel, and then press the SEND key.

SUPERSET 3DN Telephones in headset operation:

On an established call:

- Press the TRANS/CONF key - transfer dial tone is returned, the current party (second party) is on Consultation Hold and hears music, if provided.
- Dial the number of the third party.
- Press the switch-hook to transfer the call; then press the CANCEL key to return your set to idle, or,
  If desired, wait for the third party to answer and provide an introduction (if the number is busy or does not answer, press the CANCEL key to return to the held call).

Transfer Dial Tone

Description

The transfer dial tone feature supplies a tone to indicate that an extension has a call on consultation hold. Transfer dial tone is returned when an extension places an established call on hold to consult with another party or to transfer the call. Transfer dial tone is 350/440 Hz, three bursts of 100 ms on, 100 ms off, followed by continuous tone. Regular dial tone is 350/440 Hz continuous tone.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- The console is not affected.
- COS Option 701 (No Dial Tone) must be disabled.
- Mitel display telephones have indicators on the display to show that there is a consultation hold in progress.
- This is the same tone used for discriminating dial tone.

Programming

Enable COS Option 251, Transfer Dial Tone for the extension.

Operation

Establish a call.

Place call on consultation hold. Transfer dial tone is returned to indicate consultation hold.

Transfer Security (Recall)

Description

This feature is designed to prevent the dropping of mishandled calls. If an extension, during transfer, hangs up before completing dialing, or if the transfer is not allowed, the call that was placed on hold by the original extension flashing, automatically calls back to that extension. This also applies to conference calls.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- See Inhibit Trunk Ring-me-back During Dialing for an exception for this feature.
- MITEL telephones with displays indicate the special ring-me-back calls.
- The ring back only affects the party being rung back - other appearances of the line do not ring.
- The features available to extensions ringing an extension (callback, messaging) are available to the extension ringing back.
- The parties in a conference ringing back do not hear any tones.
- There is no time limit on the conference ringing back.
- Forwarding - no answer and recall operate for a single caller ringing back. All types of forwarding, and recall are ignored for a conference ringing back.
- The Trunk Recall Partial Inhibit feature blocks this feature in a particular situation.
- The Directed Call Pickup feature has no effect on calls ringing back.
- Other appearances of the extension being rung back cannot answer the call.
- An attempted flash to go back to a consultation hold could be interpreted by the system (if it is too long) as a hang up attempt. The industry-standard telephone user could then possibly find itself reconnected to the held party with no ringing heard at the station.
- If an illegal transfer is attempted then the transferring extension is rung back.

Programming

None.

Operation

None.

Trunk Answer From Any Station (TAFAS)

Description

This feature allows the user to answer incoming calls appearing at common alerting devices (night bells). The user can answer calls for a single tenant or for all tenants in the system. The answering extension can then invoke any feature associated with the incoming call that is normally available at that extension. TAFAS can also be used to answer calls which ring at the console during the day or night.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- Tie trunks, industry-standard telephones, Mitel telephones (including subattendants), and consoles can access the feature.
- The NIGHT BELL softkey appears at the console only if the appropriate COS options are enabled for the console and a call of the correct tenant is ringing the night bell.
- Extensions cannot have a consultation hold in progress when attempting to answer the TAFAS call.
- Night Bells have no tenant assignment; the tenant used is always that of the caller.
- Device interconnection control checks must pass between the caller and the device answering the call.
- All calls ringing all consoles, LDN keys (including those on subattendant phones), or night bells in the system are available, depending upon the tenant checks made.
- During day service, you can answer recalls to the attendant console with the TAFAS feature (COS Option 250, TAFAS Access During Day Service must be enabled on the set).
- During day or night service, you can answer calls at common alerting devices (night bells) and at the console with the TAFAS feature (COS Option 281, TAFAS Access During Day/Night Service must be enabled on the set.)
- The lamp associated with the NIGHT ANSWER feature key will be lit solid to indicate that the ringing TAFAS call be picked up with the key.

Programming

To restrict access to the device's tenant only, assign an Access Code to Feature 15, TAFAS - Local Tenant and enable COS Option 249, TAFAS Access - Tenant for this device.

To allow access to all tenants, assign an access code to Feature 14, TAFAS - Any and enable COS Option 248, TAFAS Access - Any for this device.

If desired, enable COS Option 250, TAFAS Access During Day Service; this permits users to answer calls ringing the console during the day.

If desired, enable COS Option 281, TAFAS Access During Day/Night Service; this permits users to answer calls ringing at common alerting devices (night bells) and at the console during the day or night.

Operation

Operation varies depending upon the device type as described below.

Family 2, Symbol MiNET Wireless, and Industry-standard Telephones:

To answer a TAFAS call:

- When the alerting device is heard, go off-hook.
- When dial tone is heard, dial the appropriate TAFAS code.

The incoming call is now connected to the extension.

Family 3, Family 4 (except Symbol MiNET Wireless), Family 5, SUPERSET 410, TeleMatrix 3000IP, and SUPERSET 3DN Telephones:

To answer a TAFAS call (in the local Tenant group) or any TAFAS call:

- When the alerting device is heard, go off-hook.
- Press the NIGHT ANSWER feature key.

The incoming call is now connected to the extension.

Family 1 Telephones:

To answer a TAFAS call (in the local Tenant group) or any TAFAS call:

- When the alerting device is heard, go off-hook.
- Press the NIGHT ANSWER softkey.

The incoming call is now connected to the extension.

Attendant Console:

To answer a TAFAS call at the console:

- When you hear the alerting device, press the NIGHT BELL softkey.

The incoming call is now connected to the console.

Trunk Circuit Descriptor Options

Description

Trunk circuit descriptors specify the programmable hardware parameters of each trunk circuit. The system supports a maximum of 25 different trunk circuit descriptors. Each trunk in the system must have a trunk circuit descriptor number with an associated set of selected options.

Conditions

None.

Programming

Select Trunk Circuit Descriptors in CDE Form 13 -Trunk Circuit Descriptors. If the system is to provide pseudo answer supervision, set an appropriate time for the time-out period via System Option 49 - Pseudo Answer Supervision Timer. See the Program CDE section.

Operation

None.

Trunk Dial Tone Detection

Description

After accessing a trunk the system tries to detect dial tone on it. If dial tone is detected before time-out, the system begins sending digits. If no dial tone is detected after the time-out period and limited wait is specified, the system automatically begins sending digits. Otherwise it removes the trunk from service and indicates an alarm. Dial tone detection, without the alarm, can also occur in the middle of trunk dialing; see Automatic Route Selection.

Conditions

The following conditions apply to this feature:

- Enabling System Option 48 (Limited Wait For Dial Tone) disables COS Option 805 (Trunk No Dial Tone Alarm).
- The no dial tone alarm time is set at 10 seconds.
- When the trunk is removed from service, the party dialing on the trunk ends dialing and is given busy tone.
- 50 simultaneous dial tone detections can occur in the system at any one time.
- If dial tone detection resources are not available and the alarm is enabled then the trunk is removed from service. If the alarm is not enabled then dial tone is assumed to have been supplied and dialing commences after a short pause.

Programming

Set System Option 48 (Limited Wait for Dial Tone) for 1-15 seconds.

To program a trunk to wait the limited time for dial tone, select COS Option 802 (Limited Wait for Dial Tone) for the trunk.

To enable the no dial tone alarm for a trunk, enable COS Option 805 (Trunk No Dial Tone Alarm).

Operation

None.

Trunk Groups

Description

Trunk groups are defined and used in the ARS forms in CDE to control extension access to trunks, to define trunk options, and to apply features to trunk groups. For further information, refer to the Automatic Route Selection and Toll Control section, and to the Program CDE section.

Conditions

The following conditions apply to this feature:

- SMDR is applied to trunks on a per trunk group basis; refer to the Station Message Detail Recording section.
- A maximum of 50 individual trunk groups are available.
- A maximum of 50 trunks are permitted in an individual trunk group.
- A trunk may be a member of only one trunk group.
- Individual trunks must be programmed before they are entered into trunk groups.
- Direct trunk select and private trunk line trunks must be in trunk groups before outgoing calls can be made on the trunk lines.
- Direct trunk select and private trunk line trunks are never hunted from a trunk group.

- A trunk is selected from a trunk group if it is idle, if it is not being seized by another party, if a no seize alarm is not pending and if the caller and the trunk can connect together.
- A trunk does not need to be in a trunk group to originate a call.
- Trunk groups can have circular or terminal hunting.

Programming

Program trunks into Trunk Groups via CDE Form 16 (Trunk Groups).

Each trunk group may also be given a unique name via CDE Form16 (Trunk Groups).

System Option 49 - Pseudo Answer Supervision Timer applies to all trunks in the system.

Refer to the Automatic Route Selection and Toll Control section.

Operation

None.

Trunk Operation - Direct Inward Dial (DID)

Description

The DID trunk type is one of the four trunk types, independent of the hardware actually used to support the types, used in the system.

DID trunks allow incoming trunk calls to reach extensions without attendant intervention or assistance. The length of the incoming number, the number of digits to be absorbed, and a prefix digit, if required, can also be specified through CDE programming.

Calls arriving at the PBX on DID type trunks are assumed to be outside calls. Callers therefore receive different call progress tones. Call handling differs from tie and DISA trunk type calls, which are assumed to be internal calls.

For the hardware that supports the DID trunk type operation, see Trunk Support - Direct Inward Dial (DID), Trunk Support - T1, and Trunk Support -E&M.

Also see DID/Dial-In/Tie Intercepts.

Conditions

The following conditions apply to this feature:

- If the DID trunk sends less digits than expected, the trunk receives reorder tone from the system after the inter digit time-out (15 s).
- DID trunks can dial any access code for any type of device.
- DID trunks can dial the DISA access code (the DISA access code used with DID and Tie trunks must not lead with an asterisk *, if programming ANI/DNIS as well). After this access code, the DID trunks (analog, T1, and PRI) get dial tone, and then can dial what their COS and COR allow.
- DID trunks can dial Account Codes.
- An extension with Option 226 (Inward Restriction DID) in its COS cannot receive a call directly from a DID trunk. It is treated by the system as an illegal number.
- Calls attempted using vacant or illegal numbers can be routed to answering points for completion; see Illegal Access Intercept and Vacant Number Intercept.

- DID trunks receive ringback tone when calling and camping onto a busy device. They receive busy tone if they reach a busy device and no busy intercept occurs (see DID/Dial-In/Tie Intercepts) and the trunk cannot camp on (see Campon). If it can campon, it does so immediately.
- DID trunks ignore the DND feature if enabled on the called extension if there is no Do Not Disturb Routing for the called extension's tenant; also see DID/Dial-In/Tie Intercepts.
- DID trunks are answered before being connected to a recording device.
- If there is no DID rerouting, DID trunks ignore the DND that is enabled on the called party.
- The attempt to call the destination is done only when the specified number of digits to be received on the trunk have been received. Extra digits are ignored and no matches are done for access codes that contain less than the specified number of digits.
- Digits are absorbed before the prefix digits are inserted.
- The trunk may be programmed to ignore incoming DTMF digits, and recognize only rotary digits.
- A DTMF receiver is needed when a DID trunk originates. If there is no receiver available then the trunk waits indefinitely until a receiver is available. Digits received on the trunk are stored and processed when a receiver is available. The Traffic Measurement feature can be used to track the wait for receiver.
- If a seize error occurs on a DID trunk then the trunk is given reorder tone.
- Prefix digits are used for matching access codes only after the DID trunk has dialed at least one digit that is not absorbed and the expected number of digits have been received. The prefix digits by themselves cannot be used to match an access code.

Programming

To program a trunk to ignore incoming DTMF digits during origination of the call, enable COS Option 801 (Incoming Trunk Call Rotary) for the trunk.

Program a DID trunk in CDE Form 15 (Dial-in Trunks).

Note: CDE Form 15 does not accept trunk programming changes if the T1 card or T1/E1 module is installed and the T1 link is down but still considered active. To make a programming change for a T1 card, unplug the T1 card, make the programming change and then re-insert the T1 card . To make a programming change for a T1/E1 module, insert a Peripheral Interface Card (PIC) into the software location (slots 5 or 6) for that T1 link, make the programming change, and then remove the PIC from the slot to reset the T1 link.

Operation

None.

Trunk Operation - Direct Inward System Access (DISA)

Description

The DISA feature allows an external caller to access the system by dialing the directory number of a special DISA trunk and then dialing a security code. After the code is dialed, the system returns dial tone to the caller, who may then access features in the DISA trunk's COS except those which require a switch-hook flash.

Optionally, the external caller can be forced to enter a special account code rather than the standard DISA security code. See Verified Account Codes (Special DISA).

DISA trunks can be supported on many different hardware types. See Trunk Support - T1, Trunk Operation - Tie, Trunk Support - E&M, and Trunk Support - Direct Inward Dial.

A trunk can be programmed as DISA at all times, or during night service only.

The Tie/DID trunks on the LS/GS trunk cards, LS/CLASS trunk modules, DID trunk cards, T1 cards, and PRI cards support DISA.

Conditions

The following conditions apply to this feature:

- After the trunk originates, the system waits a programmable time, and then answers the trunk; then dial tone is returned.
- After the preprogrammed wait, the system answers the trunk before dialing begins.
- The outside caller must use a DTMF telephone.
- The security code may be one to five digits in length.
- The same security code applies to all DISA calls.
- If a caller dials an invalid code, the call is dropped only after three digits have been dialed. This makes it a bit more difficult for unwanted callers to determine what the security code is. This does not apply when a verified account code is used.
- Three service modes -- DAY, NIGHT1, or NIGHT2 -- are available to tenant groups (see Night Services). If COS Option 810 (DISA During Night Service Only) is enabled for a trunk, the DISA feature is disabled for the trunk's tenant during DAY service mode. Therefore, during DAY service mode, an external caller can call in on the trunk without entering the DISA security code.
- Reorder tone is not returned to the caller when an invalid security code is dialed.
- A switch-hook flash is not possible on a DISA trunk.
- DISA trunks follow the illegal, do not disturb and vacant number routing as an internal industry-standard telephone would.
- The DISA access code used with DID and Tie trunks must not lead with an asterisk *, if ANI/DNIS is to be collected.
- The DISA access code is delivered when the calling party is a DISA DID/Tie Trunk and dials ARS with a *05.
- A DISA trunk is given the same call progress tones as an internal extension.
- The Alternate Trunk Recall feature applies to DISA trunks.
- If a loop start CO trunk is the hardware supporting the trunk then loop start interconnection rules apply and the restrictions on loop start CO trunks applies; see Device Interconnection Control and Trunk Operation - Non-dial-in.
- If the DISA trunk caller cannot complete a call then the caller must hang up and try again.
- If there is no DTMF receiver available for dialing when the DISA trunk originates a call, the initial answer delay period is extended indefinitely until a DTMF receiver is available.
- No call progress tone is provided by the system to the trunk in the initial answer delay period.
- Up to seven digits can be absorbed from the trunk.
- Access to the allowed features is controlled by the COS assigned to the trunk.
- The features available to DISA trunks include:

  -account codes

  -ARS

  -console hold slot retrieve

  -paging

  -system abbreviated dial.

Programming

If the trunk is to be a DISA trunk during Night Service only, enable COS Option 810 (DISA During Night Service Only) for the trunk, and set the DISA trunk DAY service routing in CDE Form 19 (Call Rerouting Table) for the trunk's tenant.

Select options for the DISA trunk type via CDE Form 13 (Trunk Circuit Descriptors).

Note: There is no dedicated descriptor for an LS/CLASS DISA circuit. Putting an LS/CLASS trunk in Form 15 is all that is required to make it a DISA trunk.

Select options for the specific DISA trunk via CDE Form 15 (Dial-In Trunks). Assignment of a DISA circuit descriptor to a trunk changes the trunk to a DISA trunk.

Assign a DISA access code to Feature 19 (Direct Inward System Access). See Attendant DISA Code Setup.

Note: The DISA access code used with DID and Tie trunks must not lead with an asterisk *, if programming ANI/DNIS as well. The DISA access code is delivered when the calling party is a DISA DID/Tie Trunk and dials ARS with a *05.

Select the DISA answer timer via System Option 54 (DISA Answer Timer: 1-8 Seconds).

Enable COS Option 812 (Loop Start Trunk to ACD Path Connect) to allow loop start DISA trunks to access the ACD feature.

Operation

To access the System:

Dial the required directory number from a DTMF telephone.

The system waits the DISA answer time before answering the trunk and supplying dial tone.

Dial the DISA security code - dial tone is returned again.

Dial the required feature access code or extension.

Trunk Operation - Non-Dial-in CO

Description

CO trunks usually carry calls between the local central office and the PBX. Calls arriving on CO trunks are assumed to be outside callers. Callers therefore receive different call progress tones. Call handling differs from tie and DISA trunk type calls, which are assumed to be internal calls.

CO trunks are assigned an origination point for DAY, NIGHT 1, and NIGHT 2 service. They can optionally be assigned as a dedicated line on a SUPERSET telephone.

For the hardware that supports CO trunk type operation, see Trunk Support - CO (LS/GS), Trunk Operation - Tie, Trunk Support - DID, Trunk Support E&M, and Trunk Support - T1.

The NIGHT1 or NIGHT2 service for CO trunks can be changed directly from the Attendant console; see Night Service Flexible.

Also see Direct-In Lines.

Conditions

The following conditions apply to this feature:

- A DISA trunk may be programmed to be a CO trunk type during day service. See Trunk Operation - Direct Inward System Access (DISA).
- Connection checking is done between the trunk and the destination point when the trunk originates.

- If the trunk origination is blocked then the trunk stays idle and the PBX does no further processing of the call.
- Trunks are always answered before listening to a recording device.
- The processing of the origination follows the operation of call rerouting; see Call Rerouting.
- For trunks assigned to DTS or private trunk lines, the trunk originates to the line and ignores the NIGHT/DAY points programmed.
- The audio heard by the caller before answer depends upon the hardware/circuit descriptor connected to that trunk. For CO trunks, CO tones provided by the central office are heard; for trunks, based on dial-in hardware circuits, PBX call progress tones provided by the PBX are heard.
- The trunk receives ringback tone when it camps on to a device.
- The Device Interconnection feature can be used to prevent loop start trunk interconnection.
- If a CO trunk in a bay fails to get resources required for a call when it originates, the trunk will wait for resources to become available.
- If the trunk is routed out to an external trunk call and the CO trunk has not been answered yet, the CO trunk is answered when dialing is finished on the outgoing trunk.
- CO trunks cannot dial ARS directly; however, ARS can be accessed via system abbreviated dial.
- If no routing points are programmed in form 14 and there is no key appearance of the trunk, the trunk will not ring.

Programming

Select options for the specific trunk circuits via CDE Form 14 (Non-Dial-In Trunks) for incoming calls.

Note: CDE Form 14 does not accept trunk programming changes if the T1 card or T1/E1 module is installed and the T1 link is down but still considered active. To make a programming change for a T1 card, unplug the T1 card, make the programming change and then re-insert the T1 card . To make a programming change for a T1/E1 module, insert a Peripheral Interface Card (PIC) into the software location (slots 5 or 6) for that T1 link, make the programming change, and then remove the PIC from the slot to reset the T1 link.

Enable COS Option 812 (Loop Start Trunk to ACD Path Connect) to allow loop start trunks to access the ACD feature.

Refer to the Automatic Route Selection and Toll Control section, for the selection of options for outgoing calls.

Operation

None.

Trunk Operation - Tie

Description

The Tie trunk is one of four trunk types, and independent of the hardware actually used to support the types, used in the system.

Tie trunks allow incoming trunk calls to reach extensions directly, without attendant intervention or assistance. The number of digits expected from the trunk is unknown. Digit absorption and adding prefix digits can be done.

Calls coming into the PBX on Tie type trunks are assumed to be callers from inside the company, similar to DISA trunk type calls. The callers therefore receive the same call progress tones that internal callers hear and may have access to many extension features.

For the hardware that supports the Tie trunk type operation, see Trunk Support - E&M, Trunk Support - T1, and Trunk Support - Direct Inward Dial (DID).

See Dial Tone Disable for dial tone control for the Tie trunk.

Conditions

The following conditions apply to this feature:

- Tie trunks have access to the following extension features:
  -account code
  -ARS
  -console hold slot retrieve
  -directed pickup
  -hold retrieve
  -paging
  -system Abbreviated Dial.
- For Tie intercept handling, see DID/Dial-In/Tie Intercepts.
- Tie trunks can dial the DISA access code (the DISA access code used with DID and Tie trunks must not lead with an asterisk *, if ANI/DNIS is to be collected). After this access code, the Tie trunks get dial tone, and then can dial what their COS and COR allow.
- Tie trunks dialing extensions with Do Not Disturb enabled are handled in the same way as extensions.
- Tie trunk dialing follows the same rules as for extensions.
- Up to 7 digits can be absorbed and 2 digits can be prefixed.
- The prefix digits are used for matching access codes only after the Tie trunk has dialed at least one digit that is not absorbed. The prefix digits by themselves cannot be used to match an access code.
- The limit to the number of digits that a Tie trunk can dial is the same as for an internal extension (25). This includes the prefix digits but not the absorbed digits.
- The trunk may be programmed to ignore incoming DTMF digits, and to recognize only rotary digits.
- A DTMF receiver is needed when a Tie trunk originates. If there is no receiver available then the trunk waits indefinitely until a receiver is available. Digits received on the trunk are stored and processed when a receiver is available. The Traffic Measurement feature can be used to track the wait for a receiver.
- For incoming answer supervision when directed to ARS, see Tandem Operation.

Programming

To program a trunk to ignore incoming DTMF digits, enable COS Option 801 (Incoming Trunk Call Rotary) for the trunk.

Operation

None.

Trunk Recall

Description

The feature provides an alternate recall point for trunks in the system. The alternate recall point can be specified for each tenant and each NIGHT/DAY service. Under the following conditions, trunks are rerouted to the alternative call point:

- for all trunk types, when an extension with a trunk on consultation hold is listening to reorder tone and times out, the trunk is removed from consultation hold and rerouted.
- for DISA and CO trunks, when a trunk recalls from campon or ringing an extension; see Recall.

Conditions

The following conditions apply to this feature:

- This feature does not apply to DTS or private trunks ringing into the SUPERSET telephone where the line appears.
- For the reorder tone case, the reroute is done before recalling serial trunks.
- For the reorder tone case, the tenant group of the extension is used to determine the routing point.
- For the reorder tone case, if there is no point programmed for the current NIGHT/DAY service then the trunk is dropped; Serial trunks recall.
- An LDN key cannot be the routing point.
- For recall situations, the tenant of the called party is used to determine the rerouting point. When a logical line is called, the tenant of the first appearance of the line is used. When a hunt group is called, the tenant of the first member of the hunt group is used.
- The feature does not operate when calling an LDN, Night Bell, or console.

Programming

In CDE Form 19 (Call Rerouting Table), program the recall points for Non-Dial-In Trunk Alternate Recall Point in the tenant of the called party.

Operation

None.

Trunk Support - CLASS

Description

The system supports CLASS with the LS/CLASS Trunk card and with the LS/CLASS module.

The LS/CLASS Trunk card interfaces eight analog trunk circuits to the system. LS is the acronym for Loop Start and CLASS is the acronym for Custom Local Area Signaling Services (allows the system to receive calling Line ID digits and CLASS name on incoming CLASS trunks). The LS/CLASS Trunk card is a low power, peripheral interface card. It can be installed into slots one to eight in a SX-200 rack mount cabinet.The LS/CLASS Trunk card provides the circuit descriptors that is required by the card.

Conditions

The LS/CLASS Trunk card and the LS/CLASS module provide Loop Start not Ground Start.

Programming

Enter the trunk into the system's physical configuration table via CDE Form 01 (System Configuration). Define the LS/CLASS Trunk card as an "8 CCT CLASS". The LS/CLASS module is programmed as a "4 CCT CLASS".

Assign a separate COS in CDE Form 03 for the CLASS trunks. To have Calling Line ID digits in the SMDR enable COS Option 806 (SMDR - Record Incoming Calls) and COS Option 814 (SMDR - Record ANI/DNIS/CLASS). Enable COS Option 702, SMDR - Overwrite Buffer. To have CLASS name reported in the SMDR records, enable COS Option 246 (SMDR - Extended Record) and COS Option 814 (SMDR - Record ANI/DNIS/CLASS). If required enable COS Option 613, Display ANI Information Only.

Define the COS in CDE Form 03 for the Mitel telephones. To display the CLASS name before the Calling Line ID on Mitel 5020, 5220 and 5224 IP Phones, and SUPERSET 4025, SUPERSET 4125, or SUPERSET 420 telephones, enable COS Option 503, Display CLASS Name.

In CDE Form 13 (Trunk Circuit Descriptors) assign the CLASS trunk (8-CCT CLASS for the LS/CLASS Trunk card or 4-CCT CLASS for the LS/CLASS Module) with the circuit descriptors. The option, Post Call Metering, is for 8-CIRCUIT CLASS TRUNKS only.

Program the CLASS trunks in CDE Form 14 (Non Dial-in Trunks) or CDE Form 15 (Dial-in Trunks). In Form 15 the CLASS trunk becomes a DISA trunk.

Program the CLASS receivers for the LS/CLASS module in CDE Form 04. Programming CLASS receivers for the LS/CLASS Trunk card is not necessary.

Operation

None.

Trunk Support - CO (LS/GS)

Description

The system supports CO (LS/GS) trunks with the LS/GS trunk card.

For loop start CO trunks that do not provide release supervision, take caution when routing to auto-answer telephones or to UCD/ACD applications that do not provide termination for a music/recording sequence. In these cases, since the system does not disconnect the trunk, the call may stay up indefinitely.

Conditions

None.

Programming

Enter the trunk into the system's physical configuration table via CDE Form 01 (System Configuration).

Select options for the CO trunk type via CDE Form 13 (Trunk Circuit Descriptors).

Select options for the specific trunk circuits via CDE Form14 (Non-Dial-In Trunks) for incoming calls.

Refer to the Automatic Route Selection and Toll Control section, for the selection of options for outgoing calls.

Operation

None.

Trunk Support - Direct Inward Dial (DID)

Description

The following types of DID trunks are supported:

- Wink Start
- Delay Dial
- Immediate Dial.

DID trunks support Tie, CO, DID, and DISA operation.

Conditions

The following conditions apply to this feature:

- DID trunk cards are incoming only.
- The DISA access code used with DID and Tie trunks must not lead with an asterisk *, if ANI/DNIS is to be collected.

Programming

Enter the trunk into the system's physical configuration table via CDE Form 01 (System Configuration).

Select options for the DID trunk type via CDE Form 13 (Trunk Circuit Descriptors).

Select options for the specific trunk circuits via CDE Form 15 (Dial-In Trunks).

Operation

None.

Trunk Support - E&M

Description

E&M trunks are supported with the E&M Trunk module on the Universal Card.

The signaling schemes supported include: type I and type V, 2-wire or 4-wire.

E&M trunks support Tie, CO, DID, and DISA operation.

Conditions

None.

Programming

Enter the trunk into the system's physical configuration table via CDE Form 01 (System Configuration).

Select options for the E&M trunk type via CDE Form 13 (Trunk Circuit Descriptors).

Select options for the specific trunk circuits via CDE Forms 14 (Non-Dial-In Trunks) and CDE Form 15 (Dial-In Trunks).

Operation

None.

Trunk Support - IP

Description

IP trunks carry voice and signaling messages between networked systems. All IP trunk calls are routed through the Ethernet Switch to the WAN. The SX-200 ICP CX/CXi supports up to 16 and the MX /AX supports up to 30 IP trunks.

Conditions

- Campon is not supported over PRI/IP trunks programed as Tie.
- Fax and modem calls routed over IP trunks fail to negotiate if route compression is enabled on the IP trunk route in Form 23, Subform Show IP.
- IP trunks are programmed as rotary dial trunks in Form 13, which limits dialing to digits 0-9; # and * are not supported.
- The SX-200 ICP can function as an end node only in a network, which means that it can be connected to only one other network node.
- Centralized voice mail over IP trunks requires System Option 113 (Centralized Attendant / Voice mail) in Form 4.

Programming

See Network Mitel IP-PBXs

Operation

None

Trunk Support - T1

Description

T1 trunks are supported using T1/D4 channel associated signaling (CAS), also referred to as DS-1. The frame formatting—Superframe or Extended Superframe (ESF)—supported is different for T1 trunk cards and T1 modules: the cards support Superframe only; whereas the modules, which reside in the MMC slots and on the BCCIII card, support ESF.

T1 trunk cards support DID, tie, CO and DISA operation on a per circuit basis. For each circuit, the circuit descriptor can be programmed through CDE to alter the signaling scheme to one of E&M, DID loop-tie, CO (loop and ground start), DISA E&M, DISA DID loop-tie, DISA CO (loop or ground start).

The system provides for a Stratum 3 or Stratum 4 source. The system can be used in master mode to serve as a clock source for the network, or in slave mode to use the network as its clock source. In slave mode the system prevents data losses due to clock rate differences by adjusting its internal T1 clock module to remain in phase with the incoming frame clock rate.

T1 Trunk Maintenance

The system automatically monitors various errors on a link and maintains a 24-hour sliding window for each link. Thresholds are programmable on a link basis for these errors. When the thresholds are exceeded, based upon the threshold type (maintenance, service, or network synchronization), the system generates maintenance logs, removes the link from service, or causes the link to not be used as a network synchronization source. Once the link is removed from service or is removed from network synchronization, a programmable timer starts. When this time is elapsed, if the link does not exceed a specified number of errors, the link is again made available for use. All of this activity is logged in maintenance and the cumulative error counts (24-hour basis) are available in maintenance error reports (refer to the RS-232 Maintenance Terminal section).

There are two different states for a T1 trunk that is out of service, a yellow or red alarm. A yellow alarm is the result of a link-based event. The link transmits a yellow alarm condition on the link, which signals the far end that the link is out of service. This is the action taken on the service threshold limits or the reception of a yellow alarm condition from the far end.

A red alarm is based upon loss of synchronization on the link. Maintenance logs are generated for any change in alarm status, and can be seen by using the maintenance SHOW STATUS command.

A link monitoring tool is available for troubleshooting T1 links, and for specifying a manual network synchronization source. Refer to the, RS-232 Maintenance Terminal section.

T1 Trunk Synchronization

The system provides for Stratum 3 source, and can be used in master or slave mode. An optional Stratum 4 source is also available. There is a programmable list of links to be used in network synchronization, with each link backing up the other links, should the previous link exceed the error thresholds (Slip Rates, Bit Error Rates (BER), and Framing Losses) programmed in the link descriptors. There are three network synchronization modes in the system: Auto, Manual, and Freerun. Auto mode is when the system chooses the network synchronization source automatically. Manual mode is when a link is specifically selected as the network synchronization source (this only lasts for 24 hours, at which point the system reverts to auto or freerun mode). Freerun mode is when there is no clock source or no link to be used as a network synchronization source. The system changes modes based on user actions, thresholds, and events on the T1 links. Maintenance logs notify the installer of these changes.

Conditions

The following conditions apply to this feature:

- There are 12 link descriptors available.
- There can be up to 8 network synchronization sources for network synchronization.
- One link provides 24 trunks.
- If a link descriptor is not assigned (it is recommended that one is assigned) then the default values in the link descriptors will apply. A link cannot be a network synchronization source without a link descriptor.
- If the last available network synchronization source specified is removed from service then the system is put into free-run mode.
- When a network source link is returned to service or is again ready for use as a network synchronization source, then the system will review the list of network synchronization sources and possibly select the returned link (depending upon the current network synchronization source and the order of the links in the list).
- A link is removed from service immediately, regardless of activity on the link, when its service threshold limits are exceeded.
- The T1 Link Monitor feature is not available in maintenance until at least one circuit is programmed (CDE forms 14 or 15) on a link in the system.
- The T1 Clock Module clock rate is adjusted by the system if there are entries made in CDE Form 44 (Network Synchronization) and one of the links there is available. If no network synchronization source is available, then the clock rate will not be adjusted by the system.
- The threshold error counts are cleared when the T1 card is plugged in or any of the values in the link descriptor assigned to the link are changed in CDE Form 42 (T1 Link Descriptors). A change in link status has no effect.
- Changing the entries in CDE Form 44 (T1 Network Synchronization) will override manual mode and change the PBX mode to freerun or auto.
- The term link is synonymous with a T1 trunk card.
- The normal maintenance busy-out feature applies to a circuit on a link and not the link itself. The link remains active but the circuit(s) are disabled.

- If the system contains two Dual T1/E1 Framer modules (one in each MMC slot), they must both be assigned the same bay number in Form 53.

Programming

For the onboard Dual T1/E1 Framer module(s), assign a bay number in Form 53 (Bay Location).

Enter the trunk into the system's physical configuration table in Form 01 (System Configuration).

Enable COS Option 802, Limited Wait for Dial Tone in Form 03 (COS Define).

In Form 04 (System Options and Timers), program option 48, Limited Wait for Dial Tone, and option 96, Number of Links (0-8). The recommended value for option 48 is one second.

Specify options for the T1 trunk types in Form 13 (Trunk Circuit Descriptors).

Specify options for specific trunk circuits in Forms 14 (Non-dial-in Trunks) and 15 (Dial-in Trunks).

Specify options for the T1 link descriptors in Form 42 (T1 Link Descriptors).

Note: CDE Form 14 and 15 do not accept trunk programming changes if the T1 card is installed and the T1 link is down but still considered active. To make a programming change for a T1 card, unplug the T1 card, make the programming change and then re-insert the T1 card .

Assign link descriptors to the various T1 links via CDE Form 43 (T1 Link Assignment).

If the T1 clock module is used, assign the primary and backup T1 links to be used as sources for network synchronization via CDE Form 44 (T1 Network Synchronization). The order that the links are programmed in the form gives the order in which the links are used as sources.

Operation

None.

Unified Messaging

Description

The Unified Messaging feature package enables the SX-200 ICP to manage e-mail messages using SMTP (Simple Mail Transfer Protocol) and/or IMAP (Internet Message Access Protocol). SMTP is a standard component of the feature package; IMAP requires the purchase of a Mitel Managed Application Server with the Unified Messaging Blade enabled. Mitel's implementation of IMAP is known as Standard Unified Messaging.

IMAP support (Standard Unified Messaging)

When it is programmed to support Standard Unified Messaging, the SX-200 allows users to manage their voice messages with one or more IMAP-enabled e-mail clients such as MS Outlook or Outlook Express. With IMAP, users do not download their entire mailbox of pending messages. Instead, they request only the messages they need—either as complete e-mails or as headers. And each message remains on the server until it is deleted. This allows users to download copies of the original message to multiple e-mail clients, and to synchronize message status throughout the system. For example, when a user opens an e-mail and listens to a voice message, the message status is updated in the local e-mail client. Then, when message synchronization occurs, the message status is updated in the user's mailbox, in the IMAP server, and in any other e-mail clients that have received a copy of the message. Synchronization also ensures that the message waiting indictor on the user's telephone always reflects the actual message status for the mailbox.

Conditions

The following conditions apply to this feature:

- FOR Option 126, Email Messaging.
- FOR Option 87, Record a Call (if used).
- Embedded voice mail for forwarding of voice mail to e-mail.
- Feature requires Advanced Certification to implement.
- For IMAP, each user needs an account on the Unified Messaging blade installed in the 6000 MAS. Also, the 6000 MAS must be programmed to interact with the SX-200 ICP.

Programming IMAP (Standard Unified Messaging)

Enabling IMAP support

1. Form 04, System Options/System Timers
   - Enable System Option 126, Email Messaging.
   - for Option 81, Enter offset from GMT (+/-hh:mm), enter the difference in hours and minutes between the time zone that the SX-200 ICP is in and Greenwich Mean Time (GMT). The offset, which appears in the timestamp on the 911 Notification email, is important for recipient(s) to know in case the email originated from another time zone.
     Note: Manual time changes (as opposed to automatic changes that the system makes to adjust for daylight savings) will require a manual update to the GMT offset.
2. Form 49, Voice Mail Options
   - Enter the IP address of the IMAP server.

Programming voice mail to e-mail

1. Complete the IMAP programming above.
2. Form 50, Mailboxes
   - In the Email subform, select a mailbox in the FWD column, then press the UNIFIED FWD softkey.

Notes:

1. When a voice message is first received, its status is "new" in the user's voice mail mailbox. The status remains new, and the user's message waiting status indicator continues to flash, even after the mailbox is synchronized with the IMAP server and clients. When the user listens to or opens the message, the status changes to "read" in IMAP and "saved" in the mailbox. If there are no other new messages, the message waiting indicator stops flashing on the user's telephone. When the user deletes the message, the status changes to "marked" in IMAP and "deleted" in the mailbox. When the "marked" message is purged from IMAP, the status changes to "expunged" in the IMAP server and the message is erased from all IMAP clients. IMAP clients must periodically synchronize themselves with the server to share message status information. The IMAP server and the voice mail system are synchronized automatically every 30 seconds.
2. To reassign a mailbox:
   - Disable e-mail forwarding for the mailbox in the Email subform for Form 50.
   - Assign the mailbox to a new user account in the MAS. To preserve existing messages, copy the contents of the voice mail folder to the new user account.
   - Enable e-mail forwarding for the mailbox in the Email subform for Form 50.
3. For an internal call that is forwarded to email, the subject and from fields display the extension number and name of the caller as programmed in Form 50. For an external call that is forwarded to email, the subject and from fields display "unknown caller" information.

Programming the Managed Applications Server

In the Managed Applications Server (MAS) Server Manager web interface:

1. On the Standard Unified Messaging panel
   - enable the application
   - enter the IP address of the SX-200 ICP

- enter the system ID (from Form 04, Option 101) of the SX-200 ICP

2. On the Users panel, select Add user account and complete the form. Make sure the extension number matches the mailbox number programmed in the SX-200 ICP, and the password matches the value programmed in the user's e-mail client. Repeat for each new user.

Note: When adding a user account to the MAS, enter the extension number in the last portion of the Phone number field. See the MAS documentation for details.

Example voice mail synchronized using IMAP

From: 106 <voicemail@do-not-reply>
Sent: Tuesday, April 13, 2004 10:03 AM
To: 107
Subject: Voicemail message from 106 <ID: 1081872898>

This e-mail contains a voice message.
Double click on the attached file to listen.

From line contains the identity of the message sender.

- Format: <name-mailbox | mailbox | CLID | unknown caller> <voicemail@do-not reply>

Examples:

- TOM-106 <voicemail@do-not-reply>
- 106 <voicemail@do-not-reply>
- 5994535 <voicemail@do-not-reply>
- unknown caller <voicemail@do-not-reply>

Sent line contains the time and date that the voice message arrived in the user's mailbox.

To line contains the destination mailbox number.

Subject line contains the message type, the identity of the call originator, and a unique call ID.

- Format: <message-type> from <name-mailbox | mailbox | CLID | unknown caller> <ID: uniqueID>

Examples:

- Voicemail message from TOM-106 <ID: 3935415736>
- Record Call from 106 <ID: 3935415737>
- Voicemail message from 5994535 <ID: 3935415738>
- Voicemail message from unknown caller <ID: 3935415739>

The subject and body will be in the language of the message recipient's voice mailbox.

Operation

Refer to Voice Mail User Guides.

Uniform Call Distribution (UCD)

Description

Uniform Call Distribution (UCD) concentrates incoming trunk traffic onto one or more special agent hunt groups. Trained operators (agents) answer the calls. If all agents are busy, the caller camps on and may be connected to a recording hunt group, where the caller hears recorded announcements. The caller retains his position in the queue. If the agents are still busy when the recording ends, the system connects the call to Music-on-Hold (if provided). After a pre-determined time, the unanswered call is rerouted to a designated answering point.

The call rings an agent immediately when one is available.

Agents can log in and out of the agent hunt group to control the arrival of calls from the agent hunt group.

When a caller is connected to a Recorded Announcement Device (RAD), full two-way audio is provided. This allows the caller to leave a message with the RAD device (if desired), while waiting for the UCD group.

See RAD Support for details on recording devices.

For night service handling, trunk calls can be routed to a recording device directly; see Intercept To Recorded Announcement.

Feature Availability

Click here to see a list of the phones that can use this feature.

Conditions

The following conditions apply to this feature:

- If there is no hold timeout point and recordings are given, then for loop start trunks that do not provide release supervision, the system does not disconnect the trunk; the trunk remains on Music-on-Hold. It is important to provide a routing point for these unanswered calls to ensure that the loop start trunk is either answered, or timed out for no answer and then released.
- Agent hunt groups are special hunt groups; hunt group conditions plus some additional features apply to them.
- Circular hunting is advised for the agent hunt group to distribute the calls in the group. There is no distribution of calls based on workload or waiting time; see Automatic Call Distribution (ACD).
- Campon to agent hunt groups follows the normal campon to hunt groups. The recording and reroute features are available once a caller has camped on to the agent hunt group.
- Features available for hunt groups, such as Trunk Campon Warning Tone and Swap Campon, are available to agent hunt groups.
- The recording hunt group is selected from the tenant of the first programmed member of the hunt group.
- If there is no recording programmed, then no music is provided; it is only provided after the recording.
- The system provides a recording for the caller as soon as a recording device is available.
- At least one hunt of the agent hunt group is done before the waiting timeout timer is started (regardless of the waiting timeout length).
- If the waiting timeout time is set to 0 seconds then the timeout effectively is an overflow busy point for the agent hunt group.
- A reroute is done for the waiting timeout.
- If the caller is listening to a recording at the time of the waiting timeout, the recording is terminated before the reroute is done. If the waiting timeout occurs while listening to music and if a campon is done on the reroute point then music is maintained while waiting for the busy reroute point. Otherwise it is removed and normal call progress tones provided.
- The waiting timeout is ignored if the caller is ringing an agent at the time.
- If the waiting timeout occurs while ringing a recording device then the ringing recording device stops ringing and the caller is rerouted to the timeout point.
- By default all agents are logged into an agent hunt group.
- Logging out of an agent hunt group only prevents calls from being presented to the agent set from the agent hunt group. Other features are not affected.
- An agent that is logged out will not be selected to receive campon beeps and the swap campon capability for the hunt group; see Swap Campon And Campon Warning Tone.

- When an agent becomes available and is rung by a waiting caller, the recording or music is removed and the caller hears ringback tone.
- UCD agent hunt groups and ACD agent groups are completely separate and different entities. The first is a collection of devices, and the second is a collection of agent id codes.
- The campon warning tone can be turned off for members of the agent hunt group; see Campon Warning Tone.

Programming

Enter a series of extension numbers into a hunt group in CDE Form 17 (Hunt Groups). Press the GROUP TYPE softkey, followed by the AGENT softkey.

Program a recording hunt group in CDE Form 17 (Hunt Groups); see RAD Support.

Enter the access code of the recording device hunt group into CDE Form 19 (Call Rerouting Table), under “UCD Recording Routing For This Tenant”. The tenant number is that of the first programmed member of the agent hunt group.

Enter an On Hold Time-out period via COS Option 256, UCD Music-on-Hold Timer, for each incoming trunk.

Enter an On Hold Time-out answering point extension into CDE Form 19 (Call Rerouting Table), under “UCD Time-Out Routing For This Tenant”. The tenant number is that of the first programmed member of the agent hunt group.

Assign an access code to Feature 38 (UCD Login/Logout Code) in CDE Form 02 (Feature Access Codes).

Enable COS Option 301 (Campon) for the incoming trunk to camp onto the UCD group.

Operation

To log in at an agent set:

- Dial the UCD Login/Logout Code, followed by 1.

To log out at an agent set:

- Dial the UCD Login/Logout Code, followed by 2.

Vacant Number Intercept

Description

Calls to unassigned (vacant) access codes can be routed to a given answering point for completion. This point can be an LDN position on the Attendant Console (see Console LDN Keys) or any valid routing point. Vacant number intercept points can be programmed to be different or the same for DAY, NIGHT1, and NIGHT2 modes of system operation.

See DID/Dial-In/Tie Intercepts for the same feature for DID and Tie trunks.

Conditions

The following conditions apply to this feature:

- If the required programming is not done, such calls receive reorder tone.
- Telephones, DISA trunks, and CO trunks only are routed to the answerpoint.
- See DID/Dial-In/Tie Intercepts for vacant number handling for DID and Tie trunks.
- If the call is routed to a console, the call is shown as an intercept call at the console.

Programming

To cause all calls to vacant numbers to be routed to a specific answering point, program CDE Form 19 (Call Rerouting Table) with the desired answering point access code, in the appropriate column for the "Station Vacant Number Routing For This Tenant" call type.

Operation

None.

Voice Mail Support

The SX-200 ICP includes an embedded voice mail application. The system also supports the following voice mail functionality:

- Centralized Voice mail
- Voice mail on DNIC Ports
- Voice mail on ONS Ports
- Mitel Express Messenger
- Single Button Transfer to Voice mail
- Softkey Support for embedded voice mail and Mitel NuPoint Messenger

Embedded Voice Mail

Description

See the SX-200 ICP General Information Guide available on the Mitel Edocs website (http://edocs.Mitel.com)

Conditions

None

Programming

See CDE Programming Overview for Embedded Voice Mail.

Operation

Refer to Voice Mail User Guides

Centralized Voice mail

Description

Centralized voice mail allows one voice mail device to service several networked SX-200 ICP systems.

New information elements are sent across the network to provide the centralized Voice mail system with call forward data and call forward reason.

Voice mail users on a remote SX-200 ICP can retrieve their messages either by the "message" key on the set or by dialing the "Call Message Sender of Oldest Message" feature access code. The caller information is passed over the network to the Voice mail system so that the caller is routed to the correct mailbag. This is done by digits programmed in a system abbreviated dial index and via ARS modified digit strings.

Conditions

Refer to Centralized Voice mail section.

Programming

Refer to Centralized Voice mail section.

Operation

Refer to Centralized Voice mail section.

Voice mail on DNIC Ports

Description

Voice mail devices may use the Mitel DNIC interface.

Conditions

For the message waiting lamp to function, check the configuration report of mail system and ensure that the feature access code for send message matches the feature access code programmed in the PBX for send message (feature access code 41).

Note: The SX-200 ICP feature access code requires an additional 1 for turning the message waiting lamp on or a 2 for turning the lamp off.

Integration with Mitel NuPoint Messenger requires key 3 to be programmed as a DND key (CDE Form 9, Expand Set Subform).

Programming

Program the Voice mail Port(s) with the following Class of Service Options:

COS Option 212 - Can Flash if Talking to an Incoming Trunk

COS Option 213 - Can Flash if Talking to an Outgoing Trunk

COS Option 216 - Data Security

COS Option 229 - Voice mail Port

COS Option 238 - Override Security

COS Option 259 - Message Sending

COS Option 301 -Camp-on

COS Option 502 - Display ANI/DNIS/CLASS Information

COS Option 508 - Station/Set: Show Internal Number on My Phone

COS Option 604 - Telephone-Automatic Outgoing Line

COS Option 606 - Telephone-Enhanced Answering Position

COS Option 609 - Telephone - Night Service Switching

Note: Disable all other options including Call Forwarding

Define Hunt Group in CDE Form 17 (Hunt Groups) by programming the following fields:

EXT NUM Enter the extension number of each voice mail port that is a member of the voice mail hunt group.

ACCESS CODE The access code is the system access number for the VX Voice Processing system. Define the access code of the voice mail hunt group by using the ACCESS CODE softkey (softkey 7).

NAME Enter the name of the voice mail hunt group. Enter a name for the voice mail hunt group by using the OPTIONS softkey (softkey 4).

Note: Do not program an overflow destination for this hunt group. This will ensure that calls can camp on to the busy group.

Operation

Refer to the Voice mail system documents.

Voice mail on ONS Ports

Description

This feature integrates an SX-200 ICP system with an ONS Voice mail system. The integration is based on the use of system abbreviated dial numbers. This eliminates several dialing steps involved in the sending and retrieving of voice mail messages.

Special codes allow the system operation to be customized to suit the operation of the particular voice mail system. Refer to the programming instructions for Form 31 - System Abbreviated Dial Entry, for a description of these special codes.

Conditions

The following conditions apply to this feature:

- This feature does not conflict with other ONS voice mail systems - they may be installed simultaneously.
- The *1, *6, *9, codes are only operational with the ONS Voice mail feature. If the abbreviated dial number dials anything but an ONS voice mail hunt group, the codes are ignored.
- The *4 and *8 codes are used with the NuPoint Messenger Enhanced Inband Signaling.
- Callers hear ringback while the digits are pulsed to the voice mail system. Audio is not cut through until after the last pause has been executed.

Programming

Form 01 - System Configuration

Program the ONS/CLASS Line Card.

Form 02 - Feature Access Codes

Assign access codes to:

- Feature 24 - Abbreviated Dial Access
- Feature 41 - Send Message

When appropriate for use with *4, assign access codes to:

- Feature 3 - Call forwarding - All Calls
- Feature 4 - Call forwarding - Internal Only
- Feature 5 - Call forwarding - External Only

Form 03 - COS Define

In ONS voice mail port COS, enable

- COS Option 208 - Call Forwarding - External
- COS Option 212 - Can Flash if Talking to an Incoming Trunk
- COS Option 213 - Can Flash if Talking to an Outgoing Trunk
- COS Option 216 - Data Security
- COS Option 229 - Voice mail Port
- COS Option 238 - Override Security
- COS Option 261 - ONS Voice mail Port
- COS Option 301 - Camp-On

In Message Waiting/Pager Port COS, enable:

- COS Option 216 - Data Security
- COS Option 220 - Do Not Disturb
- COS Option 235 - Originate Only
- COS Option 238 - Override Security
- COS Option 259 - Message Sending
- COS Option 265 - Voice mail system Speed Dial Index

Note: Include option 265 in the COS of the first member of the hunt group for Messaging - Call Me Back feature.

In the COS of the mailbox holder, enable:

- COS Option 231 - Message Waiting Setup - Bell, or
- COS Option 232 - Message Waiting Setup - Lamp
- COS Option 264 - Half Fwd NA timer for DID call when VM msg on (This is a toll saving option, allowing external DID callers who are calling in to check their messages to hang up if the cfna timer is not shortened, knowing they have no messages to retrieve.)

Form 04 - System Options/System Timers

Enable

- System Option 21 - Incoming to Outgoing Call Forwarding
- System Option 22 - Last Party Clear Dial Tone

Form 09 - (Desktop Device Assignments).

Program ONS lines and assign appropriate COS.

Ensure Voice mail Ports are in a separate COS from the Message Waiting Ports.

Form 14 - Non-Dial-In Trunks

If required, enter the Voice mail port hunt group access code for DAY, Night1 and/or Night2 for the trunk to go to the Receptionist portion of the VM system.

Form 17 - Hunt Groups

Program a hunt group for the Message Waiting port.

Program a hunt group containing the ONS voice mail ports.

Note: The Message Waiting Hunt Group is NOT the same as the ONS Hunt Group.

Form 19 - Call Rerouting Table

For Message Port Tenant

- DND Intercept Routing for this tenant for Day, Night1, and Night2 will be routed to the Voice mail hunt group access code.
- Station Illegal Number Routing for this tenant for Day, Night1, and Night2 will be routed to the Voice mail hunt group access code.

To program call forwarding

- Call Forward Busy Number For This Tenant: press Esc-1 to select SPEED DIAL and enter the index number from Form 31 that contains the forwarding string.
- Call forward No Answer Number For This Tenant: press Esc-1 to select SPEED DIAL and enter the index number from Form 31 that contains the forwarding string.

Form 31 - System Abbreviated Dial Entry

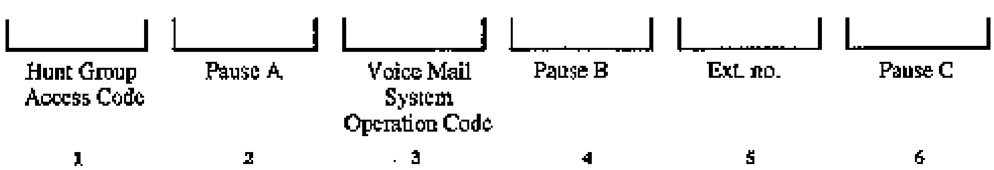
Special codes entered in this form allow the operation to be customized to suit the operation of the particular voice mail system. These special codes eliminate many of the dialing steps involved in the sending and retrieving of voice mail messages. The integration of the PBX and voice mail system is based on the abbreviated dial numbers shown below.

Code	Description
*3 XX	Insert manual dialed digits (XX) 2 digits expected
*4	Send call forward condition to Voice mail (See Note)
*6	Tone out caller extension number
*8	Send digits of calling party to Voice mail (See Note)
*9	1 second pause
**	DTMF digit *
#	DTMF digit #
0-9	DTMF digits 0 - 9

Note: *4 and *8 are used with NuPoint Messenger Enhanced Inband Signaling.

Event Timing

The following example describes the feature's timing requirements:



1: Access code of the voice mail port hunt group.
2: Length of time required for the voice mail system to prepare to receive an optional operation code after answering the call.

3: Digit(s) indicating the type of operation to be performed voice mail system (record or playback message).
4: Length of time required for the voice mail system to receive a mail box number, after receiving the operation code.
5: Usually *6: For a forwarded call, *6 translates into the originally called party's extension number. For non-forwarded calls, *6 translates into the caller's extension number.

Set up the desired call forwarding for the telephones using this voice mail feature.

- Place all ONS voice mail ports in the same hunt group in Form 17 (Hunt Groups) - assign an access code to the group.
- If the ONS voice mail system requires dial tone on hang up to release the port, enable System Option 22 (Last Party Clear - Dial Tone).
- If trunk calls are to reach the ONS voice mail system via a forwarding system abbreviated dial number, enable System Option 21 (Incoming to Outgoing Call Forward) and COS Option 208 in the trunk COS.

For Message Forward:

- Enter an index number for message forward in the INDEX NUMBER field of Form 31 - System Abbreviated Dial Entry
- Enter the following in the DIGIT STRING field:
  - the ONS voice mail hunt group access code,
  - any number of the *9 codes combining to result in an answer pause suitable to the voice mail system,
  - the voice mail system access code for the type of call forwarding which corresponds to the mailbox owner's call forwarding,
  - *6
  - the shortest *1 / *9 combination to ensure that the ONS voice mail port receives all of the digits.

For Message Retrieve:

- Enter an index number for message retrieve in the INDEX NUMBER field of Form 31 - System Abbreviated Dial Entry
- Enter the following in the DIGIT STRING field:
  - the ONS Voice mail hunt group access code,
  - any number of the *9 codes combining to result in an answer pause suitable to the voice mail system,
  - the voice mail system access code for message retrieval,
  - *6
  - the shortest *1 / *9 combination to ensure that the ONS voice mail port receives all of the digits.

Operation

Forward message

- Dial extension number and leave message.

Retrieve message

- Dial the system abbreviated dial access code and the message retrieve abbreviated dial index number.

The SX-200 ICP dials out feature access code (3, 4, or 5) plus the index number.

Valid index numbers are

1 - Forward Always
2 - Forward on Busy
3 - Forward No Answer
4 - Forward Busy/No Answer

The SX-200 ICP dials out the ANI digits if the call originated over a trunk. If ANI digits are not allowed or not available, then no digits are sent. When the call is internal, the extension's access code is dialed out.

- To retrieve a message from an ONS set that is programmed for Direct to ARS and has been programmed to access an internal dial tone with ARS leading digits
  - dial the programmed access code (e.g. *59).
    You hear internal dial tone.
  - dial the voice mail extension number.

Mitel Express Messenger

See Program Feature Packages - Mitel Express Messenger.

Whisper Announce

Description

Allows a party to place a directed page to a busy Family 1, Family 4, Family 5, or 5207 telephone. A short burst of ringing precedes the voice announcement, advising the busy party that an announcement is following. The announcement is heard through the handset or headset, only by the paged party. The other party will hear silence.

After initiating a Whisper Announce, if the paged party has COS Option 501, Override Announce, enabled, the paging party hears a short burst of ring-back tone and can talk immediately after the burst of ringing.

After initiating a Whisper Announce, if the paged party has COS Option 501, Override Announce, disabled, the paging party hears a short burst of busy tone and must wait for the paged party to respond.

Feature Availability

Click here to see a list of the phones that can use this feature.

Condition

The following conditions apply to this feature:

- A whisper announce can only be made to Family 1, Family 4 , Family 5 (except the SUPERSET 4DN and the Symbol MiNET Wireless) and 5207 telephones.
- A whisper announce can only be made to a telephone that is in the talking state, and is not in handsfree mode.
- A whisper announce can be made from a SUPERCONSOLE 1000 or a 5540 IP Console.
- A whisper announce cannot be made to a set that has DND enabled.
- A whisper announce cannot be made to a set that is involved in a conference.
- A whisper announce cannot be made to a set that has a call on hold, or that is talking to a party that has a call on hold.

Programming

Enable COS Option 501 (Override Announce) and disable COS Option 615 (Off-Hook Voice Announce) in the paged party's COS.

Program the Direct Paging feature access code (48) in Form 02, or program a Direct Paging feature key in the Expand Set Subform of CDE Form 09 (Desktop Device Assignments).

Program a RESPOND feature key to 5207 telephones in the Expand Set Subform of CDE Form 09 (Desktop Device Assignments).

Operation

To Initiate an Announce:

Family 3, TeleMatrix 3000IP, and SUPERSET 410 Telephones:

- Dial the Direct Paging feature access code, or press the DIRECT PAGE feature key.
- Dial the extension number, or press the DSS key of the party to be paged.
- Paging party gets a burst of ringback tone if the paged party has COS Option 501 enabled and can then speak and be heard.
- Paging party gets a burst of ringback tone if the paged party has COS Option 501 disabled and must wait for the paged party to respond.
- Paged party always gets a burst of ringback tone and its display changes to indicate someone is trying to page and the RESPOND key appears.
- If the paged party has COS Option 501 enabled, you may make your announcement immediately following the burst of ringback tone. If the COS is not enabled, you will hear a burst of busy tone and must wait for the paged party to respond.

Family 1, Family 4, Family 5 and 5207:

- Dial the Direct Paging feature access code, or press the DIRECT PAGE feature key.
- Dial the extension number, or press the DSS key of the party to be paged.
- If the paged party has COS option Override Announce enabled, you may make your announcement immediately following the burst of ringback tone. If the COS is not enabled, you will hear a burst of busy tone and must wait for the paged party to respond.

SUPERCONSOLE 1000:

- Dial the Direct Paging feature access code, or press the SET PAGE key.
- Dial the extension number, or press the DSS key of the party to be paged.
- If the paged party has COS option Override Announce enabled, you may make your announcement immediately following the burst of ringback tone. If the COS is not enabled, you will hear a burst of busy tone and must wait for the paged party to respond.

To Answer an Off-hook Voice Announce:

Family 1, Family 4, Family 5 or 5207 telephones which have COS option Override Announce enabled:

- You will hear a short burst of ringback tone, then the paging party's announcement.
- Press and hold the RESPOND softkey (or feature key on the 5207) if you wish to respond to the paging party. Your existing caller will be placed on hold.
- Release the RESPOND softkey (or feature key on the 5207) to end the connection with the paging caller and return to your held call.

Family 1, Family 4, Family 5 or 5207 telephones without the COS option Override Announce enabled:

- You will hear a short burst of ringback tone, but the paging party cannot be heard until you press and hold the RESPOND softkey (or feature key on the 5207).
- Your existing caller is placed on hold, and you are connected to the paging party.
- Release the RESPOND softkey (or feature key on the 5207) to end the connection with the paging caller and return to your held call.

Program Embedded Voice Mail

Introduction

This section describes the embedded voice mail features of the SX-200 ICP.

This section provides

- a description of the voice mail features
- programming requirements
- testing and troubleshooting information

Setup and maintenance are done primarily through CDE. The only setup work that cannot be done through CDE is the recording of auto attendant greetings. Recording is done by telephone through the System Administration Mailbox (number 999 by default). Some maintenance (such as editing mailbox settings) can also be done via the Administrator Mailbox. For more information, see Programming via the Administrator's Mailbox.

Additional Documentation

Refer to the following documentation for additional information about using voice mail:

- User Guide - describes all activities that are performed by users to set up, use, or change their mailbox. This guide is provided as a PDF file on the e-Docs website (http:\\edocs.mitel.com).
- Wallet Card - describes activities that are frequently performed by a mailbox user. It includes a record of dialing codes and access numbers. The Wallet Card can be ordered from Mitel Customer Service. A PDF version is also available on the e-Docs website.
- Front Desk User Guide and Guest Messaging Guide - in hotel or motel applications, the front desk attendant should have a front desk user guide. This guide provides instructions on how to administer guest mailboxes. The Guest Messaging Guide explains to guests how to use voice mail. Both guides are available on the e-Docs website. The Guest Messaging guide can also be ordered by calling Customer Service

All of the above documents are available in English, French, and Spanish.

Note: Information that is provided in the above documents is not repeated within this section.

About Mailbox Types

The voice mail system uses different types of mailboxes. Some are reserved for system functions and others are available for general use. Each mailbox type has different characteristics and uses.

Reserved Mailboxes

The Operator

Mailbox 0 is reserved for the Operator's mailbox. Callers will leave messages in this mailbox if they,

- do not know who to contact,
- are calling from a rotary dial telephone and cannot access menus or mailboxes.

Check for messages left in the Operator's mailbox several times each day. The default passcode for the operator's mailbox is 1234. If the passcode length has been changed, omit the last digit or add the appropriate number of zeros to the end of the passcode; for example, 123400.

The Directory (Phonebook)

Mailbox 9 is used for the personnel directory (also called the Phonebook). It cannot take messages. Callers use it to reach an employee by dialing the first six letters of the employee's first or last name (as defined). To prevent confusion between dialing a mailbox that starts with 9 and dialing 9 for the Directory, an inter-digit timer of 4 seconds is started when the first digit entered is 9. If no other digits are entered during these four seconds, the user is transferred to the Directory.

Administrative Functions

Mailboxes 90 through 97 or 990 through 997 or 9990 through 9997 are reserved for self-administrative functions.

The System Administrator's Mailbox

Mailboxes 99, 999, or 9999 (depending upon mailbox length) is reserved for the system administrator. It is not associated with an extension and does not take messages. Any touch-tone telephone in your company can access this mailbox. The default passcode is 8642. If the passcode length has been changed, omit the last digit or add the appropriate number of zeros to the end of the passcode; for example, 86420. To maintain system security, change this passcode at first login.

Note: To prevent confusion between dialing a mailbox that starts with 9 and dialing 9 for the Directory feature, an inter-digit timer of 4 seconds is started when the first digit entered is 9. If no other digits are entered during these four seconds, the user is transferred to the Directory

General Usage Mailboxes

There are four types of general usage mailboxes:

- Extension
- Message-Only
- Transfer-Only
- Information-Only

Note: The Auto Attendant Park and Page mailbox option is present in the TUI Administrator but these mailbox types are not supported in this release.

Each type serves a different purpose and has its own characteristics.

Extension Mailbox

An extension mailbox is the default mailbox. For hotel/motel installations, the default mailbox type is Guest. The mailbox number is usually the same as the subscriber's extension number. The default passcode for extension and guest mailboxes is 1111. If the passcode length has been changed, omit the last digit or add the appropriate number of ones to the end of the passcode; for example, 11111.

The call processing sequence for an Extension mailbox is a follows:

1. An outside caller calls the company telephone number.
2. The system answers the call and plays the company greeting.
3. The caller enters and extension number on the keypad (or uses the Directory to call by name).
4. The extension rings.
5. If the line is busy or unanswered, the system asks the caller if they wish to leave a message.

6. The system records a message in the associated mailbox.
7. The system turns on the telephone message light (if available).

For telephones without message lights, you can set up message notification to the subscriber's extension number. See Message Notification.

Message-Only Mailbox

A message-only mailbox takes messages and turns on the message light of the associated extension. Unlike an extension mailbox, the phone does not ring after the caller enters the extension number. Instead, the system asks the caller to leave a message.

A message-only mailbox does not require an associated extension. In this case, the system cannot turn on a message light; instead, the mailbox owner must periodically check for messages. Message-only mailboxes are for people who do not have telephones or who are not in the office very often.

Examples:

Out-of-town sales representatives
Delivery drivers
Part-time and seasonal employees
Frequent customers.

The call processing sequence for a Message-Only mailbox is a follows:

1. An outside caller calls the company telephone number.
2. The system answers the call and plays the company greeting.
3. The caller enters and extension number on the keypad.
4. The caller bypasses the telephone and goes directly to the voice mail system.
5. The caller is asked if they wish to leave a message.
6. The system records a message in the associated mailbox.
7. The system turns on the telephone message light (if available).

For telephones without message lights, you can set up message notification to the subscriber's extension number. See Message Notification.

Transfer-Only Mailbox

A transfer-only mailbox is associated with an extension but does not take messages. When a caller dials this extension he will hear, "You are being transferred to <mailbox name>." If the extension is busy or unanswered, the system returns the caller to the company greeting. No message is taken.

Add transfer-only mailboxes for areas in your company where you need a telephone but don't want messages taken.

Examples:

Conference rooms
FAX machines
Modem hookups
Lab areas.

The call processing sequence for a Transfer-Only mailbox is a follows:

1. An outside caller calls the company telephone number.
2. The system answers the call and plays the company greeting.
3. The caller enters and extension number on the keypad.
4. The extension rings.

5. If the line is busy or unanswered, no message is taken - the system returns the caller to the company greeting.

Information-Only Mailbox

An information-only mailbox, also referred to as a bulletin board, is not associated with a specific extension number and does not take messages. Instead, it plays a greeting that provides information to callers. The information could be any message that your company wants customers to hear but does not need a person to say.

The call processing sequence for an Information-Only mailbox is a follows:

1. An outside caller calls the company telephone number.
2. The system answers the call and plays the company greeting.
3. The caller enters and extension number on the keypad.
4. The system plays the mailbox greeting, which is information for the caller.
5. After listening to the message the caller can hang up, dial 0 to reach the Operator, or dial another extension number.

Auto-Attendant Park and Page Mailboxes

The Auto Attendant Park and Page mailbox option is present in the TUI Administrator but these mailbox types are <u>not</u> supported at this time.

Single-Digit Mailboxes

Mailboxes 1 through 8 are referred to as single-digit mailboxes, and can be any one of the four types of general use mailboxes. They are most effective when referenced in the company greeting, so that the caller needs to enter only one number to receive the information they want or to reach the most frequently dialed extensions. Three typical uses for single-digit mailboxes follow. For the first two, assume callers hear this company greeting, in which the numbers 1 and 2 refer to single-digit mailboxes.

"Thank you for calling ABC Industries. If you know the number of the person you wish to reach, enter it now. For a personnel directory, press 9. For business hours, press 1. To reach customer service, press 2. If you wish to speak to an operator, press 0 or hold on the line."

Example 1:

Mailbox 1 is an Information-Only mailbox. A typical greeting for mailbox 1 could be:

"ABC Industries is open for business Monday through Friday from 9 am to 6 p.m. and on Saturdays from 9am to 1 p.m."

The caller listens to the information and hangs up when they are done.

Example 2:

Mailbox 2 is an Extension mailbox. For this example, we'll assume extension 214 is the Customer Service telephone. When the caller presses 2, extension 214 rings. The caller might hear the following greeting if the telephone is busy or not answered.

"All of our Customer Service representatives are busy at the moment. Please leave your name, telephone number and a brief message. Someone will get back to you as soon as possible."

You need to record the appropriate name and greeting for mailbox 2 for this setup to work correctly. If extension 214 also belongs to a subscriber, such as the customer service secretary, callers can dial 214 directly to reach that person. The secretary can have a personal greeting identifying himself or herself as the owner of the mailbox but only if Mailbox 2 is programmed as a Transfer-Only mailbox. All department and personal messages are saved in mailbox 214.

A subscriber can serve as the destination for any number of single-digit mailboxes.

Note: Always add a new single digit mailbox first before changing the company greeting to refer to it.

After a caller enters a single digit at the greeting, the system will wait to see if another digit follows. To prevent this slight pause, assign single-digit mailbox numbers that are not the first digit of other mailbox or extension numbers.

Example 3:

Mailbox 8 is programmed as the Language Change mailbox, a special mailbox used to present callers with auto attendant voice prompts in a second language. In this example, when a caller presses 8, all subsequent auto attendant prompts the caller hears will be in French.

"Thank you for calling ABC Industries. For service in French, press 8."

You need to record prompts for mailbox 2 in the appropriate language for this setup to work correctly. For more information, see Recording System Greetings.

Note: The language change applies to the current auto attendant session only. If the caller is returned to the voice mail system during the same call, the system will again prompt the caller to select a language.

Summary Mailbox Functions

The table below summarizes each mailbox type and the functions it supports.

Function	Mailbox Type						
Extension	**Message Only**	**Transfer Only**	**Information Only**	**Menu Tree**	**Guest**	**Front Desk**	
Transfers caller to the associated extension	Yes	No	Yes	No	Yes	Yes	Yes
Allows transfers to the operator	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Plays the mailbox greeting or information	Yes	Yes	No	Yes	Yes	Yes	Yes
Records a message	Yes	Yes	No	No	No	Yes	Yes
Notifies users of messages	Yes	Yes	No	No	No	Yes	Yes

Voice Mail Capacities

Voice mail ports: 4 (expandable to 16 on the CX and to 24 on the MX)

20 for AX

Mailboxes: 20 (expandable to 748)

Hours of voice storage: 4 hours minimum (expandable by adding hard drive)

Note: Installation of a hard drive is strongly advised for systems with more than eight voice mail ports. The drive must be purchased from Mitel (see Install

	FRUs for part number); drives obtained elsewhere are not supported.
Concurrent voice mail or auto attendant sessions:	24
Message storage per mailbox	100 maximum (programmable)
Message retention	From one day to indefinitely for saved messages; indefinitely for unread messages. (Programmable per mailbox)
Prompt languages:	English, French, Spanish with support for bilingual prompting (Note: Bilingual support is a chargeable option.)

Programming Overview

The following is an overview of the CDE and other programming requirements for the embedded voice mail application. Much of the programming is provided completed as part of the default database.

1. In CDE Form 04 (System Options/System Timers) enter the number of mail box licenses purchased (Option 125).
2. (Optional) In CDE Form 04 (System Options/System Timers) enable Voice Mail License for Bilingual Prompts (Option 121).
3. In CDE Form 01 (Configuration), program a voice mail card into the IP bay (default is bay 1), slots 11 and or 12

   Note: The availability of slot 12 is dependent on the number of DSP resources installed. See CDE Form 01 for more information.
4. In CDE Form 02 (Feature Access Codes), program the following access codes as required:
   - **3** Call Forwarding - All Calls (COS Option 260, Internal/External Split Call Forwarding must be DISABLED)
   - **4** Call Forwarding - Internal Only (COS Option 260, Internal/External Split Call Forwarding must be ENABLED)
   - **5** Call Forwarding - External Only (COS Option 260, Internal/External Split Call Forwarding must be ENABLED)
   - **11** Extension General Attendant Access
   - **24** Abbreviated Dial Access
   - **32** Automatic Wake-up (if required for the Hospitality option)
   - **41** Send Message
5. In CDE Form 03 (Class of Service Options), set up two classes of service (one for voice mail users and one for the voice mail ports) with the required options.

Form 03 - COS Define	
COS Option Number and Name	**Status**
Voice mail Users	
206 - Call Forwarding - Busy	ENABLED
207 - Call Forwarding - No Answer	ENABLED
208 - Call Forwarding - External	ENABLED
212 - Can Flash if Talking to an Incoming Trunk	ENABLED

Form 03 - COS Define	
COS Option Number and Name	**Status**
213 - Can Flash if Talking to an Outgoing Trunk	ENABLED
232 - Message Waiting Setup - Lamp	ENABLED
245 - Abbreviated Dialing Access	ENABLED
246 - SMDR - Extended Record	ENABLED
259 - Message Sending	ENABLED
260 - Internal/External Split Call Forwarding	ENABLED
264 - Half Fwd NA Timer for DID Call When VM msg on	ENABLED
277 - Automatic Mailbox Creation	ENABLED/DISABLED (See Note below)
702 - SMDR - Overwrite Buffer	ENABLED
804 - SMDR - Drop Incomplete Outgoing Calls	ENABLED
806 - SMDR - Record Incoming Calls	ENABLED
For Softkey support	
267 - Softkey support for Voice mail	ENABLED
To set Wake-up calls	
202 - Alarm Call	ENABLED
322 - Confirm Wake-up by Offhook (required for display sets only)	ENABLED
Voice mail Ports	
212 - Can Flash if Talking to an Incoming Trunk	ENABLED
213 - Can Flash if Talking to an Outgoing Trunk	ENABLED
216 - Data Security	ENABLED
229 - Voice mail Port	ENABLED
238 - Override Security	ENABLED
245 - Abbreviated Dialing Access	ENABLED
259 - Message Sending	ENABLED
301 - Camp-on	ENABLED
502 -Display ANI/DNIS/CLASS Information	ENABLED
604 - Telephone - Automatic Outgoing Line	ENABLED
606 - Telephone - Enhanced Answering Position	ENABLED
609 - Telephone - Night Service Switching	ENABLED

6. In CDE Form 04 (System Options), enable the required System Options.

Form 04 - System Options		
System Options	**Status**	**Option Number**
For Softkey Support		
DTMF ON Timer	9	69
Support Softkey Access to Voice mail	ENABLED	97
To Set Wake-up Calls		
Automatic Wake-up	ENABLED	11
Automatic Wake-up Alarm	ENABLED	12
Automatic Wake-up Print	ENABLED	13
Automatic Wake-up Music	ENABLED	14

7. In Form 17 (Hunt Groups) program the voice mail ports as members of one or more STN/SET type hunt groups. Set the hunt group type to "terminal". Assign an access code to each hunt group.
8. In Form 19 (Call Rerouting Table) program Station Dial 0 Routing to route Dial 0 calls to the console or other destination.

   Note: For multi-tenant applications that require different Dial 0 destinations for each tenant, program Priority Dial 0 Routing instead of Station Dial 0 Routing and enable COS Option 239 (Priority Dial 0) in the extensions' Class of Service.
9. In Form 49, (Voice Mail Options) set up operational parameters such as prompt language and system greetings schedule.
10. In Form 50, (Mailboxes) customize settings for individual mailboxes and add any special purpose mailboxes (e.g., transfer-only, message-only) as required. Use the subforms to set up Message Notification, Auto-Forward of Voice Mail to Email, and routing for Multi-Level Auto Attendant and Personal Contacts as required.

    Note: Mailbox creation and setup can be done automatically for each new extension added in Form 09 by enabling Option 277 in the extension's Class of Service (Form 03). Form 50 is used to customize mailbox following initial setup. If System Option 124, 107, or 108 are enabled in Form 04, the any mailboxes that the system creates will be GUEST type mailboxes.
11. Assign Call Forwarding to phones. Set Call Forward/No Answer and/or Call Forward/Busy to forward calls to the voice mail hunt group access number.
12. Record system greetings, and assign them to voice mail hunt groups in Form 17.
13. (Optional) In Form 51, (Voice Mail Distribution Lists), set up distribution lists for system-wide use, and then record names for the lists.
14. (Optional), set up PMS integration and Hospitality options for hotel/motel.
15. (Optional), set up RAD ports.
16. (Optional), set up Mailbox Keys.
17. (Optional), set up
18. Test voice mail operation.

Programming by Telephone

The Administrator's mailbox allows you to perform voice mail programming from a telephone. Programming by phone cannot completely replace CDE programming; however, it is convenient for certain functions such as editing mailbox settings and changing passcodes required to access the Administrator's mailbox.

The passcode used to log into the Administrator's Mailbox determines which functions can be accessed:

- Manager Passcode:

  Allows the user to perform all the system administration, mailbox management, and maintenance functions except changing the Administrator passcode or passcode length.
- Administrator Passcode:

  Allows the user to perform all the manager functions plus changing the Administrator passcode and passcode length; and re-installing the voice mail software from the external flash card.

Default passcodes vary depending on the passcode length programmed in Form 50. The factory-set passcodes are 8642 for the Administrator and 6483 for the Manager. The passcodes adjusted for the various passcode lengths are as follows:

Passcode Type	**Passcode Length**			
3	**4**	**5**	**6**	
Administrator	864	8642	86420	864200
Manager	648	6483	64830	648300

IMPORTANT: The Manager passcode should be given to the person who performs the day-to-day operations, such as adding mailboxes and changing greetings. The Manager passcode provides limited access to the database thereby reducing the likelihood of novice users causing serious disruptions to the system. (Note that users are still prompted to begin a new installation when they enter the Manager Passcode but get the response "Invalid Selection" when they attempt to do so.) To protect system security, change the passcodes at first login and keep them confidential.

Default passwords for the operator and all other mailboxes are 1234 and 1111 respectively. The passcodes adjusted for the various passcode lengths are as follows:

Passcode Type	**Passcode Length**			
3	**4**	**5**	**6**	
Operator	123	1234	12340	123400
All other mailboxes	111	1111	1111	11111

Accessing the System Administrator's Mailbox

Follow these steps to log in to the administrator's mailbox.

1. From any internal DTMF telephone, lift the handset and obtain dial tone.
2. Dial the voice mail hunt group number. The system answers and plays the usual greeting.
3. Press * when you hear the greeting.
4. When prompted for a mailbox number, enter the system administrator mailbox number: three to six 9s depending on the mailbox length.
6. When prompted for a passcode enter the Adminstrator's passcode or the Manager's passcode. (See above for defaults.)

You are now logged in to the system administrator's mailbox. Follow the voice prompts for instructions using the following sections for for reference.

Figure: Administrator Menu

Editing or Deleting Mailboxes

Note: Do not use a telephone to change a mailbox's e-mail protocol from IMAP (Unified Messaging) to SMTP. Instead, use the CDE. For details, see Unified Messaging.

To edit a mailbox, follow these steps:

1. Access the System Administrator's Mailbox.
2. Press 2 for the Mailbox menu.
3. Press 2 for the Edit Mailbox menu.
4. Enter the mailbox number to edit.
5. Follow the prompts and enter the changes for each parameter.

   Note: Press # if you do not need to change a parameter. Press * if you want to reset to the default parameter.

   Note: The Auto Attendant Park and Page mailbox option is present in the TUI Administrator but these mailbox types are <u>not</u> supported in this release.
6. The system returns you to the Mailbox menu after you have altered or skipped the parameters.

To delete a mailbox, follow these steps:

1. Access the System Administrator's Mailbox.
2. Press 2 for the Mailbox menu.
3. Press 3 for the Delete Mailbox menu.
4. Enter the mailbox number to delete.
5. Press # when done.

Setting Call Forward to Voice Mail

1. Access the System Administrator's Mailbox.
2. Press 2 for the Edit Mailbox menu.
3. Enter the mailbox number to edit.
4. Press # at each prompt until you hear the "Technician Function Code" prompt.
5. Enter the desired Non-busy greeting option number from the Non-busy Options table below for the default condition.
6. The Technician's function code prompt repeats. Enter the desired Busy option from the Busy Options table below.
7. Repeat for other mailboxes as required.

Non-Busy Options

The following table describes the behaviour when bilingual operation is disabled and the mailbox is set up—that is, a name and personal greeting have been recorded by the owner.

If the mailbox is NOT setup, the fixed system message, "I'm sorry, the person you're trying to reach is unavailable at this time" plays instead of the personal greeting and the mailbox number plays instead of the mailbox name.

Greeting Option	Plays...
0 (default)	a personal greeting (as recorded by the mailbox user) followed by the fixed system prompt, "Begin speaking after the tone, then hang-up when you are finished or press any key for further options.
1	a personal greeting as per Option 0, followed by the leave-message tone, and then silence.

Greeting Option	Plays...
2	the fixed system message, "I'm sorry, the person you're trying to reach is unavailable at this time" followed by the fixed system prompt, "Begin speaking after the tone, then hang-up when you are finished or press any key for further options."
3	the fixed system message as per Option 2, followed by the leave-message tone, and then silence.
4	a standard greeting (as recorded by the administrator or other person), followed by the mailbox owner's name (as recorded by the owner),and then the fixed system prompt, "Begin speaking after the tone, then hang-up when you are finished or press any key for further options."
5	a standard greeting as per Option 4, followed by the leave-message tone, and then silence.
6	a standard greeting as per Option 5 minus the mailbox user's name, followed by the fixed system prompt, "Begin speaking after the tone, then hang-up when you are finished or press any key for further options."
7	a standard greeting as per Option 6, followed by the leave-message tone, and then silence.

Busy Options

The following table describes the behaviour when the mailbox is set up—that is, a name and personal greeting have been recorded by the mailbox owner.

If the mailbox is NOT setup, the fixed system message, "I'm sorry..." followed by the mailbox number and "...is currently on the phone" plays instead of the mailbox name and personal greeting.

Greeting Option	Plays...
0 (default)	the following fixed system message in both the primary and alternate languages if Bilingual operation is enabled and the caller has not chosen a language: "I’m sorry, the person you are calling is currently on the phone" followed by the mailbox owner's personal greeting, then the fixed system prompt, "Begin speaking after the tone, then hang-up when you are finished or press any key for further options. If Bilingual operation is disabled, the following plays: "I’m sorry…" followed by the mailbox owner's name, then "...is on the phone" and the fixed system prompt, "Begin speaking after the tone, then hang-up when you are finished or press any key for further options."
1	"I'm sorry…" followed by the mailbox owner's name, then "...is on the phone" and the fixed system prompt, "Begin speaking after the tone, then hang-up when you are finished or press any key for further options."
2	the fixed system prompt ("I'm sorry...") as per Option 1, followed by silence instead of "Begin speaking after the tone...".

Greeting Option	Plays...
3	the fixed system message, "I'm sorry, the person you are calling is currently on the phone." followed by the fixed system prompt, "Begin speaking after the tone, then hang-up when you are finished or press any key for further options."
4	the fixed system message, "I'm sorry, the person you are calling is currently on the phone." followed by silence.
5	the currently selected Non-Busy Greeting option. For example, if the Non-Busy Option is 0, then callers encountering a busy extension will hear the same greeting heard when the extension is not answering or call forwarded always to voice mail
6	a standard greeting as per Option 5 minus the mailbox user's name, followed by the fixed system prompt, "Begin speaking after the tone, then hang-up when you are finished or press any key for further options."
7	a standard greeting as per Option 6, followed by the leave-message tone, and then silence.

Conditions

- For proper operation, the phones should be programmed for Call Forward - No Answer and Call Forward– Busy conditions for both internal and external calls. If programmed for Call Forward - Always, the voicemail system will not know the state of the phone and use the Call Forward – No Answer prompts.
- Mailboxes configured not to play a personal greeting still offer the user the option to record one via the User Options menu. The personal greeting, if recorded, will be ignored.

Setting Business Hours

Follow these steps to set business hours:

1. Access the System Administrator's mailbox.
2. Press 3 for the 'Date/Business Hours' menu.
3. Press 2 for the 'Set Business Hours' menu.
4. The system cycles through the days of the week asking you to input the opening and closing times for each day, beginning with Sunday. If your business will not be open on the day listed, enter [0000] for both the opening and closing times.
   Enter the opening time (in 24-hour format): [hhmm].
   Enter the closing time (in 24-hour format): [hhmm].

Changing the Administrator Passcode

To change the administrator passcode:

1. Access the System Administrator's Mailbox.
2. Press 6 for the Passcode menu.
3. Press 1 to change the passcode.
4. Enter your new passcode: [nn...n]. You may use any keys on the keypad except * and #.
5. When prompted, enter the passcode again: [nn...n]

Changing the Administrator Passcode Length

1. Access the System Administrator's Mailbox.
2. Press 6 for the Passcode menu.
3. Press 2 to change the passcode length.
4. Enter the new passcode length; valid choices are between 3 and 6 digits.

Note: All passcodes on the system will be this new length, the administrator passcode as well as user passcodes. Notify all subscribers before making this change.

Changing the Manager Passcode

1. Access the System Administrator's Mailbox.
2. Press 6 for the Passcode menu.
3. Press 3 to change the manager passcode..
4. Enter your new passcode: [nn...n]. You may use any keys on the keypad except * and #.
5. When prompted, enter the passcode again.

Creating and Managing Distribution Lists

To create a new distribution list or to add mailbox numbers to an existing list:

1. Access the System Administrator's Mailbox.
2. Press 7 for the Distribution Lists menu.
3. Enter a 3-digit distribution list number, 001 to 049.
4. Press 1 to add mailbox numbers to this list.

   After you enter each mailbox number, the system confirms that the number has been added. Continue to add mailbox numbers until all are entered.

   If you enter a mailbox number that already exists, the system tells you the number is a duplicate but does not enter the number twice.

5. Press # when you enter the last mailbox number to stop adding.

To review an existing distribution list:

1. Access the System Administrator's Mailbox.
2. Press 7 for the Distribution Lists menu.
3. Enter a 3-digit distribution list number, 001 to 049.
4. Press 2 to review the mailbox numbers in this list.

   Provided the list contains mailbox numbers, the system itemizes the mailboxes contained in the list by speaking the name associated with each mailbox. If no personal names are recorded, the system reads all the mailbox numbers to you.
5. Press any key to interrupt the review and return to the Distribution Lists menu.

To delete a mailbox from an existing distribution list:

1. Access the System Administrator's Mailbox.
2. Press 7 for the Distribution Lists menu.
3. Enter a 3-digit distribution list number, 001 to 049.
4. Press 3 to delete a mailbox number from this list.
5. Enter the number of the mailbox to be deleted.

To record a name for a distribution list:

1. Access the System Administrator's Mailbox.
2. Press 7 for the Distribution Lists menu.
3. Enter a 3-digit distribution list number, 001 to 049.

4. Press 4 to record a name for this list.
5. Press any key to stop recording.

Enabling FAX Tone Detection

1. Access the System Administrator's Mailbox.
2. Press 5 for the 'System Parameters' menu.
3. Press 3 for the 'Set FAX Extension' menu.'
4. To enable the FAX feature, press [1]; otherwise, press [2].
5. If you enable the FAX feature, enter the FAX extension number.

Setting up Transfer Options

In the System Parameters menu, the TUI administrator may offer a selection to change transfer option to "Unsupervised". This option is not supported at this time. We recommend that you use the default setting of "Supervised".

Setting up RAD Greeting and Ports

To set up RAD greetings:

1. Access the System Administrator's Mailbox.
2. Press 8 to select the RAD Greetings menu.
3. Press 2 to configure a RAD greeting set.
4. Select the number of the RAD greeting set to configure (10 - 39).

   The first of the five greetings in the set plays (if previously recorded) followed by prompts to assign a different first greeting or to skip to the next greeting in the set.

5. When prompted, specify the number of times from 1 to 99 that you want the greeting set to play. Enter 99 to play the set continuously until the caller hangs up.

To assign RAD greeting sets to ports:

Note: Port assignments made by telephone are lost following a system reset. Use Form 17 to assign the greetings instead.

1. Access the System Administrator's Mailbox.
2. Press 5 to select the System Parameters menu.
3. Press 2 to assign greeting sets to be used by each port
4. When prompted, for each port enter a System Greeting number from 1 to 9 or a RAD greeting set number from 10 to 39.

Performing a New Installation

Performing a new installation removes all mailboxes in Form 50, except those belonging to the operator and the Administrator. All other voice mail configuration data is restored to factory defaults. However, any previously recorded company greetings are preserved and may be re-recorded during the new installation.

To re-install the voice mail software:

1. Access the System Administrator's mailbox.
2. Select option number 1 from the main administration menu, then follow the prompts.
3. Reboot the SX-200 ICP.

Recording System Greetings

System Greetings play for outside callers. There are two types of system greetings: Primary and Alternate.

The Primary greeting plays on all voice mail ports by default. You can record up to eight alternate greetings, and assign them to play on different voice mail ports in the Hunt Group Options Subform (Form 17). This feature provides flexibility by allowing, for example, multiple businesses to share a single SX-200 ICP system, or a single business to have specialized greetings their 1-800 numbers.

Both the primary and alternate greetings actually consist of three separate greetings: an open greeting, a closed greeting and a temporary greeting. When the greetings play is determined by options in Form 49, Voice Mail Options. One option plays greetings based on whether the system is in Day or Night service: in Day service, the open greeting plays; in Night service, the closed greeting plays. The other option plays greetings based on business hours specified in the form.

The temporary greeting is typically for holidays. It overrides the open and closed greetings for the number of days you specify in Form 49.

Bilingual systems require a welcome greeting and a set of primary, alternate, and temporary greetings in each of the two languages. The welcome greeting includes an instruction for callers to enter the Language Change digit if they want to hear prompts in the other language. The digit is defined in Form 49.

Systems that have the Voice Mail Property Management System option enabled in Form 04 are provided with a Guest Login Greeting. Guests hear the greeting when they log into the voice mail system.

Primary Greeting Set

Follow these steps to record the primary greeting set:

1. Access the System Administrator's mailbox.
2. Press [4] for the System Greetings menu.
3. Press [1] to set the primary greeting.
4. If prompted, press [1] to assign greetings in the default system language or [2] for the alternate language.
5. Follow the instructions below for assigning Open, Closed or Temporary greetings for your primary greeting set.

Alternate Greeting Set

Follow these steps to record an alternate greeting set:

1. Access the System Administrator's mailbox.
2. Press [4] for the System Greetings menu.
3. Select a number, 2 through 9, for an alternate greeting set.
4. Follow the instructions below for making Open, Closed, or Temporary greetings for your alternate greeting set.

Open or Closed Greetings

To set open or closed greetings for a primary or alternate greeting set:

1. After making your selection for either a primary or an alternate greeting (from the previous page):
2. Press [1] for an Open greeting.

   -or-
3. Press [2] for a Closed greeting.

4. If prompted, press [1] to assign greetings in the default system language or [2] for the alternate language.
5. Record the greeting speaking clearly into a handset, not a speaker phone. The following are typical open and closed greetings:

   "Thank you for calling ABC. If you know the number of the person you want to contact, enter it now. For a company directory, press 9. For assistance, press 0 or hold for the operator."

   "You have reached ABC Industries. We are closed for the day, but if you wish to leave a message, enter an extension number now. For a company directory, press 9."

   For advice on recording greetings, see Tips for Recording System Greetings.
6. Press any key to stop recording.
7. After recording, select one of the following options:

   [1] Accept
   [2] Review
   [3] Re-record
   [*] Cancel

If you are recording bilingual greetings, repeat the above procedure beginning at step 2 for the other language.

Temporary Greetings

To set temporary greetings for a primary or alternate greeting set:

1. Choose the primary or alternate greeting set and then press [3] to create a temporary greeting.
2. Enter the number of days for the greeting to play: [nn]
3. Enter the number of days from 1 (entered as 01) to 99 that the greeting should play. To cancel an existing temporary greeting, enter 00 for the number of days.
4. If prompted, press [1] to assign greetings in the default system language or [2] for the alternate language.
5. Record the greeting, speaking clearly into a handset, not a speaker phone. The following is an example temporary greeting:

   "Happy Holidays from all of us at ABC Company. We are closed from Friday, December 23 until Monday, January 2. To leave a message, enter the number of the person you wish to reach or press 9 for the directory."

   For advice on recording greetings, see Tips for Recording System Greetings.
6. Press any key to stop recording.
7. After finishing, select one of the following options:

   [1] Accept
   [2] Review
   [3] Re-record
   [*] Cancel

To disable an existing temporary greeting, first record something over the current greeting, then select '00' for the number of days.

If you are recording bilingual greetings, repeat the above procedure beginning at step 2 for the other language.

Bilingual Welcome Greeting

Follow these steps to record the Bilingual Welcome Greeting:

1. Access the System Administrator's mailbox.
2. Press [4] for the System Greetings menu.

3. Press [1] to set the primary greeting.
4. Press [4] to set the bilingual welcome greeting.
5. Record the greeting, speaking clearly into a handset, not a speaker phone. The following, is a typical bilingual greeting in English and French:

   "Thank you for calling ABC Industries. Merci d'appeler les Industries ABC. Pour le service en francais, composez 8."
6. Press any key to stop recording.
7. After finishing, select one of the following options:

   [1] Accept
   [2] Review
   [3] Re-record
   [*] Cancel

Guest Login Greeting

Follow these steps to record the Guest Login Greeting:

1. Access the System Administrator's mailbox.
2. Press [4] for the System Greetings menu.
3. Press [9] to set the Guest Login greeting.

   Record the greeting, speaking clearly into a handset, not a speaker phone. For bilingual systems, record the greeting in both languages.
4. Press any key to stop recording.
5. After finishing, select one of the following options:

   [1] Accept
   [2] Review
   [3] Re-record
   [*] Cancel

Tips for Recording System Greetings

- Make the greeting brief. No one enjoys listening to 10 minutes of instructions just to learn the choice they wanted was the first option.
- Inform users that they can immediately dial an extension, then offer other options. This helps to reduce frustration for callers who dislike interacting with the automated attendant.
- Include the "dial 9" prompt which allows callers to dial by name rather than extension number.
- Always give callers the option to press a key to speak with a live operator. This makes the system foolproof and, again, less frustrating for callers who can't tolerate the automated attendant. Make sure customers can then speak to the live operator. Don't allow these transfers to go to a mailbox. Instead route them to a hunt group of several extensions to make sure someone answers the call.

  Note: If the first option on the menu is to press a digit to speak with an operator, be prepared that the most of your callers will bypass the automated attendant and speak with an operator. This will greatly reduce the benefits of having an automated attendant.
- Provide open and closed greetings. Callers need to know that the business is closed and transferring to an extension will probably not result in the call being answered. Unless your business is large, it is probably not necessary to offer a full set of menu options from the closed greeting. It is more efficient to offer a single keypress transfer to "company mailbox" and then have a receptionist retrieve messages the next day and transfer to the appropriate person.

Programming Voice Mail Features

Multiple Language Support

The embedded voice mail system supports English (North American), French (North American), and Spanish (North American).

The system is shipped with the voice mail prompts in English. Additional languauges can be selected during software upgrade. Specify the language to use in Form 49, Voice Mail Options.

You can set up the system to operate either as a unilingual or bilingual system. A unilingual system prompts users in any one of the above language (default is English). A bilingual system provides prompts in two of the three available languages.

Note: Support for bilingual prompts is a purchasable FOR option.

To set up a bilingual system:

1. In Form 04, enable Voice Mail License for Bilingual Prompts (Option 121)
2. Complete the CDE programming for Embedded Voice Mail.
3. In Form 49, set the default and alternate languages plus the Language Change Number.

   Callers reaching the auto attendant or a subscriber's mailbox can dial the Language Change number to hear subsequent prompts in the alternate language.
4. Log into the Administrator's mailbox and using the Greetings menu verify existing greetings, and then record new greetings in the alternate language, including the bilingual welcome greeting. For more information, see Bilingual Welcome Greeting.

Multi-level Auto Attendant (MLAA) Menu Tree Functionality

Multi-level Auto Attendant (MLAA) allows you to set up a hierarchical menu structure that provides callers with better self-service access to the department or person they're trying to reach. In an MLAA system, callers reaching the Auto Attendant are routed from the main menu through to one or more additional sub menus (menu nodes) until their call is answered.

You can program up to 10 multi-level menus, each with its own greeting and prompts. The greeting at the top level is the primary company greeting with the usual prompts to dial an extension number, "0" for an operator, and other digits to reach destinations such as Sales or Technical Support. At the subsequent levels (the sub menu nodes) the greetings prompt callers to make further selections. The greeting for a Technical Support node might offer choices based on product type as follows:

"For SX-200 ICP technical support, press 1. For 3100 ICP technical support, press 2. For all other products, press 3. To return to the previous menu, press star. To repeat the choices, press pound."

From this menu, dialing 1 or 2 transfers the caller to a technical support representative. Dialing 3 might route the caller to yet another sub menu with more choices.

Note: The star and pound prompts help ensure that the system is easy to use. Be sure to include them in your greetings, particularly the star prompt if you have multiple menu levels.

Note: Destinations programmed in Menu Tree mailboxes must have mailboxes associated with them. (For example, Menu 2 is assigned access code 1234, which is a hunt group in the SX-200 ICP. If a caller is in a Menu Tree mailbox and is instructed to "Dial 2", then you must have a mailbox 1234 or the call will not complete.)

Menu Node Mailbox Time Out Handling

For each menu node, you can determine the system's behavior if a caller does not (or cannot) enter anything:

- divert the caller to the operator (default), or
- terminate the call

Diverting the call to the operator (default):

The greeting is repeated three times, then the call is forwarded to

1. The menu node operator or its mailbox (programmed in Subform 50, Contacts or Menus).
2. If the menu node operator is not programmed, the call is diverted to the system operator or its mailbox (programmed in Form 50 - Mailboxes).
3. If the system operator is not programmed, the call is sent to mailbox 0.

Note: If both the menu node operator and the system operator are programmed, preference is given to the menu node operator.

Terminating the call:

The greeting is repeated three times, then the call is terminated.

To set up MLAA:

1. In Form 50, Mailboxes, create a single-digit mailbox for each menu choice in the main menu. Using the following greeting as an example,

   "Thank you for calling ABC company. If you know the extension of the person you are calling, enter it now. For sales, press 1, for customer support, press 2, for assistance press 0, or stay on the line."

   create three mailboxes: one for sales, one for customer support and one for the operator extension (0):
2. For each mailbox that branches to a sub menu (mailboxes 1 and 2 in the example), set the mailbox type to Menu Tree.

   Note: You do not need to assign an extension to Menu Tree mailboxes.
3. After all the required mailboxes have been added and with the cursor positioned in one of them, press the MENUS softkey.
4. Enter a destination for the key prompts in the second level menu. (The form has room to display only five fields at a time; use the SHOW MORE softkey to display the others.)

   Using the following greeting as an example,

   "For SX-200 ICP technical support, press 1. For 3100 ICP technical support, press 2. For all other products, press 3."

   enter extension numbers (hunt group numbers may be more appropriate) for the 1 and 2 keys. The 3 key could also be assigned to an extension or hunt group number or it could be Menu Tree mailbox for a third level menu.

   Note: A single-digit mailbox could be used for the 3 key but because there are so few of them, consider using a regular extension mailbox instead.
5. After all the required key prompts have been defined, press the QUIT softkey to return to the previous form.
6. Repeat steps 3 to 5 for each additional Menu Tree mailbox.
7. From any extension in the system, log into each Menu Tree mailbox, using the mailbox's number and passcode.
8. Record a greeting with the required prompts.

   Note: For each sub menu greeting, include a prompt to press star to return to the previous menu.
9. After all the menus have been set up, test the prompts to ensure that the call flow is valid.

To change the menu node mailbox time-out behaviour

1. Access the Administrator's Mailbox.
2. Press 2 to edit or delete mailboxes.

3. Press 2 for the Edit Mailbox menu.
4. Enter the mailbox number to edit.
5. Press # until you get to the Menu Node Time Out option. The system specifies the active Menu Node Time Out behavior:
   - "Disconnect on menu mailbox time out is disabled" – callers are sent to the mailbox operator upon time out.
   - "Disconnect on menu node time out is enabled" – callers are disconnected upon time out.
6. To disconnect callers upon time out, press 1 (enable disconnect).
   To send callers to the mailbox operator upon time out, press 2 (disable disconnect).

Note: The default is "Disconnect on menu mailbox time out is disabled".

7. When you are prompted to "Enter Technician's Function Code", press * for each of the two prompts.
8. The system returns you to the mailbox.

Message Notification

Message Notification allows users to be notified whenever messages arrive in their mailbox. The system supports five types of notification:

- To an internal extension (that does not have a message indicator).
- To an outside telephone number.
- To a message pager.
- To a tone-only pager (or beeper).
- To a digital pager.

A notification number can be assigned for each mailbox subscriber.

Note: Embedded voice mail Release 6.22 (included in SX-200 ICP Releases 2.0 and later) now supports notification on every new message regardless of whether or not notification for previous messages has already been answered. This feature is normally deactivated by default and may only be enabled or disabled by accessing the Administrator's mailbox.

How it Works

For internal extension numbers and outside phone numbers:

The system calls the notification number when the subscriber's mailbox receives its first new message. It does not call for subsequent messages.

The system action depends on the response at that number:

If the notification number is busy or does not answer, the system tries calling again at 15 minute intervals. This procedure is repeated twice for no answer status, three times for busy status. The message light on the user's phone remains lit until the user saves or deletes all new messages.

If the phone is answered, the system prompts for the user's passcode. The user can listen to their message immediately.

If two mailboxes are associated with an extension and one of those mailboxes is the operator's, when that mailbox receives a new message, it is always the operator's mailbox that is notified first. If call notification is not enabled for the operator's mailbox, no call is made to the notification number.

For pagers:

Every time the subscriber's mailbox receives a new message, the system calls the notification number. The number is called only once, regardless of whether it is busy or does not answer.

The message light on the user's phone remains lit until the user saves or deletes all new messages.

The voice mail system starts the paging procedure again every time a new message arrives.

Using Pagers for Message Notification

Pager Types

Message Notification works with three types of pagers: a beeper (or tone-only), a messaging type, and a digital type. The following table describes, in general, how the system works with each pager type. The process can change depending on the pager number parameters you enter for individual mailboxes.

Pager Type	How the system works
Tone-only	Dials the phone number of the pager.
Message	Dials the phone number of the pager and announces: "(Name) ... You have new messages. Please access your mailbox through the pager speaker."
Digital	Dials the phone number of the pager and displays the Digital Pager Callback Number from CDE form 49 then the mailbox number.

Pager Number Parameters

Pager numbers can vary in length, but some pagers require pauses or other actions to occur within the pager number. The [*] key plus a digit define these specific actions as shown in the following chart.

Command	Function	Audio Playback	Displays in SX-200 ICP...
*1	Pause 1 second	“comma”	,
*3	Dials the [#] key	“pound”	#
*4	Wait for dial tone	“star 4”	*4
*5	Wait for answer	“star 5”	*5
*6	Do not wait for answer	“star 6”	*6
*7n	Pause n seconds, where n is between 1 and 9 seconds	“star 7 (1-9)”	*7(1-9)
**	Dials the [*] key	“star”	*

Helpful Tips for Setting Up Pager Notification

- Each pager needs its own distinct number sequence to work correctly. When you set up the pager number in a mailbox, the subscriber should supply you with the characteristics and requirements of their pager.
- There is a 35-character limit to the total number of digits in a pager number.
- The *5 (wait for answer) parameter takes affect at the point where the SX-200 ICP first encounters it in the pager number. Any subsequent *5 parameters are ignored. If the call is answered and the pager is digital, the system dials the remainder of the pager number followed by the call back number (555-1212 unless changed) and mailbox number.

- Use the *6 (do not wait for answer) parameter for time-based pagers. The SX-200 ICP system does not wait for any signals to perform the action dictated by the pager type: play a prompt or display the call back number. You must place the *6 parameter at the end of the pager number.
- In the absence of a *5 or *6 parameter in the pager number, the SX-200 ICP system dials the entire mailbox pager number and waits for an answer. If the call is answered, it performs the action dictated by the pager type: plays a prompt or displays the call back number.
- If you have a digital pager and decide to put the callback number and mailbox number in the mailbox notification number, include a ‘Do not wait for answer’ command (*6) in the notification number and make it a beeper type (‘tone-only’) pager. Doing this prevents the SX-200 ICP system from dialing the system-wide callback number.

Pager Examples

Assume that the pager number is 123-4567, the subscriber’s pin number is 54321, and the mailbox number is 201.

To specify this action on a digital pager...	**Enter this pager number...**	**Subscriber sees...**
Dial the pager number Wait for an answer Display the call back number and mailbox number	1234567	5551212*201
Dial the pager number Wait for an answer Pause 2 seconds Dial a pager company access code or PIN number Pause 3 seconds Display the call back number and mailbox number	1234567*5*7254321*73	5551212*201
Dial 9 for outside number Dial the pager number Wait 22 seconds Send the callback number and mailbox number	91234567*79*79*74	5551212*201
Dial 9 for outside line Dial the pager number Do not wait for an answer but wait three seconds Display the call back number and mailbox number	91234567*73*6	5551212*201

Setting Up Message Notification

Before message notification can take place, either:

1. Enable "Send Notification Calls" in Form 49 and enter a callback number (i.e., the number users dial to hear their messages) to display on digital pagers.
2. For each user requiring notification, enter the notification type and the digit "9" plus the number (e.g., 95551234) in the "Other" subform of Form 50. This causes notification to be on 24 hours a day for the user.

   Note: Each user can turn their notification off or adjust the notification schedule as needed. Refer users to the Voice mail User Guide for these procedures.

   -OR-

   You can allow individual users to modify their own mailbox type and number by enabling the Notification Access option in the Voice Mailboxes form.
3. Enter the notification type and number as follows:

   Notification Type. Indicate where notification is to take place:

   - extension
   - telephone (external phone number)
   - message pager
   - tone-only pager (beeper)
   - digital pager

   Notification Number. The user supplies this number to you. If the number is a pager number, use the information given in Pager Number Parameters, for entering pager numbers.

   Notification Access. Give the user the ability to modify his or her own notification type and number.

Note: Guest mailboxes do not provide users with user access to message notification.

Voice Mail Notification On Every New Message

To activate voice mail notification on every new message (which can only be performed using the telephone user interface):

1. Access the System Administrator's mailbox.
2. When prompted for a passcode, enter the Technician's passcode (default is 9731) instead of using either the Adminstrator's passcode or the Manager's passcode.
3. Press 9.
4. Enter 20301 to enable notification on every new message.

To deactivate voice mail notification on every new message (the default setting):

1. Access the System Administrator's mailbox.
2. When prompted for a passcode, enter the Technician's passcode (default is 9731).
3. Press 9.
4. Enter 20302 to disable notification on every new message.

Recorded Announcement Devices

The embedded voice mail system can provide recorded announcement device (RAD) functionality, eliminating the need for external tape machines or other audio-playing devices. RADs are commonly used to automatically answer lines and deliver pre-recorded messages such as, "All of our representatives are busy helping other callers, please continue to hold to maintain your call priority." When the RAD message finishes playing, the caller usually hears music-on-hold while waiting for an agent to become available. RAD messages may also give the caller information, which answers their questions, thus resulting in a 'good' abandoned call. They may also provide advertising or promotional information to callers while they're waiting for someone to take their call.

Any number of voice mail ports can function as RAD ports. (See RAD-Only Operation for considerations concerning the use of all voice mail ports for RAD.) Each RAD port is assigned a set of up to five messages (called "greetings") that play one after another. When the entire greeting set has played, the port will either hang up or replay the entire set the number of times specified. Up to 30 RAD greeting sets can be recorded for a total of 150 greetings (30 sets multiplied by 5 greetings per set). RAD greetings can be recorded in multiple languages, although callers cannot choose the language they want to hear as they can for other greetings in systems that are set up for bilingual operation.

Two or more ports can share the same set or subset of greetings. Shared greetings are common in ACD applications. For instance, ACD groups, each serviced by a separate RAD port, can all use the RAD message "Please hold to maintain your call priority," By comparison, only a Sales ACD group can use the RAD message "Please hold to speak with a sales representative." Combining generic and application-specific messages from different greeting sets in this fashion effectively reduces the number of RAD ports required.

Setting up RAD

Your knowledge of the overall call flow and approximate duration of the individual messages in it reduces the amount of time to program the RAD and ACD groups. Before programming the RAD, determine the intended RAD usage and expected call flow. Verify your message text and repeat it several times to ascertain the amount of time it takes to speak the complete the message.

The following table details the constraints on programming RAD ports.

	With Default Dual (2) DSP	With an additional Dual (2) DSP = 4 DSP	With an additional Quad (4) DSP = 6 DSP	Default Dual replaced with Quad + another Quad = 8 DSP
Business Option 1 (CX/CXi)	4 VM ports - combo of VM or RAD ports	16 VM ports - combo of VM or RAD ports	16 VM ports - combo of VM or RAD ports	N/A
Business Option 1 (MX)	4 VM ports - combo of VM or RAD ports	12 VM ports - combo of VM or RAD ports	18 VM ports - combo of VM or RAD ports	24 VM ports - combo of VM or RAD ports
Hospitality (AX)	N/A	20 VM ports combo of VM or RAD ports (Base Configuration on AX)	20 VM ports combo of VM or RAD ports	20 VM ports combo of VM or RAD ports
Business Option 2 (MX only)	8 VM ports - combo of VM or RAD ports	18 VM ports - combo of VM or RAD ports	24 VM ports - combo of VM or RAD ports	24 VM ports - combo of VM or RAD ports
Hospitality Option (MX only)	8 VM ports - combo of VM or RAD ports)	12 VM ports - combo of VM or RAD port	18 VM ports - combo of VM or RAD ports	24 VM ports - combo of VM or RAD ports

Notes:

1. This table indicates resource requirements for voice mail ports; it does not indicate overall DSP requirements. For example, Business Option 1 with Dual DSP does not have the resources for Compression.
2. Voice mail ports, whether they are used for voice mail or RAD, are enabled contiguously -- for example, VM port 1 (default extension 301) to port 8, (default extension 308). Business Option 1 with Dual DSP will allow 2 VM ports and 2 RADs or 3 VM ports and 1 RAD or 1 VM port and 3 RADs or 4 VM ports and 0 RADs.
3. Installation of a hard drive is strongly advised for systems with more than eight voice mail ports and for systems with heavy Record a Call usage. The drive must be purchased from Mitel (see Install FRUs for part number); drives obtained elsewhere are not supported.

4. When configuring voice mail ports, ensure that some of the last ports are not configured as RAD ports to avoid any potential conflicts with Message Notification. It is recommended that at least two or three ports be left free for Message Notification dependant upon system usage. Embedded voice mail will attempt to use the last programmed voice mail port that is available for Message Notification. If the last port is not available, the second last port will be attempted and if that port is busy, the third last port and so on. Embedded voice mail will continue in this fashion until it finds a free port but Message Notification will not function properly on a port assigned to RAD.

Setting up RAD ports is a four-step process:

1. Enable the RAD option and assign the RAD ports to a hunt group.
2. Record RAD greetings.
3. Configure RAD greetings sets.

   In this step you define up to five RAD greetings which constitute the RAD greeting set and define how many times the set should play. You may configure up to 30 RAD greeting sets.
4. Assign RAD greeting sets to voice mail hunt groups of type RECORDING.

   In this step you specify which RAD greeting set plays when a call arrives on a particular voice mail port.

In addition, there is an optional step, assigning RAD greeting sets to voice mail ports via telephone.

Step 1: Enable RAD Option

- Enable System Option 134, Recorded Announcement Devices in Form 04.

Step 2: Record RAD greetings

To record RAD greetings:

1. Access the System Administrator's mailbox. Note that you must have the default Password.
2. Press 8 to select the RAD Greetings menu.
3. Press 1 to record a greeting.
4. Select the number of the RAD greeting to record (1- 200).

   If the selected RAD greeting already exists, you will hear it along with prompts to accept, review or re-record it.
5. Record the greeting, speaking clearly into a handset (do not use a handsfree or speaker phone).
6. Follow the voice prompts to record each additional greeting set.

   Note: Leave fields for any greetings that are not required blank.

Step 3: Configure RAD greeting sets

Use the following procedure to setup RAD greeting sets and then proceed to Step 4. To program the RAD greeting sets, via the Voice Mail System administrator, refer to step 3a.

1. Go to Form 49 (Voice Mail Options) and open RAD Setup.
2. Configure RAD greeting sets, assuming that a greeting set can consist of the entire message. For example, Greeting 1 can be "Welcome to Company X" (this message can be reused). Greeting 2 can be "Your have reached the Sales department" (this message can be used by only one department). Greeting 3 can be "All of our agents are busy" (this message can be reused).

   Note: The Recording Failure to Hangup Timer (COS option 404 for the RAD ports in Form 03) must be long enough that it does not expire while messages are playing. The timer should not start until the Message Length timer in Form 17 (Recording Hunt Group Options subform) expires or the caller hangs up.
3. Specify how often each set should play by entering a number in the range 1 to 98 in the TIMES TO PLAY field. Press the FOREVER softkey to play the set continuously until the caller hangs up. Verify that disconnect supervision is provided by the CO.

Step 3a: Configure RAD greeting sets via the Administrator's Mailbox

This step is not necessary if the greeting sets are programmed via SX-200 ICP CDE.

Use the following procedure, to configure RAD greeting sets using a telephone.

1 Access the System Administrator's mailbox.
2. Press 8 to select the RAD Greetings menu.
3. Press 2 to configure a RAD greeting set.
4. Select the number of the RAD greeting set to configure (10 - 39).

   The first of the five greetings in the set plays (if previously recorded) followed by prompts to assign a different first greeting or to skip to the next greeting in the set.
5. When prompted, specify the number of times from 1 to 99 that you want the greeting set to play. Enter 99 to play the set continuously until the caller hangs up.

Step 4: Assign the RAD ports to a hunt group

1. Refer to Form 17 to determine which VM ports, and how many of them, are being used for VM. This step allows you to select VM ports (301 - 304), that are not being used, when you wish to program RAD ports. Note that VM Hunt groups are set for Circular Station/Set.
2. Assign the RAD ports (310 - 315) to one or more hunt groups in Form 17. Set each RAD hunt group type to RECORDING.
3. For each hunt group that contains a RAD port:
   a. Give the RAD Hunt Group a name. (for example, Greeting 1)
   b. Set the Message Length timer, in Form 17 (Recording Hunt Group Options subform), to the time length of the RAD greeting set (Greeting 1 + Greeting 2 + Greeting 3) multiplied by the amount of times repeated plus three seconds.
   c. Set the RAD Greeting set (number 10 - 39) as programmed in Form 49.

Notes:

1. When the Message Length timer expires, the Recording Failure to Hangup Timer starts. If the RAD port is still off-hook when this timer expires, the port is placed into Do Not Disturb state making it unavailable to answer calls. The port remains unavailable until Do Not Disturb is disabled. The Recording Failure to Hangup Timer is Option 404 in Form 03, COS Define.

2. Put RADs in a COS of their own, with only COS option 223 (Flash Disable) enabled.

Optional step: Assign RAD greeting sets via the Administrator's Mailbox

1. Access the System Administrator's mailbox.
2. Press 5 to select the System Parameters menu.
3. Press 2 to assign greeting sets to be used by each port.
4. When prompted, for each port enter a System Greeting number from 1 - 9 or a RAD greeting set number from 10 - 39.

RAD-Only Operation

All embedded voice mail ports can operate as dedicated RAD ports. However, a least one non-RAD port is required to record RAD greetings; to administer the voice mail system by telephone; and for message notification by pager. To get around this requirement, temporarily assign a System greeting set to one of the ports. Then, log into the administrator's mailbox and record the RAD greetings by telephone. After the greetings are recorded, use Express Manager or the telephone to reassign a RAD greeting to the port.

Distribution Lists

Distribution lists can be set up for global (system-wide) use to make it easier to send messages to a group of people. Users can also set up distribution lists for their personal use.

The system provides forty-nine programmable global lists numbered 001 to 049. A fiftieth list (the Broadcast list) numbered 000 contains all programmed extension-type mailboxes. The system creates the list automatically; it cannot be modified. All mailbox owners can use the global lists but only the administrator can change the programmable ones. Personal distribution lists are numbered 050 to 059.

Global distribution lists can be created and managed by telephone or via CDE. Personal lists can be created and managed by telephone only - see the Voice Mail User Guide for details.

Creating and Editing a Global Distribution List

To create a new global distribution list or edit an existing list via CDE:

1. In Form 51 (Distribution Lists), press the DISTRIB LIST softkey, and then enter a list number (1-49). If creating a new list, press the DISTRIB NAME softkey, type a descriptor, and then press QUIT.
2. Press the INSERT softkey to add mailboxes to the list or DELETE to delete the mailbox at the cursor position.
3. Press QUIT when done
4. Follow the next procedure to record a name for the distribution list.

To record a name for a distribution list (the name identifies the list in the telephone user interface):

1. Access the System Administrator's mailbox.
2. Press 7 for the Distribution Lists menu.
3. Enter a 3-digit distribution list number, 001 to 049.
4. Press 4 to record a name for this list.
5. Press any key to stop recording.

To create a new distribution list or to add mailbox numbers to an existing list by telephone:

1. Access the System Administrator's mailbox.
2. Press 7 for the Distribution Lists menu.
3. Enter a 3-digit distribution list number, 001 to 049.
4. Press 1 to add mailbox numbers to this list or 3 to delete them from the list.

   After you enter each mailbox number, the system confirms that the number has been added or deleted. Continue to add/delete mailbox numbers until all are entered/removed.
5. Press # when you enter the last mailbox number to stop adding/deleting.
6. Complete the preceding procedure to record a name for distribution list.

Reviewing a Global Distribution List by Telephone

To review an existing distribution list:

1. Access the System Administrator's mailbox.
2. Press 7 for the Distribution Lists menu.
3. Enter a 3-digit distribution list number, 001 to 049.
4. Press 2 to review the mailbox numbers in this list.

   Provided the list contains mailbox numbers, the system itemizes the mailboxes contained in the list by speaking the name associated with each mailbox. If no personal names are recorded, the system reads all the mailbox numbers to you.
5. Press any key to interrupt the review and return to the Distribution Lists menu.

Personal Contact Numbers

Personal Contact Numbers allow mailbox users (including users of guest and front desk-type mailboxes) to store alternate numbers where callers can contact them instead of leaving a message. Callers are prompted in the users' personal greeting to press a key to have their call transferred to the alternate number—they are never told the number. The following is a greeting that includes a prompt to contact the mailbox owner by cellular phone:

"You have reached Tom Murphy. I can't answer the phone right now. Please leave your name, number, and a short message -- I will get back to you as soon as possible. To try to reach me on my cell phone, press [2]. If you would like to speak with an operator, press [0] at any time."

Users can set up their own contact numbers by telephone or the system administrator can do it for them via CDE programming. Up to ten contact numbers can be assigned to keys [0] through [9]. Key [2] is reserved for cellular phone numbers, key [3] for fax numbers, and key [7] for pager numbers. Key [0] is usually used to transfer to the operator but it can be used for contact numbers. The same applies to the [8] key which is used to change languages in a bilingual system.

To set up Personal Contact Numbers by telephone:

1. Access your mailbox.
2. Press 8 for User Options.
3. Press 2 for Change Notification
4. Press 2 for Personal Contact Numbers.
5. Follow the prompts to assign your numbers.

Note: For external numbers, omit the digit that is normally dialed to access an outside line. The system dials the digit automatically.

6. Re-record your greeting to include prompts for the new numbers that you assigned.

To set up Personal Contact Numbers via CDE programming:

1. In Form 03, (COS Define), enable options 313 to 319 to allow trunk-to-trunk connections.
2. In Form 04 (System Options/System Timers), enable option 122, Voice mail License for Personal Contact Numbers.
3. In Form 30 (Device Interconnection), allow trunk-to-trunk interconnection.
3. In Form 49 (Voice Mail Options), enable Personal Contact Numbers.
4. In Form 50 (Mailboxes), select a mailbox to program, and then press the CONTACTS softkey.

Note: Personal Contact Numbers can be assigned to extension, guest and front desk type mailboxes only.

5. Assign telephone numbers to keys as required. For external numbers, include the ARS leading digit(s) required to access an outside line.
6. Repeat for other extensions.
7. Press the QUIT softkey when done.
8. Re-record your greeting to include prompts for the new numbers that you assigned.

Hospitality Features for Hotel or Motel Applications

The Hospitality Features provide two types of mailboxes for hotel/motel applications: guest mailbox and front desk mailbox.

A guest mailbox provides a guest with basic voice mail functionality. Guests can

- play messages
- create their own greetings
- set up their own wake-up calls.

A front desk mailbox allows the front desk attendant to administer the guest mailboxes. The front desk attendant can

- help guests access their messages
- select the language that the guest prefers to access the mail system
- set the status of a guest mailbox to checked-in
- set the status of a guest mailbox to checked-out
- move a guest's messages to another mailbox
- access a guest's mailbox
- use the standard mailbox features (the Front Desk Menu provides access to the Main Menu of the voice mailbox).

SX-200 ICP Programming

The following CDE programming is required to use the Hospitality option:

- In Form 04, System Options, enable Automatic Wake-up options 11 through 14.
- In Form 02, Feature Access Codes, program an access code for setting wake-up calls (Feature #32).
- In Form 03, COS Define, enable options number 202 and 322

Voice Mail Integration with PMS Integration

Voice Mail and PMS (Property Management System) integration automates the management of the SX-200 ICP embedded voice mail system through the hotel's property management system. PMS integrations are available for HIS and Hyatt Encore PMS systems.

The following tables summarize the structure of the protocol that SX-200 ICP and HIS and Hyatt Encore Property Management Systems (PMS) use to communicate with each other. The tables also show which protocol functions (check in, check out, message waiting status etc.) are supported. (Y = supported; N = unsupported.).

Table: HIS Protocol Message Structure

HIS Message (See below for message descriptions)	**PMS > VM**	**VM > PMS**	**0**	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**	**10**	**11**	**12**	**13**	**14**	**15**
Resynchronize	Y	Y	STX	"1"	ETX	LRC												
Check In	Y		STX	"2"	Mailbox number						ETX	LRC						
Check Out	Y		STX	"3"	Mailbox number						ETX	LRC						
Message Waiting Status		Y	STX	"4"	Mailbox number						New		Urgent		ETX	LRC		
Bad Mailbox Address		Y	STX	"5"	Mailbox number						ETX	LRC						
Query MW Status	Y		STX	"6"	Mailbox number						ETX	LRC						
Move Mailbox	Y		STX	"7"	*Mailbox number*						*Destination mailbox*						*ETX*	*LRC*
Modify FCOS	Y		STX	"8"	*Mailbox number*						*FCOS*		*ETX*	*LRC*				
Text Message MW	Y		STX	"9"	*Mailbox number*						*Unread*		*ETX*	*LRC*				
Check in with Passcode	N		STX	"A"	*Mailbox number*						*Passcode*						*ETX*	*LRC*
Change Passcode	N		STX	"B"	*Mailbox number*						*Passcode*						*ETX*	*LRC*
Merge Messages	N		STX	"C"	*Mailbox number*						*Destination mailbox*						*ETX*	*LRC*
Change ID	Y		STX	"D"	*Mailbox number*						*New ID (dial by name)*						*ETX*	*LRC*
Merge Status		Y	STX	"E"	*Mailbox number*						*Merge Status Text*						*ETX*	*LRC*

Table: Hyatt Encore Protocol Message Structure

Encore Message (See below for message descriptions)	**PMS > VM**	**VM > PMS**	**0**	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**	**10**	**11**	**12**	**13**	**14**	**15**
Resynchronize	Y	Y	STX	"1"	SP	SP	SP	SP	SP	SP	SP	SP	SP	SP	SP	SP	ETX	LRC
Check In	Y		STX	"2"	Mailbox number						SP	SP	SP	SP	SP	SP	ETX	LRC
Check Out	Y		STX	"3"	Mailbox number						SP	SP	SP	SP	SP	SP	ETX	LRC
Message Waiting Status		Y	STX	"4"	Mailbox number						New		Urgent		SP	SP	ETX	LRC
Bad Mailbox Address		Y	STX	"5"	Mailbox number						SP	SP	SP	SP	SP	SP	ETX	LRC
Query MW Status	Y		STX	"6"	Mailbox number						SP	SP	SP	SP	SP	SP	ETX	LRC
Move Mailbox	Y		STX	"7"	*Mailbox number*						*Destination mailbox*						*ETX*	*LRC*
Modify FCOS	Y		STX	"8"	*Mailbox number*						*FCOS*		*SP*	*SP*	*SP*	*SP*	*ETX*	*LRC*
Text Message MW	Y		STX	"9"	*Mailbox number*						*Unread*		*SP*	*SP*	*SP*	*SP*	*ETX*	*LRC*
Check in with Passcode	N		STX	"A"	*Mailbox number*						*Passcode*						*ETX*	*LRC*
Change Passcode	N		STX	"B"	*Mailbox number*						*Passcode*						*ETX*	*LRC*
Merge Messages	N		STX	"C"	*Mailbox number*						*Destination mailbox*						*ETX*	*LRC*
Change ID	Y		STX	"D"	*Mailbox number*						*New ID (dial by name)*						*ETX*	*LRC*
Merge Status		Y	STX	"E"	*Mailbox number*						*Merge Status Text*						*ETX*	*LRC*

Message Descriptions

Message	**Description**
Resynchronize	Request for database swap. Resynchronizes the PMS and voice mail systems in the event that either system goes down.
Check In	Indicate that a guest has checked in to a room.
Check Out	Indicates that a guest has checked out of a room.
Message Waiting Status	Indicates that the guest has unread message(s) in his/her mailbox
Bad Mailbox Address	Indicates an invalid mailbox number
Query MW Status	Request for message waiting status. Allows the front desk staff to check for messages on a per mailbox basis.
Move Mailbox	Used to transfer voice mail messages and wake-ups when a guest changes rooms.
Modify FCOS	

Used to set the language of the voice mail prompts:

"00" - use system default ("01" = North American English)
"01" - North American English
"02" - North American Spanish
"03" - North American French (Quebecois)

Text Message MW	INot used
Check in with Passcode	Not used
Change Passcode	Not used
Merge Messages	Not used
Change ID	Not used
Merge Status	Not used

Integration Features

- Automatic voice mail set-up at check-in and check-out
  Check-in: Open hotel guest mailbox on the voice mail system and remove any messages from the previous hotel guest.
  Check-out: Close the hotel guest mailbox from the voice mail system while retaining the played messages until the next check-in of this mailbox or for any number of days as determined by the system programming (see Form 49, Voice Mail Options). The unplayed messages are also retained

until the next check-in with no time limit on deletion. This is used to provide a service to the hotel guest who may want to review a message that was left in the mailbox after they have already checked out. While in a closed state (checked-out), calls forwarded to this mailbox will be redirected to the Dial 0 location.

- Automatic language selection
  If a preferred guest language is entered in the PMS and the Bilingual option is enabled in Voice Mail Prompt Language Assignment form, the mailbox is automatically set-up in the preferred language.
- Personal greeting
  Guests are allowed to record a personal greeting (not compulsory)
- Room change
  The PMS automatically sends room changes to the voice mail system. The voice mail system will then transfer voice messages along with the guest password, personal greeting etc. to the new mailbox for the new room.

Physical Connection

An RS232-to-IP serial port converter, such as the Precidia Technologies Ether232 or iPocket232 (available from the vendor or its resellers) is required to connect the customer's PMS computer to the LAN which in turn connects to the SX-200 ICP controller. Follow the instructions supplied with the converter to configure its Ethernet and serial port settings. Precidia programming is provided below. The Ethernet settings (converter IP address, Subnet Mask and Gateway IP address) to use depend on the LAN configuration. The serial port settings to use are as follows:

Setting	Value
Protocol	Telnet (cln)
Port Settings	Hyatt Oncore: 1200/7/E/1 No flow control
	HIS: 2400/7/E/1 No flow control
Connection control	Net-Linked
Terminal Type	vt-100
Local Port	0
Remote IP	SX-200 RTC IP Address
Remote Port	6830
Fallback IP	0.0.0.0
Fallback Port	0

Setting up the Precidia Ether232

1. Connect cables (power, Ethernet from Layer 2 switch to Ether232, and serial from PC to Ether232)
   Note: The serial cable connecting the PC to the Ether232 must be a Null Modem (RS232-Crossover) cable.
2. Start a terminal program on the PC (such as Hyperterminal) and set the configuration to 9600-8-N-1.
3. Press and hold the Configure button on back panel of Ether232 for several seconds.
4. Assign the Ether232 a valid IP Address with Subnet Mask and Gateway IP Address.
5. Configure the Serial Port for the Ether232 with the settings in the above table.
6. Set the remote IP address. This is the IP address of the SX-200 ICP.
7. Set the remote IP port to 6830.

8. Ensure that all remaining entries are set to zero. If they are not 0 set them to 0.
9. Save the configuration.
10. Unplug the PC from the Ether232 and connect the PMS computer in its place.
11. Press any key to establish communication with the SX-200 ICP.

Additional SX-200 ICP Programming

Enable System Option 124 (Voice mail Property Management System) in Form 04 and select the PMS protocol to use in Form 49, Voice Mail Options. The default is Hyatt Encore.

Note: Enabling Option 124 requires a system reset for the option to take effect. See Resetting the System for more information.

Record a Call

Record a Call allows the user to record an internal or external two-party conversation at their set. The recording process may start automatically or manually. If manual, the user presses a Record a Call softkey or a hardkey (depending on the set) to initiate the recording. While the conversation is recording, the user may pause, resume, save, or erase the recording.

Users of all Mitel 4000-series and 5000-series telephones (except the SUPERSET 4001 and the 5201 IP telephone) and the SUPERCONSOLE 1000 can use this feature.

The recording is stored as a recorded conversation in the user's voice mail mailbox.

See Record a Call for programming and usage information.

Mailbox Key

A Mailbox Key is the line key on a telephone or programmable key module (PKM) that has been programmed with an extension and an associated mailbox. Subscribers can use Mailbox Keys to receive notification of new messages, and to access their voice mailboxes to listen to messages. For more information, refer to Mailbox Key.

Mailbox Lockout

If this feature is enabled, the system locks the mailbox after three failed attempts to log in with an invalid password. When lockout occurs, the system plays a message that instructs the user to contact the system administrator in order to reset the password. This option is disabled by default. For more information, refer to Mailbox Lockout.

Unified Messaging

The Unified Messaging feature package enables the SX-200 ICP to manage e-mail messages using SMTP (Simple Mail Transfer Protocol) and/or IMAP (Internet Message Access Protocol). For more information, refer to Unified Messaging.

Directory Dial by Name

See Phonebook

Softkey Support

All Mitel telephones that have softkeys can use them instead of dialing codes to select voice mail menu options. For example, to listen to message, a user can press the Play Message softkey instead of dialing the digit 7. Note that softkeys are not available for all menu options.

Softkey support for voice mail requires Option 97 in Form 04 (System Options). Refer to Softkey Support for instructions on how to program voice mail softkey support. You require a Mitel Options Password to enable this option.

Auto Attendant Park and Page

Note: This mailbox option is present in the TUI Administrator but these mailbox types are not supported in this release.

Testing Voice Mail Operation

Perform the following tests to verify that voice mail is working properly.

Because system programming frequently blocks users from accessing an outside line and then dialing back in to the main number (tying up two Central Office trunks in the process), parts of this test may need to be performed from off-site. The System Administrator could call in from home and then verify the message upon arrival.

1. Call the business using an outside line. Verify that you hear the company open or closed greeting (as appropriate) and that the message is clear and understandable.
2. Call the business using several outside lines. Verify the number of rings allowed before the system answers.
3. Call the business using an outside line. When the auto atttendant answers, press 0 and verify that the system transfers the call to the attendant or other "dial 0" destination programmed in Form 19.
4. Call the voice mail hunt group using an internal line. Verify that you do not hear the company greeting. Verify the number of rings allowed before answer.
5. Call the voice mail hunt group using an internal line. Verify that you do not hear the company greeting. When the system answers, press *0 and verify that the system transfers the call to the attendant or other "dial 0" destination.
6. Set up call forward busy/no answer to voice mail on several extensions and then call them directly (not through auto attendant) to verify that the calls are forwarded to the correct mailbox.
7. Leave a message at an extension to verify that messaging waiting indicator (light) comes on.
8. (Optional) Call the voice mail hunt group from a fax terminal and verify that the Auto Attendant answers and transfers the call to the fax extension.
9. (Optional) Call an extension that has message notification enabled and verify that it is working correctly.
10. (Optional) Leave a message at an extension and verify that the message is forwarded to the user's e-mail account.

Note: Once a message has been deleted, it cannot be recovered.

Program Feature Packages

Note: References to the following SUPERSET telephones also apply to the following IP Phones:

- References to the SUPERSET 4001 apply to the Mitel 5201 IP Phone.
- References to the SUPERSET 4015 apply to the Mitel 5010, 5212 and 5215 IP Phones.
- References to the SUPERSET 4025 apply to the Mitel 5020, 5220 and 5224 IP Phones, and the Symbol MiNET Wireless Phone.

Station Message Detail Recording

Introduction

This document describes the Station Message Detail Recording (SMDR) feature of the SX-200 ICP System. The document includes a detailed description of the feature and its operational parameters.

The Station Message Detail Recording or "call detail recording" feature is an integral part of the system. It generates a descriptive call record for every incoming and outgoing trunk call made via the system. These call records can be routed to an RS-232 port for processing or printing. They allow the customer to evaluate the use of the system's trunks and determine whether the quantity and type of trunks are the most economical mix for the traffic being handled by the system. In addition, the customer can analyze the use of the trunk network by corporate personnel. Misuse can then be corrected through modifications to the toll control assignment.

++Data SMDR provides a record of each data call which is made within the system so that all data-related traffic may be analyzed.

Detailed Description - Trunk SMDR

The SMDR data collection process is initiated every time a trunk (incoming or outgoing) is seized. The collection process forms part of the system's call processing routines; as such, data is collected on a per-call basis for the duration of each call. The data is formatted into an SMDR record and is routed to a printer output port. The records can be routed directly to the output port using a reporting system.

When SMDR (outgoing and incoming) is selected, a record is generated for every trunk call regardless of the call duration, the identity of the originating party or whether the call is completed. When two or more trunks are involved in a call, a separate record is generated for each trunk, allowing each trunk to be analyzed for costing purposes. When a station which is involved in a trunk call invokes a transfer to another station, only one record is generated; however, the number of the second station appears in the SMDR call record. A call may be transferred several times but only the first and second party is recorded. If account codes are entered, a record is generated for each account code.

An SMDR record is not generated for:

- calls which encounter busy trunks,
- internal calls between stations or between a station and the Attendant,
- calls made from stations or going to trunks whose class of service includes SMDR disable.

Note: Automatic Number Identification (ANI), Custom Local Area Signaling Services (CLASS), and Dialed Number Identification Service (DNIS) information can also be reported in an SMDR trunk record. Enable COS Options 814, SMDR - Record ANI/DNIS/CLASS, and 806, SMDR - Record Incoming Calls. Please refer to the Program Features Description section for a complete operational description of ANI/DNIS and CLASS.

Trunk Call Record Buffers

The system has 200 buffers which store call record information. If all buffers become full, there are two options: overwrite the oldest record, or do not allow trunk calls until buffers are available.

Recorded Information

Each SMDR call record occupies a single 85 character line (88 if a 3-digit system identifier is included). The information which may be included in a call record is as follows:

- Call Start time and date
- Calling party
- Called party
- Call duration
- Call completion status (e.g., called number busy)
- Digits dialed on the trunk (maximum 26 digits)
- Meter pulses (optional)
- Outgoing and incoming trunk numbers
- Long calls identified (optional)
- Time to answer incoming calls
- Identifies the second station in a transfer or in a conference
- Identifies conferences and transfers
- Indicates when the Attendant was involved in the call
- ARS leading digits
- Account code (optional)
- System identifier (optional)
- ANI digits (optional)
- CLASS digits (optional)
- DNIS digits (optional)
- Room extension
- CLASS name (optional)

The definition of the data and its position in the record is given in the Summary of Fields in Trunk SMDR Records Table. The table has five columns: the first identifies the data; the second defines the data's position within the record; the third indicates the format of the data; the fourth describes the data; and the fifth provides additional notes.

A description of the call record fields detailed in the Summary of Fields in Trunk SMDR Records Table is as follows:

Note: Five-digit SMDR is described later in this section; see 5-DIGIT SMDR OPTION.

Long Call Indicator (z): This optional field contains a dash (-) for calls 5 to 9 minutes, 59 seconds, a percent symbol (%) for calls 10 to 29 minutes, 59 seconds, or a plus symbol (+) for calls of 30 minutes or more. This is useful when records are to be scanned manually.

Date (mm/dd): The date is reported numerically as a 2-digit month followed by a 2-digit day separated by a (/) slash. The year is not reported.

Start Time (hh:mmp): The start time of a call is reported in hours and minutes. System option "24 Hour Clock" determines whether a 12-hour or 24-hour format is used. The letter "p" indicates pm in 12-hour format.

Duration of Call (hh:mm:ss): The call duration is reported in hours, minutes and seconds with leading zeroes being output (maximum time that can be recorded is 18 hours, 12 minutes, 15 seconds).

Calling Party (pppp): This is the identity of the party that originated the call. If a 5-digit extension numbering plan is used, only the last four digits will be used by SMDR to identify the calling party. It may be a station, the attendant, or an incoming trunk. In a hotel/motel guest suite environment, this prime line number also represents calls made by secondary telephones associated with this prime line. The extension number of the secondary telephone is found in columns 114 to 118.

Station Number as Calling Party (cccc): A station number may be one to four digits (0-9, *, #) which are left-justified; i.e., no leading zeroes.

Attendant as Originating Party: Calls originated by the attendant which do not involve a third party report a calling party of the console directory number. If the attendant calls an outside party on behalf of a station or trunk, that station or trunk is reported as the caller but the Attendant Flag symbol (*) appears in the “Attendant was Involved” field.

Trunk Number as Calling Party (Tnnn or Xnnn): When the originating party is an incoming CO trunk, “Tnnn” appears on the record, where “nnn” is the number of the trunk. When the originating party is an incoming non-CO trunk, “Xnnn” appears in the trunk record. The “T” or “X” ensures that CO Attendant trunks may be distinguished from TIE trunks. The trunk number is the trunk ID specified during customer data entry in the Trunk Assignment tables.

Attendant (f): This 1-digit field contains an asterisk (*) when a call is originated by or initially answered by the Attendant. This flag will not appear when a call is transferred to the attendant.

Time to Answer (ttt): This is the number of seconds from the time the incoming trunk is seized until the call is answered. If the call is never answered, this field displays ***. It applies to incoming calls only. Leading zeroes are output and the field remains at 255 when an overflow is reached.

Leading Digits (up to 5 ARS leading digits): This field applies to outgoing calls. For incoming calls, this field reports Time to Answer (see above). Leading digits correspond to digits programmed in the ARS digit string form during CDE. Leading digits reported may be from one to four digits long (0-9, *, #) (only the first 4 of 5 digits are reported). The field is left-justified and space filled.

Digits Dialed on the Trunk (xxx---x): The maximum number of digits (0-9, *, #) recorded is 26. When the SMDR Meter Pulse On option is selected, this number is reduced to 20. This field does not include the trunk group access code on outgoing calls. The digits recorded are the actual digits outpulsed on the trunk after digit modification has been performed. On dial-in trunk calls, the digits dialed in on the trunk are recorded. When more than 26 digits are dialed, only the first 26 are recorded and the rest are ignored.

Meter Pulses (mmmmm): The number of reversals (i.e., meter pulses) received from an outgoing trunk can be recorded when COS Option 247 (SMDR - Record Meter Pulses) is enabled. However, the maximum number of digits dialed on a trunk that are recorded is reduced from 26 to 20. The range of the count of meter pulses is 00000 to 65535, with leading zeroes being output. Meter pulses are used most frequently in Hotel/Motel applications, where each call generates a pulse, and for outside North America, where a number of meter pulses is generated for each toll call, proportional to the distance and duration of the call. Refer to meter pulses in the Program Features section.

Call Completion Status (h) (Outgoing Calls): This field is used to report the completion status of an outgoing call in so far as the system is able to determine it. When the trunk group is programmed to receive “Answer Supervision” and a supervision is received, an “A” is reported.

Call Completion Status (h) (Incoming Calls): The system can monitor the outcome of the call and provide a comprehensive report on the call’s completion. From a dial-in trunk, but not a direct-in-line trunk, if the station or hunt group to which the call is directed is busy, a “B” is recorded. When an incoming dial-in trunk dials an invalid number and receives reorder tone, an “E” is reported. The field is blank for incomplete calls. A “T” is reported if the incoming trunk is answered with TAFAS. When an incoming call is forwarded by the Attendant to a busy station, a “B” appears in the call completion status field. Recall no answer is indicated by an “N” or an “R”; an “N” indicates that a transferor did not answer a recall, and an “R” indicates that the transferor did answer the recall.

Speed Call or Call Forward Flags (C,R, or F): This field contains a “C” when the number is speed dialed and an “F” when the call is forwarded through the external call forward feature. Otherwise, “R” will appear (routed via ARS, the default for outgoing trunk calls) unless the trunk is programmed as a Private, DTS or CO line, in which case no "R" appears.

Called Party (qqqq): This is the party to whom the call is directed. It may be a station number, the Attendant, or the trunk number for outgoing calls. The format in which the called party is output is identical to that used for the calling party. See Calling Party (pppp). For incoming calls to the attendant, the called party is recorded as the attendant unless the attendant transfers it to a station. For direct-in lines, it would be the station number. On outgoing calls handled by the attendant, the called party would be the trunk number which the call went out on.

Transfer/Conference Call (K): This field identifies calls that involve three or more parties. It contains a “T” for supervised transfers, “X” for unsupervised transfers (i.e., dead transfer or transfer into busy) and a “C” for 3-way conversations and conferences.

Third Party (rrrr): The third party field contains the number of the station to which a trunk call has been transferred. When several transfers take place during a trunk call, the first party is the only one reported. The format is identical to that of the Calling Party (pppp).

Account Code (aa...a): Account codes are typically used to charge the cost of calls either to internal departmental cost centers or to project accounts for billing to specific projects. An extension may have the option or be forced to enter an account code for trunk calls. The account code may be 1-12 digits (the default value is six digits). If COS Option 246 (SMDR - Extended Record) is enabled, up to 12 digits of the Account Code are recorded. Otherwise, only the first eight digits of the account code are recorded.

System Identifier (iii): This optional 3-digit field may contain values from “000” to “999”. “000” indicates that no identifier has been entered. The system identifier is programmed at the System level and is printed only if COS Option 246 (SMDR - Extended Record) is enabled.

ANI/CLASS Digits (nn....n): This optional field may contain up to 10 digits. If COS Options 806 (SMDR - Record Incoming Calls) and 814 (SMDR - Record ANI/DNIS/CLASS) are enabled, the ANI or CLASS digits received from an incoming trunk will be recorded. If no digits are received, this field will be blank.

DNIS Digits (dd....d): This optional field may contain up to 10 digits. If COS Options 806 (SMDR - Record Incoming Calls) and 814 (SMDR - Record ANI/DNIS/CLASS) are enabled, the DNIS digits received from an incoming trunk will be recorded. If no DNIS digits are received, this field will be blank.

Room Extension: This optional field shows the prime line number of the telephone that made the call. This number identifies the secondary telephone in cases where many secondary telephones are associated to the Primary DN/Line of a primary telephone in a guest suite.

CLASS Name: This optional field may contain up to 15 characters. If COS Options 246 (SMDR - Extended Record) and 814 (SMDR - Record ANI/DNIS/CLASS) are enabled, the CLASS name received from an incoming trunk will be recorded. If no name is received, this field will be blank.

Fields in Trunk SMDR Records

Table: Summary of Fields in Trunk SMDR Records

Name	Column	Format	Definition	Notes
Long Call (Optional)	1	z		

- = 5-9 min
% = 10-29 min

+ = 30 or more min				
Date	2-6	mm/dd	mm = Month dd = Day	mm = 01-12 dd = 01-31
Spacer	7			
Start Time	8-13	hh:mmp		

hh = Hours
mm = Minutes

p = PM (12-hour clock)	00-2300-59			
Spacer	14			
Duration of call	15-22	hh:mm:ss	hh:mm:ss = duration in hours:minutes:seconds	

hh = 00-18
mm = 00-59

ss = 00-59 maximum = 18:12:15				
Spacer	23			
Calling Party	24-27	pppp		

cccc = Extension Number Tnnn = Trunk Number (CO)

Xnnn = Trunk Number (Non-CO) mmmm = Attendant Console Directory Number

c = 0-9, *, #nnn = 001-200

m = 0-9, *, #				
Spacer	28			
Attendant	29	f		

* = Attendant

-- = Attendant not involved	Attendant answered or initiated the call, then transferred it to an extension.			
Leading Digits	30-33	cccc	cccc = Access Code (outgoing and tandem calls only)	c = 0-9, *, #, left-justified
Time to answer (Alternate)	30-32	ttt		

ttt = time in seconds

*** = Call unanswered	ttt = 000 - 255, leading zeroes output, incoming calls only			
Digits dialed on the trunk	34-59	xx x	Up to 26 (20 if metering) digits dialed on the trunk	x = 0-9, *, or #; private speed call numbers are not recorded
Meter (Optional)	55-59	mmmmm	mmmmm = number of meter pulses	mmmmm = 00000 to 65535, leading zeroes output
Call Completion Status	60	h		

A = Answer Supervision
B = Callee is Busy

E = Caller Error T = TAFAS answered R = Incoming call recalled and was answered by transferer N = Incoming call recalled and was not answered by transferer

Outgoing/Incoming

Direct/Dial-In Incoming/Dial-In Incoming Incoming/Outgoing		
Speed Call or Call Forward Flags	61	C,R, or F

C = Number was Speed called (ARS implied)

F = Forwarded through External Call Forward R = default (ARS implied)	Outgoing - All trunk calls are ARS by default.	
Called Party	62-65	qqqq

cccc = Extension Number

Tnnn = Trunk Number (CO) Xnnn = Trunk Number (Non-CO) mmmm = Attendant Console Directory Number

c = 0-9, *, #

nnn = 001-200 m = 0-9, *, #				
Transfer/ Conference Call	66	K		

T = Supervised Transfer
X = Unsupervised Transfer

C = 3-Way or Conference				
Spacer	67			
Third Party	68-71	rrrr		

cccc = Extension Number

Tnnn = CO Trunk Number Xnnn = Non-CO Trunk Number mmm = Attendant

c = 0-9, *, #

n = 001 - 200 m = 0 - 9, *, #				
Spacer	72			
Account Code (Optional)	73-84	aa .... a	Length of 1 to 12 digits	a = 0-9, space-filled
Spacer (Optional)	85			
System Identifier (Optional)	86-88	iii	Programmed at System level	

i = 0-9

iii = 000-999 000 = no code entered				
Spacer (Optional)	89			
ANI/CLASS Digits (Optional)	90-99	nn .... n	Up to 10 digits from an incoming ANI/DNIS or CLASS trunk	n = 0-9, *, #
Spacer (Optional)	100-102			
DNIS Digits (Optional)	103-112	dd .... d	Up to 10 digits or the last 10 digits from an incoming ANI/DNIS trunk	d = 0-9, *, #
Spacer (Optional)	113			

Room Extension	114-118		The prime line number of the telephone that made the call.	Requires COS 246, SMDR - Extended Record enabled.
CLASS Name (Optional)	114-128	15 char's	Up to 15 characters from an incoming CLASS trunk	

Call Timing

The timing functions which apply to incoming and outgoing calls are described as follows:

Incoming Call - Three timing aspects of an incoming call are recorded on an SMDR call record: the date, the time taken for the called party to answer and the duration of the call. The time to answer is the difference between the time when the called device is seized and the time when the called party answers. The duration of the call is the difference between the time when the call is answered and the time when the call is released; i.e., call cleardown.

Outgoing Call - For an outgoing call, the date, the call start time and the call duration are recorded on an SMDR call record. The call start time is recorded as either the time when the called device is seized, or, in the case of answer supervision, the time when the called device answers. Call answer is determined by an answer supervision signal provided by the trunk. The call duration is the difference between the time when the call is answered and the time when the call is released; i.e., call cleardown. To be recorded, calls must either be answered or exceed the duration specified by the Pseudo Answer Supervision Timer, unless SMDR is specifically programmed. Otherwise; see SMDR Programming and Control.

Trunk SMDR Call Record Examples

Examples of typical trunk SMDR call records, excluding account codes and system identifiers, are shown below:

2-PARTY OUTGOING CALL

```
         1         2         3         4         5         6         7         8         9
123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
-06/13 11:42 00:08:29 214   9   16135552122             ART054                    000

                                                                                 CC0892
```

On June 13th at 11:42 AM, extension 214 obtained trunk number 54 and dialed “1-613-555-2122”. Answer supervision was provided. The conversation lasted 8 minutes, 29 seconds.

2-PARTY OUTGOING CALL

```
         1         2         3         4         5         6         7         8         9
123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
 05/17 10:51 00:01:52 213   9  201                      ARX082                    000

                                                                                 CC0893
```

On May 17th at 10:51 AM, extension 213 accessed an identified trunk group, then dialed 201 to obtain an extension in the other PBX. The other PBX provided answer supervision and the conversation lasted 1 minute, 52 seconds. The trunk number was 082.

2-PARTY INCOMING CALL

```
          1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
 01/30  15:10   00:02:22   T102   008 201                              201                     000
```

CC0894

On January 30th at 3:10 PM, incoming Direct-in Trunk Number 102 rang in to extension 201. The extension answered after 8 seconds and they talked for 2 minutes, 22 seconds.

2-PARTY INCOMING CALL

```
          1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
 03/12  09:11   00:01:12   X116   078 63                               224                     000
```

CC0895

On March 12th at 9:11 AM, Dial-in Tie Trunk 116 dialed hunt group with access code “63”. Extension 224 answered after 7 seconds, and the conversation lasted 1 minute, 12 seconds.

ATTENDANT-HANDLED CALL - OUTGOING TRUNK

```
          1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
+01/30  15:27   00:35:11    201    *9   1655556951                    ART052                   000
```

CC0896

On January 30th, extension 201 dialed the attendant and asked for an outside line. The attendant dialed 1-654-555-6951. At 3:27 PM, the other party answered and the conversation lasted 35 minutes, 11 seconds. Trunk number 52 was used.

ATTENDANT-HANDLED CALL - INCOMING TRUNK

```
          1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
 04/05 13:42    00:00:31   T090   009                                 1234                     000
```

CC0897

On April 5th at 1:42 PM, trunk 90 rang into the attendant. After 9 seconds, the attendant at 1234 answered. The trunk party spoke to the attendant for 31 seconds, then hung up.

CALLING EXTENSION TRANSFER CALL

```
          1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
 04/02  09:36   00:04:55    103    9    5552122                       RT162T 100               000
```

CC0898

On April 2nd at 9:36 AM, extension 103 dialed ARS leading digit code followed by 555-2122. The called party answered and after conversing, the caller transferred the called party to extension 100. After further conversation, extension 100 hung up. The total period for both conversations was 4 minutes, 55 seconds. Trunk number 162 was used for the call.

CALLED EXTENSION TRANSFER CALL

```
          1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
 03/12  07:42   00:03:06    T162   *003                                241T    215              000
```

CC0899

On March 12th at 7:42 AM, trunk 162 rang the console and requested to speak to extension 241. The attendant took 3 seconds to answer the call. After speaking to extension 241, the latter extension then transferred the call to extension 215. The total conversation lasted 3 minutes, 6 seconds.

ANALOG NETWORKING - OUTGOING CALL

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
  01/16   09:52   00:02:21   2664   9   7082664#95551022           RX124
```

On January 16th at 09:52 AM, extension 2664 accessed a tie trunk and dialed 95551022. The call lasted 2 minutes and 21 seconds. The Analog Networking feature caused the analog network access code (708 as defined in Modified Digits Table) and the calling extension (2664) to be passed onto the trunk.

ANALOG NETWORKING - INCOMING CALL

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
  01/13   15:26   00:01:18   4300   004  *94300#2777                 2777
```

On January 13th at 3:26 PM, extension 2777 received a call from an incoming tie trunk. The incoming digits dialed on the trunk were 2777 (the called extension); the Analog Networking feature added *9 (network feature access code as defined in CDE) and 4300 (the calling party identification for that tie trunk). The call lasted 1 minute, 18 seconds.

CLASS SMDR Call Record Examples

Examples of typical CLASS SMDR call records, excluding account codes and system identifiers, are shown below:

2-PARTY INCOMING CALL WITH CLASS NAME AND NUMBER

```
         1         2         3         4         5         6         7         8
12345678901234567890123456789012345678901234567890123456789012345678901234567890
 04/10  03:44P  00:01:04  T002   006  *5925089*                     1506
     001  5925089          Out-of-Area
                                                                          CC0889
```

On April 10th at 3:44 PM, incoming Direct-in trunk Number 002 was accessed by a caller with Listed directory number 592-5089 with name "Out-of-Area". The call rang to extension 1506, which answered after 6 seconds, and the conversation lasted 1 minute and 4 seconds.

2-PARTY INCOMING CALL WITH TRANSFER FROM CLASS TRUNK

```
         1         2         3         4         5         6         7         8
12345678901234567890123456789012345678901234567890123456789012345678901234567890
 03/30  03:44P  00:03:25  T008   005  *5922122*                     1506T 2201
     001 5922122           J.SMITH
                                                                          CC0890
```

On March 30th at 3:44 PM, incoming Direct-in trunk number 008 was accessed by a caller with listed Directory number 592-2122 and name "J.SMITH". The call rang to extension 1506 and was answered after 5 seconds. Extension 1506 then did a supervised transfer to extension 2201. The total call duration was 3 minutes and 25 seconds.

2-PARTY INCOMING CLASS TRUNK CALL WHICH WAS TRANSFERRED

```
         1         2         3         4         5         6         7         8
12345678901234567890123456789012345678901234567890123456789012345678901234567890
 03/30  08:46P  01:10:18  T132  004 *5925000*                 7000X  1501
     001 5925000          L.PASCUAL-PEREZ
                                                                          CC0891
```

On March 30th at 8:46 PM, incoming Direct-in trunk number 132 was accessed by a caller with listed Directory number 592-500 and name L.PASCUAL-PEREZ". The call rang to extension 7000 and was answered after 4 seconds. Extension 7000 then did an unsupervised transfer to extension 1501. The call was established for 1 hour, 10 minutes and 18 seconds. The name in this example shows the maximum allowable number of characters (15).

5-Digit SMDR Option

For those environments where a five digit record of parties in a call is required, the calling, called, and transferred parties may be identified by five digits instead of four digits. The Trunk SMDR record is modified as detailed in the SMDR Fields Changed for 5- digit Trunk SMDR Table. The remainder of the record remains the same as for four digit numbers. System Option 08, Five Digit SMDR, must be enabled; otherwise, regular 4-digit trunk SMDR is in effect.

Calling Party (ppppp): This is the identity of the party that originated the call. It may be a station, the attendant, or an incoming trunk, as described below.

Called Party (qqqqq): This is the party to whom the call is directed. It may be a station number, the attendant, or the trunk number for outgoing calls. The format in which the called party is output is identical to that used for the calling party. See Calling Party (ppppp).

Third Party (rrrrr): The third party field contains the number of the station to which a trunk call has been transferred. When several transfers take place during a trunk call, the first party is the only one reported. The format is identical to that of the Calling Party (ppppp).

Table: SMDR Fields Changed for 5- digit Trunk SMDR

Name	Columns	Format	Definition	Notes
Calling Party	24-28	ppppp	ccccc = extension numberTnnn = trunk number (CO) Xnnn = trunk number (Non-CO) mmmmm = attendant console aaaaa = agent ID number	ccccc = 0-9, *, # nnn = 001-200 mmmmm = 0-9, *, # aaaaa = 0-9, *, # left-justified
Called Party	62-66	qqqqq	ccccc = extension number Tnnn = trunk number (CO)	

Xnnn = trunk number (Non-CO)
mmmmm = attendant console
aaaaa = agent ID number
ppppp = ACD path number

ccccc = 0-9, *, #

nnn = 001-200
mmmmm = 0-9, *, #
aaaaa = 0-9, *, #
ppppp = 0-9, *, #
left-justified

Transfer/ Conference Call	67	K

T = supervised transfer

X = unsupervised transfer
C = 3-way or conference

Third Party	68-72	rrrrr

ccccc = extension number

Tnnn = CO trunk number
Xnnn = non-CO trunk number
mmmmm = attendant
aaaaa = agent ID number
ppppp = ACD path number

ccccc = 0-9, *, #

n = 001 - 200
mmmmm= 0 - 9,*, #
aaaaa = 0-9, *, #
ppppp = 0-9, *, #
left-justified

Note: The changes in data for columns 28, 66, 67, and 72 may affect call costing machines connected to the PBX, unless the call costing machines are programmed for these new fields.

Examples of typical trunk SMDR call records (with 5-digit fields) are shown below:

OUTGOING CALL WITH CONFERENCE

```
0         1         2         3         4         5         6         7         8         9
123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
  06/13  11:42    00:08 29   1701   9    5562122                     RT001  C10065
```

On June 13th at 11:42 AM, extension 1701 dialed 555-2122 via trunk 001 and added extension 10065 to make a conference call. The conversation lasted 8 minutes, 29 seconds.

OUTGOING CALL PLACED VIA ATTENDANT

```
0        1         2         3         4         5         6         7         8         9
123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
 05/17 10:51 00:01:52 10065 *9   5552122                           RT002
```

On May 17th at 10:51 AM, extension 10065 had the attendant place a call to 555-2122 via trunk 002. The conversation lasted 1 minute, 52 seconds.

INCOMING CALL WITH TRANSFER BY ATTENDANT AND THEN BY CALLED EXTENSION

```
0        1         2         3         4         5         6         7         8         9
123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
 04/02 09:36   00:04:55 T002 *004                             10065T 1701
```

On April 2nd at 9:36 AM, a call is received on trunk 002 and answered by the attendant. The call is transferred to extension 10065, who then transfers it to extension 1701. The total period for both conversations was 4 minutes, 55 seconds.

Detailed Description - Data SMDR

The Data SMDR feature allows the printing out of detailed records for external and internal data calls in a format similar to existing Trunk SMDR printouts.

The major differences between a Data SMDR record and a Trunk SMDR record are: the month and day are separated by a dash (-) rather than a slash (/) and the “Digits Dialed on a Trunk” field is replaced by three fields: Data Call Indicator, Type of Data Call, and Reason for Disconnect. When an external data call uses a pooled modem to interface the internal data device to analog trunks, the Data SMDR record shows the call between the data device and the pooled modem, while the Trunk SMDR record shows the call between the trunk and the pooled modem. An internal modem pooling call shows the Data SMDR record between the pooled modem and the data device.

The Data SMDR function operates with the trunk SMDR feature in the system. Its printouts may be directed to a specific printer port by programming it in CDE Form 34, Directed IO. The Data SMDR record format is such that an application processor can print the records with other call records but not cost them.

Recorded Information

Each Data SMDR call record occupies a single line of 85 characters (88 if it includes a 3-digit system identifier). The information included in a Data SMDR record includes:

- Long Call Indicator (optional)
- Date
- Call Start Time
- Duration of Call
- Calling Party
- Calling Party Disconnect
- Data Call Identifier
- Data Call Type
- Party Which Caused the Disconnect
- Call Completion Status
- Indication If Call Was Speed Dialed
- System Identifier
- Called Party
- Called Party Disconnect
- External Indicator
- Trunk SMDR Indicator
- Account Code

A summary of the fields printed in a Data SMDR record is shown in the Summary of Fields in a Data SMDR Record Table. Many of these fields are the same as in a Trunk SMDR record.

A description of the fields printed in a Data SMDR record follows:

Long Call Indicator (z): This optional field contains a dash (-) for calls of 5 to 9 minutes, 59 seconds, a percent symbol (%) for calls of 10 to 29 minutes, 59 seconds, a plus symbol (+) for calls of 30 or more minutes, or a space ( ) for durations less than five minutes. This is useful when records are to be scanned manually.

Date (mm-dd): The date that the data call was initiated is reported numerically as a 2-digit month followed by a 2-digit day separated by a (-) dash. The year is not reported.

Start Time (hh:mmp): The start time of a call is reported in hours and minutes. System Option “24 Hour Clock” determines whether a 12-hour or 24-hour format is used. The letter “p” indicates pm in a 12-hour format.

Duration of Call (hh:mm:ss): The call duration is reported in hours, minutes and seconds with leading zeroes being output (maximum time that can be recorded is 99 hours, 59 minutes, and 59 seconds).

Calling Party (ppppp): This is the identity of the party that originated the call. A 5-digit extension numbering plan is used. The extension number is a data device (either a data station or a pooled modem) originating a data call. The pooled modem’s extension number is for an incoming modem pooling call. It may be a station or an incoming trunk, as described below:

Station Number as Calling Party (ccccc): A station number may be one to five digits (0-9, *, #). If less than five digits, they are left-justified; i.e., no leading zeroes.

Trunk Number as Calling Party (Tnnn or Xnnn): When the originating party is an incoming CO trunk, “Tnnn” appears on the record, where “nnn” is the number of the trunk. When the originating party is an incoming non-CO trunk, “Xnnn” appears in the trunk records. The “T” or “X” ensures that CO Attendant trunks may be distinguished from TIE trunks. The trunk number is the trunk ID specified during customer data entry in the Trunk Assignment tables.

Calling Party Disconnect: The carat character (^) indicates that the calling party caused the call to be disconnected.

Data Call Indicator: (DATA) indicates a Data SMDR record.

Type of Data Call: This field has four possible values: DTRX, ADL, HM, or ACD. HM indicates that a Monitor HM session originated from DTRX occurred. ACD indicates that a Monitor ACD session originated from DTRX occurred. This field may be left blank if the data call is an incoming or internal modem pooling call without DTRX access.

Reason For Disconnect: This field reports the reason the data call was terminated. The Data Call Disconnect Reasons Table gives possible reasons for disconnecting a data call.

Call Completion Status: Two values are available: B (Busy) or E (Error).

The busy flag “B” is set if the called party is busy and the caller queues for completion to the called data device. If the caller terminates while in the queue, the busy flag will be set. If the call is completed after being queued, the busy flag will not be set.

The error flag “E” is set if the error was “user preventable”. User preventable errors include:

- dialing the extension of a non-data device,
- originator’s baud rate not compatible with destination,
- originator, programmed for autobaud, not entering the auto-baud character within 30 seconds,

Speed Call Indicator: A flag is set if the data call is dialed via system abbreviated dial or personal speed call on a Mitel telephone.

Called Party (qqqqq): This is the party to whom the call is directed. It may be a 5-digit data station number or a pooled modem number for outgoing modem pooling calls (the called party may not appear in a Data SMDR record which has an error flag set). The format in which the called party is output is identical to that used for the calling party.

Called Party Disconnect: The carat character (^) indicates that the called party caused the call to be disconnected.

External Data Call Indicator: "EX" is displayed in this field when the data call involves a pooled modem and a trunk, regardless of the direction of the call (incoming or outgoing). This indicator is not dependent on the existence of a trunk SMDR record.

Only external data calls involving trunks and a pooled modem are flagged. Internal data calls using a pooled modem are not indicated in this field.

Trunk SMDR Indicator: This field displays an asterisk (*) when there is a trunk SMDR record associated with this data call. This facilitates the matching of data SMDR records with SMDR records.

Speed Call or Call Forward Flags (S): This field contains an "S" when the number is speed dialed.

Account Code (aa...a): Account codes are typically used for security purposes to charge the cost of calls either to internal departmental cost centers or to project accounts for billing to specific projects. The account code may be 1-12 digits (the default value is six digits).

System Identifier (iii): This optional 3-digit field may contain values from "000" to "999". "000" indicates that no identifier has been entered. The system identifier is programmed at the system level and is printed if COS Option 246 (SMDR - Extended Record) is enabled.

Fields in a Data SMDR Record

Table: Summary of Fields in a Data SMDR Record

Field	Column	Format	Definition
Long Call (Optional)	1	z	- = 5 to 9 minutes, 59 seconds % = 10 to 29 minutes, 59 seconds + = 30 or more minutes
Date	2-6	mm-dd	mm = Month dd = Day
Spacer	7		
Start Time	8-13	hh:mm	hh = Hours mm = Minutes
Spacer	14		
Duration of Call	15-22	hh:mm:ss	hh = Hours mm = Minutes ss = Seconds
Spacer	23		
Calling Party	24-28	ppppp	ppppp = Extension Number
Calling Party Disconnect	29	^	Indicates that the calling party caused the disconnect.
Spaces	30-33		
Data Call Indicator	34-37	DATA	Flags a data call.
Spacer	38		
Type of Data Call	39-42	XXXX	XXXX = DTRX or ADL or ACD or HM or blank (blank only for incoming or internal modem pooling call).
Spaces	43-44		

Table: Summary of Fields in a Data SMDR Record

Field	Column	Format	Definition
Reason for Disconnect	45-58	14 char's	14 character explanation - see the Data Call Disconnect Reasons Table
Spacer	59		
Call Completion Status	60	h	B = Callee is busy E = Caller error
Speed Call Indicator	61	S	S = Number was speed called
Called Party	62-66	qqqqq	qqqqq = Extension Number
Called Party Disconnect	67	^	Indicates that the called party caused the disconnect.
Spacer	68		
External Indicator	69-70	EX	Indicates this was an external data call.
Trunk SMDR Indicator	71	*	Indicates that there is an associated Trunk SMDR report.
Spacer	72		
Account Code	73-84	aa. . . a	a = 0-9 space filled; length of 1 -12 digits
Spacer	85		
System Identifier (optional)	86-88	iii	i = 0 to 9 iii = 000 to 999 000 = no code entered

Data Call Disconnect Reasons

Table: Data Call Disconnect Reasons

Reason	Explanation
ADL DISCONNECT	ADL disconnect code dialed.
ANSWER: DTR HI	The device is programmed for Auto Answer and has DTR high.
ANSWER: DTR LO	The device is not programmed for Auto Answer and has DTR low. This may appear on an originator also.
ATTN BUTTON	Call/attn button on the DATASET or call/attn key on digital SUPERSET was pressed.
ATTN CHARACTER	Programmed attention character detected.
BREAK DETECTED	Break character detected.

Table: Data Call Disconnect Reasons

Reason	Explanation
BUSIED OUT	The destination data device is busied out.
BUSY	The called party is already involved in a call. If the busy indicator (column 60) does not accompany this reason, then the state of the device may be transient. Try again. If this reason persists with no busy indicator, reset the device.
CALL DUR OVFLW	The duration of the call timer has overflowed. The call has not been terminated (the SMDR record has been printed) but this is to inform the system that there has been a call of long duration (exceeding the timer limit of 99 hours, 59 minutes, 59 seconds). It might be advisable to check with the originator.
CARD REMOVED	The digital line card was removed.
DIALER HANGUP	DATA originator hangs up the voice line before the data call was connected.
DISC BUTTON	Disconnect button on the DATASET or data disconnect key on digital SUPERSET was pressed.
DTR DROPPED	DTR for one or both parties dropped for longer than the programmed interval.
HDLC DISC	Communication has been lost to the DATASET.
INACTIVITY	Session activity timer expired; refer to CDE Form 11.
INCOMPATIBLE	The two data devices are not compatible; one is synchronous, the other is asynchronous.
INTERCONNECT	The two data parties are prevented from being connected by the interconnection table.
INVALID ACCT	The originator used an invalid account code when making a data call.
INVALID DEVICE	The pooled modem data set is not a DATASET 2102 or a DATASET 2103. The attached device is not a valid DATASET; possibly it is a digital SUPERSET. If the device is valid, reset the device.
INVALID NUMBER	The originator of the data call dialed a wrong number OR the originating data device has an invalid hotline number programmed.
LINK ABORT/ LINK FAILURE	An error has occurred in the data link layer of the DATASET communication. Repeated occurrences may indicate faulty hardware, wiring or electrical interference.
MODEM HUNG UP	The pooled modem hung up.
NO ACCESS	The user failed to enter an access code when System Option Forced Account Code is on.
NO ADAPTER	The pooled modem is missing its modem adapter. This reason may also appear if the adapter is not a Mitel Modem Adapter.
NO ANSWER	The dialed destination has not responded to the programmed auto-answer stimulus. Check the Data Circuit Descriptor; the called modem did not respond

Table: Data Call Disconnect Reasons

Reason	Explanation
	with answer tone.
NO AUTOBAUD	The originator, who is programmed for autobaud, did not enter a carriage return.
NO CHANNEL	There are no available channels. This would indicate a busy system.
NO SEIZE ACK	The trunk used in this outgoing data call did not receive a ‘seize ack’ from the far end telephone office.
ORIGINATE ONLY	The called party has the “Originate Only” Class Of Service enabled.
PM: DCD LO	The pooled modem is programmed to have DCD as “Communication Established Indicator” and DCD has NOT come HIGH 15 seconds after the offhook.
PM: DSR LO	The pooled modem is programmed to have DSR as “Communication Established Indicator” and DSR has NOT come HIGH 15 seconds after the offhook.
PM: NO OFFHOOK	The pooled modem has not gone offhook after 10 seconds of waiting. **outgoing -** DTR has been raised and the MI/MIC leads have been toggled with a 2.5 sec ON, 2.5 sec OFF cycle. **incoming -** RI has been detected, DTR has been raised.
PM: RI LO	The pooled modem has had ringing voltage applied to its analog side for 10 seconds and has not detected RI on the digital side. Possible faulty RS-232 connection between the modem and the DATASET.
POWERED OFF	The dialed destination is not powered up.
RECEIVER FAIL	At some point in the data call, especially when queuing is involved, a receiver was not available or did not function as expected.
SET FAILURE	The set failed, dropping the call. See Maintenance Logs for details.
SET POWER UP	The DATASET went through a power up sequence. This occurs if there is a power supply problem to the dataset.
SET UNPLUGGED	Twisted pair connection broken or power removed.
SETUP FAILURE	The call has failed during setup. The cause is unknown and probably obscure. In this case it is best that the user try the call again. If this persists, the DATASET should be reset. Further occurrence of this reason code may indicate a faulty data device or RS-232 connection.
SPEED MISMATCH	Ranges of baud rates of the two DATASETs are not compatible.
SYSTEM DISC	The system has disconnected for one of several reasons, none of which is a problem. - The destination is busy and the originator cannot (or did not) queue, - The connection is not allowed due to invalid tenant interconnection, - The connection is not allowed due to type of devices involved.

Table: Data Call Disconnect Reasons

Reason	Explanation
TENANTING	The two data parties are in different tenant groups.
UNDECODABLE	Reason unknown.

Data SMDR Call Record Examples

Examples of typical Data SMDR call records are shown below:

DTRX CALL FROM ANOTHER SET WHICH PRESSED BREAK KEY

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
 03-13  11:25  00:01:24  1411 ^      DATA DTRX   BREAK DETECTED     1410
```

On March 13th at 11:25, an originating DTRX call from 1411 was answered by 1410. Extension 1411 pressed the BREAK key, disconnecting the call. The call lasted 1 minute, 24 seconds.

DTRX CALL NOT ANSWERED - TERMINAL POWERED DOWN

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
 03-13  11:25  00:00:04  1411 ^      DATA DTRX   ANSWER: DTR LO B  1402
```

On March 13th at 11:25, extension 1411 dialed extension 1402 (which was busy); extension 1411 was queued. Extension 1411 dropped DTR (terminal powered down).

DTRX CALL TO A WRONG OR INVALID NUMBER

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
 03-13  11:25  00:00:03  1411 ^      DATA DTRX   INVALID  NUMBER E 1412
```

On March 13th at 11:25, extension 1411 called extension 1412, which was an invalid number. The call lasted 1 minute, 3 seconds.

DTRX CALL TIMES OUT AFTER BEING ESTABLISHED

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
 03-13  11:25  00:01:01  1411       DATA DTRX   INACTIVITY          1412  ^
```

On March 13th at 11:25, Extension 1411 called Extension 1412. The call timed out after one minute.

DTRX CALL DROPPED WHEN DATASET UNPLUGGED

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
 03-13  11:25  00:00:30  1411       DATA DTRX   SET UNPLUGGED       1412  ^
```

On March 13th at 11:25, extension 1411 called extension 1412. Extension 1412 removed power from its dataset. The call lasted 1 minute, 30 seconds.

DTRX CALL DROPS DTR

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
 03-13  11:25  00:00:08  1411     DATA DTRX  DTR DROPPED      1412  ^
```

On March 13th at 11:25, extension 1411 called extension 1412. Extension 1412 powered off its terminal and dropped DTR. The call lasted 8 seconds.

CALL ANOTHER SET AND PRESS THE ATTN KEY

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
 03-13  11:25  00:01:24  1411 ^   DATA DTRX  ATTN BUTTON      1410
```

On March 13th at 11:25, extension 1411 called extension 1410. The call was terminated when extension 1411 pressed the ATTN key. The call lasted 1 minute, 24 seconds.

DURING A DTRX CALL, THE CALLING SET PRESSED THE DISC KEY

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
 03-13  11:25  00:01:24  1411 ^   DATA DTRX  DISC BUTTON      1410
```

On March 13th at 11:25, extension 1411 called extension 1410. During the call, extension 1411 pressed the DISC key. The call lasted 1 minute, 24 seconds.

DURING A DTRX CALL, THE CALLED EXTENSION PRESSES THE DISC KEY

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
 03-13  11:25  00:01:24  1411     DATA DTRX  SET UNPLUGGED    1410  ^
```

On March 13th at 11:25, extension 1411 called extension 1410. Extension 1410 pressed the DISC key. The call lasted 1 minute, 24 seconds.

DURING A DTRX CALL, THE CALLED EXTENSION PRESSES THE BREAK KEY

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
 03-13  11:25  00:01:24  1411     DATA DTRX  BREAK DETECTED   1410  ^
```

On March 13th at 11:25, extension 1411 called extension 1410. Extension 1410 pressed the BREAK key. The call lasted 1 minute, 24 seconds.

OUTGOING DTRX CALL DISCONNECTED WHEN THE FAR END HUNG UP

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
+03-13  11:25  02:14:46  1410     DATA DTRX  MODEM HANGUP      898  ^ EX*
```

On March 13th at 11:25, extension 1410 made an outgoing DTRX call. The far end modem hung up which then caused the pooled modem to also hang up. The call lasted 2 hours, 14 minutes, 46 seconds. There is a corresponding trunk record.

OUTGOING ADL CALLER HANGS UP PREMATURELY

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
 03-13  11:25  00:00:17  1410 ^   DATA ADL   DIALER HANGUP     S898    EX
```

On March 13th at 11:25, extension 1410 made an ADL call to an abbreviated dial number which was an outgoing number. Pooled modem 898 was selected from the default pool. Before the call was completed, the ADL originator hung up. The call lasted 17 seconds.

ADL CALL DISCONNECTED BY CALLING PARTY

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
 03-13   11:25   00:01:24   1410 ^       DATA ADL      ADL DISCONNECT      1411
```

On March 13th at 11:25, extension 1410 made an ADL call to extension 1411 and disconnected the call after 1 minute, 24 seconds.

ADL CALL AND CALLED PARTY PRESSES ATTN KEY

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
 03-13   11:25   00:01:24   1410         DATA ADL      ATTN BUTTON         1412 ^
```

On March 13th at 11:25, extension 1410 made an ADL call to extension 1411. The call was terminated when extension 1411 pressed the ATTN key. The call lasted 1 minute, 24 seconds.

ADL CALL DIALED AN INVALID NUMBER

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
 03-13   11:25   00:00:01   1410 ^       DATA ADL      INVALID NUMBER    E 1412
```

On March 13th at 11:25, extension 1410 dialed an invalid number. The system disconnected the call. The call lasted 1 second.

ADL CALL CAMPS ON BUT CALLED TERMINAL IS TURNED OFF

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
 03-13   11:25   00:01:24   1410         DATA ADL      ANSWER:  DTR LO     1411 ^
```

On March 13th at 11:25, extension 1410 camped a call onto extension 1411. Extension 1411 turned its terminal off. The camp-on lasted 1 minute, 24 seconds.

ADL CALL AND CALLED PARTY UNPLUGGED DATASET LINE

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
 03-13   11:25   00:00:24   1410         DATA ADL      SET UNPLUGGED       1411 ^
```

On March 13th at 11:25, extension 1410 called extension 1411. Extension 1411 unplugged its dataset from the Digital Line Card. The call lasted 24 seconds.

ADL CALL TO ANOTHER DATASET, THEN DROPPED BY PRESSING DISCONNECT BUTTON

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
 03-13   11:25   00:01:24   1410 ^       DATA ADL      DISC BUTTON         1411
```

On March 13th at 11:25, extension 1410 established a call with extension 1411. Extension 1410 presses its Disconnect Button to disconnect the call. The call lasted 1 minute, 24 seconds.

ADL CALL - NO HANGUP AFTER PRESSING <CR>

```
0         1         2         3         4         5         6         7         8         9
123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890
 03-13   11:25   00:00:22   1410 ^       DATA ADL     SETUP FAILURE      1411
```

On March 13th at 11:25, extension 1410 made an ADL call to extension 1411 but did not hang up after pressing <CR> (Carriage Return). The call lasted 22 seconds.

Data Call Record Buffers

There are 128 buffers which are dedicated for Data SMDR. Data SMDR records can be directed to any IO port, independent of other directed IO, including trunk SMDR. The one error that can occur is a lack of available buffers which occurs when records are being queued for printing while the printer is not printing.

If the port to which the Data SMDR records are directed is not guaranteed to print (refer to CDE Form 34, Directed IO), and no Data SMDR buffers are available, the oldest records are automatically overwritten, irrespective of the state of COS Option 908, Data SMDR - Overwrite Buffer.

If the port to which the Data SMDR records are directed is guaranteed to print, and no Data SMDR buffers are available, the state of COS Option 908, Data SMDR-Overwrite Buffer will determine the action. If OVERWRITE is enabled in CDE, the oldest buffer in the queue will be overwritten; therefore, all calls are not guaranteed to be recorded. If the OVERWRITE option is disabled, data calls will be barred, due to lack of resources, once the buffers are full.

An external data call causes a Data SMDR record and a trunk SMDR record to be printed.

ACD TELEMARKETER Reporting System SMDR

General

A new SMDR record is created every time an ACD path is seized. At the completion of the call, the data is formatted and routed to an RS-232 output port. The contents of this record describe how the call progressed through the ACD path. This information is required by the *ACD TELEMARKETER* Reporting System.

If the call is transferred to an ACD path, either by the attendant or from another telephone, a second SMDR record is created. The first record is the usual incoming trunk SMDR record that shows the call being transferred to the ACD path. The second record contains details about how the call progressed through the ACD path.

If an ACD call interflows out of an ACD path without being answered, a second call record is created.

Two system options must be enabled to allow creation of SMDR reports for the *ACD TELEMARKETER* feature:

- System Option 104, Maximum ACD Agents, must be set to a non-zero value to allow programming of ACD paths.
- System Option 44, ACD Reports, must be enabled to generate the SMDR reports for the reporting system (System Option 104 must be set up first).

Recorded Information

Each *ACD TELEMARKETER* Reporting System SMDR call record occupies a single 85 character line. The information which may be included in a call record is as follows:

- Call Start time and date

- Calling party
- Called party
- Call duration
- Call completion status (e.g., called number busy)
- Long calls (optional)
- Time to answer incoming calls
- Conferences and transfers
- Second station in a transfer or a conference
- Attendant involvement in a call

The following table defines the SMDR record data and its position in the record. The table has five columns: the first identifies the data; the second defines the data's position within the record; the third indicates the format of the data; the fourth describes the data; and the fifth provides additional notes.

A description of the call record fields follows the table.

Summary of Fields in ACD SMDR Records

Name	Columns	Format	Definition	Notes
Long Call (Optional)	1	z	- = 5-9 min % = 10-29 min + = 30 or more min	
Date	2-6	mm/dd	mm = Month dd = Day	mm = 01-12 dd = 01-31
Spacer	7			
Start Time	8-13	hh:mmp	hh = Hours mm = Minutes p = pm (12-hour format)	hh = 00-23 mm = 00-59
Spacer	14			
Duration of call	15-22	hh:mm:ss	hh:mm:ss = duration in hours:minutes:seconds	hh = 00-18, mm = 00-59 ss = 00-59 maximum = 18:12:15
Spacer	23			
Calling Party	24-27	pppp	cccc = Extension Number Tnnn = CO Trunk Number Xnnn = Non-CO Trunk Number mmmm = Attendant Console Directory Number aaaa = Agent ID Number	c = 0-9, *, # nnn = 001-200 m = 0-9, *,# a = 0-9, *, #
Spacer	28			
Time to answer	30-32	ttt	ttt = time in seconds (000-255) *** = Call unanswered	leading zeroes output, incoming calls only
ACD Call Information	34-59	xxx...x	Incoming ACD Calls: P = path identifier XXXX = ACD path	Displays path of call in the ACD system (ACD path followed by agent group numbers).

Summary of Fields in ACD SMDR Records

Name	Columns	Format	Definition	Notes
			G1 = primary agent group number O1 = first overflow group number O2 = second overflow group number O3 = third overflow group number A1 = agent group number that answered call I = call interflowed YYYY = ACD path number to which call interflowed	With 5 digit SMDR, only the last four digits are recorded.
Called Party	62-65	qqqq	cccc = Extension Number Tnnn = Trunk Number (CO) Xnnn = Trunk Number (Non-CO) mmmm = Attendant Console Directory Number aaaa = Agent ID Number pppp = ACD Path Number	c = 0-9, *, # nnn = 001-200 m = 0-9, *, # a = 0-9, *, # p = 0-9, *, #
Transfer/ Conference Call	66	K	T = Supervised Transfer X = Unsupervised Transfer C = 3-Way or Conference	
Spacer	67			
Third Party	68-71	rrrr	cccc = Extension Number Tnnn = CO Trunk Number Xnnn = Non-CO Trunk Number mmmm = Attendant aaaa = Agent ID Number pppp = ACD Path Number	c = 0-9, *, # nnn = 001-200 m = 0-9, *, # a = 0-9, *, # p = 0-9, *, #
System Identifier (Optional; programmable on MX systems only)	86-88	iii	Programmed at System level	i = 0-9 iii = 000-999 000 = “No Code entered”
Spacer (Optional)	89			
ANI Digits (Optional)	90-99	nn .... n	Up to 10 digits from an incoming ANI/DNIS trunk	n = 0-9, *, #
Spacer (Optional)	100-102			
DNIS Digits (Optional)	103-112	dd .... d	Up to 10 digits from an incoming ANI/DNIS trunk	d = 0-9, *, #

The following paragraphs define each of the fields listed in the table above:

Long Call Indicator (z): This optional field contains a dash (-) for calls of 5 to 9 minutes, 59 seconds, a percent symbol (%) for calls of 10 to 29 minutes, 59 seconds, or a plus symbol (+) for calls of 30 or more minutes. This is useful when records are to be scanned manually.

Date (mm/dd): The date is reported numerically as a 2-digit month followed by a 2-digit day separated by a (/) slash. The year is not reported.

Start Time (hh:mmp): The start time of a call is reported in hours and minutes. System Option "24 Hour Clock" determines whether a 12-hour or 24-hour format is used. The letter "p" indicates pm when a12-hour format is used.

Duration of Call (hh:mm:ss): The call duration is reported in hours, minutes and seconds with leading zeroes being output (maximum time that can be recorded is 18 hours, 12 minutes, 15 seconds).

Calling Party (pppp): This is the identity of the party that originated the call. If a 5-digit extension numbering plan is used, only the last four digits will be used by SMDR to identify the calling party. It is usually an incoming trunk but may be a station or the Attendant. When the originating party is an incoming CO trunk, "Tnnn" appears on the record, where "nnn" is the number of the trunk. When the originating party is an incoming non-CO trunk, "Xnnn" appears in the record.

Time to Answer (ttt): This is the number of seconds from the time the call is presented to the ACD system until the call is answered. If the call is never answered, this field displays ***. It applies to incoming calls only. Leading zeroes are output and the field remains at 255 when more than 255 seconds are required.

ACD Call Information: (PXXXX G1 O1 O2 O3 A1 or PXXXX O1 O2 O3 IYYYY). O1, O2, or O3 only appear if they are programmed and the call overflows to these groups. If the Interflow point is not an ACD path access code, only "I" appears after the last overflow group number. If the Interflow point is an ACD path, the "I" is followed by the 4-digit path access code.

Called Party (qqqq): This is the agent to whom the call is directed. The format in which the called party is output is identical to that used for the calling party. See Calling Party (pppp). For direct-in lines (typical ACD application), it is the agent ID number. If an agent answers an incoming call, the agent ID appears in this field.

Transfer/Conference Call (K): This field identifies calls that involve three or more parties. It contains a "T" for supervised transfers, "X" for unsupervised transfers (i.e., dead transfer or transfer into busy) and a "C" for 3-way conversations and conferences.

Third Party (rrrr): The third party field contains the number of the station to which a trunk call has been transferred. When several transfers take place during a trunk call, the first party is the only one reported. The format is identical to that of the Calling Party (pppp). When a call is transferred to an ACD system, the ACD Path Access Code appears in this field.

System Identifier (iii): This optional 3-digit field may contain values from "000" to "999". "000" indicates that no identifier has been entered. The system identifier is programmed at the System level and is printed only if COS Option 246, SMDR - Extended Record, is enabled.

ANI Digits (nn .... n): This optional field may contain up to 10 digits. If COS Options 806 (SMDR - Record Incoming Calls) and 814 (SMDR - Record ANI/DNIS) are enabled, the ANI digits received from an incoming trunk will be recorded. If no ANI digits are received, this field will be blank.

DNIS Digits (dd .... d): This optional field may contain up to 10 digits. If COS Options 806 (SMDR - Record Incoming Calls) and 814 (SMDR - Record ANI/DNIS) are enabled, the DNIS digits received from an incoming trunk will be recorded. If no DNIS digits are received, this field is blank.

ACD Call Record Examples

The following subsections show examples of typical SMDR call records for various types of ACD calls. The examples omit system identifiers.

ACD CALLS ANSWERED BY AGENTS

```
0         1         2         3         4         5         6         7         8         9
12345678901234567890123456789012345678901234567890123456789012345678901234567890123456789
 07/01 08:41   00:00:08  T001  002 P2123 01 01                1101
```

CO Trunk 1 comes in on path 2123 and rings agent group 1. Call is answered by agent 1101 of agent group 1.

```
0         1         2         3         4         5         6         7         8         9
12345678901234567890123456789012345678901234567890123456789012345678901234567890123456789
 07/01 08:44   00:08:29  T002  009 P0456 05 06 05             1515
```

CO trunk 2 comes in on path 456 and rings agent group 5 and overflows to agent group 6. Call is answered by agent 1515 of agent group 5.

```
0         1         2         3         4         5         6         7         8         9
12345678901234567890123456789012345678901234567890123456789012345678901234567890123456789
 07/01 08:50   00:00:22  T003  012 P2324 01 02 02             1205
```

CO trunk 3 comes in on path 2324 and rings agent group 1 and overflows to agent group 2. Call is answered by agent 1205 of agent group 2.

```
0         1         2         3         4         5         6         7         8         9
12345678901234567890123456789012345678901234567890123456789012345678901234567890123456789
 07/01 08:52   00:00:05  T001  006 P2123 01 07                1703
```

CO trunk 1 comes in on path 2123 and rings agent group 1. Call is answered by agent 1703 of agent aroup 7 (via call pickup).

```
0         1         2         3         4         5         6         7         8         9
12345678901234567890123456789012345678901234567890123456789012345678901234567890123456789
 07/01 09:53   00:00:18  T003  005 P2324 01 02 02             1202X 1861
```

CO trunk 3 comes in on path 2324 and rings agent group 1 and overflows to agent group 2. Call is answered by agent 1202 of agent group 2. Call is transferred (unsupervised) by 1202 to 1861 (non-agent).

```
0         1         2         3         4         5         6         7         8         9
12345678901234567890123456789012345678901234567890123456789012345678901234567890123456789
 07/01 08:53   00:00:12  T001  006 P2123 01 01                1103T 1515
```

CO trunk 1 comes in on path 2123 and rings agent group 1. Call is answered by agent 1103 of agent group 1. 1103 transferred (supervised) call to agent 1515.

```
0         1         2         3         4         5         6         7         8         9
12345678901234567890123456789012345678901234567890123456789012345678901234567890123456789
 07/01 09:54   00:00:10  T001  006 P2123 01 01                  1101T 1512
```

CO trunk 1 comes in on path 2123 and rings agent group 1. Call is answered by agent 1101 of agent group 1. Agent 1101 transferred (supervised) call to path 456. Agent 1512 answered call.

ACD CALLS ANSWERED BY NON-AGENTS

```
0         1         2         3         4         5         6         7         8         9
123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890123456789
 07/01 09:01   00:00:03  T001  004 P2123 01 00                2101
```

CO trunk 1 comes in on path 2123 and rings agent group 1. Call is answered by extension 2101 (via call pickup).

```
0         1         2         3         4         5         6         7         8         9
123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890123456789
 07/01 09:02   00:00:08  T003  004 P2324 01 02 00             2678
```

CO trunk 3 comes in on path 2324 and rings agent group 1. Call overflowed to agent group 2. Call is answered by extension 2678.

ACD CALLS ABANDONED

```
0         1         2         3         4         5         6         7         8         9
123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890123456789
 07/01 09:11   00:00:10  T002  *** P0456 02
```

CO trunk 2 comes in on path 456 and rings agent group 2. Call is abandoned.

```
0         1         2         3         4         5         6         7         8         9
123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890123456789
 07/01 09:14   00:00:14  T002  *** P0456 02 03
```

CO trunk 2 comes in on path 456 and rings agent group 2. Call overflowed to agent group 3. Call is abandoned.

```
0         1         2         3         4         5         6         7         8         9
123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890123456789
 07/01 09:15   00:00:07  T002  *** P0456
```

CO Trunk 2 comes in on path 456 and the caller hung up in delay for ringback before the call is presented to the primary agent group.

ACD CALLS INTERFLOWED

```
0         1         2         3         4         5         6         7         8         9
123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890123456789
 07/01 09:04   00:00:14  T002  *** P0456 05 06 07 08 I
```

CO trunk 2 comes in on path 456 and rings agent group 5. Call overflows to agent groups 6, 7, 8 and is interflowed. Call is dropped.

```
0         1         2         3         4         5         6         7         8         9
123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890123456789
 07/01 09:05   00:00:16  T001  *** P2324 01 02 04 10 I2311
 07/01 09:05  00:00:02  T001 *009                             2311
```

CO trunk 1 comes in on path 2324 and rings agent group 1. Call overflows to agent groups 2, 4,10 and is interflowed to the attendant 2311.

CO trunk 1 rings 2311. Call is answered by 2311.

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890

 07/01 09:07   00:00:26   T002   ***  P0456 05 06 07 08 I2324
 07/01 09:07   00:00:07   T002   008  P2324 01 01                   1102
```

CO trunk 2 comes in on path 456 and rings agent group 5. Call overflows to agent groups 6, 7, 8 and is interflowed to path 2324.

CO trunk 2 comes in on path 2324 and rings agent group 1. Call is answered by agent 1102 of agent group 1.

UNSUPERVISED TRANSFER OF CALLS TO THE ACD SYSTEM

Note: Each call generates two records.

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890

 07/01 09:20   00:00:22   T003   000  P2324 01 02 02              1201X 456
 07/01 09:20   00:00:10   T003   004  P2324 05 05                 1501
```

CO trunk 3 comes in on path 2324 and rings agent group 1. Call overflowed to agent group 2 and is answered by agent 1201 of agent group 2. Agent 1201 then transferred the call to ACD path 456.

Trunk 3 comes in on path 456 and rings agent group 5. Call is answered by agent 1501 of agent group 5.

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890

 07/01 09:22   00:00:04   T003   015  P2324 01 01                 1101X 456
 07/01 09:22   00:00:02   T003   ***  P0456 05
```

CO trunk 3 comes in on path 2324 and rings agent group 1. Call is answered by agent 1101 of agent group 1. Agent 1101 then transferred the call to ACD path 456.

Trunk 3 comes in on path 456 and rings agent group 5. Call is abandoned.

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890

 07/01 09:24   00:00:04   T007  *005                              2324
 07/01 09:24   00:00:05   T007   009  P2324 01 01                 1104
```

CO trunk 7 rings the attendant, the attendant answered the call and transferred it to ACD path 2324.

Trunk 7 comes in on path 2324 and rings agent group 1. Call is answered by agent 1104 of agent group 1.

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890

 07/01 09:27   00:00:07   T008   005                              5220X 2324
 07/01 09:27   00:00:05   T008   009  P2324 01 01                 1102
```

CO trunk 8 rings extension 5220. Call is answered by 5220 and is transferred to path 2324.

Trunk 8 comes in on path 2324 and rings agent group 1. Call is answered by agent 1102 of agent group 1.

SUPERVISED TRANSFER OF CALLS TO THE ACD SYSTEM

Note: Each call generates two records.

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890

  07/01 09:51     00:00:10   T013   015                                 3201T  1456
  07/01 09:51     00:00:20   T013   009  P1456  01  01                  1201
```

CO trunk 13 rings extension 3201. Call is answered by 3201 (non-agent). 3201 dialed path 1456. Agent 1201 answered the call. 3201 transferred CO trunk 13 to 1201.

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890

  07/01 09:54     00:00:14   T008  *009                                 1456
  07/01 09:54     00:00:20   T008   010  P1456  01  01                  1104
```

CO Trunk 8 rings an attendant. Call is answered by the attendant. The attendant dialed path 1456. Call is answered by agent 1104.

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890

  07/01 09:57     00:00:15   T003   010  P2324  01  01                  1107T  1456
  07/01 09:57     00:00:30   T003   005  P1456  01  01                  1502
```

CO trunk 3 comes in on path 2324 and rings agent group 1. Call is answered by agent 1107 of agent group 1. 1107 dialed path 1456. Agent 1502 answered call.

```
0         1         2         3         4         5         6         7         8         9
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890

  07/01 09:58     00:00:12   T001   012  P2324  01  01                  1101T  1606
  07/01 09:58     00:00:15   T001   020                                 1606
```

CO trunk 1 comes in on path 2324 and rings agent group 1. Call is answered by agent 1101 of agent group 1. 1101 dialed agent 1606. Agent 1606 answered call and 1101 transferred CO trunk 1 to 1606.

ACD CALLS INCLUDING ANI/DNIS INFORMATION

Note: Assume an 80-column printer is used to record the SMDR. If a larger printer is used, the ANI/DNIS information will appear on the extreme right-hand side of the page.

```
0         1         2         3         4         5         6         7         8         9         0
12345678901234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890

  07/01 08:41     00:00:08   T001   002  P2123  01  01                  1101                           017230456    9485763
```

CO trunk 1 comes in on path 2123 and rings agent group 1. Call is answered by agent 1101 of agent aroup 1. ANI and DNIS digits received from CO Trunk 1 are respectively 017230456 and 9485763.

```
0         1         2         3         4         5         6         7         8         9         0
12345678901234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890

  07/01 08:42     00:00:15   T001   002  P2123  01  01                  1102                           2782212
```

CO trunk 1 comes in on path 2123 and rings agent group 1. Call is answered by agent 1102 of agent group 1. ANI digits received from CO trunk 1 are 2782212. No DNIS digits were received.

```
0         1         2         3         4         5         6         7         8         9         0
1234567890123456789012345678901234567890123456789012345678901234567890123456789012345678901234567890123456789
  07/01 08:43    00:00:10   T001   003  P2123  01  01                 1103                                          5922124
```

CO trunk 1 comes in on path 2123 and rings agent group 1. Call is answered by agent 1103 of agent group 1. DNIS digits received from CO trunk 1 are 5922124. No ANI digits were received.

External Devices

The data formatting of SMDR records is an integral part of the system which does not require additional hardware or hardware modification. Installation is simply a matter of connecting a compatible printer or terminal to the output port of the system.

SMDR records are output from the system in standard RS-232 format. Any RS-232 compatible device can be connected directly to the output port for the storage of records.

Printer

A summary of the parameters which must be available from an SMDR printer is as follows:

- 132 character line length
- 300 baud minimum, 300 default, 19,200 maximum
- standard ASCII character set

The baud rate of the printer port or data port is selectable from the Maintenance Terminal (refer to the RS-232 Maintenance Terminal section). Ensure the baud rate of the printer matches that selected at the Maintenance Terminal.

Tape Machine

A tape machine which is capable of operating over an RS-232 interface can be connected to the SMDR output port of the PBX.

Cabling

The printer, terminal or storage device should be located within 15.2 m (50 ft) of the RS-232 port. Failure to observe this precaution may cause the SMDR record to be corrupted.

SMDR Programming and Control

The operation of the SMDR feature is determined during initial installation of the system. Programming the SMDR feature is part of the System and Class of Service Options programming known collectively as customer data entry (CDE). The operation can be modified at any time by the maintenance person, using the CDE terminal. The CDE-Selected SMDR COS and System Options Table shows the Trunk SMDR options involved and the CDE-Selected Data SMDR COS and System Options Table shows the Data SMDR options involved. A brief description of these options is given below. A complete description of the CDE forms is given in the Program CDE section. For an outgoing call record to be generated, SMDR must be enabled in CDE Form 16, Trunk Groups. A device accessing a trunk group with SMDR enabled will generate an SMDR record unless COS Option 700, SMDR - Does Not Apply, is enabled in that device's Class of Service.

TRUNK SMDR and ACD SMDR Programming Options

The Trunk SMDR and ACD SMDR programming options are as follows:

- **Record Incoming Calls:** This COS option enables SMDR for incoming trunk calls.

Note: For modem pooling, if the Record Incoming Calls option is enabled, and the SMDR buffers become full, new incoming CO calls will receive busy tone and new incoming TIE, DID, or DISA trunks will receive reorder tone. These calls will not be accepted by the PBX.

- **Record Meter Pulses:** This COS option causes meter pulses generated by the central office to be counted and then reported in the SMDR record.
- **Drop Calls Less Than n Digits:** When this COS option is enabled, outgoing calls of less than n digits are not reported (where "n" is programmed to be between 0 and 11 and 0 disables the option).
- **Drop Incomplete Outgoing Calls:** When this COS option is enabled, incomplete outgoing calls are not recorded.

Notes:

1. If the call duration is less than the programmed value of the system timer "Pseudo Answer Supervision Timer", the call is considered an incomplete call.
2. If Trunk Circuit Descriptor Option "Far End Gives Answer Supervision" is set to YES, and ANSWER SUPERVISION is not received, an SMDR record is not generated.

    - **Extended Record:** When this COS option is enabled, the length of the SMDR record is extended from 80 to 88 columns. This allows the last four columns of 12-digit account codes and the System ID to be reported.

Note: If Cos Option 814, SMDR - Record ANI/DNIS/CLASS, is enabled and there are ANI/DNIS/CLASS digits to be printed, the extended record will be printed even if Cos Option 246, SMDR - Extended Record, is disabled.

    - **Overwrite Buffer:** When this COS option is DISABLED and all SMDR buffers are in use, outgoing calls requiring SMDR will not be allowed. When ENABLED, the OLDEST SMDR buffer waiting to be printed will be overwritten with the new outgoing call SMDR information.

Note: If the Overwrite Buffer option is disabled and the SMDR buffers become full (for example, printer out of paper), outgoing calls are prohibited until the buffers are emptied.

    - **Does Not Apply:** When this COS option is enabled, no calls will be recorded for the Class of Service in which it is enabled.
    - **Indicate Long Calls:** When this System Option is enabled, calls of 5 minutes or longer are flagged in the SMDR record.

- **24-Hour Clock:** When this System Option is enabled, the start time within the SMDR record will be recorded in 24-hour format. When disabled, the start time is recorded in 12-hour format (p = pm).
- **ACD Reports:** This System Option, when enabled, provides ACD information to be output to the reporting system as part of the SMDR reports.
- **Analog Networking SMDR:** When this System Option is enabled, the calling party field contains the caller's extension number as provided by Analog Networking, instead of the incoming trunk number. When disabled, the calling party field is not overwritten with the Analog Networking information.
- **Five Digit SMDR:** When this System Option is enabled, five digit numbers will be recorded for Calling party, Called party and Third party.
- **Record ANI/DNIS/CLASS:** To record the ANI and/or DNIS and/or CLASS digits received from an incoming trunk, COS Option 806 (SMDR - Record Incoming Calls) and COS Option 814 (SMDR - Record ANI/DNIS/CLASS) must be enabled.

CDE-Selected SMDR COS and System Options

Table: CDE-Selected SMDR COS and System Options

Option	Type	Option Number
SMDR - Extended Record	COS	246
SMDR - Record Meter Pulses	COS	247
SMDR - Does Not Apply	COS	700
SMDR - Overwrite Buffer	COS	702
SMDR - Drop Calls < n Digits (n = 0...11, disable = 0)	COS	803
SMDR - Drop Incomplete Outgoing Calls	COS	804
SMDR - Record Incoming Calls	COS	806
SMDR - Record ANI/DNIS/CLASS	COS	814
24-Hour Clock	System	01
Analog Networking SMDR	System	06
Five Digit SMDR	System	08
SMDR - Indicate Long Calls	System	28
ACD Reports	System	44
Pseudo Answer Supervision Timer	System	49

Data SMDR Programming Options

Three Class of Service (COS) options and one System Option, as listed in the CDE-Selected Data SMDR COS and System Options Table, are available with Data SMDR.

- **Data SMDR - Does Not Apply:** This option, when enabled, prevents Data SMDR records from being printed.
- **Data SMDR - Extended Record:** This option, when enabled, adds the system identifier to the Data SMDR record.
- **Data SMDR - Overwrite Buffer:** This option, when enabled, will prevent a data call from being blocked due to a lack of Data SMDR buffers. However, with buffers being overwritten, there is no guarantee of recording all data calls. If disabled, data calls will be blocked if there are no available Data SMDR buffers.
- **Data SMDR - Indicate Long Calls:** This timer, when enabled, provides an identifying character in column 1 to indicate the approximate length of the call.

Note: If the Overwrite Buffer option is disabled and the Data SMDR buffers become full, data calls are prohibited until the buffers are emptied.

CDE-Selected Data SMDR COS and System Options

Table: CDE-Selected Data SMDR COS and System Options

Option	Type	Number
DATA SMDR - Does Not Apply	COS	906
DATA SMDR - Extended Record	COS	907
DATA SMDR - Overwrite Buffer	COS	908
DATA SMDR - Indicate Long Calls	System	39

Operational Parameters

The SMDR feature is transparent to the end user. There are no operational procedures to be employed by the Attendant or station user. A summary of the operational parameters which are described earlier in this document follows.

Non-Recording Conditions

SMDR is not initiated under the following conditions:

- Busy tone is obtained by the Attendant or a station when a trunk is dialed (because all trunks in the group are busy),
- The calling or called party has a class of service which disables SMDR,
- Reorder tone is obtained by the caller,
- The Attendant intercepts a station attempting to access a trunk group,
- During a power failure condition, no SMDR records are made.

Attendant-Handled Calls

The following conditions are reported as shown when the Attendant handles a call:

- When the Attendant dials a trunk with no station or trunk involved, the calling party is the Attendant.
- Direct Trunk Access by the Attendant is reported. The Leading Digits field is left blank.

- When the Attendant answers a trunk call and does not transfer it to a station, the called party is the Attendant.
- When the Attendant dials a trunk while it has a station as its source, the calling party is reported as the station and an (*) appears in the “Attendant was Involved” field.
- When the Attendant connects a previously held station to a trunk, the calling party is the station and an (*) appears in the “Attendant was Involved” field.
- When the Attendant has a trunk as Source, and then connects a station to the trunk, the calling party is the trunk, the called party is the station, and an (*) appears in the “Attendant was Involved” field.

Incoming Calls

When SMDR is enabled for incoming calls, the following conditions are reported:

- Digits dialed on incoming DID, DISA or dial-in TIE trunks are reported in the “Digits Dialed on the Trunk” field. When the dial-in trunk dials an illegal or vacant number or hangs up before completing the number, the call is reported. The called party is the station dialed. The DISA Security Code is not reported.
- The called party is always the Attendant, except when the Attendant forwards the call to a station. The station then becomes the called party and an (*) is reported in the “Attendant was Involved” field.
- Direct-in trunks show the station number as the called party (i.e., dial-in trunks). However, the “Digits Dialed” field is blank. When the trunk is directed to a Hunt Group, the station that answered the call is reported.
- On incoming calls, an “E” is reported when the trunk hangs up while listening to reorder tone, or a “B” is reported when the trunk hangs up while listening to busy tone. A “T” is reported when the incoming call is answered with TAFAS.

Data SMDR

All internal and external data calls are recorded if this option is enabled. External data calls generate a trunk SMDR record as well as a DATA SMDR record.

Account Codes

Description

Account codes uniquely identify SMDR call records for billing purposes. Two types of account code length options (fixed length and variable length) can be programmed in the System Options form during customer data entry. A fixed length account code is preprogrammed to be of a length between 4 and 12 digits. A variable length account code can be of any length between 1 to 12 digits. Variable length account codes require a terminator digit (see Account Code Terminator). When no account code length is specified during CDE programming, the default length is six digits.

The Verifiable Account Code feature is an option of SMDR which provides unique codes with assigned COS and/or COR options. Dialing a verifiable account code (which has been assigned a COS or a COR) will override the COS and/or COR currently assigned to a telephone. A new SMDR record is generated each time a new verified account code is entered.

Account Code Terminator

The variable length account code terminator is the # digit. It is dialed at the end of a variable-length account code to indicate to the system that the account code is complete. A variable length account code cannot be dialed from rotary telephones since they have no # digit. If variable length is enabled, rotary telephones default to 6-digit account codes. The terminator is not required should a variable length account code of the full 12 digits be dialed.

Forced Account Codes

COS Option 200, Account Code, Forced Entry - External Calls, may be assigned to stations in order to bar trunk access from those stations, unless the access attempt is preceded by a valid account code. COS Option 201, Account Code, Forced Entry - Long Distance Calls, may be assigned to stations in order to bar certain long distance calls (identified in the ARS Digit Strings CDE form), unless the access attempt is preceded by a valid account code.

Forced account codes can also be assigned to DISA trunks via the COS of the special DISA trunk in order to force a validation check on incoming DISA calls. System dial tone is not returned to the caller until a valid independent account code is received.

Operation

Account Code entry at the start of an outgoing call:

- Lift handset - dial tone is returned.
- Dial the feature access code followed by the account code and terminator (#); the terminator is not required with fixed length account codes.
- Dial trunk access code (usually 9).
- Dial the outgoing number.

Account code entry during a call on a Mitle 5020 IP, 5220 IP, 5224 IP, SUPERSET 4125, SUPERSET 4025 or a SUPERSET 420 telephone:

- Press SUPERKEY.
- Press NO softkey until ACCOUNT CODE? appears.
- Press YES softkey
- Enter Account Code number.
- Press SAVE softkey.

Account code entry during a call on a SUPERSET 4150 SUPERSET 430, or SUPERSET 4DN telephone:

- Press ACCOUNT CODE softkey.
- Enter Account Code number.
- Press SAVE softkey.

Tenanting

Introduction

Tenanting is a feature which allows up to 25 tenants to share features and capabilities of the PBX. Some system features are enabled for each tenant individually through customer data entry (CDE) while other features are shared by all tenants.

General Description

Tenanting is a very powerful and flexible feature. The system may be operated as a single tenant or in a multiple customer or multiple tenant mode, by sharing services such as attendants or trunks. A typical installation may have several tenants, each with its own trunks, stations and attendant. Members of different tenants may be programmed to access each other (but not necessarily each other's features) without going through the local central office, or may be totally independent of each other. The customer (or customers) can share the economies of a larger system, such as specialized trunks, leased services, and other features. SMDR, described in the Station Message Detail Recording section, allows equitable assessment of costs among all users.

Up to 25 tenants (or customers) may be defined within the system with a maximum of 11 consoles. A maximum of 25 enhanced subattendant positions may be defined. Please refer to CDE Form 09 (Desktop Device Assignments) in Program CDE for information on the subattendant position.

Multiple Tenant Application

A typical multiple tenant application is a company which occupies a large building and has attendants and/or subattendants and/or enhanced subattendants in different areas, such as each floor or department. Some features, such as attendant recall, access the local attendant instead of the main attendant position, but other features such as WATS, ARS, Tie lines, or abbreviated dialing are available to all users.

Multiple Customer Application

A typical multiple customer application is a building served by one PBX, with telephone service for each customer (tenant) going through the PBX independently. When one customer calls another, the calling party must access one of its CO trunks and dial the other customer's listed directory number. The call will go through the CO and then appear as an incoming CO call at the called customer.

Data Users Application

The PBX tenanting features are also available for data users. A typical application is to restrict access to certain computers, printers, and other data devices to data users in one or more specific internal departments, by placing all members into the same tenant and denying all other tenants access. Any computer resources which must be available to all users can be put into a tenant group to which everyone has access.

Independent Consoles, Subattendants, Trunks, and Stations

The PBX is divided into distinct tenants; all stations, Mitel telephones, trunks, subattendant/enhanced subattendant telephones, and attendant consoles are assigned to a tenant as they are programmed into the PBX. Trunks and Dial 0 calls are programmed to ring to only their associated attendant/subattendant/enhanced subattendant position. Outgoing calls seize only trunks within their same tenant as shown in the figure below.

Figure: Tenant Connections to PBX

Tenant Interconnection

The system may be programmed to allow certain tenants to connect to each other internally within the system or to allow for sharing of services such as an attendant console or trunks.

Each tenant may or may not be allowed to connect directly to any other tenant. For example, an attendant console to be shared by tenants 1, 2, and 3 can be assigned to tenant 25 and programmed to connect internally to tenants 1, 2, and 3. Tenants 1, 2, and 3 may still be programmed to NOT connect to each other as shown in the Tenant Interconnection with Shared Console (Triple Play) Figure.

Note the exception to the case when tenant 1 can call tenant 2 but tenant 2 cannot call tenant 1. A SUPERSET 4150 telephone A in tenant 1 calls SUPERSET 4150 telephone B in tenant 2, and upon receiving no answer, sends the message to call set A. Set B can press MSG, READ MSG, and CALL softkeys, and return the call to set A, even though tenant 2 (Set B) is normally restricted from calling tenant 1 (Set A).

Figure: Tenant Interconnection with Shared Console

Identifying Tenants on Consoles and Mitel Telephone

Attendant consoles or Mitel telephones may be programmed to provide a common answering point for incoming calls to all tenants, as well as calls that are unanswered by any tenant. Each tenant may not call the other tenants or access each other's trunks, but all of their calls will be directed to the common answering point. The incoming calls may be DID, DIL, or regular CO trunks.

When recalls are answered at a common answering point, the name of the tenant that did not answer is displayed as a NO ANSWER so that the recall may be answered with an appropriate response. If the called party has a Mitel display telephone, that party's name is displayed. The calling party's trunk group may be programmed to display the company name. If the trunk has not been programmed with a name, the trunk group name is displayed. If the trunk group has not been programmed with a name, the trunk number is displayed.

The call rerouting point for all tenants may be to Mitel telephones with each tenant having its own key for reroutes, as shown in this Figure. A reroute to a busy line will camp on to that line, allowing queuing for each tenant's calls.

Figure: SUPERSET 4150 Telephone Serving Several Tenants

Local Night Switching

Tenants sharing the system may each require different system operation with respect to Day/Night mode. The system allows each tenant to switch into Night service independently and to operate independently when it has switched to Night service. This Figure shows two typical examples.

Figure: Night Switching Options

An attendant with a console may switch to Night 1 or Night 2 by pressing the FUNCTION key followed by the appropriate softkey.

A tenant with a SUPERSET 420 telephone presses the Superkey and the NO softkey until NIGHT SERVICE? appears in the display area of the telephone. Pressing the YES softkey will display the current service mode to the user and pressing the CHANGE softkey will allow the user to switch to another mode.

To switch into night service using a SUPERSET 4150, SUPERSET 430, or SUPERSET 4DN telephone, the user presses the Superkey and the MORE softkey until the NIGHT ANSWER softkey appears. Pressing this softkey will select the new mode of operation. To retain the current mode of service, press the BACKUP softkey or the Superkey.

The display area for each SUPERSET mentioned displays the current mode of operation, either NIGHT 1, NIGHT 2 or DAY service. In the case of DAY service, the display reverts to showing the time and date after a short period of time. Please refer to the Program Features section for complete programming information.

Since call rerouting destinations and incoming CO trunks are programmed for Day, Night 1, and Night 2, switching to Night service will automatically reroute all calls for that tenant, provided that the tenant is properly programmed in CDE Form 06, Tenant Night Switching Control.

It may be desirable to have a console or a SUPERSET 4150, SUPERSET 430, and SUPERSET 4DN telephone (such as a night security desk) able to switch all tenants into Night 1 or Night 2 service. The security desk console may be programmed as a tenant that controls the Night status of tenants. When the security guard sets Night 2, all associated tenants switch into Night 2. If a tenant forgot to switch to Night service, and calls were not being rerouted, the status is corrected when the night security guard comes on duty and sets Night 2 to reroute calls to the security desk.

Night Bells and Night Answer

The system allows incoming calls to be redirected to a Night bell. A non-display Mitel telephone in that tenant may dial the TAFAS - LOCAL TENANT feature access code to answer the first call in that tenant's queue. Mitel 5010, 5212 and 5215 IP Phone, and SUPERSET 3DN, SUPERSET 410, SUPERSET 420, SUPERSET 4015, SUPERSET 4025, and SUPERSET 4125 users press a NIGHT ANSWER feature key; SUPERSET 4DN, SUPERSET 430, or SUPERSET 4150 users press the NIGHT ANSWER softkey; and the attendant console will press the NIGHT BELL softkey. Any extension may dial the TAFAS - Any feature access code to pick up a call in any tenant to which it is allowed to connect. Please refer to the Program Features section for complete programming information.

Dial 0 Routing

Any tenant group may operate with a central attendant, subattendant or enhanced subattendant position handling incoming calls and Dial 0 calls, or may program its own Day/Night directory numbers to route Dial 0 calls. This Figure shows two examples.

The feature access code for attendant access (usually 0) is programmed systemwide for all stations. Each tenant may then specify the Day, Night 1, and Night 2 answer points to route callers who dial this access code.

Call Rerouting Features and Answer Point

Each tenant may determine its method of rerouting Dial 0 calls, intercepts for illegal access, calls that are not answered or that reach busy parties. The answer point may be an attendant console, subattendant LDN, enhanced subattendant LDN, Mitel telephone line, station, hunt group, or night bell. The caller is automatically camped on to a busy station or Mitel telephone. Calls may also be routed to a staffed night answering desk for the PBX, or to an outside answering service, by routing to a system abbreviated dial key programmed to an external number.

Figure: Dial 0 Call Routing

Rerouted calls arrive at a console as NO ANSWER or BUSY recalls. From the display, the attendant can identify which tenant originated the recall. Calls rerouted to a Mitel telephone may ring into a different key for each tenant, thereby identifying the company that originated the recall. If the line is busy, a recall will automatically camp on to the Mitel telephone.

Several examples of Call Rerouting follow and a complete list is given in the Customer Data Entry section. Call Rerouting is described in detail in the Program Features section.

Direct Inward Dial Rerouting

In some cases DID trunks may be shared among tenants. The block of numbers obtained is used by all the tenants, each with its own phone numbers listed in the telephone directory. The DID trunks are placed into one tenant which is programmed to be able to connect to the other tenants that share the DID trunks. If no one answers an incoming call, or if the number is busy, it may be rerouted to a specified answer point as a No Answer/Busy recall.

Direct-In Line Rerouting

Incoming DIL trunks are directed to ring destinations in any tenant that they can connect to. An unanswered incoming call may be rerouted to a specified answer point (Day, Night 1, or Night 2) as a No Answer recall, from the tenant into which the trunk rang.

Subattendants/Enhanced Subattendants and Message Centers

A company may set up its system with a main attendant console for receiving incoming calls and routing them to stations in the system. Usually the attendant who handled an incoming trunk call receives the No Answer recall. Any department wanting to handle its own recalls (for example, to take messages) may program alternate recall points for its calls. The department is programmed as a tenant and then uses call rerouting for its calls.

The person can take a message and set message waiting on the absent party’s set. Members of a department may call their own “message center” by dialing 0. Since each tenant programs its own dial 0 answering point, people do not have to remember how to call their message desk. They can call another department’s message desk by dialing its extension number.

Numbering Plan

Since tenants are allowed to connect to each other, extension numbers must be unique across the entire system. For example, there may be one and ONLY ONE extension 201 in the PBX. Similarly, feature access codes must be the same throughout the system. SMDR records may be sorted for each tenant by their unique extension numbers.

Automatic Route Selection with Multiple Tenants

Automatic Route Selection (ARS) enables the system to access, in a specific sequence, possible routes to a given destination. Routes are listed within the system in the order in which they are to be tried.

Route definition is based on trunk groups. Routes may be organized into lists of up to six route choices to arrive at a destination (refer to CDE Form 24, ARS: Route Lists). Since some tenants may be allowed to access only specific trunks, the ARS package checks the tenant interconnection table to verify if a caller may access a particular trunk. If the first choice trunks are not available to a particular tenant, the system skips them and takes the next choice trunks instead. The system automatically searches available trunks from first choice to last choice as it attempts to find a free trunk.

If tenant interconnection is allowed, shared trunks should be accessible from all participating tenants. ARS will optimize all calls made from these tenants. The administrator must collect and sort the SMDR reports for billing to each tenant.

Programming and CDE

This section describes how the customer data entry (CDE) package is programmed to set up the various functions already described in this document. The forms are more fully explained in the Program CDE section.

Tenant Assignments

Stations, sets, data sets, consoles, subattendants, enhanced subattendants, and trunks are assigned to a tenant when they are programmed into the system during CDE. The field labelled TEN is the tenant assignment field. Enter the tenant number (1 to 25) to which the station or Mitel telephone belongs. For example, the Form 09 below shows that extension 1301 and 1702 have both been assigned to Tenant 1.

Figure: Programming Form

Interconnection Between Tenants

Each tenant may be allowed or not allowed to call any other tenant, as specified in the Tenant Interconnection Table. On each horizontal line, an asterisk (*) indicates tenants that may be connected, and a period (.) indicates tenants that may not be connected from one tenant to another. The tenant being examined (horizontal row) can connect with each tenant that has an asterisk (*) in its column.

The diagonal of the matrix is labelled 0 because a tenant may not be prohibited from calling within itself. Connection control is unidirectional; if tenant 1 may connect to tenant 2, tenant 2 may not necessarily be able to connect to tenant 1. This allows for a master tenant who could call everyone but who may not be called by everyone.

This Figure shows an example where tenant 1 may call any tenant except tenants 5 and 7.

The Tenant Interconnection Table can provide security for data devices connected to the PBX through data sets by restricting access only to data sets within the same tenant group. Each group of data users who are to be able to access only each other are placed within the same tenant. Data set calls between tenants are then restricted by programming Form 05, Tenant Interconnection Table.

Figure: Tenant Interconnection Table

Call Rerouting

The Call Rerouting Table Figure specifies where each tenant is to route different types of calls in Day, Night 1, or Night 2 Service. If no number is specified, the caller receives reorder tone.

For Dial-in and DID trunks, forwarding and intercepting is for calls that are dialed into the tenant currently being displayed. DID trunks, which can access multiple tenants, are routed by the tenant whose local directory number was dialed.

Night bells and Attendant/Subattendant/Enhanced Subattendant LDN keys are programmed into the system with directory numbers for reference and may be specified here along with stations, Mitel telephones, and logical lines.

Figure: Call Rerouting Table

Note: For programming CDE Form 19, Call Rerouting Table, DID and Dial-In Tie Trunk Attendant Access Night Points only work when the tie trunks are sent to the Attendant's extension number and not to an LDN key on the console.

The following table defines which tenant controls the recall point for incoming calls that are not answered by the destination station:

TYPE OF CALL	CONTROLLING TENANT GROUP
Automatic Wake-up Routing for this Tenant	Destination Station
DID Recall Points on Busy	Destination Station
DID Recall Points on No Answer	Destination Station
DID Routing for Calls into this Tenant	Destination Station
DID Illegal # Intercept for this Tenant	Calling Trunk
DID Vacant Number Routing for this Tenant	Calling Trunk
DID Attendant Access Night Points	Calling Trunk
Note: Will not recall on no answer if the ringing destination is the trunk's Night 1 answering point.	
Non-Dial-In Trunks Alternate Recall Points	Destination Station
Dial-In Tie Recall Points on Busy	Destination Station
Dial-In Tie Recall Points on No Answer	Destination Station
Dial-In Tie Routing for Calls into this Tenant	Destination Station
Dial-In Tie Illegal # Intercept for this Tenant	Destination Station
Dial-In Tie Vacant Number Routing for this Tenant	Destination Station
Dial-In Tie Attendant Access Night Points	Destination Station
DND Intercept Routing for this Tenant	Destination Station
Station Dial 0 Routing	Destination Station
Priority Dial 0 Routing	Destination Station
UCD/Attendant Recording for this Tenant	Destination Station
UCD on Hold Timeout for this Tenant	Destination Station
DISA Day Service Routing for this Tenant	Destination Station
Station Vacant Number Routing for this Tenant	Destination Station
CO Line Routing Points on No Answer	Destination Station

DMP Music Sources for this Tenant	Destination Station
Station Illegal Number Routing for this Tenant	Destination Station

Night Switching

A Tenant Night Switching Control matrix allows one or more tenants (such as a night security desk console, subattendant, enhanced subattendant or Mitel telephone) to switch all tenants into night service. This matrix is similar to the Tenant Interconnection Table.

The tenant being examined (horizontal row) will night-switch each tenant programmed with an (*) in its column. This matrix is unidirectional. Tenant 1 may be programmed to night-switch tenant 2 but tenant 2 may not be programmed to night-switch tenant 1. This Figure shows an example where tenant 1 will night-switch all tenants into night service but all other tenants will only night-switch themselves. Tenant 1, therefore, is the “master tenant” for night switching control.

Figure: Tenant Night Switching Control

Traffic Measurement & IP Trunk Performance

Introduction

This section describes principles of telephone traffic measurement for a PBX, including programming and operating procedures for obtaining Traffic Measurement data. It also describes statistics collection for measuring the performance of IP trunks in the system.

Note: Traffic reports for features (such as ACD and Automated Attendant) are available only if the feature has been purchased.

Traffic Analysis

Traffic Analysis involves two activities: collecting data about the system (measurement) and interpreting this data (analysis) to optimize system performance. Once traffic measurement has been started in the PBX, it continues automatically until changed or stopped. Traffic measurement produces a single report for the system. The report includes all tenants, if a tenant service is provided.

Traffic measurement accumulates data in the form of peg counts and usage over a specified period of time. A peg count is the total number of times a facility (device, service, or feature) is accessed, regardless of the length of each access. Usage is the length of time or duration for which each facility is used. In certain applications, the peak value of facility usage during the period is also recorded. Call holding time is the average length of each call.

$$\text{usage (Erlangs)} = \frac{\text{peg counts (per hour) X call holding time (in seconds)}}{\text{3600 (seconds per hour)}}$$

$$\text{usage (CCS)} = \frac{\text{peg counts (per hour) X call holding time (in seconds)}}{\text{100 (seconds per CCS period)}}$$
(CCS = centi-call second)

$$\text{usage (Erlangs)} = \frac{\text{usage (CCS)}}{36} \quad \text{or,} \quad 1 \text{ Erlang} = 36 \text{ CCS}$$

Examine traffic measurement results to determine the adequacy of equipment provisioning, and the effectiveness of programmed options and features. Analyze the results to identify changes which can be implemented by reprogramming and/or reprovisioning to improve system performance.

Traffic Measurement Methods

Traffic measurement data is accumulated in periods of 1 to 60 minutes in length. The start time, which is specified to the nearest minute, and the duration (required number of periods) identify the daily time when measurements are collected. These parameters are entered from the console or maintenance terminal. Once set, traffic measurement will begin at the same time each day and continue for the same duration, until changed or stopped.

The system clock times the length of each period; however, a guard timer also monitors each period and can terminate the period if it times out before the system clock. This situation would occur only if the system clock has been changed during a measurement period. The period length is not guaranteed if the clock is changed or the system is reset during a measurement period.

Data is accumulated in active registers during each measurement period. At the end of each period, data is transferred to storage registers and the active registers are set to zero in preparation for receiving data from the next period. The data held in these storage registers can be printed, or written to magnetic tape or a similar storage device. At the end of each period, the data in the storage registers is replaced by the data accumulated in the active registers during the period. If the data was not retrieved from the storage registers during a period, new data received at the end of that period overwrites the data, and it is lost. This sequence then repeats for the specified duration.

Types of Traffic Counts

Two types of traffic counts are used in traffic measurement: peg counts and usage counts.

Peg Count

Each time a facility (device, service, or feature) is used, the call processing software increments its register by one count. A peg count is not concerned with the length of time of usage.

Usage Count

Usage counts can be divided into normal usage and maximum value counts.

Normal Usage: The amount of time for which a facility is used can be calculated from the usage count. At 10-second intervals, the call processing software scans each facility and increments the register if it is in use. The usage count is directly proportional to the time that a circuit is in use.

Each accumulated usage count is converted into CCS or Erlangs for the traffic measurement report. The maximum usage is 6554 CCS or 182 Erlangs which will occur if 182 devices in a facility are all busy for 1 hour. The consoles are scanned every second to increase accuracy in the average waiting time for an answered call.

Maximum Value Count: This type of count is obtained in a similar manner to usage count, except that the value obtained at each scan does not increment the register; instead, it is compared with the register's value and updates the register only if its value exceeds that in the register. This value reflects the scanned maximum count of the usage of a resource. Since this is a scanned value, it is possible that a busy peg can exist without the maximum count reaching the number available.

Types of Data Accumulated

The following data is accumulated during traffic measurement:

System Activity

System activity data indicates the extent of activity on the basic system (maximum peg count = 65535). Counts include:

1 s Dial Tone: Incremented every time an extension or dial-in trunk has to wait 1 or more seconds for dial tone. This peg is not incremented by the Automated Attendant feature.

2 s Dial Tone: This is incremented each time an extension or dial-in trunk has to wait 2 or more seconds for dial tone. This peg is not incremented by the Automated Attendant feature.

3 s Dial Tone: Incremented every time an extension or dial-in trunk has to wait 3 or more seconds for dial tone. This peg is not incremented by the Automated Attendant feature.

Console Calls: A count of all calls directed to any LDN (including those on subattendant sets) whether answered or not.

Console Orig.: A count of all console originations.

Dataset Orig.: This count is incremented each time a data call is originated either by ADL or DTRX.

Dial 0 Calls: A count of all dial 0 calls that are answered at any console. It includes priority dial 0 calls and all calls directed to Console directory numbers that are not answered.

Ext Origin.: Each time an idle extension goes off-hook, it causes this register to increment by one count. It does not increment when a ringing extension goes off-hook.

Intercepted: A count of all intercepted calls that are answered at all consoles.

Recall: A count of all console-answered recalls.

Activity: Each time a device has activity, this count is incremented.

Illegal Calls: This count is incremented whenever an extension, console or dial-in trunk dials a vacant or illegal number.

Features

Feature data indicates the activity of those features which have been programmed for the installation (maximum peg count = 65535). Counts include:

ADL Access: Successful ADL feature accesses.

Callback: Callbacks that have been set up.

Call Forward: All call forwarding set-ups (programming) at any extension or console.

Call Hold: Number of hard holds by an extension.

Call Park: Number of call parks.

Call Pickup: Legally dialed call pick-ups, including local pick-ups or directed pick-ups from extensions.

Camp-on: Camp-ons to any device (including trunk groups and hunt groups) by another device for extension-to-trunk, extension-to-extension, and trunk-to-extension.

Console Conf.: The number of times consoles use the conference softkey.

Console Hold: The number of all call-holds made at the console.

DND: Do not disturb set-ups.

DTRX Session: Successful DTRX session accesses.

Ext. Conf.: Extension conferences.

Flash Hold: Transfers or flash and holds from an extension or console.

Guest Room: Guest Room softkey depressions.

Hold Pickup: Successful hold pick-ups (calls held at the console in a hold slot and picked up using the dialed console and hold slot number).

Maid in Room: Number of accesses to Feature Access Code 35, Maid in Room.

Msg Waiting: Applied message waiting, from any source in any manner.

Override: Completed busy overrides.

Paging: Successful pager accesses.

Serial Call: Console serial calls set up by all consoles.

TAFAS: TAFAS (night answer) answered from dialed feature access code or Mitel softkey.

Wake-Up: Call wake-up set-ups.

UCD Loginout: UCD agent logins and logouts.

Receivers

This data records the activity on receivers within the system (maximum peg count = 65535). Sample traffic measurements have been included in the Reports Chapter .The following counts are recorded:

Peg: The total number of times that the receivers were accessed during the measurement period.

Usage: Represents the total usage (Erlangs or CCS) for the receivers during the measurement period. Does not include busied-out receivers (max = 6553.5 ccs).

Busy Peg: The number of times a call could not immediately be allocated a receiver due to busy conditions.

Maximum in Use/Available: The scanned maximum count of the number of receivers in use during the period and the number that are available for use. Does not include busied-out receivers.

Channel Usage And Local Switching

This data records activity between bays and within each peripheral bay. The following counts are recorded:

Bay Number: Identifies the bay for which data is being accumulated.

Channel Usage: Usage of channels available to the bay. Does not include channels allocated for music or tones (maximum = 6553.5 ccs).

Max Channel/Available: The scanned maximum number of channels in use and the number available for use. Does not include channels allocated for music or tones.

Local Switching Usage: Usage of analog peripheral bay local switching (maximum = 6553.5 ccs).

Max Local Switching: The scanned maximum number of local switches found in an analog peripheral bay.

Dynamic Records for Hunt Groups, Trunk Groups, and Trunks

The number of system, feature, receiver and channel pegs or records is fixed for each system. The number of console, hunt group, trunk group, and trunk records can vary. At the beginning of each period, records are allocated for consoles, trunk groups, and trunks. Each console, hunt group, and trunk group that is programmed in CDE is allocated one record. The programmed trunks are allocated the remaining records, in ascending trunk number order. A total of 300 records is allocated.

Example: The following traffic could be recorded:
- 4 consoles
- 8 hunt groups
- 15 trunk groups
- 70 trunks, numbered 1 to 70

97 records (total) 203 records still available (98 - 300)

The following data is accumulated during traffic measurement:

Console

This data records activity of each console on the system. Console activity includes the following information:

Console Number: Directory number of the console. The console directory number is determined at the time of CDE Entry in Form 07, Console Assignments. If this console has been deleted, it will display ???? indicating that the directory number could not be determined.

Usage: Non-idle use of the console. The console is non-idle when it is using a channel. The console is idle when it is ringing, using the application key, or using the select features key (maximum = 36 CCS).

Call Waiting Usage: Usage of the Console queue for all calls presented to this console, either answered or unanswered (maximum = 655.35 CCS).

Calls Answered: Calls answered using the console. Includes softkey and ANSWER key answers (maximum peg count = 65535).

Avg Waiting Time: Average waiting time for a call directed to the console to be answered by the console.

Hunt Groups

The activity of each hunt group is recorded. Hunt group activity includes the following information:

Hunt Group Number: Programmed hunt group number.

Peg: Number of accesses to a hunt group, including busy attempts (maximum = 65535).

Skip Peg: Number of failures to get a receiver, which resulted in skipping the recording and routing to the default destination (for Automated Attendant Hunt Groups only).

Usage: Usage of all devices in the hunt group (maximum = 6553.5 CCS).

Busy Peg: Number of times the hunt group was found busy by a caller.

Max in Use/Available: Scanned maximum number of devices in the hunt group that were busy (busied-out devices are not included) and the number of devices available for use in the group.

Trunk Groups

This data records the activity of each trunk group. Trunk group activity includes the following information:

Trunk Group Number: Programmed trunk group number.

Peg: Number of accesses to a trunk group, including busy attempts (maximum = 65535).

Usage: Usage of all trunks in the trunk group making outgoing calls, measured in CCS or Erlangs (maximum = 6553.5 CCS).

Busy Peg: Number of times the trunk group was busy (maximum = 255).

Max in Use/Available: Scanned maximum number of trunks in the trunk group that were busy (busied-out trunks are not included). The number of available trunks in a trunk group is determined at the time of the traffic measurement report and is not in the storage registers.

Trunks

The activity of each trunk is also recorded. Trunk activity includes the following information:

Trunk Number: List of all trunk numbers that were accessed.

Peg in: Total number of times the trunk was seized for incoming calls (maximum = 255).

Usage In: Represents the usage (in Erlangs or CCS) for incoming calls on the trunk (maximum = 36 ccs).

Peg Out: Total number of times the trunk was seized for an outgoing call on the trunk (maximum = 255).

Usage Out: Represents the usage (in Erlangs or CCS) for outgoing calls on the trunk (maximum = 36 ccs).

Register Count Examples

A call which lasts for 100 seconds has a value of 1 CCS (one hundred call-seconds). One Erlang equals 36 CCS or 3600 call seconds. Usage is measured in 10-second units; for example, a usage count of 128 (128 x 10 s) represents 1280 seconds of usage, equivalent to 12.8 CCS or 0.3556 Erlang (36 CCS equals 1 Erlang). The following example illustrates these “count” methods.

Extension Originations Peg Count - Each time an idle extension goes off-hook, it causes the register to increment by one count. Since it is a measure of the number of calls originated by the extensions, it does not increment when a ringing extension goes off-hook to answer a call. For example, if this register has a value of 858, the number of call originations (or off-hook originations) during the period totalled 858.

Trunk Group Usage - Each time the system scans the members of this trunk group, at 10 second intervals, it counts the number of members that are busy. A register value of 273 indicates that members were busy for 273 X 10 = 2730 seconds (27.3 CCS) during the measurement period.

Usage Measurement - Device usage is measured every ten seconds; therefore, each usage measurement is accurate to +/- 10 seconds (0.1 CCS or 0.0028 Erlang). Console usage measurement is accurate to +/- 1 second (0.01 CCS or 0.00028 Erlang) because they are scanned every second.

Power Failure

If system power fails, the current traffic measurement will be lost, and the latest traffic report will also be lost if it had not been output. When power is restored, traffic measurements will restart and continue until the end of the scheduled period. The new start time will begin at system power-up.

Traffic Measurement Commands

Maintenance Terminal and Console

The maintenance terminal or console is used to enter the data required to start traffic measurements, print measurements, monitor status, or change traffic measurement parameters. Refer to the RS-232 Maintenance Terminal section for instructions on using the terminal. At the maintenance terminal or console, select the MAINTENANCE application. From the MAINTENANCE main menu, select TRAFFIC_MEAS (softkey 5).

Traffic Measurement Commands

1- SET	2- SHOW	3- PRINT	4- READ	5-IP MEAS
6- QUIT	7-	8-	9- STOP	0-

SET	Set traffic report information
SHOW	Status of current traffic report that contains information defined by the SET command
PRINT	Print the latest traffic report
READ	Read the latest traffic report
IP MEAS	Set up collection of IP trunk statistics
STOP	Stop current traffic measurements or stop printing report

SET Command

1- UNITS	2- PERIOD	3- DURATION	4- AUTOPRINT	5- CANCEL
6-	7-START_TIME	8- CONDENSED	9-	0-

UNITS	CCS or Erlangs
PERIOD	1 to 60 minutes (default = 60)
DURATION	Number of periods (default = 8)
AUTOPRINT	Print report automatically after each period (toggles ON or OFF) **Note:** Autoprint does not apply to IP traffic measurement.
CANCEL	Quit SET commands
START TIME	Presented in hours:minutes format (hh:mm)
CONDENSED	Condensed report (toggles ON or OFF)

Select the SET command to enter or change any traffic measurement parameters. Use the prompts to change required data and to exit from the TRAFFIC_MEAS function. Setting START_TIME to a specific time prepares the traffic data collection facility. Data collection is only started when that time is reached. Until then, the PERIOD, DURATION, and START_TIME parameters can be changed. Once traffic measurements are started, a STOP command must be issued first, followed by other commands. The AUTOPRINT and CONDENSED parameters can be changed during a traffic measurement period. At the beginning of each period of current traffic measurement, the pegs and dynamic records are initialized. If a new trunk group was set up by CDE, it would not be included in the traffic report until the next period.

STOP Command

When the STOP softkey is pressed, two new softkeys appear (TRAFFRPT and PRINT), allowing you to select the activity you wish to stop.

TRAFFRPT **-** Stops the traffic measurement which is currently active. No report will be made of the incomplete collected data.

PRINT **-** Stops a printout from being sent to the PRINTER port.

Note: The STOP command will not stop an AUTOPRINT in progress. The AUTOPRINT command must be set to OFF between printouts to stop printouts.

SHOW Command

When you press the SHOW softkey, you will see a STATUS softkey displayed. Pressing the STATUS softkey will provide a summary report on the current traffic measurement parameters.

STATUS:	Activated/OFF
START TIME:	hh:mm (hours:minutes)
DATA COLLECTION:	Running/OFF
PERIOD:	mm (minutes)
DURATION:	nn (periods)
AUTOPRINT:	ON/OFF
UNITS:	Erlangs/CCS
CONDENSED REPORT:	ON/OFF

START TIME and DATA COLLECTION are not displayed if STATUS is set to OFF.

To set STATUS to off:

press STOP
press TRAFFRPT
press ENTER

The DATA COLLECTION field indicates whether data is currently being collected (if traffic is being measured). When the field is set to OFF, either the daily start time has not been reached, or, the required number of periods have been completed since the start time. In either case, no data is currently being collected.

PRINT Command

Pressing the PRINT softkey causes the latest traffic report held in the system to be printed to the output device specified in Form 34, Directed IO.

READ Command

This softkey command causes the latest traffic report in the system to be displayed on the maintenance terminal where it can be read by the user. READ is only available at the maintenance terminal. If the console softkey is pressed, the following message appears on the LCD display:

IP MEAS Command

Pressing the IP MEAS softkey displays additional softkeys used to collect IP trunk statistics.

Note: IP statistics collection is off by default and must be restarted following a system reboot.

Use the SET softkey to set up the following collection parameters:

SAMPLE	can be programmed for 2, 5, 10 or 15 minute intervals (default = 5) during the period of the traffic report. This means that if you have the sample programmed for two minute intervals and the period was set for 30 minutes, then you should see that 15 samples were requested during the report.
PERIOD	determines how long you want each report to be. If it was programmed to do two minute samples and a 30 minute period, then after 30 minutes, a report would be generated to show what the results were for each IP trunk route programmed in the SX-200 ICP. The period length can be set for 10, 30 or 60 minutes in length.
DURATION	determines how many periods you would like the traffic report to run. A Duration of 20, for example, would generate an output every 30 minutes for the next 10 hours. 30 minute period X 20 periods = 10 hours. The duration can be set from 1 to 99 periods.
MAX DELAY	the value used to determine if the sample is Slow or not. It can be programmed from 100 ms to 500 ms.
AUTOPRINT	Not supported for IP traffic measurement.

Changes made to the collection parameters take effect after the START softkey is pressed to begin collecting.

You can print the last report (to the printer) or read the last report using softkeys 3 and 4. To see the whole report you can send it to the FTP Server using softkey 8. The FTP server must be set up in Form 47 (IP Networking), Subform 01 (System IP).

Below is a sample traffic report.

THU SEP 25 12:14:25 20 TO THU SEP 25 12:24:25 2003 (sample rate 2 mins)

		Delay(ms)		Samples		
Node	Status	Ave	Max	#Req	#Conn	#Slow
3	Connected	7	10	5	5	0
6	Connected	10	10	5	5	0
8	Connected	5	10	5	5	0
102	Connected	1165	2190	5	2	1

112	Local	0	0	0	0	0
117	Disconnected	55	160	5	4	0
119	Connected	13	30	5	5	0

The traffic report above shows that the local SX-200 ICP has IP trunking to six other locations. 112 is the IP Node number of the local SX-200 ICP. **(Note**: IP Nodes can be numbered from 0 - 255 but the maximum number of nodes that can be assigned is 100.) The most important information is the # of Slow samples. In an ideal network, this should always be 0. The timestamp at the top indicates that the report is of a 10 minute period. Since the sample rate is two minutes there should be five samples requested (#Req) in this period. The #Conn is the actual number of samples that were completed -- i.e, received a response from the remote PBX.

For Node number 3, the SX-200 ICP sent five samples, received five responses and the average delay time was seven milliseconds and the longest delay was 10 milliseconds. For Node 102, the SX-200 ICP sent five samples, only received one response. One response was beyond the Max Delay timer and the average time for the two samples was 1165 milliseconds with the longest delay being 2190 milliseconds.

There are several Status states.

"Connected" means the network is up and working.
"Disconnected" means the IP trunking is programmed and the link exists but no response is being received.
"Local" means this is the local Node so it will always contain 0's.
"No Link" means that there is no link between the two nodes and no attempts will be made to test the link.
"Config Error" means the link is programmed but no route exists for this link in Form 23.

TO SERVER Command

The TO_SERVER softkey transfers the IP statistics measurement logs to the FTP server specified in Form 47 (IP Networking), Subform 01 (System IP). The log is a text file containing the reports generated at the end of each statistics measurement session.

QUIT Command

This softkey command allows the user to exit from Traffic Measurement mode at the console or maintenance terminal.

Installation

To meet Traffic Measurement requirements during system installation, you must

1. determine the required traffic parameters
2. determine the required output device
3. install the output device
4. program the traffic parameters and output device for this installation.

Connection Requirements, Local Printer

The Installation Procedures section details the installation. Refer to the General Maintenance Information section to reference System Set Commands. This section explains how to assign printers during installation and to perform initial maintenance programming. The Traffic Measurement data may be output to a printer, magnetic recording device, or the maintenance terminal. If required, a Data Demultiplexer may be used and the Traffic Measurement data directed to one output of it.

When a backplane printer port is used, it should be located as near as possible to the PBX and connected to the PBX data port with a 25-conductor connectorized cable, not longer than 4.5 m (15 ft) in length.

A Printer may be specified in CDE Form 34, Directed IO, which will allow all Traffic Measurement printouts to be sent to this printer which is connected to a port equipped with an asynchronous data set. Refer to the Form 34 - Directed IO in the Program CDE section for further details.

Note: Ensure that the length of time the printer requires to print one report is less than the length of time specified for a traffic measurement period; otherwise, reports will be generated faster than the printer can print them.

Reports

The standard traffic measurement and IP trunk performance reports provide a printed record of the data in the storage registers (usually the measurements taken during the preceding period).

Report Heading Description

Port size and system number are unique to each system and are programmed when the system is initially configured. In the following typical report, the Generic is 100Y, where Y is the last digit of the software Generic installed in the PBX. ZZZZZZ is an optional field which defines software loads that include special features; for example, an ACD load displays ACD PC. The port size is represented by XXXP. The system identifier is represented by nnn (if one has not been programmed, 0 appears). The CHANNELS data for a maximum system configuration is printed. Local Usage and Max Local apply only to analog bays.

Typical Traffic Measurement Report

In this report, traffic from two bays is printed.

SX-200_ICP_2.0.4.19 TRAFFIC REPORT

SYSTEM 337
17-AUG-04
12:24 to 12:44

SYSTEM ACTIVITY:

Is dial tone	0	2s dial tone	0	3s dial tone	0
Console calls	0	Console orig.	0	Dataset orig.	0
Dial 0 calls	0	Ext. origin.	0	Intercepted	0
Recall	0	Activity	0	Illegal calls	0

FEATURES:

ADL access	0	CallBack	0	Call hold	0
Call Forward	0	Call Park	0	Call Pickup	0
Campon	0	Console conf.	0	Console hold	0
DND	0	DTRX session	0	Ext. conf.	0
Flash hold	0	Guest Room	0	Hold pickup	0
Maid in Room	0	Msg Waiting	0	Override	0
Paging	0	Serial call	0	TAFAS	0
Wakeup	0	UCD Login/out	0	Rec-call conf	0

DTMF RECEIVERS:

Peg	Usage	Busy Peg	Max/Avl
0	0.00 ccs	0	0/16

PSEUDO DTMF RECEIVERS:

Peg	Usage	Busy Peg	Max/Avl
0	0.00 ccs	0	0/32

CLASS RECEIVERS:

Info	No Info	No Rcvr	Total Avl
0	0	0	0

CHANNELS:

Bay	Usage	Max/Avl	Local Usage	Max local
1	1.20 ccs	2/184		
2	0.00 ccs	0 / 0		
3	0.00 ccs	0 / 0		
4	0.00 ccs	0 / 0		
5	0.00 ccs	0 / 0		
6	0.00 ccs	0 / 0		
7	0.00 ccs	0 / 0		
8	0.00 ccs	0 / 0		
9	0.00 ccs	0 / 0		
10	0.00 ccs	0 / 0		
11	0.00 ccs	0 / 0		
12	0.00 ccs	0 / 0		
13	0.00 ccs	0 / 0		
14	0.00 ccs	0 / 0		
15	0.00 ccs	0 / 0		

CONSOLE:

Directory Number	Usage	Calls Answered	Calls Waiting Usage	Avg. Waiting Time (sec)
1805	0.00 ccs	0	0.00 ccs	0.00

CLASS GENERATORS

Bay	Peg	No Gen	Total Avl
1	0	0	0
2	0	0	0
3	0	0	0
4	0	0	0
5	0	0	0
6	0	0	0
7	0	0	0
8	0	0	0
9	0	0	0
10	0	0	0
11	0	0	0
12	0	0	0
13	0	0	0
14	0	0	0
15	0	0	0

HUNT GROUPS:

Number	Peg	Skip	Usage	Busy Peg	Max/Avl
1	0		0.00 ccs	0	0 / 4
5	0		0.00 ccs	0	0 / 1
10	0		0.00 ccs	0	0 / 1

TRUNK GROUPS:

Number	Peg	Usage	Busy Peg	Max/Avl
1	0	0.00 ccs	0	0 / 6
7	0	0.00 ccs	0	0 / 5

TRUNKS:

Number	Peg In	Usage In	Peg Out	Usage Out
1	0	0.00 ccs	0	0.00 ccs
2	0	0.00 ccs	0	0.00 ccs
3	0	0.00 ccs	0	0.00 ccs
4	0	0.00 ccs	0	0.00 ccs
5	0	0.00 ccs	0	0.00 ccs
6	0	0.00 ccs	0	0.00 ccs
177	0	0.00 ccs	0	0.00 ccs
178	0	0.00 ccs	0	0.00 ccs
179	0	0.00 ccs	0	0.00 ccs
180	0	0.00 ccs	0	0.00 ccs
181	0	0.00 ccs	0	0.00 ccs
182	0	0.00 ccs	0	0.00 ccs
183	0	0.00 ccs	0	0.00 ccs
184	0	0.00 ccs	0	0.00 ccs
185	0	0.00 ccs	0	0.00 ccs
186	0	0.00 ccs	0	0.00 ccs
187	0	0.00 ccs	0	0.00 ccs
188	0	0.00 ccs	0	0.00 ccs
189	0	0.00 ccs	0	0.00 ccs
190	0	0.00 ccs	0	0.00 ccs
192	0	0.00 ccs	0	0.00 ccs
193	0	0.00 ccs	0	0.00 ccs
194	0	0.00 ccs	0	0.00 ccs
195	0	0.00 ccs	0	0.00 ccs
196	0	0.00 ccs	0	0.00 ccs
197	0	0.00 ccs	0	0.00 ccs
198	0	0.00 ccs	0	0.00 ccs
199	0	0.00 ccs	0	0.00 ccs
200	0	0.00 ccs	0	0.00 ccs

Typical IP Trunk Performance Report

In this report, statistics for a large network of IP systems is printed.

SX-200_ICP_2.0.4.19   IP TRAFFIC REPORT
SYSTEM 337

WED AUG 18 13:53:04 20 TO WED AUG 18 14:03:09 2004 (sample rate 2 mins)

		Delays (ms)		Samples		
Node	**Status**	**Ave**	**Max**	**#Req**	**#Conn**	**#Slow**
1	Connected	7	10	5	5	0
22	Connected	7	20	7	5	0
73	No Link	0	0	0	0	0
81	Connected	3	10	5	5	0
89	Local	0	0	0	0	0
222	No Link	0	0	0	0	0

Explanation of fields:

Node is the number assigned in Form 48 to each SX-200 ICP (or other Mitel IP PBX) in the network.

Status is the status at the end of the sampling period. Possible states are as follows:

Connected -- connected to remote node.

Local -- local node doing the measurements

Config Error -- connected to remote node but some configuration information is missing. Either the node is not present in Form 48 or there is no route to this node in Form 23.

Disconnected -- disconnected from remote node.

Delay (ms) is the average and maximum end-to-end signaling delay between nodes.

Samples

#Req: Number of samples requested
#Conn: Number of samples where a connection was discovered
#Slow: Number of samples where the connection was above the acceptable maximum signalling delay.

The values reported for #Req and #Conn should be identical. If the network is well engineered and the system is not under unusual stress, the number of slow samples should also be 0

The above report could be interpreted as follows:

Connection to Node 21: OK

Connection to Node 1: has excessive network delays and should be investigated

Connection to Node 41: was disconnected for the entire report and remained so at the end of the report.

Connection to Node 146: Node 146 was disconnected for part of the report (possibly was reset).

Connection to Node 200: Either node 200 does not appear in Form 48 correctly or there is no route to node 200 in Form 23.

Condensed Traffic Report

The condensed report contains the data only, in decimal form. Register numbers and assignments are known for each particular software Generic. The dynamic section of the report requires a header for each group of data and the device number beside the device data. A program which analyses the data knows the format in which the condensed report is structured, and can interpret the data accordingly.

Analyzing Traffic Reports

From reading the Traffic Report, the following total traffic values can be determined:

Total Traffic	= Sum of channel usage - Receiver usage
Number of Calls	= Extension Originations + Console Originations + sum of incoming trunk pegs

The following table lists typical business traffic:

Traffic Type	Number of Ports	Traffic / Port (CCS)	Total CCS
Light	250	1.4	350
	300	1.4	420
	350	1.4	490
	400	1.4	560
	480	1.4	672
Medium	250	3.0	750
	300	2.9	870
	350	2.9	1015
	400	2.9	1160
	480	2.9	1392
Heavy	250	5.5	1375
	300	5.4	1620
	350	5.4	1890
	400	5.4	2160
	480	5.4	2592

Hotel/Motel

Introduction

The hotel/motel feature package integrates standard business features with custom hotel/motel features. The SX-200 ICP can also interface with a property management system (PMS).

The SX-200 ICP can operate with either the Hotel/Motel Package or the Property Management System package (not to be confused with the Voice mail Property Managment System Option), but not both packages on the same system.

Description

Hotel/Motel Feature Package

The Hotel/Motel feature package is provided by Mitel and is supported by the SX-200 ICP system.

The Hotel/Motel feature package is designed specifically for use in the hotel environment. The features facilitate guest-related activities such as checking guests in and out, providing automatic wake up calls and taking messages. The Hotel/Motel feature package can be used to configure the telephone activities that guests are allowed (such as blocking calls between rooms), to provide direct-to-room dialing, and to record what use is made of room telephones. It also assists with the administration of hotel activities such as keeping an accurate record of room occupancy and tracking housekeeping activity and status.

Attendant Activities

Using the Hotel/Motel feature package, the attendant can:

- Check the current room status
- Modify room status (assign guest rooms)
- Program multiple automatic wakeup calls that occur once or repeat daily
- Program personal wakeup calls that occur once or repeat daily for VIP guests
- Request reports of all current wakeup calls, message register counts, and status of rooms
- Enable direct inward dialing; outside callers can reach guests without front desk assistance.
- Inhibit room to room calling
- Assign outgoing call privileges on a room to room basis
- Request a message register audit.

Staff/Management Activities

Hotel/Motel staff or management can:

- Record the number of calls made from each guest room (automatic message register for guest rooms)
- Receive automatic room status conversion
- Access maid in room updates from guest room
- Access maid status display from Mitel telephone
- Access room status display from Mitel telephone
- Interface to a property management system (PMS)
- Generate single-line printouts for events such as unanswered wakeup calls, set up or canceled wakeup calls, and message register overflow.

Guest Activities

A hotel/motel guest can:

- Receive calls directly from outside callers without front desk assistance
- Set their own wakeup calls (either on the guest room telephone or by request to the attendant)
- Refuse telephone calls (by setting Do Not Disturb on the guest room telephone or by request to the attendant)

- Access special features and speed dial capabilities through guest room Mitel telephone key programming
- Observe the calling name and number on analog telephones with CLASS functionality.

Feature Access

Hotel/Motel features can be accessed from:

- Attendant console
- Front desk terminal
- Mitel display telephones with softkeys
- Industry-standard telephones and SUPERSET telephones without softkeys.

The SUPERSET 4150, SUPERSET 430, SUPERSET 4DN, and the Mitel 5340 IP telephones programmed as a subattendant can set up a wake-up alarm call that rings the guest room. SUPERSET 4150, SUPERSET 430, and the Mitel 5340 IP telephones programmed as a subattendant may also activate the Call Blocking feature.

Analog telephones with a display show the calling name and number while the analog telephone is ringing and while the user is on a call (campon). Clearing of the stored calling names and numbers occur during checkout. See CLASS for Analog Sets.

In addition, there are several features that can only be accessed directly from the SX-200 ICP System.

Attendant Console Access

Feature Access. Attendant console access to the Hotel/Motel Feature Package integrates telephony with guest room functions. The attendant can be in conversation with a guest room while performing hotel/motel features. The SX-200 ICP System can support several (a maximum of 11) consoles. However, each console operates independently to change guest room information.

The capabilities of the hotel/motel features as accessed by the attendant console are shown in the figure below.

Figure: Hotel/Motel Features Available on the Console

Access Mechanism. Hotel/motel features are accessed through the FUNCTION hardkey and the GUEST ROOM softkey on the attendant console. The features are then available on softkeys. One feature, Call Block (blocking calls between rooms, see Call Blocking) is accessed through the BLOCK hardkey.

Figure: Attendant Console Access to Hotel/Motel Features

The hotel/motel features can be accessed whether the attendant console is idle or in the process of handling a call. When handling a call from a guest room, after the FUNCTION hardkey and GUEST ROOM softkey are pressed, the attendant console display shows information about the source on the top line of the display, room information on the second line, and the softkeys on the remaining two lines. Note that information about the source is presented without pressing the FUNCTION and GUEST ROOM keys. See the figure below.

Figure: Guest Room Display During a Call from the Guest Room

Pressing the GUEST ROOM softkey while the attendant console is not engaged in a call produces the display shown in the figure below.

Figure: Guest Room Display with Idle Console

Once a room number (extension number) has been entered, the display changes to the format shown in this Figure showing Guest Room Display During a Call from the Guest Room with a ROOM NUMBER softkey displayed in the F0 position. Pressing EXIT at any time returns the attendant console to its prior state (idle or busy call processing).

The following softkeys appear on the attendant console display after pressing the GUEST ROOM softkey:

Figure: Console Display after Pressing GUEST ROOM Softkey

When one of the choices has been selected, the display is changed as follows:

Figure: Console Display after Pressing VAC/DIRTY RM Softkey

The total number of rooms of the specified status, e.g. VAC/DIRTY, appears in the top left corner of the display if there is at least one room which fits that description. A maximum of 6 room numbers will be shown on the same display. Pressing the MORE softkey will display the remainder of the room numbers.

When there are no VAC/DIRTY rooms, the following is displayed:

Figure: Console Display if no VAC/DIRTY Rooms

The console display could also show no occupied and clean rooms, no vacant and clean rooms, no occupied and dirty rooms or no reserved and clean rooms.

Attendant Console Key Functions

The functions of the hotel/motel keys on the attendant console are shown in the Attendant Console Key Functions Table below.

Table: Attendant Console Key Functions

Console Hardkeys	Console Softkeys	Function
BLOCK		Blocks calls between rooms.
FUNCTION		Accesses GUEST ROOM softkey.
	GUEST ROOM Softkey	Accesses all hotel/motel features other than call blocking.
	MAID	Produces display of all rooms with maids in them.
	RES/CLEAN RM	Produces display of all reserved and clean rooms.
	VAC/CLEAN RM	Produces display of all vacant and clean rooms.
	OUT OF SERV.	Produces display of all out of service rooms.**
	AUDITS	Accesses audit softkeys. Generate printouts of:
	ROOM STATUS WAKEUPS PRINT DID*** MSG REGISTER	- Room Status - Wakeup calls - DID Number - Message Register counts.
	NO DISTB *	Alternately sets up and cancels Do Not Disturb on an extension.
	SEND MSG * CLEAR MSG * READ MSG *	Sets up Message Waiting for a room. Cancels Message Waiting for a room. Reads the message for a room.

Table: Attendant Console Key Functions

Console Hardkeys	Console Softkeys	Function
	WAKE-UPS*	Sets up Wakeup Calls.
	MSG REG * CLR REG *	Displays message register. Clears message register.
	ROOM NUMBER*	Allows input of a new room number.
	STATUS *	Accesses occupancy, condition, and call restriction softkeys.

* Must enter Room Number first.

** The OUT OF SERV. softkey appears in two different contexts:

- When the GUEST ROOM softkey is pressed, the OUT OF SERV. softkey appears (used for searching out of service rooms).
- When the STATUS softkey is pressed, the OUT OF SERV. softkey appears (used to set the room status to out of service).

*** PRINT DID output example:

Name: JOHN SMITH
<blank line>
Room: 2000
<blank line>
Direct: 6135432559

- Name must be entered in Form 9 (Desktop Device Assignment) to appear in DID printout.
- If a DID number is not assigned to the Room Extension, the "Direct" display shows "Not Available".

Front Desk Terminal Access

Feature Access. The capabilities of the Hotel/Motel Feature Package as accessed by the front desk terminal are shown in the figure below.

Figure: Hotel/Motel Features Available on the Front Desk Terminal

Up to four VT100 terminals can access the Hotel/Motel Feature Package simultaneously. The VT100 terminal can access more hotel/motel features than the attendant console. For example, using a VT100-compatible terminal to input guest information permits the entry of alphanumeric characters (such as a guest name) and allows more information to be displayed at once.

A terminal interface provides a low-cost alternative to a property management system (PMS) by providing faster access to information but no room billing. The terminal provides no telephony function. Telephone activities must be performed through the attendant console or some other telephony device.

Access Mechanism. When the Hotel/Motel Feature Package is running on a front desk terminal, there are four modes of operation: house statistics, search by room or name, room updates, and audits.

House statistics. After successful login using a password, a house statistics screen appears. It contains a snapshot of the current status of the guest rooms or gives the time that the last statistics had been calculated (or when the REFRESH softkey is pressed). The information is only updated when the screen is entered. The house statistics screen is the only entry point when logging on to the application. It is also the only point the application can be exited from. Five softkeys give access to the other modes of operation. A sixth softkey is used to recalculate totals. See the figure below.

Figure: House Statistics Screen

Search by room or name. Using a terminal allows guest first and last names to be entered as part of the check-in procedure. A search facility allows searching by last name by pressing the GUEST SEARCH softkey on the house statistics screen. A partial or complete text string can be entered and all names matching the input string are displayed.

There are nine other types of room searches. These are based on room type, room number, and room status. They are accessed by pressing the ROOM SEARCH softkey on the House Statistics screen as seen in this Figure).

Any search other than room type or room number requires that a room code be entered to specify the type of room, such as all single rooms, that should be searched. The digit 0 can be used to search all room types meeting the search criteria. See the figure below.

Figure: Room Search Mechanism

Room updates. Changes to the guest room can only occur in this mode. It can be entered from the house statistics screen, or the room search or guest search mode by pressing the ROOM UPDATE softkey. Guest name, room occupancy status, room condition, and call privilege can be changed; wakeup times, message waiting, and Do Not Disturb can be set or cleared; and the message register can be cleared from this screen. The softkeys and prompts change according to the highlighted field.The screen shows the three wakeup timers. A daily wakeup call shows an asterisk beside the time. A personal wakeup call has "Pers" after the time. See the figure below.

Figure: Vacant Room Template

Audits. Various kinds of audits (hard copy reports) can be generated on demand. Audit mode is entered from the house statistics screen by pressing the AUDITS softkey. Each kind of audit appears as a corresponding softkey. The system option to allow the audit must be enabled to allow the softkey to appear. The types of audits available are: message register, wakeup, room type, and room status.

Front Desk Terminal Softkey Functions

The functions of the hotel/motel softkeys on the front desk terminal are shown in the Front Desk Terminal Softkey Functions Table below.

Table: Front Desk Terminal Softkey Functions

Mode	Softkey	Function
House Statistics	Audits	Enters the Audits mode of operation.
	Room Updates	Enters the Room Updates mode of operation.
	Guest Search	Enters the Guest Search mode of operation.
	Room Search	Enters the Room Search mode of operation.
	Refresh	Recalculates the totals on the screen.
Guest Search		N/A
Room Search	Room Type	Searches for rooms of a specified room type. Displays nine rooms at a time.

Table: Front Desk Terminal Softkey Functions

Mode	Softkey	Function
Room Number	Searches for a specified room and displays the next eight rooms as well.	
Vacant	Searches for all vacant rooms when a room code (room type) is specified or the wild card (0) is used.	
Vacant/Clean	Searches for all vacant/clean rooms when a room code (room type) is specified or the wild card (0) is used.	
Maid in Room	Searches for all rooms with a maid in them when a room code (room type) is specified or the wild card (0) is used.	
Out of Service	Searches for all rooms that are out of service when a room code (room type) is specified or the wild card (0) is used.	
Dirty	Searches for all dirty rooms when a room code (room type) is specified or the wild card (0) is used.	
Reserved	Searches for all reserved rooms when a room code (room type) is specified or the wild card (0) is used.	
Guaranteed	Searches for all guaranteed rooms when a room code (room type) is specified or the wild card (0) is used.	
Page Down	Presented in search mode on the first and subsequent screens if there are more than nine rooms meeting the search requirements. Is not displayed on the last screen. Displays subsequent (additional) rooms, nine at a time.	
Page Up	Presented in each search mode on the second and subsequent screens if there are more than nine rooms meeting the search requirements. Is not displayed on the first screen. Displays previous rooms, nine at a time.	
Room Updates	Show Choices	When highlight bar is in the Occupancy field, presents the Occupancy softkeys. When highlight bar is in the Condition field, presents the Condition softkeys. When highlight bar is in the Call Privilege field, presents the Call Privilege softkeys.
Check Out	Only available if room occupancy is occupied. Sets room occupancy to vacant, room condition to dirty, call restriction to the default setting. Confirmation (the Enter key) is required to set this status. Enter softkey blanks out guest name, disables Do Not Disturb (if set), clears message waiting, wakeup time and message register.	
Set Message	When pressed, notifies the guest room by bell or lamp that there is a message waiting for the guest.	
Clear Message	Appears when there is a message waiting. Clears the message waiting indication in the guest room and at the front desk terminal.	
Occupancy -Vacant	Sets room occupancy to vacant and the call restriction to the vacant/reserved default* setting; unassigns DID number if applicable.	

Table: Front Desk Terminal Softkey Functions

Mode	Softkey	Function
-Occupied -Reserved -Guaranteed		Sets room occupancy to occupied and the call restriction to the occupied default setting; assigns DID number if applicable; Sets room occupancy to reserved and the call restriction to the vacant/reserved default setting; unassigns DID number if applicable. Sets room occupancy to guaranteed and the call restriction to the vacant/reserved default setting; assigns DID number if applicable.
Condition -Clean -Dirty -To Inspect -Out of Serv		Sets the room condition to clean. Sets the room condition to dirty. Sets the room condition to to be inspected. Sets the room condition to out of service.
Call Privilege -Internal -Local -Long Distance		Sets the call privilege to internal. Sets the call privilege to local. Sets the call privilege to long distance.
Wakeup Time -Enter -Cancel -PM -Daily/Once -Personal/Auto		Allows wakeup time to be entered. Clears wakeup time once it is set. Indicates that time is PM in 12-hour format. Allows the wakeup time to be repeated daily Sets a personal wakeup call to the VIP.
Do Not Disturb		Toggling key that sets/clears Do Not Disturb.
Clear Msg Reg		Appears when there is a message register count. Clears the count. Confirmation (the Enter key) is required to clear the message register.
Audits	Message Register	Generates printout of message register counts that are greater than zero.
Wakeup		Generates printout of wakeup times.
Direct Inward Dialing		Generates a printout of room numbers and their associated DID number
Room Type		Generates printout of all rooms of a specified type showing the occupancy of each room.
Room Status -Vacant -Occupied -Reserved -Guaranteed -All Rooms -To Clean -To Inspect		Generates printout of: vacant rooms by room type or all rooms occupied rooms by room type or all rooms reserved rooms by room type or all rooms guaranteed rooms by room type or all rooms status of all rooms rooms to be cleaned by room type or all rooms rooms to be inspected by room type or all rooms.

Table: Front Desk Terminal Softkey Functions

Mode	Softkey	Function
* Defaults are programmed in CDE in Form 04, System Options/System Timers, option numbers 57 and 58.		

Front Desk Terminal Hard Key Functions

The following rules apply when the keyboard is used to enter non-softkey data such as a name, room number room code, password or terminal type. No softkeys are provided while this type of data is being entered. The RETURN hard key or the arrow keys are used to signal the end of data entry. See the Front Desk Terminal Hard Key Functions table below.

Table: Front Desk Terminal Hard Key Functions

Hard Key	Function
Typing keys	All alphanumeric characters that are allowed for Mitel telephone displays are allowed in the Guest Name field.
Numeric keys	Wake up times and room numbers can be entered with numeric keys only.
RETURN	**Room Updates Mode:** If no characters have been entered, moves the shade bar to the next field leaving the previous field unchanged. If characters have been entered, moves the shade bar to the next field and leaves the changed information in the field. **Search Display Mode:** Functions like a down arrow key.
Down, Right Arrow keys	**Room Updates Mode:** If no characters have been entered, moves the shade bar to the next field leaving the previous field unchanged. If characters have been entered, moves the shade bar to the next field leaving the previous field unchanged.
Up, Left Arrow keys	**Room Updates Mode:** If no characters have been entered, moves the shade bar to the previous field leaving the last field unchanged. If characters have been entered, moves the shade bar to the previous field leaving the last field unchanged.
Down or Up Arrow Keys	**Search Display Mode:** Moves the highlight bar up or down the list of entries.
Enter softkey	Enters the information in the field into the database. (User must exit the field first.) The Enter softkey is provided whenever a change has been made.
DELETE	If no characters have been entered, the previous softkey menu is displayed.

Table: Front Desk Terminal Hard Key Functions

Hard Key	Function
Control R or Control W	Causes the screen currently displayed to be redrawn. If data is currently being entered when Control R is used, the data for the field is lost and the original data for the field is redisplayed.
Control Y or Control C	Ends the session and drops the data call. Similar to pressing the Quit softkey on the House Statistics screen.
Control D (if D is defined as the PBX attention character for the dataset in CDE Form 11 in the Programming section.)	Ends the session. Using Control D is not recommended because it may leave the terminal in graphics mode.

Using the Front Desk Terminal

Chart: Accessing the Hotel/Motel Feature Package from a Front Desk Terminal

1.	Enter: M[onitor] HM (letters shown in [...] are optional.)	The screen displays: Ringing Connected Once connected, the screen displays: 1 - VT 100 Compatible 2 - IBM PC SELECT A TERMINAL TYPE
2.	Enter a 1 or a 2 and press Enter.	The display shows: ENTER A PASSWORD
3.	If you have made the wrong choice, use the Delete key, or drop the data call by using a special key if one is provided or waiting for the inactivity timeout to occur (approximately 10 seconds).	
4.	Enter the password of the CDE supervisor (if you are going to enter or change guest information) or CDE attendant (if you are going to request audits, look at information, or do a room search).	If a password is not entered accurately, an error message states: Incorrect Password Entered. The password must contain numeric characters only. The House Statistics screen is displayed is this Figure. If the attendant-level password was used, the Room Update softkey is not displayed.

Note: Once the SX-200 ICP System is operational, access to the hotel/motel application from the front desk terminal is via the DTRX monitor command. (Some messages may vary depending on systems options programming.)

Figure: House Statistics Screen after Log On

Chart: Refreshing the House Statistics Screen		
1.	On the House Statistics screen, press the **Refresh** softkey.	If there has been a change to the occupancy summary, room condition summary, or features usage summary, the totals are recalculated and displayed. The date and time are updated to show the current date and time.

Chart: Logging Off	
1.	Return to the House Statistics screen.
2.	Press the **Quit** softkey.

Mitel Display Telephone Access

Feature Access. Mitel display telephones with softkeys can be used as guest room telephones. They can also be used as supervisory telephones, for instance by housekeeping supervisors to monitor the progress of hotel cleaning staff. The SUPERSET 4150, SUPERSET 430, SUPERSET 4DN, and Mitel 5340 IP telephones programmed as a subattendant can set up wake-up alarm calls that ring the guest room. The SUPERSET 4150, SUPERSET 430, and Mitel 5340 IP telephones programmed as a subattendant can also activate the call blocking feature. As guest room telephones or as supervisory sets, they provide access to a limited range of hotel/motel features. For more information about using the Mitel 5340 IP Phone as a subattendant, refer to *Subattendant User Guide for the Mitel 5340 IP Phone* available at www.mitel.com.

The hotel/motel application uses the guest room telephone extension number as the room number. These extension numbers are assigned through Customer Data Entry (CDE).

Figure: Hotel/Motel Features Available on a Mitel Display Telephone

Access Mechanism. Guests can use several access mechanisms themselves on the Mitel display telephones: softkeys, hardkeys and access codes.

Softkeys. Pressing a softkey associated with the feature name on the display activates the feature (e.g., Message Waiting Clear).

Hardkeys. Pressing the Do Not Disturb key on a Mitel display telephone activates or deactivates the Do Not Disturb feature. Telephony features (such as speed call numbers) could be assigned to this row of programmable hardkeys. (Blocks of hardkeys on these telephones can be programmed as speed dial and feature access keys. See Guest Room Key Programming.)

Access Codes and Feature Codes for Information Input. Dialing a code gives the user access to a feature to input information. For instance, a code (for example, 111) gives the maid access to the maid in room feature. A number from 1 to 4 inputs information such as whether a maid is working in the room. See Maid in Room.

Access Codes for Information Display. Dialing a code gives the user access to information that is shown on the Mitel display; for example, room condition and occupancy status or maid in room status.

1476	OCC / CLEAN

The table below lists the hotel/motel features available on the Mitel display telephones related to their feature access mechanisms.

Table: Mitel Display Telephone Access Functions

Feature	Access Mechanism	Function
Set Maid in Room Status	Dial Maid in Room access code followed by the code for room activity.	Maid or supervisor can input guest room status (maid in room, no maid in room, clean, inspected) from the room telephone.
Display Room Occupancy and Condition	Dial Room Status access code followed by the room number.	Room occupancy and condition status can be displayed.
Arrow softkey	Occupancy and condition status of next rooms in numeric order can be displayed.	
Display Maid in Room Status	Dial Maid in Room Status access code.	Room numbers of all rooms with maids in them are displayed.
Arrow softkey	Room numbers of next rooms in numeric order can be displayed.	

Industry-Standard Telephone Access

Feature Access. Any other type of telephone (industry-standard or SUPERSET non-display telephone) can access an applicable subset of the hotel/motel features. See the figure below.

Figure: Hotel/Motel Features Available on Industry-Standard, SUPERSET non-display Telephones

Access Mechanism. Most Hotel/Motel features are accessed by dialing feature access codes. In addition, blocks of hardkeys on the Mitel 5010, 5212 and 5215 IP Phones, and SUPERSET 4015, SUPERSET 401+, SUPERSET 410, TeleMatrix 3000IP, and SUPERSET 3DN telephones can be programmed as speed dial and feature access keys.

The Industry-Standard Telephone Access Functions Table below shows the Hotel/Motel features available on industry-standard telephones related to their feature access codes.

Table: Industry-Standard Telephone Access Functions

Feature	Access Mechanism	Function
Message Waiting Indication	Indicator on display or audible bell.	Lighted lamp or special ring indicates there is a message waiting for the guest.
Set Maid in Room Status	Dial Maid in Room access code followed by code for room activity.	Maid or supervisor can input guest room status (maid in room, no maid in room, clean, inspected) from the room telephone.

PBX Access

The PBX provides direct access to some hotel/motel features. These are system-level features that require no user interaction.

Figure: Hotel/Motel Features Available Only from the PBX

Access Mechanism. These features are programmed through CDE.

The PBX Access Functions Table below shows the hotel/motel features accessed directly through CDE programming on the PBX and not through any terminal.

Table: PBX Access Functions

Feature	Access Mechanism	Function
Single-Line Reports	Automatic	Creates a single-line printout for certain events, such as a wakeup call going unanswered.
PMS	Automatic	Interfaces the PBX to a PMS package.
Auto Room Status Conversion/Auto Wakeup Print	Automatic	Prints a report of all rooms with a wakeup time set, and changes the room condition of occupied rooms from clean to dirty at a particular time each day.
Message Lamp Test	Automatic	Performs a test of message waiting lamps whenever the room status changes from occupied to vacant and there are no messages waiting. The message lamp test does not work on CLASS sets that have COS Option 505, ONS Stations Support CLASS enabled.

List of Features

A list of the features available on the SX-200 ICP System and an indication of the features available through each terminal are shown in the Hotel/Motel Features Table below. For information about using the Mitel 5340 IP phone as a subattendant, refer to the 5340 Subattendant User Guide available at Mitel OnLine.

Table: Hotel/Motel Features

Feature Name	Feature is available on...				
	Attendant Console	**Front Desk Terminal**	**Mitel 5020 IP Mitel 5220 IP Mitel 5224 IP Mitel 5324 IP Mitel 5330 IP SUPERSET 4025 SUPERSET 4125 SUPERSET 4150 SUPERSET 420 SUPERSET 430 SUPERSET 4DN**	**Mitel 5340 IP with Guest Room feature key programmed**	**Other Telephones [6]**
Attendant Do Not Disturb	Yes	Yes	No	No	No
Attendant Message Waiting Setup and Cancel	Yes	Yes	No	No	No
Audits					
Attendant Message	Yes	Yes	No	No	No

Table: Hotel/Motel Features

Feature Name	**Feature is available on...**				
Register Audit					
Direct Inward Dialing Number Assignment Audit	Yes	Yes	Yes	Yes	Yes
Room Status Audit	Yes	See All Room Status Audit	No	Yes	No
Vacant Room Audit	No	Yes	No	No	No
Occupied Room Audit	No	Yes	No	No	No
Reserved Room Audit	No	Yes	No	No	No
Guaranteed Room Audit	No	Yes	No	No	No
To be Inspected Room Audit	No	Yes	No	No	No
Dirty Room Audit	No	Yes	No	No	No
All Room Status Audit	See Room Status Audit	Yes	No	No	No
Room Type Audit	No	Yes	No	No	No
Wakeup Audit	Yes	Yes	No	Yes	No
Auto Room Status Conversion/Auto Wakeup Print[3]	No	No	No	No	No
Wakeups	Yes	Yes	Yes, if the SUPERSET 4150, SUPERSET 430, SUPERSET 4DN telephone is a sub attendant	Yes	No
Guest Room Wakeup	No	No	Yes[7]	Yes	Yes
Call Blocking	Yes	No[3]	SUPERSET 430 and SUPERSET 4150 sub attendant telephones	Yes	No
Call Restriction	Yes	Yes	No	No	No
Check Out (single key)	No	Yes	No	No	No
CLASS for Analog Sets					Yes
Guest Names	Yes[4]	Yes	Yes	Yes	No
Guest Room Key Programming	N/A	N/A	Yes	Yes	Yes[5]
Maid in Room	No	No	Yes	Yes	Yes
Maid in Room Display	Yes	Yes	Yes	Yes	No
Message Lamp Test	No	No	No		Yes
Message Register Display and Clear	Yes	Yes	No	Yes	No
Multi-user	Yes	Yes	Yes	Yes	Yes
Passwords	No	Yes	No	No	No

Table: Hotel/Motel Features

Feature Name	**Feature is available on...**				
PMS Interface	No	No	No	No	No
Room Condition Display					
Clean	Yes	Yes	Yes	Yes	No
Dirty	Yes	Yes	Yes	Yes	No
Out of Service	Yes	Yes	Yes	Yes	No
To be Inspected	Yes	Yes	Yes	Yes	No
Room Occupancy Display					
Vacant	Yes	Yes	Yes	Yes	No
Occupied	Yes	Yes	Yes	Yes	No
Reserved	Yes	Yes	Yes	Yes	No
Guaranteed	Yes	Yes	Yes	Yes	No
Room Status Display	Yes	Yes	No/See next feature.	Yes	No
Room Types and Room Codes	No	Yes	No	No	No
Searches					
Guest Search	No	Yes	No	No	No
Room Number	Yes	Yes	No	No	No
Room Type	No	Yes	No	No	No
Maid in Room Search	Yes	Yes	Yes	Yes	No
Room Status Search - Vacant/clean - Vacant - Out of service - Dirty - Reserved - Guaranteed - Reserved and clean	 Yes No Yes No No No Yes	 Yes Yes Yes Yes Yes Yes No	 No No No No No No No	 Yes Yes Yes Yes Yes Yes Yes	 No No No No No No No
Single Line Reports	No	No	No	No	No

1 Telephones can be programmed with this feature.

2 The PBX can be programmed to produce this audit and change room status at a particular time each day.

3 Front desk terminal displays indication that call blocking has been set, but cannot set call blocking.

4 Consoles can display guest names, but cannot enter them.

5 Mitel 5010 IP, Mitel 5212 IP, Mitel 5215 IP, SUPERSET 4015, SUPERSET 410, TeleMatrix 3000IP, and SUPERSET 3DN telephones only.

6 Mitel 5201, SUPERSET 4001, SUPERSET 4015, SUPERSET 401+, SUPERSET 410, TeleMatrix 3000IP, SUPERSET 3DN, and industry-standard telephones. The message lamp test does not work on CLASS sets that have COS Option 505, ONS Stations Support CLASS enabled.

7 This is actually in the form of a Reminder. See Reminder in the Program Features section.

Hardware Configuration

Basic hardware is described in the Hotel/Motel Hardware Table below . The following figure shows a Hotel/Motel System Configuration (Triple Play).

Figure: Hotel/Motel System Configuration

Table: Hotel/Motel Hardware

Hardware	Description
PBX	The PBX can be any hardware configuration.
Attendant Console	Provides telephony as well as access to hotel/motel features. A system supports 11 attendant consoles; however, hotel/motel applications usually use one console only. If more than one console is available on the system, each operates independently. That is, two consoles could change the same information for a room at the same time. In that case, the console that left the room last would have its changes captured by the system. If an attendant console and a front desk terminal are configured on a system, the attendant console can access a room already accessed by a front desk terminal and change the same information at the same time. However, a front desk terminal cannot access a room already accessed by a console. If the console leaves the room last, its changes are captured. The attendant console can be used in place of a VT100 terminal for maintenance and/or CDE programming. Four lines of a CDE form can be displayed at a time: three lines of information and the active line.
Guest Room Telephones	Any type of industry-standard telephone can be configured on the system. Industry-standard guest room telephones allow guests access to certain Hotel/Motel features. The number is limited by the hardware configuration of the PBX. Guest rooms using analog sets with a display can show the calling name and number if the CLASS option is enabled. See CLASS for Analog Sets.
Front Desk Terminal	The system can support up to four VT100 terminals as front desk terminals. Each terminal must connect to an IP socket in the SX-200 ICP either through a Telnet session or an RS232-to-IP serial port converter. If more than one terminal is available on the system, only one of them can make changes to the same guest room at a time. If a front desk terminal attempts to access a guest room already accessed by an attendant console, the terminal is blocked from access.
VT100 Type Terminal or VT100 Terminal Emulation Package	Used for maintenance and/or CDE programming. It displays full-screen CDE forms facilitating data entry. The maintenance terminal can be associated with a system printer to print alarm logs.
System Printer	Delivers hard copies of alarm logs and prints Hotel/Motel audits and single line reports. Separate printers can be used.
Mitel 5020 IP/5220 IP/5224 IP/5324 IP/ 5330 IP/ 5340 IP/ SUPERSET 420 / 4150/4125/ 4025/ 430/ 4DN/ Telephones	Because these telephones can display room status information, they can be used as supervisory sets by hotel staff. They can also be used as guest room telephones. Programming allows guests single-key access to designated features. For more information about using the Mitel 5340 IP Phone as a subattendant, refer to the Subattendant User Guide for the Mitel 5340 IP Phone available at www.mitel.com.

Mitel non-display Telephones	Programming non-display telephones allows guests single-key access to designated features.

Property Management System (PMS)

The PBX interfaces to IBM-compatible personal computers with the Lodgistix PMS software package (or a package that follows the same protocol). Hotel/Motel information is stored in the PBX and in the PMS. The information is split between the PMS and the PBX as follows:

PBX Database	PMS Database
	Room Status
Message Register	Message Register
Message Waiting	Message Waiting
Automatic Wakeup	Automatic Wakeup
Guest Names	Guest Names
Do Not Disturb	

Note that the PBX will not store room status information when PMS is enabled. Refer to the chart Programming for PMS for programming information.

PMS-Specific Features

The PMS interface maintains the following information between the PBX and the PMS:

- maid status
- message registration of outgoing trunk calls
- check in/out
- name (guest name)

Check In/Check Out. When a guest checks in (PMS Check In), the room telephone is enabled to allow outgoing trunk calls. Upon check out (PMS Check Out), the phone is disabled from making calls and the PMS clears message register, message waiting, Do Not Disturb and automatic wakeup from the guest room database.

The attendant may restrict the room phone to internal calls, local calls, or long distance calls by using the Outgoing Call Restriction feature described in the Attendant Console Guide.

Message Registration of Outgoing Trunk Calls. This feature provides the PMS with the number of trunk calls made from a room (local and long distance). A call-accounting device connected to the PBX monitors SMDR reports for long distance calls. The charge for these calls is automatically added to the guest’s bill at check out time. Call-costing equipment may be attached to the PMS, to allow the PMS to handle call costing.

Maid in Room Status. This feature allows the maid to change the room status (clean/dirty) from the room telephone. The maid in room status is also indicated on the PMS terminal. This feature is functionally identical to that of the PBX’s attendant room status; however, the displaying and monitoring of room status is completely controlled from the PMS terminal.

Wakeups. This feature allows the front desk PMS terminal to set up and cancel wakeup requests for guest rooms. If programmed in the PBX, a printout will be produced upon setup of a valid time to verify that the request was committed. As well, a printout occurs when the wakeup is honored to verify the receipt of the call.

Name. This feature allows the guest room name in the PMS to be sent to the PBX when a guest checks in and be stored against the room extension. This Figure provides a breakdown of the Add Name command.

Figure: Breakdown of the Add Name Command

Message Waiting. This feature allows the message center PMS terminal to setup and cancel a message waiting indication on a guest room telephone. If programmed in the PBX, a printout will be produced to verify that the message was sent and the time of the message. When the message is canceled from the PMS side, another printout is produced to verify the cancelation.

Feature Interactions

The following conditions apply to the PBX/PMS interface:

- Because all room status updates are applied against the PMS database, the following options must be disabled:
  - System Option 33 (Room Status)
  - System Option 27 (Room Status Audit)
  - System Option 34 (Auto Room Status Conversion and Wakeup Print)
  - COS option 244 (Room Status Applies)
  - COS option 608 (Telephone - Room Status Display)
- Maintenance logs record any errors found in the PMS interface. Refer to the Troubleshooting section for details.
- Automatic Wakeup Feature Access Code must not be programmed in Form 02.

Keyboard Commands

The PMS Keyboard Commands table below lists the keyboard commands for the PMS personal computer.

Table: PMS Keyboard Commands

Commands	PMS Keyboard Keys	Comments
Enquire (ENQ)	^E	^ = CTRL key
Acknowledged (ACK)	^F	
Not Acknowledged (NAK)	^U	
Start Transaction (STX)	^B	
End Transaction (ETX)	^C	
Check In	^BCHK1 nnnnn^C	nnnnn is the extension number. If the extension number is not five digits in length, use leading spaces in place of the digits.
Check Out	^BCHK0 nnnnn^C	
Add Name	^BNAM1 a-z nnnnn^C	a-z is the extension name. The extension name must be 20 characters in length followed by a single space, followed by the extension number. If the extension name is not 20 characters in length, use trailing

Table: PMS Keyboard Commands

Commands	PMS Keyboard Keys	Comments
		spaces in place of the characters. Only the first 10 characters of the name are displayed.
Replace Name	^BNAM2 a-z nnnnn^C	
Delete Name	^BNAM3 a-z nnnnn^C	
Send Message	^BMW 1 nnnnn^C	Insert a space before and after the digit 1.
Delete Message	^BMW 0 nnnnn^C	Insert a space before and after the digit 0.
Set Wakeup	^BWKPTTTTnnnnn^C	TTTT is the time in hours:minutes. Use a 24-hour clock. The nnnnn are five digits for the extension number. If the extension number is not five digits in length, use leading spaces in place of the digits.
Delete Wakeup	^BWKPSSSSnnnnn^C	This format cancels a wakeup. The four S characters indicate the cancel function.

Feature Availability

The following restrictions also apply to Lodging and Property Management System (PMS) features.

- Room Status (Option 33) cannot be enabled if PMS is enabled
- Room Status (Option 33) cannot be enabled if Outgoing Call Restriction (Option 32) is enabled
- Outgoing Call Restriction (Option 32) cannot be enabled if Room Status (Option 33) is enabled
- PMS cannot be enabled if Room Status (Option 33) is enabled
- PMS cannot be enabled if Lodging is enabled
- PMS cannot be enabled if Automatic Wakeup feature code is enabled
- Lodging cannot be enabled if PMS is enabled
- Options 4, 9, 12, 13, 23, 24, 25, and 27 cannot be enabled unless Lodging or PMS is enabled
- Lodging or PMS cannot be disabled if one or more of Options 4, 9, 12, 13, 23, 24, 25, or 27 is enabled.

Front desk terminals support the list of features in the Feature Availability table below.

Table: Feature Availability

Attendant Do Not Disturb
Guest Room Do Not Disturb
Attendant Message Waiting Setup and Cancel
Guest Message Waiting Cancel
Audits
Attendant Message Register Audit
Room Status Audit
Vacant Room Audit
Occupied Room Audit
Reserved Room Audit

Table: Feature Availability
Guaranteed Room Audit
To be Inspected (To Inspect) Room Audit
Dirty (To Clean) Room Audit
All Room Status Audit
Room Type Audit
Wakeup Audit
Auto Room Status Conversion/Auto Wakeup Print
Automatic Wakeup
Guest Room Wake Up
Call Blocking
Call Restriction
Check Out (single key)
Direct Inward Dialing Number
Guest Names
Guest Room Key Programming
Maid in Room
Maid in Room Display
Maid in Room Status Display - Mitel 5020 IP/5220 IP/5224 IP/5340 IP//SUPERSET 4025/4150/420/430/4DN Telephones
Message Lamp Test (available on the attendant console only)
Message Register Display and Clear
Multi-user
PMS Interface
Room Condition Display
Clean
Dirty
Out of Service
To be Inspected
Room Occupancy Display
Vacant
Occupied
Reserved
Guaranteed
Room Status Display
Room Types and Room Codes
Searches
Direct Inward Dialing Number
Guest Search
Room Number
Room Type
Maid in Room Search
Room Status Searches

Table: Feature Availability
- Vacant/clean
- Vacant
- Out of service
- Dirty (To Clean)
- Reserved
- Guaranteed
- Reserved and Clean
Single Line Reports

Installing the Hardware

PBX

The basic hotel/motel package is available with a PBX.

Peripherals

Hardware installation for the hotel/motel system involves installing the following peripherals:

- Attendant console(s)
- Front desk terminal(s)
- An RS232-to-IP serial port converter for front desk terminal and PMS interface (not required if using Telnet to connect to an IP socket in the SX-200 ICP)
- Printer(s)
- Mitel telephones as staff supervisory sets
- Mitel or industry-standard telephones as room telephones.

Chart: Installing Hardware

Step	Action	Comments
	Install the PBX	The PBX may already be installed at the site.
1.	Install the PBX.	Installation Information section.
	Install the Attendant Console(s)	If this is an existing installation, the attendant console(s) may already be installed.
2.	Install the attendant console(s).	Install a SUPERCONSOLE 1000
	Install the Front Desk Terminal(s)	
3.	Install up to four front desk terminals.	Documentation provided with the terminal(s).
4.	Install an RS232-to-IP serial port converter for each terminal installed in Step 3. (Not required if connecting via Telnet to an IP socket in the SX-200 ICP)	Click here for information about RS232-to-IP serial port converter.

Chart: Installing Hardware

Step	Action	Comments
	Install Printer(s)	
5.	Install printer(s).	Installation Information section.
	Install Telephones	
6.	Install Mitel telephones.	Install Mitel IP Phones and SUPERSET Telephones sections.
7.	Install industry-standard telephones.	Documentation provided with the telephones.

Programming the Hotel/Motel Application

Note: This information is in addition to the programming required for the PBX.

Chart: Programming the Hotel/Motel Application

Step	Action	Comments
1.	**Form 02: Program System Access Codes**	Access codes can range from one to five digits.
	Set Access Code for Do Not Disturb. Assign an access code for Do Not Disturb.	COS option 220 on Form 03 must also be set. The same access code sets and cancels Do Not Disturb at a guest room telephone.
	Set Access Code for Automatic Wakeup for Room Phones. Assign a one- to five-digit code for Feature 32, Automatic Wakeup.	COS options 202 and 244 on Form 03 and System Option 11 on Form 04 must also be set. System Option 1 set to 24-hour format.
	Set Access Code for Maid in Room Feature. Assign a code for Feature 35, Maid in Room.	Maid in Room allows maids to dial a telephone number when they enter a room for housekeeping and dial another number when they have finished the room.
	Set Access Code for Mitel Telephone Room Status Display. Assign a code for Feature 36, SUPERSET 4 Room Telephone Status Display.	Room Status Display allows an authorized Mitel display telephone to display the status of guest rooms. The Room Status parameters are:Vacant or Occupied or Reserved; Clean or Dirty or Out of Service or To Be Inspected.
	Set Access Code for Telephone Maid in Room Status Display. Assign a code for Feature 40, SUPERSET 4 Telephone Maid in Room Status.	Telephone Maid in Room Status Display allows an authorized Mitel display telephone to determine which guest rooms have maids in them.
	Set access code for retrieving a message from a telephone. Assign a code for Feature 42, Call	Allows a guest room telephone to call the attendant to retrieve a message.

Chart: Programming the Hotel/Motel Application

Step	Action	Comments
	Message Sender of Oldest Message.	
	Enable DID Number Display if applicable.	Enables DID Number audit for sub-attendants.
	Form 03: To Program Room Types (Front Desk Terminal)	The hotel guest rooms can be divided into 50 groups by putting each room type into a separate COS. Each COS has a name associated with it. Therefore, the room type can be identified by the COS name.
2.	**Create a COS for each different room type you have.** Assign a name to it by using the COS Name softkey.	For instance, COS 1 SINGLE single rooms COS 2 DOUBLE double rooms COS 3 NONSMOKS nonsmoking single COS 4 NONSMOKD nonsmoking double The name must be: -unique -up to eight characters in length
	Define the Characteristics of each Room	
3.	On each Form 03 you have, set the COS options for each type of room. Enable:	
	101 OG Restriction/Room Status Setup	

Allows access to room status (occupancy, condition, call restriction). It provides the STATUS softkey to the console.

Note: The front desk terminal always has access to room status. Only System Option 33 on Form 04 has to be set.		
	105 Guest Room Key	This COS option gives access to the hotel/motel features. This COS option does not have to be set for the front desk terminal. This COS option must be enabled in the 5340 subattendant phone COS for the Guest Room feature key.
	113 Attendant (Console only) Call Block Key	**Attendant Console.** This COS option allows the attendant to inhibit room-to-room calls. COS option 204 and System Option 09 must also be set. This COS option does not have to be set for the front desk terminal.
	202 Alarm Call	**Room Telephones.** This COS option allows guests to set up a wakeup alarm call from their extension. It also allows the console or the front desk to set up wakeup calls for guest rooms. See also COS option 244.

	204 Call Block Applies (Performed by Attendant Console only)	**Room Telephone.** Room telephones are prevented from calling each other when call blocking is enabled. Error (reorder tone) is heard. This COS option must be set for each room telephone call block applies to. COS option 113 and System Option 09 must also be set.
	220 Do Not Disturb	**Room Telephone.** Do Not Disturb must be enabled for any telephone that requires use of the feature. When Do Not Disturb is active, the telephone appears busy to incoming calls.
	230 Message Register Overflow Alarm	**Extension or Trunks.** Optional. This COS option should be set for the extension(s) or trunk(s) for which there is to be a message register overflow alarm indication at the attendant console. The alarm icon flashes and a maintenance log is generated. See also COS option 703. See also the following system options: System Option 49 System Option 04 System Option 23 System Option 40
	231 Message Waiting Bell	**Room Telephones**. This COS option rings the bell on a room telephone when there is a message waiting. See also System Option 02 (Form 04). COS options 231 and 232 cannot be enabled at the same time. See Note.
	232 Message Waiting Lamp	

Room Telephones. This COS option lights the message lamp on a room telephone when there is a message waiting. See also System Option 02 (Form 04). COS options 231 and 232 cannot be enabled at the same time.

Note: COS option 231 rings a message bell only. COS option 232 lights message lamps.		
	244 Room Status Applies	**Room Telephones.** This COS option must be enabled to allow guests to set up a wakeup alarm call from their extension. See also COS option 202. It must be enabled to allow the attendant to display and change the status of a room and produce a room status audit. See Form 04, System Option 33. It must be enabled for SUPERSET 4150 or SUPERSET 430 room telephones that are to be programmed with speed dial and feature access keys. See COS option 610 and Form 37.

	320 Transparent Multi-Console Operation	**Attendant Consoles.** Enables more than one console on the system to respond to recalls, message waiting, and to read messages. Can also allow multi-consoles to view and remove messages for guest room phones. COS option 320 is necessary for consoles only. It is not necessary for front desk terminals.
	608 Telephone - Room Status Display	**Mitel Supervisory Telephones.** This COS option allows room status to be displayed on a Mitel IP or SUPERSET supervisory telephone.
	610 Telephone - Guest Room Template	

Mitel Room Telephones. This COS option allows a block of guest room telephones (Mitel IP or SUPERSET) to be programmed with speed dial and feature access keys.

NOTE: Applying a Guest Room template to 53xx series phones will not override keys that have been programmed using the Superkey or Applications key.		
		See also COS option 244. See also Form 37: Guest Room SUPERSET Keys Template.
	623 Automatic DID Number	**Direct Inward Dialing**.This option only applies to devices that can be checked in or out. When enabled, DID number is automatically assigned at check-in.
	703 Message Register Applies	**Attendant Console, Telephones, Mitel Telephones, Trunks.** This COS option provides the PBX the ability to keep track of the number of completed calls, and identifies the trunks on which counts apply on a per phone basis. See COS option 230. See System Option 49 in Form 04 and optional message register features programming: System Option 04 System Option 23 System Option 40.
	Setting System Options	
4.	**On Form 04, set the characteristics of the system.**	

	01 Clock Format	Option 01 sets one of three clock formats. The three clock format options appear on CDE softkeys: 24 HOUR, 12 HOUR, and 12 HOUR AM. The 24 HOUR enables a 24-hour clock (example 16:00). The 12 HOUR enables a 12-hour clock (example 4:00). The 12 HOUR AM enables a 12-hour clock with an "a" to represent a.m. (example 4:00a). There will be no letter to represent the p.m.
	02 Message Lamp Test Enable	System option 02 allows the message waiting lamps on room telephones to be tested from the console. See also COS option 232 (Form 03). The message lamp test does not work on CLASS sets that have COS Option 505, ONS Stations Support CLASS enabled.
	04 Message Waiting and Message Register Clear Print	Optional. Prints a single line report to the default printer each time the message register changes, including the case where the message register is cleared by the console. This system option also requires System Option 107 or 108 to be enabled. See also COS option 230 COS option 703. See also the following system options: System Option 49 System Option 23 System Option 40. See also System Option 13, Auto Wakeup Print, for enabling single line reports of wakeups.
	09 Attendant Call Block	This system option allows the attendant to inhibit room-to-room calls. COS option 113 must also be set. This system option also requires System Option 107 or 108 to be enabled.
	11 Automatic Wakeup	Allows the attendant or guest to set up a wakeup alarm call. COS option 202 and 244 must be enabled for all extensions that are to use automatic wakeup.
	12 Automatic Wakeup Alarm	**Optional.** Allows the console alarm icon to be flashed in the event that a wakeup call is not answered within three attempts. Also generates a maintenance log. No effect on SUPERSET 420, SUPERSET 430, and SUPERSET 4DN telephones.
	13 Automatic Wakeup Print	**Optional.** Allows wakeup audits and one-line messages to be printed on the system printer whenever a wakeup call is set up, canceled, answered, or an attempt is made to honor the wakeup. This system option also requires System Option 107 or 108 to be enabled.
	14 Automatic Wakeup Music	**Optional.** Allows music to be heard by an extension that answers a wakeup call. Requires a music source.
	19 DID Server Application	**Optional**. Enables the Direct Inward Dialing (DID) Server Application which supplies automatic DID number assignment.

	23 Message Reg. Count Additional Supervisions	Counts meter pulses after supervision is received on an outgoing trunk. This system option also requires System Option 107 or 108 to be enabled.
	24 Message Register Audit	Allows the attendant to request a message register audit (either from the attendant console or the front desk terminal). See System Option 25. This system option also requires System Option 107 or 108 to be enabled.
	25 Message Register Zero After Audit	All message registers are set to zero when an audit is requested, including instances where the maximum number of audits is exceeded or the printer is unavailable. See System Option 24. This system option also requires System Option 107 or 108 to be enabled.
	27 Room Status Audit	**Console.** Produces a printout of the current status of all guest rooms. Only rooms with COS option 244 enabled are displayed by the audit. See COS option 244, Form 03. This system option also requires System Option 107 to be enabled. **Terminal.** Produces seven different types of audits (vacant, occupied, reserved, guaranteed, to clean, to inspect, all rooms).
	33 Room Status	Allows the status of a room to be displayed and changed. Provides the three call privileges: internal, local and long distance. See also: COS option 244 (for all extensions for which room status is to apply), System Option 34. If System Option 02 is enabled, when a room’s status is changed from vacant to occupied or occupied to vacant, the system performs a message lamp test on the telephone in that room. If the test finds out a lamp is defective, it raises a minor alarm. The message lamp test does not work on CLASS sets that have COS Option 505, ONS Stations Support CLASS enabled.
	34 Auto Room Status Conversion/Auto Wakeup Print	**Optional.** Produces an automatic audit of all wakeups in the system every day. This option also changes the room status. For instance, all occupied/clean rooms can be changed to occupied/dirty at a set time. The time of day for the printout must be set in System Option 56 . The default is 00:00 (midnight). See System Option 56.
	40 Message Register Follows Talker	Charges the extension or console last talking to a trunk with the message register count.
	49 Pseudo Answer Supervision Timer	This value is applied to trunks in the system that do not provide answer supervision.

	56 Auto Room Status Conversion/Auto Wakeup Print Timer	**Optional.** Sets the time of day for the automatic daily audit of all wakeups in the system. The default is 00:00 (midnight).Only applies if System Option 34 is enabled. See System Option 34. The same time is also used to convert all occupied/clean rooms to occupied/dirty.
	57 Vacant/Reserved Room Default Call Restriction	By default, vacant rooms (and rooms that become vacant) have their call restriction set to Internal, the most restrictive option. This setting prevents unauthorized calls from vacant/reserved rooms.
	58 Occupied Room Default Call Restriction	By default, occupied rooms have the call restriction set to Internal, the most restrictive option. Typically when a room becomes occupied, the guest is provided with a less restrictive setting.
	102 Feature Level (level 1 or greater)	Supports daily, multiple, and personal wakeup calls and call blocking from a sub attendant.
	107 Lodging	
	Note:	
	32 Do not enable System Option 32, Outgoing Call Restriction, for Hotel/Motel.	Prevents unauthorized trunk calls after a guest checks out. However, it cannot be enabled at the same time as System Option 33, Room Status. Since System Option 33 provides the three call restrictions, System Option 32 is not necessary in the hotel/motel context. If a PMS is used, System Option 32 must be enabled.
	To Set Up the Attendant Console	
5.	Form 07: Console Assignments specifies the physical location of each attendant console.	Described in the Program CDE section.
6.	Form 08: This form specifies the listed directory numbers (LDN) for each attendant console.	Described in the Program CDE section.
7.	Form 19: This form defines the LDN for the personal wakeup calls for the attendant consoles and subattendants.	Described in the Program CDE section.
	To Set Up Telephones	
8.	Form 09 assigns stations and Mitel telephones to the system. See Note 2 for extension numbering instructions.	

Described in the Program CDE section.

SUPERSET 430, SUPERSET 4DN, and SUPERSET 4150 telephones can be programmed as subattendants and can then program wakeup calls. Personal wakeup calls require routing of the LDN key in Form 19.

Notes:

1: If a front desk terminal is configured on the system, the names of guests in occupied rooms are displayed in the Name field on Form 09. Only the first 10 characters of the last name can be displayed.

2: The hotel/motel application uses the guest room extension number as the room number. A simple pattern for relating an extension number to a room number should be used. For example, if four-digit extension numbers are used, they could be assigned by using format xfrr:

where:
- x is always a 1
- f is the floor number
- rr is the room number on the floor.

Therefore:
- Room 1 on floor 1 is 1101
- Room 2 on floor 1 is 1102
- Room 1 on floor 2 is 1201.

To Set Up the Front Desk Terminal

9. Form 12 sets up IP Sockets (Allows IP-enabled terminals to connect to the SX-200 ICP via Telnet; terminals requiring a serial connection must use a 3rd-party RS232-to-IP serial port converter to connect to the network).

Described in the Program CDE section.

See RS232-to-IP Serial Port Converter Setting for additional programming requirements.

Form 11: Data Circuit Descriptors.	Described in the Program CDE section.
Form 11: Data Circuit Descriptor Options Subform.	Described in the Program CDE section.

		Note: When programming the PBX attention character to be used with the front desk terminal, Control R, Control Y and Control C should not be used. The Front Desk Terminal application has predefined functions for these key sequences. No checking is done to prevent the use of these characters.
	To Set Up Access Restriction	
10.	Form 28: Access Restriction form.	Described in the Program CDE section. The CDE supervisor-level password allows editing privileges. The CDE attendant-level password allows no editing privileges.
	To Set Up Printer(s)	
11.	Form 34: Directed I/O.	Described in the Program CDE section. The hotel/motel wakeup printer prints single line reports. The hotel/motel audit printer prints wakeup, message register, room type, room status, and wakeup/room condition audits.
	To Set Up Mitel Telephones Block Programming	
12.	Form 37: Guest Room Key Template	

This form provides three preprogrammed sets (templates) of speed dial and feature access keys for hotel/motel guest room phones.

COS options 244 and 610 must be programmed in the COS of all guest rooms using Mitel display telephones with speed dial and feature access keys.

NOTE: Applying a Guest Room template to 53xx series phones will not override keys that have been programmed using the Superkey or Applications key.

Programming for PMS

Chart: Programming for PMS

Notes:

1. The PBX can be interfaced to a Property Management System.
2. An RS232-to-IP serial port converter, such as the Precidia Technologies Ether232 or iPocket232 (available from the vendor or its resellers) is required for PMS applications that only support serial connectivity to the SX-200 ICP. IP-enabled applications can connect via Telnet.
3. Maintenance logs record any errors in the PMS interface. Refer to the Troubleshooting section for details.
4. Message Waiting Setup must be enabled in the COS for the guest room phone if it is an industry-standard telephone. This restriction does not apply to the Mitel IP telephones and to SUPERSET 400 and 4000 series telephones.
5. PMS Terminal. Dial 0 calls are routed to a console (Form 19) or a console LDN or sub- attendant LDN (Form 08).

Chart: Programming for PMS

This routing makes sure there is a single place the guest must call to receive messages. The dial 0 point for the room must be a console or a console LDN or subattendant LDN for the message softkey to work properly

6. A list of the PMS Keyboard Commands is provided in the PMS Keyboard Commands table.

Step	Action	Comments
	Setting Feature Access Codes	
1.	On Form 02, Programming Feature Access Codes:	
	Assign an access code to Feature 35 (Maid in Room).	To allow room status to be changed from room telephones.
	Assign an access code to Feature 10 (Do Not Disturb).	Allows a room telephone to set up and cancel Do Not Disturb. See also COS option 220.
	Assign Feature Access Code 32 (Automatic Wakeup) if required.	Assign the Feature Access Code if the phones will be setting wakeups through Embedded Voice Mail.
	Setting COS Options	
2.	Enable COS option 105 (Attendant Guest Room Key).	To allow the attendant to set wakeups. (see also System Option 11, Form 04.) Also allows the attendant to set up or cancel Do Not Disturb. Also allows the attendant to set up or cancel message waiting for a room telephone. See also COS option 231 (Bell) or 232 (Lamp).
	Enable COS option 202 (Alarm Call).	Enabling this COS option allows the attendant to program wakeup alarm calls for rooms from the PMS PC. See also COS option 244.
	Enable COS option 220 (Do Not Disturb).	Allows guests to set up and cancel Do Not Disturb from their extension. See System Option 10 (Do Not Disturb), Form 04.
	Enable COS option 231 (Bell).	**Optional.** Indicates a message waiting at a room telephone. See also COS option 105.
	Enable COS option 232 (Lamp).	Indicates a message waiting at a room telephone. See also COS option 105.
	Disable COS option 244 (Room Status Applies).	All room status updates are applied against the PMS database.
	Disable COS option 608 (Telephone - Room Status Display).	All room status updates are applied against the PMS database.
	Enable COS option 703 (Message Register Applies).	To charge guests for outgoing trunk calls.
	Setting System Options, Form 04 (System Options/System Timers):	
3.	Enable System Option 04 (Message Waiting and Message Register Clear	The message register is cleared in the PBX on checkout. System Option 04 produces a record of it.

Chart: Programming for PMS

	Print).	Also produces a record of messages still waiting on the room telephone.
	Enable System Option 11 (Automatic Wakeup).	To allow the attendant to set automatic wakeups. See also COS option 105, Form 03.
	Enable System Option 13 (Automatic Wakeup Print).	For a record of wakeup cancelations at checkout time.
	Disable System Option 27 (Room Status Audit).	All room status updates are applied against the PMS database.
	Enable System Option 32 (Outgoing Call Restriction).	To prevent unauthorized trunk calls after the guest has checked out.
	Disable System Option 33 (Room Status).	All room status updates are applied against the PMS database.
	Disable System Option 34 (Auto Room Status Conversion and Wakeup Print).	All room status updates are applied against the PMS database.
	Enable System Option 108 (Property Management System).	
4.	**Define the Data Circuit**	

Form 11: Define a data circuit descriptor to match the RS-232 interface of the PMS in CDE Form 11 (Data Circuit Descriptor Options Subform).

Note: If using a 3rd-party RS232-to-IP serial port converter, program the converter to match the descriptor. See the converter documentation for further details.	The PMS documentation has further details. A typical RS-232 interface for the PMS port is: 7 ASCII data bits No parity One stop bit 1200 baud.	
5.	Form 12: Define the data circuit (i.e., the IP socket) used for connection to the PMS in CDE Form 12 (Data Assignment).	Described in the Program CDE section.
6.	**Form 19 (Dial 0 calls)**	

If Dial 0 calls are routed for a guest room, program Form 19. Enter Day, N1 and N2.

The Dial 0 point must be a console, a console LDN or subattendant LDN.

Day, N1 and N2 can be the same number.

Note: If Day, N1 and N2 numbers are not provided, it is impossible to set message waiting from the PMS terminal. However, it can be done from the attendant console.		
7.	Form 34: Configure the IP socket port for PMS application in CDE Form 34 (Directed I/O).	Described in the Program CDE section.

Hotel/Motel Features

Each Hotel/Motel feature is listed alphabetically below with the sections:

- Description
- Conditions
- Programming
- Operation

Where possible, features have been associated into groups. For example, the various types of audits are described under the main heading Audits.

Audits

An audit is a printed report that gives a record of guest room activities or conditions. There are several types of audits. Most are requested by the attendant, but one (the Auto Room Status Conversion/Auto Wakeup Print) is programmed to be printed automatically each day. All audit requests are directed to the printer assigned to the Hotel/Motel Audit printout type in CDE Form 34. The Audits by Terminal Type Table below shows the audits that can be requested from each terminal.

Table: Audits by Terminal Type

Audit Types	Terminal	Operation
Attendant message register audit Room status audit Wakeup audit	Attendant Console	The audits mode of operation is entered through the AUDIT softkey.
Attendant message register audit DID Number Display Room status audits - Vacant room audit - Occupied room audit - Reserved room audit - Guaranteed room audit - Dirty (To clean) condition audit - To be inspected (To inspect) condition audit - All rooms audit Room type audit Wakeup audit	Front Desk Terminal	The Audit mode of operation is entered from the main menu by pressing the Audits softkey. The softkeys are updated to show the various audit types. System options must be enabled to allow the various types of audits, or the corresponding softkey is not provided.

Table: Audits by Terminal Type

Audit Types	Terminal	Operation
Auto Room Status Conversion/Auto Wakeup Print	N/A	Programmed to be printed automatically each day.

Attendant Message Register Audit

Description

The message register can be printed for any room that has a message register count greater than zero. The audit report lists the rooms in order of their room number (lowest to highest) and the current register count is given for each. The message register audit is shown in the ACD SUPERVISION (Screens) Figure. For more information, see Message Register.

Figure: Message Register Audit

Conditions

- A console cannot request an audit if it is active in a call. A console can request any number of audits at one time, but only the first three are printed. The requests are processed in a first requested, first printed sequence.
- If System Option 25 (Message Register Zero After Audit) is enabled, all message registers are set to zero when the audit is requested. This includes the instance where the maximum number of audits is exceeded or the printer is unavailable for any reason.
- Printing the message register for rooms depend upon them having a message register count greater than zero.

Programming

See the programming for Message Register.

In CDE Form 03, enable COS Option 703, Message Register Applies, for the extensions.

In CDE Form 04, select System Option 24 Message Register Audit. Select System Option 25, Message Register Zero After Audit, to reset all message registers to 0 after an audit request.

In CDE Form 34, Directed IO, assign a printer to the “Hotel/Motel Audit” printout type.

Operation

Requesting a Message Register Audit - Attendant Console

- Press **FUNCTION**. You must not be involved in a call.
- Press the **GUEST ROOM** softkey. The AUDITS softkey appears.
- Press the **AUDITS** softkey. The AUDIT softkeys appear.
- Press the **MSG REGISTER** softkey.
- When finished, press the **EXIT** softkey.

Requesting a Message Register Audit - Front Desk Terminal

- From the House Statistics screen, press the **Audits** softkey.The Audit softkeys are displayed (MSG REG and WAKEUP).
- Press the **Msg. Reg.** softkey. A report of all guest rooms with a message register greater than zero is printed.

- After you have finished, press the **Quit** softkey.

Direct Inward Dialing Audit

Description

The Direct Inward Dialing (DID) audit produces a report of room numbers and their assigned DID numbers.

Figure: DID Audit - (Attendant Console)

Figure: DID Audit (Sub-attendant Display)

Conditions

- For attendant console and subattendants, a printer must be assigned to the Hotel/Motel DID type in CDE Form 34 (Directed IO).

Programming

In CDE form 34, add a new printer using the HM DID softkey. By default, there are no Hotel/Motel DID printers defined.

Notes:

- The DID prompt is only displayed if system option DID Server Application is Enabled.
- If the DID Server Application and ACD are both enabled, the HM DID softkey does not appear on the main screen. Press the **MORE** softkey to access the HM DID softkey in this instance.

Operation

Requesting a DID Number Audit - Attendant Console

- Press **FUNCTION**. You must not be involved in a call.
- Press the **GUEST ROOM** softkey. The AUDITS softkey appears.
- Press the **AUDITS** softkey. The Audit softkeys appear.
- Press the **Print DID** softkey.
- When finished, press the **EXIT** softkey

Requesting a DID Number Audit - Sub-attendant

- Enter the feature access code for DID Number followed by the guest room number. The DID Number appears and the PRINT DID softkey displays.
- Press PRINT DID to send the DID number to the Hotel/Motel DID printer. **Note**: This softkey only appears if a DID number is assigned to the selected room.

Room Status Audit(s)

The attendant can request a report of the current status of all guest rooms. The attendant console produces one audit that contains information on all rooms. The front desk terminal can produce two types of room status audits: room occupancy audits (vacant, occupied, reserved and guaranteed audits) and room condition audits (to clean and to inspect audits). Each of these audits can be requested by room type (see Room Type) or for all rooms. In addition, an all rooms audit can be requested from the front desk terminal. This request is the equivalent of the attendant console room status audit (see the Room Status Audits Table below). Also see Room Occupancy and Room Condition.

Each audit (console or front desk terminal) shows the call restriction status of the room (see Call Restriction) and whether there is a maid in the room (see Maid in Room). The attendant console audit is shown in four parts: Room Audit For all Vacant Rooms, Room Audit for all Occupied Rooms, Room Audit for all Reserved Rooms, and Room Audit for all Guaranteed Rooms. A front desk terminal audit shows the Room Status Audit for All Vacant Rooms. For more information see Room Status Display.

Table: Room Status Audits

Terminal	Name	Content	Categories
Attendant Console	Room Status Audit	All rooms by status and condition.	Four parts: vacant, occupied, reserved, or guaranteed rooms. Each part is divided by room condition: clean, dirty, out of service, or to be inspected.
Front Desk Terminal	**Room Occupancy Audits (All Rooms or by Room Type)**		
Vacant Room Audit	All guest rooms with room occupancy status of vacant.	Listed by condition: clean, dirty, to be inspected, out of service.	
Occupied Room Audit	All guest rooms with room occupancy status of occupied.	Listed by condition: clean, dirty, to be inspected, out of service.	
Reserved Room Audit	All guest rooms with room occupancy status of reserved.	Listed by condition: clean, dirty, to be inspected, out of service.	
Guaranteed Room Audit	All guest rooms with room occupancy status of guaranteed.	Listed by condition: clean, dirty, to be inspected, out of service.	
Room Condition Audits (All Rooms or by Room Type)			
To Clean Room Audit	All guest rooms with room condition of dirty.	Listed by occupancy: vacant, occupied, reserved, guaranteed.	
To Inspect Room Audit	All guest rooms with room condition of to be inspected.	Listed by occupancy: vacant, occupied, reserved, guaranteed.	
Room Occupancy and Condition			
All Rooms Audit	All rooms by status and condition.	Listed by occupancy: clean, dirty, to be inspected, out of service categories for each of: vacant, occupied, reserved, and guaranteed rooms.	

Conditions

- A four-part audit report is printed on the system printer showing all extensions with room status. One part lists vacant rooms, one lists occupied rooms, one lists reserved rooms, and one lists guaranteed rooms. Each contains a subpart for clean rooms, dirty rooms, inspected rooms, and out of service rooms, and shows the call restriction and maid in room status of each room.

- This audit displays only rooms with COS option 244, Room Status Applies, enabled.
- A printer must be assigned to the “Hotel/Motel Audit” printout type in CDE Form 34 (Directed IO).
- Pressing the Audits key with the Front desk terminal does not update the terminal screen. The softkeys are updated to show the various audit types.
- A maintenance log is generated if the Hotel/Motel printer fails.

Programming

See the programming for Room Status Display.

In CDE Form 03, select COS Option 244 (Room Status Applies) for all extensions for which room status is to apply.

In CDE Form 04, enable System Option 27 (Room Status Audit) to allow room status printouts.

Operation

Requesting a Room Status Audit - Attendant Console

- Press **FUNCTION**. You must not be involved in a call.
- Press the **GUEST ROOM** softkey. The AUDITS softkey appears.
- Press the **AUDITS** softkey. The AUDIT softkeys appear.
- Press the **ROOM STATUS** softkey.
- When finished, press the **EXIT** softkey.

Requesting a Room Status Audit - Front Desk Terminal

- On the House Statistics screen, press the **Audits** softkey. The Room Status softkey is displayed.
- Press the **Room Status** softkey. The room status softkeys are displayed. For each of these audits, you are prompted to enter the code for the room type, or 0 for all rooms.
  - If you want to print a Vacant Room Audit, press the **Vacant** softkey. The vacant room audit is printed.
  - If you want to print an Occupied Room Audit, press the **Occupied Room** softkey. The Occupied room audit is printed.
  - If you want to print a Reserved Room Audit, press the **Reserved Room** softkey.The Reserved room audit is printed.
  - If you want to print a Guaranteed Room Audit, press the **Guaranteed Room** softkey. The Guaranteed room audit is printed.
  - If you want to print a To Clean Audit, press the **To Clean** softkey.The Rooms to be Cleaned audit is printed. The report lists all rooms that are currently dirty.
  - If you want to print a To Inspect Audit, press the **To Inspect** softkey.The Rooms to be Inspected audit is printed. The report lists all rooms with status set to Inspect.
- Type the room code and press **Return.**
- To print an All Rooms Audit, press the **All Rooms** softkey. The vacant, occupied, reserved, and guaranteed room audits are combined into a single printout. Each part has the same format.
- To return to the main Audits menu, press the **Quit softkey.**

Room Type Audit

Description

The Room Type audit produces a report of all guest rooms of a particular type; it gives their room number, occupancy, and whether there is a maid in the room. This report can only be requested from a front desk terminal.

Figure: Room Type Audit (Front Desk Terminal)

Conditions

- The Room Type audit can only be requested from a front desk terminal.

Programming

None.

Operation

To print a Room Type Audit from the Front Desk Terminal.

- From the House Statistics screen, press the **Audits** softkey. The Audit softkeys are displayed.
- Press the **Room Type** softkey. When a Room Type Audit is requested, you are prompted to enter a room code for the type of room you want information on.
- Enter a valid room code. It is not possible to print an all room types audit by using 0.
- Press the **Return** hardkey. An audit of all rooms of that type is printed.
- When you have finished, press the **Quit** softkey.

Wakeup Audit

Description

The Wakeup Audit produces a report of all guest rooms that have wakeup calls enabled. The format of the printout is identical, whether requested by the attendant console or front desk terminal. The wakeup audit shows the wakeup times for all extensions with active wakeups set. The audit can also be printed automatically daily at a predetermined time. The audit report lists all room numbers and the wakeup times currently set.

Figure: Automatic Wakeup Audit

Conditions

- The format of each report is the same as the format of a report when a console is used to request a report from the basic hotel/motel system.
- Pressing the associated softkey starts the audit print. The screen displays a message to indicate that printing has begun.
- A printer must be assigned to the “Hotel/Motel Audit” printout type in CDE Form 34 (Directed IO)

Programming

In CDE Form 04, select System Option 13 (Automatic Wakeup Print) to allow wakeup audit printouts and one-line messages to be printed on the system printer whenever a wakeup call is set up, canceled, answered, or an attempt is made to honor the wakeup.

Operation

Requesting a Wakeup Audit - Attendant Console

- Press **FUNCTION**. You must not be involved in a call.
- Press the **GUEST ROOM** softkey. The AUDITS softkey appears.
- Press the **AUDITS** softkey. The AUDIT softkeys appear.
- Press the **WAKEUP** softkey.

- When finished, press the **EXIT** softkey.

Requesting a Wakeup Audit - Front Desk Terminal

- From the House Statistics screen, press the **Audits** softkey.The Audit softkeys are displayed (MSG REG and WAKEUP).
- Press the **Wakeup** softkey. A report of all guest rooms with a wakeup call enabled is printed.
- When finished, press the **Quit** softkey.

Wakeup/Room Condition Audit

Description

The Wakeup/Room Condition audit produces a report of all guest rooms that have wakeup calls enabled. At the same time, the occupancy and condition of all occupied/clean rooms are changed to occupied/dirty. The PBX is programmed through CDE to print the audit and change the occupancy/condition automatically at a preset, user-selected time each day.

Conditions

- System Option 34 produces an automatic audit of all wakeups in the system every day. This option also changes the room status. All occupied/clean rooms can be changed to occupied/dirty at a set time. (This room status is the only status that can be changed automatically). The time of day for the printout and the condition change must be set in System Option 56. The default is 00:00 (midnight).
- System Option 56 sets the time of day for the automatic daily audit of all wakeups in the system. The default is 00:00 (midnight). This only applies if System Option 34 is enabled.

Programming

In CDE Form 04, enable option 34, Auto Room StatusConversion/Auto Wakeup Print and option 56, Auto Room Status Conversion/Auto Wakeup Print Timer.

Operation

None.

Call Blocking

Description

The Call Blocking feature can be set from attendant consoles or from SUPERSET telephones or Mitel IP Phones that are programmed as subattendants. Call Blocking allows the attendant or subattendant to inhibit room-to-room calls. Calls to the attendant, subattendant, or to extensions without the call blocking COS option selected can be made normally. Attempted calls between restricted extensions are treated as illegal numbers and the calling party hears reorder tone. A call blocking indication is displayed on the House Statistics screen of the front desk terminal.

Users of SUPERSET telephones and IP Phones programmed as subattendants can activate and deactivate Call Blocking with the feature key, Call Block. The Call Block feature key will toggle the Call Block condition on and off in the same way as the attendant console hard key does.The LED asssociated with the Call Block feature key toggles on and off along with the Call Block condition.

Conditions

- For attendant consoles with the feature enabled, the LED associated with the Block hardkey is updated each time call blocking is turned on or off. When the LED is on, call blocking is enabled. Consoles without the feature enabled do not have an operational key and LED.
- Tenanting does not apply to the operation of this feature. All consoles with the feature enabled can enable or disable call blocking throughout the system.
- If Call Rerouting is used, blocked calls are treated as Station Illegal Number Routing. Call rerouting can be used to intercept blocked calls to an appropriate destination such as the attendant console.
- SUPERSET 430 and SUPERSET 4150 telephones, and Mitel 5020, 5220, 5224, 5330, and 5340 IP Phones programmed as subattendants must have the following options enabled:
  System Option 9 (Attendant Call Block)
  COS Option 113 (Attendant Call Block Key)
  FOR Option 102 (Feature Level) set to level 1 or greater.
  The Call Block feature must be programmed with CDE Form 09 or with the SUPERKEY on the subattendant telephone.

Programming

In CDE Form 03, enable COS option 113 (Attendant Call Block Key) for the attendant console or the subattendant telephone which can apply the call block. Enable COS option 204 (Call Block Applies) for the extensions to which call block will apply.

In CDE Form 04, enable System Option 09 (Attendant Call Block). Enable FOR Option 102 (Feature Level) to level 1 or greater for the subattendant telephones that want to activate call blocking.

For users of SUPERSET telephones and IP Phones programmed as subattendants, program a Call Block feature key in CDE Form 09. If not programmed in CDE Form 09, users of these telephones must program a Call Block feature key. See Feature Keys.

Operation

Attendant Consoles

To Activate Call Blocking:

- Press the BLOCK hardkey. The associated LED is lit. All calls are blocked between all extensions with the call block option (204) enabled in their COS.

To Deactivate Call Blocking:

- Press the BLOCK hardkey. The associated LED is turned off. The call blocking is removed.

SUPERSET 430, SUPERSET 4150 telephones (Subattendants)

To Activate Call Blocking:

- Press the Call Block feature key. The associated LED is lit. All calls are blocked between all extensions with the call block option (204) enabled in their COS.

To Deactivate Call Blocking:

- Press the Call Block feature key. The associated LED is turned off. The call blocking is removed.

5020, 5220, 5224, 5324, 5330, and 5340 IP telephones (Subattendants)

To activate Call Blocking:

- Press the Call Block feature key. The associated LED is lit. All calls are blocked between all extensions with the call block option (204) enabled in their COS.
- Press the STATIONS? softkey.

To Deactivate Call Blocking:

- Press the Call Block feature key. The associated LED is turned off. The call blocking is removed.

Call Restriction

Description

The Call Restriction feature is used to assign a level of calling privilege to each guest room. There are three levels:

- internal
- local
- long distance

The attendant can assign any of the levels to a particular room telephone, or to all the telephones in a guest suite. In addition, the system automatically sets the call restriction for a room (to a programmable value) when an occupancy change to either vacant reserved, guaranteed or occupied occurs. On the front desk terminal screen, this feature is called Call Privilege.

For more information about Call Restriction in Guest Suites, see Suite Services.

Conditions

- System Options 32 (Outgoing Call Restriction) and 33 (Room Status) are mutually exclusive. System Option 32 cannot be used in programming Hotel/Motel.
- A user using PMS may restrict the room phone to internal calls, local calls, or long distance calls using the Outgoing Call Restriction feature described in the Attendant Console Guide.
- By default, vacant rooms (and rooms that become vacant) have their call restriction set to Internal, the most restrictive option. This setting prevents unauthorized calls from vacant rooms.
- By default, occupied rooms have the call restriction set to Internal, the most restrictive option. Typically, when a room becomes occupied, the guest is provided with a less restrictive setting.
- When System Option 32 (Outgoing Call Restriction) is enabled, the Redial key will not appear on phones that have an internal restriction applied to them via PMS or the Attendant Console. To restore Redial functionality to these phones, set local and long distance permissions permanently.

Programming

In CDE Form 03, select COS option 101 (Attendant O/G Restriction/Room Status Setup) for the attendant console

In CDE Form 04, enable System Option 33 (Room Status).

In CDE Form 04, set a default Vacant Room Status Call Restriction type via System Option 57 (Vacant Room Default Call Restriction).

In CDE Form 04, set a default Occupied Room Status Call Restriction type via System Option 58 (Occupied Room Default Call Restriction).

Operation

Attendant Consoles

To Check/Change the Status of a Room (Call Restriction, Room Occupancy, Room Condition, Maid)

Note: The status of the room can be checked from: an idle postion beginning with the first step, or while in a converstation with the room to be checked by pressing the Status softkey.

- Press **FUNCTION**.

- Press the **GUEST ROOM** softkey.
- Enter the extension (room) number. The STATUS softkey is displayed. The second line of the display shows the room's status. For example, Status = Vac/Dirty/Int/Maid means the room is vacant, dirty, has internal telephone privileges, and has a maid in it.
- Press the **STATUS** softkey. The Room Status softkeys are displayed. These softkeys change according to the status of the room. For instance, if the room is dirty, the CLEAN softkey appears. If the room is clean, the DIRTY softkey appears.
  The second line of the display shows the room's status.

If you want to change an element of the status, press the softkey for the desired status. The MORE... softkey returns you to the previous display.

Front Desk Terminal

To Change the Call Privilege

- On the Room Update screen, move the shaded bar to the call privilege field and press the **Show Choices softkey.** The call privilege softkeys are displayed:
  Internal: Most restrictive setting. Can only make internal calls.
  Local: Can only make local calls.
  Long Distance: Least restrictive setting. Can make long distance, local and internal calls.
- Press the softkey for the call privilege setting you want. The shaded bar is automatically moved to the next field that can be changed.
- Continue making changes, or

To end and save all changes, press the **Enter** softkey, then the **Quit** softkey.

To end and not save the changes, press the **Quit** softkey and follow the instructions given.

If no changes are necessary**,** press the **Quit** softkey.

Check Out

Description

The Check Out feature is available to the front desk terminal only. The Check Out softkey provides a simple checkout procedure. It is only provided if the occupancy setting of the guest room is Occupied. It has these functions:

- Sets the room occupancy field to vacant. Sets the room condition to dirty.
- Clears all calling names and numbers stored on the analog telephone in the guest room.
- Clears the message register, message waiting, DND settings, individual call forwarding, and wakeups for the primary and secondary telephones in a suite. The Guest Name is only deleted on the Primary telephone.
- Sets the call restriction to the setting specified in System Option 57 (Vacant/reserved default call restriction).

Conditions

- Once the check out key is pressed, the softkeys are updated to present **Cancel** and **Enter** keys only. **Cancel** resets the room conditions as they were originally. When **Enter** is pressed, the guest name is erased on the primary telephone only, and on the primary and secondary telephones the DND is turned off (if it had been set), the message waiting is turned off, the message register is cleared, the individual call forwarding programmed is cleared, and the wakeup is turned off (if it had been set).
- The check out softkey is a simple method of checking out a guest. It is only provided when the occupancy setting of the guest room is Occupied.

- The clearing of the individual call forwarding (programmed on the set) on check out does not affect the tenant-based call forwarding. On check out, the front desk clears all individual call forwarding information from the room phone. For example, the check out clears the forwarding to a cell phone, but not the tenant-based forwarding to voice mail that was assigned in CDE Form 19.
- The clearing of the individual call forwarding requires Feature Level 3 or greater.

Programming

None.

Operation

Front Desk Terminal

To Check Out (Single Key)

- On the Room Update screen, move the shaded bar to the Occupancy field. The Check Out softkey is displayed.
- Press the **Check Out** softkey. The room occupancy field is set to Vacant. The room condition is set to Dirty. The call restriction is set to the setting specified in System Option 57 (vacant/default call restriction). The Cancel softkey and the Enter softkey appear.

  A prompt appears: Press Enter to complete check out.
- To cancel the checkout, press the **Cancel** softkey. All original information is displayed.
- To complete the check out**,** press the **Enter** softkey. The guest name is blanked out (on the primary telephone only).
  Do Not Disturb is disabled (if set).
  Message waiting is cleared.
  Wakeup time is cleared.
  Message register is cleared.
  Individual Call Forwarding is cleared.

CLASS (station side) for Analog Telephones

CLASS (station side) for Analog Telephones allows analog telephones to show the calling party name and number on their display. For more information, refer to CLASS for Analog Telephones in Program Features.

DID Server

See Direct Inward Dialing For Guests

Direct Inward Dialing For Guests

Description

The Direct Inward Dialing (DID) application is a purchaseable option that allows outside callers to reach guests without going through the front desk. The DID Server application automatically assigns DID numbers to guests when they are checked in with the Property Management System (PMS). Upon check-out, DID numbers are un-assigned. Assignment is performed in a "First In-First Out" circular manner.

Assigned DID numbers can be viewed and printed by attendant consoles and sub-attendants.

Conditions

- The DID Server Application is mutually exclusive to the MEM DID Server. They can not both be active at the same time.
- DID numbers can not be deleted (except at check-out), modified, or manually assigned.

Programming

In CDE Form 2, assign a Feature Access Code to Feature 69 (DID Number Display).

In CDE Form 3, select COS option 623 (Automatic DID Number Assignment).

In CDE Form 4, enable system options 19 (DID Server Application) and 108 (Property Management System). **Note**: You need an AMC password to enable these options.

In CDE Form 15, program the X field with Feature Access Code 67 (Digit Translation Table Access).

In CDE Form 55, program digit translation and routing.

In CDE Form 34, add a Hotel/Motel DID printer type.

Operation

Automatic. For audits, refer to Direct Inward Dialing audit.

Note: CDE Form 55 displays the destinations that have been assigned a DID number.

Do Not Disturb

Description

The Do Not Disturb feature blocks calls from ringing at a guest's telephone. Outgoing calls are not affected. See the Do Not Disturb Table. Callers to a telephone with Do Not Disturb active hear reorder (error) tone. The message Do Not Disturb appears on display sets. The system can be programmed to reroute these calls to a predetermined answering point. The console, and SUPERSET 4150, SUPERSET 430, SUPERSET 4DN, and Mitel 5340 IP telephones can override Do Not Disturb.

Do Not Disturb can be applied to a single telephone or to all the telephones in a guest suite. The attendant console and the front desk terminal have softkeys DND All On and DND All Off that set all the room telephones (the primary and the secondary telephones in a suite).

The attendant console can set and clear Do Not Disturb for a guest room telephone whether or not the console is connected to the guest room phone. The front desk terminal can set Do Not Disturb in Room Update mode while the highlight bar is on the DND field.

A guest can set and clear Do Not Disturb from a guest room telephone.

Table: Do Not Disturb

Terminal	Description	Display/Confirmation
Attendant Console		

Setting and clearing Do Not Disturb are controlled by the NO DISTB softkey. When DND is set, the NO DISTB softkey does not appear. When DND is not set, the softkey appears.

DND can be applied to a single telephone or to all the telephones in a guest suite. If a guest calls an attendant to set a DND, the attendant can use the THIS PHONE softkey to select the extension that the caller is actually using for the call.

DND On: DND status flag appears on console status line.
DND Off: There is no DND status flag on the status line of the attendant console.

In cases where the guest has multiple telephones (at least one secondary telephone associated with a primary telephone), the attendant console display identifies all of the telephones showing the primary DN with an "A>". Four secondary telephones can be listed. A (w) indicator beside the secondary telephone shows that there is at least one wakeup programmed for that telephone; a (d) indicator shows that the telephone is in Do Not Disturb; (wd) shows that both exist.

Softkeys DND All On and DND All Off can apply DND to all of the telephones in the room.

Front Desk Terminal	Setting and clearing Do Not Disturb are controlled by DND softkeys. DND can be applied to a single telephone or to all the telephones in a guest suite.

In cases where the guest has multiple telephones (at least one secondary telephone associated with a primary telephone), the Guest Room Update screen shows "Other Phones". Four secondary telephones can be listed. A (w) indicator beside the secondary telephone shows that there is at least one wakeup programmed for that telephone; a (d) indicator shows that the telephone is in Do Not Disturb; (wd) shows that both exist. The DND field appears on the Guest Room Update screen on the primary telephone screen and on the secondary telephone screen. Softkeys DND All On and DND All Off appear only when the DND field is selected on the primary telephone screen.

When DND is set, ON is displayed in the field. When DND is not set, the field is blank.

Industry-standard Telephones	Setting and clearing Do Not Disturb from a guest room telephone is done by dialing a feature access code plus 1 (to set) and 2 (to cancel) DND.	Dial tone indicates DND has been set or canceled.
Mitel 5201, SUPERSET 4001 / 4015 / 410 / 401+/ 3DN/ Telephones	Setting and clearing Do Not Disturb from a guest room telephone is done by pressing the Do Not Disturb feature key. The Mitel 5201, SUPERSET 4001, and SUPERSET 401+ do not have feature keys. They follow the same procedure as the industry standard telephones.	Dial tone indicates DND has been set or canceled.
Mitel 5010/5020/ 5215/5220/5330/ 5340/SUPERSET 4025/4125/4150/ 420 / 430/ 4DN/ Telephones	To set or clear Do Not Disturb, toggle the Do Not Disturb feature key or use the Superkey menu.	Dial tone indicates DND has been set or canceled.

Conditions

- For DND to be set on a guest room phone, whether it is done from the attendant console, the front desk terminal, or the guest room phone, COS option 220 must be enabled for that particular guest room phone.
- On the attendant console, if Do Not Disturb is set on an extension, DND appears on the top line of the display, for example, COS 12 COR 01 DND.
- In cases where the guest suite has primary and secondary telephones, the sub attendant telephone will only show the primary extension (room number). However, if the sub attendant user knows all of the extension numbers, any one of the extensions can be selected and a DND can be programmed.

Programming

In CDE Form 02, set Access Code for Do Not Disturb. Assign an access code for Do Not Disturb. The same access code sets and cancels Do Not Disturb at a guest room telephone

In CDE Form 03, enable COS option 220, Do Not Disturb, for each extension Do Not Disturb will apply to.

Operation

Attendant Console

Note: If you are connected with the extension, begin with the fourth step in these procedures.

To set up Do Not Disturb for a guest room with single phone

- Press **FUNCTION**.
- Press the **GUEST ROOM** softkey.
- Enter the extension (room) number. The NO DISTB softkey appears.
- Press the **NO DISTB** softkey. DND appears in the top line of the display.
- When you are finished, press the **EXIT** softkey.

To set up Do Not Disturb for a guest room with multiple associated phones

- Press **FUNCTION**.
- Press the **GUEST ROOM** softkey.
- Enter the extension (room) number. The NO DISTB softkey appears.
- Press the **NO DISTB** softkey. To program all the phones, press the **DND ALL ON** softkey. To program one of the phones, press the appropriate softkey that identifies the phone, and press the **NO DISTB** softkey.
- When you are finished, press the **EXIT** softkey.

To cancel Do Not Disturb for a guest room with single phone

- Press **FUNCTION**.
- Press the **GUEST ROOM** softkey.
- Enter the extension (room) number. The NO DISTB softkey appears.
- Press the **NO DISTB** softkey. DND disappears from the top line of the display.
- When you have finished, press the **EXIT** softkey.

To cancel Do Not Disturb for a guest room with multiple associated phones

- Press **FUNCTION**.
- Press the **GUEST ROOM** softkey.
- Enter the extension (room) number. The NO DISTB softkey appears.

- Press the **NO DISTB** softkey. To program all the phones, press the **DND ALL OFF** softkey. To program one of the phones, press the appropriate softkey that identifies the phone, and press the **NO DISTB** softkey.
- When you are finished, press the **EXIT** softkey.

Front Desk Terminal

To set Do Not Disturb for a guest room with single phone

- On the Room Update screen, move the shaded bar to the DND field and press the **DND** softkey.The Do Not Disturb field shows ON.

To set Do Not Disturb for a guest room with multiple associated phones

- On the Room Update screen, move the shaded bar to the DND field.
- To program all the phones, press the **DND ALL ON** softkey. To program the primary phone, press the **DND** softkey.The Do Not Disturb field shows ON. To program one of the other phones, move the shaded bar to the extension of the other phone, press the **Select** softkey, and press the **DND** softkey. The Do Not Disturb field shows ON.

To clear Do Not Disturb for a guest room with single phone

- On the Room Update screen, move the shaded bar to the DND field and press the **DND** softkey.The Do Not Disturb field goes blank.

To clear Do Not Disturb for a guest room with multiple associated phones

- On the Room Update screen, move the shaded bar to the DND field.
- To program all the phones, press the **DND ALL OFF** softkey. To program the primary phone, press the **DND** softkey.The Do Not Disturb field goes blank. To program one of the other phones, move the shaded bar to the extension of the other phone, press the **Select** softkey, and press the **DND** softkey. The Do Not Disturb field goes blank.

Guest Names

Description

Guest names can be entered and stored from the front desk terminal and from Mitel display telephones only.

The front desk terminal has two data fields: last name and first name. The last name field accepts up to 15 alphanumeric characters. The first name field accepts up to six characters. Names are displayed in uppercase, even if they are entered in lowercase. The order is LAST NAME, FIRST NAME. The Mitel display telephone can enter and display one 10-character name, or can display up to 10 characters of a name entered from a front desk terminal.

Guest names can be displayed on an attendant console if entered from the front desk terminal or a Mitel display telephone, but they cannot be entered or searched for. See the Guest Names Table below.

Table: Guest Names

Terminal		Description Display
Attendant Console	Guest names can be displayed on an attendant console with a front desk terminal configured on the system, but they cannot be entered or searched for.	First 10 characters of the last name field as entered from the front desk terminal. (If the name entered on the front desk terminal is longer than 10 characters, the name displayed on the console is truncated at 10.)

Table: Guest Names

Terminal		Description Display
		Ten-character name as programmed from a Mitel 5020 IP, 5220 IP, 5224 IP, SUPERSET 420, SUPERSET 430, SUPERSET 4125, SUPERSET 4025, SUPERSET 4150 telephone.
Front Desk Terminal	Guest names can be entered and stored, as well as displayed. There are two data fields: last name and first name.	The last name field accepts up to 15 alphanumeric characters. The first name field accepts up to six characters. They are displayed in uppercase, in the order LAST NAME, FIRST NAME.

Conditions

- When room status is set to vacant, the name fields are cleared. Both the Front Desk Terminal and the attendant console operate in this way.
- If the name field is not programmed, the data field is blank.
- If a copy database is in progress when a name update is requested from the Front Desk Terminal (the Enter key is pressed), the name is not saved. A warning message is displayed. Any other information entered on the form is saved.
- With the exception of name updates, a copy database is transparent to the user. All other front desk terminal functions can be carried out because the information is saved until it can be executed.
- A guest name cannot be stored if the Occupancy field is set to Vacant.
- The name fields are automatically blanked out when the Check Out softkey is used.

Programming

None.

Operation

Front Desk Terminal

To enter a guest name

- Display the Room Update screen. On the Room Update screen, the guest name field is highlighted if the occupancy status of the room is vacant. You are prompted to enter the last name.
- Enter the last name (up to 15 alpha/numeric characters) in the Guest Name field. While a name is being entered, no softkeys are provided. All names are stored and displayed as uppercase, even if they are entered in lowercase.

Note: If you use an arrow key to move to the next field, the data you entered is lost.

- Press the **Return** hardkey. The new data is displayed. The shade bar moves to the beginning of the next field. You are prompted to enter the guest's first name, and then press Return.
- Enter the first name (up to six characters) in the next field.
- Press the **Return** hardkey. The shade bar moves to the beginning of the next field.The new data is displayed.
- Press the **Enter** softkey. The information is entered in the database. The Enter softkey is displayed only if the Occupancy field is not set to Vacant.

To remove a guest name in either guest name field

- Go to the Room Update screen.
- Use the cursor keys to move the shaded bar to the name field to be blanked out.
- Press the **spacebar**.The name is blanked out.
- Press the **Return** hardkey.
- Press the **Enter** softkey.The name is removed from the database.

Guest Room Enhanced

See Subattendant - Guest Room Enhanced.

Guest Room Key Programming

Description

A block of guest room Mitel telephones can be programmed with Speed Dial and Feature Access keys only be done from CDE. Up to three separate blocks (templates) can be programmed at one time.

Conditions

- A block of guest room Mitel telephones can be programmed with speed dial and feature access keys through CDE. It is possible to program up to three separate blocks of telephones with unique speed dial and feature access keys.
- Keys programmed as Speed Dial keys or Feature Access keys in a Guest Room Key Template cannot be programmed as Line Appearances in CDE Form 09 (Desktop Device Assignments).
- Guest Room set users cannot program speed dial keys when a guest room template is enabled for the extension.
- Applying a Guest Room template to 53xx series phones will <u>not</u> override keys that have been programmed using the Superkey or Applications key.

Programming

In CDE Form 03, enable COS option 244 (Room Status Applies) in the set's Class of Service. Enter the COS Template Number as well.

In CDE Form 37, enter the desired speed dial numbers and feature access keys for the appropriate template.

Operation

None.

Internal Number Block

Internal Number Block provides set/station number confidentiality between calls. The hotel operator can choose whether or not the room number of a guest should appear on another guest's telephone. For more information, see Internal Number Block in Program Features.

Maid in Room

Description

This feature allows the maid to enter a code from the guest room telephone to indicate that the room is currently being cleaned. Upon leaving the room, the maid can enter a second code indicating that the room is ready for inspection or that it is clean. See the Maid in Room table below.

Table: Maid in Room

Code	Maid in Room Status	Meaning
1	Maid is in room	Maid has entered the dirty room and has dialed the room access code and the maid in room code. The Room Condition does not change.
2	No maid in room	Maid has dialed the room access code and the no maid in room code. Maid has decided to leave the dirty room after dialing in the Maid in Room code. Room has not been cleaned. The Room Condition does not change.
3	Room clean, maid not in room	The room is clean. If there is a supervisor, supervisor dials this code to indicate room has been inspected and is ready to be rented out. If there is no supervisor, maid dials this code to indicate that the room is clean and ready to be rented out. Maid or supervisor has dialed the room access code and the room clean code.
4	Room to be inspected, maid not in room Maid has finished cleaning the room.	The room is ready for inspection by a supervisor. Maid has dialed the room access code and the room to be inspected code.

The attendant console or the front desk terminal can display this information. A Mitel telephone with the proper COS option can also display this information. See Maid in Room Status Display. A Room Status single line report is generated each time that the Maid in Room feature is used. See Single Line Reports. If the maid in room status changes for a room currently in room update mode, the new status is not posted until the current display (room update screen on the front desk terminal, or current display on the attendant console) is exited and reentered. See the Maid in Room Table below.

Table: Maid in Room

Status	**Description of Display**		**Changing the Display from a(n)...**		
			Attendant Console	**Front Desk**	**Terminal Telephone**
	Attendant Console	**Front Desk Terminal**			
Maid in room	Maid in room status is displayed on the attendant console display	The status "**Maid in Room**" in bold letters appears in the Maid in	Maid in Room status cannot be changed from the console.	Maid in Room status cannot be changed from the front desk terminal.	Maid dials the room access code and 1.

Table: Maid in Room

Status	Description of Display		Changing the Display from a(n)...		
			Attendant Console	Front Desk	Terminal Telephone
	Attendant Console	Front Desk Terminal			
	room status line as "Maid".	Room field on the room updates form.			
No maid in room	If there is no maid in the room, "Maid" is not displayed.	If there is no maid in the room, the field is blank.	Maid in Room status cannot be changed from the console.	Maid in Room status cannot be changed from the front desk terminal.	Maid dials the room access code and 2.
Room to be inspected	To be inspected condition is displayed on the attendant console display room status line as "Insp".	The status "**Insp**" in bold letters appears in the Condition field on the room updates form.	Attendant presses the Insp softkey.	On the Room Update screen, Attendant presses the Show Choices softkey and then the Insp softkey.	Maid dials the room access code and 4.
Room clean	Clean condition is displayed on the attendant console display room status line as "Clean".	The status "**Clean**" in bold letters appears in the Condition field on the room updates form.	Attendant presses the Clean softkey.	On the Room Update screen, attendant presses the Show Choices softkey and then the Clean softkey.	Maid or supervisor dials the room access code and 3.

Conditions

- The Maid in Room status can only be changed from the guest room telephone. This feature allows the maid to enter a code from the guest room telephone to indicate that the room is currently being cleaned. Before leaving the room, the maid can enter a second code to indicate that the room is ready for inspection, or that it is clean. The attendant console, front desk terminal, or a Mitel telephone with the proper COS option can display this information.
- Only the Room Condition status can be modified by the maid. This feature cannot change the Room Occupancy status of the guest room.
- Feature Access Code 35, Maid in Room, affects all the rooms in the suite when the suite has secondary telephones associated with a primary telephone. This access code can be dialed from any telephone in the suite.

Programming

In CDE Form 02, assign an access code to Feature 35 (Maid In Room).

In CDE Form 03, enable COS option 244 (Room Status Applies) for the extension.

Operation

Attendant Consoles

To Check the Status of a Room (Call Restriction, Room Occupancy, Room Condition, Maid)

Note: The status of the room can be checked from: an idle postion beginning with the first step, or while in a converstation with the room to be checked by pressing the Status softkey.

- Press **FUNCTION**.
- Press the **GUEST ROOM** softkey.
- Enter the extension (room) number. The STATUS softkey is displayed. The second line of the display shows the room's status. For example, Status = Vac/Dirty/Int/Maid means the room is vacant, dirty, has internal telephone privileges, and has a maid in it.
- Press the **STATUS** softkey. The Room Status softkeys are displayed. These softkeys change according to the status of the room. For instance, if the room is dirty, the CLEAN softkey appears. If the room is clean, the DIRTY softkey appears.
  The second line of the display shows the room's status.

Front Desk Terminal

To check the Maid in Room status

- On the Room Update form, check the field below the Guest name fields (this field does not have a title). If this field has Maid in Room displayed, the maid is currently in that room. Nothing is displayed if the maid is not in the room.

Telephones

To change the Maid in Room Status from the telephone

- The maid enters the room, picks up the handset and dials the Maid In Room access code and the code for Maid in Room, then hangs up. For example, 111 could be the Maid in Room access code, 1 means maid is in room. The number to be dialed is 1111.
- When the maid finishes the room, the maid picks up the handset and dials the Maid in Room access code and the code for Room to be Inspected, Maid not in Room. For example, 4 means room to be inspected, maid is not in room.The number to be dialed is 1114.
- If the maid leaves the room without finishing the job, the maid dials the code for No Maid in Room. For example, 2 means no maid is in room. The number to be dialed is 1112. The maid hangs up.
- The supervisor can then enter the room and checks it. If satisfactory, enter the Maid in Room access code and the code for Room Clean, Maid not in Room, then hang up. For example, 3 means room clean, maid is not in room. The number to be dialed is 1113.

Maid in Room Status Display - Mitel Display Telephones

Description

This feature allows an authorized Mitel display telephone to determine which guest rooms have maids in them. See the Maid in Room Status Display table.

Table: Maid in Room Status Display

Status	Description of Display	Changing the Field from a(n)....
Attendant Console	**Front Desk terminal**	**Room Telephone**

Mitel 5020 IP, 5220 IP, 5224 IP, 5324 IP, 5330 IP, SUPERSET 4125, SUPERSET 4025 and SUPERSET 420 Telephone				
Maid in room	Rooms with a maid in them are displayed two at a time in numeric order. If there are more rooms with maids in them, a left arrow softkey appears. Pressing this softkey displays the other rooms, two at a time. See Maid in Room Search.	Maid in Room status cannot be changed from the console.	Maid in Room status cannot be changed from the front desk terminal.	Maid or supervisor dials an access code and a status code from a guest room phone.
SUPERSET 4150, SUPERSET 430, SUPERSET 3DN, and Mitel 5340 IP Telephone				
Maid in room	Rooms with a maid in them are displayed 10 at a time in numeric order. If there are more rooms with maids in them, a right arrow softkey appears. Pressing this key displays the other rooms, 10 at a time. See Maid in Room Search.	Maid in Room status cannot be changed from the console.	Maid in Room status cannot be changed from the front desk terminal.	Maid or supervisor dials an access code and a status code from a guest room phone.

Conditions

- This feature is available on Mitel 5020 IP, 5220 IP, 5224 IP, 5324 IP, 5330 IP, 5340 IP, SUPERSET 4025, SUPERSET 4125, SUPERSET 4150, SUPERSET 420, SUPERSET 430, and SUPERSET 4DN telephones only.

Programming

In CDE Form 02, assign an access code to Feature 40 (SUPERSET Telephone Maid In Room Status Display).

In CDE Form 03, enable COS option 608 (Telephone - Room Status Display) in the set's Class of Service.

Operation

Mitel 5020 IP, 5220 IP, 5224 IP, 5324 IP, 5330 IP, SUPERSET 4025, SUPERSET 4125, SUPERSET 420 telephones

Monitoring Maid in Room Status

- Dial the access code for Maid In Room Status Display. The LCD displays the extension numbers of the first two rooms that have maid in room codes entered. If there are more rooms, the arrow softkey appears.
- Press the **arrow** softkey to display these other guest room extension numbers.The rooms are displayed two at a time. When the last room number is displayed, the arrow softkey no longer appears.
- When you have finished, press the **Hangup** softkey.

SUPERSET 4150, SUPERSET 430, SUPERSET 4DN, and Mitel 5340 telephones

Monitoring Maid in Room Status

- Dial the access code for Maid In Room Status Display. The LCD displays the extension numbers of the first 10 rooms that have maid in room codes entered. If there are more rooms, the arrow softkey appears.
- Press the **arrow** softkey to display these other guest room extension numbers. The rooms are displayed 12 at a time. When the last room number is displayed, the arrow softkey no longer appears.
- When you have finished, press the **Hangup** softkey.

Message Lamp Test

Description

The message waiting lamp on a guest room telephone is automatically tested each time the room status changes. The test will run when a room status change is made from a front desk terminal, a PMS terminal, or an attendant console. There is no indication at the front desk terminal that the test has run. However, if there is a failure, notification is through the alarm icon at the console. The test verifies lamp operation and confirms that the telephone is still connected in the room. The test does not verify bell operation.

Conditions

- The message lamp test only applies to telephones connected to ONS cards in digital bays. The message lamp test does not work on CLASS sets that have COS Option 505, ONS Stations Support CLASS enabled.
- COS Option 232 (Message Waiting Lamp) and System Option 02 (Message Lamp Test Enable) must be enabled.

Programming

- In CDE Form 03, enable COS option 232 (Message Waiting Lamp) for the telephones to be tested. This COS option lights the message lamp on a room telephone when there is a message waiting. When this COS option is enabled, COS option 231 cannot be enabled. They are mutually exclusive.
- In CDE Form 04, enable System Option 02 (Message Lamp Test Enable). This option allows the message waiting lamps on room telephones to be tested automatically.

Operation

None.

Message Register

Description

The Message Register feature keeps track of the number of completed external calls or call units for each extension. This feature applies to consoles, industry standard telephones, and Mitel telephones. The extension or console which answers an incoming trunk call or places an outgoing trunk call is charged with the count on the trunk while the trunk is active in the system, even if the extension is no longer involved in the call. This operation can be changed so that the count is applied to the current extension or console talking to the trunk. If the attendant handles the trunk call first (incoming or outgoing) and transfers the trunk call to an extension (supervised or unsupervised), the console's message register is transferred to the extension. Supervision on an outgoing trunk is counted as one message register count.

There are two modes of operation. The first counts the number of external calls made by each room. The second keeps track of meter pulses being sent from the far end to the associated trunk circuit. These pulses can in turn be charged against the guestroom making the call. The attendant console or the front desk terminal displays the current value of the message register for a room each time a room number is entered. The message register can be cleared by the attendant from the console or terminal, byt the 5340 subattendant, or automatically upon requesting an audit. (See Attendant Message Register Audit.) Clearing the message register can be recorded as a single line report on a system printout - the system printer prints the message register count before it is cleared. This feature is only used in cases where COS Option 703, Message Register Applies, has been enabled for some extensions.

The maximum message register count is 50,000. The counter resets to zero when this count is reached (an alarm can also be generated). The alarm icon on the console can be set to flash when the message register overflows.

Meter pulses are recorded in SMDR. See the Station Message Detail Recording section.

Conditions

- Message registration allows the system to accumulate the number of completed call units (local and long distance) made from an extension. An alarm can be generated at the attendant console when a message register overflows. Message registration can apply to extensions and trunks.
- If an extension with Message Registration connects to an outgoing trunk which does not have Message Registration, Message Registration does not apply to the extension.
- When the trunk does not provide answer supervision (trunk circuit descriptor option “Far-end Gives Answer Supervision” is not enabled), a completed outgoing call is determined by the pseudo answer supervision timer. This timer is started when a trunk seizure is acknowledged and when the timer expires, a meter pulse is counted (a pseudo answer supervision) giving an indication of a completed external call.
- A printer must be assigned to the “Hotel/Motel Audit” printout type in CDE Form 34 (Directed IO) to define where the single-line reports are printed.
- Clearing the register may result in printing a single-line report (if programmed).
- The Message Register field on the Front Desk Terminal is only displayed if COS 703, Message Register Applies, is enabled for the guest room. The highlighted Message Register field on the Front Desk Terminal causes the Clear Msg Reg softkey to appear. This softkey resets the message register to zero.
- The Message Register field shows the total number of message registers for the primary and secondary phones when the guest room has multiple phones associated.

Programming

In CDE Form 03, select COS option 703 (Message Register Applies) for the extensions for which counts are to be maintained and for the trunks on which counts apply.

Optional.
In CDE Form 03, select COS option 230 (Message Register Overflow Alarm) for the extension(s) or trunk(s) for which there is to be a message register overflow alarm indication at the attendant console. The alarm icon on the console flashes and a maintenance log is generated.

Optional
To enable the "Read MWI" feature for Mitel 5340 IP phone subattendants, in CDE Form 03, select COS option 320 (Transparent Multi-Console Operation) in the COS of the 5340 IP phone.

In CDE Form 04, select a value for System Option 49 (Pseudo Answer Supervision Timer) to be applied to trunks in the system that do not provide answer supervision.

Optional.
In CDE Form 04, select System Option 04 (Message Waiting and Message Register Clear Print) to print a

single-line report to the default printer each time that the message register changes, including the case when a message register is cleared by the console.
Select System Option 23 (Message Reg. Count Additional Supervisions) to count meter pulses after supervision is received on an outgoing trunk.
Select System Option 40 (Message Register Follows Talker) to charge the extension or the console last talking to a trunk with the message register count.

Operation

Attendant Console

To display the Message Register for an Extension

- Press **FUNCTION**.
- Press the **GUEST ROOM** softkey.
- Enter the extension (room) number. The message register count is shown in the first position on line 2 of the display.

To clear the Message Register for an Extension

- Press **FUNCTION**.
- Press the **GUEST ROOM** softkey.
- Enter the extension (room) number. If there is a message register count, the CLR REG softkey is displayed.
- Press the **CLR REG** softkey.

Front Desk Terminal

To clear the Message Register

- On the Room Update screen, move the shade bar to the message register field. The shade bar only highlights this field if the count is greater than zero.
- Press the **Clr Msg Reg** softkey. Yes and Cancel softkeys appear.
- To clear the register, press **Yes**. The message register is reset to zero.
- If you decide that you don't want to clear the register, press **Cancel**. The message register is not cleared.
- Press **Enter**. The message register is cleared. The shade bar moves to the next field.

Message Waiting

Description

The Message Waiting feature allows the attendant to inform guest room telephones that there is a message waiting. The message waiting indication may take the form of

- A message on the display of an analog telephone (CLASS). See CLASS for Analog Sets.
- A message on the display of a Mitel display telephone.
- A continuously flashing lamp (if equipped) on the telephone (Mitel telephones have a message lamp).
- A distinctive ringing pattern (three short bursts every 20 minutes). The telephone rings with this distinctive ringing pattern if the extension has been busy, or has Do Not Disturb set, or until message waiting is canceled.

Conditions

The following conditions apply to this feature:

- The setting and clearing of a message waiting indication can be printed from the console on a system printer. A single-line report can be generated, giving the room number, date, time and status change, and who changed it. See Single-Line Reports.
- When the guest returns and calls the attendant, the “MSW” indicator appears on the attendant console display to indicate that there is a message waiting for that extension.
- The Cancel softkey on a display Mitel telephone can cancel message waiting. Telephones with no softkeys must call the attendant to retrieve the message. It is then canceled by the attendant. See also Message Lamp Test.
- For Front Desk Terminals, dial 0 calls are routed to a console (Form 19) or a console LDN or subattendant LDN (Form 08). This routing makes sure there is a single place the guest must call to receive messages. If the dial 0 point for the room is not a console, or a console or subattendant LDN, the message softkey is not provided on the front desk terminal.
- Message Waiting Setup must be enabled in the COS for the guest room phone that is an industry-standard telephone. This restriction does not apply to Mitel telephones.
- COS options 231 and 232 cannot be enabled at the same time
- The attendant can inform guests that there is a message waiting and cancel the message waiting indication after guests call for their message.
- The attendant can read messages for an extension sent by another guest room extension or from another console.
- On the Front Desk Terminal, setting and clearing messages are mutually exclusive. If there is no message waiting for a room, the Set Message softkey is displayed. If there is a message waiting for a room, the Clear MSW softkey is displayed. If there is already a message waiting for a room (the message set from another device such as a console or another terminal), and the user of the front desk terminal presses the Set Message softkey, then the user will be informed that the room has a message waiting and the present message cannot be sent.
- Set-to-set messages are transparent to the front desk terminal. If a Mitel guest room telephone has left a message for another Mitel guest room telephone, it is not shown as a message indication on the front desk terminal.
- On industry-standard telephones and Mitel non-display telephones, a message waiting indication is a continuous flashing lamp on the extension (if equipped), and/or a distinctive ringing pattern every 20 minutes, or a discriminating (stutter) dial tone when the station goes offhook. The set rings with the distinctive ringing pattern if the extension had been busy, had Do Not Disturb set, or until Message Waiting is canceled. The discriminating dial tone is only available to ONS and OPS stations with COS 219, Discriminating Dial Tone enabled.

Programming

In CDE Form 03, enable COS option 105 (Attendant Guest Room Key) for the console to access message waiting from the Guest Room softkey. Enable COS option 219, Discriminating Dial Tone for guest room stations (ONS and OPS), or enable COS option 231 (Message Waiting Bell) for the guest room extension, or enable COS option 232 (Message Waiting Lamp).

Notes:

1. COS option 219, the discriminating dial tone (stutter dial tone) will alert the guest that a message is waiting when they go offhook.
2. COS option 231, Message Waiting Bell rings the bell on a guest telephone when there is a message waiting. COS option 231 only rings message bells. COS option 232 lights message lamps.
3. COS option 232, Message Waiting Lamp, lights the message lamp on a guest telephone when there is a message waiting. See also System Option 02, Message Lamp Test Enable (CDE Form 04).
4. COS options 231 and 232 cannot be enabled at the same time.

In CDE Form 04, select System Option 04 (Message Waiting and Message Register Clear Print). This option also enables printing of single-line reports for message register changes.

In CDE Form 19, program, if dial 0 calls are routed for a guest room. Enter Day, N1 and N2. Day, N1 and N2 can be the same number. If Day, N1 and N2 numbers are not provided, it is impossible to set message waiting from the front desk terminal. However, it can be done from the attendant console.

Operation

Attendant Console

To Set Up Message Waiting

- Press the **FUNCTION** key.
- Press the **GUEST ROOM** softkey.
- Enter the extension/room number.
- Press the **SEND MSG** softkey.
- Press the **EXIT** softkey.

To Cancel Message Waiting

- Press the **FUNCTION** key.
- Press the **GUEST ROOM** softkey.
- Enter the extension/room number.
- Press the **CANCEL MSG** softkey.
- Press the **EXIT** softkey.

If the attendant console is connected to the guest's extension

- Press the **CANCEL MSG** softkey.
- Press the **EXIT** softkey.

Front Desk Terminal

To Set Up Message Waiting

- From the House Statistics screen or a search display screen, press the **Room Update** softkey.
- Enter the room number. The Room Update screen appears.
- On the Room Update screen, press the **Set Message** softkey. A Clear MSW softkey appears.

To Clear Message Waiting

- From the House Statistics screen or a search display screen, press the **Room Update** softkey.
- Enter the room number. The Room Update screen appears.
- On the Room Update screen, press the **Clear MSW** softkey. The Message Waiting field is blanked and the Set Message softkey appears.

Industry-standard Telephones, Mitel non-display Telephones

To Retrieve Message Waiting

- Call the attendant for the message. The attendant reads the message and cancels the message waiting indication.

Mitel 5020 IP, 5220 IP, 5224 IP, 5324 IP, 5330 IP, 5340 IP, SUPERSET 4025, SUPERSET 4125, SUPERSET 4150, SUPERSET 420, SUPERSET 430, and SUPERSET 4DN Telephones

To Retrieve Message Waiting

- Press the **Message** key or the **MSG** softkey.
- Follow the display instructions to retrieve your message.

Multi-User

Description

Four front desk terminals can run the hotel/motel application at the same time. However, two terminals cannot edit information for the same room at the same time.

The front desk terminal also checks that the room is not being accessed by an attendant console. The message “Room being accessed by another user. Try again later.” appears on the screen if a console is currently displaying information for the room. If more than one front desk terminal is logged in with the supervisor-level password, each terminal checks that the room is not being accessed by another terminal or by the console.

The front desk terminal and a Mitel display telephone, or several display telephones can enter, change or delete a guest name. The entry that is done last is the one that is stored.

Up to 11 consoles can be configured on one system. Two (or more) consoles can access a guest room at the same time. The user that leaves the guest room last updates the database last, and that information is the valid data for the room. If two users are changing different fields at the same time, all the information should be captured because only fields that have been modified will be updated when the EXIT softkey is pressed.

Conditions

- Entering or changing guest room information from the front desk terminal can be controlled by passwords. When first installed, all passwords are the same, and different levels must be enabled. A CDE attendant-level password allows no editing capability. A CDE supervisor-level password allows editing. The user of an attendant password can read information about rooms, request audits, and conduct searches. The user of a supervisor password can, in addition, enter and change information about a guest room, because this password presents the Room Update softkey.

Programming

See Progam the Hotel/Motel Application.

Operation

See Using the Front Desk Terminal.

PMS Interface

Description

The PBX can interface to IBM-compatible personal computers with the Lodgistix PMS software package (or a package that follows the same protocol). See Property Management System.

Conditions

See Feature Interactions.

Programming

See Program for PMS.

Operation

None.

Room Condition

Description

Room Condition indicates the current housekeeping condition of a guest room, and can be set from an attendant console or a front desk terminal (Room Condition Table). See Room Status Display. Some conditions can be set from a guest room telephone. See Maid in Room. Room condition can be displayed as part of the room status display on the Mitel display telephone. See Room Status Display.

Table: Room Condition

Condition	Meaning	Description of Display	Changing the Field from a(n)...		
			Attendant Console	**Front Desk Terminal**	**Room Telephone**
Attendant Console					
Clean	Room has been inspected and is ready for occupancy.	"**Clean**" is displayed on the attendant console display room status line.	Attendant presses the STATUS softkey, then the CLEAN softkey.	On the Room Update screen, Attendant presses the **Show Choices** softkey, then the **Clean** softkey.	Maid or supervisor dials **Clean** code from a guest room phone. (Maid in Room feature.)
Dirty	Room has not been cleaned by a maid.	"**Dirty**" is displayed on the attendant console display room status line.	Attendant presses the STATUS softkey, then the DIRTY softkey.	On the Room Update screen, Attendant presses the **Show Choices** softkey, then the **Dirty** softkey.	Cannot be changed.
Out of Service	Room is not in use.	"**Serv.**" is displayed on the attendant console display room status line.	Attendant presses the STATUS softkey, then the OUT OF SERV softkey.	On the Room Update screen, Attendant presses **Show Choices** and **Serv** softkeys. .	Cannot be changed
To be Inspected	Room has been cleaned by a maid but has not yet been inspected.	"**Insp.**" is displayed on the attendant console display room status line.	Attendant presses the STATUS softkey, then the TO INSPECT softkey.	On the Room Update screen, Attendant presses the **Show Choices** softkey, then the **Insp** softkey.	Maid or supervisor dials the **To Be Inspected** code from a guest room telephone. (Maid in Room feature.)
Front Desk Terminal					
Clean	Same as above.	"**Clean**" appears in the Condition field on the room updates form.	Attendant presses the **clean** softkey.	On the Room Update screen, Attendant presses the **Show Choices**	Maid or supervisor dials **Clean** code from a guest room phone. (Maid in Room feature.)

Table: Room Condition

Condition	Meaning	Description of Display	Changing the Field from a(n)...		
			Attendant Console	Front Desk Terminal	Room Telephone
Attendant Console					
				softkey, then the **Clean** softkey.	
Dirty	Same as above.	"**Dirty**" appears in the Condition field on the room updates form.	Attendant presses the **Dirty** softkey.	On the Room Update screen, Attendant presses the **Show Choices** softkey, then the **Dirty** softkey.	Cannot be changed.
Out of Service	Same as above.	**"Serv"** appears in the Condition field on the room updates form.	Attendant presses the **Serv** softkey.	On the Room Update screen, Attendant presses the **Show Choices** softkey, then the **Out of Serv** softkey.	Cannot be changed.
To be Inspected	Same as above.	"**Insp**" appears in the Condition field on the room updates form.	Attendant presses the **Insp** softkey.	On the Room Update screen, Attendant presses the **Show Choices** softkey, then the **To Inspect** softkey.	Maid or supervisor dials the **To Be Inspected** code from a guest room telephone. (Maid in Room feature.)

Conditions

- If System Option 02 (Message Lamp Test Enable) is enabled, when a room's status is changed from vacant to occupied or from occupied to vacant, the system performs a message lamp test on the telephone in that room. If the test finds the lamp defective (i.e., burnt-out or missing), it raises a minor alarm. The message lamp test does not work on CLASS sets that have COS Option 505, ONS Stations Support CLASS enabled.
- System Option 56 (Room Status Conversion/Auto Wakeup Print Timer) only applies if System Option 34 (Auto Room Status Conversion/Auto Wakeup Print) is enabled.

Programming

In CDE Form 03, select COS option 244 (Room Status Applies) for all extensions for which room status is to apply.

In CDE Form 04, enable System Option 02 (Message Lamp Test Enable) to allow message lamp testing. This global option applies to the attendant console only. Enable System Option 33 (Room Status).
Optional: Select System Option 34 (Auto Room Status Conversion/Auto Wakeup Print).

Operation

Attendant Console

To Check/Change the Status of a Room (Call Restriction, Room Occupancy, Room Condition, Maid)

Note: The status of the room can be checked from: an idle postion beginning with the first step, or while in a converstation with the room to be checked by pressing the Status softkey.

- Press **FUNCTION**.
- Press the **GUEST ROOM** softkey.
- Enter the extension (room) number. The STATUS softkey is displayed. The second line of the display shows the room's status. For example, Status = Vac/Dirty/Int/Maid means the room is vacant, dirty, has internal telephone privileges, and has a maid in it.
- Press the **STATUS** softkey. The Room Status softkeys are displayed. These softkeys change according to the status of the room. For instance, if the room is dirty, the CLEAN softkey appears. If the room is clean, the DIRTY softkey appears.
  The second line of the display shows the room's status.
- To change an element of the status, press the softkey for the desired status. The MORE... softkey returns you to the previous display.

Front Desk Terminal

To Change the Room Condition

- From the House Statistics screen or a search screen, press the **Room Update** softkey. The Room Update screen is displayed.
- On the Room Update screen, move the shaded bar to the Condition field.
- Press the **Show Choices** softkey. The screen changes. The four states are presented as softkeys. They are:
  Clean Room has been cleaned by a maid.
  Dirty Room has not been cleaned.
  Out of Serv Room is not in service.
  To Inspect Room has been cleaned by a maid but must be inspected by a supervisor.
- Select one of the choices by pressing the key associated with it. The field is updated and the shaded bar passes to the next field.
  An Enter softkey appears.
  Or, if you decide that you don't want to make a change, press the **Cancel** softkey.
- If you have other changes to make, continue to make changes in the data fields.
- If you have no other changes to make, press the **Enter** softkey. The changes are saved in the database.

Subattendant

Refer to the Subattendant User Guide for the Mitel 5340 IP Phone available at Mitel OnLine.

Room Occupancy

Description

This feature indicates the current occupancy of a guest room. The room occupancy can be set from a console or a front desk terminal (Room Occupancy Table). See Room Status Display.

Table: Room Occupancy

Occupancy	Meaning	Description of Display	Changing the Field from a(n)...		
			Attendant Console	**Front Desk Terminal**	**Room Phone**
Attendant Console					
Vacant	Room is not occupied.	"**Vac**" is displayed on the attendant console display room status line.	Attendant presses the STATUS softkey, then the VACANT softkey.	On the Room Update screen, Attendant presses the **Show Choices** softkey, then the **Vacant** softkey.	Cannot be changed.
Occupied	Room is occupied by a guest.	"**Occ**" is displayed on the attendant console display room status line.	Attendant presses the STATUS softkey, then the OCCUPIED softkey.	On the Room Update screen, Attendant presses the **Show Choices** softkey, then the **Occupied** softkey.	Cannot be changed.
Reserved	The guest room is reserved for a guest, but not paid for. If the guest does not pay for the room by a particular time, it can be rented out to another guest.	"**Res**" is displayed on the attendant console display room status line.	Attendant presses the STATUS softkey, then the RESERVED softkey.	On the Room Update screen, Attendant presses the **Show Choices** softkey, then the **Reserved** softkey.	Cannot be changed.
Guaranteed	The guest room is reserved for a guest and paid for, although the guest has not checked in yet.	"**Gua**" is displayed on the attendant console display room status line.	Attendant presses the STATUS softkey, then the GUARANTEED softkey.	On the Room Update screen, Attendant presses the **Show Choices** softkey, then the **Guaranteed** softkey.	Cannot be changed.
Front Desk Terminal					
Vacant	Same as above.	"**Vacant**" appears in the Occupancy field on the room updates form.	Attendant presses the VACANT softkey.	On the Room Update screen, Attendant presses the **Show Choices** softkey, then the **Vacant** softkey.	Cannot be changed.
Occupied	Same as above.	"**Occupied**" appears in the Occupancy field on the room updates form.	Attendant presses the OCCUPIED softkey.	On the Room Update screen, Attendant presses the **Show Choices** softkey, then the **Occupied** softkey.	Cannot be changed.

Table: Room Occupancy

Occupancy	Meaning	Description of Display	Changing the Field from a(n)...		
			Attendant Console	**Front Desk Terminal**	**Room Phone**
Reserved	Same as above.	"**Reserved**" appears in the Occupancy field on the room updates form.	Attendant presses the RESERVED softkey.	On the Room Update screen, Attendant presses the **Show Choices** softkey, then the **Reserved** softkey.	Cannot be changed.
Guaranteed	Same as above.	"**Guaranteed**" appears in the Occupancy field on the room updates form.	Attendant presses the GUARANTEED softkey.	On the Room Update screen, Attendant presses the **Show Choices** softkey, then the **Guaranteed** softkey.	Cannot be changed.

Conditions

- If System Option 02 (Message Lamp Test Enable) is enabled, when a room's status is changed from vacant to occupied or from occupied to vacant, the system performs a message lamp test on the telephone in that room. If the test finds the lamp defective (i.e., burnt-out or missing), it raises a minor alarm. The message lamp test does not work on CLASS sets that have COS Option 505, ONS Stations Support CLASS enabled.
- System Option 56 (Room Status Conversion/Auto Wakeup Print Timer) only applies if System Option 34 (Auto Room Status Conversion/Auto Wakeup Print) is enabled.

Programming

In CDE Form 03, select COS option 244 (Room Status Applies) for all extensions for which room status is to apply.

In CDE Form 04, enable System Option 02 (Message Lamp Test Enable) to allow message lamp testing. This global option applies to the attendant console only. Enable System Option 33 (Room Status).
Optional: Select System Option 34 (Auto Room Status Conversion/Auto Wakeup Print).

Operation

Attendant Console

To Check/Change the Status of a Room (Call Restriction, Room Occupancy, Room Condition, Maid)

Note: The status of the room can be checked from: an idle postion beginning with the first step, or while in a converstation with the room to be checked by pressing the Status softkey.

- Press **FUNCTION**.
- Press the **GUEST ROOM** softkey.

- Enter the extension (room) number. The STATUS softkey is displayed. The second line of the display shows the room's status. For example, Status = Vac/Dirty/Int/Maid means the room is vacant, dirty, has internal telephone privileges, and has a maid in it.
- Press the **STATUS** softkey. The Room Status softkeys are displayed. These softkeys change according to the status of the room. For instance, if the room is dirty, the DIRTY softkey appears. The second line of the display shows the room's status.
- To change an element of the status, press the softkey for the desired status. The MORE... softkey returns you to the previous display.

Front Desk Terminal

To Change the Occupancy Field

- From the House Statistics screen or a search screen, press the **Room Update** softkey. The Room Update screen is displayed.
- On the Room Update screen, move the shaded bar to the Occupancy field.
- Press the **Show Choices** softkey. The screen changes. The four states are presented as softkeys. They are:
  Vacant Room is not occupied.
  Occupied Room is occupied.
  Reserved Room is being held until a certain time. After that, the room can be rented out to someone other than the person who had reserved it.
  Guaranteed Room has a guaranteed reservation.
- Select one of the choices by pressing the key associated with it. The field is updated and the shaded bar passes to the beginning of the next field. The Enter softkey appears.
  Or
  if you decide that you don't want to make a change, press the Cancel softkey.
- If you have other changes to make, continue to make changes in the data fields.
- If you have no other changes to make, press the **Enter** softkey. The changes are saved in the database.

Subattendant

Refer to the Subattendant User Guide for the Mitel 5340 IP Phone available at Mitel OnLine.

Room Status Display

Description

This feature allows the attendant to display and change the status of a room(s). Room status consists of: room occupancy, room condition, telephone privileges (call restriction) and maid in room. Each (except maid in room) can be set independently by the attendant. See the Room Status Display table.

The system can be programmed to change the status of all "occupied/clean" rooms to "occupied/dirty" at a predetermined time. At the same time, an audit of all auto wakeups is generated. See Wakeup/Room Condition Audit.

Table: Room Status Display

Terminal Type	Description of Display
Attendant Console	When a room number is selected, the status from the above list is displayed on the second line of the attendant console LCD. For example: Status= Vac/Dirty/Int/Maid

Table: Room Status Display

Terminal Type	Description of Display
	means the room is vacant, dirty, has only internal telephone privileges and has a maid in it. The last field is blank if the maid is not present.
Room status for a number of rooms can be displayed by doing a room search on the basis of: - Maid in room - Vacant/clean - Reserved and clean - Out of service See Room Status Search.	
Front Desk Terminal	A. Room status for a particular room is displayed on the room update screen. occupancy, condition, call privilege, and maid are displayed as separate fields.
B. Room status for a number of rooms can be displayed by doing a room search on the basis of: - Vacant/clean - Vacant - Maid in room - Out of service - Dirty - Reserved - Guaranteed See Room Status Search.	

The abbreviations and softkeys used in the attendant console display are shown in the Room Status table below.

Chart: Room Status

Status	Abbreviation shown on display line	Softkey Name	Meaning
Vacant	Vac	VACANT	Room is not rented to a guest.
Occupied	Occ	OCCUPIED	Room is rented to a guest.
Clean	Clean	CLEAN	Room has been cleaned by maid.
Dirty	Dirty	DIRTY	Room has not been cleaned by maid.

Chart: Room Status

Status	Abbreviation shown on display line	Softkey Name	Meaning
Internal	Int	INTERNAL	Room can only make internal telephone calls.
Local	Loc	LOCAL	Room can make internal and local calls.
Long Distance	LD	LONG DIST	Room can make internal, local, and long distance calls.
Maid in Room	Maid	No Softkey	At present a maid is in the room.
Reserved	Res	RESERVED	Room is being held for a certain time. After that time, it can be rented out to someone other than the person who reserved it.
Guaranteed	Gua	GUARANTEED	Room is guaranteed to be paid for but not yet occupied.
Out of Service	Serv.	OUT OF SERVICE	Room is not in service.
To be Inspected	Insp.	TO INSPECT	Room has been cleaned by a maid, but is to be inspected by the supervisor.

Conditions

- If System Option 2 (Message Lamp Test Enable) is enabled, when a room's status is changed from vacant to occupied or from occupied to vacant, the system performs a message lamp test on the telephone in that room. If the test finds the lamp defective (i.e. burnt-out or missing), it raises a minor alarm.
  The message lamp test does not work on CLASS sets that have COS Option 505, ONS Stations Support CLASS enabled.
- System Option 56 (Room Status Conversion/Auto Wakeup Print Timer) only applies if System Option 34 (Auto Room Status Conversion/Auto Wakeup Print) is enabled.

Programming

- In CDE Form 03, select COS option 244 (Room Status Applies) for all extensions for which room status is to apply.
- In CDE Form 04, enable System Option 02 (Message Lamp Test Enable) to allow message lamp testing. This global option applies to the attendant console only. Enable System Option 33 (Room Status).
  Optional: Select System Option 34 (Auto Room Status Conversion/Auto Wakeup Print).

Operation

Attendant Console

See the procedure for checking and changing the status in the operation section of Room Occupancy.

See the procedure for searching for rooms by status in the operation section of Room Status Search.

Front Desk Terminal

See the procedure for changing the occupancy field in the operation section of Room Occupancy.

See the procedure for checking the Maid in Room status in the operation section of Maid in Room.

See the procedure for searching for particular rooms in the operation section of Maid in Room Search.

See the procedure for changing call privilege in the operation section of Call Restriction.

Subattendant

Refer to the Subattendant User Guide for the Mitel 5340 IP Phone available at Mitel OnLine.

Room Types and Room Codes

Description

Hotel guest rooms can be divided into 50 different types, such as single, double, queen, smoking, nonsmoking. This division is done through customer data entry programming from the CDE terminal by putting each room type in a separate class of service. Since each class of service can have a different name associated with it, the type of room is identified by the class of service name. The name can be eight characters long. By default, it can be identified by the code (COS number) associated with the class of service name. Searches and audits can be requested by room type or code.

Table: Room Types (Examples)

Room Type	COS Name	Room Code (COS Number)
All single rooms	Single	1
All regular doubles	Double	2
All doubles facing pool	Doub_VIP	3
All suites no smoking	NoSmk_Su	4

Conditions

- The COS name must be unique and have less than nine characters in length.

Programming

See Programming the Hotel/Motel Application, Step 2.

Operation

Attendant Console

None.

Front Desk Terminal

See the procedure for searching for a particular room type in the operation section of Maid in Room Search.

Searches

Several different types of searches can be conducted:

- Guest search (see note)
- Room number search
- Room type search (see note)
- Maid in room search
- Room status search

Note: Front Desk Terminal only.

Guest Search

Description

Guest Search searches all guest rooms for a text string that matches a guest's name.

Conditions

- The string entered must match with the first letters of the last name field. For instance, if BLACK were entered, the name would match with BLACK or other names containing these letters such as BLACKBURN.
- All matches are displayed in alphabetical order using the standard front desk search display format.
- If last names are identical, first names are used to determine the order.
- If both names are identical, the guest who checked in last is displayed first.
- This feature applies to the Front Desk Terminal only.

Programming

None.

Operation

Front Desk Terminal

- From the House Statistics screen, press the **Guest Search** softkey. The display does not change. The softkeys are blanked. You are prompted to enter the last name being searched for.
- Enter the name. You can enter part of a name, for instance, BLA. All the names with BLA as the first letters would be displayed.
- Press the **Return** hardkey. If a match is found, the room that is matched, or the room that is the closest matched, is displayed first, followed by the next eight rooms in alphabetical order.
- To display the next nine names meeting the search criteria, press the **Page Down** softkey. This key appears if there are any more names in the system. They are displayed in alphabetical order.
- To display the previous nine names meeting the search criteria, press the **Page Up** softkey. This key only appears if there is at least one name that has scrolled off the top of the display
- To search for other names, use the **Guest Search** softkey. This softkey can be used as a "Guest Phone Book" service.
- To return to the House Statistics screen**,** press the **Quit** softkey.
- To go to the room update screen that displays information about the indicated room, press the **Room Update** softkey.

Room Number Search

Description

A search can be conducted for information about a particular room number. For suites with primary and secondary telephones, room searches exclude secondary phones and step only through Primary Directory Numbers. See the Room Number Search table below.

Table: Room Number Search

Terminal Type	Operation	Display
Attendant Console	Press the Guest Room softkey and enter a room number.	The room occupancy, condition, call restriction, whether a maid is in the room , and message register count is displayed. Attendant consoles associated with a front desk terminal application display the guest name and room type.
Front Desk Terminal	**A.** Beginning from the House Statistics screen, press the Room Update softkey and enter the room number.	**Room Update Screen.** The name of the guest, room occupancy, condition, call restriction, whether a maid is in the room, wakeup time, room type, whether there is a message waiting, whether Do Not Disturb is set, and the message register count is displayed on the Room Update screen.
B. From the Room Search menu, press the Room Number softkey and enter the room number.	**Search Display Screen.** The name of the guest, room type, room occupancy, condition, call restriction, whether a maid is in the room, message waiting indication, and wakeup time are displayed for that particular room number, and the next eight rooms. A plus (+) sign after the wakeup time indicates that more than one wakeup time has been set. When a guest has more than one wakeup time set, the time shown is the one that is due next.	

Programming

None.

Operation

Attendant Console

To search for a particular room by number

- Press the **GUEST ROOM** softkey.
- Enter the room number you want to check. The status information is provided on the second line of the LCD display in the status section.

Front Desk Terminal

To search for a guest room by number

- On the House Statistics screen, press the **Room Search** softkey. The softkeys show the various kinds of searches that can be selected.
- Press the **Room Number** softkey. You are prompted to Enter the Room Number.
- Type the room number.
- Press the **Return** hardkey. The specified room is listed first, followed by up to eight following room numbers. If the room number entered is not valid, the following prompt appears: Room not found. Press Enter to search for another.
- To search for another room, or to do a different type of search, press the **Quit** softkey.

Room Type Search

Description

A search can be conducted on the basis of room type. The Room Type softkey is used to search for a particular room type, such as single rooms or double rooms. The class of service number for the room type is used as the room code for search purposes. There is a wild card entry for searching all room types.

Searches can also be conducted on the basis of room status and room type. See Room Type.

Conditions

- Room Type Search is available on the front desk terminal only.
- Each type of room has a distinct COS programmed in CDE. This COS number is the code for that type of room. There can be up to 50 COS.

Programming

None.

Operation

Front Desk Terminal

To search for a particular room type

- From the House Statistics screen, press the **Room Search** softkey. The softkeys show the various kinds of searches that can be selected.
- Press the **Room Type** softkey. You are prompted to enter the code for the kind of rooms you want to search: for instance, single rooms.
- Enter the code for the room type, or enter 0 for all room types. If 0 is entered, all guest rooms are displayed in order of room number.
- Press the **Return** hardkey. All rooms of the required type are displayed in numerical order.
- To display the next nine rooms, press the **Page Down** softkey. This key only appears if at least one more room matching the search criteria exists.
- To display the previous nine rooms meeting the search criteria, press the **Page Up** softkey. This key appears only if at least one room has scrolled off the top of the display.
- To return to the House Statistics screen, press the **Quit** softkey twice.
- To go to the room update screen displaying information about the indicated room, press the **Room Update** softkey. If the room code entered is not valid, the following prompt appears: No Rooms Found. Enter Choice.If the room code entered is over 50, the terminal beeps and the code is not accepted.

- To search for another room type, press the Enter softkey.

Maid in Room Search

Description

A search can be conducted from Mitel display telephones equipped with softkeys, from the attendant console, or from a front desk terminal for rooms with a status of Maid in Room (Maid in Room Search table below). Rooms with a maid in them are displayed in sequence from lowest to highest room number. See Room Status Search.

Table: Maid in Room Search

Terminal Type	Operation	Display
Attendant Console	Search is initiated by pressing the MAID softkey.	The first 10 room numbers of rooms with a maid in them are displayed. If there are more rooms being cleaned, the MORE... softkey appears. This softkey displays the other guest room extension numbers, 10 at a time. When the last room number is displayed, the MORE... softkey disappears. Only room numbers are shown.
Front Desk Terminal	Search is initiated by pressing the Maid in Room softkey.	The first nine room numbers of rooms with a maid in them are displayed. Guest name, type of room, occupancy, condition, call restriction, message waiting indication, and wakeup time are also displayed. If there are more rooms being cleaned, the PAGE DOWN softkey appears. This softkey displays the other guest room extension numbers, nine at a time. When the last room number is displayed, the PAGE DOWN softkey disappears, and the PAGE UP softkey appears. The PAGE UP softkey allows paging backwards, nine rooms at a time. This softkey is not available on the first screen of nine rooms. If no rooms are found that meet the requirements, the user is informed and the main search menu softkeys are presented.
Mitel 5020 IP Mitel 5220 IP Mitel 5224 IP Mitel 5324 IP Mitel 5330 IP SUPERSET 4125 SUPERSET 4025 SUPERSET 420 Telephones	Search is initiated by dialing the Maid in Room status display access code.	The first two room numbers of rooms with a maid in them are displayed. If there are more rooms being cleaned, the arrow softkey appears. This softkey displays the other room numbers, two at a time. When the last room number is displayed, the arrow softkey disappears.
SUPERSET 4150 SUPERSET 430 SUPERSET 4DN Mitel 5340 IP Telephones	Search is initiated by dialing the Maid in Room status display access code.	Search is initiated by dialing the Maid in Room access code. The first 10 room numbers of rooms with a maid in them are displayed. If there are more rooms being cleaned, the arrow softkey appears. This softkey displays the other room numbers, 10 at a time. When the last room number is displayed, the arrow softkey disappears.

Conditions

With the Front Desk Terminal, rooms can be searched on the basis of: vacant/clean, vacant, maid in room, out of service, dirty, reserved, or guaranteed.

See the condition section in Maid in Room and Maid in Room Status Display - Mitel Display Telephones.

Programming

See the programming section in Maid in Room and Maid in Room Status Display - Mitel Display Telephones.

Operation

Attendant Console

See the operation section in Maid in Room.

Front Desk Terminal

To search for a particular room

- From the House Statistics screen, press the **Room Search** softkey. The softkeys show the various kinds of searches that can be selected.
- Press the appropriate softkey. You are prompted to enter the code for the kind of room that you want to search: for instance, all single rooms.The wild card (0) is used to search all room types.
- Enter the room code, or 0. All rooms meeting the search requirements are displayed in a lowest to highest sequence by room number.
- To display the next nine rooms meeting the search criteria, press the **Page Down** softkey. This key appears only if at least one more room matches the search criteria.
- To display the previous nine rooms meeting the search criteria, press the **Page Up** softkey. This key appears only if at least one room has scrolled off the top of the display
- To return to the main room search menu, press the **Quit** softkey.
- To go to the room update screen that displays information about the indicated room, press the **Room Update** softkey.

Mitel Telephones

See the operation section of Maid in Room Status Display - Mitel Display Telephones.

Room Status Search

Description

The attendant can search for guest rooms that have certain status combinations. Different types of searches can be conducted from the attendant console or from the front desk terminal. All room searches can be divided into two phases: searching and displaying. The search mode provides softkeys to select the type of search to be done. See the Room Status Search table below.

Table: Room Status Search

Terminal Type	Types of Room Status Searches	Operation	Display
Attendant Console	- Maid in room - Vacant/clean - Reserved	Search is initiated by pressing the	When displaying all rooms with a particular status, as selected by softkeys, the LCD displays the numbers of up to 10 rooms that have the selected

Table: Room Status Search

Terminal Type	Types of Room Status Searches	Operation	Display
	and clean - Out of service	appropriate softkey.	status. If there are more rooms, the MORE... softkey appears. It displays the next group of up to 10 room numbers. Rooms are displayed in order from highest to lowest room number. Only room numbers are shown.
Front Desk Terminal	- Vacant/clean - Vacant - Maid in room - Out of service - Dirty - Reserved - Guaranteed	Search is initiated by pressing the appropriate softkey. For each, room type (or the wild card entry 0 for all rooms) must be specified.	If the search is successful, (i.e., a room meeting the search criteria exists) the first nine room numbers of rooms with that particular status are displayed. Room number, guest name, room type, occupancy, condition, whether there is a maid in the room, call restriction, message waiting indication, and wakeup time are displayed. If there are more rooms, the PAGE DOWN softkey appears. This softkey displays the other room numbers, nine at a time. When the last room number is displayed, the PAGE DOWN softkey disappears. The PAGE UP softkey allows paging backwards, nine rooms at a time. This softkey is not available on the first screen of nine rooms. If no rooms are found which meet the requirement, the user is informed and the main search menu softkeys are presented.
Mitel 5020 IP **Mitel 5220 IP** **Mitel 5224 IP** **Mitel 5324 IP** **Mitel 5330 IP** **SUPERSET 4125** **SUPERSET 4025** **SUPERSET 420**	Search is on a per room basis. See Room Status Display.	Search is initiated by dialing the room status display access code followed by a room number.	The LCD displays the status (occupancy and condition only) of the room: 1101 VAC/CLEAN The arrow softkey displays the status of the room with the next higher room number: 1102 RES/CLEAN When the last room number is displayed, the arrow softkey no longer appears.
SUPERSET 4150 **SUPERSET 430** **SUPERSET 4DN** **Mitel 5340 IP**	Search is on a per room basis. See Room Status Display.	Search is initiated by dialing the room status display access code followed by a room number.	The LCD displays the status (occupancy and condition only) of the room: 1101 VAC/CLEAN The arrow softkey displays the status of the room with the next higher room number: 1102 RES/CLEAN When the last room number is displayed, the arrow softkey no longer appears.

Conditions

See the conditions section in Room Status Display.

Programming

See the programming section in Room Status Display.

Operation

Attendant Console

See the procedure for checking and changing the status of a room in the operation section of Call Restriction.

To search for rooms by status (Room status, Room Condition Maid)

- Press **FUNCTION**.
- Press the **GUEST ROOM** softkey. Search selections are displayed.
- To list rooms with Maid in Room status, press the **MAID** softkey. The LCD displays the numbers of up to 10 rooms that have this status. If there are more rooms, the MORE softkey appears.
- To list rooms by status, press the **VAC/CLEAN** softkey, the **RES/CLEAN** softkey, or the **OUT OF SERV.** softkey. You are prompted to enter the room number. A valid room number results in the 10 rooms following the entered number being displayed.
- To list rooms by status beginning at a particular room number, press the **ROOM NUMBER** softkey. You are prompted to enter the room number. A valid room number results in the 10 rooms following the entered number being displayed.
- Press the **MORE...** softkey to see the next group of room numbers.

Front Desk Terminal

See the procedure to search for a particular room in the operation section of Maid in Room Search.

Mitel Display Telephones

To monitor Room Status from Mitel Display Telephones

- Dial the access code for Mitel Room Status Display. This feature applies to Mitel display telephones.
- Dial the extension number of the required guest room's telephone. The LCD displays the room's status (e.g., VAC/CLEAN).
- Press the arrow softkey for the status of the next room (in order of extension number). When the last room is displayed, the arrow softkey no longer appears.
- When you have finished, press the **Hangup** softkey.

Single Line Reports

Description

Single line reports are types of audits used to record changes in status for an individual room. See the Single Line Reports table. These reports are generated automatically (under control of System Options 04, 13, and 24) and provide hard-copy evidence that a change occurred. The printouts produced by single line reports are limited to 40 characters in length. The report printouts begin with a standard prefix, containing extension number, date and time. This prefix occupies the first 19 characters of the print line, as follows:

xxxxx mm/dd hh:mmP

The xxxxx field refers to an extension number (five-digit maximum). The mm/dd field refers to a two-digit month and a two-digit day. The hh:mmP field refers to a two-digit hour, and two-digit minute, where P indicates PM when a 12-hour clock is used.

There are three categories of single line reports:

- Automatic Wakeup. Sixteen suffixes are available for automatic wakeup single line reports. Three apply to attendant console operations, three to front desk terminal operations, three to station operations (guest room telephones), and seven to wakeup attempts. Each contains the programmed wakeup time.

If the FOR Option 102, Feature Level is set to level 1 or greater, automatic wakeup calls may be programmed as single daily (one timer repeated at the same time every day),or multiple wakeups (up to three timers that may also be repeated at the same time every day). The asterisk (*) following the time indicates a daily wakeup time. The WU-1, WU-2, or WU-3 indicates the wakeup number of the timer. The suffixes are summarized in Single Line Reports table.
A sample automatic wakeup report is given below:

12345 01/23 11:20P WU-1 7:00 * SET BY ROOM

- Message Registration. Message registration single line reports have only one printout suffix; it is generated whenever a room's message register is cleared from the attendant console. The register count field is five digits in length, and all leading zeros are displayed. The format of the suffix is:

  REG. CLEARED AT nnnnn

nnnnn represents the register contents prior to being cleared. A sample message registration report is given below:

12345 01/23 11:40P REG. CLEARED AT 00012

- Message Waiting. Two suffixes are available for Message Waiting single line report printouts. One is generated when Message Waiting is turned on for an extension; the other is generated when it is turned off. The format of the suffixes is:

  MESSAGE WAITING ON or MESSAGE WAITING OFF

A sample message waiting report is given below:

12345 01/23 11:45P MESSAGE WAITING ON

Note: When an attendant console sets 'message waiting' on or off, a Message Waiting report prints out. Printouts of Message Waiting reports occur for the attendant consoles only.

- Room Status. Room Status single line reports are generated each time the Maid in Room feature is accessed from a room phone. Room Status reports are generated whether the Property Management System is enabled or not.

A sample room status report is given below:

12345 01/23 11:45P RS INSP MAID

Table: Single Line Reports

Operation Type	Suffix	Definitions & Notes
Console Operations	WU-1 hh:mmP SET BY CONS	WU-1 = Wakeup Timer 1 hh:mmP = time and PM indicator SET BY CONS = Set by Attendant Console
	WU-1hh:mmP CHG BY CONS	CHG BY CONS = Changed by Attendant Console
	WU-1hh:mmP CAN BY CONS	CAN BY CONS = Canceled by Attendant Console
Front Desk Terminal Operations	WU-1 hh:mmP SET BY DESK	WU = Wakeup Timer 1 hh:mmP = time and PM indicator SET BY DESK = Set by Front Desk Terminal
	WU-1 hh:mmP CHG BY DESK	CHG BY DESK = Changed by Front Desk Terminal
	WU-1 hh:mmP CAN BY DESK	CAN BY DESK = Canceled by Front Desk Terminal
Room Operations	WU-1 hh:mmP SET BY ROOM WU-2 hh:mmP * SET BY ROOM	WU-1 = Wakeup Timer 1 hh:mmP = time and PM indicator SET BY ROOM = Set by guest room telephone WU-2 hh:mmP * = Wakeup Time 2 that repeats daily

Table: Single Line Reports

Operation Type	Suffix	Definitions & Notes
	WU-1 hh:mmP CHG BY ROOM	CHG BY ROOM = Changed by Guest Room Telephone
	WU-1 hh:mmP CAN BY ROOM	CAN BY ROOM = Canceled by Guest Room Telephone
Wakeup Attempts	WU -1hh:mmP aaaaaaa	aaaaaaa = either: ANSWERED NO ANS 1 NO ANS 2 NO ANS 3 * * BUSY 1, BUSY 2, BUSY 3 * * *** *** Third-attempt failures for NO ANS and BUSY generate three CTRL G characters separated by seven nulls, along with the asterisks. A CTRL G character rings bell.
Room Status	RS yyyyyzzzz	RS = Room Status yyyyy = Room Condition zzzz = Maid in Room - this field is blank when the maid is not in the room.

Conditions

- These reports are generated automatically under control of System Options 04, 13, and 24.

Programming

In CDE Form 04, enable the system option 04, Message Waiting and Message Register Clear Print . This option prints a single-line report to the default printer each time the message register changes, including the case where the message register is cleared by the console. See also: COS option 230, COS option 703, System Option 49, System Option 23 and System Option 40.

In CDE Form 04, enable the system option 13, Auto Wakeup Print. This option allows wakeup audit printouts and single-line reports to be printed on the system printer whenever a wakeup call is set up, canceled, answered, or an attempt is made to honor the wakeup. This option has no effect on Mitel 5020 IP, 5220 IP, 5224 IP, SUPERSET 4150, SUPERSET 420, SUPERSET 430, and SUPERSET 4DN telephones.

Chart: Enabling Single-Line Reports

Step	Action	Comments
	Form 04	
1.	Enable the following system options:	
	04 Message Waiting and Message Register Clear Print	Prints a single-line report to the default printer each time the message register changes, including the case where the

Chart: Enabling Single-Line Reports

Step	Action	Comments
		message register is cleared by the console. See also: COS option 230 COS option 703. See also the following system options: System Option 49 System Option 23 System Option 40.
	13 Auto Wakeup Print	Allows wakeup audit printouts and single-line reports to be printed on the system printer whenever a wakeup call is set up, canceled, answered, or an attempt is made to honor the wakeup. No effect on SUPERSET 4150, SUPERSET 420, SUPERSET 430, and SUPERSET 4DN telephones.

Operation

None (automatic through PBX once programmed).

Suite Services

Description

Suite Services is a purchasable option that provides the ability to group a number of telephone lines through interconnected hotel or motel rooms (suites), for the purposes of billing and shared telephone services. For example, two bedrooms and a common area can make up a suite. A suite usually has a telephone in each bedroom, and one or more telephones in the common areas.

One of the suite telephones is regarded as the primary telephone or primary extension. The prime number of the primary telephone is the Primary DN for the suite. The other telephones (secondary telephones) in the suite are associated with the Primary DN.

All the telephones (DNIC or ONS) in the suite are grouped together for the following services.

- **Basic Call Handling**: When a call comes in to the primary number, all of the telephones will ring and all of the telephones can be used to answer the call. If a telephone is free, it can be used to answer another call or make a call. Multiple simultaneous calls are supported.

  All calls from the suite have the same Caller ID. A call made from any DNIC set in the suite will by default, select the primary line and be identified as the name and number of the primary DN. The primary number substitution is done for everything except when displaying 911 emergency calls to the console. An ONS telephone is associated to the primary DN in CDE Form 9 and this gives the ONS set a preferred line as a hidden multi-call line for the primary DN.
- **Message Waiting Indication**: A voice mail message left for the primary number will light the message lamps on all of the telephones in the suite. The message lamps will not all be in the same state when there is a message specific for the set. Examples of messages that are specific to the set are messages from the Message key, attendant console, front desk, or from a non-voice mail device using the Send Message feature access code
- **Voice mail Messages and Record a Call Recordings**: The voice mail messages and Record a Call recordings for all of the telephones in the suite are stored in one mailbox only (the primary telephone). Messages from the Message key, attendant console, or front desk are sent to the specific set only.

- **Call Forwarding**: Call forwarding for the suite will follow what is programmed for the primary telephone. Guests that want to forward their calls to another number; for example, a cell phone, program the call forwarding on the guest room telephone with feature access codes or request the call forwarding from the person at the front desk. On checkout, all the guest call forwarding information is cleared. The only number that can be forwarded is the primary number. Using the Call Forward toggle key on the primary telephone simplifies the on and off action. If forwarding is programmed for a secondary telephone this programmed forwarding is ignored. To discourage the guests from programming the DNIC suite sets, the Forwarding softkey will be hidden and the SUPERKEY functionality will be disabled. All ONS telephones in the suite can set call forwarding via access codes but the call forwarding will still only be applied to the primary DN.

  Call forwarding busy and no answer destinations can be programmed for an entire tenant group with Form 19. Users can still program their sets to forward. If programmed at the set, the set forwarding takes priority over the Form 19 destination (tenant-based forwarding). When the user turns forwarding off at their set, the Form 19 destination takes effect. See Call Forwarding.
- **Fax Services**: The FAX machine is called directly and will bypass the Do Not Disturb programmed to the suite. The number given out for the FAX machine should be the primary DN of the ONS port that connects to the FAX machine. Calls to the primary number of an ONS secondary telephone will bypass DND and will ring the device regardless of the DND. Outgoing calls on the FAX will still be recorded in the SMDR under the primary DN.

  If the user on a secondary phone is receiving a fax and quickly goes off hook/then on hook, he or she may receive dial tone. In this case the Class of Service and Class of Restriction should be setup so that a guest cannot call out by bypassing toll restrictions; making free long distance calls.
- **SMDR**: The Primary DN identifies calls from all the telephones in the suite. No matter what line the secondary telephone uses, the toll control will be of of the primary telephone.
- **Check-in and check-out**: The status change of the Primary Number includes the status change for all the telephones in the suite, applied from an attendant console, front desk terminal or a PMS system. To maintain privacy, the CLID information is cleared on checkout.The checking out also clears set based call forwarding, primary name, DND, messages waiting, wakeups, call logs, the message register, and call privileges for all the telephones in the suite.
- **Caller ID**: All telephones in the suite are identified (name and number) as the Primary DN whenever the identification is displayed on a set or console, except when displaying 911 calls on the attendant console. In this case, the actual extension number will appear.
- **Call Privileges**: The call restrictions or privileges assigned to the Primary Number can apply to all the telephones in the suite.

Conditions

- This feature requires the purchasable System Option 84, Multiple Guest Suite Phones.
- A suite can support one primary telephone and a maximum of four secondary telephones. If more than four are programmed, only four of them will function as secondary telephones.The choice of the four will be random.
- The Primary DN should be aligned with the number of the hotel room. The staff at the front desk use the Primary DN to represent all the telephones in the suite.
- This feature does not support Mitel 5201, SUPERSET 3DN , SUPERSET 4DN, SUPERSET 401 telephones or SUPERSET 4001 telephones as secondary or primary telephones. The primary telephone must be an ONS or a multi-line DNIC telephone; the secondary telephone a DNIC or ONS.
- The primary telephone must not have a multi-call line as the preferred line.
- The Primary DN cannot have any key-line appearances in the system. The Primary DN appears as a multi-call line on the secondary telephone DNIC sets and is their preferred line. The primary DN appears in the ASSOC field in CDE Form 09 for secondary ONS telephones.
- COS Option 262, Ignore Forward Busy with Free Appearance, enabled on a primary telephone allows a second call to ring a secondary set when one of the secondary telephones is free.

- Wakeups remain on individual telephones. The telephone that sets the wakeup is the telephone that receives the wakeup. If the guest calls for a wakeup, the staff using a front desk terminal or an attendant console can see which telephones are associated with each other, ask the guest to specify which telephone should receive the wakeup, and select the appropriate telephone for the wakeup. For more information on wakeups, see Attendant- Wakeups.

  Subattendant telephones do not provide information regarding secondary telephones associated with a primary telephone in a guest suite. However, if the subattendant knows the extensions of the secondary telephones, the subattendant can use the extension number of the secondary telephone to program a wakeup.

  Staff using a PMS system, have no way of programming wakeups on secondary telephones. Wakeups will only be directed towards the primary telephone. However, if the extension of the secondary telephone is known, then a wakeup can be programmed on that extension.
- Call Forwarding programmed by the attendant or from a telephone in the suite only applies to the primary DN. The Call Forward Toggle key will only work on the primary telephone. The Forwarding softkey is hidden on all of the DNIC suite sets.
- The SUPERKEY functionality is disabled for primary and secondary DNIC sets.
- Multi-call lines on DNIC telephones can be programmed for no ring, immediate ring, or delayed ring.
- Messages to secondary telephones that are sent via the Send Message feature access code are discarded.
- Secondary telephones are able to receive calls (regular or paged) on their prime line. These calls follow normal call processing rules.
- Call privileges on all the telephones can be associated together providing the installer assigns each telephone in the suite to the same COS, COR, and tenant group.
- Do Not Disturb can be applied to a single telephone (prime number) or to all the telephones (one prime number and the rest as secondary numbers) in a guest suite. The attendant console and the front desk terminal have softkeys DND All On and DND All Off that set all the room telephones. Secondary telephones can have DND applied so they do not ring when the prime number is dialed. Separate calls to the secondary telephones can still be made when in DND because DND is not honored when the secondary numbers have a prime line. Calls to the FAX machine bypass DND. To allow FAX machines to always receive a call, DND will be ignored on a prime line for a secondary ONS telephone.
- For billing disputes, SMDR provides the prime-line number of the telephone that made the call. This prime-line number appears in the Room Extension field. To have this number appear for outgoing calls, COS Option 246, SMDR - Extended Record must be enabled.
- All reports showing wakeup data will include wakeups on the primary telephone and the secondary telephones.
- Message register audits include all calls made by the room.
- Data generated at the Front Desk terminal and attendant console lists data only for the primary DN for the suite. For the Feature Usage Summary, there is one count each time a feature is used (primary and secondary telephones included) except for the non-zero Message Requests when there is one count per suite.
- Room searches exclude secondary phones and step only through Primary Directory Numbers.
- Feature Access Code 35, Maid in Room, affects the whole suite and is logged against the Primary DN. This code can be dialed from any telephone in the suite.
- Twin Phone COS Option 276 and the Guest Suite Extension COS Option 272 are incompatible and must not be both enabled in the same Class of Service.

Programming

In CDE Form 04, enable Option 84, Multiple Guest Suite Phones.

For the Primary Telephone (ONS or DNIC)

In CDE Form 09, program the primary telephone. Make the prime line the preferred line. Ensure that the primary telephone has no key-line appearance in the system and that the primary telephone does not have a multi-call line as its preferred line.

For DNIC Secondary Telephones

In CDE Form 03, enable Option 272, Guest Suite Extension.

In CDE Form 09, program the secondary telephone. In the Expand Set Subform, assign a multicall line key to the primary DN and make the multiline key the preferred line (LINE PREF softkey) for incoming and outgoing calls.

For ONS Secondary Telephones

In CDE Form 03, enable Option 272, Guest Suite Extension (ensure that COS Option 607, Telephone Associated Modem Line is not enabled). Recommendations are that all secondary sets in the suite have the same COS, COR, and Tenant Group.

In CDE Form 09, program the secondary telephone. Program the Primary DN in the ASSOC field.

Operation

None.

Mitel Telephone Room Status Display

Description

The display of an authorized Mitel telephone can show a combination of occupancy and condition (one of Vacant/Occupied/Reserved/ Guaranteed and one of Clean/Dirty/To be Inspected/Out of Service).

Table: Mitel Telephone Room Status Display

Description of Display			
Occupancy		**Condition**	
Vacant	VAC appears as the first word in the telephone display area.	Clean	CLEAN appears as the second word in the telephone display area.
Occupied	OCC appears as the first word in the telephone display area.	Dirty	DIRTY appears as the second word in the telephone display area.
Reserved	RES appears as the first word in the telephone display area.	Out of Service	SERV appears as the second word in the telephone display area.
Guaranteed	GUA appears as the first word in the telephone display area.	To be Inspected	INSP appears as the second word in the telephone display area.

Conditions

- Maid in Room is displayed separately as part of the Maid in Room feature.
- Call Restriction is not displayed.

- This feature applies to Mitel 5020 IP, 5220 IP, 5224 IP, 5330 IP, 5340 IP, SUPERSET 4025, SUPERSET 4125 SUPERSET 4150, SUPERSET 420, SUPERSET 430, and SUPERSET 4DN telephones.

Programming

In CDE Form 02, assign an access code to Feature 36 (SUPERSET Room Status Display). The display will show one room at a time.

In CDE Form 03, enable COS option 608 (Telephone - Room Status Display) in the set's Class of Service.

Operation

See the telephone procedure for changing the Maid in Room Status in the operation section of Maid in Room.

See the telephone procedure for monitoring Maid in Room Status in the operation section of Maid in Room Status Display - Mitel Display Telephones.

See the telephone procedure for monitoring Room Status in the operation section of Room Status Search.

For information about the Mitel 5340 IP phone acting as subattendant, see the *Subattendant User Guide for the Mitel 5340 IP Phone* available at Mitel OnLine.

Wakeups

The Wakeup feature provides automatic wakeup calls and personal wakeup calls to the guest room. The attendant, sub- attendant, or the guest can set up an automatic wakeup alarm call that rings the guest room at a prearranged time. The attendant or a subattendant can set up **personal** wakeup calls that alert the staff to personally call the guest room or **automatic** wakeup calls.

Wakeups consist of three types:

- **Single wakeup** - The user programs one automatic wakeup timer.
- **Daily wakeup** - The user programs an automatic wakeup timer to be repeated at the same time every day until canceled.
- **Multiple wakeup** - The user programs up to three simultaneous wakeup timers (automatic or personal) with the option of repeating daily until canceled.

Notes:

1. Guests can not set up personal wakeup calls.
2. Daily (repeated at the same time every day) wakeups and multiple wakeups are dependent on the FOR Option 102, Feature Level being set to level 1 or greater.
3. SUPERSET 3DN and 4DN telephones only support single automated wakeups.

On answering a wakeup alarm call, the guest receives a special tone (100 ms ON, 400 ms OFF, repeated), or music, or a recorded announcement. The recorded announcement can be changed for night service.

When a wakeup call is not answered within six rings, or when the extension is busy, two more attempts, at 5-minute intervals, are made to ring the extension. Each wakeup attempt can generate a single-line report on a system printer. If the call is still unanswered, the wakeup is canceled and the attendant can be notified that the wakeup was not honored.

Note: The wakeup call to a Mitel telephone is treated as a timed reminder (it rings once and sets a reminder prompt). (See Reminder in the Program Features section.) A wakeup attempt on a Mitel display telephone does not generate a single-line report.

If COS Option 322, Confirm Wakeup by Offhook, is enabled, the Mitel display telephones will ring until the handset goes offhook (or the ringing timeout expires).

Recorded announcements are provided through recording hunt groups. When an extension answers the wakeup call, the recording device is rung. The extension hears silence until the recording device answers. If a recorded announcement is used, a wakeup is only honored when there is a free recording announcement device available. Ensure that there are sufficient RADs to provide adequate wakeup service.

Attendant - Wakeups

Description

The attendant can set, change, audit, verify or cancel the wakeup time.

The attendant sets the wakeup on individual telephones. If a guest is staying in a guest suite with multiple telephones (secondary telephones associated with the DN of the primary telephone), the guest should specify which telephone requires the wakeup timer. The attendant sees all the telephones that belong to the guest suite on screen and can select the appropriate telephone.

Wakeup times are taken as AM unless they are specifically set for PM. The attendant can print a list of all the current wakeups. See Wakeup Audit. A maintenance log is created if the Hotel/Motel Wakeup printer fails.

Table: Attendant - Wakeups

Terminal	Description	Display
Attendant Console		

Softkeys on the console allow you to set and cancel wakeups, as well as, select the telephone if there are more than one displayed.

If the FOR Option 102, Feature Level is set to level 1 or greater, the attendant can set up to three automatic or personal wakeups that may or may not be repeated daily.
Personal wakeups behave similar to a Callback set by the attendant console. When the personal timer expires, a call rings on the LDN (defined in CDE Form 19) on the attendant console. When the attendant answers the call on the LDN, the guest's telephone rings. The attendant personally gives the wakeup call.

If the attendant or subattendant does not answer the callback on the LDN for the personal wakeup call, the personal wakeup call converts to a regular automatic wakeup call to the guest with

- multiple rings if COS Option 322 is enabled
- one ring with a flash reminder if COS Option 322 is disabled.

If the guest does not answer the personal wakeup call, the callback to the guest can be retried to the guest three times in five minute intervals.

The attendant can also be programmed to provide wakeup alarm notification when a guest does not respond to a wakeup call.

In each case, a printout automatically occurs at the associated printer if the system option to print these changes has been enabled.

Pressing WAKE-UPS, can show two different screens depending on the telephone configuration in the guest room.

- If the guest room has one telephone, the display shows the three wakeups that can be programmed on that telephone.
- If the guest room (suite) has secondary telephones associated with a primary telephone, the display identifies all of the telephones in the suite so the attendant can select the telephone to program. Position one identifies the primary telephone. Positions two to five identify the secondary telephones. A (w) indicator beside the identifed telephone indicates that there is at least one wakeup programmed for that extension; the (d) indicator shows that the telephone is in Do Not Disturb; (wd) shows that both are set.

  After the attendant selects the appropriate telephone, the display shows the three wakeups that can be programmed for that telephone.

The following are examples of three wakeups

Wk-up1 = 09:35AM
If the wakeup time is AM in a 12 hour format,

Wk-up2 = *03:30PM
If the wakeup time is PM in a 12 hour format, the asterisk (*) indicates that the wakeup is repeated daily

Wk-up3 = None
No wakeup time is set.

After pressing the SET WAKE-UP<1,2 or 3> softkey, further information on the second line of the display is shown. ONCE is for a wakeup call not to be repeated. AUTO is for an automatic call. PERSONAL is for a personal wakeup call. DAILY is for a wakeup call that repeats daily.

Front Desk Terminal	If no wakeup time is set for a room, the Set Wake Up softkey is displayed. If a wakeup time is set for a room, a Set Wake Up and a Clear Wake Up softkey are displayed. When the Set Wake Up softkey is pressed, all softkeys are cleared and a prompt to enter the wakeup time is displayed. Only the number keys, up/down arrow keys, and the Delete and Return hardkey are valid when setting the time. Time must be entered as four digits (hh:mm).

Wakeup times are entered, changed, and displayed in the three Wake Up Time fields on the Guest Room Update screen.

In cases where the guest has multiple telephones (at least one secondary telephone associated with a primary telephone), the Guest Room Update screen shows "Other Phones". Four secondary telephones can be listed. A (w) indicator beside the secondary telephone shows that there is at least one wakeup programmed for that telephone; the (d) indicator shows that the telephone is in Do Not Disturb; (wd) shows that both are set.The Message Register field shows the total message registers for the primary and secondary telephones.

The user can program three wakeup times. The wakeup times can be set to repeat daily. An asterisk (*) beside the time indicates a repeated wakeup call. The wakeup time can also be set as a personal call. "Pers" beside the time indicates a personal call.

If the 24-hour clock system option is enabled, the wakeup time is displayed in a 24-hour format. Otherwise, the wakeup time is displayed in 12-hour format with an AM/PM indication.

The Search Display screen displays the wakeup time for a guest room (if set).
A plus (+) sign after the wakeup time indicates that more than one wakeup time has been set. When a guest has more than one wakeup time set, the time shown is the one that is due next.

Each digit is validated as it is entered. Any digit is accepted that could be a valid time setting. A 24-hour setting is accepted as valid in the 12-hour mode: it is automatically converted to a 12-hour format when displayed. Any time entered in a 12-hour format is taken to be AM and is displayed with AM appended.

If time is entered in the 12-hour format, a PM softkey is displayed. This display allows time to be identified as morning or afternoon. Any time entered in the 12-hour format is assumed to be morning time unless the PM softkey is pressed. The key toggles the display from AM to PM. Cancel moves the shade bar to the next programmable field and leaves the wakeup time the same as it was when the form was displayed. The Enter softkey commits the wakeup time to the database.

Conditions

- If COS Option 322, Confirm Wakeup by Offhook, is enabled, the Mitel display telephones will ring until the handset goes offhook (or the ringing timeout expires).
- Daily (repeated at the same time every day) wakeups and multiple wakeups are dependant on the FOR Option 102, Feature Level being set to level 1 or greater.
- SUPERSET 3DN and 4DN telephones only support single automated wakeups.
- The personal wakeup call overrides COS Option 322, Confirm Wakeup by Offhook, if this option is disabled on the set.
- System Option 13 can be enabled to print changes to wakeup status. Enabling this option causes a SET BY CON or CHG BY CON message (attendant console) or a SET BY DESK or CHG BY DESK (front desk terminal) to be printed to indicate how the change was made.
- On the attendant console, if system time is in 12-hour format, a PM softkey is available for afternoon wakeup times. If system time is in 24-hour format, the PM softkey does not appear. Time must always

be entered as four digits (hh:mm). The system does not accept an invalid setting, but gives no error message.

- If a guest room telephone is programmed for Automatic Wakeup Routing, the Automatic Wakeup Music option has no effect.
- Recorded announcements are provided through recording hunt groups.
- A printer must be assigned to the “Hotel/Motel Wakeup” printout type in CDE Form 34 (Directed IO) for the status messages for wakeup activity. Audits of wakeups from the console are directed to the Hotel/Motel Audit printer.
- The attendant console identifies all the telephones in a guest suite if the suite has secondary telephones associated with the Primary DN for that suite. The extension names that appear are taken from CDE Form 9. If a name is not programmed in Form 9, then the extension number of the telephone will appear.
- If the LDN for a personal wakeup call is deleted after programming a personal wakeup call, the personal wakeup is converted to an automatic wakeup, tries three times and then expires.

Programming

To Set the System Time

- In CDE Form 04, enable System Option 01, 24-hour Clock. This option sets the system time in 24-hour format; otherwise the system time is in 12-hour format.

To program wakeups

- In CDE Form 03, enable COS option 101 (Attendant O/G Restriction/Room Status Setup) for the console. Enable COS option 202 (Alarm Call) and COS option 244 (Room Status Applies) for the extension.
- In CDE Form 04, enable System Option 11 (Automatic Wakeup Enable).

  CDE Form 04, Optional:

  Select System Option 12 (Automatic Wakeup Alarm) to allow the console alarm icon to be flashed, in the event that a wakeup call is not answered within three attempts. This option also generates a maintenance log.

  Enable System Option 13 (Automatic Wakeup Print) to print single-line reports. System Option 13 (Automatic Wakeup Print) allows wakeup audit printouts and one- line messages to be printed on the system printer whenever a wakeup call is set up, canceled, answered, or an attempt is made to honor the wakeup.

  Enable System Option 14 (Automatic Wakeup Music) if requested.

  Select System Option 34 (Auto Room Status Conversion/Auto Wakeup Print) to have an automatic audit of all wakeups in the system every day. This option also changes the room status. Set the time of day for the printout in System Option 56 (Auto Room Status Conversion/Auto Wakeup Print Timer). The default is 00:00 (midnight).
- In CDE Form 19, program Personal Wakeup Routing For This Tenant and Automatic Wakeup Routing For This Tenant

To program wakeup alarm notification

See Wakeup Alarm Notification.

Operation

Attendant Console

To set a wakeup time

Note: To set a wakeup time while you are talking to the guest begin at Step 4, but press the SET WAKE-UP softkey instead.

1. Press **FUNCTION**.
2. Press the **GUEST ROOM** softkey.
3. Enter the extension (room) number**.** The WAKE-UPS softkey appears.
4. Press the **WAKE-UPS** softkey.
   - If the guest room has one telephone, the display shows the three wakeups that can be programmed on that telephone.
   - If the guest room (suite) has at least one secondary telephone associated with a primary telephone, the display identifies all of the telephones in the suite so the attendant can select the telephone to program. If the attendant is talking to the guest, a THIS PHONE softkey appears. This softkey acts as a short-cut to select the extension that the caller is actually using for the call.
5. If the guest room is a suite with associated telephones, press the softkey corresponding to the appropriate telephone that you wish to program.

   Softkeys to set the three wakeup timers appear. Softkeys to clear the wakeup timers appear if the wakeup timers are programmed.The BACKUP softkey lets you return to the previous screen. The status of the three wakeups show on the screen. An asterisk prior to the programmed wakeup time indicates that the wakeup repeats daily until canceled. AM and PM indicators only appear if the system time is in 12 hour format.
6. Press the wakeup timer softkey that you want to set (SET WAKE-UP1, SET WAKE-UP2, or SET WAKE-UP3). The screen prompts you to enter the wakeup time. The Backup and Same Time softkeys appear. The Backup softkey allows you to return to the previous screen.The Same Time softkey appears if a time is already programmed and allows you to reset the timer with the previous time.
7. Enter the wakeup time. If the time is not in 24-hour format, you can toggle between the AM and PM softkey. The default in 12-hour format is AM. A valid wakeup time has the format HH:MM where HH is between 00 and 23 for 24-hour clocks and 00 and 11 for 12-hour clocks, and MM is 00 to 59.
8. Press the **DAILY** softkey for a wakeup that repeats everyday until canceled. The default setting is ONCE. ONCE means that the wakeup only occurs one time.
9. Press the **PERSONAL** softkey for a wakeup that you deliver personally. The default setting is AUTO-WKUP (not a personal wakeup).
10. Press the **SET** softkey. When you are finished, press the EXIT softkey. To check if a wakeup is programmed as Personal or Automatic, press the waker timer softkey (e.g.SET WAKEUP1), press SAME TIME, and the display shows the programmed time and type. To change the time use the back arrow key.

To cancel a wakeup time

1. Press **FUNCTION**.
2. Press the **GUEST ROOM** softkey.
3. Enter the extension (room) number.
4. Press the **WAKE-UPS** softkey. If there are multiple telephones, press the appropriate softkey for the telephone you wish to program. Softkeys to clear the wakeup timers appear if the wakeup timers are programmed.

5. Select the wakeup softkey that you want to cancel (CLR WAKE-UP1, CLR WAKE-UP2, or CLR WAKE-UP3).

To verify wakeup notifications

You can print out the wakeup status to the SMDR socket. This is requested from the console. Ensure that in Form 4, option 13 is enabled, and in Form 34, Output HM wakeup to SMDR socket is enabled.

After a guest in Room 108 set up a wakeup for 8:33am on May 5th at 8:30pm:

SMDR output:

108 05/05 20:30 WU-1 08:33 SET BY SUB

Cancelled from the room phone 108:

108	05/05	20:33	WU-1	08:33	CAN BY SUB

To audit wakeup notifications

You can audit wakeup notifications if they are not answered. Ensure that COS 322 is enabled for the room phone if it is a DNIC or IP set.

1. Press **FUNCTION**.
2. Press the **GUEST ROOM** softkey.
3. Press the **AUDIT** softkey.
4. Press the **WAKEUPS** softkey.

The wakeup notification rings the room phone 3 times every 5 minutes, 6 rings each time.

Generated SMDR Output: Wakeup not answered as well as console alarm

108	05/05	21:17	WU-1	21:17	NO ANS1
108	05/05	21:17	WU-1	21:17	NO ANS2
108	05/05	21:17	WU-1	21:17	NO ANS3 **

To access alarm notification data:

1. Press **FUNCTION**
2. Press the **ALARM** softkey. All alarm information is displayed.

Front Desk Terminal

To set a wakeup time

1. From the House Statistics screen, press the **Room Update** softkey.
2. Enter the guest room number. The Guest Room Update screen is displayed.
   If the guest room (suite) has at least one secondary telephone associated with a primary telephone, the display identifies the secondary telephones so you can select the telephone to program.
3. If you want to program a wakeup on a secondary telephone, move the shaded bar to the extension of the secondary telephone and press the Select softkey. The Guest Room Update screen shows the Phone ID as programmed in CDE Form 09 and the three wakeup fields.
4. On the Guest Room Update screen, move the shaded bar to the wakeup field and press the **Set Wakeup** softkey. The prompt Enter Wakeup Time (hh:mm) is displayed. The softkeys disappear as shown in this Figure.
5. Enter the time, and press the appropriate softkeys to define the type of wakeup call. Only the number keys and the Delete hardkey are valid when setting the time. Time must be entered as four digits. The Delete hardkey deletes the digits entered.
   The prompt Press Enter to confirm Wakeup Time appears.
   Once four digits are entered, Cancel, Enter, AM/PM (if the 12-hour clock option is enabled),Daily/Once, Auto Wk-Up/Personal softkeys appear.
   The default setting for the wakeup is Once and Auto Wk-Up providing the System Option 102, Feature Level is enabled to level 1 or greater.
6. For 12-hour formats, press the **PM** softkey if you are setting an afternoon time. Any time entered in the 12-hour format is assumed to be morning time unless the PM softkey is pressed.
   A 24-hour setting is accepted as valid in the 12-hour format. It is automatically converted to a 12-hour format. The AM/PM softkey toggles between AM and PM. AM is the default.

Note: Use the **Delete** hardkey to delete the last digit of the time (digit by digit).

If you are setting the wakeup time and decide not to, press the **Cancel** softkey. You must have entered the complete time (the full four digits). The shade bar moves to the next programmable field, and leaves the wakeup field unchanged. To continue making changes, press the **Return** hardkey. To end, press the **Enter** softkey.

Figure: Wakeup Time Prompt

To cancel a wakeup time

Note: When a wakeup time is set, a Clear Wakeup softkey is displayed.

1. On the Guest Room Update screen, move the shaded bar to the wakeup field and press the **Clear Wakeup** softkey. The wakeup is cleared. The shade bar moves to the next field that can be modified. If there are multiple telephones on the Guest Room Update screen, press the appropriate softkey for the telephone you wish to program, and then move the shaded bar to the wakeup field and press the **Clear Wakeup** softkey.
2. To continue making changes, press the **RETURN** hardkey.
   To end, press the **Enter** softkey.

Subattendant - Wakeups

Description

This Wakeup feature allows users of SUPERSET 4150, SUPERSET 430, SUPERSET 4DN, or Mitel 5340 IP telephones, programmed as subattendants, to set up an automatic wakeup alarm call that rings the guest room telephone at a prearranged time. If the FOR option, Feature Level, is set to level 1 or greater, users of SUPERSET 4150 and SUPERSET 430 telephones can also program a daily automatic wakeup or multiple (up to three in a 24-hour period) automatic or personal wakeups with the option of repeating daily. The programmed wakeup times can be changed or canceled.

A printout or audit shows the room number, wakeup date and time, and the time the wakeup call is set, changed, or cancelled.

Conditions

- SUPERSET 4DN telephones only support a single wakeup (TIMER 1). SUPERSET 4DN telephones do not support daily wakeups, multiple wakeups, or personal wakeups.
- FOR Option 102, Feature Level must be set to level 1 or greater to program daily wakeups, multiple wakeups, or personal wakeups on the SUPERSET 4150 and SUPERSET 430 subattendant telephones.
- Personal wakeups behave similar to a Callback set by the subattendant. When the personal timer expires, a call rings on the LDN (defined in CDE Form 19) of the sub attendant's telephone. When the subattendant answers the call on the LDN, the guest's telephone rings. The subattendant personally gives the wakeup call.
- Personal wakeup calls override COS Option 322, Confirm Wakeup by Offhook if this option is disabled on the set.
- If the attendant or subattendant does not answer the callback on the LDN for the personal wakeup call, the personal wakeup call converts to a regular automatic wakeup call to the guest with
  - multiple rings if COS Option 322 is enabled
  - one ring with a flash reminder if COS Option 322 is disabled.
- If the guest does not answer the personal wakeup call, the callback to the guest will be retried three times at five minute intervals. You can also program the attendant to receive alarm notification for missed wakeup calls.
- Wakeup settings are not lost with system upgrades.
- An asterisk beside an active time on a set display indicates a timer that repeats daily at the same time until canceled.
- Do Not Disturb has no effect on the automatic wakeup feature.

- Do not enable Auto-Answer on telephones that are to receive an automatic wakeup. If Auto-Answer is enabled on the telephone, the automatic wakeup will not be presented to the set.
- If the wakeup call is not acknowledged, an automatic wakeup will give three 5 ring calls at five minute intervals. The automatic wakeup call expires after the third set of rings and generates a #125 alarm code on the maintenance log.
- When COS Option 322, Confirm Wakeup by Offhook, is enabled, the recipient of the automatic wakeup shall respond to the wakeup call by going offhook.
- If COS Option 322, Confirm Wakeup by Offhook, is disabled, the set will give individual rings at minute intervals until the recipient responds.The recipients responding to the wakeup call shall respond differently on the following telephones:
  - Mitel 5020 IP, 5220 IP, 5224 IP, 5324 IP, SUPERSET 4025, SUPERSET 4125 and SUPERSET 420 users press Confirm
  - SUPERSET 4150, SUPERSET 430, and SUPERSET 4DN users press Acknowledge
  - Non-display telephone users go offhook.
- System Option 13, Automatic Wakeup Print, must be enabled in order to get printouts of the automatic wakeup logs.
- Most subattendant telephones do not provide information regarding secondary telephones associated with a primary telephone in a guest suite. However, if the subattendant knows the extensions of the secondary telephones, the sub- attendant can use the extension number of the secondary telephone to program a wakeup. **Note**: The 5340 IP phone, when programmed as a subattendant, can acess and program wake-ups for all phones in a suite.
- If the LDN for a personal wakeup call is deleted after programming a personal wakeup call, the personal wakeup is converted to an automatic wakeup, tries three times and then expires.

Programming

In CDE Form 02, assign an access code to Feature 32, Automatic Wakeup. SUPERSET 4150, SUPERSET 430, or SUPERSET 4DN telephones programmed as a subattendant can use the feature access code or the Superkey to set up a wakeup call.

In CDE Form 04, enable System Option 11 (Automatic Wakeup). Set System Option 102 (Feature Level) to level 1 or greater for daily wakeups,multiple wakeups, and personal wakeups.

Optional: In CDE Form 04, enable System Option 13 (Automatic Wakeup Print) to print single-line reports. Assign a Hotel/Motel Wakeup printer in CDE Form 34 (Directed IO). Enable System Option 14 (Automatic Wakeup Music) if requested.

In CDE Form 19, program Personal Wakeup Routing For This Tenant and Automatic Wakeup Routing For This Tenant.

Operation

SUPERSET 4150 and SUPERSET 430 telephones (Subattendants)

To set up a single wakeup calls using the Superkey.

Note: FOR Option, Feature Level is disabled. This procedure reflects the 12-hour clock format programmed with Option 1 (Clock Format) in CDE Form 04. If 24-hour format is set, then the AM and PM are not included.

- Press **Superkey**.
- Press **Stations**.
- Dial the extension number of the set that will receive the wakeup call. In case of error use the ← key.
- Press **Wake-up** .The display showing TIMER 1 appears.
- Dial the desired time in a 12-hour format (e.g. 01:45). In case of error, use the ← key.
- Specify AM or PM by pressing the **AM/PM** softkey.

- Press **Save**.

To set up multiple wakeups (up to 3 timers) using the Superkey.

Note: Programming multiple wakeups is dependant on the FOR Option, Feature Level, being set to level 1 or greater. This procedure reflects the 12-hour clock format programmed with Option 1 (Clock Format) in CDE Form 04. If 24-hour format is set, then the AM and PM are not included.

Wakeup calls may be programmed with four possible combinations: Once/Auto, Once/Personal, Daily/Auto, and Daily/Personal.

- Press **Superkey**.
- Press **Station**.
- Dial the extension number of the set that will receive the wakeup call. In case of error use the ← key.
- Press **Wake-up**. The display showing TIMER 1 appears. Timer 2, Timer 3, Backup, and Same Time softkeys appear. The Backup softkey allows you to return to the previous screen.The Same Time softkey appears if you wish to reset the timer with the previous time.
- Dial the desired time in a 12-hour format (e.g. 01:45). In case of error, use the ← key. AM, ONCE, and AUTO appear after the time is entered. These are the default settings.
- Press **PM** if required. The default is AM.
- Press **Daily** if you wish to have the timer repeated daily (the DAILY softkey only appears if the FOR option Feature Level is set to level 1 or greater). DAILY-AUTO appears after pressing the Daily softkey.
- Press **Personal** if you wish the wakeup to be a personal wakeup call instead of an automatic wakeup call.
- Press **Save**.
- Repeat the first four steps.
- Press **Timer 2** to access the TIMER 2 display.
- Program the wakeup time for TIMER 2 using the same procedure for TIMER 1.
- Repeat the first two steps again to access the TIMER 3 display.
- Program the wakeup time for TIMER 3 using the same procedure for TIMER 1.
- Press **Save**.
- Press **Superkey** to return to normal display.

To display the wakeup settings using the Superkey.

- Press **Superkey**.
- Press **Stations**.
- Dial the extension number of the set that will receive the wakeup call.
- Press **Wake-up**. The wakeup setting for TIMER 1 is displayed. DAILY appears if the wakeup repeats on a daily basis. ONCE appears if the wakeup is set for a one time occurrence. PERS displays if the wakeup is a personal wakeup. AUTO appears if the wakeup is not a personal wakeup.
- Press **Timer 2** or **Timer 3** to display the wakeup settings on the respective timers.
- Press **Superkey** to return to normal display.

To cancel a wakeup before it occurs using the Superkey.

- Press **Superkey**.
- Press **Stations**.
- Dial the extension number of the set that will receive the wakeup call.
- Press **Wakeup**. The display showing the time setting for TIMER 1 appears. Pressing the TIMER 2 or TIMER 3 softkey displays the respective time setting for the chosen timer.
- Press **Cancel**.

To change the current wakeup setting using the Superkey.

- Press **Superkey**.
- Press **Stations**.
- Dial the extension number of the set that will receive the wakeup call.
- Press **Wakeup**. The display showing the time setting for TIMER 1 appears. Pressing the TIMER 2 or TIMER 3 softkey displays the respective time setting for the chosen timer.
- Dial the desired wakeup time in a 12-hour format (e.g. 01:45). In case of error, use the ← key. AM, ONCE, and AUTO appears after the time is entered. These are the default.
- Press **PM** if required. The default is AM.
- Press **Daily** if you wish to have the timer repeated daily (the Daily softkey only appears if the FOR option Feature Level is set to level 1 or greater). DAILY-AUTO appears after pressing the Daily softkey.
- Press **Personal** if you wish the wakeup to be a personal wakeup call and not an automatic wakeup. Wakeup calls may be programmed with four possible combinations: Once/Auto, Once/Personal, Daily/Auto, and Daily/Personal.
- Press **Save**.
- Press **Superkey** to return to normal display.

To set up a single wakeup call using the keypad (access codes)

- Obtain dial tone.
- Dial the Automatic Wakeup access code and immediately dial the extension number of the set that will receive the wakeup call. A dial tone is returned if all conditions are set to receive a wakeup time.
- Dial the desired time in 24-hour format. A dial tone is returned if all conditions are set.

To set up a daily, single wakeup call using the keypad

- Obtain dial tone.
- Dial the Automatic Wakeup access code and immediately dial the extension number of the set that will receive the wakeup call. A dial tone is returned if all conditions are set to receive a wakeup time.
- Dial in an asterisk (*). The asterisk specifies a wakeup time that will repeat at the same time on a daily basis.
- Dial the desired time in 24-hour format. A dial tone is returned if all conditions are set.

To set up multiple wakeups (up to three) using the keypad

- Obtain dial tone.
- Dial the Automatic Wakeup access code and immediately dial the extension number of the set that will receive the wakeup call. A dial tone is returned if all conditions are set to receive a wakeup time.
- Dial in a pound character and the number of the timer (#1, #2, or #3). If you wish to repeat this timer daily, dial in an asterisk (*) after the timer number.
- Dial in the wakeup time. The time is specified by dialing the hour and minutes in 24-hour time format.
- Reorder tone is returned for an invalid time; dial tone is returned for a valid time.

To cancel a single wakeup (or Timer 1) using the keypad

- Obtain dial tone.
- Dial the Automatic Wakeup code and immediately dial the extension number of the set that receives the wakeup call. A dial tone is returned if all conditions are set.
- Dial four 9's. A dial tone is returned. The wakeup time is canceled.

To cancel multiple wakeups (Timer 1, Timer 2 or Timer 3) using the keypad

- Obtain dial tone.
- Dial the Automatic Wakeup code and immediately dial the extension number of the set that receives the wakeup call. A dial tone is returned if all conditions are set to receive a timer number.

- Dial in the pound character followed by the timer number (#1, #2, or #3).
- Dial four 9's. A dial tone is returned. The wakeup time is canceled.

SUPERSET 4DN (Subattendants)

To set up an automatic wakeup call using the SUPERKEY.

Note: This procedure reflects the 12-hour clock format programmed with Option 1 (Clock Format) in CDE Form 04. If 24-hour format is set, then the AM and PM are not included.

- Press the SUPERKEY.
- Press the STATIONS softkey.
- Dial the extension number of the set that will receive the wakeup call. In case of error use the ← key.
- Press the WAKE-UP softkey.The display showing TIMER 1 appears.
- Dial the desired time in a 12-hour format (e.g. 01:45). In case of error, use the ← key.
- Specify AM or PM by pressing the AM/PM softkey.
- Press the SAVE softkey.
- Press SUPERKEY to return to normal display.

To display the current wakeup setting

- Press the SUPERKEY.
- Press the STATIONS softkey.
- Dial the extension number of the set that will receive the wakeup call.
- Press the WAKE-UP softkey. The current wakeup time is displayed.
- Press the SUPERKEY to return to normal display.

To change the current wakeup setting

- Press the SUPERKEY.
- Press the STATIONS softkey.
- Dial the extension number of the set that will receive the wakeup call.
- Press the WAKE-UP softkey. The current wakeup time is displayed.
- Dial the desired wakeup time.
- Press the SAVE softkey.
- Press the SUPERKEY to return to normal display.

To cancel the automatic wakeup

- Press the SUPERKEY.
- Press the STATIONS softkey.
- Dial the extension number of the set that will receive the wakeup call.
- Press the WAKE-UP softkey. The current wakeup time is displayed.
- Press the CANCEL softkey.

To set up an automatic wakeup call using the keypad

- Obtain dial tone.
- Dial the Automatic Wakeup access code and immediately dial the extension number of the set that will receive the wakeup call. A dial tone is returned if all conditions are set to receive a wakeup time.
- Dial the desired time in 24-hour format. A dial tone is returned if all conditions are set.

To cancel a single wakeup (or Timer 1) using the keypad

- Obtain dial tone.
- Dial the Automatic Wakeup code and immediately dial the extension number of the set that receives the wakeup call. A dial tone is returned if all conditions are set.

- Dial four 9's. A dial tone is returned. The wakeup time is canceled.

Mitel 5340 IP Phone programmed as Subattendant:

Note: COS Option 202 (Alarm Call), must be enabled in the COS of the guest room.

To set, change, or cancel a wake-up call time for a guest:

- Press the **Guest Room** feature key.
- Dial the extension number of the set that will receive the wakeup call.
- Press the **Wake-up** softkey. The display shows any previously set wake-up call time and, if available, Timer 2, Timer 3, Delete and Change.
- Enter the desired time in a 12-hour format (e.g. 01:45). In case of error, use the <-- key.
- Specify AM or PM by pressing the AM/PM softkey.
- Specify a system or personal call by pressing the **Auto Wkup** or **Personal** softkey.
- Specify a repeating or one-time only call by pressing the **Daily** or **Once** softkey
- Press **Save**.
- Repeat for Timers 2 and 3 to set additional wake-up calls for the guest.
- To change or delete a wake-up call, press the desired Timer softkey (not required to change/delete Timer 1), followed by the **Delete** or **Change** softkey.

Guest Room - Wakeups

Description

Guests can set and cancel wakeups. The wakeups can be programmed to go off once or daily. Up to three wakeups can be set in a 24-hour period. For the EMEM feature, only one wakeup can be scheduled per 24-hour period.

Guests can set the wakeup times with an access code using the key pad or by using the menu under Superkey. If using the Superkey, the wakeups are programmed as reminders.

Key Pad Access for the Guest

The Guest Room Wakeup feature allows a guest to set and cancel a wakeup call from any room telephone by dialing an automatic wakeup access code. Wakeup time must be in 24-hour format and is entered by dialing from the telephone. Dial tone indicates that a wakeup time has been set or canceled successfully.

If a guest room cancels a wake up by dialing the automatic wakeup access code followed by the digit "9" four times (9999), a single line report CAN BY ROOM is printed.

If the FOR Option 102, Feature Level is set to level 1 or greater, a guest may program the telephone (SUPERSET 4DN telephone excluded) to support a single daily timer, or up to three timers that may be repeated daily.
To program a daily timer, the guest uses the asterisk (*) .The guest dials the automatic wakeup access code, the asterisk, and then the time in 24-hour format.
To program three simultaneous timers, the guest uses the pound and the number 1, 2, or 3 (#1, #2, #3) to identify Timer 1, Timer 2, or Timer 3. The guest dials the automatic wakeup access code, the pound and number of the timer, and then the time in 24-hour format (example 26#11425) To program the timer to repeat daily, the guest dials an asterisk before dialing in the time (example 26#1*1425).
To cancel the timer, the guest dials the automatic wakeup access code, the pound and number of the timer, and then the digit "9" four times (9999). Dialing the automatic wakeup access code and then the digit "9" four times (excluding the pound and the number of the timer) cancels the wakeup time for Timer 1.

Key Pad Access for the Guest for EMEM

The Guest Room Wakeup feature allows a guest to set and cancel a wakeup call from any room telephone by dialing the voice mail hunt group number. Wakeup time must be in 24-hour format and is entered by dialing from the telephone. Dial tone indicates that a wakeup time has been set or canceled successfully. Only one wakeup can be set.

From the room phone:

- Dial the voice mail hunt group number
- Dial the "W" (the 9 key) to set up the wakeup.
- If no wakeup was set on this room phone, enter the wakeup time in 24-hour format. (Press # if you need to cancel your wakeup call).
- If a previous wakeup was set on the room phone, the guest hears "The current wakeup time is XX date, XX time. Press # to cancel the wakeup.
- Enter the wakeup time in 24-hour format.

Superkey Access for the Guest

Mitel 5020 IP, 5220 IP, 5224 IP, 5324 IP, 5330 IP, 5340 IP, SUPERSET 4025, SUPERSET 4125, SUPERSET 4150, SUPERSET 420, SUPERSET 430, and SUPERSET 4DN telephones support automatic wakeup calls in the form of reminders. If the FOR Option 102, Feature Level is set to level 1 or greater, users of these telephones (SUPERSET 4DN telephone excluded) may program their single timer to repeat daily, or may program up to three timers that may be repeated daily. The Superkey on Mitel 5020 IP, 5220 IP, 5224 IP, 5324 IP, 5330 IP, SUPERSET 4025, SUPERSET 4125, and SUPERSET 420 telephones provide access to TIMED REMINDER? The Superkey on Mitel 5340 IP, SUPERSET 4150, SUPERSET 430, and SUPERSET 4DN telephones provide access to Reminder. See the feature, Reminder in the Program Features section.

Conditions

- If a guest room telephone is programmed for Automatic Wakeup Routing, the Automatic Wakeup Music option has no effect.
- Do Not Disturb is ignored on extensions when the wakeup is honored.
- If the Wakeup call time is set by both the system and by the room occupant, the time used is the wakeup time most recently set.
- System Option 56 (Room Status Conversion/Auto Wakeup Print Timer) only applies if System Option 34 (Auto Room Status Conversion/Auto Wakeup Print) is enabled.
- In the event of a power failure, all expired wakeups are honored within 5 minutes of a system reset.
- Recorded announcements are provided through recording hunt groups.
- If a recording hunt group is used for wakeups, there can only be one wakeup call per recording announcement device at any given time. If multiple wakeups have been set for the same time, only one can proceed. All the others are shifted. The amount of this shift is variable depending on:
  - the time it takes for each preceding extension to answer/acknowledge the wakeup
  - the 1-minute period that is between wakeups being scanned.

Programming

In CDE Form 02, assign an Access Code to Feature 32 (Automatic Wakeup).

In CDE Form 03, enable COS Option 202 (Alarm Call) and COS Option 244 (Room Status Applies) for the extension.

In CDE Form 04, enable System Option 11 (Automatic Wakeup Enable) for Industry-Standard, Mitel 5201, SUPERSET 4001, SUPERSET 4015, SUPERSET 401+, TeleMatrix 3000IP, and SUPERSET 410 Telephones. System Option 11 is not required for Mitel 5020 IP, 5220 IP, 5224 IP, SUPERSET 4025,

SUPERSET 4125, SUPERSET 4150, SUPERSET 420, SUPERSET 430, and SUPERSET 4DN telephones.

In CDE Form 04, enable System Option 13 (Automatic Wakeup Print) to print CHG BY ROOM, SET BY ROOM, CAN BY ROOM single-line reports.

In CDE Form 04, set System Option 102 (Feature Level) to level 1 or greater to support daily, multiple, and personal wakeup calls.

Operation

Industry-Standard, Mitel 5201, SUPERSET 4001, SUPERSET 4015, SUPERSET 401+, TeleMatrix 3000IP, and SUPERSET 410 Telephones

To program a single wakeup

- Dial the automatic wakeup access code and then the time in 24-hour format. Dial tone indicates that a wakeup time has been set successfully.

To program a daily wakeup

The asterisk identifies the wakeup as daily.

- Dial the automatic wakeup access code, the asterisk, and then the time in 24-hour format. Dial tone indicates that a wakeup time has been set successfully.

To program three simultaneous wakeups

The pound sign and the number 1, 2, or 3 (#1, #2, #3) identifies Timer 1, Timer 2, or Timer 3.

- Dial the automatic wakeup access code, the pound and number of the timer, and then the time in 24-hour format (example 26#11425). Dial tone indicates that a wakeup time has been set successfully.

  To program the timer to repeat daily, the guest dials an asterisk before dialing in the time (example 26#1*1425).

To cancel the wakeup

- Dial the automatic wakeup access code, the pound and number of the timer, and then the digit “9” four times (9999). Dialing the automatic wakeup access code and then the digit “9” four times (excluding the pound and the number of the timer) cancels the wakeup time for Timer 1. Dial tone indicates that a wakeup time has been canceled successfully.

Mitel 5020 IP, 5220 IP, 5224 IP, 5324 IP, 5330 IP, 5340 IP, SUPERSET 420, SUPERSET 430, SUPERSET 4025, SUPERSET 4125, and SUPERSET 4150 telephones

Wakeup calls function as reminder calls. See the operation section of Reminder.

Wake-up Alarm Notification

Description

When wake-up calls are not answered after three attempts, the attendant can be programmed to receive wake-up alarm notification. The WkUp Alarm feature key is used for visual wakeup alarm notification on all display sets that have softkeys.

Conditions

Wakeup alarm notification can be accessed by only one subattendant user at a time. Subsequent users receive an error message stating that the feature is in use.

Programming

In CDE Form 04 enable System Option 12 (Automatic Wake up Alarm).

In CDE Form 03 enable COS Option 102 (Attendant Display of System Alarms) in the COS of the attendant.

For visual wakeup alarm notification on display sets:

In CDE Form 03, enable COS option 624 (Wakeup Alarm Notification) in the COS of the display set.

Operation

Attendant Console:

When activated by a missed wakeup call, the text "ALARM: flashes on the lower right-hand corner of the console display. The attendant can press FUNCTION and then select the ALARM softkey to access alarm data. **Note**: alarm data includes all alarms set by the system.

Subattendant

The LED of the WkUp Alarm feature key flashes when there is one or more alarms.

This feature on display sets does not interact with the attendant feature. If your system has both console and display sets programmed to receive wakeup alarm notification, they operate independently. Deleting the wakeup alarm notification on the console will have no effect on the subattendant, and vice versa. Console and display set can access wakeup alarms at the same time. The most recent transaction made by either user overrides the data set by a previous user.

After displaying the alarm data, use the Next softkey to navigate through alarms; the Delete softkey to delete the alarm; or the Backup key to exit the display.

Property Management System (PMS)

Introduction

A Property Management System (PMS) provides a center for managing a hotel business. The PMS system can provide reservation control, centralized accounting and billing, and call logging.

IP-enabled PMS applications can communicate with the SX-200 ICP via a Telnet connection. Applications that require a serial interface must use a third-party Serial-to-IP port converter to connect to the network.

When information about a guest is changed at the PMS system, messages are sent to the PBX via the PMS. Similarly when information about any guest is changed on the PBX, messages are sent via the PMS to the PMS system (see the Figure below). In brief, the PMS is an intermediary for passing messages from the PBX to the PMS system and from the PMS system to the PBX.

PBX Database	**PMS Database**
	Room Status
Message Register	Message Register
Message Waiting	Message Waiting
Automatic Wakeup	Automatic Wakeup
Guest Names	Guest Names
Do Not Disturb	

PBX - PMS Interface

The PBX's PMS interface collects and sends Hotel/Motel information to a customer's PMS computer through an IP socket. The PMS computer communicates with the PBX via Telnet or an RS232-to-IP serial port converter (customer-supplied) that connects the computer's serial port to the network. The PBX interfaces to IBM-compatible personal computers with the Lodgistix PMS software package (or a package that follows the same protocol). A PMS message is a string of ASCII text characters.

The PMS interface exchanges the following information between the PBX and the PMS:

- Maid status and room status
- Guest check-in and check-out
- Guest name
- Auto wakeup and message waiting
- PMS related maintenance logs.

Hotel/Motel information is stored in the PBX and in the PMS as follows:

Room Status

Room status (also called Maid in Room status or Maid status) allows the maid to change the status of the guest room from the room telephone and to indicate to the PMS terminal which room the maid is in. Monitoring and displaying of room status is completely controlled from the PMS terminal. The digits dialed by the maid have no meaning to the SX-200 ICP and are only used by the PMS.

Message Registration Of Outgoing Trunk Calls

Message Registration Of Outgoing Trunk Calls provides the PMS with a count of how many outgoing trunk calls were made from a room. This feature treats long-distance calls the same as local calls; the hotel guest is charged the same amount for either one. Extra charges for long distance calls are handled by a call-accounting device which may be connected to the PMS or integrated into it. When the hotel guest checks out, the PMS will have automatically added the cost of all calls to the bill.

Check In/Out

Check in / check out allows the PMS terminal to enable outgoing trunk calls from the room phone when a guest has checked into the hotel, and to disable these calls when a guest has checked out. This prevents hotel staff from making unauthorized calls from rooms.

Upon guest check-out, the PMS Interface clears the following items in the guest room database for the primary and secondary telephones: the Message Register, Message Waiting, Do Not Disturb, Automatic Wakeup, and Call Forwarding. The Guest Name is cleared for the primary telephone only.

Automatic Wakeup

From PMS terminal: Automatic wakeup allows the PMS terminal to set up or cancel a wakeup request for a guest room. If programmed in the SX-200 ICP a printout will be produced upon a setup of a valid time to verify that the request was committed as well when the wakeup is honored a printout will be produced to verify the receipt of the wakeup call.

From room telephone: To set, modify, or cancel the current reminder (wakeup) setting from the room telephone see Reminder in the Program Features section.

Notes:

1. Enable COS Option 202 (Alarm Call) and COS option 322 (Confirm Wakeup by Offhook) in the extension's COS.

2. Enable System Option 11 (Automatic Wakeup) and assign an access code to Feature 32 (Automatic Wakeup).

Message Waiting

Message waiting allows the message center PMS terminal to set up and cancel a message waiting indication for a guest room phone. If programmed in the SX-200 ICP a printout will be produced to verify that the message waiting was sent and the time it was sent. When the message is cancelled from the PMS system another printout will be produced to verify the cancellation.

In Form 19, Call Rerouting: program the “Station Dial 0" answer point for all tenants to be a Console LDN or subattendant LDN. To allow the customer’s PMS to set message waiting, you must program Day, Night 1, and Night 2 answer points to the same LDN.

Programming the SX-200 ICP PMS Interface

Feature Interactions

The following conditions apply to the SX-200 ICP/PMS interface:

- Disable the following options because all room status updates are applied against the PMS database:
  - System Option 33 (Room Status)
  - System Option 27 (Room Status Audit)
  - System Option 34 (Auto Room Status Conversion and Wakeup Print)
  - COS option 244 (Room Status Applies)
  - COS option 608 (Telephone - Room Status Display)
- Maintenance logs record any errors found in the PMS interface. Refer to Maintenance Logs and to Troubleshooting for details.
- Message Waiting must be enabled in COS Option 232 for the guest room phone if it is an industry-standard telephone. This restriction does not apply to Mitel 5020 IP, 5220 IP, 5224 IP, SUPERSET 4025, SUPERSET 4125, SUPERSET 4150, SUPERSET 420, SUPERSET 430, and SUPERSET 4DN telephones.
- System Option 32 (Outgoing Call Restriction) must be enabled to allow all rooms to have emergency 911 access

1. Form 11, Circuit Descriptor: Determine the customer’s PMS communications protocol requirements: baud rate, parity, character length, and number of stop bits. Program to match the protocol; other options remain at default values.

Option Name	Value
Session Inactivity Disconnect Timer	0
Guard Timer	2
Minimum Baud Rate	match customer’s PMS
Default Baud Rate	match customer’s PMS
Maximum Baud Rate	match customer’s PMS
Always use Default Baud Rate when called	YES
DTR Off Disconnect Timer	5
DTR to CTS Delay Timer	100
DTR Forced High	YES

Option Name	Value
RTS Forced High	YES
DSR Is held High when device is Idle	YES
CTS Is held High when device is Idle	YES
Originate a DTRX Call with Low - High transition of DTR	NO
Action taken if the Idle DTE has DTR Low (Auto-Answer)	REFUSE
ASYNC: Keyboard Origination Allowed (Auto Baud)	DISABLE
ASYNC: ADL Auto Baud	DISABLE
ASYNC: Flow Control	XON/XOFF
ASYNC: Parity	match customer's PMS
ASYNC: Character Length	match customer's PMS
ASYNC: Number Of Stop Bits	match customer's PMS
DS2100: Operating Mode	ASYNCHRONOUS

2. Form 12, Data Assignment: Program an IP SOCKET type data device to an available PLID in the following range:

PLID	Port
1/13/20	61320
1/13/21	61321
1/13/22	61322
1/13/23	61323
1/13/24	61324
1/13/25	61325
1/13/26	61326
1/13/27	61327
1/13/28	61328

   Assign the Circuit Descriptor COS, COR, and Tenant.

   If using a RS232-to-IP serial port converter, program it with the Port Number in Form 12. See RS232-to-IP Serial Port Converter Settings for other programming requirements.

3. Form 02, Feature Access Codes: Assign access codes as follows:

Feature Access Codes	
10 Do Not Disturb	
32 Automatic Wakeup	
35 Maid in Room	

4. Form 04, System Options: program as follows:

System Options / Timers	Status
02 Message Lamp Test	ENABLE

System Options / Timers	Status
04 Message Waiting and Message Register Clear Print	ENABLE
11 Automatic Wakeup	ENABLE
13 Automatic Wakeup Print	ENABLE
32 Outgoing Call Restriction To allow all rooms emergency 911 access this option should be disabled.	ENABLE
27 Room Status Audit	DISABLE
33 Room Status	DISABLE
34 Auto Room Status Conversion / Wake Up Print	DISABLE
45 Disable PMS Logs	DISABLE
108 Property Management System	ENABLE

5. Form 03, Class of Service Define: program the appropriate device classes of service as follows:

Device	COS Option	Status
Attendant Consoles	101 Attendant Outgoing Restriction / Room Status Set Up	ENABLE
105 Attendant Guest Room Key	ENABLE	
Room Telephones	202 Alarm Call	ENABLE
220 Do Not Disturb	ENABLE	
232 Message Waiting Setup - Lamp	ENABLE	
322 Confirm Wakeup by Offhook	ENABLE	
703 Message Register Applies	ENABLE	
239 Priority Dial 0 (optional - see step 6)	ENABLE	
244 Room Status Applies	DISABLE	
608 Telephone - Status Display	DISABLE	

6. Form 19, Call Rerouting: program “Station Dial 0" routing
   - The “Station Dial 0" answer point used for all tenants must be a Console LDN or Subattendant LDN. To allow the customer’s PMS to set message waiting, you must program Day, Night 1, and Night 2 answer points to the same LDN

[Tenant _____] Type Of Call	Day	N1	N2
Station Dial 0 Routing	Same LDN	Same LDN	Same LDN

- If you require different “Dial 0" answer points for Day, N1, and N2 use “Priority Dial 0" as an alternate method.

7. Form 34, Directed I/O: program the PMS

EXT NUM	PRINTOUT	PRINTOUT TYPE	GUARANTEED
From Form 12	PMS	AUTOPRINT	NO

Note: When the Printout type of an IP socket extension is changed, the Telnet session needs to be disconnected, and then reconnected for to restore functionality.

8. Depending on how the PMS computer is connected to the network, either

   Start a Telnet session on the PMS computer to the IP address of the SX-200 ICP and the socket port number programmed in Form 12.

   Or

   Connect the PMS computer to the Layer 2 switch using the RS232-to-IP serial port converter. Then, program the converter with the settings

Setting	**Value**
Protocol	Telnet (cln)
Port Setting	9600 bps 8n1 [no]
Connection control	Net-Linked
Terminal Type	vt100
Local Port	0
Remote IP	SX-200 RTC IP Address
Remote Port	As programmed in CDE Form 12
Fallback IP	0.0.0.0
Fallback Port	0

9. Verify that the customer’s PMS computer communicates with the SX-200 ICP.

   For serially-connected PMS computers, you will see an exchange of ENQ (Enquire) and ACK (Acknowledge) commands at the customer’s PMS computer if it is communicating with the SX-200 ICP. If they are not communicating, see then test the PBXs PMS interface (refer to Testing The PBX’s PMS Interface).

Keyboard Commands

The PMS Keyboard Commands table below lists the keyboard commands for the PMS personal computer.

Table: PMS Keyboard Commands

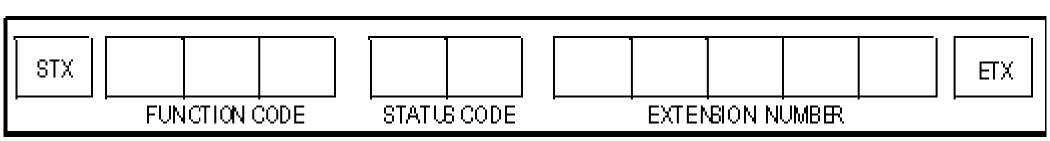
Commands	PMS Keyboard Keys	Comments
Enquire (ENQ)	^E	^ = CTRL key
Acknowledged (ACK)	^F	
Not Acknowledged (NAK)	^U	

Table: PMS Keyboard Commands

Commands	PMS Keyboard Keys	Comments
Start Transaction (STX)	^B	
End Transaction (ETX)	^C	
Check In	^BCHK1 nnnnn^C	nnnnn is the extension number. If the extension number is not five digits in length, use leading spaces in place of the digits.
Check Out	^BCHK0 nnnnn^C	
Add Name	^BNAM1 a-z nnnnn^C	a-z is the extension name. The extension name must be 20 characters in length followed by a single space, followed by the extension number. If the extension name is not 20 characters in length, use trailing spaces in place of the characters. Only the first 10 characters of the name are displayed.
Replace Name	^BNAM2 a-z nnnnn^C	
Delete Name	^BNAM3 a-z nnnnn^C	
Send Message	^BMW 1 nnnnn^C	Insert a space before and after the digit 1.
Delete Message	^BMW 0 nnnnn^C	Insert a space before and after the digit 0.
Set Wakeup	^BWKPTTTTnnnnn^C	TTTT is the time in hours:minutes. Use a 24-hour clock.
Cancel Wakeup	^BWKPSSSSnnnnn^C	The four S characters indicate the cancel function. The nnnnn are five digits for the extension number. If the extension number is not five digits in length, use leading spaces in place of the digits.

Messages

Messages have the following general format:



CC0872

Each box represents 1 byte. Each data message is prefaced with an STX, the ASCII character for start-of-text, and is followed with the character ETX, the ASCII character for end-of-text. The characters mentioned in the following messages are all in upper case unless otherwise stated.

Each message is preceded by an ENQ message which must receive an ACK response within 3 seconds. Then the STX - message - ETX is sent. Finally the recipient of the message responds with an ACK to indicate that the message has been successfully received.

Extension numbers of less than 5 digits in length are padded out using space characters (ASCII 32, HEX 20) not zeros (ASCII 48, HEX 30). For messages sent from the SX-200 PBX to the PMS, spaces are padded on the RIGHT side. For messages sent from the PMS to the SX-200 PBX, spaces are padded on the LEFT side.

Check In/Out Messages

The PBX enables the room phone when a guest checks in to the hotel, and disables the room phone when a guest checks out.

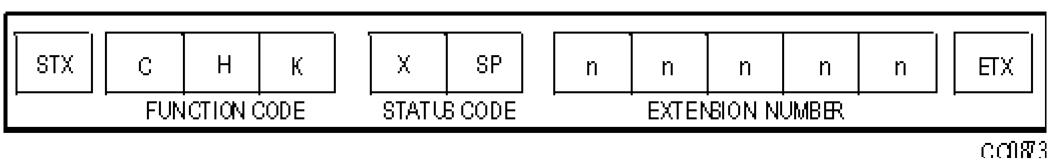
Check In operations include:

- changing the Call Restriction by following settings in Form 04, Option 58
- updating the Wakeup and Name fields according to information received from PMS.
- resetting the Message Registration to zero

Check Out operations include:

- deleting the name associated with the primary telephone
- resetting the Message Registration to zero
- setting the Call Restriction to the Check Out value defined in System Option 57
- clearing any pending Wakeup calls
- clearing Do Not Disturb, if set
- clearing the Message Waiting lamp

The Check In/Check Out Message has the following format:



where:

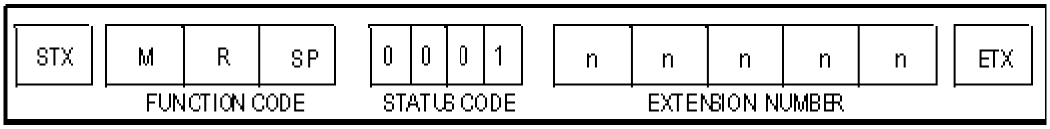
X = the Check In/Out status code:
ASCII character 1 for Check In
ASCII character 0 for Check Out
SP= ASCII space character
n = extension number digit

If the PMS is down, no check-in or check-out can be done, although the four PBX-related items can still be cleared one at a time from the attendant console if COS Option 105, Guest Room Key is enabled. If the PMS is up, but the message from the PMS is judged to be incorrect by the PBX, the PBX returns NAK and generates an appropriate maintenance log. Refer to Maintenance Logs for more details.

Message Registration Message

Each time a hotel extension makes a trunk call, the SX-200 system will send a message to the PMS to update the total count of outside calls made against the guest room. No distinction is made between local and long-distance calls.

Output Format:



CC0874

where:

SP= ASCII space character
n = extension number digit

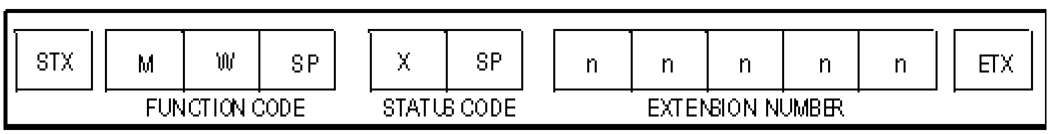
The status code in this case is a peg count. This is the only status code that is 4 bytes instead of 2 bytes in length. For North America, the status code is always 0001.

If the PBX is unable to establish proper communication with the PMS, the PBX will generate a maintenance log. Refer to Maintenance Logs for more details.

Message Waiting Message

This message format is similar to the check in/out message where a binary status code provides the new state of the message waiting lamp.

The Message Waiting Message has the following format:



CC0875

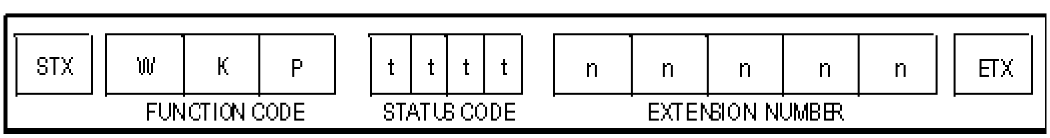
where:

SP= space character
X = the status of the message waiting lamp
ASCII character 1 for Lamp On
ASCII character 0 for Lamp Off
n = extension number digit

If the PMS is down, Message Waiting can be cleared from the console if COS option 320, Transparent Multi-Console operation, is enabled in the console's class of service.

Wakeup Messages

The PMS system can inform the PBX to set a wakeup call at a particular time for a specific guest extension using the following format:



CC0876

where:

t = the wakeup time
n = extension number digit

The wakeup time is specified in 24 hour time.

If the extension number has less than a five digits, the leading digits must be spaces.

To delete a wakeup message, replace the **t** character with an ASCI space for all four spaces.

If the PMS is down, the Automatic Wakeup can be cleared from the console if System Option 11, Automatic Wakeup, and COS Option 105, Attendant Guest Room key, are enabled for the console.

Name Message

The name message is sent from the PMS to program the PBX telephone directory with the guest's name.

Input Format:

CC0877

where:

NAM	= the name function code
x	= first byte of the status code, one of: 1 for addition 2 for replacement 3 for deletion
c	= a character of the name (maximum field size is 21 characters)
n	= extension number digit

The length of the name can be up to 20 characters. The name is left-justified, with spaces used for padding. The characters can be upper case, lower case, or numeric. In the name string, both a last and first name may be given. If only one name is input, then it is taken to be the last by default. The string sequence must be: last name or surname, comma, first name (the two names are separated by a comma). The comma may be anywhere except the first and 21st location. If only one name is given then the 21st character is a space.

Room Status Message

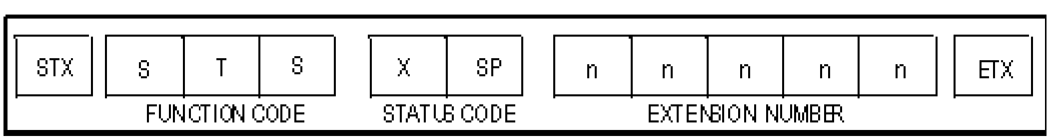
The maid dials an access code and a one-digit status code from the room telephone. When the status code is dialed, the PBX returns dial tone, translates the status code to ASCII and sends it as the first byte of the status code to the PMS.

The code dialed by the maid only changes the condition of the room as stored in the PBX's database; it never changes the occupancy. When a code is dialed, the PBX sends a function code and status code to the PMS, which may be interpreted as shown in the example that follows:

Message	Occupancy Condition
STS1	maid present
STS2	clean
STS3	not clean
STS4	out of service
STS5	to be inspected

If the PMS has additional room conditions, the maid can dial any room condition digit (0-9) but the PBX will only recognize digits 1-5.

The Room Status Message has the following format:



CC0878

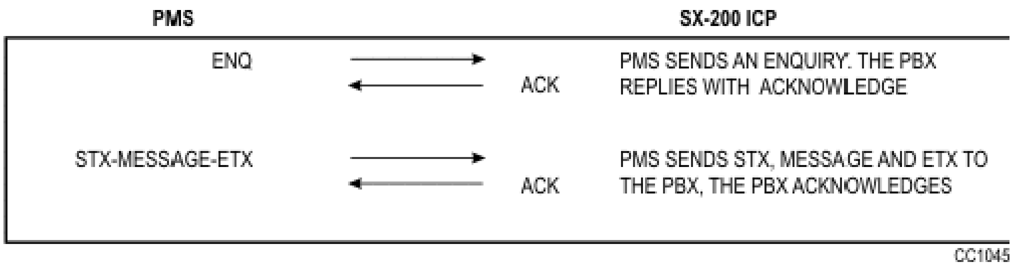
where:

- **X** = the room status code
- **SP** = ASCII space character
- **n** = extension number digit

The PBX sends off the transaction to the PMS (it cannot determine if the number dialed by the maid is incorrect). PMS will acknowledge the transaction to the PBX but will subsequently ignore it. If the PBX cannot get an acknowledgment from the PMS, the PBX will regard the PMS as out-of-order and will generate a maintenance log. Refer to Maintenance Logs for more details.

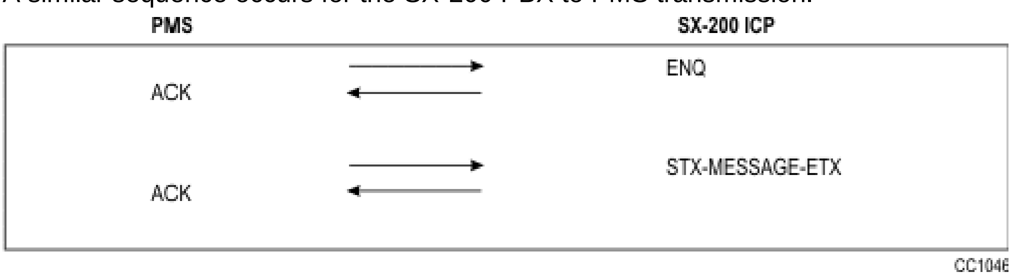
PMS/SX-200 ICP Communication Protocol

The PMS/SX-200 PBX link uses the ENQ /ACK/STX-text-ETX/ACK protocol. The link is bi-directional (though half-duplex) and is used for both SX-200 ICP to PMS and PMS to SX-200 ICP transmission. The PMS to SX-200 ICP transmission sequence is:

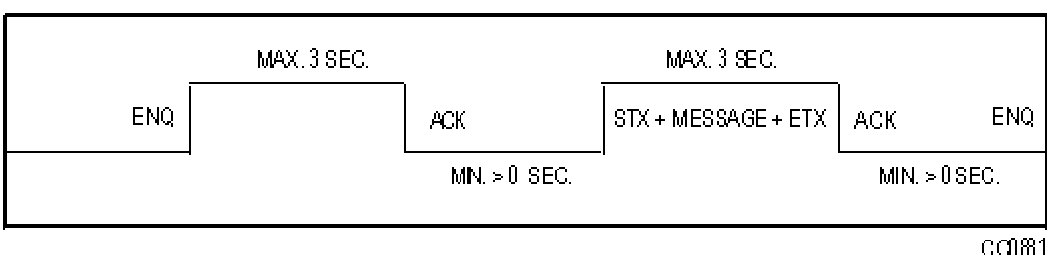


The transmission of the message is complete.

A similar sequence occurs for the SX-200 PBX to PMS transmission.



The timing restriction on transmission sequences is to wait a maximum time of 3 seconds for the ACK after an STX + message + ETX transmission.



PMS/PBX Communication Messages

The following communications messages are sent between the PMS and the SX-200 ICP.

Communications Messages

PMS to PBX:	**PBX to PMS:**
Are You There?	Request to Initialize
General Reset	Enquiry
End	Acknowledgment
Enquiry	Negative Acknowledgment
Acknowledgment	
Negative Acknowledgment	

Are You There?

STX	A	R	E	Y	U	T	H	E	R	E	ETX

CC0082

General Reset

STX	G	R	S	SP	SP	SP	SP	SP	SP	SP	ETX

CC0083

End (message used to indicate the end of the general reset process)

STX	E	N	D	SP	SP	SP	SP	SP	SP	SP	ETX

CC0084

Request To Initialize

STX	R	Q	I	N	Z	SP	SP	SP	SP	SP	ETX

CC0085

SX-200 PBX to PMS Transmission Error-Handling

PBX sends an ENQ character to the PMS and awaits a reply.

If an ACK (acknowledgment) is received within three seconds of an ENQ, the transmission continues.

> If no response or if a NAK (negative acknowledgment) character is received within the time-out period of 3 seconds, the ENQ is retried 19 times before abandoning the transmission of the message (a total of 20 times and an elapsed time of 1 minute). The PBX generates a log and an alarm to indicate that the PMS is down, and then attempts to re-establish communication by sending an ENQ every 3 seconds.
>
> When the PMS is able to respond with an ACK, the PBX will generate a log for 'PMS is up'.

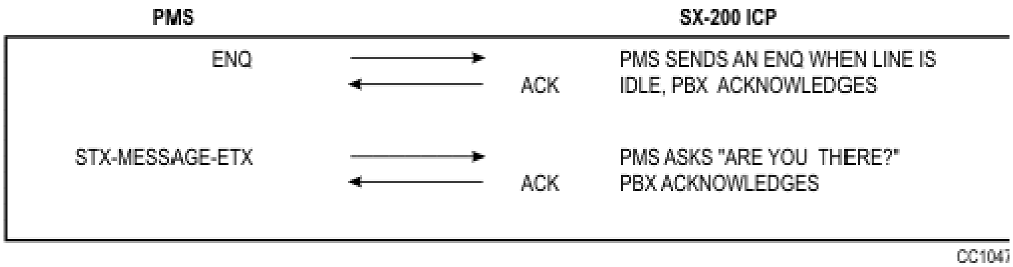
The PBX sends a STX (start-of-text) character, the message text, and an ETX (end-of-text) character to the PMS, and then awaits an ACK from the PMS.

If an ACK is received, transmission is completed successfully.

If no response or a NAK character is received within the time-out period (3 seconds) the entire sequence, beginning with ENQ, is retried 4 times (for a total of 5 attempts and an elapsed time of 15 seconds) before abandoning transmission. The PBX generates a log and an alarm to indicate that the PMS is down. The transaction is then given up and discarded.

Typical Idle State Transmission

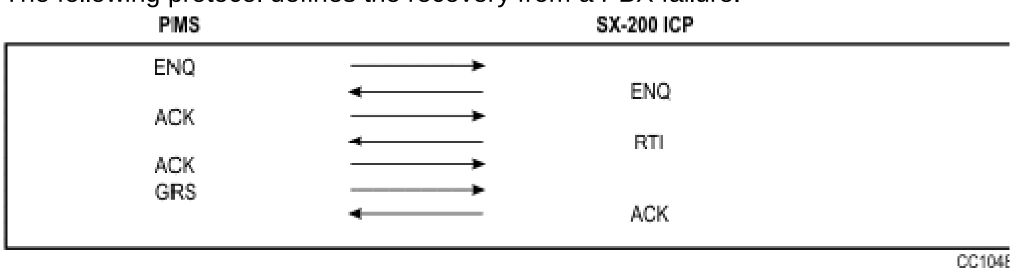
When the communication link is not busy between the two machines, the PMS will automatically send out an ENQ to the PBX every 10 seconds to ensure that the PBX is still up and functioning correctly. The PBX is expected to reply within 3 seconds. If the PBX replies with an ACK, the PMS sends an ‘Are You There?’ message. The PBX is expected to ACK within three seconds. For example:



Recovery From SX-200 PBX Failure

If the PBX fails to acknowledge, the PMS will continue to send an ENQ every three seconds until it receives a reply from the PBX.

The following protocol defines the recovery from a PBX failure.



At this point, the PMS accesses its database and for each room sends the PBX a message that the room is either checked in or checked out.

Recovery From PMS Failure

When the PMS is down, the PMS terminal cannot be informed of changes in Room Status or Maid in Room status.

The PBX normally keeps a transaction buffer with up to 50 transactions. A transaction may be either Message Registration or Room Status messages.

While the PMS is down, the PBX will:

- create a maintenance log indicating the PMS is down
- keep track of the number of trunk calls made by a hotel extension while the PMS is down
- maintain only the Message Registration transactions left in the transaction buffer before the PBX determined the PMS was down

When the PMS system returns to service, the PBX will create a maintenance log indicating PMS has returned to service. It will also use its database to update the PMS system database with Message Registration and Room Status messages since the PMS was down.

When the PBX and PMS Cannot Communicate

If the PBX's PMS interface and the customer's PMS computer cannot communicate, the system will generate PMS related maintenance log messages. Typically, PMS maintenance log messages will identify:

- operating status of the PMS
- invalid **STX** (start of text) and **ETX** (end of text) message characters
- invalid **function** code message characters
- invalid **status** code message characters
- invalid **room number** message characters.

Testing The PBX's PMS Interface

PMS problems typically occur in these areas:

- **PMS Protocol settings:** Ensure the communication parameters between systems match
- **PMS Programming:** Verify that all recommended PMS and RS232-to-IP serial port converter (if using) programming is complete
- **PMS Cabling:** Use a straight-through cable between the customer's PMS computer and the RS232-to-IP serial port converter (if using).

Test the PBX's PMS interface as follows:

1. Disconnect the customer's PMS computer from the RS232-to-IP serial port converter.
2. Connect a personal computer or terminal to the converter RS-232 port.
    - Ensure that the communications parameters of the personal computer / terminal match those expected by the PBX's PMS interface
    - Verify that you receive the **ENQ**uire character from the PBX's PMS interface. If not, verify that all programming and connections are correct.
3. Type the PMS message required to light a telephone's message waiting lamp, using CAPITAL letters only.

    ^BMW 1 nnnnn^C (space required before and after the digit 1)
4. Verify that the telephone's message waiting lamp is lit.
    - If the lamp is not lit, check the PMS-related logs in maintenance. The logs will indicate which section of the PMS message is invalid.
5. Type the PMS message required to turn off he telephone's message waiting lamp.

    ^BMW 0 nnnnn^C (space required before and after the digit 0)
6. Verify that the telephone's message waiting lamp is off.

Turning a telephone's message waiting lamp on and off from a personal computer confirms that the PBX's PMS interface works correctly.

If problems continue when you reconnect the customer's PMS computer, check the maintenance logs for PMS related information which customers may need to identify problems with their PMS hardware/software.

Maintenance Logs

The logs generated by PMS fall into the Alarm Fault category. The template for this type of log is two lines as follows:

MMM-DD-YY HH:MM:SS   NIL PLID failed at 00 00 00 00
Actual Message Goes Here   Alarm Code=XXX

The only useful information from the first line is the date and time. Ignore the rest of the first line and the alarm code on the second line (it is just an error message number).

If the message sent from the PMS to the PBX is judged to be incorrect, the PBX will automatically generate a maintenance log and send a NAK signal back to the PMS. Possible maintenance logs for invalid messages are:

No STX from PMS
No ETX from PMS
Bad PMS function
Bad PMS status
Bad PMS room number
PMS requires Console
Bad PMS time

Set System Option 45 to disable PMS logs to prevent filling the logs buffer with PMS logs.

Communication Problems

In addition to the seven logs mentioned above, there are also four logs related to communication problems.

If the PMS refuses to accept a transaction after five tries, the PBX will generate the following log:

Cannot send PMS message

When the PBX receives no reply after one full minute of enquiry, it will generate the following log:

PMS is down

While the PMS is down, the PBX can still accept up to 50 transactions. At this point, the circular buffer of transactions is full so the PBX will generate the following log:

PMS buffer is full

If the PMS remains down, all subsequent transactions will be lost. When the PMS recovers and sends an ACK signal, the PBX will generate the following log:

PMS is up

As the PMS acknowledges receipt of the oldest transaction, the PBX frees up the corresponding slot in the circular buffer. This slot is used to hold the next transaction which arrives at the PMS Interface from Call Processing.

ACD TELEMARKETER

Introduction

The ACD TELEMARKETER® Applications Package is a software option that must be purchased. The following conventions are used in the description of this software option:

Key Names

Names of keys on a keyboard are shown in bold, within brackets.

Example: <**Return>**

Softkey Names

Names of softkeys used with monitors or to program are shown in uppercase.

Example: GRP_SUMMARY

Data Entry

Data to be entered by the user is shown in bold uppercase.

Example: **EXIT**

Data Entry Instructions

The following conventions have been used throughout this manual when presenting data entry instructions to the user:

Type: Type the information (usually a single key) as shown without pressing the Return key.

Example: Type **A**

Enter: Type the information as shown and press the **<Return>** key.

Example: Enter **HOTEL**

Press: Press the indicated key(s) with no return.

Example: Press <**CTRL> <C>**

ACD Overview

This section gives a general overview of Automatic Call Distribution (ACD) and describes the basic components found in ACD systems. If you are familiar with ACD concepts, see ACD TELEMARKETER Feature for details on the implementation of the ACD TELEMARKETER feature for the SX-200 ICP.

Automatic Call Distribution

Automatic Call Distribution (ACD) offers uniform distribution of incoming calls to station users (agents). Calls are routed to groups of agents as determined by the type of information or service required by the caller. The agents are trained and equipped to provide the particular information or service that the caller is requesting. If calls cannot be handled immediately, the caller is usually provided with recorded announcements and/or music until an agent is available.

Most ACD systems generate one or more reports listing call handling statistics and ACD traffic levels. The system administrator uses these reports when determining optimum staffing levels, acceptable caller delay times, and use of system resources.

ACD Applications

Typical ACD applications include airline reservation offices, telephone order desks for department stores, and customer service departments of telephone or cable TV companies. In all cases, the caller is attempting to reach an individual who can supply a service, answer a question, take a reservation, or accept a purchase order.

Call Queuing

To ensure optimum use of personnel and system resources, historical calling patterns are often used to determine staffing levels for the agents. Most ACD installations set staffing at levels which ensure that the average number of callers equals or exceeds the number of agents. During peak periods when all agents are busy, callers are placed in a queue to wait for the first available agent.

While a caller is waiting in the queue, the ACD system can provide recorded announcements and music at predetermined intervals. The first recording typically advises the caller that all agents are busy, and that an agent will answer as soon as possible. If an agent is unavailable after a programmed interval, additional recordings can inform the caller about call progress, or advise the caller of information that will be required when the agent answers.

If calls arrive when some of the agents are free, the system may be programmed to equalize the work load by directing the next incoming call to the agent who has been idle the longest.

Staffing an ACD System

In most ACD applications, the person handling ACD calls is referred to as an agent. Agents are trained to deal with the caller's problems or requests.

A supervisor normally oversees the ACD operation by monitoring the activity of the agents, reassigning agents to handle overload conditions, and dealing with unusual situations.

Reporting

ACD systems normally provide a reporting mechanism that allows tracking of key items such as the number of calls handled during a specific time period, the length of the calls, and the number of calls abandoned (caller hangs up before an agent answers). From these reports, the supervisor can determine optimum staffing levels and track the performance of individual agents.

Monitoring

While reports give a hard copy record of events over a period of time (such as a shift), a monitor gives a snapshot of conditions in the system at any instant. By monitoring the ACD system, the supervisor is aware of the current situation and can quickly reassign agents to handle overload conditions.

ACD TELEMARKETER Feature

The ACD TELEMARKETER Application Package is an advanced Automatic Call Distribution (ACD) system that is fully integrated with the Mitel SX-200 system, and designed with the power and performance needed to ensure satisfaction in the most demanding telemarketing environments.

This section provides information on the ACD TELEMARKETER system components:

- **ACD Path -** This innovative call routing design guides incoming calls through the system. The ACD path defines all information required for each type of call, including how the system will handle queued callers.
- **ACD Call Flow -** Describes the handling of a typical ACD call arriving at the system.
- **ACD Sets -** Mitel 5010 IP, 5207 IP, 5215 IP, 5020 IP, 5220 IP, 5224 IP, 5324 IP SUPERSET 4025, SUPERSET 4125, SUPERSET 4150 ,SUPERSET 430 and SUPERSET 420 telephones may be used in the senior supervisor, supervisor, or agent positions with the ACD TELEMARKETER feature package. Mitel 5212 IP, 5215 IP, 5010 IP, Symbol MiNET Wireless, SUPERSET 4015, and SUPERSET 410 telephones may be used in the agent position only.
- **ACD Positions -** The ACD TELEMARKETER feature package structures the personnel handling ACD calls into a hierarchy of ACD positions. The ACD package supports three types of positions: senior supervisors, supervisors, and agents.
- **Recorded Announcements -** The ACD TELEMARKETER feature uses recorded announcements to tell callers about the progress of their call while they wait in the queue for the first available agent.

ACD Path

The ACD TELEMARKETER feature is built around the “ACD Path”, a call routing mechanism which provides all information required for handling an ACD call. Use of the ACD Path gives users unmatched flexibility during initial programming and when they add new features.

Up to 99 ACD paths may be programmed to allow customized routing for a wide variety of incoming calls. Each path is assigned a priority and is given a unique access code and descriptive path name (optional). This information determines how the system handles queued callers, system resources to be used, when the call will be answered, and who will answer the call.

Upon entering the ACD system, a call is allocated a path and is assigned the parameters of that path. These parameters remain with the call for its duration.

Path Access Code

Incoming trunks carrying ACD calls are routed to a path access code. This code points the trunk to the ACD agent groups and recorded announcements appropriate for the type of call.

Path Priority

Each path is assigned a priority level in the range 1 to 99, with 1 being the highest priority. Calls arriving on high priority paths move directly to the front of the call queues for servicing ahead of calls which entered on a lower priority path.

Path priority can be an effective tool for reducing communications costs and improving customer service. For example, call queue time can be reduced by directing expensive incoming trunks, such as long distance collect or INWATS, to a high priority path. Customers can be assured of prompt service if their calls are routed through a high priority path.

Path Access

All devices have unrestricted access to ACD Paths except loop start CO and loop start DISA (Direct Inward System Access) trunks. The Class of Service (COS) Option “Loop start trunk to ACD path connect” (COS Option 812) controls ACD access for loop start trunks. By default this option is disabled, blocking loop start trunks from entering ACD.

Path Rerouting

The ACD path access code can be placed in the call rerouting table to link existing routing schemes (such as DID trunk routing points) to the ACD system. Rerouting to ACD paths is set up as follows:

- For dial-in trunks, the system uses the current routing (defined in CDE Form 19) for incoming calls to send calls to an ACD path.
- For non-dial-in trunks, one or all of the Day/Night1/Night2 answering points is programmed as an ACD path.

Calls entering the system on different trunk types can be routed to the same ACD path.

The rerouting scheme means that a trunk does not have to be dedicated to ACD. The day answering point may be an ACD path but the Night1 and Night2 answering points may be an attendant console or any other valid routing point.

Service Level

The service level for a path defines a standard time to answer that becomes the criteria for measuring path performance. Service level is programmable within the range from 0 seconds to 54 minutes.

When an ACD call is answered by any group in a path, software compares the actual time to answer with the programmed Service Level. The system creates a record indicating if the time to answer was:

- less than or equal to the service level time
- greater than the service level time.

This information is stored for statistical analysis and can be viewed from the ACD Path Monitors and Group/Path Summary Reports.

Overflow

Higher priority paths are given special treatment when placed in overflow queues. Predictive overflow is another key element of the ACD TELEMARKETER feature. The system uses overflow queues to keep call queueing time to a minimum. The system performs a load calculation when each new call arrives at an agent group or when the status of an agent changes. If the system predicts that a call will not be answered before the normal overflow time, it forces an immediate overflow.

Priority calls entering an overflow queue are placed ahead of non-priority calls in the same queue. The non-priority calls maintain their position in relation to each other, but follow the priority calls.

Each path is assigned one primary agent group and up to three overflow groups. Timers programmed in CDE for each agent group determine how long a call waits on a group before overflowing. If the system predicts that a call will not be answered before the timer expires, the system forces an immediate overflow without waiting for the timer to expire.

Interflow

Unlike overflowed calls, interflowed calls are rerouted from ACD to an alternate answer point. Each path has a programmable interflow timeout field that specifies the maximum period that an unanswered call can wait in a path before the system routes the call to an interflow point. The interflow point can be:

- a listed directory number, station, console, night bell, ACD path, station/set hunt group, Automated Attendant group, or UCD agent group
- a logical line
- a system abbreviated dial number.

To limit the time that a caller remains waiting for an agent, the system can also be programmed to drop interflowed calls.

The system can be programmed so that any callers dialing in to an ACD path will interflow immediately when no agents are logged in. This interflow takes place regardless of the status of the “Interflow Timeout,” or the option “Allow Overflow to Interflow Point Before Timeout,” or the “Interflow Point Access code” having a value of DROP CALL. “Interflow Enabled” is required for the immediate interflow to work.

A tenant number can be assigned to the ACD Path. When a tenant number is assigned, then DID and TIE trunks which dial into the ACD Path directly will follow the routing for this tenant as defined in Form 19 - Call Rerouting Table.

In addition, the customer can program the system to overflow to the interflow point as soon as the system determines that the call is unlikely to be answered by the last agent group in the path.

Calls interflowed to a system abbreviated dial number are treated by the system as an external call forward with the interflow requiring a receiver. If a receiver is unavailable when the interflow occurs, the call is dropped. The only indication of the dropped call is a receiver unavailable peg in the traffic report. The call appears in the ACD reports as an interflowed call.

If Automatic Route Selection (ARS) is busy when a call interflows to a system abbreviated dial, the system attempts a camp-on to ARS.

The path from which the caller interflows is set up as the original forwarding destination for the caller. When the interflow point is an internal device, such as a SUPERSET 420 telephone, the display indicates that the call is being forwarded from a path.

Music Between Recordings

Between each recording on an ACD path, the incoming caller, by default, listens to the system music source (if programmed). An alternate music source may be specified between each recording and after the last recording.

Alternate Music Source

The alternate music source is an off hook ONS port that connects callers to a listen-only conference. The customer decides what is supplied on the ONS port - music or endless loop recordings. If there is neither system music nor an alternate music source, then callers hear silence between RAD messages.

The audio playback device must simulate an off hook and allow connection to an ONS port (an impedance matching transformer (8/600 ohms) may be required). There are no restrictions on how paths share alternate music sources.

Once the device is programmed as an ONS port in Form 09 and then as an alternate music source in Form 41, it may be necessary to disconnect and reconnect the device to simulate on-hook / off-hook so that the system initially recognizes it.

Note: Depending upon country of installation, the alternate music source must be either an FCC Part 68 or Industry Canada approved voice coupler, or voice connecting arrangement to an ONS circuit.

ACD Call Flow

The following paragraphs describe the handling of a typical ACD call arriving at the system on an incoming trunk. Included is a description of what the caller hears at each stage of the call.This Figure shows the system action when determining what the caller hears while in the queue.

ACD Caller

The system considers an ACD caller to be anyone who is on, or has been answered by, an ACD path. Once answered, the ACD caller status remains while the caller is talking to an agent, on hold by an agent, or in the process of being transferred by an agent.

After an ACD caller has been answered by an agent, the SX-200 ICP reverts to normal call handling but provides additional tracking for ACD session timing, ACD hard-hold timing, and the caller's identification as an ACD caller. When an ACD caller reaches the console through either a supervised or unsupervised transfer, the ACD caller status ends. If, during a supervised transfer, the person performing the transfer remains on the line after the console answers, the ACD session is terminated.

Typical Call Handling

If multiple agents are free when an ACD call is presented to a group, the system sends the call to the longest idle agent. To select the longest idle agent, the system gives a number to the first agent finishing an ACD call. The next agent to finish an ACD call is given the next higher number, and so on. When a call arrives at the group, the system sends the call to the agent with the lowest number. The number does not change if the agent makes a non-ACD call.

Figure: ACD Call Progress - All Agents Busy

1. ACD Call arrives at the path specified as the answer point for the trunk.
   - As shown in this Figure, the caller hears ringback until the Ringback Delay timer expires. This timer ensures that the caller hears at least one ringback before an agent answers.
2. The incoming call queues on the primary agent group for the path.
   - If an agent is available, the call rings an agent (See Note).
   - If multiple agents are available, the call rings the longest idle agent; if not, the caller waits for first recorded announcement.

Note: Once the agent set begins to ring, the call must be answered. If the called agent fails to answer within the period programmed for the Forward Timer in the agent's COS, the system forces the agent's set into Make Busy and routes the call to another agent in the group. This operation is transparent to the caller.

3. The system connects caller to the first available RAD in the first recording group defined for the path.
   - The caller listens to first recording. Call remains queued on first agent group.
4. When the RAD message ends, system connects call to music-on-hold (MOH) source or to first alternate source as defined for the path.
   - The caller hears music or alternate source.
5. After time interval programmed in the path for Recording 2 starts, system connects call to first available RAD in second recording group defined for the path.
   - The caller listens to second recording. The call remains queued on first agent group.
6. The system connects call to MOH source or to second alternate source as defined for the path.
   - The caller hears music or alternate source.
   - The caller continues listening to music and recorded announcements until an agent is available. Timing is set in CDE.
   - Up to four recordings can be programmed for each path.
7. If the call remains queued against the first agent group for a period exceeding the overflow time programmed for the group, the system adds the first overflow group defined for the path.

- The caller is now queued on two groups.
- The caller continues listening to music and recorded announcements until an agent is available.
- The caller retains position in queue for primary agent group.
- Path priority determines position of call in overflow group.

The system can add up to three overflow groups if a call remains unanswered. Overflow times are programmed individually for each agent group. This Figure shows how overflow groups are added as the caller waits in the queue.

8. The system performs a load calculation when each new call arrives at an agent group, or when the status of an agent changes. If the system predicts that a call will not be answered before the overflow timer expires, the system forces an immediate overflow. This predictive overflow is always enabled.
9. As shown in this Figure, if the call remains unanswered for a period exceeding the Interflow Timeout programmed for the path, the system routes the call to the interflow point which can be an internal or external destination. The call is handled as a call reroute.

Figure: Overflow/Interflow

ACD Sets

The Mitel 5010 IP, 5212 IP, 5312 IP, 5215 IP, 5020 IP, 5220 IP, 5224 IP, 5324 IP, SUPERSET 4015, SUPERSET 410, SUPERSET 4025, SUPERSET 4125, and SUPERSET 420 telephones used by ACD positions provide call status and progress information about agent groups and individual agents. A supervisor's set (Mitel 5220 IP, 5020 IP, 5224 IP, 5324 IP, SUPERSET 4025, SUPERSET 4125, SUPERSET 4150, SUPERSET 430 and SUPERSET 420) provides agent reports and queue status reports for the supervisor's agent groups. An agent's set (Mitel 5010 IP, 5212 IP, 5215 IP, SUPERSET 4015, SUPERSET 4125, SUPERSET 4150, SUPERSET 430, SUPERSET 410, SUPERSET 4025 or SUPERSET 420) provides queue information for the agent's group.

Feature Availability

Click here to see a list of the phones that can use this feature.

Mitel 5324 IP, 5312 IP, 5220 IP, 5224 IP, 5212 IP, 5215 IP, SUPERSET 4025, SUPERSET 4125, SUPERSET 4150, SUPERSET 410, SUPERSET 420, SUPERSET 430 telephones used with the ACD TELEMARKETER Feature Package offer:

- Single key feature activation
- Supervisor monitoring of agent calls with or without agent notification
- Agent help requests during a conversation - transparent to ACD callers (not applicable to SUPERSET 4015, SUPERSET 410 telephones and Mitel 5207 IP, 5212 IP and 5215 IP telephones)
- Handset/handsfree/headset operation. (COS Option number 612 must be enabled in the user's COS prior to operation. The telephone handset should remain off-hook when headsets are in use.)
- LCDs load status information
- Auto answer
- Time and date display (Mitel 5220 IP, 5224 IP, 5324 IP, 5312 IP, 5212 IP, 5215 IP, 5207 IP, 5020 IP, 5010 IP, SUPERSET 4015, SUPERSET 4025, SUPERSET 4125, SUPERSET 4150, SUPERSET 430 and SUPERSET 420 telephones)
- Path name is displayed when calls are presented to the agent. (Not applicable to SUPERSET 410 telephones.) When COS Option 654 - ACD Display Path Always is enabled, the path name remains on the display for the duration of the call.
- Make Busy keys to temporarily block ACD calls from agents sets
- A programmable name for every ACD entity: paths, RADs, groups, agents and supervisors
- A programmed set of speedcall keys (via CDE programming).

NOTE: Mitel 5224 IP phones have 24 programmable keys but keys 16 to 24 are not supported for ACD.

Automatic Number Identification (ANI) and Dialed Number Identification Service (DNIS) are available through COS options programmed during Customer Data Entry (CDE). ANI provides the telephone number of the calling party; DNIS provides the telephone number dialed by the calling party. Refer to the Program Features section for details.

Conditions:

- Applying a Guest Room template to 53xx series phones will not override keys that have been programmed using the Superkey or Applications key.

ACD Positions

The ACD TELEMARKETER feature package supports three types of positions: senior supervisors, supervisors, and agents. This Figure shows an example of the ACD hierarchy.

ACD calls entering the system normally terminate on agent positions (SUPERSET 410, SUPERSET 420, SUPERSET 430, SUPERSET 4015, SUPERSET 4150, Mitel 5212 IP, 5215 IP, 5312 IP, 5010 IP, 5207 IP telephones). Agents handling similar types of calls are arranged in agent groups. Supervisors and senior supervisors (SUPERSET 420, SUPERSET 430, SUPERSET 4150, SUPERSET 4025, SUPERSET 4125, Mitel 5220 IP, 5020, 5224, 5324 IP telephones) monitor agent and system performance, but do not handle ACD calls.

As shown in this Figure,every supervisor, senior supervisor, and agent has an ID number. This is a 1- to 5-digit number assigned during customer data entry. Before they can receive ACD calls, the agent or supervisor must log in to the system by dialing an access code followed by the appropriate ID number. Agent groups and the three ACD positions are described below.

The SX-200 ICP treats the ID number assigned to each position as an access code. This number can be directly dialed by other devices in the system as normal extension numbers.

Note: An ACD agent is considered not available for an ACD call if the agent is on its prime key or any other line key appearing at the telephone.

Figure: Hierarchy of ACD Positions

Agent Group

Agent groups are included with the ACD Agents option that you purchase. An agent group consists of one or more persons, called agents, that handle incoming ACD traffic. Each agent group must contain at least one member; the maximum number of agents in a group is 99. The ACD system accepts up to 50 agent groups.

As shown in this Figure, each agent group must be set up to report to either a supervisor or a senior supervisor (never both). Supervision requirements are determined by the customer and are usually dictated by the size of the group.

Agent groups are created through Customer Data Entry (CDE) by entering in the ACD agent groups form (Form 39) an agent group number in the range from 1 through 50. An optional name can also be given to the group to assist in identifying the group's function. Members are added to the group though CDE by entering a 1- to 5-digit ID number and an optional name for each agent.

Each agent group has timers that govern events such as:

- the time granted to an agent for completing paperwork after an ACD call
- the length of time a call will remain unanswered in the group before overflowing
- turning on and off visual indicators which show that calls have been unanswered for a time exceeding a programmed threshold level.

Refer to programming in this section for details about creating agent groups and the fields on the agent groups form.

Agent

The ACD TELEMARKETER feature terminates ACD calls at agent positions. In most ACD installations, all callers routed to an agent are requesting similar information or seeking a similar service. Agents can then be trained and equipped to provide the information or service requested by the caller.

The system routes calls to an agent only after the agent logs in to the ACD system. Once the agent has logged in, the system recognizes the agent as a member of a specific agent group.

The Mitel telephones used by the agent are each equipped with a feature key to temporarily block ACD calls from ringing the set. Other feature keys provide information about the current status of the agent group.

In many cases, an agent must be given the flexibility of moving between agent groups. If, for example, the ACD agent group handling long distance trunks is suddenly overloaded and calls are overflowing to an alternate group, significant financial gain could result by reassigning Agents to the busy groups until the traffic subsides.

Because the ACD system uses the ID number to determine the members of an agent group, providing the agents more than one ID number allows the agents to be members of more than one group. To move between groups, the agent logs in by using the ID appropriate to the group. Only the ID number must be unique; an agent name can appear in any number of groups.

Supervisor

The supervisor ACD position is for individuals who “supervise” the agent groups. Each supervisor is responsible for at least one agent group and reports to a senior supervisor. Supervisors do not answer ACD calls. The Mitel telephone used by the supervisor is equipped with special feature keys that allow the supervisor to view individual agent activities or to view agent group activities .

Senior Supervisor

The senior supervisor oversees the supervisors and is, therefore, the highest level in the hierarchy of ACD positions. In smaller installations, however, where a supervisor is not required between the agent group and the senior supervisor, agent groups may report directly to the senior supervisor. The senior supervisor does not answer ACD calls. The Mitel telephone used by the senior supervisor is equipped with special feature keys, similar to the supervisor set, with emphasis on queue activity.

Recorded Announcements

A recorded announcement device (RAD) is a digital or endless-loop tape unit that can store one or more pre-recorded messages. The required RADs are designed for connection to ONS circuits and appear as a standard telephone to the SX-200 ICP. The RAD’s message is played when the unit is triggered by ringing current. In the ACD TELEMARKETER feature, the recorded messages are given while callers are waiting in the queue for a free agent.

The SX-200 ICP supports both intelligent and dumb RADs. An intelligent device hangs up when the message is finished. A dumb device provides a fixed-length recording (such as a tape) and the system must hang up on the device to prevent callers from listening to a long period of silence at the end of the message.

Recording Groups

The recorded announcement feature is implemented by using one or more RADs programmed into a specialized hunt group called a recording group. Each RAD in the group contains the same announcement.

Recording groups are formed by using hunt groups of regular ONS ports and are defined during CDE in Form 17, Hunt Groups. Refer to the programming section for details.

Recording Group Operation

When a call rings a recording group, the first available idle RAD answers the call and connects its recording. If all RADs in a recording group are busy, the caller camps onto the group to wait for a free recording. When a recording becomes available, the system connects all waiting callers to a listen-only conference with the recording. (The listen-only conference does not use any SX-200 ICP conference resources.) When the recording finished, the callers are removed from the conference and are connected to music or silence as defined in the ACD Path programming form (Form 41).

Notes:

1. The system does not use a special CODEC gain setting for listening to a recording. The gain is the same as for ringback, or set to no gain.
2. A RAD is always rung with the standard ringing cadence.
3. Callers are never connected after the RAD starts its message.

RAD Failure

The ACD system handles four types of RAD failures:

Failure to Answer: The system considers the RAD to have failed if it does not answer within the fixed interval of 30 seconds. The system clears ringing, puts the RAD into Do Not Disturb (DND), turns on the console alarm icon, and creates a maintenance log entry as shown in the following example:

```
1997-JAN-15 12:47:04    Recording dev test    failed at  01 01 01 00
                        Failure to answer     Alarm code = 123
```

Failure to Hang Up: The system detects failure to hang up when the system ends the recording. The hangup time is set by COS option 404 - Recording Failure to Hangup Timer, which has a range of 1 through 255 seconds. The timer starts after the SX-200 ICP system hangs up on the RAD. The RAD must clear down within the programmed interval. Otherwise, the system puts the RAD into DND, turns on the console alarm icon, and creates an entry in the maintenance log as shown in the following example:

```
1997-JAN-15 12:47:04    Recording dev test    failed at  01 01 01 00
                        Fail to hang-up.  Alarm code = 123
```

False Origination: If a RAD generates a false origination, the system puts the RAD into a suspended state. After the suspended timer expires, the RAD is placed into a lockout state. If the RAD goes on hook while in either suspended state or lockout state, the RAD is returned to idle and is immediately available to the system.

Card Failure: If the system detects a card failure, such as the card being unplugged or the bay going down, the RAD is placed into a busy-out state. Any callers listening to the RAD are handled as though the RAD had gone on hook. The RAD is not placed in DND unless it was ringing at the time (which is treated as a ring-no-answer). When returned to service, the RAD is in idle state.

Removing DND from RAD

The system places a RAD into DND whenever the RAD fails to answer or fails to hang up. In systems with the embedded voice mail application (CX/CXi and MX controllers), DND can be cleared from the maintenance terminal by using the Clear Features key. In all other systems, DND can be removed from a RAD by accessing the attendant console stations feature, dialing the RAD, and pressing the DND softkey.

Removing DND from a RAD from the maintenance terminal generates the following maintenance log entry:
2004-DEC-06 14:15:51 VOICE MAIL CARD passed at 01 11 07 01 Ext 307
Recording dev test Alarm Code = 123

Removing DND from a RAD from the attendant console generates the following maintenance log entry:
2005-JAN-15 12:47:04 Ons Card passed at 01 01 01 00 ext 1101
Recording device test Alarm code 123

Configuring an ACD System

The communications manager planning the installation of an ACD TELEMARKETER system may find that the information in this section can help in determining the final system configuration. Because thorough planning can ensure maximum performance from the ACD system, the following guidelines have been developed to help customers define their requirements.

Incoming Calls

Because the most critical element of an ACD system is the timely handling of incoming calls, the communications manager must first consider the level of traffic that the system will receive and then determine the types of calls and any trunking details that could influence the importance of the call. For example, long distance charges can be kept to a minimum by assigning these calls to high priority paths. The following questions serve as examples of those areas to be addressed when categorizing the ACD callers:

- Are service departments involved?
- What traffic is anticipated for each department?
- What priority is given to service calls?
- Do any service departments require a customer complaint area?
- Is there local service only or service also to out-of-town clients?
- Will there be revenue generating calls? Unless the company holds a monopoly on service, these callers should be highlighted for priority paths. Are there general information calls?
- Will agents receive long distance calls....collect?
- Does the company offer INWATS, foreign exchange, or any specialized trunking?

Grouping the Agents

Using the caller information that was collected, begin grouping the agents. Use the following questions and comments as a guide:

- Are any agents capable of handling various types of calls? For example, will any agents be common to more than one service department?
- Which type of calls will this group specialize in? List the types of calls this group could handle as an overflow point.
- Which groups will require a wrap up time?

Recorded Announcement Planning

Used properly, recorded announcements are a valuable tool in the ACD TELEMARKETER system. The following suggestions can help you gain the most from the recordings:

- Supply a company introduction to the caller. “Thank you for calling .....”, followed by reassurance that the first available agent will answer the call.
- Consider the advertising potential while the caller is waiting for service. Use the recorded announcements to promote new products, specials, or services.
- Refer to the list of callers as a guide when defining the RADs. Because out-of-town callers would be frustrated by local promotions, adjust your recorded message to the caller’s needs.

- Is there any information that the agent will require from the caller? Recordings can be used to minimize time with an agent if callers have prior notice of information they should have ready, such as account numbers, credit cards, or postal codes.

In addition, the Automated Attendant feature can be used to pre-screen ACD calls into the system. Refer to the Automated Attendant Application Package section.

Planner Sheets

The agent group planner and the path planner sheets illustrated on the following pages can aid the ACD system designer when laying out the agent groups and the routing for incoming ACD calls. The planner sheets identify all major elements that must be addressed while setting up the system.

The agent group planner is completed first because it identifies the various agent groups that are required for the system. After setting up the agent groups, one path planner is completed for each path to show the ACD call handling. Each path planner includes the primary agent group, recorded announcements, overflow groups, and interflow conditions.

Agent Group Planner

The agent group planner shown in this Figure assists in planning the distribution of workload between agent groups. After doing the initial sizing to determine the number of agent groups required to handle the calls, use this planner to assign the parameters to each group. This information will be used later during the CDE programming of the system.

The agent planner form contains space for eight agent groups. The fields shown in the box for each group are described below.

Agent Group

The top field in each box, labeled Agent Group #, specifies the number of the agent group. This number will be used later when assigning primary and overflow agent groups to the ACD paths.

Figure: Agent Group Planner

Name

The Name field specifies the name of the agent group. During CDE, transfer this information from the agent group planner sheet to the Name field on the ACD Agent Groups form (Form 39).

Overflow Time

The overflow time specifies the maximum length of time that a waiting ACD call remains at this group before overflowing. The timer range is 0 seconds to 54 minutes. The use of this field is optional.

The system performs a load calculation when each new call arrives at an agent group, or when the status of an agent changes. If the system predicts that a call will not be answered before the timer expires, the system forces an immediate overflow.

During CDE, transfer this information from the agent group planner sheet to the Overflow Timer field on the ACD Agent Groups Subform (Subform 39).

1st Threshold

The 1st Threshold field specifies the time period for the first call waiting threshold. If calls are waiting beyond this time period, the LCD symbol beside the Queue Status key on the SUPERSET telephones changes (see the Queue Status Indicators table). During CDE, transfer this information from the agent group planner sheet to the First Status Threshold field on the ACD Agent Groups Subform (Subform 39).

2nd Threshold

The 2nd Threshold field specifies the time period for the second call waiting threshold. If calls are waiting beyond this second time period, the LCD symbol beside the Queue Status key on the SUPERSET telephones changes again (see the Queue Status Indicators table). During CDE, transfer this information from the agent group planner sheet to the Second Status Threshold field on the ACD Agent Groups Subform (Subform 39).

After Work Timer

The Afterwork Timer field specifies the time allocated to an agent for completing paperwork following an ACD call. During this time, the agent will not receive ACD calls. The timer range is 0 seconds to 15 minutes. This time is included as part of each call in reports and statistics. It is recommended that the timer be set to a minimum of five seconds.

During CDE, transfer this information from the path planner sheet to the Afterwork Timer field on the ACD Agent Groups Subform (Subform 39).

Paths Using This Group

This box allows the system planner to note the paths using this group. Refer to this box when transferring agent group information to the path planner sheets.

Path Planner

The path planner sheet illustrated in this Figure is used in conjunction with the agent group planner when laying out the routing for incoming ACD calls. The planner identifies all major elements that must be addressed while setting up the system.

Once the path planner has been completed to the ACD system designer’s satisfaction, the information is transferred to the CDE forms for system programming.

The CDE forms that pertain to the ACD TELEMARKETER feature are described in the Programming section. For a description of all system CDE forms, refer to the Program CDE section. The Planner Sheets Appendix of this section contains additional blank copies of the Path Planner. This Figure shows a blank path planner sheet. The following subsections describe the fields on this sheet. Examples later in this section illustrate the use of a Path Planner and trace calls through completed path planners. Unless mentioned otherwise, all fields on the path planner have corresponding fields on one of the CDE forms used in programming the system.

Purpose of This Path

The Purpose of this Path field allows the designer to summarize in a few words the intention of this path. The information in this field is for information only and is not programmed on any form during CDE.

Name

The Name field contains a descriptive name that identifies the function of the path. This name appears in the ACD path monitor displays and on the agent’s telephone when the set is presented with a call. Agents that handle calls for more than one path can answer the caller with an appropriate greeting.

During CDE, transfer this information from the path planner sheet to the ACD Path field on the ACD Path form (Form 41).

Path Access Code

The Path Access Code field identifies the path to the rest of the system. The path access code can be a destination in the Non-Dial-In Trunks form and in the Call Rerouting Table. The path access code can also be attached to a Dial-In Trunk, or it can be entered in another path planner as an interflow point and programmed as a forwarding destination for a MOtel 5212 IP, 5215 IP, 5010 IP, 5207 IP, SUPERSET 410, SUPERSET 420, SUPERSET 430, SUPERSET 4015, SUPERSET 4025, SUPERSET 4125, or SUPERSET 4150 station.

During CDE, transfer the information in this field of the path planner sheet to the “Access Code for This ACD Path” field on the ACD Path form (Form 41).

Priority

The Priority field sets the relative priority for all calls that arrive on this path. The priority remains with the call for its duration, regardless of the overflow handling programmed for the path. Priorities range from 1 through 99, with 1 being highest.

During CDE, transfer this information from the Path Planner sheet to the Priority field on the ACD Path form (Form 41).

Path Number

The Path Number field identifies the path number in the range 1 through 99. This number is used to sort the paths on the Path Summary display.

Figure: Path Planner

During CDE, transfer this information from the Path Planner sheet to the ACD Path field on the ACD Path form (Form 41).

Delay to Answer

The Delay to Answer field contains the value of the Delay for Ringback timer. This value should be set high enough to ensure the caller hears ringback from the CO or SX-200 system before the agent answers. In some situations, caller confusion may arise if the agent answers the call before the caller hears ringback. This field can be set to any value from 1 second through 54 minutes. The system default is 3 seconds.

During CDE, transfer this information from the Path Planner sheet to the Delay For Ringback field on the ACD Path form (Form 41).

Recording 1

The Recording 1 box contains the following three fields to capture the parameters for the first recording group:

RAD: This field contains the access code of the RAD group that contains Recording 1. During CDE, transfer this information from the Path Planner sheet to the Recording 1: Access Code field on the ACD Path form (Form 41).

Start: The Start field specifies the time between the end of the Delay for Ringback timer and the start of the first recording. During CDE, transfer this information from the Path Planner sheet to the “Recording 1: Start Time” field on the ACD Path form (Form 41).

Name: The Name field specifies the name of the RAD Group for Recording 1. During CDE, transfer this information from the Path Planner sheet to the Name field on the Hunt Group form.

MOH or Ext

The MOH or Ext # field allows the designer to specify what callers hear after Recording 1 is finished. The designer can give callers music on hold (MOH) from the system music source, or an alternate music source from an ONS port. Circle MOH to indicate music from the system music source, or enter the ONS port directory number that supplies the alternate music. If no music is connected, the caller hears silence.

During CDE, if an alternate music source has been selected, transfer this information from the path planner sheet to the Recording 1: Music Source Following field on the ACD Path form (Form 41). If MOH has been selected, no CDE action is required because the system connects to the default music source.

Recording 2 through 4

Use the boxes labeled Recording 2 through 4 to specify the parameters for the additional recordings supplied to the caller while waiting for an agent. The MOH or Alternate fields can be used to specify different music sources following each recording.

During CDE, transfer the information from these fields as described under Recording 1.

Queued Calls

The Queued Calls above each agent group block represent the calls queued against the agent group. No information is required in this block and there are no corresponding entries on CDE forms.

Primary Agent Group

Transfer the number from the Agent Group Planner of the agent group designated as Primary for this path.

Overflow Group Exists

For the box labeled Overflow Group Exists, circle Yes if the first overflow group is to be assigned to this path, or No if no overflow group is assigned. If Yes was selected, repeat the planning steps above for each of the overflow groups.

Interflow

Next to the box labeled Interflow, circle Yes if an interflow point is to be assigned to this path. Circle No if no interflow is to be assigned.

Interflow Point

The Interflow Point # field specifies the access code for the interflow device. This access code can be a listed directory number for a station, console, nightfall, ACD path, station/set hunt group, UCD agent group, or system speedcall number.

If Select Drop is left blank, the system drops the call rather than allow the call to interflow. If interflow is allowed for this path, enter the directory number of the interflow point.

During CDE, transfer this information from the Path Planner sheet to the Interflow Point Access Code field on the ACD Path form.

Interflow Timeout

The Interflow Timeout field specifies the waiting time for an ACD call before the system routes the call to an interflow point outside the ACD system. The timer range is from 1 second to 54 minutes.

During CDE, transfer this information from the path planner sheet to the Interflow Timeout field on the ACD Path form (Form 41).

Allow Overflow to Interflow

The Allow Overflow to Interflow Point Before Timeout field specifies whether the system can force calls to the interflow point as soon as the system determines that the call is unlikely to be answered, without waiting for the Interflow Timeout timer to expire.

During CDE, transfer this information from the path planner sheet to the Allow Overflow to Interflow Point Before Timeout field on the ACD Path form (Form 41).

ACD Agent Sets

This section describes the ACD TELEMARKETER features on SUPERSET and Mitel IP telephones used by ACD agents. Descriptions of the following features are provided:

- ACD agent login and logout
- Agent functions
- Special feature keys, set displays and/or indicators.

The information in this section is aimed at persons planning an ACD installation, setting up an ACD system, and operating the sets in an existing system.

ACD Agent Login/Logout

All ACD positions are linked to software, not hardware, so the system recognizes a login from any SUPERSET 4015, SUPERSET 4025, SUPERSET 4125, SUPERSET 4150, SUPERSET 410, SUPERSET 420, SUPERSET 430, Mitel 5010 IP, 5020 IP, 5207 IP, 5212 IP, 5215 IP, 5220 IP, 5224 IP, 5312 IP, 5324 IP, 5330 IP, 5340 IP, Symbol MiNET Wireless and SpectraLink Netlink Wireless telephones within the SX-200 ICP, and immediately transforms the set to the user’s preprogrammed specifications. The system routes calls to an agent only after the agent logs in to the ACD system.

When a position logs in, the set’s name, COS, speedcall, and feature keys are replaced by those assigned to the position in CDE. Call forwarding, DND, redial, reminders, callbacks or messaging are not affected by position login.

While agents are logged in to the ACD system, they can’t program their personal keys.

Login

To log in, the agent dials an access code followed by the ID number assigned through CDE. Dial tone indicates a successful login. In addition, the status indicator beside the Make Busy key turns on solid and ACD LOGIN appears briefly on display sets.

Login Conditions

The following conditions must be met before an ACD position can log in:

- The position must not be already logged in.
- The position must be logging in to a SUPERSET 4015, SUPERSET 4025, SUPERSET 4125, SUPERSET 4150, SUPERSET 410, SUPERSET 420, SUPERSET 430, Mitel 5010 IP, 5020 IP, 5207 IP, 5212 IP, 5215 IP, 5220 IP, 5224 IP, 5312 IP, 5324 IP, and Symbol MiNET Wireless or SpectraLink Netlink Wireless telephone.
- The position will be unable to log in when there are line appearances of a prime line, or when there are key definitions in the position’s ACD Keys Template for keys which are not physically present on the set. An “INVALID KEY” message will appear in this case.
- The position must have the ACD template for the set type being programmed enabled in its Class of Service (COS).
- The Mitel IP or SUPERSET telephone must not have appearances of its prime line programmed elsewhere in the system, except as a Direct Station Select (DSS) key or Busy Lamp Field (BLF) indicator. The Mitel IP or SUPERSET telephone may have a Programmable Key Module (PKM), PKM 48, or PKM 12 associated with it.

NOTE: The position will be unable to log in if the set is a member of a ring group.

Logout

To log out, the agent dials the login/logout access code. ACD LOGOUT appears briefly on display sets. Dial tone indicates a successful logout. To log in to another group, the agent dials the access code followed by the ID number for the second group.

Logout Conditions

An agent cannot log out while on an ACD call. If an agent on an ACD call (in progress or on softhold) attempts to log out, the agent receives reorder tone.

If a position is logged in at an ACD SUPERSET telephone and the user changes the set at that extension to an illegal ACD device, the position will be automatically logged out.

ACD Agent Functions

Each logged in agent uses a Mitel telephone that is normally programmed with one line select key or personal key assigned as a Make Busy key, and a second as a Queue Status key. Mitel 5220 IP, 5020 IP, 5224 IP, SUPERSET 4025, SUPERSET 4125, SUPERSET 4150, SUPERSET 420, or SUPERSET 430 agents also have access to a HELP softkey. The following subsections describe the functions of these keys and their operation.

Make Busy Key

Purpose

Every agent set should be preprogrammed with one personal key allocated as a Make Busy feature key.

Activating the Make Busy feature prevents ACD calls from ringing the set. Normal operation of the set is not affected and calls in progress are not disrupted. An agent can press the Make Busy key when the set is idle or during a call. If the agent presses the Make Busy key during a call, the set is automatically placed in Make Busy state when the agent goes on hook. The set remains in the Make Busy state until canceled by the agent.

Operation

Press the Make Busy feature key. To cancel Make Busy, press the Make Busy key again. When calls ringing the set are not answered within the period specified by the Call Forward No Answer Timer in the agent's COS, the system places the set in Make Busy, and causes the LCD beside the Make Busy key to flash to advise the agent of the Make Busy state (see below). The agent must press the Make Busy key to cancel the Make Busy state.

Make Busy Indicators

The indicator beside the Make Busy key will flash flash (fast flash - 150 msec on, 150 msec off) when the set is in the Make Busy state. The indicator remains on solid when the set is not in Make Busy mode.

Queue Status Key

Purpose

The Queue Status key and the indicator beside the key show the agent the current status of the call waiting queue and the load condition of the queue.

The following operational information applies only to the Mitel 5220 IP, 5020 IP, 5224 IP, SUPERSET 4125, SUPERSET 4025, SUPERSET 420, and SpectraLink Netlink Wireless telephones. Pressing the Queue Status key on the SUPERSET 4015 or SUPERSET 410 telephone does not register the time of the Queue Status. Note, however, that the key indicators described below apply to all five telephones.

Operation - Mitel 5220 IP, 5020 IP, 5224 IP, SUPERSET 4125, SUPERSET 4025, SUPERSET 420, and SpectraLink Netlink Telephones

When the Queue Status key is pressed, the main display on the set shows the queue number, the number of ACD calls waiting in the queue, and the longest call waiting time. The figure below shows a typical display. In this example, group number 1 is displayed, showing 10 calls in the queue waiting to be answered. The oldest call has been waiting for 3 minutes and 16 seconds.

Figure: 5220 IP, 5020 IP, 5224 IP, SUPERSET 4125, SUPERSET 4025 and SUPERSET 420 Queue Status Display

At this point, the agent can press the SuperKey to terminate the Queue Status display. A NAME softkey will appear if the group had been assigned a name in CDE. Pressing this softkey displays the group name.

Operation - SUPERSET 4150 and SUPERSET 430 Telephones

When the Queue Status key is pressed, the main display on the set shows the queue number, the number of ACD calls waiting in the queue, and the longest call waiting time. This Figure shows a typical display. In this example, group number 1 is displayed, showing 2 calls in the queue waiting to be answered. The oldest call has been waiting for 1 minute and 2 seconds.

While on an ACD call, the HELP softkey is available to agents who use Mitel 5020 IP, 5220 IP, 5224 IP, SUPERSET 4125, SUPERSET 4025 or SUPERSET 420 telephones. See the HELP prompt paragraph for details of the HELP function.

Figure: SUPERSET 4150/430 Queue Status Display

At this point, the agent can press the Superkey to terminate the Queue Status display. A NAME softkey will appear if the group had been assigned a name in CDE. Pressing this softkey displays the group name.

HELP Softkey

While on an ACD call, a HELP softkey is available to agents who use Mitel 5020 IP, 5220 IP, 5224 IP, SUPERSET 4025, SUPERSET 4125, SUPERSET 4150, SUPERSET 420 or SUPERSET 430 telephones. See the HELP function paragraph for more details.

Queue Status Key Indicators

The indicator beside the Queue Status key on the SUPERSET telephones indicates the load condition of the agent queue. The LCD is off when there are no calls waiting for an idle agent. When ACD calls are waiting to be answered, the LCD lights to indicate the queue status according to predefined threshold levels for the agent's group.

The LCD is updated periodically to indicate when a call remains waiting in the queue beyond an assigned threshold time. Two status threshold times are programmed in CDE for each agent group. As the timers expire, the LCD is updated to inform the agent of the workload. The Queue Status Indicators table describes the indicators associated with the Queue Status key.

Table: Queue Status Indicators

Key Indicator Flash Rate	Meaning
OFF	no calls waiting
ON	calls waiting before first threshold period
SLOW FLASH (750 msec on / 750 msec off)	calls waiting between first and second threshold periods
PULSED FLASH (600 msec on / 150 msec off)	calls waiting longer than second threshold period
FAST FLASH (150 msec on / 150 msec off)	calls have overflowed

HELP Softkey

Purpose

During an ACD call, Mitel 5020 IP, 5220 IP, 5224 IP, SUPERSET 4025, SUPERSET 4125, SUPERSET 4150, SUPERSET 420, or SUPERSET 430 agent sets display the HELP prompt. Help allows the agent, while involved in an ACD call, to request that a supervisor monitor the call. The agent may also tape the call by pressing the HELP softkey and dialing the directory number of a recording device. This recording device must be a member of a hunt group.

The HELP function is not available on Mitel 5212 IP, 5215 IP, 5207 IP, SUPERSET 4015 or SUPERSET 410 telephones.

Operation - Mitel 5020 IP, 5220 IP, 5224 IP, SUPERSET 4025, SUPERSET 4125, and SUPERSET 420 Telephones

To request help while involved in an ACD call, the agent presses the HELP softkey. The display changes to CALL SUPERVISOR (or CALL SENIORS. if the agent is reporting directly to a senior supervisor).

Three choices are then presented to the agent. To complete the help request call to the supervisor, the agent may press the YES softkey. If the agent decides to terminate the help request, the SuperKey is pressed.

When the NO softkey is pressed, an ENTER NUMBER prompt appears. The agent may select another help destination by either dialing a valid destination number or by pressing a programmed speed call. A valid destination number in this case is the ID for the supervisor or senior supervisor, or an access code for a recording hunt group.

Once a valid number has been entered, the agent presses the CALL softkey to complete the call. HELP REQUESTED appears on the agent set when the destination begins to ring.

When the help request is answered, the agent set display changes to “XXXXX INTRUDING” where “XXXXX” is the helper’s extension number. If the destination is unavailable, the agent set displays DESTINATION BUSY. The ENTER NUMBER prompt is shown again to enable the Agent to redirect the request.

The person or recording device that responds to the help call is automatically placed in a “listen only” state. A supervisor or senior supervisor may break into the conversation by pressing the TRANS/CONF key.

Operation - SUPERSET 4150 and SUPERSET 430 Telephones

To request help while involved in an ACD call, the agent presses the HELP CALL softkey. The display changes to CALL SUPERVISOR or DIAL DIGITS. A SUPERVISOR softkey appears if the supervisor is logged in (otherwise, digits must be dialed to complete the call).

The agent may also tape the call by pressing the HELP softkey and dialing the directory number of a recording device. This recording device must be a member of a hunt group.

The Supervisor’s set display shows HELP: nnnn (nnnn is the calling agent).

Once a call to the supervisor has been established, the display on the agent’s telephone shows nnnn INTRUDING.

The HANG-UP softkey is always present at the agent set.

After Work Timer

Purpose

When an agent completes an ACD call, a programmable “After Work Time” period is allotted during which the agent can complete work generated by the ACD call.

Agents using Mitel 5220 IP, 5020 IP, 5224 IP, SUPERSET 4150, SUPERSET 4125, SUPERSET 4025, SUPERSET 420, or SUPERSET 430 telephones can cancel the “After Work Timer” if their work is completed before the timer expires. Canceling the timer allows the agent to take the next ACD call.

Displays are provided on Mitel 5212 IP, 5215 IP, 5010 IP, 5020 IP, 5220 IP, 5224 IP, SUPERSET 4015, SUPERSET 4025, SUPERSET 4125, SUPERSET 4150, SUPERSET 420, or SUPERSET 430 telephones to indicate that the After Work Timer is running. The SUPERSET 410 does not provide an After Work Timer display and an agent using this set is unable to cancel the work timer. An agent using a 5207 IP or SUPERSET 4015 set is unable to cancel the work timer.

Operation

To cancel the After Work Timer, press the RESUME softkey.

Auto Answer

The agent’s set can be programmed with a COS option to auto-answer when a call arrives at the set. Auto Answer can be forced to be turned on when the agent logs in. The auto-answer process is described below:

1. Call arrives at free agent.
2. The agent’s telephone gives a burst of ringing.
3. The agent’s telephone answers the call and connects the two parties
4. At the completion of the call, the external party hangs up.
5. Agent’s telephone gives a hang up tone (Miscellaneous tone). See Note below.
6. The After Work Timer starts.
7. When After Work Timer expires, a new call is waiting.
8. Agent’s telephone gives a burst of ringing, and the sequence repeats for all new calls.

Note: Agents occasionally mistake the hang-up tone, which indicates the end of a call, for a burst of ringing which indicates a new call. This can lead to confusion because the agent is actually on the After Work Timer rather than answering a new call.

Agents using a Mitel 5212 IP, 5215 IP, 5010 IP, or SUPERSET 4015 telephone must operate in Headset mode to use the Auto Answer feature.

ACD Supervisor and Senior Supervisor Sets

This section describes the ACD TELEMARKETER features on SUPERSET 4025, SUPERSET 4125, SUPERSET 4150, SUPERSET 430, SUPERSET 420, or Mitel 5020 IP, 5220 IP and 5224 IP telephones used by ACD supervisors and senior supervisors. SUPERSET 4015, SUPERSET 410, Mitel 5010 IP, 5212 IP and 5215 IP telephones cannot be used in the supervisor or senior supervisor positions. Descriptions of the following features are provided:

- Senior supervisor and supervisor login and logout
- Senior supervisor and supervisor functions
- Feature keys, reports, set displays and/or indicators
- Call monitoring
- Help calls

The number of agents involved determines whether there is a need for both supervisors and senior supervisors to oversee ACD operations.

The information in this section is aimed at persons planning an ACD installation, setting up an ACD system, and operating the sets in an existing system.

Login/Logout

The system recognizes a supervisor only after the supervisor logs in to the ACD system. Once logged in, the set takes on the properties assigned to the supervisor through CDE.

When a position logs in, the set's name, COS, speedcall, and feature keys are replaced by those assigned to the position in CDE. Call forwarding, DND, redial, reminders, callbacks or messaging are not affected by position login.

While supervisors are logged in to the ACD system, they cannot program their personal keys.

Login

To log in, the supervisor dials an access code followed by the ID number assigned through the ACD Supervisor Form in CDE. ACD LOGIN appears briefly in the display, and the supervisor hears dial tone

Login Conditions

The following conditions must be met before an ACD position can log in:

- The position must not be already logged in.
- The position must be logging in to a SUPERSET 4025, SUPERSET 4125, SUPERSET 4150, SUPERSET 430, SUPERSET 420, or a Mitel 5020 IP Phone, 5220 IP Phone, or 5224 IP Phone.
- A SUPERSET telephone user will be unable to log in when key definitions are in the position's ACD Keys Template for keys which are not physically present on the set. An “INVALID KEY” message will appear.
- The position must have an ACD template enabled in its Class of Service (COS).

Logout

To log out, dial the access code again. ACD LOGOUT appears briefly in the display and the supervisor hears dial tone. Press HANGUP or dial a number to make a new call.

If a position is logged in at a Mitel ACD telephone and the user changes the set at that extension to an illegal ACD device, the position will be automatically logged out.

ACD Supervisor Functions

The ACD supervisor position is reserved for the individual responsible for supervising one or more agent groups.

Supervisors are unable to answer ACD calls. Instead, they are assigned pre-programmed keys which allow them to display status reports for agent queues and individual agents, establish a call monitor on an agent, or respond to a help request from an agent.

The way in which a supervisor obtains status reports for agent queues and individual agents differs slightly depending on whether the supervisor is responsible for one or more than one agent group.

Two scenarios are described in the following sections: supervisors with only one agent group (starting at the Overview paragraph), and supervisors with more than one agent group (starting at the Purpose paragraph).

ACD Senior Supervisor Functions

The ACD supervisor position is reserved for the individual responsible for supervising one or more agent groups.

Senior supervisors are unable to answer ACD calls. Instead, they are assigned pre-programmed keys which allow them to display status reports for agent queues and individual agents, establish a call monitor on an agent, or respond to a help request from an agent.

Senior supervisors obtain status reports for agent queues and individual agents in much the same way as supervisors with more than one agent group. The information provided in the Purpose paragraph onward applies to both groups.

Supervisor Set With One Agent Group: Overview

The supervisor responsible for only one agent group will use the Queue Status, Agent Status, and Shift feature keys. The supervisor’s telephone requires only one Queue Status key. Remaining feature keys can be assigned as Agent Status keys.

The Queue Status key provides ACD call queue information. The Agent Status key provides status reports for individual agents. The Shift key allows more than one agent to be assigned to a single Agent Status key. These keys are described in more detail below.

Queue Status Key: Supervisor Set With One Agent Group

Purpose

The Queue Status key and the indicator beside the key serve two functions in showing the supervisor the current status of the call waiting queue and the load condition of the queue.

Operation - Mitel 5220 IP, 5020 IP, 5224 IP, SUPERSET 4125, SUPERSET 4025 and SUPERSET 420 Telephones

The Queue Status key is pressed to display a summary of queue activity.

The set displays the agent group number, the number of ACD calls in the queue waiting to be answered, and the length of time the oldest call has been waiting.

This Figure shows a typical Mitel 5220 IP, 5020 IP, 5224 IP, SUPERSET 4125, SUPERSET 4025 and SUPERSET 420 queue status display. The display indicates that queue number 1 has 4 calls waiting to be answered. The longest waiting call has been in the queue for 2 minutes and 4 seconds.

Figure: Mitel 5220 IP, 5020 IP, 5224 IP, SUPERSET 4125, SUPERSET 4025 and SUPERSET 420 Queue Status Display

At this point, the supervisor can press the SuperKey to terminate the Queue Status display.

A NAME softkey will appear if the group was assigned a name during CDE. Pressing this softkey will display the group name.

To display the next report, the supervisor can scroll forward or backward by using the Volume ↓ key and the Volume ↑ key.

A HELP softkey displays a prompt PRESS KEY 0-9. As each dial pad key is pressed, a help message is displayed on the set to remind the user which report is associated with that dial pad key. The Supervisor and Senior Supervisor Reports - Queue Status table below lists the supervisor queue status reports and the key that displays each report.

A CANCEL softkey appears after the HELP softkey is pressed. Pressing CANCEL returns the supervisor set to the queue status mode.

Table: Supervisor and Senior Supervisor Reports - Queue Status

Key Number	Sample Set Display	Meaning
0	21 2 1 0 2 8 (See Note below.)	Condensed queue status report: -ACD calls answered -Number of abandoned ACD calls -Number of logged in agents -Number of times agents made busy -Number of non-ACD calls handled by the group.
1	W A I T T I M E 1 0 : 2 3	The average waiting time, in minutes and seconds, of logged in Agents in an Agent Group.
2	# A C D C A L L S 2 1	Number of ACD calls the group has answered.
3	A C D C A L L 0 2 : 2 3	Average duration of ACD calls.
4	# N O N A C D 8	Number of non-ACD calls made/answered by the group.

5	N O N A C D 0 1 : 2 3	Average duration of non-ACD calls.
6	# M A D E B U S Y 2	Number of times agents made busy.
7	A V G B U S Y 0 1 : 0 3	Average duration of make busy.
8	# O N H O L D 2	Number of ACD calls put on hard hold.
9	A V G H O L D 0 0 : 5 5	Average duration of ACD calls on hard hold.
Note: A senior supervisor pressing the 0 Key will receive a different report (an example of which is shown below). # AGT LOGIN 6 This report indicates the number of ACD agents logged in.		

Operation - SUPERSET 4150 and SUPERSET 430 Telephones

Press the Queue Status key to display a summary of queue activity.

The set displays the agent group number, the group name, the number of ACD calls in the queue waiting to be answered, and the length of time the oldest call has been waiting.

The figure below shows a typical SUPERSET 4150, and SUPERSET 430 queue status display. The display indicates that queue number 1 has 2 calls waiting to be answered. The longest waiting call has been in the queue for 1 minute and 2 seconds.

Figure: SUPERSET 4150/430 Queue Status Display

The supervisor can display additional reports by pressing the Reports softkey. The supervisor can scroll forward or backward by using the Next Report and Prev Report softkeys. The SUPERSET 4150/430 Agent Status Reports table below lists the supervisor queue status reports.The figure below shows the options for selecting additional agent status reports or queue status reports.

Figure: Selecting Other Agents or Reports

Pressing the EXIT softkey returns the supervisor set to the queue status mode.

Table: SUPERSET 4150/430 Agent Status Reports

Set Display	Meaning
AVERAGE WAITING TIME 01:55	Average waiting time for the agent.
NUMBER OF ACD CALLS ANSWERED 25	Number of ACD calls answered by the agent.
AVERAGE DURATION OF ACD CALLS 02:25	Average duration of ACD calls.
NUMBER OF NON-ACD CALLS HANDLED 2	Number of non-ACD calls made/answered by the agent.
AVERAGE DURATION OF NON-ACD CALLS 01:46	Average duration of non-ACD calls.
NUMBER OF TIMES AGENTS MADE BUSY 2	Number of times the agent made busy.

Table: SUPERSET 4150/430 Agent Status Reports

Set Display	Meaning
AVERAGE DURATION OF BUSY STATE 02:58	Average duration of make busy state.
NUMBER OF ACD CALLS PUT ON HOLD 2	Number of ACD calls put on hard hold.
AVERAGE HOLD DURATION OF ACD CALLS 00:47	Average duration of ACD calls on hard hold.

Queue Status Key Indicators

The indicator beside the Queue Status key continuously shows the load condition of the agent group reporting to the supervisor. The indicator is off if there are no calls waiting for an idle agent. When ACD calls are waiting to be answered, the indicator lights to indicate the queue status that is based on predefined threshold levels defined for the group.

The indicator is updated periodically to show when a call remains waiting in the queue beyond an assigned threshold time. Two status threshold times are programmed in CDE for each agent group. As the timers expire, the indicator is updated to inform the agent of the workload. The Queue Status Indicators table defines the indicators for queue status on the Mitel ACD telephones.

Table: Queue Status Indicators

Key Indicator Flash Rate	Meaning
OFF	no calls waiting
ON	calls waiting before first threshold period
SLOW FLASH (750 msec on / 750 msec off)	calls waiting between first and second threshold periods
PULSED FLASH (600 msec on / 150 msec off)	calls waiting longer than second threshold period
FAST FLASH (150 msec on / 150 msec off)	calls have overflowed

Agent Status Key: Supervisor Set With One Agent Group

Purpose

The Agent Status key and the indicator beside the key serve two functions in showing the supervisor the current status of an individual agent, and reporting on the performance of the agent.

An agent can be in any one of the following states:

LOG OUT	WAITING
ACD CALL	ACD WORK

NON ACD or DND MAKE BUSY	ACD HOLD

Operation - Mitel 5220 IP, 5020 IP, 5224 IP, SUPERSET 4125, SUPERSET 4025 and SUPERSET 420 Telephones

Press the Agent Status key to start the report displays, beginning with the current state of the first agent in the group.

Pressing subsequent keys allows the supervisor to identify the name of the agent and to obtain further agent status information. Ten categories of agent reports are available to the ACD supervisor.

It is not necessary to exit one Agent Status display before moving on to check the status of the next agent. Pressing the Agent Status key while already in agent status mode allows the supervisor to start the report display for the next agent.

This Figure shows a sample display when the Agent Status key is pressed.

Figure: Agent Status Display

The supervisor is then presented with a number of options. Pressing the SuperKey terminates the Agent Status display.

Pressing the NAME softkey identifies the agent associated with the displayed agent status. Additionally, when these softkeys are pressed, the CALL softkey is displayed. This key allows the supervisor to call the agent without dialing the Agent ID number or extension number.

To display the next report, the supervisor can scroll forward or backward by using the Volume ↓ key and the Volume ↑ key.

The FWD or BACK softkeys allow the supervisor to move on to the next or to the previous agent in the group.

To access the HELP softkey, the user must press an Agent Status key and then dial 0 on the dial pad. The HELP softkey prompts the user to 'PRESS KEY 0-9'. Pressing a dial pad key displays a help message to remind the user which report is associated with the dial pad key. See the Senior Supervisor and Supervisor Reports - Agent Status table below for a list of dial pad keys and examples and explanations of the agent reports available.

Table: Senior Supervisor and Supervisor Reports - Agent Status

Key Number	Sample Set Display	Meaning
0	398 BOB	Agent identification number and name.
1	W A I T T I M E 0 1 : 5 5	Average waiting time for the agent.
2	# A C D C A L L 2 5	Number of ACD calls answered by the agent.
3	A C D C A L L 0 2 : 2 5	Average duration of ACD calls.
4	# N O N A C D 2	Number of non-ACD calls made/answered by the agent.
5	N O N A C D 0 1 : 4 6	Average duration of non-ACD calls.
6	# M A D E B U S Y 2	Number of times the agent made busy.
7	A V G B U S Y 0 2 : 5 8	Average duration of make busy state.

Table: Senior Supervisor and Supervisor Reports - Agent Status

Key Number	Sample Set Display	Meaning
8	# O N H O L D 2	Number of ACD calls put on hard hold.
9	A V G H O L D 0 0 : 4 7	Average duration of ACD calls on hard hold.

Operation - SUPERSET 4150 and SUPERSET 430 Telephones

Press the Reports softkey to access Senior Supervisor and Supervisor reports of Queue status. Press the Next Report or Prev Report softkeys to view the complete set of reports, as shown in this Figure.

Press the Agent Status softkey to access Senior Supervisor and Supervisor reports of Agent Status. Press the Next Agent or Prev Agent softkeys to view the complete set of reports, as shown in this Figure.

The figure below shows a typical supervisor menu for selecting Agent Status or Queue Status Reports.

Figure: Typical SUPERSET 4150/430 Agent or Report Selection

The SUPERSET 4150/430 Supervisor Agent Status Reports table below defines set displays of supervisor reports

Table: SUPERSET 4150/430 Supervisor Agent Status Reports

Set Display	Meaning
AVERAGE WAITING TIME 01:55	Average waiting time for the agent.
NUMBER OF ACD CALLS ANSWERED 25	Number of ACD calls answered by the agent.
AVERAGE DURATION OF ACD CALLS 02:25	Average duration of ACD calls.
NUMBER OF NON-ACD CALLS HANDLED 2	Number of non-ACD calls made/answered by the agent.
AVERAGE DURATION OF NON-ACD CALLS 01:46	Average duration of non-ACD calls.
NUMBER OF TIMES AGENTS MADE BUSY 2	Number of times the agent made busy.
AVERAGE DURATION OF BUSY STATE 02:58	Average duration of make busy state.
NUMBER OF ACD CALLS PUT ON HOLD 2	Number of ACD calls put on hard hold.
AVERAGE HOLD DURATION OF ACD CALLS 00:47	Average duration of ACD calls on hard hold.

Agent Status Key Indicators

The indicator beside the Agent Status key continuously shows the call-status of the agent assigned to the key. The indicator is off when the agent is logged out. The display changes to reflect changes in the status of the ACD agent. The Agent Status Indicators table below describes the agent status indicators on a SUPERSET telephone.

Table: Agent Status Indicators

Key Indicator Flash Rate	Meaning
OFF	agent logged out
PULSED FLASH (600 msec on / 150 msec off)	agent logged in - no calls waiting
FAST FLASH (150 msec on / 150 msec off)	agent in Make Busy status
ON	agent on ACD call
SLOW FLASH (750 msec on / 750 msec off)	agent on non-ACD call or in DND
ON	ACD call on hold
ON	after-call work timer

Shift Key: Mitel 5220 IP, 5020 IP, 5224 IP, SUPERSET 4125, SUPERSET 4025 and SUPERSET 420 Only

Agent Status - Supervisor Set With One Agent Group

If an ACD supervisor has more agents than available keys, the system provides a Shift key. The Shift key allows the supervisor’s set to accommodate more than one agent on a single Agent Status key.

If an ACD supervisor has more agents than available keys, the system provides a Shift key. The Shift key allows the supervisor’s set to accommodate more than one agent on a single Agent Status key.

The Shift key allows one Queue Status key to address more than one agent group. This key has no effect when the number of Queue Status keys programmed is greater than the number of agent groups.

Agent Status Operation

If there are three agents in a group and two Agent Status keys information for agents one and two is displayed by Agent Status keys one and two respectively.

Reports for the third agent are viewed on the first Agent Status key and are accessed by pressing the Shift key followed by the Agent Status key. At this point, the name and ID of Agent 03 are displayed and the indicator shows the status of the third agent.

The example below provides a configuration with three Agent Status keys and seven agents. Pressing Agent Status key 1 displays the status of Agent 1. Pressing the Shift key at this point displays Agent 4. Agents 2 and 3 are associated with Agent Status keys 2 and 3 respectively.

Table: Using the Shift Key to Check Agent Status

KEY NAME	Agent Status Key #1 display	Agent Status Key #2 display	Agent Status Key #3 display
Agent Status	Agent 1	Agent 2	Agent 3
Shift	Agent 4	Agent 5	Agent 6
Shift	Agent 7	none	none
Shift	Agent 1	Agent 2	Agent 3

Queue Status Operation

Senior supervisors and supervisors who are responsible for more than one agent group use one Queue Status key for each agent group, plus the Shift key and the AGENT softkey. Senior supervisors and supervisors who are responsible for more than one agent group cannot have Agent Status keys programmed on their sets.

Each Queue Status key provides information about one agent group. The Shift key allows access to more than one agent group on a single Queue Status key. The AGENT softkey provides individual agent status information.

If there are three agent groups and two Queue Status keys, information for agent groups one and two is displayed by Queue Status keys one and two respectively.

Reports for the third agent group are viewed on the first Queue Status key; they are accessed by pressing the Shift key and then the Queue Status key. At this point the name and ID of Agent Group 3 is displayed and the indicator reflects the status of the third agent group.

The example below shows a configuration with three Queue Status keys and seven agent groups. Pressing the Queue Status key 1 displays the status of Agent Group 1. Pressing the Shift key at this point displays Agent Group 4.

Table: Shift Key Operation

KEY NAME	Queue Status Key #1 display	Queue Status Key #2 display	Queue Status Key #3 display
Queue Status	Agent Group 1	Agent Group 2	Agent Group 3
Shift	Agent Group 4	Agent Group 5	Agent Group 6
Shift	Agent Group 7	none	none
Shift	Agent Group 1	Agent Group 2	Agent Group 3

Queue Status Key: Senior Supervisors and Supervisors With More than One Agent Group

Purpose

The Queue Status key and the indicator beside the key provide the senior supervisor or supervisor with queue and load condition information for one agent group.

In addition, the AGENT softkey, which appears after the Queue Status key has been pressed, provides reports on individual agents within the displayed group.

Operation - Mitel 5220 IP, 5020 IP, 5224 IP, SUPERSET 4125, SUPERSET 4025 and SUPERSET 420 Telephones

Press the Queue Status key of the desired agent group. The set displays the agent group number, the number of ACD calls in the queue waiting to be answered, and the length of time that the oldest call has been waiting.

The figure below shows a typical queue status display. The display indicates that queue number 5 has 10 calls waiting to be answered. The longest waiting call has been in the queue for 2 minutes and 4 seconds.

Figure: Agent Status Display

Several options are available to the senior supervisor or supervisor.

- Press the SuperKey to terminate the Queue Status display.
- Press the NAME softkey to display the name of the agent group, if a name had been assigned during CDE.
- Press the HELP softkey to display the prompt PRESS KEY 0-9. As each dial pad key is pressed, a help message is displayed on the set to remind the user which report is associated with that dial pad key. The Supervisor and Senior Supervisor Reports - Queue Status table lists the queue status reports and the key that displays each report.
- Press the Volume Ø key and the Volume ¦ key to scroll forward or backward to the next or previous agent report.
- Press the AGENT softkey (which appears after the Queue Status key has been pressed) to access a variety of reports for individual agents within the displayed group.

The identification of the first agent in the group is displayed initially. From this point, the senior supervisor or supervisor can scroll backward or forward through agent reports, call the agent by pressing the CALL softkey, request help on agent reports (see the Senior Supervisor and Supervisor Reports - Agent Status table), move onto the next or previous agent in the group (FWD and BACK softkeys) or CANCEL the agent reports and return to Queue Status mode.

Operation - SUPERSET 4150 and SUPERSET 430 Telephones

Press the Queue Status key of the desired agent group. The set displays the agent group number, the number of ACD calls in the queue waiting to be answered, and the length of time that the oldest call has been waiting.

The figure below shows a typical queue status display. The display indicates that queue number 1 has 2 calls waiting to be answered. The longest waiting call has been in the queue for 1 minute and 2 seconds.

Figure: SUPERSET 4150/430 Agent Status Display

Several options are available to the senior supervisor or supervisor.

- Press the SuperKey to terminate the Queue Status display.
- Press the Prev Report or Next Report softkeys to scroll forward or backward to the next agent report.

- Press the AGENT softkey (which appears after the Queue Status key has been pressed) to access many reports for individual agents within the displayed group.

The identification of the first agent in the group is displayed initially. From this point, the senior supervisor or supervisor can scroll backward or forward through agent reports, call the agent by pressing the CALL softkey, request help on agent reports

(see the Senior Supervisor and Supervisor Reports - Agent Status table), move onto the next or previous agent in the group (Next Agent and Prev Agent softkeys) or Exit the agent reports and return to Queue Status mode.

Call Monitoring

Purpose

The Call Monitoring feature allows the senior supervisor or supervisor to listen in on an agent's conversation. During a call monitor, the system gives the supervisory set a one-way audio path, preventing the agent and the caller from hearing the supervisor.

Restrictions

Monitoring can be performed on any line and on any agent conversation that can be overridden. Monitoring is not permitted, for example, on 5-party calls, held calls and conferences. Keyline privacy is ignored for the call monitor.

Programming

To enable monitoring:

- In Form 02, Features Access Codes, assign an access code to Feature 45 (Silent Monitoring).
- In Form 04, System Options and Timers, enable System Option 42, Silent Monitoring.

If agents are to be notified when a monitor is in progress, enable System Option 43, ACD Silent Monitoring Beeps. When the monitoring starts, all parties in the call hear beeps except for parties on PRI/BRI/QSIG/IP trunks. The agent display changes to show the extension number of the monitoring set followed by "INTRUDING" but only after the supervisor presses the TRANS/CONF key to change the call from a monitored call to a regular three-party conference call.

Operation

The senior supervisor or supervisor initiates a call monitor by dialing the ACD Monitor access code, followed by the agent's ID code.

- If the agent set is idle and COS option 655 is disabled in the supervisor's class of service, the supervisory set indicates the agent's set is idle.
- If the agent is in a call and call monitoring begins, the supervisory set displays the extension number of the agent's set.
- An agent may be monitored by only one supervisor at a time. A senior supervisor or supervisor attempting to monitor an agent who is already being monitored receives busy tone and the normal busy display.
- An illegal user attempting to set up a monitor causes an ACCESS DENIED message to appear on the set.
- An attempt to monitor an agent who is not logged in results in a set displaying that the number is invalid.
- Attempting to monitor an agent who has Do Not Disturb activated and is idle results in the supervisory set displaying that the set has Do Not Disturb activated.

At any time while monitoring, the senior supervisor or supervisor may enter the conversation by pressing the TRANS/CONF key.

If COS option 655 is disabled in the supervising set's class of service, that supervisor will be disconnected from monitoring when the agent places the caller on hold, transfers the call, or the call is terminated by the agent or the ACD caller. The supervisory set displays "DISCONNECTED" and the system gives re-order tone.

If COS option 655 is enabled in the supervising set's class of service, that supervisor will not be disconnected regardless of the call state, unless the agent logs off. If the agent set places the caller on hold, performs a transfer, or the agent or ACD caller terminate their call, the supervisor's set goes into waiting state and displays "WAITING".

The supervisor ends monitoring by pressing the CANCEL key.

Help Call Feature

The Help Call feature is initiated by an agent who needs assistance from a supervisor. A supervisor or senior supervisor receiving a help request gets an audible and visual indication on the Mitel 5020 IP, 5220 IP, 5224 IP, SUPERSET 4025, SUPERSET 4125, SUPERSET 4150, SUPERSET 430 or SUPERSET 420 telephone.

Operation

When an agent initiates a help request, the supervisory set rings and the display changes to "HELP" followed by the agent's name and ID number. When the senior supervisor or supervisor lifts the handset, a monitor begins. To form a conference with the agent and ACD caller, the supervisor/senior supervisor must press the TRANS/CONF key.

If a help request is sent to a busy supervisory set, the set rings and displays "HELP" with the agent's name and ID. The supervisory set can only handle the request after the current call is terminated.

If auto answer is enabled at a supervisory telephone, it will be ignored. The telephone will ring until the handset is lifted.

Reports

The ACD TELEMARKETER package generates a series of printer directed reports that list call and performance information for agents, paths, and groups. The Agent Shift Summary Record covers the agent's logon period and is automatically printed when an agent logs off. The Path and Group Summary reports present information collected over a pre-defined period.

The reporting parameters may be selected through the maintenance terminal or the console. Refer to Report Commands for descriptions of the parameters and how they are selected.

Agent Shift Summary Record

An Agent Shift Summary Record prints automatically whenever an agent logs out. Once enabled, this report requires no predefined time parameters. Refer to the SET Command section for instructions on enabling the Agent Shift Summary Record.

The Agent Shift Record - Example 1 Figure and the Agent Shift Record - Example 2 Figure provide examples of Agent Shift Summary Records. The fields in the record are defined in the Agent Shift Summary Record Fields Table.

Figure: Agent Shift Record - Example 1

The display in this Figure shows that on September 27, 1995, agent 349 of agent group 18 at extension 1432 logged on at 13:28 for a period of 6 hours and 35 minutes.

The agent answered 485 ACD calls which lasted a total of 3 hours, 34 minutes and 51 seconds.

Sixteen non-ACD external outgoing calls lasting a total of 1 hour, 25 minutes and 26 seconds were made by this agent during the shift. The agent's extension was in Make Busy state once for 15 minutes and 2 seconds.

Figure: Agent Shift Records - Example 2

The example in this Figure shows that on November 12, 1994, agent 24157 of agent group 1 at extension 5211 logged on at 08:26 for a period of 7 hours and 45 minutes.

The agent answered 36 ACD calls which lasted a total of 7 hours, 1 minute and 23 seconds.

Eight non-ACD external outgoing calls with a total duration of 31 minutes and 52 seconds were placed by the agent during the shift. The agent's extension was in Make Busy state three times and it spent a total of 35 minutes and 10 seconds in Make Busy.

Table: Agent Shift Summary Record Fields

Softkey Label	Function
DATE	Month/day/year (mm/dd/yy).
GROUP	Agent group (4 digits).
AGENT	Agent ID (5 digits).
EXTN	Extension number (5 digits).
LOGIN	Login time (hh:mm).

Table: Agent Shift Summary Record Fields

Softkey Label	Function
SHIFT	Agents' shift length (hh:mm).
CALLS	Number of ACD calls answered by the agent (4 digits).
TIME	Total time spent by the agent on ACD calls (hh:mm:ss).
EXT-OUT	Number of non-ACD external outgoing calls made (4 digits).
TIME	Total time the agent spent on non- ACD external outgoing calls (hh:mm:ss).
MAKE BUSY	Total number of times that the agent went Make Busy.
TIME	Total time the agent's extension was in Make Busy (hh:mm:ss).

Path Summary Report

A Path Summary Report is printed only for programmed entities. This report includes:

- total counts for traffic entering the path during the specified time period
- the number of those calls that were answered, abandoned, and interflowed.

This Figure provides an example of a Path Summary Report. Fields in the report are defined in the Path Summary Report Fields Table.

Figure: Sample Path Summary Report

The example in this Figure shows that path number 23 received 1732 calls and, of these, 1578 were answered within an average of 15 seconds.

Twenty calls were abandoned after waiting an average of three minutes and eight seconds. The 134 calls that interflowed out of the path did so after an average time of three minutes and 27 seconds. No calls were answered within the Path Service Level time.

Table: Path Summary Report Fields

Softkey Label	Function
PATH	Path number.
ENTERED	Total number of calls entering this path.
ANSWERED	Total number of calls entering this path that were answered and the average time to answer.
ABANDONED	Number of calls which entered this path that were abandoned and average time the caller waited before abandoning.
INTERFLOWED	Number of calls which interflowed out of this path and the average time before interflowing.
SERVICE	Number of callers that were answered within the path service level in this time frame.

Group Summary Report

A summary record for the Group Reports is printed for programmed entities only. The information in the Group Summary Report is categorized as:

- offered
- answered
- non-ACD external calls
- non-ACD other calls
- the average number of agents logged in during the period.

This Figure provides an example of a Group Summary Report. The Group Summary Report Fields table defines the fields in this report.

Figure: Sample Group Summary Report

The example above shows that agent group number 12 was offered 174 calls, 131 of which the group handled. Each agent spent an average of two minutes and 21 seconds speaking to the caller.

The 15 non-ACD external calls placed by the group had an average duration of four minutes and nine seconds. Other non-ACD calls, a total of 21, lasted an average of one minute and 57 seconds. An average of seven agents were logged in for this report period.

Table: Group Summary Report Fields

Softkey Label	Function
GROUP	Group number being summarized.
OFFERED	Total number of calls offered to the group during the reporting period from all paths.
ANSWERED	Total number of ACD calls that were answered by this group and the average length of time agents spent talking to the caller.
NON-ACD EXTERNAL	Total number of external calls made by this group and the average time of these calls.
NON-ACD OTHER	Total number of calls, other than non-ACD external calls and ACD calls, made and received by the group and the average time of these calls.
AVERAGE LOGGED IN	Average number of agents logged in during the period. The number logged in is calculated every time that the queue status keys are updated, and these counts are then averaged.

ACD Real Time Event

The ACD Real Time Event option is a purchasable option that monitors and records the activity of the entire ACD operation as it happens. The system transmits call status messages that reflect all changes of state of the line or device, to the host computer. Real time events are divided into two groups: call events and group statistics events. Call events report on individual ACD agent activity, and group statistics events provide a cumulative report for hunt group congestion.

Interface

The event records are output using an RS232 port. The physical location identifier of this port can be a Dataset 2103, an IP socket. The IP socket or dataset and its extension number are programmed in Form 12, Data Assignment. The data is directed to this extension number in Form 34, Directed I/O. Records that are output from this port are directed to the report generation package running on a PC.

Conditions

ACD Real Time Event requires the Mitel 6100 Contact Center Solutions application.

Deprogramming the Directed I/O, Form 34 stops the generation of records. If there are any failures in the link, the SX-200 system does not report them.

If you change a data descriptor option during installation or troubleshooting, you will need to reset the dataset for the option to take affect.

Records will be lost if

- The dataset port is disconnected and the buffered records are purged by the system
- A software restart is executed on the dataset
- The link between the main controller and the dataset goes down
- The port assignment is changed while the link is in use.

After a reset, the record number field will start at 0.

Programming

In Form 04, System Options and Timers, enable the purchasable FOR system option, ACD Real Time Event, Option 90.

In Form 12, Data Assignment, program the dataset and assign an extension number.

In Form 34, Directed I/O, program the new data stream to the dataset.

Note: Mitel 6100 Contact Center Solutions may be programmed to the RS-232 port on the MX controller (no other printout types can share the same RS-232 port) or to an IP socket connection (MX, AX or CX/CXi) using an allocated unique socket number.

- Select Add.
- Enter the extension number of the dataset.
- Tab to the PRINTOUT column and select MORE.
- Select ACD EVENTS.
- Tab to the PRINTOUT TYPE column and select AUTOPRINT.

Record Formats

The basic call format for call events is shown below. The STX and ETX characters are not printable. Therefore, you will not generally see these characters. Note that all ASCII characters are enclosed in single quotes.

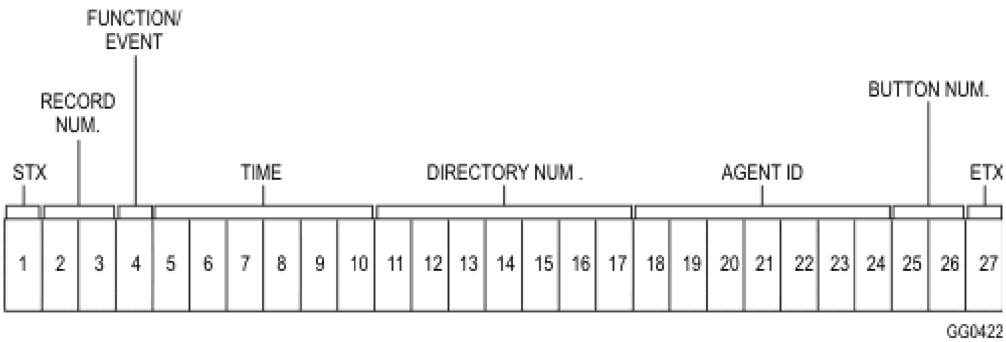


Figure: Basic Call Format

The agent status format is shown below.

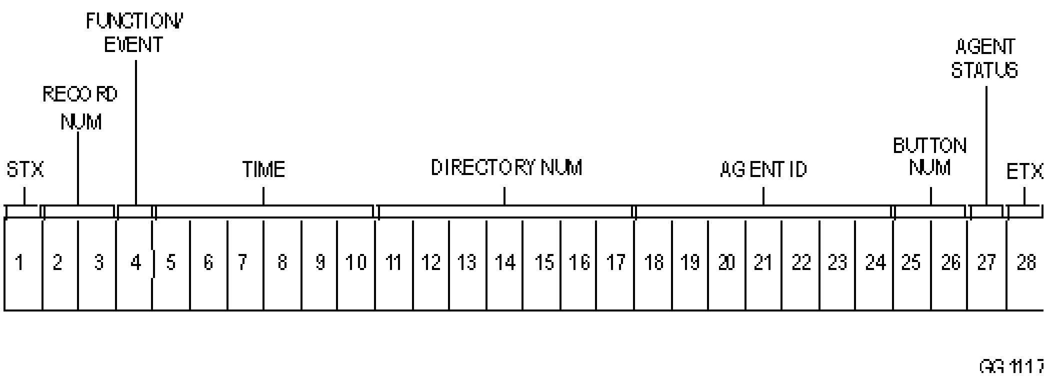


Figure: Agent Status Format

The Agent Group report is generated each time the caller enters or leaves an ACD queue and when the agent answers a call.The basic format for the agent group report is shown below.

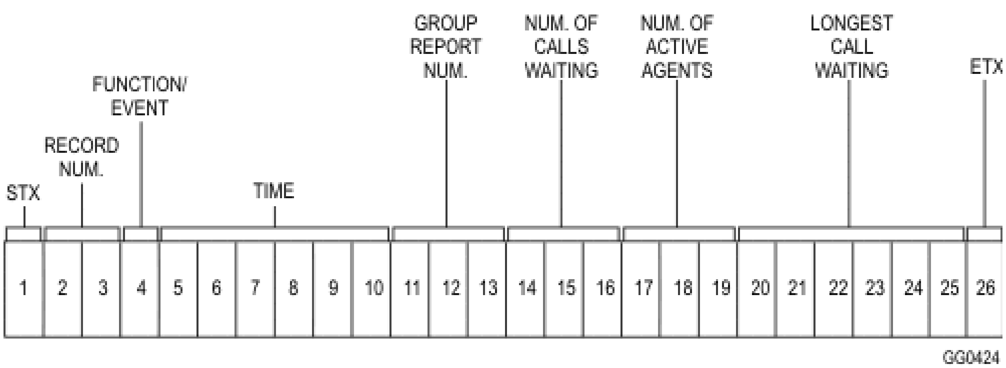


Figure: Agent Group Format

The ACD Path report is generated under the same conditions as the Agent Group report, that is, each time the caller enters or leaves an ACD queue and when the agent answers a call.The basic format for the ACD Path report is shown below.

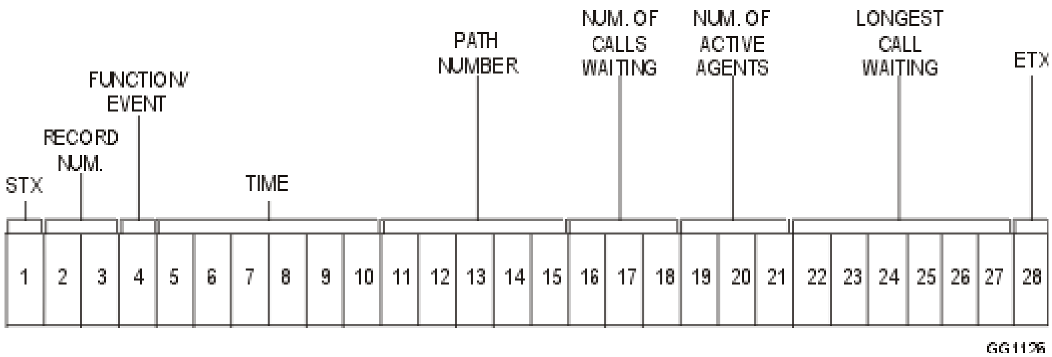


Figure: ACD Path Format

The Time report is sent every 60 seconds. The basic format for the Time report is shown below.

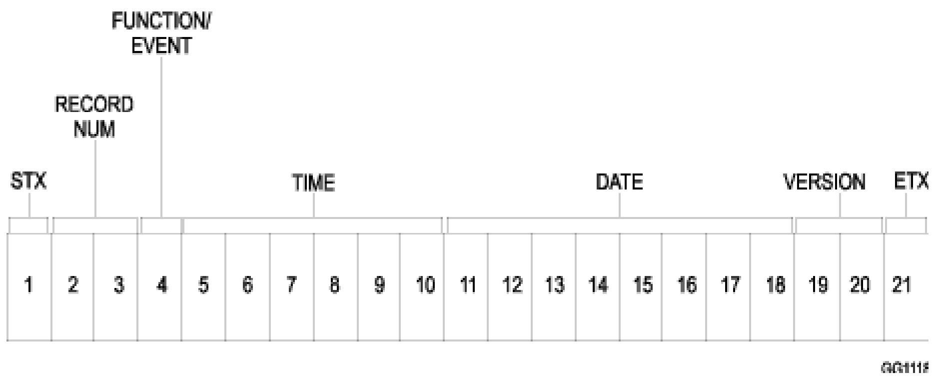


Figure: Time Report Format

STX

ASCII start of text. Hex value 2.

REC NUM

This is a two character field that contains the record number. This field goes from ’00’ to ’99’. After the ’99’th record, the numbers cycle back to ’00’.

FUNCTION/EVENT

The following function codes are used for event records:

’A’ agent login
’B’ agent logout
’C’ set Do Not Disturb (DND)
’D’ remove DND
’E’ set make busy
’F’ remove make busy
’G’ answer ACD call
’H’ answer non-ACD call
’I’ originate a call
’J’ agent status
’K’ agent group report

'L' work timer start
'M' work timer expire
'N' hold activate
'O' hold retrieve
'P' hold abandon
'Q' path report
'R' time report
TIME

This six-character field displays the time in 24-hour format.

DIRECTORY NUMBER

The directory number is a seven-character field.

AGENT ID

This seven-character field displays the agent ID. Blanks in this field means an agent is not logged in.

BUTTON NUMBER

This field is the button number of the line appearance that initiated the event. A button number of one designates the prime line. A button number of 99 designates line key numbers on PKMs that are above 99. All line key numbers above 99 will default to 99.

Events that are line appearance independent, have a button number of one. The following events always have a button number of one:

- Agent log in
- Agent log out
- Work timer start
- Work timer end
- Set make busy
- Remove make busy
- Set DND
- Remove DND event.

The call events that have button numbers are:

- Answer ACD call
- Answer Non-ACD
- Originate call
- Agent status
- Call hold
- Call hold retrieve
- Call hold abandon.

AGENT STATUS

The agents status has four codes:

Idle = '0'

Make busy = '2'

DND = '1'

Make Busy and DND = '3'

VERSION

The version field has two characters that display the version number for the ACD reporting functionality.

ETX

End of text. Hex value 3.

Call Events

Call events are generated only for ACD agents. A report is generated with the following agent activities:

- Agent log in
- Agent log out
- Set do not disturb
- Remove do not disturb
- Set make busy
- Remove make busy
- Answer ACD call
- Answer Non-ACDl call
- Originate a call
- Agent status
- Work timer start
- Work timer expire
- Call hold
- Call hold retrieved
- Call hold abandoned.

Agent Log In

This record is generated each time an agent successfully logs on. The extension number is the prime line of the ACD set that an agent logs on to.

Agent Log Out

This record is generated whenever an agent successfully logs out. The extension number is the prime line of the ACD set that an agent is logs out of.

Set Do Not Disturb

This record is generated whenever an agent sets DND. The records do not differentiate between setting DND locally and remotely (for example, an attendant). The extension number reported is always the prime line of the ACD extension.

Remove Do Not Disturb

This record is generated whenever an agent removes DND. The extension number is always the prime line of the ACD extension.

Set Make Busy

This record is generated whenever an extension is placed on the ‘Make Busy’ state. The extension number is always the prime line of the ACD extension.

Remove Make Busy

This record is generated whenever an extension has removed the "Make Busy" state.The extension number is always the prime line of the ACD extension.

Answer ACD Call

This record is generated whenever an agent answers a call that was directed to the ACD hunt group to which the call was placed.The event record contains Agent ID and extension number (hunt group number that the call was placed to).

Answer Non-ACD Call

All non-ACD hunt group calls are included in this category. The extension number is always the prime line of the ACD extension that answered.

Originate Call

This record is generated whenever an agent originates a call and enters a conversation. A record is not generated under the following conditions; entering a feature access code, dialing an invalid number, or hanging up before the called party answers. The extension number of the agent is any line on the ACD set that originates the call.

Agent Status

Agent Status is generated when the agent completes an originated call or when the agent completes a non-ACD call (no work timer has been started).

Work Timer Start Event

This record is generated when an agent terminates an ACD call and the work timer is started for that agent.

Work Timer Expire Event

This record is generated when the work timer expires for that agent.

Call Hold Event

When an agent places a call on hold, a record is generated for that line appearance. If the agent places another call or answers a call on a second line, the call event record for the new line will not cause confusion for the reporting package.

Hold Retrieve Event

When the held call is retrieved, a record is generated for that line indicating that the agent is now busy on that line.

Hold Abandon Event

When the held call is abandoned (the caller hangs up), a record is generated for that line.

Call Event Record Examples

This following examples show a real time event in a modified format to enhance readability. Path 001 has a directory number of 8800, agent 1007 is logged in on set directory number 2007. A work timer is programmed.

In this example, agent 1007 answers a call for path 8800 and completes the call.

STX	01	Answer ACD Call	12:23:07	8800	1007	01	ETX
STX	02	Work Timer Start	12:23:37	2007	1007	01	ETX
STX	03	Work Timer Expire	12:23:47	2007	1007	01	ETX

Group Statistics Events

This record is generated for each agent group assigned real time events in the ACD Agent Group form.

The group statistics event is modified to identify the agent group by its ID number.

Group Statistics Event Example

STX 01 Group Statistics 12:23:27 003 5 1 001133 ETX

The group statistics event indicates agent group id 003, has 5 calls waiting, 1 free agent and the longest waiting caller has been queued for 0 hours, 11 minutes, 33 seconds.

Path Statistics Events

This record is generated each time there is an activity on the path.

The path statistics event is modified to identify the path by its ID number.

Path Statistics Event Example

STX 01 Path Statistics 12:23:27 123 3 00 002408 ETX

The path statistics event indicates path 123, has 3 calls waiting, 0 free agents and the longest waiting caller has been queued for 0 hours, 24 minutes, 8 seconds.

Time Report Event

This record is generated once at the start of each new cycle of agent group and path statistic events. A record is generated every minute.

Time Report Event Example

STX 01 Time Report 12:23:27 19991231 00 ETX

The time report event signifies the start of new cycle of group and path statistic events. The event provides the current system date (i.e. in the example Dec. 31,1999) as data and also states the version number (00) of the ACD reporting functionality.

Printing Reports

Before the reports can be printed, the printer assignment must be completed as follows:

CDE Form 34, Directed IO

- assign the ACD AGT SUM and ACD GRP SUM printouts.

Refer to the Program CDE section for programming details.

Report Commands

Either the maintenance terminal or console can be used to enter the softkey commands and parameters needed to start the ACD Reports. Refer to the RS-232 Maintenance Terminal section for instructions on using the terminal. The following subsections describe the report commands.

Accessing Reports

To access the reports, log in to maintenance at the console or maintenance terminal using the correct USERNAME and PASSWORD. The Accessing Reports: Softkey Labels Figure displays the softkey labels that appear after maintenance login.

To access the ACD Reports subform, press the ACD_REPORTS softkey. The softkey labels at this level are shown in this Figure. The SET softkey sets up the various parameters for the summary reports. The SHOW softkey shows the current setup of the ACD printouts. QUIT exits the user from the ACD Report mode. These softkey commands are described in more detail below.

Figure: Accessing Reports: Softkey Labels

SET Command

Use the SET command to enter or change any ACD Report parameters. After pressing the SET softkey, the prompts change as shown in the figure below.

Figure: Softkey Labels, SET Softkey Subform

Use the softkeys to enter the required data and to exit from the ACD REPORT function. The softkey functions are described in the SET Softkey Subform Functions table below.

Table: SET Softkey Subform Functions

Softkey Label	Function
AGENT_SHIFT	Enables/disables printing of the agent shift summary reports. Enabled by default.
PERIOD	Sets the length of each reporting period from 10 minutes to 1 hour in 10-minute intervals. The default is 10 minutes.
DURATION	Sets the number of reporting periods. The duration must be a minimum of one period. The product of the duration multiplied by the period must be less than or equal to 24 hours. The default duration is 6 periods.
AUTOPRINT	Enables/disables printing of an intermediate report at the end of each reporting period. Enabled by default.
CANCEL	Terminates the SET command.
START_TIME	Sets the time of day, at 10-minute intervals, when the shift data collection starts. The start time must be defined before the group summary reports can be enabled. The default start time is 08:00.
GRP_SUMMARY	Enables/disables printing of the group summary reports. Disabled by default. A start time must be defined before the report is enabled.

Notes:

1. The DURATION, PERIOD, GRP_SUMMARY, and the START_TIME cannot be changed once the report is running. Stop the report before attempting to change these parameters.
2. The start time is used to clear the shift buffers, even if no printouts are enabled. See ACD MONITORS section.

SHOW Command

Use the SHOW command to display the status of the ACD reports. After the SHOW softkey is pressed, the labels change as shown in the figure below .

Figure: Softkey Labels for SHOW Softkey Subform

Use the softkeys to enter the required data. The SHOW subform softkey functions are described in the SHOW Softkey Subform Functions table below.

Table: SHOW Softkey Subform Functions

Softkey Label	Function
STATUS	The STATUS softkey displays the following information in the screen window: GROUP SUMMARY : ON/OFF STATUS : Active/Inactive AUTO PRINT : ON/OFF START TIME : hh:mm PERIOD : nn minutes DURATION : nn periods AGENT SHIFT : ON/OFF
CANCEL	Cancels the show command.

QUIT Command

This softkey command allows the user to exit from the ACD REPORT mode at the console or terminal.

ACD Monitors

The ACD Monitors act as a “window” to the ACD system by giving ACD supervisors an event-display that is updated after the completion of each ACD activity. This section describes the purpose and content of each monitor and defines the monitor fields and the keys available in each display.

Four Types of Monitors

The monitors allow access to four areas of the ACD TELEMARKETER system. The supervisor may view current information for:

System Activity

The System Activity monitor displays the current status of the ACD system. The display shows the number of agents logged in, the number of calls in the system, and general statistics on agent performance. Refer to System Activity Monitor Display for further details.

Paths

The Path Summary monitors are a series of displays for individual paths. Displays include the CDE programmed parameters of the path, the current activity on the path, and a brief statistical analysis for the path. See Accessing the Path Summary Display.

Agent Groups

The Agent Groups Summary monitors are a series of displays for individual agent groups. Displays include the CDE programmed data for each agent group, the current activity of the group, and a brief current and historical analysis of statistical data for the group. Refer to Monitoring Groups Overview section.

Agents

The Agent Information monitors are a series of displays for individual ACD agents. Displays include CDE programmed data for each agent in the system, the current activity of the agents, and a brief current and historical analysis of statistical data for the agents. More details are provided in Monitoring Agents Overview.

Hierarchy of Monitor Displays

The monitor data is presented in a series of displays on a standard VT100 compatible terminal. Most displays show a summary of performance over the last hour of operation, or from the beginning of an agent's shift. The monitors are arranged in a hierarchy as shown in the figure below.

Figure: Monitor Hierarchy

Accessing Monitors

Restrictions

The monitors are accessed through a Telnet session or a third-party RS232-to-IP serial port converter that connects the monitor terminal to the network.

As many as four users can access the monitors simultaneously. While most displays are updated every 5 to 10 seconds, the update rate will vary according to the number of users currently requesting monitor displays.

Entering Monitor Mode

Perform the following steps to gain access to the monitors through a VT100-type terminal:

1. If the terminal has a TCP/IP connection to the network, start a Telnet session and connect to the SX-200 ICP using the IP address of the SX-200 ICP and the IP socket port number assigned in Form 12.

   If the terminal is serially-connected, press <**Return**> key. The system responds with:

   Welcome to Mitel SX-200 Data Switching Note:1.
2. To request the monitors, enter: **MONITOR ACD** or **M ACD**. The display shows call progress as follows:

1 - VT100 compatible
2 - IBM PC
select terminal type: 1
enter password: 1000

3. If the limitation of four simultaneous monitors has been exceeded, the system response following the MONITOR ACD request changes to:

   Ringing
   System Busy, Try Again Later

Notes:

1. The system response is a programmable herald that is selected during the programming of the Data Transceiver's Class Of Service. Refer to the Program CDE section for details.
2. The default password is 1000; however, the password is programmable. Contact your communications manager if the password needs to be changed. New passwords are assigned by accessing CDE Form 28 from a maintenance terminal or console. The level of access for monitors is softkey number 2, SUPERVISOR.

Once logged in to the system, the user is presented with the SYSTEM ACTIVITY screen, described in detail in System Activity Monitor Display.

Monitoring System Activity

System Activity Monitor Display

After selecting MONITOR ACD, the supervisor is presented with the SYSTEM ACTIVITY monitor as shown in this Figure. This display contains a summary of the entire ACD system showing all current call activity and agent activity as well as a summary of system performance over the past hour.

The fields in the System Activity Monitor display are described in the Terms Used In System Activity Display table.

Figure: System Activity Monitor Display

System Activity Monitor Softkeys

The System Activity monitor display is accompanied by softkeys used to enter the second level in the monitor hierarchy. The softkeys presented with this display provide access to detailed displays of performance of specific paths, agent groups or agents. The System Activity Monitor Softkeys table below explains the various softkeys and their purpose.

Table: System Activity Monitor Softkeys

Softkey Label	Function
PATHS	Prompts the user to specify a path ID. Displays the Path Monitors for the selected Path. Refer to the "access the Path Summary display" paragraph for details.
AGENT GROUPS	Prompts user to specify an Agent group number. Displays the Agent Group monitors for the selected group. Refer to the "forms provide" paragraph for details.
AGENTS	AGENTS prompts for an Agent Id number. Displays the Agent Monitors showing current agent activity and the last hour's statistics. Refer to the Figure: Agent Activity Display for details.

Table: System Activity Monitor Softkeys

Softkey Label	Function
LANGUAGE	LANGUAGE provides a FRANCAIS softkey; when pressed, it converts softkey prompts and the command line to French. When prompts are in French, the softkeys toggle to LANGUAGE and ENGLISH to allow selection of English prompts.
QUIT	Logs the user out of the Monitor ACD application and terminates the session.
PRINT	Prints the System Activity form on the printer used for the ACD Monitor Print.

Table: Terms Used In System Activity Display

Term	Meaning
Caller Activity	
Callers	The number of callers within the ACD system including callers talking to agents and callers waiting for agents, but not including callers in the delay for ringback.
Calls Wtg	The number of callers queued up waiting for an agent to become available, including those listening to silence, music, alternate music, or a recorded announcement.
Longst Wtg	The duration, in minutes and seconds, of the call that has been waiting longest in the queues.
Under 1st	The percentage of callers in the system that have been waiting less than the first threshold time programmed for the primary group of the path.
Between	The percentage of callers in the system that have been waiting longer than the first threshold time but less than the second threshold time programmed for the primary group of the path.
After 2nd	The percentage of callers in the system that have been waiting longer than the second threshold time programmed for the primary group of the path.
On Recrdng	The number of calls that are listening to a recorded announcement while waiting in the queues.
Held Calls	Indicates the number of ACD callers that have been placed on hold.
Agent Activity	
On ACD	The number of agents currently on ACD calls.
Ready	The number of agents currently ready. Those agents are not on any type of call and are available to receive ACD calls.
Make Busy	The number of logged in agents that are in MAKE BUSY. These agents receive no ACD calls.

Table: Terms Used In System Activity Display

Term	Meaning
DND	The number of logged in agents that have DO NOT DISTURB activated. These agents receive no ACD or non-ACD calls.
On Non ACD	The number of agents currently involved in incoming non-ACD calls or agent originated calls.
Logged On	The number of agents currently logged into ACD.
Logged Off	The number of agents currently NOT logged in to the ACD system.
Summary - Last Hour	
Entered	The total number of ACD calls that have entered a path in the ACD system over the past hour.
Time To Ans	The average time before a call is answered by an agent.
SVC Level	The summary - from all paths in the system - of the path service level statistics. The first field shows the number of calls answered within the paths' service time; the second is the percentage of calls answered outside of the service time.
Ans By Agt	The number of calls received over the past hour that have been answered by an agent and the average duration of the calls.
Ans By Agt	The percentage of all ACD calls that entered over the past hour that were answered by an agent.
On Non ACD	The number of calls over the past hour that were either incoming non-ACD calls answered by an agent or agent originated calls, and the average duration of these calls.
Abandoned	The number of callers who abandoned before being answered by an agent and the average time a caller waited before abandoning.

Monitoring Paths

Accessing the Path Summary Display

To access the Path Summary display, the user presses the PATHS softkey at the system level. The system responds by asking for a path access code. After the user enters a valid path access code, the screen changes to the Path Summary display shown in the figure below.

Figure: Path Summary Display

Path Summary Display

The system displays path information on four forms. Entry into the Path monitor sub-level begins with the Path Summary display. See the Figure) which shows activity on the requested path.

The chosen path is the top entry on the display. If the user presses the RETURN key without first entering a path number, the lowest number path is displayed.

Entering an illegal number, such as an unprogrammed or out-of-range path, results in an error message on the work-line. The system offers the CANCEL softkey. The **<Return>** key also cancels the error message.

The Terms Used In Path Summary Display table below describes the fields in the Path Summary Display.

Table: Terms Used In Path Summary Display

Term	Meaning
Current Information	
Path Num	The path number. Paths are displayed in ascending order by path numbers (range = 1 - 50).
Path Name	The name of the path as programmed in CDE.
Access Code	The access code of the path (1 - 5 digits).
Num Calls Wtg	The number of ACD calls which originated on this path that are currently queued against any of the groups programmed in the path.
Conn To Agts	The number of callers from this path currently talking to agents of any of the groups programmed in this path.
Summary - Last Hour	
Entered	The number of calls that entered this path. See Note below.
Answered	The first entry is the number of calls answered by all groups in the path. The second entry is the average time to answer for those calls.
Abandoned	The first entry is the number of callers who abandoned while waiting for a group in this path. The second entry is the percentage of the calls offered that this represents, and the third is the average time a caller waited before abandoning.
Interflowed	The first entry is the number of callers who interflowed out of this path. The second entry is the average time to interflow for those calls.

Note: The Entered field on the Path Summary shows the number of times a call entered the path. In cases where a path interflows to itself or to another path, each call that interflows increments the entered count. Therefore, one call into the system may have “entered” many times.

Path Summary Softkeys

From this point, softkeys allow the supervisor to access detailed information in three categories: CDE programmed data, statistics gathered on the path over the past hour, and current path activity.

The softkeys for the Path Summary monitor are described in the following Path Summary Display Softkeys table. The displays that result from pressing the PROGRAMMING, STATISTICS and ACTIVITY softkeys are discussed in the following paragraphs.

Table: Path Summary Display Softkeys

Softkey Label	Function
PROGRAMMING	Displays the Path Programmed Data form that shows information programmed in CDE for the path currently bracketed by the work-line arrows. Refer to the Path Programmed Data form description
STATISTICS	Accesses the Path Statistics form that contains a statistical overview of activity on the path at the present moment, as well as a summary of statistics collected over the past hour.
ACTIVITY	Accesses the Path Activity display that contains a frequently updated view of current traffic on the selected path including waiting callers and idle agents. See the Path Activity form.
CANCEL	Returns the user to the System level display.
PRINT	Prints the Path Summary form on the printer used for the ACD Monitor Print directed printout.
PAGE UP	If there are more path summary lines than can fit on the screen, this key accesses paths with lower Path numbers than displayed on the screen. This key only appears when an upward scroll can occur.
PAGE DOWN	If there are more path summary lines than can fit on the screen, this key accesses paths with higher Path numbers than displayed on the screen. This key only appears when a downward scroll can occur.

Path Programmed Data Display

Pressing the PROGRAMMING softkey in the Path Summary window displays the Path Programmed Data form as shown in this Figure. This display contains the data entered during CDE for the selected path.

The fields in this display are described in the Terms Used In Path Programmed Data Display table below.

Figure: Path Programmed Data Display

Table: Terms Used In Path Programmed Data Display

Term	Meaning
Primary	Number and name of the primary agent group for the path.
Overflow 1	Number and name of the first overflow group for the path.
Overflow 2	Number and name of the second overflow group for the path.
Overflow 3	Number and name of the third overflow group for the path.
Interflow	Access code of the interflow point.
I/F Enabld	Yes in this field indicates that interflow is activated.
O/F to I/F	Yes in this field indicates that the call can flow from the last programmed overflow to the Interflow point before the Interflow Timeout occurs. The I/F

Table: Terms Used In Path Programmed Data Display

Term	Meaning
	ENABLD flag must also be “YES”.
I/F Time	The time, in minutes and seconds, before the call interflows out of the ACD system. Applies only if the flag I/F ENABLD is “YES”.
SVC Time	The time, in minutes and seconds, that defines the service level for this path. The time to answer of all calls answered on this path is compared to SVC Time and the results are shown as the service level data in the monitors.
Ans Delay	The time, in minutes and seconds, that the caller is allowed to hear ringback before the ACD system attempts to answer the call.
Priority	The priority of the ACD call. The highest priority is 1 and the lowest is 99.
RADs 1 through 4	
ID	The access code and name of the Recorded Announcement Device (RAD) hunt group.
Music	The access code of the music source to be used after the recording has been heard. If no access code has been programmed then ‘system’ music is displayed. When no system music is available this field is blank.
Start	The time, in minutes and seconds, that a caller is connected to the specified recording after entering this path.

The following Path Programmed Data Softkeys table describes the softkeys presented with the Path Programmed Data display.

Table: Path Programmed Data Softkeys

Softkey Label	Function
CANCEL	Returns the user to the System level display.
PREVIOUS	This softkey is present if paths exist that have lower access codes than the path being displayed. Pressing this key displays the programmed data for the next lower ACD Path.
NEXT	This softkey is present if paths exist that have higher access codes than the path being displayed. Pressing this key displays the programmed data for the next higher ACD Path.
RETURN	The RETURN key (either hard-key or softkey) returns the user to the PATH SUMMARY form.

Path Statistics Display

Pressing the STATISTICS softkey in the Path Summary window displays the Path Statistics display as shown in this Figure. This display provides a statistical overview of the path’s performance for the current instant, as well as a summary of activity over the past hour. The Terms Used In Path Statistics Display table below describes the fields in this display.

Figure: Path Statistics Display

Table: Terms Used In Path Statistics Display

Term	Meaning
Programmed Information And Current Summary	
Group Num	The agent group number or recording group number, as assigned during CDE.
Group Name	The name of the agent group or recording group, as assigned during CDE.
Num Agt/Rec	The number of logged in agents in the group. If a recording group, the number of RAD ports assigned to the group.
Num Calls Wtg	The number of calls from this path waiting for the agent group or recording group.
Conn Agt/Rec	The number of callers from this path currently talking to agents in the group, or listening to RADs.
Summary - Last Hour	
Offered	The number of ACD calls offered to the group from this path, and the percentage of the total calls offered on the path that this number represents.
Answered	The number of ACD calls answered by agents in the group from this path only, and the percentage of the total calls answered on the path that this number represents.
Overflowed	The number of calls that overflowed from the group to the next overflow point on this path, and the percentage of calls offered to the group which overflowed. Calls which overflowed to interflow are not counted.
SVC Level<	The number of ACD calls answered on the path within the path's service time, and the percentage of calls answered that this number represents.
SVC Level>	The number of ACD calls answered on the path outside the path's service time and the percentage of calls answered that this number represents.

The Path Statistics Display Softkeys table below describes the softkeys that are presented with the Path Statistics Display.

Table: Path Statistics Display Softkeys

Softkey Label	Function
CANCEL	Returns the user to the System level.
PRINT	Prints the Path Statistics at the printer used for the ACD Summary Reports directed printout.
PREVIOUS	This softkey is present if paths exist that have lower access codes than the path

Table: Path Statistics Display Softkeys

Softkey Label	Function
	being displayed. Pressing this key displays the statistics for the next lower ACD Path.
NEXT	This softkey is present if paths exist that have higher access codes than the path being displayed. Pressing this key displays the statistics for the next higher ACD Path.
RETURN	The Return key (either hardkey or softkey) returns the user to the Path Summary form.

Path Activity Display

Pressing the ACTIVITY softkey in the Path Summary window presents the Path Activity display as shown in this Figure. This display provides a continuously updated picture of the traffic on a given path. Information relates to the callers queued for the path as well as any ready agents waiting for calls from the path.

Figure: Path Activity Display

All queued callers that originated on the specified path are identified by trunk number or extension access code. The longest waiting caller for each group is at the bottom of the list, immediately above the horizontal line.

As the number of waiting calls increases or decreases, the information is updated in that group's column, and the horizontal bar below the first call in the queue shifts up or down the screen accordingly. The screen displays up to twelve waiting calls.

When there are no calls waiting in a queue, the sub-title under the group name changes to AGENT name and READY time. As many as 12 idle agents can be listed below the horizontal bar, beginning with the longest idle agent.

Overflow groups that were programmed but have no callers waiting and no agents logged in to a group, are identified by the group name in the column title. If an overflow group was not assigned during the customer data entry, the column remains blank.

The fields in the display are defined in the Terms Used In The Path Activity Display table below.

The Path Activity Display Softkeys table below describes the softkeys presented with the Path Activity Display.

Table: Terms Used In The Path Activity Display

Term	Meaning
Caller	The trunk number or extension access code of the caller waiting for the agent group.
No.	The queue position of the caller waiting for this group. The numbers are ordered but may not be sequential if there are callers waiting for this group who originated on another path. These other callers hold a queue position for this group but are shown on the activity display for their path.
Wtg	The time, in minutes and seconds, that the caller has been waiting in this path.

Table: Terms Used In The Path Activity Display

Term	Meaning
Agent	The name of a ready agent in the displayed agent group. If the agent does not have a name assigned in CDE, the agent's ID appears.
Ready	The time, in minutes and seconds, that the agent has been available and ready to accept an ACD call.

Table: Path Activity Display Softkeys

Softkey Label	Function
CANCEL	Returns the user to the System level.
PREVIOUS	This softkey is present if paths exist that have lower access codes than the path being displayed. Pressing this key displays the activity for the next lower ACD Path.
NEXT	This softkey is present if paths exist that have higher access codes than the path being displayed. Pressing this key displays the activity for the next higher ACD Path.
RETURN	The Return key (either hardkey or softkey) returns the user to the Path Summary form.

Monitoring Groups

Overview

Four forms provide information about the agent groups programmed in the ACD system and are described as follows:

- Group Summary form displays important information about each agent group.
- Group Programmed Data form displays the data programmed in CDE for each agent group.
- Group Statistics form provides statistics gathered on the agent group over the past hour and since the beginning of the shift.
- Group Activity form shows current caller and agent activity for the group.

Accessing the Agent Group Summary Form

To access the Agent Group Summary form, the user presses the AGENT GROUPS softkey at the system level. The system responds by asking for an agent group number. After the user enters a valid agent group number, the screen changes to the display shown in this Figure. The chosen group is at the top of the screen. The fields on the display are defined in the Terms Used In Group Summary Display table.

Pressing the RETURN key without first entering a group number also produces the display shown in this Figure. In this case, however, groups are listed in ascending order by group number.

Entering an unprogrammed or out-of-range group number results in an error message on the work-line. The CANCEL key allows deletion of the entry. Alternately, the user may press the RETURN hardkey and re-enter a group number.

Figure: Agent Group Summary Form

Table: Terms Used In Group Summary Display

Term	Meaning
Grp Num	The agent group's number.
Group Name	The agent group's name as programmed in CDE.
Current State	
AGENTS LOGGED ON	The number of agents in this group currently logged on.
NUM CALLS WTG	The number of ACD calls queued up, from all paths, for agents in this group.
ON ACD CALLS	The number of agents in this group currently active on ACD calls.
Summary - Last Hour	
OFFERED	The number of incoming ACD calls offered to the group from all paths.
ANSWERED	The number of incoming ACD calls answered by this group and the average duration of those calls.
OVERFLOWED	The number of calls that overflowed in any path while queued for this group as the primary group.

Agent Group Monitor Softkeys

Softkeys which appear in the Agent Group Monitor window allow the supervisor to access detailed information on CDE programmed data, statistics gathered on the agent group over the past hour, and current group activity. The softkeys for the Agent Group monitor are described in the Group Summary Form Softkey Labels table below. The displays that result from pressing the PROGRAMMING, STATISTICS and ACTIVITY softkeys are discussed in greater detail in the following paragraphs.

Table: Group Summary Form Softkey Labels

Softkey Label	Function
PROGRAMMING	Displays the Group Programmed Data form that shows the programmed CDE entries for the agent group currently bracketed by the work line arrows.
STATISTICS	Accesses the Group Statistics form that provides a statistical overview of the group's activity at the present moment, and a summary of statistics collected over the past hour.
ACTIVITY	Accesses the Group Activity display that shows a frequently updated view of current status of the agent group.
CANCEL	Returns the user to the System level. See System Activity Monitor Display for details.
PRINT	Starts a dump of this form to the printer used for the ACD Monitor Print.

Table: Group Summary Form Softkey Labels

Softkey Label	Function
PAGE UP	Accesses groups with lower numbers if there are more agent group summary lines than fit on one screen. The key appears only when an upward scroll can occur.
PAGE DOWN	Accesses groups with higher numbers if there are more agent group summary lines than fit on one screen. The key appears only when a downward scroll can occur.

Group Programmed Data Display

Pressing the PROGRAMMING softkey in the Agent Group Monitor window displays the Group Programmed Data form (as shown in this Figure). This form displays the data programmed during CDE for the agent group.

The table below describes the softkeys presented with the Group Programmed Data Display.The Terms Used In Group Programmed Data Display Table describes the fields in the display.

Figure: Group Programmed Data Display

Table: Group Programmed Data Softkey Labels

Softkey Label	Function
CANCEL	Returns the user to the System level.
PREVIOUS	This softkey is present if groups exist that have lower access codes than the group being displayed. Pressing this key displays the activity for the next lower ACD Group.
NEXT	This softkey is present if groups exist that have higher access codes than the group being displayed. Pressing this key displays the activity for the next higher ACD Group.
RETURN	The RETURN key (either hardkey or softkey) returns the user to the Group Summary form. Refer to the Monitoring Groups Overview paragraph for details.

Table: Terms Used In Group Programmed Data Display

Term	Meaning
Super Name	The name of the group’s supervisor, as programmed in CDE.
Super ID	The ID number of the group’s supervisor, as programmed in CDE.
Sen’r Name	The name of the group’s senior supervisor, as programmed in CDE.
Sen’r ID	The ID number of the group’s senior supervisor, as programmed in CDE.

Table: Terms Used In Group Programmed Data Display

Term	Meaning
Prog Agts	The number of agents programmed as members of this group.
Threshld 1	The first threshold time for a call waiting for the group. This time is used to indicate the overall caller load on the group. The flashing icons on agent and supervisory sets are updated to reflect the status derived from this time.
Threshld 2	The second threshold time for a call waiting for the group. This time is used to indicate the caller load on the group. The flashing icons on agent and supervisory sets are updated to reflect the status derived from this time.
O/F Timer	The setting of the ‘overflow’ timer for the group.
After Work	The setting of the after work timer for the group.

Group Statistics Display

Pressing the STATISTICS softkey in the Agent Group Monitor window displays the Group Statistics form (as shown in this Figure). This display provides a summary of statistics collected over the past hour and shift totals for the group.

Figure: Group Statistics Display

The table below describes the softkeys presented with the Group Statistics display. The Terms Used In Group Statistics Display Table describes the fields in the display.

Table: Group Statistics Display Softkey Labels

Softkey Label	Function
CANCEL	Returns the user to the System level.
PRINT	Starts a dump of this form to the printer used for the ACD Monitor Print.
PREVIOUS	This softkey is present if groups exist that have lower access codes than the group being displayed. Pressing this key displays the statistics for the next lower ACD Group.
NEXT	This softkey is present if groups exist that have higher access codes than the group being displayed. Pressing this key displays the statistics for the next higher ACD Group.
RETURN	The RETURN key (either hardkey or softkey) returns the user to the Group Summary form.

Table: Terms Used In Group Statistics Display

Term	Meaning
Ans By Grp	The number of ACD calls answered by the group both over the time interval and the average duration of those calls.
Offered	The number of ACD calls offered to the group from all paths over the time interval.
Overflowed	The number of calls that have overflowed from this group over the time interval when this group is the primary group of a path.
Logins	The number of logins which occurred in this group over the time interval.
Avg Agents	The average number of agents logged in over the time interval. The calculation is an average of counts taken every time the queue status indicators are updated.
Make Busy	The number of times members of this group went into Make Busy during the interval, and the average duration of the Make Busy states.
Outgoing	The number, and average duration, of external outgoing calls made by the agents in this group over the time interval.
Non ACD	The number, and average duration, of internal calls made and non-ACD calls received by group members over the time interval.

Note: The field definitions in the Terms Used In Group Statistics Display table above apply to the LAST HOUR field and the SHIFT TOTAL field. The shift start time is programmed from the System Maintenance facility via the ACD_REPORTS softkey. The group shift buffers are cleared at that time.

Group Activity Display

Pressing the ACTIVITY softkey in the Agent Group Monitor window displays the Group Activity form (as shown in this Figure). This form provides a continuously updated display of the activity of callers and agents within the group.

Figure: Group Activity Display

The table below describes the softkeys presented with the Group Activity display. The Terms Used In Group Activity Display Table defines the fields in the display.

Table: Group Activity Softkey Labels

Softkey Label	Function
CANCEL	Returns the user to the system level. See System Activity Monitor Display for details.
PREVIOUS	This softkey is present when a group exists that has a lower access codes than the group being displayed. Pressing this key displays the activity for the next lower ACD Group.
NEXT	This softkey is present when a group exists that has a higher access codes than the group being displayed. Pressing this key displays the activity for the next higher ACD Group.

Table: Group Activity Softkey Labels

Softkey Label	Function
RETURN	The RETURN key (either hardkey or softkey) returns the user to the Group Summary form.

Table: Terms Used In Group Activity Display

Term	Meaning
Caller Activity	
Calls Wtg	The number of callers queued for this group from all paths.
Longst Wtg	The waiting time of the longest waiting caller queued for this group.
Under 1st	The number of callers queued for this group who have been waiting for less than the first threshold timer value.
Between	The number of callers queued for this group who have been waiting for longer than the first threshold timer value but less than the second threshold timer value.
After 2nd	The number of callers queued for this group who have been waiting for longer than the second threshold timer value.
Overflowed	The number of callers queued for this group who have overflowed to the next group in the path, if programmed to do so.
Held Calls	The number of callers on hard hold by agents in this group.
Agent Activity	
Logged On	The number of agents currently logged on in this group.
Ready	The number of agents in this group ready to take an ACD call.
After Work	The number of agents who have just completed an ACD call, and their After Work Timer is active.
On ACD	The number of agents currently on incoming ACD calls.
Make Busy	The number of agents currently in Make Busy.
DND	The number of agents with Do Not Disturb activated.
On Non ACD	The number of agents currently on non-ACD incoming calls or outgoing calls.

Monitoring Agents

Overview

The following forms provide information about the agents who are currently logged in, and who are programmed in the ACD system. The primary form within the agent sub-level is the Agent Summary. This form provides an overview of current activity as well as a synopsis of statistical data collected over the past hour.

A set of softkeys in this form provides access to additional forms which supply detailed information about the agent and the agent's performance. These softkeys are described in the Agent Summary Form Softkey Labels table.

The additional forms are:

- Agent Programmed Data display showing all information entered during customer data entry that affects this agent.
- Agent Statistics display with details of the agent's performance over the last hour, and a comparison between the performance of this agent and the entire group.
- Agent Activity display showing the current status of any specified agent.

These forms are described in detail below.

Accessing the Agent Summary Form

After pressing the AGENTS softkey at the system level, the user is prompted to enter an agent ID number. After entering a valid agent ID, the user is presented with the screen shown in this Figure.

Information about the chosen agent appears at the top. The Terms Used In Agent Summary Display table defines the fields in the display.

Pressing **<Return>** without first entering an agent ID number also displays the screen shown in this this Figure, but in this case the agents are listed by group number and ID number from lowest to highest ID number as programmed during customer data entry.

Entering an unprogrammed or out-of-range agent number results in an error message on the work-line. The CANCEL key allows deletion of the entry. Alternately, the user may press **<Return>** and re-enter an agent number.

Figure: Agent Summary Form Display"

Table: Agent Summary Form Softkey Labels

Softkey Label	Function
PROGRAMMING	Displays the Agent Programmed Data form showing the CDE information for the agent currently bracketed by the work-line arrows. Refer to the Agent Programmed Data form.
STATISTICS	Accesses the Agent Statistics form that shows a statistical overview of the agent's activity at the present moment, and a summary of statistics collected over the past hour.
ACTIVITY	Accesses the Agent Activity display containing a frequently updated view of current status of the agent and the agent's set. See Agent Activity form.
CANCEL	Returns the user to the system level

Table: Agent Summary Form Softkey Labels

Softkey Label	Function
PRINT	Prints the Agent Summary form at the printer used for the ACD Monitor Print.
PAGE UP	Accesses agents with lower group numbers if there are more agent summary lines than fit on one screen. The key appears only when an upward scroll can occur.
PAGE DOWN	Accesses agents with higher group numbers if there are more agent summary lines than fit on one screen. The key appears only when a downward scroll can occur.

Table: Terms Used In Agent Summary Display

Term	Meaning
Agent Name And Current State	
Grp Num	The agent group's number.
Agent Name	The agent's name as programmed in CDE.
Agent ID	The agent's ID or access code.
ACD State	The current state of the agent as one of the following states: acd — on an ACD call dnd — the agent's set has Do Not Disturb active hold — agent has an ACD caller on hold makeb — agent has entered the make busy state nonacd — agent is on a non-ACD or outgoing call ready — agent is ready to accept an ACD call afterw — agent is in after work timer state
Summary - Last Hour	
Answered	The number and average duration of incoming ACD calls answered by this agent.
Cmp To Grp Av	The first percentage is the ratio of the number of incoming ACD calls answered by this agent compared to the total number of calls answered by the group over the same period. Agents logging on and off during the period affect the numbers. The second percentage is the ratio of the average call duration of incoming ACD calls answered by this agent compared to the average duration of calls answered by the group over the same period.
Non ACD	The number and average duration of non-ACD calls received and placed by the agent.
Make Busy	The number and average duration of times the agent was in a Make Busy state.

Agent Programmed Data Form

Pressing the PROGRAMMING softkey displays the Agent Programmed Data form as shown in the figure below. This form displays the data programmed for the agent by the installer during customer data entry. The softkeys and fields on the form are described in the following table and in the Terms Used In Agent Programmed Data Display Table.

Figure: Agent Programmed Data Display

Table: Agent Programmed Data Softkey Labels

Softkey Label	Function
CANCEL	Returns the user to the System level.
PREVIOUS	Displays the previous agent sorted by group number and agent ID. The softkey appears only when there is a preceding agent.
NEXT	Displays the next agent sorted by group number and agent ID. The softkey appears only when there is an additional agent.
RETURN	The RETURN key (either hardkey or softkey) returns the user to the Agent Summary form.

Table: Terms Used In Agent Programmed Data Display

Term	Meaning
Group Name	The group’s name as programmed in CDE.
Group Num	The group’s number as programmed in CDE.
Super Name	The name of the agent’s supervisor.
Super ID	The ID of the agent’s supervisor.
Sen’r Name	The name of the agent’s senior supervisor.
Sen’r ID	The ID of the agent’s senior supervisor.
Agent COS	The Class of Service assigned to the agent. When this agent logs in, this COS is transferred to the set, overwriting the set’s existing COS.

Agent Statistics Form

Pressing the STATISTICS softkey displays the Agent Statistics form as shown in the figure below. This form provides a summary of agent performance statistics collected over the past hour.

The Agent Statistics Form Softkey Labels table below describes the softkeys presented with the Agent Statistics display. The Terms Used In Agent Statistics Display table defines the fields on the display.

Figure: Agent Statistics Display

Table: Agent Statistics Form Softkey Labels

Softkey Label	Function
CANCEL	Returns the user to the system level.
PRINT	Prints this form to the printer used for the ACD Monitor print.
PREVIOUS	Displays the statistics, sorted by group number and agent ID, for the previous agent. The softkey appears only when there is a preceding agent.
NEXT	Displays the statistics, sorted by group number and agent ID, for the next agent. The softkey appears only when there is an additional agent.
RETURN	The RETURN key (either hardkey or softkey) returns the user to the Agent Summary form.

Table: Terms Used In Agent Statistics Display

Term	Meaning
Agent Data	
Group Name	The group's name as programmed in CDE.
Group Num	The group's number as programmed in CDE.
Login Time	The time the agent logged into ACD.
Login Date	The date the agent logged into ACD.
Shift Dur	The duration of the agent's shift since login.
Last Hour	
Ans By Agt	The number and average duration of incoming ACD calls answered by the agent.
Ready	The number of times the agent became ready to receive an ACD call and the average duration of each ready interval.
Make Busy	The number of times the agent was in Make Busy and the average time spent in the Make Busy state over the past hour.
Outgoing	The number of external outgoing calls placed by the agent and the average duration of these calls.
On Non ACD	The number of non-ACD calls handled by the agent. It includes both incoming and outgoing external calls, and all internal calls.
Shift Total	
Ans By Agt	The number and average duration of incoming ACD calls answered by the agent since login.

Agent Activity Form

Pressing the ACTIVITY softkey displays the Agent Activity form (shown in this Figure). "This form provides a continuously updated display of ACD information for the agent, plus status for the agent's set and the current call.

The following table describes the softkeys presented with the Agent Activity display. The Terms Used In Agent Activity Display Table defines the fields on the display.

Figure: Agent Activity Display

Table: Agent Activity Form Softkey Labels

Softkey Label	Function
CANCEL	Returns the user to the system level. Refer to the "Cancel softkey" paragraph for details.
PREVIOUS	Displays the statistics for the previous agent, sorted by group number and agent ID. The softkey appears only when there is a preceding agent.
NEXT	Displays the statistics for the next agent, sorted by group number and agent ID. The softkey appears only when there is an additional agent.
RETURN	The RETURN key (either hardkey or softkey) returns the user to the Agent Summary form.

Table: Terms Used In Agent Activity Display

Term	Meaning
Group Name	The group's eight character name as programmed in CDE.
Group Num	The group's number as programmed in CDE.
Set Ext	The extension number of the Mitel telephone where the agent is logged in.
Set Loc'n	The physical location (bay, slot, circuit, and sub circuit) of the Mitel telephone where the agent is logged in.
Set State	The software state of the set where the agent is logged in. The state will be one of: busy busyout campon dialing dnd held idle intoACD locked talking paging reorder ringback ringing select suspend
Call Type	The type of call the agent is on, either two-party or conference.
DND	Indicates that the agent's set is in Do Not Disturb mode.
Auto Answr	Indicates that the set will automatically answer ACD calls to the set without agent intervention.

Table: Terms Used In Agent Activity Display

Term	Meaning
English	Indicates the language of the set: English or French.
ACD State	Indicates the type of activity the agent is involved in: suspnd - agent is ending a call locked - agent's set has been left off-hook acd - on an ACD call dnd - the agent's set has Do Not Disturb active hold - agent has an ACD caller on hold makeb - agent has entered the make busy state nonACD - agent is on a non-ACD call ready - agent is ready to accept an ACD call afterw - agent is in after work time
Make Busy	Indicates that the agent is in a Make Busy state.
On ACD	The agent is on an ACD call.
Time On ACD	The duration of the current call.
ACD On Hold	The agent has an ACD caller on hard hold.
Caller	The trunk number or extension access code of the caller waiting for the agent group. Note that callers waiting for a group may not have originated on this path.
Orig Path	Indicates the path access code where this call originated.
Prime Grp	Indicates whether this agent's group is the prime group of the caller's path.

Programming

The ACD TELEMARKETER Applications Package is a software option that must be purchased.

Customer Data Entry for the ACD TELEMARKETER Applications Package involves specifying the routing of incoming ACD calls by entering data into a network of programming forms. Information in the forms is linked through a series of indexes and pointers. The forms which make up the ACD network are:

- System Options (CDE Form 04)
- Feature Access Codes (CDE Form 02)
- ACD Agent Groups (CDE Form 39)
- Agent Group Subform (CDE Subform 39)
- ACD Keys Template (CDE Form 38)
- COS Define (CDE Form 03)
- ACD Supervisors (CDE Form 40)
- ACD Paths (CDE Form 41)
- Hunt Groups (CDE Form 17)
- Call Rerouting (CDE Form 19).

System Options Form

This Figure shows a portion of the System Options form containing the options related to the ACD TELEMARKETER feature package. Options are changed by editing the Status field.

Figure: System Options Form (CDE Form 04)

Four fields on the System Options form control operation of the ACD TELEMARKETER feature. The System Option Form Fields table below defines the function of each field.

Table: System Option Form Fields

Option	Function
ACD Silent Monitoring (Option 42)	By setting this option to Enabled, a supervisor can dial a programmed feature access code to monitor ACD calls.
ACD Silent Monitoring Beeps (Option 43)	Set this option to Enabled if agents are to be notified when monitoring is in progress.
ACD Reports (Option 44)	Setting this option to Enabled changes the format of the SMDR records to that required by the ACD TELEMARKETER Reporting Package. For additional information, refer to the Station Message Detail Recording section.
Maximum ACD Agents (Option 104). See Note.	Setting this option to the number of agents purchased allows access to the programming forms related to ACD.

Note: System Option 104 (Maximum ACD Agents) must be set to a valid number before programming the remaining CDE forms related to ACD. The maximum number of ACD agents per bay is 50.

ACD Agent Groups Form

ACD agent groups are included with the ACD TELEMARKETER option you purchase. This Figure shows a blank ACD Agent Groups Form (CDE Form 39). All Agents must be a member of an ACD Agent Group. The system accommodates a maximum of 50 ACD agent groups. Each group must contain a minimum of one Agent. The maximum number of agents per group is 99. The ACD system supports 100 agents logged in at the same time.

The ACD system allows CDE programming of 999 ACD positions in any combination of agents, supervisors, and senior supervisors.

The agent information entered on this form is the agent name, agent ID, and COS. The agent ID is a 1- to 5-digit access code that allows the agent to log onto the ACD system. Entries in the Agent Group form are sorted by this ID number. The ID is associated with an agent, not a particular extension, so that any SUPERSET 410, SUPERSET 420, SUPERSET 430, SUPERSET 4150, SUPERSET 4125, SUPERSET 4025, Mitel 5220 IP, 5020 IP or 5224 IP telephone that the agent uses reflects that agent's name and ID.

The ordering of agents on the form has no effect upon the agent selection algorithm.

The fields on the ACD Agent Group form contain customer-defined data. The ACD Agent Group Fields table below defines the contents of each field.

Figure: ACD Agent Groups Form (CDE Form 39)

Table: ACD Agent Group Fields

Field	Meaning
ACD GRP	The group number and name.
ACD GROUP NUMBER	A programmable field containing a 1- or 2-digit number in the range 1 through 50 that identifies an ACD group.
ACD GROUP NAME	An optional field that allows the customer to assign a name to the agent group. A maximum of eight characters may be entered. The agent group must contain at least one member before the ACD GRP NAME softkey is presented.
AGENT ID	A programmable field of up to five digits for assigning an identification number to an ACD agent. This access code must not conflict with any other access code in the database. An ID number can exist in only one Agent Group. Agents requiring access to more than one group must be given a different ID for each group. Entries on the ACD Agent Groups form are sorted numerically by Agent ID.
AGENT NAME	An optional field that assigns a name to an agent ID. The agent name is carried to the set where the agent logs on. The name may be up to 10 characters but cannot begin with an asterisk (*). The same conditions apply to position names as well as to set names.
COS	The Class of Service number of this agent. Range is 1 to 50.

Agent Groups Subform

The Agent Groups subform (CDE Subform 39) shown in this Figure is accessed through the OPTIONS key. This form can only be accessed if there is at least one agent in the agent group. Entries are changed by editing the status column. Four fields on the ACD Agent Group subform contain customer-defined data. The contents of each field are defined in the ACD Agent Groups Subform Fields table below.

Figure: ACD Agent Groups Subform (CDE Subform 39)

Table: ACD Agent Groups Subform Fields

Field	Meaning
Afterwork Timer	An optional field to give the agent a wrap-up time (after work time) following ACD calls. Timer range is 0 seconds to 15 minutes. Default is 0 (no delay time before the next call is presented at the set). Campons and callbacks do not take precedence over a waiting ACD call. We recommend a minimum time of 3 seconds for every agent. Refer to the following paragraphs for additional information.
Overflow Timer	An optional field used to specify the maximum time a waiting ACD call remains in this group before overflowing. The overflow destination is defined in the ACD Path Form (CDE Form 41). The value entered can range from 0 seconds to 54 minutes. Default value is 9 minutes. Refer to the following paragraphs for additional information.
First Status Threshold	This time (range is 0 seconds to 54 minutes) must be less than the time specified in the Second Status Threshold. The field defaults to 3 minutes. Refer to the

Table: ACD Agent Groups Subform Fields

Field	Meaning
	following paragraphs for additional information.
Second Status Threshold	This timer (range 0 seconds to 54 minutes) must be greater than the time specified in the First Status Threshold field. The field defaults to 6 minutes. Refer to the following paragraphs for additional information.

After Work Timer

In many ACD situations, the agent may require some time after each ACD call to complete paperwork before accepting the next call. This subform allows programming of a wrap-up time (Afterwork Timer). The Afterwork Timer prevents an ACD call from being presented to this set until the specified time has expired. It is recommended that the Afterwork Timer be programmed.

Overflow Timer

The Overflow Timer is programmed for each agent group in the ACD system. It specifies how long an ACD call will wait in the queue for this group before being overflowed. Overflowed calls remain in this group's queue, but are added to a queue for another agent group, to increase the chances of the call being delivered to an agent. The time selected in this form specifies the maximum time a call can wait to be answered before the call overflows. The default time is 9 minutes.

If the system predicts that a call will remain unanswered before the time out period, the system ignores the specified timer and forces an immediate overflow. The two conditions described below can cause this forced overflow:

- If the agent group form specifies an overflow time of three minutes and no agents are logged on in this group, the system ignores the timer and forces an immediate overflow to avoid an unnecessary delay to the caller.
- The second case of overflowing before the specified time out arises during an overloaded state. The system performs an algorithm for an overloaded condition each time a new caller arrives for an agent group or when the status of an agent changes. Either event causes an overflow if excess callers are waiting for the agent group.

Threshold Timers

When an ACD call is initially routed to the agent group and there are no idle agents available, any appearance of the Queue Status indicator for this group reflects a call waiting in queue. This Queue Status indicator is driven by the threshold timers assigned to this form.

The First and Second Status Threshold timers provide a visual indication on all Queue Status keys of the current work load condition for this agent group.

ACD Keys Template Form

Assigning ACD Keys

The ACD Keys Template Form (CDE Form 38) allows global programming of Mitel telephones that require common ACD feature keys.

Global programming is still possible when these telephones are mixed in an ACD system, in spite of the varying number of line select keys available on the different Mitel telephones. The installer must ensure that the ACD Keys Template is programmed to allow global programming.

When the set types are mixed, the installer must assign ACD keys within keys two to six on the Mitel telephone. If more than five keys are assigned, then the template is invalid, and the agent cannot login until the template is programmed correctly.

Assigning ACD keys in this way is especially important because Mitel ACD positions will be unable to log in when ACD keys are assigned to keys not physically present on these sets.

NOTE: Applying an ACD template to 53xx series phones will not override keys that have been programmed using the Superkey or Applications key.

ACD Key Configurations

Up to three different function key configurations may be programmed for each ACD position: agent, supervisor, and senior supervisor (for a total of nine key templates). In each COS, however, only one template for one position type can be enabled.

The template assigned to a user is portable to any Mitel 5220 IP, 5324 IP, 5312 IP, 5224 IP, 5212 IP, 5215 IP, 5020 IP, 5010 IP, 5207 IP, SUPERSET 4015, SUPERSET 4025, SUPERSET 4125, SUPERSET 410, SUPERSET 420, or SpectraLink NetLink Wireless telephone (with the exception of the 5212 IP, 5215 IP, 5010 IP, SUPERSET 4015, SUPERSET 410, or SpectraLink Netlink telephones which cannot be used by a supervisor or senior supervisor). When the user logs out, the previous template is restored on the set.

Note: Line appearance keys assigned in Form 09 (Desktop Device Assignments) have priority over ACD feature keys when a position logs in.

Assigning Non-ACD Keys

Non-ACD feature keys and speed dial numbers can also be programmed in this form. Pressing the NON-ACD KEYS softkey provides access to a sub-level of softkeys through which the user can select non-ACD feature keys for the template.

Note: Line appearance keys assigned in CDE Form 09 (Desktop Device Assignments) have priority over non-ACD feature keys when a position logs in.

ACD Keys Template Display

When selected during CDE, the ACD Keys Template form defaults to display the first Agent Keys template. The title line contains the type of template and the template number.

Figure: Agent Keys Template (CDE Form 38)

Several fields on the Keys Template form contain customer-defined data. The ACD Keys Template Fields table below defines the contents of each field.

Table: ACD Keys Template Fields

Field	Meaning
POSITION	Selectable field defining one of the following positions: Agent, Supervisor, or Senior Supervisor.
TEMPLATE NUMBER	Selectable field containing one digit in the range of 1 through 3, indicating the number of the template being programmed.

Table: ACD Keys Template Fields

Field	Meaning
KEY	A system generated field listing set key numbers in the range of 02 to 15.
TYPE	

A programmable field defining the function of the set keys. Available options are dependent upon the position selected:

Agent - Speed Dial, Make Busy and Queue Status. **Supervisor** - Speed Dial, Queue Status, Agent Status and Shift. **Senior Supervisor** - Speed Dial, Queue Status and Shift.	
SPEED DIAL NUMBER	A programmable field used to save frequently dialed numbers. This field can also be used to program feature access codes. A maximum of 25 digits may be entered in this field. This field can only be accessed if the “Type” field for that line is “Speed Dial”. In order to transfer calls to voice mail, you can program * 9 for a speed dial pause. Pauses are required to allow the voice mail to answer.
PRIVATE	A programmable field used to control the display of speed dial numbers on the set. When privacy is enabled the set does not display the speed call entry. If privacy is not requested, the speed dial entry appears on the set during dialing, or when a display key is requested.

COS Define

The COS options shown in the COS Define form in this Figure apply to the ACD feature. These ACD options are explained below. The COS options for ACD are described below in the COS Options Fields Table below.

Figure: COS Define Form (CDE Form 03)

Table: COS Options Fields

Field	Meaning
Recording Failure To Hang up Timer (COS option 404)	The values 1 through 255 are valid. Default is 30 seconds. Set this timer to a value greater than the cleardown time of the recording groups whose members have this COS. Assigning this COS timer to a recording group ensures that, in the case of the system hanging up on a RAD, the RAD eventually goes on-hook in the specified time. If it does not, the RAD is taken out of service.
Telephone - Auto Answer (COS	Set to ENABLED if the agent set is to be placed in Auto Answer mode

Table: COS Options Fields	
Field	**Meaning**
option 600)	immediately upon login. Set to DISABLED if agent is not using Auto Answer or if the agent is to manually place set in Auto Answer mode after login. Default is DISABLED.
Telephone - Headset Operation (COS option 612)	Set to ENABLED if the agent's Mitel telephone is to be used with a headset rather than a handset. Set to DISABLED if the telephone is to be used only with a handset. Default is DISABLED.
ACD - Template (Agent, Supervisor, and Senior Supervisor) (COS options 650 through 652)	Enter the template number of the appropriate ACD Keys Template Form. Three template configurations are available for each ACD position. By default they are disabled, but appear in the enabled list. All three COS options are mutually exclusive. For example, if an agent is selected, the supervisor or senior supervisor templates cannot be selected from the same COS.
ACD - Agent Always Auto-Answer (COS option 653)	When enabled in the agent COS, this option causes Auto-Answer to be activated as soon as the agent logs on. Default is DISABLED.
ACD - Display Path Always (COS option 654)	When enabled, the ACD path name is displayed on the Mitel telephone for the duration of the call. The ACD path name is programmed in Form 41. If disabled, the ACD access code is presented. Default is DISABLED.
ACD - Allow Continuous Monitor of Agent (COS option 655)	When enabled in the supervisory set COS, this option allows a supervisory set to monitor an agent set continuously, regardless of the call state. Default is DISABLED.
Loop Start Trunk to ACD path connect (COS option 812)	When enabled, this option allows a loop start trunk to connect to an ACD path.

ACD Supervisors

The ACD Supervisors form shown in this Figure records the ID number, name, and COS of each ACD Supervisor. System option 104, “Maximum ACD Agents” must be enabled before this, or any ACD related CDE form, can be accessed.

The senior supervisor's name is carried to the set when the supervisor logs on. The ID codes assigned in this form are used in the log on procedure and may also be used as an access code to call the user.

Figure: ACD Supervisor Form (CDE Form 40)

The ACD Supervisor form contains three programmable fields which must be defined by the customer. The ACD Supervisor Form Fields table below defines each field.

Table: ACD Supervisor Form Fields	
Field	**Meaning**
ID Code	A programmable field used to record senior supervisor identification numbers. Their IDs are used when logging on and as an access code to call the user. The IDs are 1-5 digit entries, and must not conflict with other access codes already assigned in the system. The form is sorted numerically by ID.
Name	An optional programmable field used to record the supervisor's name. This name

Table: ACD Supervisor Form Fields

Field	Meaning
	appears on any SUPERSET 420 telephone that the user logs onto. The same conditions that apply to supervisor names apply to set names.
COS	A 2-digit field specifying the Class Of Service number of this senior supervisor. The range is 1 through 50.

Pressing the EXPAND softkey displays the subform shown in this Figure. This form lists all groups reporting to the supervisor.

ACD Supervisor Subform

This subform is sorted by SUPER ID. If the entry has no supervisor assigned, the field is sorted by group number. When more than one group is assigned to the same supervisor, the entries reporting to the supervisor are sorted by group number.

Figure: ACD Supervisor Subform (CDE Subform 40)

The ACD Supervisor Subform Fields table below defines the fields of the ACD Supervisor Subform.

Table: ACD Supervisor Subform Fields

Field	Meaning
Heading	A system generated field that lists the senior supervisor's name and ID. The name defaults to "SENIOR" if no name was programmed in the ACD Supervisor Form.
GRPS OF	A programmable field used to define groups reporting to the senior supervisor. If the selected group reports directly to the senior supervisor, pressing the ENTER key causes the display NO SUPER to appear in the SUPER ID field. The remaining two fields are blank.
SUPER ID	A programmable field used to assign a supervisor to the agent group.
SUPER NAME	An optional field of up to ten characters. The entry cannot begin with an asterisk (*). The same conditions that apply to supervisor names apply to set names.
COS	A 2-digit field that specifies the Class of Service number of this supervisor. The range is 1 through 50.

ACD Paths

The ACD path is the major element of the ACD structure. Each path contains all the information necessary to carry an incoming call through the ACD system. Paths specify the resources used, the order in which the resources are encountered, and the timing of the steps through the path. Up to 99 paths may be assigned in the system. This Figure contains the ACD Path form. The ACD Path Form Fields table describes the fields on the ACD Path form.

ACD Path Programming

For a path to function, the primary agent group and the path access code must be programmed. Few restrictions exist on path programming. An agent group, for example, could be the primary group of three paths and the first overflow group of two other paths. The same applies to recording groups and the alternate music sources. The result is that a path can be custom tailored to the call being handled.

Each path is given a priority ranging from 1 through 99 (priority 1 being the highest priority). ACD calls entering a high priority path are serviced before calls that entered a path with a lower priority. This feature improves cost efficiency by routing to higher priority paths those trunks that incur additional expenses: for example, long distance calls or WATS.

All devices have unrestricted access to ACD paths except Loop Start CO trunks and Loop Start DISA trunks (if located on a CO Trunk card). Loop start trunks can be prevented from entering ACD through the use of the “Loop Start Trunk to ACD Path Connect” option (COS option 812). By default, this option is disabled, so by default loop start trunks may not enter ACD.

Figure: ACD Path Form (CDE Form 41)

Table: ACD Path Form Fields

Field	Meaning
ACD Path	Header field identifying the ACD path by name and number.
ACD Path Number	Programmable field containing a one or two digit number in the range of 1 through 99.
ACD Path Name	Programmable field identifying the path by name. This field cannot be accessed until the path has been assigned an access code and a primary agent group. The path name can be up to 8 characters and cannot begin with an asterisk (*).
Access Code for this ACD Path	Programmable field containing the access code for the path. This code can be used as a destination in the Non-Dial-In Trunks form (CDE Form 14) and the Call Rerouting Table (CDE Form 19), as an Automated Attendant defined destination in the Hunt Groups form (CDE Form 17), as an interflow point in another path definition, and as a call forwarding point for a Mitel telephone or station. This code allows the ACD system to tie in to existing routing schemes such as the DID trunk routing points. The connection checking between a device and ACD paths only prevents access to ACD paths.
Primary ACD Agent Group	A programmable field containing a one or two digit number in the range of 1 through 50. This entry indicates which group first receives the ACD calls on this path. The agent group must be assigned in the ACD Agent Groups Form before it can be entered in this field.
Delay For Ringback	A programmable field specifying a timer value in the range of 00:01 through 54:00. The default value is 3 seconds (00:03). All other timers connected with the ACD functions start after the Delay for Ringback timer has expired.
Recording 1: Start Time	A programmable field specifying when Recording 1 begins relative to when the caller enters the ACD system. This timer is initiated after the Delay For Ringback timer has expired. The range of the Recording 1 Start Time is 00:00 through 54:00.A 3-second minimum delay exists between recordings. During this time, the caller listens to the system or to the alternate music source.
Recording 1: Access	This programmable field is mandatory if a Recording Start Time has been specified. The access code entered in this field is defined in the Hunt Groups CDE

Table: ACD Path Form Fields

Field	Meaning
Code	form. The default value is: no recordings.
Recording 1: Music Source Following	A programmable field that directs the call to the ONS port supplying music after listening to the recording. The default music source is the system music, if provided, or silence. The entry in this field cannot have keyline or multi-call line appearances. The music source is a permanently off-hook ONS port that connects the caller in a listen-only conference. An alternate music source must be an FCC Part 68 and Industry Canada approved Recorded Announcement Device that is connected either to an ONS circuit, or to another source that is connected to an ONS circuit through an FCC Part 68 and Industry Canada approved “voice coupler” or “voice connecting arrangement”.
Recording 2 through 4	The recording fields must be edited in sequence. For example, Recording 3 Start Time cannot be edited unless Recording 1 and Recording 2 are both assigned.
Overflow 1 Agent Group	A programmable field specifying the ID of the agent group that receives overflow calls. ACD calls that overflow to this group also retain their position for the Primary Agent Group. The default value for this field is no overflow.
Overflow 2 Agent Group	A programmable field specifying the ID of the second agent group that receives overflow calls. Callers waiting for this group remain in the queue for the primary and first overflow groups. An Overflow 1 agent group must be assigned before the Overflow 2 field can be accessed. The default value is no overflow.
Overflow 3 Agent Group	A programmable field specifying the ID of the third agent group that receives overflow calls. Callers waiting for this group remain in the queue for the primary, first, and second overflow groups. The default value is NO. Overflow 1 and Overflow 2 agent groups must be assigned before the Overflow 3 field can be accessed. The default value is no overflow.

Interflow Enabled	Entering YES in this field allows the waiting ACD call to exit ACD and call a specified number. If this field is enabled, the call interflows to the Interflow Point Access Code. Default for the Interflow Enabled field is NO.
Interflow Timeout	

A programmable field that specifies when the waiting ACD call should leave the ACD system and be routed to the interflow point. The timer range is 00:01 through 54:00. The default value of this field is the maximum time of 54 minutes. Programming a value in this field ensures that unanswered calls do not remain in the system after the caller disconnects. This can occur with loop start trunks if the CO fails to send a disconnect to the SX-200 ICP.

This timer also ensures that all calls are handled within a maximum time interval. Call handling may involve routing the caller elsewhere or dropping the call.	
Interflow Point Access Code	A programmable field that contains the directory number of the interflow device. Valid interflow points are LDNs, stations, sets, consoles, night bells, ACD paths, station/set hunt groups, UCD agent hunt groups, Automated Attendant hunt groups, system speed call numbers or DROP CALL. If an access code is programmed, the DROP CALL softkey is provided.
Allow Overflow to Interflow Point before Timeout	Entering YES in this field allows an overflow to the interflow point before the Interflow Timeout.
Priority	A programmable field used to set the priority of the ACD Path. Priority range is 1 through 99 (1 is the highest priority). The field default is 99. Calls waiting for an Agent Group are serviced according to the path priority. Expensive trunks should be routed to a path with a high priority.
Service Time	

A programmable field used to establish a standard time to answer. The supervisor can use the Service Time to monitor the performance of agents answering calls on the path. The service time is programmable in the range 00:00 through 54:00.

The path level of service is calculated by comparing the actual time to answer with the programmed service level.	
Tenant	When a tenant number is assigned to the ACD Path, DID and TIE trunks (which dial into the ACD path directly) follow the routing for this tenant as defined in Form 19 - Call Rerouting Table. Enter a valid number (1 to 25). The default is blank (no tenant).

Immediately Interflow when no Agents Logged In	When set to YES, and "Interflow Enabled" is set to YES, then any callers dialing in to an ACD path will interflow immediately when no agents are logged in. This interflow takes place regardless of the status of the "Interflow Timeout," or the option "Allow Overflow to Interflow Point Before Timeout," or the "Interflow Point Access Code" having a value of DROP CALL. The default is NO (immediate interflow is not desired).

Call Routing

Dial-in trunks to ACD Paths may also be rerouted as defined in this form. The ACD Path is assigned a tenant number in Form 41. This Figure contains the Call Rerouting table. The Call Rerouting Options table describes the fields applicable to ACD.

Figure: Call Rerouting Table (CDE Form 19)

Field Descriptions

The header line displays the tenant group number being programmed.

DAY: This field designates a directory number for each type of call in Day service mode.

N1: This field specifies the extension to which calls are routed during Night1 Service Mode. If this field is blank, the call reroutes to the extension specified in the DAY field.

N2: This field specifies the extension number to which calls are routed during Night2 Service Mode. If this field is blank, the call does not reroute.

Table: Call Rerouting Options

Field	Meaning
DID Routing for Calls into this Tenant	All DID calls normally routed to extensions are routed here to allow screening of DID calls. This rerouting option is based on the destination tenant.
Dial-In Tie Routing for Calls into this Tenant	All Dial-In Tie calls normally routed to extensions are routed here to allow screening of Dial-In Tie calls. This rerouting option is based on the destination tenant.

Softkeys

TENANT: This softkey selects a tenant group. Pressing the TENANT softkey displays the ENTER TENANT GROUP NUM: prompt on the command line. The selection is completed by entering a valid number (1 to 25). The system displays the selected tenant group number on the header line.

TENANT NAME: Allows a name to be programmed for the selected tenant group. The name may have a maximum of eight characters.

The standard softkeys **CANCEL**, **DELETE**, **ENTER**, **TOP**, **BOTTOM**, and **QUIT** are also provided.

ACD Recording Hunt Groups

Recorded announcements are given to ACD callers while they wait for an idle agent. ACD callers entering the SX-200 ICP System on a path hear ringback until the “Delay for Ringback” timer specified in the Path form has expired. At this point, the system attempts to provide an agent. If all agents are busy and the caller must wait, the call is routed to a recording.

The recordings are provided by hunt groups of regular ONS ports. When a RAD answers, the system collects all callers waiting for the RAD and creates a listen-only conference. The hunting type of the group determines how callers select an idle RAD. When all recordings in a group are busy, the caller camps on to the recording group and waits for a free recording. All calls which are waiting are brought into a listen-only conference with the first available recording.

At the end of the recorded message, the callers are routed to the music source defined in the Path form. If no alternate music source is defined, the caller is given system music or silence.

The ONS ports cannot have keyline or multi-call line appearances. After changing the Hunt Groups form to a Recording Hunt Group, the Hunt Groups form changes to that shown in this Figure. Fields described below but not shown in this Figure are accessed through the OPTIONS softkey. The Hunt Groups Form Fields table below describes the fields on the Hunt Groups form.

Figure: Hunt Groups Form (CDE Form 17)

Table: Hunt Groups Form Fields

Field	Meaning
Hunt Group Number	A programmable two-digit field in the range of 1 through 99. ACD has only 99 hunt groups.
Access Code	A programmable field that contains the access code for the hunt group. This code must be a unique number that does not conflict with the system numbering plan. The entry in this field must be copied to the ACD Path form as an access code to the recordings.
Type Of Search	

Enter CIRCULAR or TERMINAL to specify the desired hunting method:

CIRCULAR: Hunting begins at the extension following the extension to which the previous call was completed, and hunts through all extensions in the hunt group in the programmed sequence. **TERMINAL:** Hunting always starts at the first extension programmed in the hunt group and terminates at the first idle extension. This Figure shows an entry of TERM in this field.	
Group Type	Softkey that allows the user to specify a RECORDING group type. This Figure shows the entry RECORD in this field on the form.
Extension Number	Extension number of the ONS port(s) connected to a recording. The maximum number of entries in this field is 50.

Bay/Slot/Circuit	A system generated field that is displayed after entering the extension number of the ONS port.
Message Length	A programmable field that defines the length of the recorded announcement. The range is one second to four minutes. Default entry is ten seconds. This timer value determines whether the SX-200 ICP or the RAD ends the recording. If the RAD is to hang up, set the Message Length at least three seconds longer than the actual recording length. This field allows for timing and message delays. In addition, the flash hook time programmed into the system detects how soon the system sees the RAD going on hook after the RAD hangs up.
Name	A programmable field that identifies the hunt group by name. This field cannot be accessed before the Hunt Group Access code has been assigned.

ACD Programming Error Messages

The error messages that may appear during programming of the ACD forms are listed, and explained, in the following table:

Table: Programming Error Messages

Error Message	Meaning
ACD agent group XX already assigned to a supervisor	The ACD agent group which has been edited or inserted into the sub-form is already programmed under some other supervisor.
Agent group XX has already been assigned elsewhere in this path	The selected overflow agent group or primary ACD agent group is a duplicate of an agent group already specified in this path.
ACD agent group XX not assigned to a supervisor	ACD group XX, requested by the FIND GROUP key, cannot be displayed because it has not yet been assigned to a supervisor.
ACD groups under XXXXX must first be deleted	This senior supervisor cannot be deleted from the first-level form by the DELETE key because there are ACD groups defined under this senior supervisor.
ACD keys template for this COS is invalid or disabled	An attempt was made to change the COS of a logged in ACD position. The new COS has the ACD keys template disabled or assigned to another ACD position.
ACD Monitor Print in progress	System found at least one printer - job has been started.
ACD position active - Cannot make template change	An attempt was made to disable or change a template COS option while the ACD position is logged in.
AGENT STATUS not	The user is attempting to program an agent status key when there is more

Table: Programming Error Messages	
Error Message	**Meaning**
allowed when multiple QUEUE STATUS keys are programmed	than one queue status key programmed for the template. Only one is allowed if an agent status key is present.
Agent XXXXX does not exist	The ID entered for the FIND ID key does not exist in the database.
Agent XXXXX is on line and cannot be deleted	The DELETE key cannot be used on an agent that is on line.
Alternate music sources cannot have keyline or multi-call line appearances.	Music sources cannot be line appearances.
An ACD position is currently logged in at the Mitel telephone	The user is attempting to delete a Mitel telephone where an ACD position is logged in. The ACD position must first log out.
An agent's set's prime line cannot have any appearances on other sets	An ACD agent is logged in at the given Mitel telephone; therefore, line appearances of that Mitel telephone cannot be programmed into the data base.
Attempting to remove an Agent Group that has calls waiting	This message is displayed if the Primary Agent group or the Overflow agent groups have calls waiting from the path, and the user is attempting to change or delete the agent group or the path.
Attempting to remove a Music Source that is currently in use	Someone is listening to the music source so it cannot be removed. This message can occur when attempting to change or delete a music source, deleting the recorded music source, or deleting the path.
Attempting to remove a Recording that is currently in use	Someone is using the Recording hunt group so it cannot be removed. This message can occur when attempting to change or delete the recording or when deleting the path.
Beep (audible sound from terminal)	The speed dial number entered is too long (maximum 25 digits).
Beeping (repeating audible sounds from terminal)	Too many digits or characters have been entered in the selected field.
Cannot change agent information at time of reassignment	The user attempted to reassign an agent to the current ACD Group AND change agent name or COS at the same time. These two steps (reassign and change) must be done separately.
Cannot change COS of an ACD position whose set is currently in use	The ACD position being updated is logged in and is using the set. To change the COS, the ACD position must be logged off or logged in to an idle set.
Cannot delete last agent while callers are waiting on ACD group XX	The user attempted to delete the last agent from ACD group XX, and thereby delete group XX itself. ACD calls waiting for group XX, so the deletion cannot be permitted at this time.
Cannot disable option when ACD programming is present	The user is attempting to disable System Option 104 (Maximum ACD Agents) when ACD programming is present in one or all of CDE Forms 38, 39, 40 and 41.

Table: Programming Error Messages

Error Message	Meaning
Checking status of printer(s)	The system is checking the status of the printer(s) - during this time the CANCEL softkey may be pressed to return the user to the previous level.
Delete ACD Group XX from ACD SUPERVISORS form before deleting last agent	The user attempted to delete the last agent from group XX, which would delete group XX itself. Group XX is referenced in the ACD SUPERVISORS form, so it must be deleted from that form first. Then the user is allowed to delete the last agent, which deletes the group.
Delete ACD Group XX from ACD PATH YY before deleting last agent	The user attempted to delete the last agent from group XX, which would delete group XX itself. Group XX is referenced in the ACD PATH form, for path number YY, so it must be deleted from that form first. Then the user is allowed to delete the last agent, which deletes the group.
Extension XXXXX can't have a key template and BLF module at the same time	A key template cannot be enabled in the COS because the extension has an associated PKM. The two features are mutually exclusive.
First Status Threshold must be start time of previous recording	Adjust the interflow timeout as indicated.
Form access disallowed, enable Automatic Call Distribution System Option	The user is attempting to program an ACD CDE form when ACD is not enabled in the system. System Option 104 (Maximum ACD Agents) must be enabled.
Invalid Interflow Point	The access code entered for the Interflow point is non-existent or illegal. Legal points are: LDNs, stations, sets, consoles, ACD paths, station/set hunt groups, UCD agent hunt groups, nightbells, and system speedcall numbers.
Key number XX has MAKE BUSY programmed	Each agent template can only have one MAKE BUSY key programmed.
Key number XX has QUEUE STATUS programmed	An agent keys template can have only one QUEUE STATUS key programmed. A supervisor keys template can only have one QUEUE STATUS key programmed if there is already one AGENT STATUS key programmed.
Key number XX has SHIFT programmed	Each template can have only one SHIFT key programmed.
Maximum Afterwork Timer is 15:00	The user entered a time that is out of range.
Maximum Time is 54:00	The user entered a time that is out of range for Start Time or for one of the thresholds.
Multiple QUEUE STATUS keys not allowed with AGENT STATUS keys	The user is attempting to program more than one queue status key for a template that has an agent status key programmed. Only one is allowed if an agent status key is present.
Must delete all appearances of XX from CALL REROUTING TABLE	This ACD path cannot be deleted because its access code is defined in the CALL REROUTING table (Form 19).
Must delete all appearances	This ACD path cannot be deleted because its access code is defined in the

Table: Programming Error Messages

Error Message	Meaning
of XX from answer points in NON-DIAL-IN trunks	NON-DIAL-IN TRUNKS form (Form 14).
No printer(s) defined for ACD	No printers for ACD have been programmed in CDE Form 34.
ONS Port access code XXXXX does not exist	The access code entered for the Alternate Music Source Between Recordings, or Music Source Following a Recording is non-existent or illegal.
Option XXX conflicts with this option	Only one of the ACD COS options (ACD agent, ACD supervisor, ACD senior supervisor) can be enabled in the same COS.
Option 104 must be enabled	Option 104 (Maximum ACD Agents) must be set to a valid number before this option can be enabled.
Option 44 must be disabled	The user is attempting to disable Option 104 (Maximum ACD Agents)
Printer(s) busy, try later	Printers are programmed, but the system cannot find at least one that is idle.
Recording access code XXXXX does not exist	The access code entered for a recording is non-existent or illegal. The code must be for a recording hunt group.
Start time of a recording must be < Interflow Timeout	Adjust the start time as indicated.
Start time of a recording must be < start time of the next recording	Adjust the recording start times as indicated.
Start time of recording must be > start time of the previous recording	Adjust the recording start times as indicated.
Supervisor XXXX does not exist	A supervisor with ID XXXX cannot be displayed, as requested via the FIND SUPER key, because this access code has not been assigned to a supervisor.
Template number must be in range (1-3)	The template number is out of range.
The access code XXXX is already assigned	

In the paths form the error is: The access code entered for this ACD Path is already assigned elsewhere in the database. In the supervisor form the error is: The ID code which has been inserted into the main form or the subform already exists in the database. (It could exist as another supervisor, senior supervisor, agent, or any other device.)

In the subform this error occurs if the user specified a different name or COS than was previously entered for this supervisor.	
The agent group XX does not exist	The specified agent group does not exist in the data base.
The maximum ACD positions allowed are already assigned	The maximum number of ACD positions, including all position types, is 999.

The Mitel telephone has a BLF module and cannot have a COS with a key template	The COS being assigned to the set has a key template enabled.
The value XX is outside valid range for ACD agent group (1-50)	The given value is outside the valid range.
XXXXX is an ACD agent ID	The specified supervisor ID from FIND SUPER is an ACD agent.
XXXXX is an ACD supervisor	The user has entered a supervisor ID for FIND AGENT.

Planner Sheets

This section contains additional blank copies of the agent group planner and the path planner sheets. Each sheet contains instructions on its use. For additional information, see Configuring an ACD System.

ACD TELEMARKETER Agent Group Planner

ACD TELEMARKETER Path Planner

Automated Attendant

Introduction

This document describes the features, operation and programming of the Automated Attendant feature for the SX-200 ICP. The Automated Attendant Application Package is a software option that must be purchased and is incompatible with the Automated Attendant feature of the SX-200 ICP embedded voice mail application.

Refer to the following sections for additional information:

- Standard Features
- ACD TELEMARKETER Application Package.

Functional Description

This section describes the basic operation of the Automated Attendant feature and the hardware and software components that make up the feature package. Included are overviews of the feature and descriptions of Auto-Attendant groups, handling of illegal calls, default destinations, DTMF receiver requirements, and resource allocation.

Overview

The Automated Attendant feature, as shown in this Figure, directs incoming calls to a recorded announcement device (RAD). The RAD message instructs callers that they can access a directory number on the system by dialing over the message. Callers choosing not to dial during the message are routed to a default answering point, such as an attendant, when the message is finished.

Figure: Automated Attendant Call Handling

Auto-Attendant Groups

The Automated Attendant feature introduces an additional hunt group type called an Auto-Attendant group. This group is similar to the recording groups used in the ACD TELEMARKETER feature. The Automated Attendant feature is accessed by either rerouting or dialing into an Auto-Attendant group. The Auto-Attendant group can contain only single-line ports (that is, ONS), and has the main features of any hunt group such as:

- hunt group number
- access code
- circular or terminal hunting.

Through customer data entry (CDE), Auto-Attendant groups also have several programmable options that include:

- name
- message length
- default destination
- prefix digits
- dialing enabled
- wait for resources time.

The Auto-Attendant group access code can be dialed by any device. This access code can be programmed as a destination in the following CDE forms:

- Form 19, Call Rerouting
- Form 14, Non Dial-In Trunks
- Form 17, Hunt Groups (hunt group overflow)
- Form 41, ACD Path (path interflow).

The Auto-Attendant group access code can also be used as a call forwarding point. The group cannot be used as a normal recording group for Uniform Call Distribution (UCD), ACD and/or Automatic Wakeup.

Basic Operation

When an internal or external caller reaches an Auto-Attendant group, the system hunts for an available RAD, connects the caller to the RAD, and connects a DTMF receiver to retrieve digits dialed by the caller. If the caller dials a number, the caller is routed to that number. If the caller does not dial a number, the caller is routed to a default answering point assigned during CDE. The following subsections describe basic call handling for calls to the Automated Attendant.

Note: The system Automated Attendant cannot detect DTMF tones for calls transmitted over IP trunks. As a result, the system will not respond to key presses entered by remote callers.

Digit Handling

The system assigns one DTMF receiver to each caller while the caller is listening to the recording. The recording is terminated as soon as the first digit is received from the caller. After dialing a valid number, the caller is routed to that number.

Conflict Dialing

When the caller dials, conflict dialing is in effect, and the normal 15-second inter-digit timeout applies. The recording is terminated when the conflict timer starts. The user listens to silence while the conflict timer is running. For additional information on conflict dialing and inter-digit timeout, refer to Conflict Dialing in the Program Features section.

Valid Destinations

The valid destinations available to a caller depend upon what the originating device (trunk or telephone) is allowed to dial during a normal operation. All the usual system dialing features apply, such as:

- DID/TIE rerouting on no answer
- DID and DND interactions
- Device/tenant interconnection restrictions
- Class of Service (COS) options for Abbreviated Dial.

Dialing capability ends as soon as the recording ends. Once a destination is attempted, the automated feature ends, and the regular call features such as campon are available.

CO Trunk Dialing

The Automated Attendant feature allows CO trunks to access several features that are normally inaccessible. A CO trunk can dial:

- account codes
- abbreviated dial numbers
- stations
- sets
- consoles
- LDNs
- night bells
- modem pools
- ACD paths
- ACD positions.

Automated Attendant also affects ARS toll control: a CO trunk can only access ARS through a system abbreviated dial number.

Prefix Digits

Each Auto-Attendant group can be programmed in CDE with a string of prefix digits. The prefix can contain up to 4 digits and is inserted in front of the digits dialed by the caller. The prefix allows the caller to dial a single digit and be routed to devices that have normal multi-digit extension numbers. The prefix is only inserted if the caller dials at least one digit.

Typical Applications

The prefix can be used to

- provide single digit menus
- reduce digit dialing
- restrict dialing to numbers that start with the prefix
- provide entry into other features that require digits (such as ARS and feature access codes).

Prefix Processing

Processing of prefix digits starts only after the first digit is dialed. If the prefix itself is a valid access code, the caller will be considered to have dialed the access code, and the system ignores the first digit dialed by the caller.

If the prefix contains a feature access code, control of the call is given to the feature after the digits are processed. If the feature returns dial tone to the caller and other digits have to be processed, the remaining digits are ignored because the feature clears the digits from the receiver.

If, for example, the system is programmed with * as the account code access code and 123 as a valid account code, setting the prefix to *123 causes the system to process the prefix as a complete account code and ignore the first digit dialed by the caller. In this case, the system returns dial tone after the processing is complete.

This situation can be avoided by ensuring that the prefix does not include enough digits to run a feature to completion. Again, assuming * is the account code access code, using a prefix of * prevents dropping any digits. The first digit dialed by the caller is the start of the account code.

Illegal Number Handling

If the dialed number is illegal, the system checks for illegal number routing using the tenant group of the first member programmed in the Automated Attendant group. If the routing point has illegal number routing programmed, the system redirects the caller to the routing point. The routing point can be another Automated Attendant group. If no illegal number routing is programmed, the caller is given reorder tone and is eventually disconnected.

Examples of illegal number conditions include:

- device interconnection
- tenant interconnection
- not valid for caller type
- feature restricted.

Each Auto-Attendant group can be assigned different rerouting points by assigning a different tenant group number to the first member of each Auto-Attendant group.

Vacant Number Routing

Handling callers that dial a vacant number, such as an unassigned access code, is similar to the illegal number handling described above. In the case of a vacant number, vacant number routing rather than illegal number routing is checked.

Front End Recording

Front end recordings present a message to the caller as soon as the call enters the system. Auto-Attendant groups can provide this feature by having dialing disabled during the recording. This provides a simple front-end recording without assigning a DTMF receiver. Digits dialed by the caller are ignored and the prefix digits have no affect. Calls are routed to the default destination as normal.

Default Destination

When a recording ends, callers who have not dialed at least one digit during the recording are routed to the default destination for the group. Failing to program a default destination means that when the recording ends, the caller is given reorder tone and eventually disconnected.

Default Answering Points

Valid default points for the Automated Attendant feature are:

- console
- LDN
- night bell
- station
- Mitel telephone
- logical line
- ACD path
- hunt group
- ACD positions (agent, supervisor and senior supervisor)
- system Abbreviated Dial
- Auto-Attendant hunt group.

Routing to Default Destination

When the caller is routed to the default destination, the system handles the call as a call reroute. Device and tenant interconnection is bypassed, DND is ignored, and the caller automatically camps on to the destination if it is busy. Refer to Call Rerouting in the Program Features section.

Fax Tone Detection

When System Option 99 is enabled, the system can recognize Fax tone on incoming calls to the Automated Attendant and route incoming Fax calls to the Fax destination. System Option 106, Automated Attendant, must be enabled first.

RAD Operation

RAD operation is similar to the RADs in the ACD TELEMARKETER applications package. The Automated Attendant feature uses the Auto-Attendant group as an enhanced recording group so that the basic recording group features apply. Refer to Recording Support, in the Program Features section for information about RADs and recording groups.

The length of the recorded message, either programmed in CDE or in the RAD itself, should be greater than the actual message to create a silent period at the end of the recording. The silent period results in a delay between the end of the message and the rerouting to the default destination to give the caller time to think about the message and to start dialing. Complex instructions in the message may require a longer delay at the end of the message.

Resource Allocation

Each call entering the Automated Attendant feature uses two primary resources: a RAD and a DTMF receiver. Usage differs between the two resources as explained in the following paragraphs:

Each time a RAD becomes free, an unlimited amount of that resource becomes available because of the unlimited number of listen-only conferees that can be serviced by that one RAD.

When a receiver becomes free, however, only one piece of that resource becomes available because only one caller can use the receiver at a time. Receiver availability becomes the primary resource limitation for the Automated Attendant feature.

DTMF Receiver Requirements

In addition to the number of receivers needed for normal PBX operation, the number of extra receivers needed for the Automated Attendant feature is approximately:

$$\frac{\text{Average number of calls per hour}}{\text{Maximum \# of messages given per hour}}$$

where:

maximum # of messages given per hour = 3600/message length (including setup and clear down time) in seconds (for groups with one RAD).

A limited number of messages can be played in one hour. Therefore, to service all callers, each message must play to a number of callers. This calculation is an estimate for a single group in the system. Additional groups require additional receivers.

As an example for a single group, assume that 100 callers per hour are accessing the group and that the message length is 20 seconds. This means 180 messages can be played per hour and 100/180 or at least one receiver is needed to service all of the callers in that hour. Round the result of this calculation up to the next whole number.

Note: The minimum quantity of receivers must be equal to the number of RADs assigned to the Automated Attendant feature so that RADs can operate concurrently.

Receiver Allocation Control

The user can place limits on the number of receivers available to the Automated Attendant feature (across all Auto-Attendant groups). Without this limit, the Automated Attendant feature can potentially use up all receivers in the system and block dialing for extended periods.

The limit is specified by programming System Option 59, Receivers Reserved for Non-Auto-Attendant Use. The Automated Attendant feature uses as many receivers as possible but it will always leave at least this number of receivers available for the rest of the system.

Receivers going out of service (for diagnostics, etc.) have no effect on this number. Reducing the number of available receivers removes receivers from the Automated Attendant feature first.

When the limit is specified, the system does not check to ensure that the number of receivers present in the system is greater than the number of receivers programmed as a limit. If the system contains fewer receivers than the limit, the Automated Attendant feature is unable to access any receivers. The system handles this as a “no receivers available” condition.

Setting the number of receivers to more than the number in the system results in all calls that are directed at Auto-Attendant groups ending up at the default destination.

Busy Recordings

If a call arrives at the Automated Attendant when all recordings are busy or unavailable, the caller is camped on to the group to wait for a recording. The wait time is programmable for each group through CDE. Unless all RADs fail, the caller wait time should be no longer than the RAD cycle time. Normal camp-on audio is returned to callers.

When a recording or DSP receiver becomes free, the system rings the RAD. When the RAD answers, the system sets up a listen only conference for all callers camped on to the Auto-Attendant group. The callers are retrieved using the normal campon priority scheme. Callers will be retrieved until there are no more waiting callers or until all available DTMF receivers in the system have been allocated (DTMF receiver allocation is subject to the receiver usage limits for the Automated Attendant feature in the system). Once all receivers are used up, the remaining callers continue to wait for resources to become free.

Busy Receivers

If no receivers are available when a RAD becomes free, the system camps the caller on to the Auto-Attendant group to wait until a receiver becomes available. The wait time is programmable through CDE.

Note: Internal callers dialing the group (including dial-in trunks and CO trunks coming from other groups) do not keep their receivers after dialing an Auto-Attendant group. These callers are allocated another receiver based upon the receiver allocation system option.

Busy RAD or Receiver Timeout

The wait for resources timer on the group controls the length of time that a caller is allowed to wait for a RAD or for a receiver to become available. When the wait timeout occurs, the caller immediately calls the default destination without listening to a recording. If the caller is ringing a RAD when the timer expires, the timeout is ignored. Traffic measurement for this group is pegged each time that a recording is skipped because of a waiting timeout. If no default destination is specified, the caller is given reorder tone and eventually is ‘put into suspended state.

Operation

The Automated Attendant feature requires no special operating instructions because calls are routed to the Automated Attendant by the system. This section describes the displays that appear on the sets of internal callers after dialing the Automated Attendant, and when receiving a call directed from the Automated Attendant. This section also describes the interaction between the Automated Attendant feature and other SX-200 system features

Caller Displays

The Mitel display telephones and the attendant console show normal ringback, busy, and talking state displays when calling a hunt group.

Called Party Displays

The display of a party called from an Auto-Attendant group is the normal call processing display for the given caller and called party. No special indication is given to show that the call came from a group, however, either the name or the access code of the group is displayed.

Transfers

Callers are unable to transfer another party to a group while ringing or listening. If a transfer is attempted during ringing or listening, the system will terminate the feature and the caller will be recalled by the held party. Extensions can transfer a party to a busy group, however, the console cannot transfer to a busy Auto-Attendant group.

Calls can be indirectly transferred to groups by calling an extension that has Call Forward No Answer programmed to a group and releasing the held caller to the forwarded extension before the call forward no answer timeout has expired.

Interaction with Other Features

The Automated Attendant feature can affect the operation of certain features in the system. The following subsections outline the interactions. Features are arranged in alphabetical order.

The Automated Attendant feature has no special call handling features built in for compatibility with ACD. A caller using the Automated Attendant feature to access ACD hears the Automated Attendant recording, dials an ACD path, and hears ringback tone before the ACD recordings are started.

The Automated Attendant feature does close off and print the current SMDR buffer before it enters ACD to preserve the Automated Attendant feature information in the SMDR record and to prevent a conflict between ACD and Automated Attendant information in the dialed digits field.

ACD callers using the Automated Attendant feature after an agent answers are handled as normal. In addition, the Automated Attendant feature does not add any digits into the dialed digit buffer to preserve the ACD information.

Account Codes

CO trunks can dial the account code access code. No checks are made to see whether the access is from the Automated Attendant feature.

DID trunks are blocked from accessing the account code feature even through Automated Attendant.

Verified account codes can provide ARS access security. With Auto-Attendant groups, this feature can be used by assigning prefix digits that are the account code access code. The user then dials the verified account code to change dialing privileges. Because CO trunks can access account codes, the Direct to ARS feature is available to CO trunks after the CO trunk user dials an account code.

The prefix for a group can be programmed as the account code access code and may contain the leading account code digits. The user then dials the remaining account code digits.

ARS

A CO trunk cannot normally dial ARS directly and must use one of three ways to access ARS: forwarding, transfer, or ACD Interflow. Because a CO trunk has no Class of Restriction (COR) number, forwarding and transfer use the COR number of the forwarder or transferrer.

For the Automated Attendant feature, as with ACD interflow, no COR number is provided for the CO trunk during dialing; therefore, there is no toll control. CO trunks, however, are unable to dial an ARS digit string directly. From the Automated Attendant feature, a CO trunk can only access ARS through a system abbreviated dial number.

Callbacks

Callbacks to a group are not allowed to Auto-Attendant groups.

Call Duration Display

When a Mitel display telephone answers a trunk routed from an Auto-Attendant group, the call duration display shows the call duration beginning when a non-recording answers the trunk. This method is consistent with the display for trunks from other recording applications (even though for Automated Attendant the time to answer in SMDR is the time to answer by the recording, not the set).

Call Forwarding

When a caller reaches an Auto-Attendant group, the caller's current call forwarding history is cleared to prevent problems with forwarding hop limits when a group is a forwarding destination. The caller can be forwarded again for the maximum number of forwarding steps.

Campon

When all RADs are unavailable, the system camps the caller on to the Auto-Attendant group. All device types except the console can be camped on to the group. Because the console is not permitted to camp on to anything, it is given busy tone and must try dialing the group again.

DID/Dial-in Trunk Busy Rerouting

The DID/Dial-in trunk busy rerouting point is not operational when calling an Auto-Attendant group (the trunk always camps on if the group is busy). The feature is operational when the caller dials from the group.

Direct to ARS

The Direct to ARS feature applies to calls after an account code is successfully dialed from a group (Direct to ARS applies to all devices). An added application is that Analog Networking passes the account code into the network. If a caller dials an account code from a group and then with Direct to ARS goes to an analog network trunk, the digits that the caller dials from the group will be passed into the network.

Recall on Default or Dialed Destination

No recall point is set up by the Automated Attendant feature. Recall on busy and no answer operate as if the feature had not been accessed, this feature acts as if the default or dialed destination had been reached directly. The answer supervision given to the trunk during the ringing has no effect on recall.

Ring Groups

If a member of a ring group has a call camped on and a call is queued to this ring group at the same time, when this member becomes free, the call queued to the ring group will be serviced first. (The member being camped on can answer the waiting call at any time using the Trade/Trade Calls softkey or the Swap/Trade feature key.

The following features are ignored by the system for a member of a ring group if a call is routed to the pilot number of the ring group:

- Auto answer
- Call forwarding (tenant or set-based)
- Intercom Mode

The following features should be avoided for ring group members. Combining ring groups with any of the following features may result in unexpected behavior:

- ONS ring group
- Phone twinning
- Guest Suites
- ACD

Users will <u>not</u> be allowed to place callbacks or camp on to a ring group.

System Abbreviated Dial

Normal system operation prevents CO trunks from accessing system abbreviated dialing except through external call forwarding. With the Automated Attendant feature, CO trunks can dial the system abbreviated dial access code. The CO trunk must have the abbreviated dial access COS option enabled as is the case with other devices.

The system allows callers to access any numbers in the system. If access is given to Automated Attendant callers, the only control available is through toll control for ARS numbers.

If necessary, the prefix feature can be used to restrict access from the Automated Attendant to specific numbers. The prefix feature limits the caller to dialing only a limited set of numbers, such as those beginning with the digits 12.

Tenanting

Tenanting can be used to restrict the dialing ability of callers who use the Automated Attendant feature. With the tenant interconnection table, callers can be put in a special tenant group and be allowed to dial only designated extensions in the system.

SMDR

Incoming SMDR records indicate that the Automated Attendant feature has been used. The SMDR Fields table below defines the significant fields.

Table: SMDR Fields

SMDR Field	Purpose
Called Party	Contains the extension number of the party that answered the caller.
Call Completion Status	Indicates call completion with regard to the group called rather than the destination dialed or routed to from the Automated Attendant feature.
Dialed Digits	Shows the group access code and the destinations dialed by the caller (even if the digits dialed are invalid) or the default destination (if taken). Dialed digits overwrite any information already in this field as a result of analog networking. The access code and digits are written to the field with a single blank between them.
Time to Answer	Indicates the time until answered by the RAD. This duration shows waiting times for receiver and RAD resources, since the trunk will only be answered when the recording is ready to be played. For UCD, ACD, and AAO (Automatic Attendant Overflow - COS option 705), the time to answer is the duration until answer by a device other than the recording.
Call Duration Time	Shows the elapsed time from the time when the RAD answers the call, not when a party in the system answers the call (unlike ACD, UCD, and AAO).

Sample SMDR Record

Below is a sample record. Trunk 001 has called in to group 123 and after 20 seconds is answered by a RAD in the group at 12:32. The trunk then dials 555 and is answered at extension 555. The trunk talks to extension 555 and then hangs up. The total duration of the trunk call was 20 minutes, 12 seconds.

```
06/28 12:32  00:20:12    T001   020  123  555  555
```

If the caller arrives at a group again or after SMDR has already recorded an answer, no special entries are made in the record. The Auto-Attendant group appears as the called destination, third party, etc., as would any hunt group.

Analog Networking

Analog networking information in the SMDR records is overwritten when a trunk that uses analog networking accesses the Automated Attendant feature and dials a number. The Automated Attendant feature information replaces the digits already stored. The SMDR record is not altered if no recording is heard; instead, normal SMDR is done.

Traffic Measurement

The normal traffic measurement statistics for hunt groups are also available for Auto-Attendant groups. As with other recording groups that have listen-only conference, the busy pegs are not very useful because an unlimited number of callers are connected each time a RAD becomes free.

The usage pegs and usage CCS indicate calls handled and RAD usage. An additional peg is present for Auto-Attendant groups to help diagnose receiver shortage problems. The skip peg shows how many failures to get a receiver resulted in a call skipping the recording and routing to the default destination. A non-zero value in this field indicates receiver shortage problems.

EXAMPLE:

HUNT GROUPS:

Number	Peg	Skip	Usage	Busy Peg	Max/Avl
1	6	1	123.00 ccs	1	2/3

Traffic measurement also records receiver usage from the Automated Attendant feature. The 1-, 2-, and 3-second receiver wait pegs are not updated. The receiver usage peg, receiver CCS, and max/avail fields are updated.

Programming

The Automated Attendant Application Package is a software option that must be purchased. The Automated Attendant feature is enabled and controlled through entries in customer data entry (CDE). This section lists the forms related to this feature and describes the entries required on each form.

System Option Form

The following fields on the System Options Form (Form 04) affect the Automated Attendant feature:

System Option 106 - Automated Attendant: This option is a purchasable option that controls the availability of this feature. This option allows programming of Auto-Attendant groups. The option cannot be disabled until all groups are deleted. The option is disabled by default.

System Option 59 - Receivers Reserved For Non-Auto-Attendant Use: A numeric field that defines the number of receivers reserved for normal call processing. Acceptable entries are 1 to 99 or ALL. The value in this field is not restricted by the number of receivers currently in the system.

By default, the value is set to UNKNOWN when the user starts programming the system. The system prevents the user from programming any Auto-Attendant groups until this value is changed to a number from 1 to 99 or ALL. “UNKNOWN” is an initial value only and can never be programmed by the user.

COS Option Form

The COS (Form 03) options that apply are those for members of recording hunt groups. Refer to the Programming section of the Program Features section for details about setting the recording group COS options.

Call Rerouting Form

The UCD Recording routing and Automatic Wakeup routing entries in the Call Rerouting Table (Form 19) cannot include an Auto-Attendant group.

Hunt Group form

The programming of the Auto-Attendant group type in Form 17, Hunt Groups, follows the programming for all other hunt group types. The distinction is in the group type and options. The group type is selected by pressing the GROUP TYPE and then the AUTO ATT softkeys.

If the user has not filled in a value for System Option 59, Receivers Reserved For Non-Auto-Attendant Feature Use, they are not permitted to create an Auto-Attendant group. When the ENTER key is pressed to change the group type, the following error message appears:

System Option 59 must be programmed before creating an Auto-Attendant group.

The user must return to the System Options form and program a value for Option 59.

The group type cannot be changed from AUTO ATT to some other group type unless all of the RADs in the group are either DND, busied-out, or idle. Once the group type is set, the OPTION softkey is used to set up options on the group. The options for Automated Attendant groups only appear when the group type is AUTO ATT.

The fields on the Auto-Attendant group form are described below. Default values are also shown. To change information in the fields, scroll to the desired field and, depending upon the field, either enter the desired value or press the appropriate softkey:

Name: Enter the name string (the same rules apply as for recording groups). Default: No name.

Message Length: Enter the message length time in minutes and seconds (the same rules apply as for recording groups). Default: message length = 10 seconds.

Default Destination: Enter an access code. The access code must already be assigned to a valid destination. Default: No default destination.

Dialing Over Recording: Select the DISABLE or ENABLE softkey that is to appear when the Dialing over Recording field is in the scroll window. Default: Dialing over recording enabled.

Fax Destination: Enter a valid destination to which to route incoming Fax calls.

Prefix Digits: Enter the digit string, containing 0 to 4 digits. Valid digits are 0-9, * and #. Default: No prefix digits.

Wait For Resources: Enter the time to wait in minutes and seconds (00:00 to 54:00). Default: Wait for resources = 1 minute.

If the group type is changed, all of the above information is deleted if the information no longer applies to the new group type. “Name” is never lost and “Message Length” is not lost if the new group type is RECORDING.

Sample Programming

Basic Automated Attendant Feature

Callers are routed to a group with a typical message:

“Thank you for calling the ABC company Automated Attendant number. If you know the extension number of the person you are trying to reach, and if you have a touch-dial telephone, you may dial the number before the end of this message. If not, someone will be with you shortly.”

In this example, an LDN is programmed as the default destination for the Automated Attendant group. The LDN is assigned the name “ABC” which shows the attendant the name of the company that the caller was attempting to reach.

An additional group could be supplied as the illegal number routing point for the tenant group of the first RAD in the first group. The second group could have a message saying:

“You have dialed an incorrect number. Please try again or stay on the line and someone will be with you shortly.”

The default destination for the second group is an LDN that indicates a caller that had already misdialed a number. The illegal number routing point for the tenant group of the first RAD in the second group would be the same group itself, so that the caller would keep looping back to the same group when an illegal number is dialed.

ACD Front-end Message

Using an ACD Front-End message, ACD callers are routed to a group with a message such as:

“Thank you for calling the ACME Supply House. If you have a touch-dial telephone, please dial 1 for housewares, 2 for seed catalogs and 9 to repeat this message. Otherwise, please stay on the line and an agent will be with you shortly.”

The default destination for this recording group is an ACD path that handles unscreened calls. The recording group also has a prefix of “123” programmed. The ACD path for housewares has the access code “1231”, the ACD path for seed catalogs has the access code “1232”, and the recording group itself has the access code “1239”.

Aided External Dialing

Using aided external dialing, callers are routed to a group with a message such as:

“Thank you for calling. Please dial 1 for the Toronto office, 2 for Vancouver and 3 for Montreal. Otherwise, please stay on the line and the attendant will be with you shortly.”

The recording group is programmed with the attendant console as the default destination with a prefix of 80.

Assuming that the access code for system speed dial is 80, the caller is selecting system speed abbreviated dial numbers. Index 1 is Toronto, 2 is Vancouver, and 3 is Montreal.

Maintenance

RAD Failure Handling

The RAD failure handling for the Automated Attendant feature is as described in the ACD TELEMARKETER Application Package section.

RADs in DND

If a caller accesses an Auto-Attendant group that has all RADs in DND or Busy-out, the caller is immediately routed to the default destination. If the last RAD in a group goes out of service, the waiting callers are processed as if they had just accessed a group in which all of the RADs are out of service.

Failure to Answer

If a RAD fails to answer, the caller ringing the RAD is routed to the default destination. The RAD is placed in DND.

Troubleshooting Guidelines

Failure to Answer

Problem: A call is never answered by a RAD. The call always routes to the default.

Action:

- Verify that the RAD is functioning (not all in DND or busy-out).
- Check that the RAD message length is not too short. The message length programmed in CDE Form 17 should be 3-5 seconds longer than the message length.
- Verify that COS Option 404, Recording Failure to Hangup Timer is programmed to be longer than the actual message length in the COS of the RAD.
- Check traffic measurement for skip pegs for the group (indicates a problem with too few RADs or too few receivers required for Automated Attendant).
- Check the “wait for resources time” for the group.

Failure to Respond

Problem: The RAD does not respond to dialed digits for a call transmitted over IP trunks.

Action:

- Do not call the Automated Attendant over IP trunks.

Call Dropped

Problem: A call is dropped after no number is dialed.

Action:

- Verify that a default destination is programmed.
- Confirm that a connection is allowed between possible callers and the default destination.

RAD Fails to Drop

Problem: Recording does not end even though digits are dialed.

Action: Ensure that the option “Dialing Over Recording” is enabled for that group.

Wrong Message

Problem: A caller receives the wrong message.

Action:

- Verify that message was recorded correctly.
- Check that the RAD is programmed correctly.
- Check that the RAD is connected to the correct line circuit.

Installation

The Automated Attendant feature may require additional receivers to prevent users from complaining about a delay to dial tone. The following subsections describe the installation requirements for the Automated Attendant feature:

Ensure that the system is provisioned with sufficient DTMF/DSP receivers. Traffic measurement can be used to monitor receiver usage and to identify failures to get a receiver.

Universal Card Receiver Modules

Each Universal Card can contain up to four receiver modules; each receiver module contains four receivers, for a total of 16 receivers. The Universal card is a high-power card. The SX-200 ICP system allows up to 4 high-power cards per bay.

Centralized Voice mail

Introduction

The Centralized Voice mail feature allows a network of IP PBXs (SX-200 ICP, 3300 ICP and SX-200 IP Nodes) to share a single voice mail facility with Message Waiting Indication at all network sites. The Centralized Voice mail feature works with any voice mail interface - ONS, DNIC, QSIG. Centralized Voice mail can also be installed on another SX-200 ICP or a Mitel 3300 ICP and accessed over IP and T1 trunks. Release 4.0 of the 3300 ICP adds support for voice mail softkeys.

Feature Description

The Centralized Voice mail feature allows a single voice mail device to service several interconnected PBXs (SX-200 ICP, 3300 ICP and SX-200 IP Nodes). The Centralized Voice mail feature works with any voice mail interface e.g., ONS, or DNIC. The Centralized Voice mail feature is a purchasable option.

When a call is forwarded to voice mail from an extension on the same PBX to which the voice mail device is connected, the caller's extension number, the forwarding extension number, and the call forward reason are passed to the voice mail system. When the forwarding extension is on another PBX, Centralized Voice mail is used to pass this information between PBXs to the voice mail system.

The transfer of information is done over trunks between each PBX.

Figure: Typical Centralized Voice mail System

Supported Trunks and Voicemail Systems

SX-200 ICP Trunks and Supported Voice Mail Systems			
Voice Mail Systems Supported (Site A)	**Trunks Supported (Site B) SX-200 ICP**		
T1	**QSIG**	**IP**	
SX-200 ICP Embedded Voice Mail	Yes(1)	Yes	Yes(2)
SX-200 ICP ONS	Yes	No	No
SX-200 ICP DNIC	Yes	Yes	Yes
SX-200 ONS	Yes	No	No
SX-200 DNIC	Yes	Yes	Yes
3300 ICP Embedded Voice Mail	No	No	Yes(2)

Notes

(1) With SX-200 ICP R1.2 and above.

(2) Voicemail softkeys are supported over IP trunks only.

Calls Forwarded to Voice mail

The PBX forwards user information to the voice mail device. When the forwarding extension is in a remote PBX (a PBX that is not connected directly to the voice mail system), the remote PBX provides the user information to the local PBX (the PBX to which the voice mail device is connected). This user information is sent to the local PBX through an ARS route using modified digits. The local PBX sends the user information to voice mail in the form of abbreviated dial digits.

Message Lamp is Lit

The voice mail system uses inband signaling.

The voice mail system dials the Send Message feature access code plus the digit 1 to light the message waiting LED on the set. Normal ARS programming is used to reach the set in the remote PBX. The host PBX recognizes the message waiting indicator code and checks all subsequent digits to determine if the code is for an extension on the host PBX or the remote PBX. Host message waiting indicators are set and removed normally. Remote PBX message waiting indicators follow the ARS routing defined for the remote PBX extension digits dialed after the message waiting indicator code. Only the message waiting code and the remote PBX extension digits can be sent to the remote PBX.

User Retrieves the Voice mail Message

The same abbreviated dial digits that were used to forward calls to voice mail are used to retrieve messages. When retrieving messages, the information sent to the voice mail device contains only the calling extension number.

Voice mail users on a remote PBX can retrieve their messages either by the "message" key on the set or by dialing the "Call Message Sender of Oldest Message" feature access code. The caller information is passed over the PBX network to the Voice mail system so that the caller is routed to the correct mailbox. This is done by digits programmed in a system abbreviated dial index and via ARS modified digit strings.

Message Lamp is Turned Off

The voice mail system dials the Send Message feature access code plus the digit 2 to turn off the message waiting LED on the set. Normal ARS programming is used to reach the set in the remote PBX.

Programming

Conditions and Recommendations

Centralized Voice mail programming requires the use of the # character as a delimiter over the network trunks. Do **NOT** use the # character in any feature access code that will be part of your abbreviated dial string in Form 31. If so, the # character will act as a delimiter and the forwarding will not take place.

QSIG and IP trunks also prohibit the use of the * character because they are programmed as rotary dial trunks. Dialing on rotary trunks is limited to digits 0-9.

The following features must have the same feature access codes for all the PBXs sharing the Centralized Voice mail System. Do NOT use the # character—and the * character if using QSIG and IP trunks—in the feature access codes for the following:

Feature 03, Call Forwarding - All Calls
Feature 04, Call Forwarding - Internal Only
Feature 05, Call Forwarding - External Only
Feature 11, Extension General Attendant Access
Feature 24, Abbreviated Dial Access
Feature 32, Automatic Wakeup
Feature 39, Analog Network Accept Caller's Extension
Feature 41, Send Messages
Feature 54, Analog Network Accept Call Forward Data
Feature 55, Analog Network Accept Call Forward Reason

System Option 36, End of Dial Character (#), in Form 04 must be disabled in all PBXs sharing the Centralized Voice mail System.

System Option 21, Incoming to Outgoing Call Forward, in Form 04 must be enabled in all PBXs sharing the Centralized Voice mail System.

System Option 113, Centralized Attendant / Voice mail, must be purchased and enabled in all PBXs sharing the Centralized Voice mail System.

Class of Service Option 265 (Voice mail System Speed Dial Index 0-255) must be the same in all PBXs sharing the Centralized Voice mail system. This option is used by the network trunks and the Voice mail ports when messages are retrieved. The index must be between 100 and 255 to prevent interference with the conflict dialing timer.

Analog networking must operate in both directions between the Host PBX and the Remote PBX.

Make sure the tie trunks in the Remote PBX can connect to each other. Incoming calls to the Remote PBX will then be able to forward back out another tie trunk to the Host PBX.

The Host PBX with the Centralized Voice mail system should be the largest PBX of the group.

The PBXs cannot have duplicated extension numbers.

ONS Voice mail Programming Over E&M Trunks

The setup for the PBX with the voice mail attached (PBX A) is the same as that shown in the system documentation under Voice mail on ONS Ports. Please make sure this is working before proceeding. Once this initial setup is complete, PBX A is then able to pass all the necessary information into the voice mail system in order to route calls to the proper mailboxes.

PBX B, the remote system, has all the necessary information to integrate with PBX A, but needs a way to send it. This is accomplished through feature access codes and some new ARS entries for Form 22. The new codes are used to send call forward information across the E&M tie trunks between the systems.

The following programming is for a two-system network. To connect more than two, all remote PBXs will have the same programming as PBX B.

The following programming is to be entered the same in both PBXs unless otherwise specified. Remember that each PBX must use a unique numbering plan for extensions, but the same feature access codes. The codes themselves may be any number that does not include the pound key (#).

In CDE Form 01 (System Configuration): Program the required E&M trunks between the PBXs (either T1 Trunk card or E&M module on a Universal Card).

In CDE Form 02 (Feature Access Codes): Program an access code for the following:

Note: These Feature Access Codes must be the same for all PBXs sharing the Centralized Voice mail system. Do NOT use the # character in the feature access codes. Feature Access Code 3 is mutually exclusive with Feature Access Codes 4 and/or 5.

- 3 Call Forwarding - All Calls (COS Option 260, Internal/External Split Call Forwarding must be DISABLED in the COS for the user dialing this code)
- 4 Call Forwarding - Internal Only (COS Option 260, Internal/External Split Call Forwarding must be ENABLED in the COS for the user dialing this code)
- 5 Call Forwarding - External Only (COS Option 260, Internal/External Split Call Forwarding must be ENABLED in the COS for the user dialing this code)
- 11 Extension General Attendant Access
- 24 Abbreviated Dial Access
- 32 Automatic Wakeup
- 39 Analog Network Accept Caller’s Extension
- 41 Send Message
- 54 Analog Network Accept Call Forward Data
- 55 Analog Network Accept Call Forward Reason

In CDE Form 03 (Class of Service Options): Enable the following:

Voice mail Users:

- COS Option 206 - Call Forwarding Busy
- COS Option 207 - Call Forwarding - No Answer
- COS Option 208 - Call Forwarding - External
- COS Option 232 - Message Waiting Setup - Lamp
- COS Option 245 - Abbreviated Dialing Access

ONS Ports - Serving the Main Hunt Group :

- COS Option 208 - Call Forwarding - External
- COS Option 212 - Can Flash If Talking To an Incoming Trunk
- COS Option 213 - Can Flash If Talking To an Outgoing Trunk
- COS Option 216 - Data Security

- COS Option 238 - Override Security
- COS Option 245 - Abbreviated Dialing Access
- COS option 259 - Message Sending
- COS Option 261 - ONS Voice mail Port
- COS Option 301 - Camp-on
- COS Option 313 - 319 (trunk to trunk connect as applicable)
- COS Option 702 SMDR Overwrite Buffer

ONS Ports - Message Waiting Lamp Ports:

- COS Option 216 - Data Security
- COS Option 220 - Do Not Disturb
- COS Option 235 - Originate Only
- COS Option 238 - Override Security
- COS Option 259 - Message Sending
- COS Option 261 - ONS Voice mail Port
- COS Option 265 - Voice mail Speed Dial Index (e.g 101)

E&M Trunks - Both PBXs:

- COS Option 208 - Call Forward External
- COS Option 245 - Abbreviated Dialing Access
- COS Option 259 - Message Sending
- COS Option 261 - ONS Voice mail Port,
- COS Option 265 - Voice mail System Speed Dial Index (e.g.101).

Notes:

1. Use the same index for all PBXs. Use an index between 100 and 255.

2. All trunks involved in centralized voice mail transmission must be in trunk groups in both PBXs. They do not necessarily have to be in the same trunk group.

In CDE Form 04 (System Options/System Timers): Enable option 113, Centralized Attendant/Voice mail for both systems (this feature package must be purchased). Also enable option 21, Incoming to Outgoing Call Forward, in both systems to allow for external call forward to abbreviated dial.

System Option 22 needs to be enabled in PBX A to present dial tone to the voice mail when a caller hangs up. This signals the voice mail system to hang up the ONS port and wait for a new call.

Note: Option 36 (End of Dial Character) must NOT be enabled when Centralized Voice mail is used.

DTMF on/off timing can be shortened from default (via system option 69 and 70) to make sending of digits across the tie trunks faster. Since this is a system option, be sure other outgoing trunks still operate properly once the DTMF timers are shortened.

In CDE Form 13 (Trunk Circuit Descriptors): For each PBX, program a circuit descriptor for the E&M trunks to be used for Centralized Voice mail. Set the Incoming and Outgoing Start Types to “WINK” and set DTMF to ‘yes’. Set "Far end gives answer supervision" to "no".The circuit descriptor options must be the same for all PBXs in the network.

In CDE Form 15 (Dial-In Trunks): Program the E&M trunks to be used for Centralized Voice mail.

Ensure that fields “N” and “M” are set to 0, and that field “X” is left blank.

In CDE Form 16 (Trunk Groups): Program the trunks used for Centralized Voice mail into a trunk group. Give the trunk group a meaningful name (appears on display sets when retrieving centralized voice mail messages).

In CDE Form 17 (Hunt Groups): For PBX A program the voice mail ports (ONS ports setup in form 9) into two hunt groups, one for message lamp ports, and one for the main voice mail ports. Give the two hunt groups access codes. In our example, the main hunt group access code is 555. At least one port from the voice mail system must be dedicated to message waiting, and be in its own Hunt Group.

Note: Most ONS voice mail systems need to have a separate hunt group set up for the message lamp ports so that the PBX does not erroneously extinguish the lamps when users call voice mail. The ports in this 'message lamp' hunt group are typically setup as DND so all calls overflow to the main voice mail hunt group via hunt group overflow. In our example, this hunt group access code is 556.

In CDE Form 30 (Device Interconnection) make sure the tie trunks in the remote PBX can connect to each other. Incoming calls to the remote PBX will then be able to forward back out another tie trunk to the host PBX .

To give PBX A the ability to connect its incoming lines to the outgoing tie trunks when the ONS Voice mail system is used as the auto attendant, Form 30 must be altered so that those trunk types may connect. Use the review key of Form 30 to see what the numbers mean, then make sure an * appears at the intersection of the incoming trunk type ROW number with Column 6 (outgoing tie trunks to PBX B).

To permit the tie trunks in PBX B to connect with each other, Form 30, row 6 column 6 needs to be *. Incoming calls to PBX B will then be able to forward back out another tie trunk to PBX A.

In CDE Form 31 (System Abbreviated Dial Entry): Users of PBX A will forward their calls to an abbreviated dial entry on their host PBX. For PBX B, users will setup call forward to another abbreviated dial string (100 is this example), which is used by PBX B ARS to transfer call details to PBX A. Users will check for their messages via abbreviated dial entry 101.

The following is a PBX A example for Form 31 .

Index Number	Digit String
100	555*9*9ZZ*4#*6#*8#
101	555*9*9YY*6#

PBX A users forward to an abbreviated dial string that sends call details via DTMF after the ONS voice mail system answers. In our example, the string is 555*9*9*4#*6#*8# where:

555 is the main hunt group access code
*9 - a one second pause (add pauses as needed)
ZZ - Special code to access mailbox greeting/take-a-message
YY - Special code for user access to mailbox (asks for PIN)
*4 - send forward reason (busy, always or no answer)
- Delimiter
*6 - send called party extension number (the set that forwarded the call)
- Delimiter
*8 - send 'calling party' extension number (the set that originally made the call)
- Delimiter

Note: Most ONS voice mail systems provide for a special code to access the 'take a message' function. Please enter that function code into the string above after the pause.

The following is a PBX B example for Form 31.

Index Number	Digit String
100	555
101	556

PBX B users forward to an abbreviated dial string that contains only the hunt group access code for the voice mail system in PBX A. In our example the string is 555. Users check for their messages via index 101.

ARS Programming

This document assumes that normal ARS programming between the switches is already complete, and is not shown here. It is further assumed that analog networking is also set-up both ways between the two switches. Please refer to Analog Networking in the documentation for that setup.

This ARS programming is for PBX B only.

In CDE Form 26 (ARS: Digit Strings): Program the leading digits that will route the caller to the extensions on the remote switches. This is also used by the Voice mail system to turn message waiting lamps on or off.

For our example, you need to send calls to 555 out its own route, as it will be using a different digit modification table than normal set to set calls.This assumes that the numbering plan for PBX A starts with the digit 5. This entry in PBX B should already be present.

Leading Digit	Return Dial Tone	Restricted COR Group
5	No	Unrestricted

In CDE Form 26 Subform (Digit Strings Subform): Program the following:

Return Dial Tone No
Digits to be Analyzed enter digits
Qty to Follow 0
Long Distance no
Termination Type route
Termination Number nn

For our example:

Digits to be Analyzed	Qty to Follow	Long Distance	Term Type and Number
55	0	No	Route 5
56	0	No	Route 6

Program the route that will be used by the Centralized Voice mail trunk group (refer to Form 22, ARS: Modify Digits Table).

In CDE Form 23 (ARS: Route Definition): Select the E&M trunk group (previously programmed in Form 16) used for Voice mail between PBXs.

For example:

Route Number	Trunk Group	Modify Digit Entry	Comments
5	As applicable	5	Forward to voice mail
6	As applicable	6	Get Msg on voice mail

In CDE Form 22 (ARS: Modify Digits table): Refer to this example for programming.

Entry	Qty to Delete	Digits to be Inserted
5	3	**39*6#**54*01#**55*02#7100
6	3	**39*67101

Because the voice mail is ONS, the above information must be presented to PBX A which then uses abbreviated dial to call the voice mail system and pass the data via DTMF to the voice mail system after it answers.

**39 FAC for analog network accept caller's extension number
(remember we must enter ** to send *)
*6 Send caller's extension number (originating node)
Delimiter
**54 FAC for analog network accept call forward data
*01 Send call forwarder extension number
Delimiter
**55 FAC for analog network accept call forward reason
*02 Send call forward reason (Internal/External, Busy or No Answer)
Delimiter

To explain this briefly, PBX B users set up their phones to forward to an abbreviated dial entry (from Form 31), which contains the entry 555. 555 is then dialed and picked up by ARS leading digits (5) and digits to analyze (55) which sends the call to route 5. Route 5 has modified digits entry 5 assigned to it. Modified digit entry 5 deletes the 555 and adds the string of feature access codes and special function codes (*6, *01, etc.) followed by the access code and bin number to access abbreviated dial in PBX A as shown above in Form 22 programming.

The effect of this is that PBX B will first send the details of the call to PBX A. Once PBX A has the required information from PBX B, PBX A's abbreviated dial bin 100 is requested. Then, after the voice mail system has answered, the call forward information is sent via DTMF tones. This allows the voice mail system to route the call to the proper mailbox and answer appropriately (i.e. user is busy or user is unavailable).

All users in PBX A setup forwarding to the abbreviated dial (7100 in our example). Users in PBX B setup their forwarding to their own abbreviated dial entry (7100).

E&M Trunk Programming

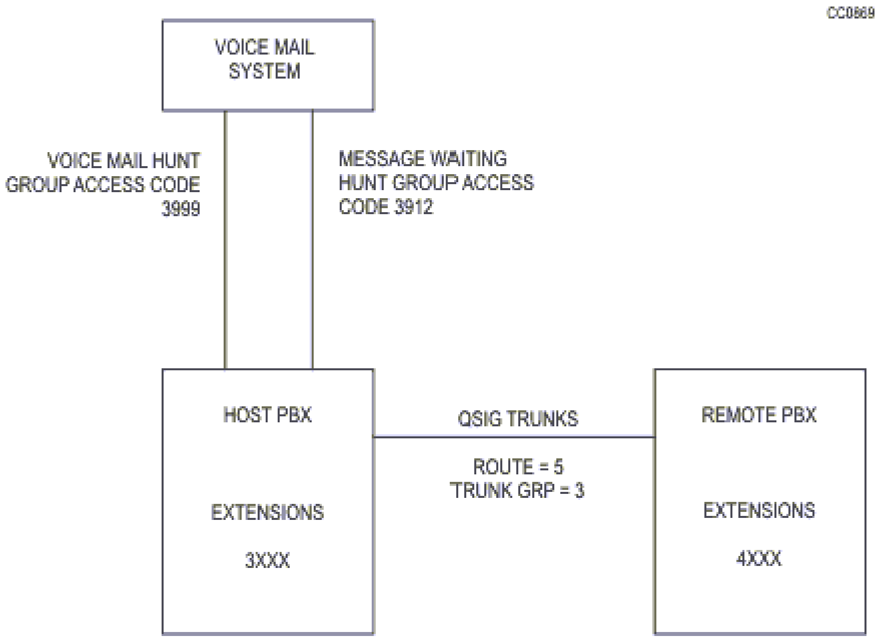
Other than the COS options identified earlier in this document, the only concerns are that the E&M trunk's associated trunk circuit descriptor (from Form 13) be setup as follows; DTMF =YES, far end gives answer supervision = NO.

Wink Start trunks work faster and may reduce call setup time.

Ensure there are sufficient trunks and receivers in the system to accommodate the anticipated traffic, and that all trunks to be used for centralized voice mail are entered into a trunk group (or groups).

In Form 15, make sure the N,M,X columns are set to 0,0,- (this sets up the E&M trunks to Tie Trunk type - this allows receipt of variable length strings).

Example - ONS Voice mail Programming over E&M Trunks



Form 02 - Feature Access Codes - all PBXs

Form 02 - Feature Access Codes		
Feature #	**Feature**	**Access Code**
3	Call Forwarding - All Calls	61
4	Call Forwarding - Internal Only	65
5	Call Forwarding - External Only	62
24	Abbreviated Dial Access	88
39	Analog Network Accept Caller's Extension	43
41	Send Message	70
54	Analog Network Accept Call Forward Data	44
55	Analog Network Accept Call Forward Reason	45

Form 03 - COS Define

Form 03 - COS Define	
	COS Options ENABLED
Voice mail Users	206 - Call Forwarding - Busy

Form 03 - COS Define	
	207 - Call Forwarding - No Answer
	208 - Call Forwarding - External
	232 - Message Waiting Setup - Lamp
	245 - Abbreviated Dialing Access
ONS Voice mail Ports on Host PBX	208 - Call Forwarding External
	212 - Can Flash if Talking to an Incoming Trunk
	213 - Can Flash if Talking to an Outgoing Trunk
	216 - Data Security
	238 - Override Security
	245 - Abbreviated Dialing Access
	261 - ONS Voice mail Port
	301 - Camp-on
ONS Message Waiting Port on Host PBX	216 - Data Security
	220 - Do Not Disturb
	235 - Originate Only
	238 - Override Security
	259 - Message Sending
	261 - ONS Voice mail Port
	265 - Voice mail System Speed Dial Index (100)
E&M Trunk Group	208 - Call Forward External
	245 - Abbreviated Dialing Access
	259 - Message Sending
	261 - ONS Voice mail Port
	265 - Voice mail System Speed Dial Index (100)

Form 17 - Hunt Groups (Host PBX only)

Form 17 - Hunt Groups		
Hunt Group's Function: ONS Voice mail Ports		
OPTIONS		
EXTENSION NUMBERS		
HUNT GROUP NUM (1-100):	ACCESS CODE: 3999	GROUP TYPE:
HUNTING (Terminal or Circular):	HUNT GROUP NAME:	
Overflow:	Msg Length (mm:ss):	
Default Destination:	Dialing Over Recording (Y/N):	
Wait For Resources (mm:ss):	Prefix Digits:	
DTRX (Y/N):	Default Modem Group (Y/N):	

Form 17 - Hunt Groups		
Hunt Group's Function: Message Waiting Ports		
OPTIONS		
EXTENSION NUMBERS		
HUNT GROUP NUM (1-100):	ACCESS CODE: 3912	GROUP TYPE:
HUNTING (Terminal or Circular):	HUNT GROUP NAME:	
Overflow:	Msg Length (mm:ss):	
Default Destination:	Dialing Over Recording (Y/N):	
Wait For Resources (mm:ss):	Prefix Digits:	
DTRX (Y/N):	Default Modem Group (Y/N):	

The ONS voice mail system must use a dedicated message sending port.

Program the voice mail system to wait for confirmation dial tone before going on hook after outpulsing the message waiting codes.

Form 31 - System Abbreviated Dial Entry

Host PBX:
The digit string invoked by index number 100 contains the information required by the voice mail system (i.e., calling extension, called extension, and reason for forwarding).

Form 31 - System Abbreviated Dial Entry	
Index Number	**Digit String**
100	3999*9*4#*6#*8#

3999 Voice mail Hunt group
*9 1 second pause
*4 Send reason for forwarding (no answer, busy)
Delimiter for reason for forwarding
*6 Send called party's extension number
Delimiter for called party's extension number
*8 Send calling party's extension number
Delimiter for calling party's extension number

Remote PBX:
The digit string invoked by index number 100 for the Remote PBX does not contain the information required by the voice mail system. The number (3999 in this example) matches the digits programmed in ARS to route the call to the Host PBX over the tie trunks.

Form 31 - System Abbreviated Dial Entry	
Index Number	**Digit String**
100	3999 88 100

Form 26 - ARS: Digit Strings

Form 26 - ARS Digit Strings				
Lead Digits: __3__ Ret. DT: __NO__ Restr COR Grp. Unrestricted				
Digits to Analyze	**Qty to Follow**	**Long distance**	**Term Type & #**	
999	0	NO	Route	5

Form 23 - ARS: Route Definition

Form 23 - ARS Route Definition			
Route #	**Trunk Grp**	**Mod Digit Entry**	**Comments**
5	3	5	Voice mail

Form 22 - ARS: Modified Digit Table

Form 22 - ARS Modified Digit Table		
Entry	**Qty to delete**	**Digits to be Inserted**
5	4	43*6#44*01#45*02#

43	FAC for Analog Network Accept Callers Extension
*6	Send caller's extension number (originating node)
#	Terminator of caller's extension
44	FAC for Analog Network Accept Call Forward Data
*01	Send call forwarder extension number
#	Terminator of the call forwarder extension number
45	FAC for Analog Network Accept Call Forward Reason
*02	Send call forward reason
#	Terminator for call forward reason

DNIC Voice mail Programming Over E&M Trunks

The setup for the PBX with the voice mail attached (PBX A) is the same as that shown in the system documentation under Voice mail on DNIC Ports. Please make sure this is working before proceeding.PBX A has all the necessary information to route the calls to the proper mailboxes, and therefore operates in a normal fashion. PBX B, the remote system, also has all necessary information, but needs a way to send this to PBX A. This is accomplished through new feature access codes programmed in each switch, which are used to pass call forward information across the E&M tie trunks between the systems.

The following programming is for a two-system network. To connect more than two, all remote PBXs will have the same programming as PBX B.

The following programming is to be entered for both PBXs unless otherwise specified. Remember to use a unique numbering plan for extensions, but the same feature access codes. The codes themselves may be any number that does not include the pound key (#).

In CDE Form 01 (System Configuration): Program required E&M trunks between the PBXs (either T1 Trunk card or E&M module on a Universal Card).

In CDE Form 02 (Feature Access Codes): Program an access code for the following:

Note: These Feature Access Codes must be the same for all PBXs sharing the Centralized Voice mail system. Do NOT use the # character in the feature access codes. Feature Access Code 3 is mutually exclusive with Feature Access Codes 4 and/or 5.

- 3 Call Forwarding - All Calls (COS Option 260, Internal/External Split Call Forwarding must be DISABLED in the COS for the user dialing this code)
- 4 Call Forwarding - Internal Only (COS Option 260, Internal/External Split Call Forwarding must be ENABLED in the COS for the user dialing this code)
- 5 Call Forwarding - External Only (COS Option 260, Internal/External Split Call Forwarding must be ENABLED in the COS for the user dialing this code)
- 11 Extension General Attendant Access
- 24 Abbreviated Dial Access
- 32 Automatic Wakeup
- 39 Analog Network Accept Caller's Extension
- 41 Send Message
- 54 Analog Network Accept Call Forward Data
- 55 Analog Network Accept Call Forward Reason

In CDE Form 03 (Class of Service Options): Enable the following:

Voice mail Users:

- COS Option 206 - Call Forwarding Busy
- COS Option 207 - Call Forwarding - No Answer
- COS Option 208 - Call Forwarding - External

- COS Option 232 - Message Waiting Setup - Lamp (for ONS sets)
- COS Option 245 - Abbreviated Dialing Access

E&M Trunks - Both PBXs:

- COS Option 208 - Call Forward External
- COS Option 245 - Abbreviated Dialing Access
- COS Option 259 - Message Sending
- COS Option 261 - ONS Voice mail Port,
- COS Option 265 - Voice mail System Speed Dial Index (e.g.100).

Notes:

1. Use the same index for all PBXs. Use an index between 100 and 255.
2. All trunks involved in centralized voice mail transmission must be in trunk groups in both PBXs. They do not necessarily have to be in the same trunk group.
3. For a PBX between the PBX A (Host) and the PBX B (Remote) enable COS Option 316, Tie Trunk to Tie Trunk Connect.

In CDE Form 04 (System Options/System Timers): Enable Option 113, Centralized Attendant/Voice mail for both systems (this feature package must be purchased). Also enable Option 21, Incoming to Outgoing Call Forward in PBX B to allow for external call forward to abbreviated dial. For enable

For Centralized Voice mail over T1 trunks where the host PBX is an SX-200 ICP, add Option 135, Voicemail Control of MWI. The option is required on the host to control the Message Waiting Indicator (MWI) lamp on remote PBXs via the transmission of DTMF digits.

Note: Option 36 (End of Dial Character) must NOT be enabled when Centralized Voice mail is used.

DTMF on/off timing can be shortened from default (via system option 69 and 70) to make sending of digits across the tie trunks faster. Since this is a system option, be sure other outgoing trunks still operate properly once the DTMF timers are shortened.

In CDE Form 09 (Desk to Device Assignments): Set up each port required to interface the DNIC voice mail system as a SUPERSET 430 subattendant.

In CDE Form 13 (Trunk Circuit Descriptors): For each PBX, program a circuit descriptor for the E&M trunks to be used for Centralized Voice mail. Set the Incoming and Outgoing Start Types to “WINK” and set DTMF to ‘yes’. Set "Far end gives answer supervision" to "no".The circuit descriptor options must be the same for all PBXs in the network.

In CDE Form 15 (Dial-In Trunks): Program the E&M trunks to be used for Centralized Voice mail.

Ensure that fields “N” and “M” are set to 0, and that field “X” is left blank.

In CDE Form 16 (Trunk Groups): Program the trunks used for Centralized Voice mail into a trunk group. Give the trunk group a meaningful name (appears on display sets when retrieving centralized voice mail messages).

In CDE Form 17 (Hunt Groups): For PBX A, program one hunt group with all voice mail ports, and assign an access code (for example 555). The Voice mail ports will perform message waiting as well as call processing. Set the hunt group up as terminal. (Testing Note: To troubleshoot the installation, put a SUPERSET 430 as the first member of group to allow viewing of display as presented to voice mail. Make sure the set has the same COS as the voice mail ports).

Note: Other voice mail systems typically need to have a separate hunt group setup for the message lamp ports so that the PBX does not erroneously extinguish the lamps when users call voice mail. The ports in this 'message lamp' hunt group are typically setup as DND so all calls overflow to the main voice mail hunt group via hunt group overflow.

In CDE Form 30 (Device Interconnection) make sure the tie trunks in the remote PBX (PBX B) can connect to each other (row 6 column 6 needs to be *). Incoming calls to the remote PBX will then be able to forward back out another tie trunk to the host PBX (PBX A).

To give PBX A the ability to connect its incoming lines to the outgoing tie trunks when the Voice mail system is used as the auto attendant, Form 30 must be altered so that those trunk types may connect. Use the review key of Form 30 to see what the numbers mean, then make sure an * appears at the intersection of the incoming trunk type ROW number with Column 6 (outgoing tie trunks to PBX B).

To permit the tie trunks in PBX B to connect with each other, Form 30, row 6 column 6 needs to be *. Incoming calls to PBX B will then be able to forward back out another tie trunk to PBX A.

In CDE Form 31 (System Abbreviated Dial Entry): For DNIC voice mail configurations, only the users in PBX B will set-up forwarding to an abbreviated dial string. Users of PBX A will forward directly to the hunt group pilot number. For PBX B, the abbreviated dial string will be used by ARS to transfer call details to PBX A.

The following is a PBX B example for Form 31 where 555 is the Voice mail Hunt Group.

Index Number	Digit String
100	555

PBX A users forward directly to the voice mail hunt group access code.

PBX B users forward to an abbreviated dial string that contains only the hunt group access code for the voice mail system in PBX A. In our example the string is 555.

ARS Programming

This document assumes that normal ARS programming between the switches is already complete, and is not shown here. See the feature, Analog Networking. It is further assumed that analog networking is also set-up both ways between the two switches.

This ARS programming is for PBX B only.

In CDE Form 26 (ARS: Digit Strings): Program the leading digits that will route the caller to the extensions on the remote switches. This is also used by the Voice mail system to turn message waiting lamps on or off.

For our example, you need to send calls to 555 out its own route, as it will be using a different digit modification table than normal set to set calls.This assumes that the numbering plan for PBX A starts with the digit 5. This entry in PBX B should already be present.

Leading Digit	Return Dial Tone	Restricted COR Group
5	No	Unrestricted

In CDE Form 26 Subform (Digit Strings Subform): Program the following:

Return Dial Tone	No
Digits to be Analyzed	enter digits
Qty to Follow	0
Long Distance	no
Termination Type	route
Termination Number	nn

For our example:

Digits to be Analyzed	Qty to Follow	Long Distance	Term Type and Number
55	0	No	Route 5

Program the route that will be used by the Centralized Voice mail trunk group (refer to Form 22, ARS: Modify Digits Table).

In CDE Form 23 (ARS: Route Definition): Select the E&M trunk group (previously programmed in Form 16) used for Voice mail between PBXs.

For example:

Route Number	Trunk Group	Modify Digit Entry	Comments
5	As required	5	Route to voice mail

In CDE Form 22 (ARS: Modify Digits table): Refer to this example for programming.

Entry	Qty to Delete	Digits to be Inserted
5	0	**39*6#**54*01#**55*02#

Because the voice mail is DNIC, the above information must be presented to the voice mail system while ringing in.

**39 FAC for analog network accept caller's extension number
*6 Send caller's extension number (originating node)
Delimiter
**54 FAC for analog network accept call forward data
*01 Send call forwarder extension number
Delimiter
**55 FAC for analog network accept call forward reason
*02 Send call forward reason (Internal/External, Busy or No Answer)
Delimiter

To explain this briefly, PBX B users set up their phones to forward to an abbreviated dial entry (from Form 31), which contains the entry 555. 555 is then dialed and picked up by ARS leading digits (5) and digits to analyze (55) which sends the call to route 5. Route 5 has modified digits entry 5 assigned to it which then adds the string of feature access codes and special number sequences (*6, *01, etc.) as shown above in Form 22 programming. The effect of this is the system will first send the details of the call. Then, while the voice mail system is being rung, the call forward information is presented in the display so that the voice mail system can route the call to the proper mailbox and answer appropriately (i.e. user is busy or user is unavailable).

All users in PBX A setup forwarding directly to the voice mail hunt group access code (555 in our example). Users in PBX B setup their forwarding to an abbreviated dial entry (7100).

E&M Trunk Programming

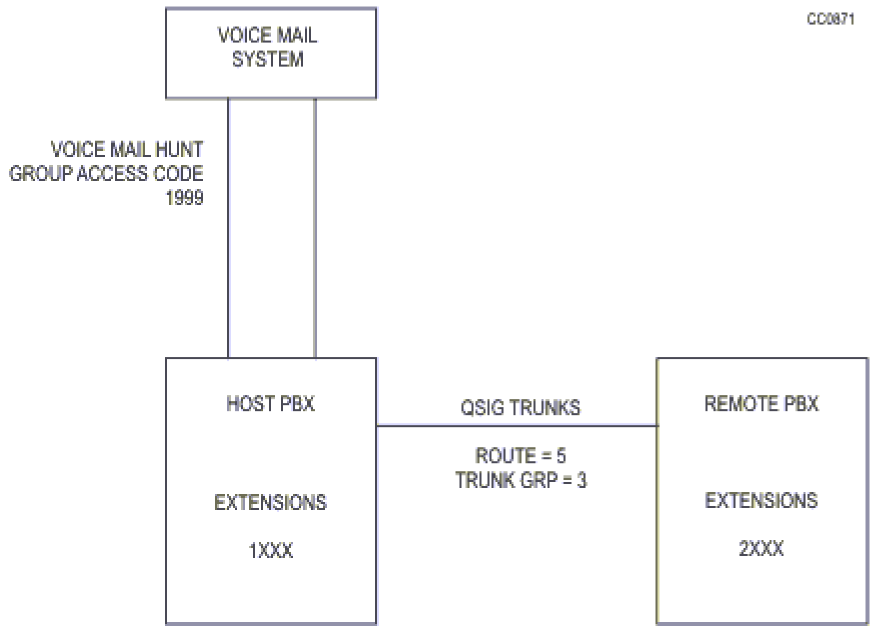
Other than the COS options identified earlier in this document, the only concerns are that the E&M trunk's associated trunk circuit descriptor (from Form 13) be setup as follows; DTMF =YES, far end gives answer supervision = NO.

Wink Start trunks work faster and may reduce call setup time.

Ensure there are sufficient trunks and receivers in the system to accommodate the anticipated traffic, and that all trunks to be used for centralized voice mail are entered into a trunk group (or groups).

In Form 15, make sure the N,M,X columns are set to 0,0,- (this sets up the E&M trunks to Tie Trunk type - this allows receipt of variable length strings).

Example - DNIC Voice mail Programming Over E&M Trunks



Form 02 - Feature Access Codes - all PBXs

Form 02 - Feature Access Codes

Feature #	Feature	Access Code
3	Call Forwarding - All Calls	61
4	Call Forwarding - Internal Only	65
5	Call Forwarding - External Only	62
24	Abbreviated Dial Access	88
39	Analog Network Accept Caller’s Extension	43
41	Send Message	70
54	Analog Network Accept Call Forward Data	44
55	Analog Network Accept Call Forward Reason	45

Form 03 - COS Define

Form 03 - COS Define	
	COS Options ENABLED
Voice mail Users - all PBXs	206 - Call Forwarding - Busy
	207 - Call Forwarding - No Answer
	208 - Call Forwarding - External
	232 - Message Waiting Setup - Lamp
	245 - Abbreviated Dialing Access
DNIC Voice mail Ports on Host PBX	208 - Call Forwarding - External
	212 - Can Flash if Talking to an Incoming Trunk
	213 - Can Flash if Talking to an Outgoing Trunk
	216 - Data Security
	229 - Voice mail Port
	261 - ONS Voice mail Port
	238 - Override Security
	301 - Camp-on
	502 - Display ANI/DNIS/CLASS Information
DNIC Msg Wtg Port on Host PBX	216 - Data Security
	220 - Do Not Disturb
	261 - ONS Voice mail Port
	229 - Voice mail Port
	235 - Originate Only
	238 - Override Security
	259 - Message Sending
	265 - Voice mail System Speed Dial Index (100)
E&M Trunk Group	208 - Call Forward External
	261 - ONS Voice mail Port
	245 - Abbreviated Dialing Access
	259 - Message Sending
	265 - Voice mail System Speed Dial Index (100)

Form 17 - Hunt Groups (Host PBX only)

Form 17 - Hunt Groups		
Hunt Group's Function: DNIC Voice mail Ports		
OPTIONS		
EXTENSION NUMBERS		
HUNT GROUP NUM (1-100):	ACCESS CODE: 1999	GROUP TYPE:
HUNTING (Terminal or Circular):	HUNT GROUP NAME:	
Overflow:	Msg Length (mm:ss):	
Default Destination:	Dialing Over Recording (Y/N):	
Wait For Resources (mm:ss):	Prefix Digits:	
DTRX (Y/N):	Default Modem Group (Y/N):	

Form 31 - System Abbreviated Dial Entry

Host PBX:
The digit string invoked by index number 100 contains the information from the remote PBX required by the voice mail system (calling extension, called extension, and reason for forwarding).

Form 31 - System Abbreviated Dial Entry	
Index Number	**Digit String**
100	1999

1999	Voice mail Hunt group
*9	1 second pause
*4	Send reason for forwarding (no answer, busy)
#	Delimiter for reason for forwarding
*6	Send called party's extension number
#	Delimiter for called party's extension number
*8	Send calling party's extension number
#	Delimiter for calling party's extension number

Remote PBX:
The digit string invoked by index number 100 for the Remote PBX does not contain the information required by the voice mail system. The number (1999 in this example) matches the digits programmed in ARS to route the call to the Host PBX over the tie trunks

Form 31 - System Abbreviated Dial Entry	
Index Number	Digit String
100	1999 88 100

Form 26 - ARS: Digit Strings

Form 26 - ARS Digit Strings				
Leadl Digits: __3__ Ret. DT: __NO__ Restr COR Grp. Unrestricted				
Digits to Analyze	Qty to Follow	Long distance	Term Type & #	
999	0	NO	Route	5

Form 23 - ARS: Route Definition

Form 23 - ARS Route Definition			
Route #	**Trunk Grp**	**Mod Digit Entry**	**Comments**
5	3	5	Voice mail

Form 22 ARS: Modified Digit Table

Form 22 - ARS Modified Digit Table		
Entry	**Qty to delete**	**Digits to be Inserted**
5	4	43*6#44*01#45*02#

43 FAC for Analog Network Accept Callers Extension
*6 Send caller's extension number (originating node)
Terminator of caller's extension
44 FAC for Analog Network Accept Call Forward Data
*01 Send call forwarder extension number
Terminator of the call forwarder extension number
45 FAC for Analog Network Accept Call Forward Reason
*02 Send call forward reason
Terminator for call forward reason

DNIC Voice mail Programming Over QSIG or IP Trunks

The setup for the PBX with the voice mail attached (PBX A) is the same as that shown in the system documentation under Voice mail on DNIC Ports. Please make sure this is working before proceeding.PBX A has all the necessary information to route the calls to the proper mailboxes, and therefore operates in a normal fashion. PBX B, the remote system, also has all necessary information, but needs a way to send this to PBX A. This is accomplished through new feature access codes programmed in each switch, which are used to pass call forward information across the trunks between the systems.

The following programming is for a two-system network. To connect more than two, all remote PBXs will have the same programming as PBX B.

The following programming is to be entered for both PBXs unless otherwise specified. Remember to use a unique numbering plan for extensions, but the same feature access codes. The codes themselves may be any number that does not include the pound key (#).

In CDE Form 01 (System Configuration): Program required QSIG or IP trunks between the PBXs.

In CDE Form 02 (Feature Access Codes): Program an access code for the following:

Note: These Feature Access Codes must be the same for all PBXs sharing the Centralized Voice mail system. Do NOT use the # character in the feature access codes. Feature Access Code 3 is mutually exclusive with Feature Access Codes 4 and/or 5.

- 3 Call Forwarding - All Calls (COS Option 260, Internal/External Split Call Forwarding must be DISABLED in the COS for the user dialing this code)
- 4 Call Forwarding - Internal Only (COS Option 260, Internal/External Split Call Forwarding must be ENABLED in the COS for the user dialing this code)
- 5 Call Forwarding - External Only (COS Option 260, Internal/External Split Call Forwarding must be ENABLED in the COS for the user dialing this code)
- 11 Extension General Attendant Access
- 24 Abbreviated Dial Access
- 32 Automatic Wakeup
- 39 Analog Network Accept Caller’s Extension
- 41 Send Message
- 54 Analog Network Accept Call Forward Data
- 55 Analog Network Accept Call Forward Reason

In CDE Form 03 (Class of Service Options): Enable the following:

Note: COS Option 613 - Display ANI Information Only must be enabled on Voice mail ports for IP Trunks only.

Voice mail Users:

- COS Option 206 - Call Forwarding Busy
- COS Option 207 - Call Forwarding - No Answer
- COS Option 208 - Call Forwarding - External
- COS Option 232 - Message Waiting Setup - Lamp (for ONS sets)
- COS Option 245 - Abbreviated Dialing Access

E&M Trunks - Both PBXs:

- COS Option 208 - Call Forward External
- COS Option 245 - Abbreviated Dialing Access
- COS Option 259 - Message Sending
- COS Option 261 - ONS Voice mail Port,
- COS Option 265 - Voice mail System Speed Dial Index (e.g.100).

Notes:

1. Use the same index for all PBXs. Use an index between 100 and 255.
2. All trunks involved in centralized voice mail transmission must be in trunk groups in both PBXs. They do not necessarily have to be in the same trunk group.
3. For a PBX between the PBX A (Host) and the PBX B (Remote) enable COS Option 316, Tie Trunk to Tie Trunk Connect.

In CDE Form 04 (System Options/System Timers): Enable option 113, Centralized Attendant/Voice mail for both systems (this feature package must be purchased). Also enable option 21, Incoming to Outgoing Call Forward in PBX B to allow for external call forward to abbreviated dial.

Note: Option 36 (End of Dial Character) must NOT be enabled when Centralized Voice mail is used.

DTMF on/off timing can be shortened from default (via system option 69 and 70) to make sending of digits across the tie trunks faster. Since this is a system option, be sure other outgoing trunks still operate properly once the DTMF timers are shortened.

In CDE Form 09 (Desk to Device Assignments): Set up each port required to interface the DNIC voice mail system as a SUPERSET 430 Subattendant.

In CDE Form 13 (Trunk Circuit Descriptors): For each PBX, program a circuit descriptor for the trunks to be used for Centralized Voice mail.

In CDE Form 15 (Dial-In Trunks): Program the trunks to be used for Centralized Voice mail.

Ensure that fields “N” and “M” are set to 0, and that field “X” is left blank.

In CDE Form 16 (Trunk Groups): Program the trunks used for Centralized Voice mail into a trunk group. Give the trunk group a meaningful name (appears on display sets when retrieving centralized voice mail messages).

In CDE Form 17 (Hunt Groups): For PBX A, program one hunt group with all voice mail ports, and assign an access code (for example 555). The Voice mail ports will perform message waiting as well as call processing. Set the hunt group up as terminal. (Testing Note: To troubleshoot the installation, put a SUPERSET 430 as the first member of group to allow viewing of display as presented to voice mail. Make sure the set has the same COS as the voice mail ports).

Note: Other voice mail systems typically need to have a separate hunt group setup for the message lamp ports so that the PBX does not erroneously extinguish the lamps when users call voice mail. The ports in this 'message lamp' hunt group are typically setup as DND so all calls overflow to the main voice mail hunt group via hunt group overflow.

In CDE Form 30 (Device Interconnection) make sure the trunks in the remote PBX (PBX B) can connect to each other (row 6 column 6 needs to be *). Incoming calls to the remote PBX will then be able to forward back out another tie trunk to the host PBX (PBX A) .

To give PBX A the ability to connect its incoming lines to the outgoingtrunks when the Voice mail system is used as the auto attendant, Form 30 must be altered so that those trunk types may connect. Use the review key of Form 30 to see what the numbers mean, then make sure an * appears at the intersection of the incoming trunk type ROW number with Column 6 (outgoing tie trunks to PBX B).

To permit the trunks in PBX B to connect with each other, Form 30, row 6 column 6 needs to be *. Incoming calls to PBX B will then be able to forward back out another tie trunk to PBX A.

In CDE Form 31 (System Abbreviated Dial Entry): For DNIC voice mail configurations, only the users in PBX B will set-up forwarding to an abbreviated dial string. Users of PBX A will forward directly to the hunt group pilot number. For PBX B, users will forward their calls to an index number in their abbreviated dial form on their host PBX. In our example, the index number represents the string 555, which is the main hunt group access code.

The following is a PBX B example for Form 31.

Index Number	Digit String
100	555

ARS Programming

This document assumes that normal ARS programming between the switches is already complete, and is not shown here.

This ARS programming is for PBX B only.

In CDE Form 26 (ARS: Digit Strings): Program the leading digits that will route the caller to the extensions on the remote switches. This is also used by the Voice mail system to turn message waiting lamps on or off.

For our example, you need to send calls to 555 out its own route, as it will be using a different digit modification table than normal set to set calls.This assumes that the numbering plan for PBX A starts with the digit 5. This entry in PBX B should already be present.

Leading Digit	Return Dial Tone	Restricted COR Group
5	No	Unrestricted

In CDE Form 26 Subform (Digit Strings Subform): Program the following:

Return Dial Tone No
Digits to be Analyzed enter digits
Qty to Follow 0
Long Distance no
Termination Type route
Termination Number nn

For our example:

Digits to be Analyzed	Qty to Follow	Long Distance	Term Type and Number
55	0	No	Route 5

Program the route that will be used by the Centralized Voice mail trunk group (refer to Form 22, ARS: Modify Digits Table).

In CDE Form 23 (ARS: Route Definition): Select the trunk group (previously programmed in Form 16) used for Voice mail between PBXs.

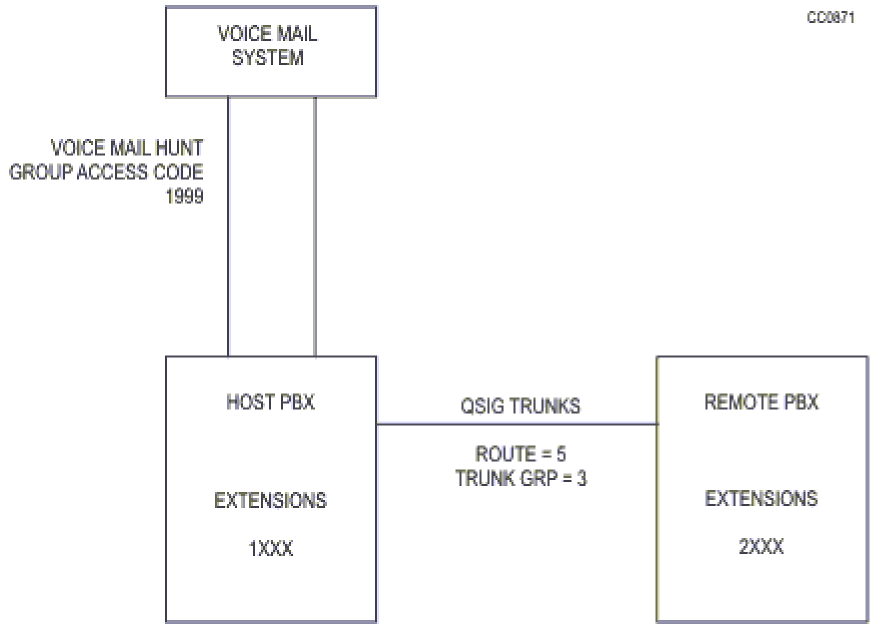
For example:

Route Number	Trunk Group	Modify Digit Entry	Comments
5	As required	5	Route to voice mail

In CDE Form 22 (ARS: Modify Digits table): Refer to this example for programming.

Entry	Qty to Delete	Digits to be Inserted
5	0	N/A

Example- DNIC Voice mail Programming Over QSIG or IP Trunks



Form 02 - Feature Access Codes - all PBXs

Form 02 - Feature Access Codes

Feature #	Feature	Access Code
3	Call Forwarding - All Calls	61
4	Call Forwarding - Internal Only	65
5	Call Forwarding - External Only	62
24	Abbreviated Dial Access	88
39	Analog Network Accept Caller’s Extension	43
41	Send Message	70
54	Analog Network Accept Call Forward Data	44
55	Analog Network Accept Call Forward Reason	45

Form 03 - COS Define

Form 03 - COS Define

	COS Options ENABLED
Voice mail Users - all PBXs	206 - Call Forwarding - Busy
	207 - Call Forwarding - No Answer

Form 03 - COS Define	
	208 - Call Forwarding - External
	232 - Message Waiting Setup - Lamp
	245 - Abbreviated Dialing Access
DNIC Voice mail Ports on Host PBX	208 - Call Forwarding - External
	212 - Can Flash if Talking to an Incoming Trunk
	213 - Can Flash if Talking to an Outgoing Trunk
	216 - Data Security
	229 - Voice mail Port
	261 - ONS Voice mail Port
	238 - Override Security
	301 - Camp-on
	502 - Display ANI/DNIS/CLASS Information
	613 - Display ANI Information Only
DNIC Msg Wtg Port on Host PBX	216 - Data Security
	220 - Do Not Disturb
	261 - ONS Voice mail Port
	229 - Voice mail Port
	235 - Originate Only
	238 - Override Security
	259 - Message Sending
	265 - Voice mail System Speed Dial Index (100)
Trunk Group	208 - Call Forward External
	261 - ONS Voice mail Port
	245 - Abbreviated Dialing Access
	259 - Message Sending
	265 - Voice mail System Speed Dial Index (100)

Form 17 - Hunt Groups (Host PBX only)

Form 17 - Hunt Groups		
Hunt Group's Function: DNIC Voice mail Ports		
OPTIONS		
EXTENSION NUMBERS		
HUNT GROUP NUM (1-100):	ACCESS CODE: 1999	GROUP TYPE:
HUNTING (Terminal or Circular):	HUNT GROUP NAME:	
Overflow:	Msg Length (mm:ss):	
Default Destination:	Dialing Over Recording (Y/N):	
Wait For Resources (mm:ss):	Prefix Digits:	
DTRX (Y/N):	Default Modem Group (Y/N):	

Form 31 - System Abbreviated Dial Entry

Host PBX:
The digit string invoked by index number 100 contains the main hunt group number.

Form 31 - System Abbreviated Dial Entry	
Index Number	**Digit String**
100	1999

1999 Voice mail Hunt group

Remote PBX:
The number (1999 in this example) matches the digits programmed in ARS to route the call to the Host PBX over the trunks

Form 31 - System Abbreviated Dial Entry	
Index Number	Digit String
100	1999

Form 26 - ARS: Digit Strings

Form 26 - ARS Digit Strings				
Leadl Digits: __3__ Ret. DT: __NO__ Restr COR Grp. Unrestricted				
Digits to Analyze	Qty to Follow	Long distance	Term Type & #	
999	0	NO	Route	5

Form 23 - ARS: Route Definition

Form 23 - ARS Route Definition			
Route #	**Trunk Grp**	**Mod Digit Entry**	**Comments**
5	3	5	Voice mail

Form 22 ARS: Modified Digit Table

Form 22 - ARS Modified Digit Table		
Entry	**Qty to delete**	**Digits to be Inserted**
5	0	N/A

Testing and Troubleshooting

The use of SMDR as an aid to testing and troubleshooting is recommended. The SMDR printout will enable you to verify that the correct digits are being sent and received in the correct sequence. Copy and complete the worksheets found in Programming Worksheets for use as a reference during the testing operation.

Testing

The following operational tests should be performed in all PBXs in the network. Refer to the completed worksheets for information on the digits programmed in each PBX.

Note: Host PBX refers to the PBX on which the voice mail device is connected. Remote PBX is the PBX that wants to share the voice mail facility and is not directly connected to the voice mail device.

Call Forwarded to Voice mail From PBX Connected to Voice mail System

From the Host PBX, leave a voice mail message for another extension in the Host PBX. The call will be forwarded to AABBB where AA is the abbreviated dial access code and BBB is the speed dial bin number.

Use a COV monitor (a modified SUPERSET 4) to observe communications between the PBX and the voice mail system. A monitor is a display telephone connected in parallel with the voice mail port so that signaling and messages can be observed on its display.

Note: COV monitoring is not available on the SX-200 ICP.

CAUTION: Connect the display telephone in parallel only when it is being used as a monitor - do not leave it connected at any other time.

The speed dial bin contains the digit string HHHH*9*4#*6#*8# where:

HHHH is the voice mail hunt group access code

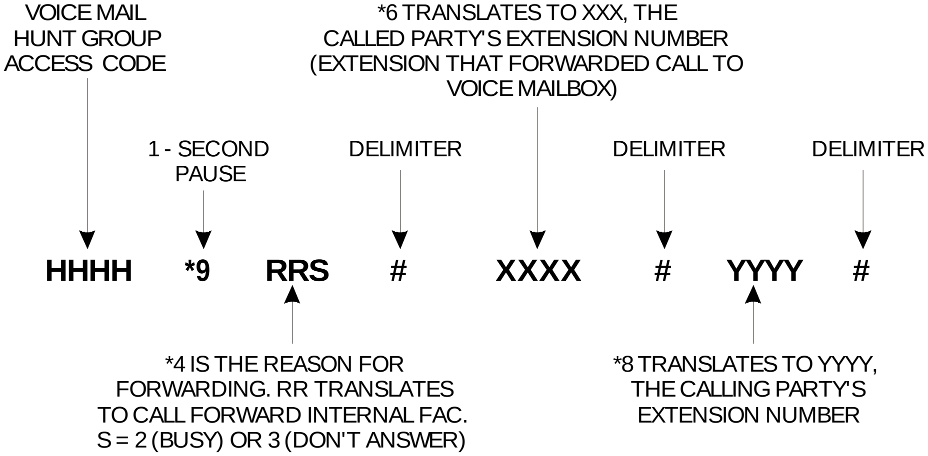
*9 gives a 1 second pause

*4 is the reason for forwarding

*6 is the called extension number

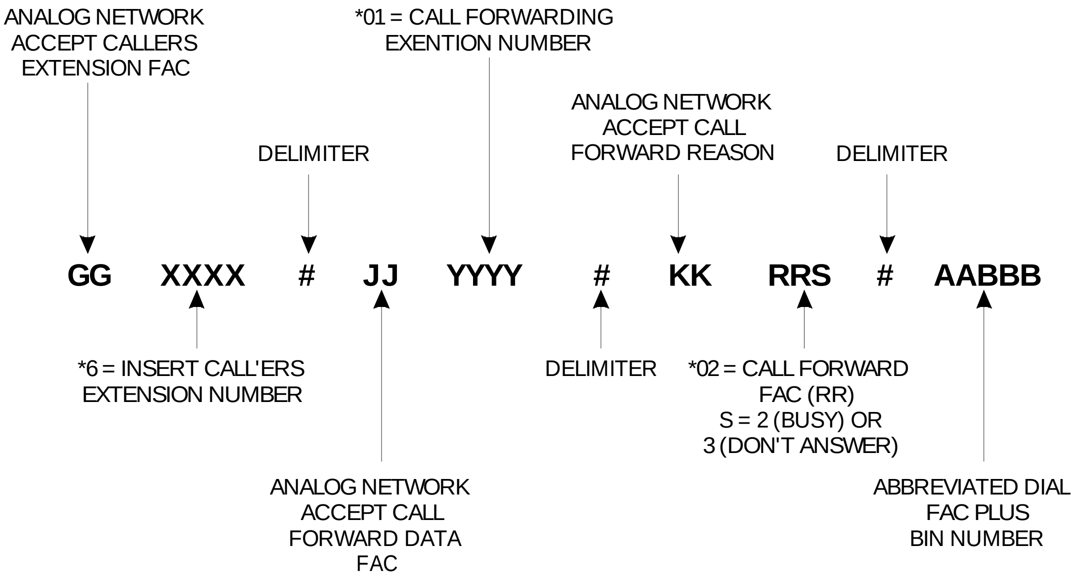
*8 is the calling extension number.

This instructs the PBX to send the following string to the voice mail system:

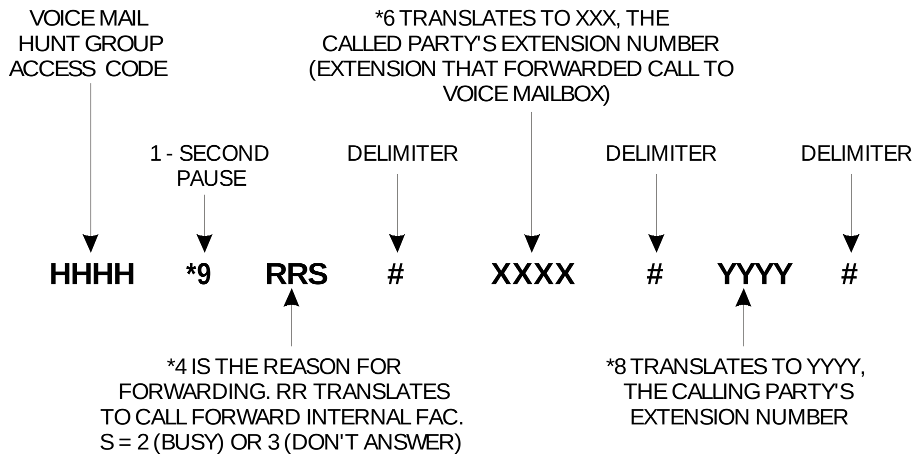


Call Forwarded to Voice mail From PBX Not Connected to Voice mail System

From the Remote PBX, leave a voice mail message for another extension in the Remote PBX. The call will be forwarded to AABBB where AA is the abbreviated dial access code and BBB is the speed dial bin number. The speed dial bin contains the digit string CDDD, where C is the ARS leading digit and DDD are the digits to follow, pointing to a modified digit entry which deletes CDDD and sends the digits GG*6#JJ*01#KK*02#AABBB where GG is the Analog Network Accept Callers Extension FAC, *6 is the called extension number, JJ is the Analog Network Accept Call Forward Data FAC, and KK is the analog network accept call forward reason. This instructs the Remote PBX to send the following string to the Host PBX:



The Host PBX now has all of the information required to pass to the voice mail system. The abbreviated dialing code (AABBB) are used to access a speed dial bin which contains the digit string HHHH*9*4#*6#*8# where HHHH is the voice mail hunt group access code, *9 gives a 1 second pause,*4 is the reason for forwarding, *6 is the called extension number and *8 is the calling extension number. This instructs the PBX to send the following string to the voice mail system:



Lighting the Message Lamp

From the Host PBX, leave a voice mail message for another extension in the Host PBX. The voice mail device will send the digits FF1XXXX to the PBX to light the lamp on the called extension. FF is the Send Message FAC, the digit 1 indicates that the lamp is to be turned on, and XXXX is the called extension number.

From the Host PBX, leave a voice mail message for an extension in the Remote PBX. Normal ARS programming routes the digits (same as above) to the Remote PBX which turns the lamp on at the called extension.

Note: In order for lamps to be lit on remote systems, the ONS voice mail system must program a post dial delay long enough to ensure the message sending operation is completed.

Note: The remote system must program IP channel 1 on the first IP trunk card programmed or MWI will not work.

Extinguishing the Message Lamp

Retrieve the voice mail messages from the extensions used above. The voice mail device will send the digits FF2XXXX to the PBX to extinguish the lamp on the called extension. FF is the Send Message FAC, the digit 2 indicates that the lamp is to be turned off, and XXXX is the called extension number.

Retrieving Messages

Retrieve the voice mail message by dialing the Call Sender of Oldest Message code or by pressing the message waiting key on the set.

When the access code is used, the request is directed to the voice mail hunt group. The system looks at COS option 265 for which speed dial index to go to for voice mail.

When the message waiting key is used to retrieve a message, the COS options listed below are required so that the request is routed correctly.

261 - ONS Voice mail Port
265 - Voice mail System Speed Dial Index

These options must be enabled in the class of service of the Message Waiting Port of the Host PBX and in the E&M trunk class of service of the Remote PBX.

Retrieving Messages - Host PBX

When the message key is pressed, the PBX checks the COS of the device which sent the message. For a message left for a user in the Host PBX this device will be the message waiting port. The COS options contain abbreviated dialing index entered in COS Option 265 so the message retrieval request is directed to the voice mail hunt group via modified digits and abbreviated dial digits.

Retrieving Messages - Remote PBX

When the message key is pressed, the PBX checks the COS of the device (trunk) which sent the message. For a message left for a user in the Remote PBX this device will be the E&M tie trunk. The COS of the E&M tie trunk contains abbreviated dialing index entered in COS Option 265 so the message retrieval request is directed to the voice mail hunt group via modified digits and abbreviated dial digits.

Troubleshooting

Problem: Unable to leave voice mail messages on extensions in the Remote PBX - SMDR shows digit strings across tie trunks are incomplete.

Action:

- Ensure that System Option 36 (End of Dial Character #) is disabled in Form 04. Check the ARS programming.
- Ensure that the trunks do not have the Far End Gives Answer Supervision option enabled.

Note: The sequence of the ARS digit strings is very important to the correct operation of the Centralized Voice mail feature.

Problem: Unable to light message waiting lamp on extensions in the Remote PBX. Lamps light normally on extensions in the Host PBX.

Action:

- When a lamp on or lamp off request is sent across tie trunks, there is a slight delay. Test from the console on the Host PBX. Do a Direct Trunk Select and dial the ‘Lamp On’ digits to the remote extension. If the lamp now lights, program the voice mail device to wait for confirmation dial tone.

Problem: *6 Unable to leave voice mail messages on extensions in the Remote PBX - SMDR shows digit strings across tie trunks are incomplete.

Programming Worksheets

Form 02 - Feature Access Codes

Host PBX ____________

Form 02 - Feature Access Codes		
Feature #	**Feature**	**Access Code**
3	Call Forwarding - All Calls	
4	Call Forwarding - Internal Only	
5	Call Forwarding - External Only	
24	Abbreviated Dial Access	
39	Analog Network Accept Caller’s Extension	

Form 02 - Feature Access Codes		
Feature #	**Feature**	**Access Code**
41	Send Message	
54	Analog Network Accept Call Forward Data	
55	Analog Network Accept Call Forward Reason	

Remote PBX ____________

Form 02 - Feature Access Codes		
Feature #	**Feature**	**Access Code**
3	Call Forwarding - All Calls	
4	Call Forwarding - Internal Only	
5	Call Forwarding - External Only	
24	Abbreviated Dial Access	
39	Analog Network Accept Caller’s Extension	
41	Send Message	
54	Analog Network Accept Call Forward Data	
55	Analog Network Accept Call Forward Reason	

Form 03 - COS Define - ONS Voice mail

Host PBX _________________ (Voice mail Connected)

Form 03 - COS Define		
	COS Option Name	**Enabled**
Voice mail Users	206 - Call Forwarding - Busy	
207 - Call Forwarding - No Answer		
208 - Call Forwarding - External		
232 - Message Waiting Setup - Lamp		
245 - Abbreviated Dialing Access		

Form 03 - COS Define		
	COS Option Name	**Enabled**
ONS Voice mail Ports	208 - Call Forwarding - External	
212 - Can Flash if Talking to an Incoming Trunk		
213 - Can Flash if Talking to an Outgoing Trunk		
216 - Data Security		
238 - Override Security		
245 - Abbreviated Dialing Access		
261 - ONS Voice mail Port		
301 - Camp-on		
ONS Message Waiting Ports	216 - Data Security	
220 - Do Not Disturb		
235 - Originate Only		
238 - Override Security		
259 - Message Sending		
261 - ONS Voice mail Port		
265 - Voice mail System Speed Dial Index (0 - 255)		
E&M Trunk Group	208 - Call Forward External	
	245 - Abbreviated Dialing Access	
259 - Message Sending		
261 - ONS Voice mail Port		
265 - Voice mail System Speed Dial Index (0 - 255)		

Form 03 - COS Define - ONS Voice mail

Remote PBX ______

Form 03 - COS Define		
	COS Option Name	**Enabled**
Voice mail Users	206 - Call Forwarding - Busy	
207 - Call Forwarding - No Answer		
208 - Call Forwarding - External		
232 - Message Waiting Setup - Lamp		
245 - Abbreviated Dialing Access		
E&M Trunk Group	208 - Call Forward External	
	245 - Abbreviated Dialing Access	
259 - Message Sending		
261 - ONS Voice mail Port		
265 - Voice mail System Speed Dial Index (0 - 255)		

Remote PBX ______

Form 03 - COS Define		
	COS Option Name	**Enabled**
Voice mail Users	206 - Call Forwarding - Busy	
207 - Call Forwarding - No Answer		
208 - Call Forwarding - External		
232 - Message Waiting Setup - Lamp		
245 - Abbreviated Dialing Access		
E&M Trunk Group	208 - Call Forward External	
	245 - Abbreviated Dialing Access	
259 - Message Sending		
261 - ONS Voice mail Port		
265 - Voice mail System Speed Dial Index (0 - 255)		

Form 03 - COS Define - DNIC Voice mail

Host PBX ________________ (Voice mail Connected)

Form 03 - COS Define		
	COS Option Name	**Enabled**
Voice mail Users	206 - Call Forwarding - Busy	
207 - Call Forwarding - No Answer		
208 - Call Forwarding - External		
232 - Message Waiting Setup - Lamp		
245 - Abbreviated Dialing Access		
267 Softkey Support for Voice mail		
DNIC Voice mail Ports	208 - Call Forwarding - External	
212 - Can Flash if Talking to an Incoming Trunk		
213 - Can Flash if Talking to an Outgoing Trunk		
216 - Data Security		
261 - ONS Voice mail Port		
229 - Voice mail Port		
238 - Override Security		
301 - Camp-on		
502 - Display ANI/DNIS/CLASS Information		
DNIC Message Waiting Ports	216 - Data Security	
220 - Do Not Disturb		
229 - Voice mail Port		
261 - ONS Voice mail Port		
235 - Originate Only		
238 - Override Security		
259 - Message Sending		
265 - Voice mail System Speed Dial Index (0 - 255)		
E&M Trunk Group	208 - Call Forward External	
	261 - ONS Voice mail Port	

Form 03 - COS Define		
	COS Option Name	Enabled
245 - Abbreviated Dialing Access		
259 - Message Sending		
265 - Voice mail System Speed Dial Index (0 - 255)		

Form 03 - COS Define - DNIC Voice mail

Remote PBX ____________

Form 03 - COS Define		
	COS Option Name	Enabled
Voice mail Users	206 - Call Forwarding - Busy	
207 - Call Forwarding - No Answer		
208 - Call Forwarding - External		
232 - Message Waiting Setup - Lamp		
245 - Abbreviated Dialing Access		
E&M Trunk Group	208 - Call Forward External	
	261 - ONS Voice mail Port	
245 - Abbreviated Dialing Access		
259 - Message Sending		
265 - Voice mail System Speed Dial Index (0 - 255)		

Remote PBX ____________

Form 03 - COS Define		
	COS Option Name	Enabled
Voice mail Users	206 - Call Forwarding - Busy	
207 - Call Forwarding - No Answer		
208 - Call Forwarding - External		
232 - Message Waiting Setup		

Form 03 - COS Define		
	COS Option Name	**Enabled**
- Lamp		
245 - Abbreviated Dialing Access		
E&M Trunk Group	208 - Call Forward External	
261 - ONS Voice mail Port		
245 - Abbreviated Dialing Access		
259 - Message Sending		
265 - Voice mail System Speed Dial Index (0 - 255)		

Form 17 - Hunt Groups

Voice mail Ports

Form 17 - Hunt Groups		
Hunt Group's Function:		
OPTIONS		
EXTENSION NUMBERS		
HUNT GROUP NUM (1-100):	ACCESS CODE:	GROUP TYPE:
HUNTING (Terminal or Circular):	HUNT GROUP NAME:	
Overflow:	Msg Length (mm:ss):	
Default Destination:	Dialing Over Recording (Y/N):	
Wait For Resources (mm:ss):	Prefix Digits:	
DTRX (Y/N):	Default Modem Group (Y/N):	

Message Waiting Ports (ONS voice mail systems only)

Form 17 - Hunt Groups		
Hunt Group's Function:		
OPTIONS		
EXTENSION NUMBERS		
HUNT GROUP NUM (1-100):	ACCESS CODE:	GROUP TYPE:

Form 17 - Hunt Groups	
HUNTING (Terminal or Circular):	HUNT GROUP NAME:
Overflow:	Msg Length (mm:ss):
Default Destination:	Dialing Over Recording (Y/N):
Wait For Resources (mm:ss):	Prefix Digits:
DTRX (Y/N):	Default Modem Group (Y/N):

Form 31 - System Abbreviated Dial Entry

Host PBX __________ (Voice mail Connected)

Form 31 - System Abbreviated Dial Entry		
Index Number	**Digit String**	**Private (Y/N)**

Remote PBX __________

Form 31 - System Abbreviated Dial Entry		
Index Number	**Digit String**	**Private (Y/N)**

Remote PBX __________

Form 31 - System Abbreviated Dial Entry		
Index Number	**Digit String**	**Private (Y/N)**

Remote PBX __________

Form 31 - System Abbreviated Dial Entry		
Index Number	**Digit String**	**Private (Y/N)**

Remote PBX ___________

Form 31 - System Abbreviated Dial Entry		
Index Number	**Digit String**	**Private (Y/N)**

Form 26 - ARS: Digit Strings

Host PBX ___________

Form 26 - ARS Digit Strings				
Lead Digits: _____ Ret. DT: _______ Restr COR Grp. _____________				
Digits to Analyze	**Qty to Follow**	**Long distance**	**Term Type & #**	

Remote PBX ___________

Form 26 - ARS Digit Strings				
Lead Digits: _____ Ret. DT: _______ Restr COR Grp. _____________				
Digits to Analyze	**Qty to Follow**	**Long distance**	**Term Type & #**	

Remote PBX ___________

Form 26 - ARS Digit Strings				
Lead Digits: _____ Ret. DT: _______ Restr COR Grp. _____________				
Digits to Analyze	**Qty to Follow**	**Long distance**	**Term Type & #**	

Form 23 - ARS: Route Definition

Host PBX ___________

Form 23 - ARS Route Definition			
Route #	**Trunk Group**	**Mod Digit Entry**	**Comments**

Remote PBX ___________

Form 23 - ARS Route Definition			
Route #	**Trunk Group**	**Mod Digit Entry**	**Comments**

Remote PBX ___________

Form 23 - ARS Route Definition			
Route #	**Trunk Group**	**Mod Digit Entry**	**Comments**

Remote PBX ___________

Form 23 - ARS Route Definition			
Route #	**Trunk Group**	**Mod Digit Entry**	**Comments**

Form 22 - ARS: Modified Digit Table

Host PBX ___________

Form 22 - ARS Modified Digit Table			
Entry Num (1-100)	**Qty to Delete**	**Digits to be inserted**	**Comments**

Remote PBX ___________

Form 22 - ARS Modified Digit Table			
Entry Num (1-100)	**Qty to Delete**	**Digits to be inserted**	**Comments**

Remote PBX ___________

Form 22 - ARS Modified Digit Table			
Entry Num (1-100)	**Qty to Delete**	**Digits to be inserted**	**Comments**

Remote PBX ___________

Form 22 - ARS Modified Digit Table			
Entry Num (1-100)	**Qty to Delete**	**Digits to be inserted**	**Comments**

Voice mail Systems

This section describes two types of voice mail system that may connect to an SX-200 ICP

Voice mail - DNIC Port

Description

Voice mail devices may use the Mitel DNIC interface.

Conditions

None.

Programming

Program the DNIC Voice mail Port(s) with the following Class of Service Options:

COS Option 212 (Can Flash if Talking to an Incoming Trunk)

COS Option 213 (Can Flash if Talking to an Outgoing Trunk)

COS Option 216 (Data Security)

COS Option 229 (Voice Mail Port)

COS Option 238 (Override Security)

COS Option 259 (Message Sending)

COS Option 261 (ONS Voice mail Ports)

COS Option 301 (Camp-on)

COS Option 604 (Telephone - Automatic Outgoing Line)

COS Option 606 (Telephone - Enhanced Answering Position)

COS Option 609 (Telephone - Night Service Switching)

COS Option 502 (Display ANI/DNIS/CLASS Information)

Note: Disable all other options including Call Forwarding

Define Hunt Group in CDE Form 17 (Hunt Groups) by programming the following fields:

EXT NUM Enter the extension number of each voice mail port that is a member of the voice mail hunt group.

ACCESS CODE The access code is the system access number for the VX Voice Processing system. Define the access code of the voice mail hunt group by using the ACCESS CODE softkey (softkey 7).

NAME Enter the name of the voice mail hunt group. Enter a name for the voice mail hunt group by using the OPTIONS softkey (softkey 4).

Note: Do not program an overflow destination for this hunt group. This will ensure that calls can camp on to the busy group.

Operation

Refer to the Voice mail system documents.

Voice mail - ONS Port

Description

This feature integrates an SX-200 ICP System with an ONS Voice mail system. The integration is based on the use of system abbreviated dial numbers. This eliminates several dialing steps involved in the sending and retrieving of voice mail messages.

Special codes allow the PBX operation to be customized to suit the operation of the particular voice mail system. Refer to the programming instructions for Form 31 - System Abbreviated Dial Entry, for a description of these special codes.

Conditions

The following conditions apply to this feature:

- This feature does not conflict with the Voice mail - Port feature - both systems may be installed simultaneously.
- This feature does not conflict with other ONS voice mail systems - they may be installed simultaneously.
- The *1, *6, *9, codes are only operational with the ONS Voice mail feature. If the abbreviated dial number dials anything but an ONS voice mail hunt group, the codes are ignored.
- The *4 and *8 codes are used with the NuPoint Messenger Enhanced Inband Signaling.
- Callers hear ringback while the digits are pulsed to the voice mail system. Audio is not cut through until after the last pause has been executed.

Programming

Form 01 - System Configuration

Program the ONS/CLASS Line Card.

Form 02 - Feature Access Codes

Assign access codes to:

- Feature 24 - Abbreviated Dial Access
- Feature 39 - Analog Network Accept Caller's Extension
- Feature 41 - Send Message
- Feature 54 - Analog Network Accept Call Forward Data
- Feature 55 - Analog Network Accept Call Forward Reason

When appropriate for use with *4, assign access codes to:

- Feature 3 - Call forwarding - All Calls
- Feature 4 - Call forwarding - Internal Only
- Feature 5 - Call forwarding - External Only

Form 03 - COS Define

In ONS voice mail port COS, enable the following:

- COS Option 208 - Call Forwarding - External
- COS Option 212 - Can Flash if Talking to an Incoming Trunk
- COS Option 213 - Can Flash if Talking to an Outgoing Trunk
- COS Option 216 - Data Security
- COS Option 238 - Override Security

- COS Option 245 - Abbreviated Dialing Access
- COS Option 261 - ONS Voice mail Port
- COS Option 301 - Camp-On

In Message Waiting/Pager Port COS, enable the following:

- COS Option 216 - Data Security
- COS Option 220 - Do Not Disturb
- COS Option 235 - Originate Only
- COS Option 238 - Override Security
- COS Option 259 - Message Sending
- COS Option 261 - ONS Voice mail Port
- COS Option 265 - Voice mail system Speed Dial Index

Note: Include option 265 in the COS of the first member of the hunt group for Messaging - Call Me Back feature.

In the COS of the mailbox holder, enable the following:

- COS Option 231 - Message Waiting Setup - Bell, or
- COS Option 232 - Message Waiting Setup - Lamp
- COS Option 264 - Half Fwd NA timer for DID call when VM msg on (This is a toll saving option, allowing external DID callers who are calling in to check their messages to hang up if the cfna timer is not shortened, knowing they have no messages to retrieve.)

Form 04 - System Options/System Timers

Enable:

- System Option 21 - Incoming to Outgoing Call Forwarding
- System Option 22 - Last Party Clear Dial Tone

Form 09 - (Desktop Device Assignments)

Program ONS lines and assign appropriate COS.

Ensure Voice mail Ports are in a separate COS from the Message Waiting Ports.

Form 14 - Non-Dial-In Trunks

If required, enter the Voice mail port hunt group access code for DAY, Night1 and/or Night2 for the trunk to go to the Receptionist portion of the VM system.

Form 17 - Hunt Groups

Program a hunt group for the Message Waiting port.

Program a hunt group containing the ONS voice mail ports.

Note: The Message Waiting Hunt Group is NOT the same as the ONS Hunt Group.

Form 19 - Call Rerouting Table

For Message Port Tenant

- DND Intercept Routing for this tenant for Day, Night1, and Night2 will be routed to the Voice mail hunt group access code.
- Station Illegal Number Routing for this tenant for Day, Night1, and Night2 will be routed to the Voice mail hunt group access code.

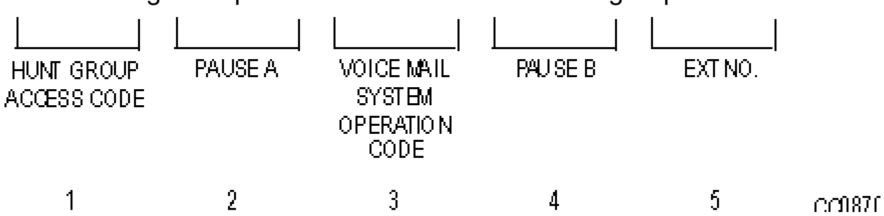
Form 31 - System Abbreviated Dial Entry

Special codes entered in this form allow the PBX operation to be customized to suit the operation of the particular voice mail system. These special codes eliminate many of the dialing steps involved in the sending and retrieving of voice mail messages. The integration of the PBX and voice mail system is based on the abbreviated dial numbers shown below.

Code	Description
*01	Tone out call forward data
*02	Tone out call forward reason
*3 XX	Insert manual dialed digits (XX) 2 digits expected
*4	Send call forward condition to Voice mail (See Note)
*6	Tone out caller extension number
*8	Send digits of calling party to Voice mail (See Note)
*9	1 second pause
**	DTMF digit *
#	DTMF digit #
0-9	DTMF digits 0 - 9
Note: *4 and *8 are used with NuPoint Messenger Enhanced Inband Signaling.	

Event Timing

The following example describes the feature's timing requirements:



1: Access code of the voice mail port hunt group.
2: Length of time required for the voice mail system to prepare to receive an optional operation code after answering the call.
3: Digit(s) indicating the type of operation to be performed voice mail system (record or playback message).
4: Length of time required for the voice mail system to receive a mailbox number, after receiving the operation code.
5: Usually *6: For a forwarded call, *6 translates into the originally called party's extension number. For non-forwarded calls, *6 translates into the caller's extension number.

Set up the desired call forwarding for the telephones using this voice mail feature.

- Place all ONS voice mail ports in the same hunt group in Form 17 (Hunt Groups) - assign an access code to the group.
- If the ONS voice mail system requires dial tone on hang up to release the port, enable System Option 22 (Last Party Clear - Dial Tone).
- If trunk calls are to reach the ONS voice mail system via a forwarding system abbreviated dial number, enable System Option 21 (Incoming to Outgoing Call Forward) and COS Option 208 in the trunk COS.

For Message Forward:

- Enter an index number for message forward in the INDEX NUMBER field of Form 31 - System Abbreviated Dial Entry
- Enter the following in the DIGIT STRING field:
  - the ONS voice mail hunt group access code,
  - any number of the *9 codes combining to result in an answer pause suitable to the voice mail system,
  - the voice mail system access code for the type of call forwarding which corresponds to the mailbox owner's call forwarding,
  - *6
  - the shortest *1 / *9 combination to ensure that the ONS voice mail port receives all of the digits.

For Message Retrieve:

- Enter an index number for message retrieve in the INDEX NUMBER field of Form 31 - System Abbreviated Dial Entry
- Enter the following in the DIGIT STRING field:
  - the ONS Voice mail hunt group access code,
  - any number of the *9 codes combining to result in an answer pause suitable to the voice mail system,
  - *6
  - the shortest *1 / *9 combination to ensure that the ONS voice mail port receives all of the digits.

Operation

Forward message:

- Dial extension number and leave message.

Retrieve message:

- Dial the system abbreviated dial access code and the message retrieve abbreviated dial index number.

The PBX dials out feature access code (3, 4, or 5) plus the index number.

Valid index numbers are:

1 - Forward Always

2 - Forward on Busy

3 - Forward No Answer

4 - Forward Busy/No Answer

The PBX dials out the ANI digits if the call originated over a trunk. If ANI digits are not allowed or not available, then no digits are sent. When the call is internal, the extension's access code is dialed out.

Mitel Express Messenger

Introduction

This section describes the Mitel Express Messenger voice mail card. The card installs in an SX-200 peripheral cabinet and uses a DNIC interface that connects directly to the backplane of the cabinet.

This section provides

- an illustration of the card (see the Express Messenger card perspective Figure)
- a description of the voice mail features
- programming requirements for the SX-200 ICP
- troubleshooting information
- removal and replacement instructions.

Only Mitel certified technicians are authorized to service the Mitel Express Messenger.

Installation and maintenance are done through the System Administration Mailbox (number 99 or 999 or 9999). There are two passcodes for the Administration Mailbox: Manager and Administrator.

Note: The person that performs the day-to-day operations and not the setting-up of the database should have the Manager passcode. The Manager passcode provides limited access to the database and therefore avoids novice users accessing the database.

- The Manager passcode lets the manager perform administrative duties (adding, deleting, editing mailbox functions, change greetings and default user mailbox passcodes) and gives minimum access to the system setup. The Default Manager passcode values are 484 (Release 1.0) and 6483 (Release 2.0 and later). The Manager passcode access still prompts users to press 1 to begin a new installation, but the prompt, when pressed says "Invalid Selection" and remains harmless.
- The Administrator passcode provides a higher level of access to the system setup. The Default Administrator passcode values are 741 (Release 1.0) and 1234 (Release 2.0 and later).

Refer to the following documentation for additional information about Mitel Express Messenger:

- System Administration Manual (PN 9109-080-025-NA)- describes all activities that are performed by the System Administrator to set up, operate, and change the Express Messenger system. This guide is provided as a PDF file that is shipped on disk with the card.
- User Guide - describes all activities that are performed by users to set up, use, or change their mailbox. This guide is provided as a PDF file is shipped on disk with the Mitel Express Messenger card. The User Guide is available in English and Spanish.
- Wallet Card - describes activities that are frequently performed by a mailbox user. It includes a record of dialing codes and access numbers.The Wallet Card is available in English and Spanish.
- Front Desk User Guide - in hotel or motel applications, the front desk attendant should have a front desk user guide. This guide provides instructions on how to administer guest mailboxes.It is available in English and Spanish.
- System Installation Sheet (PN 9109-080-024-NA) - describes how to install and set up the Express Messenger system.

Note: Information that is provided in the above documents is not repeated within this section.

Feature Description

Mitel Express Messenger allows a single voice mail card to provide either two, four, six, or eight voice mail ports. Softkey and Hospitality support is offered with Mitel Express Messenger Version 2.1 and greater.

Auto Attendant Features

Open and Closed Greetings: The company greeting can be programmed to automatically change from open business hours to closed or after hours.

Temporary Greeting: A Company Greeting can be programmed for use over holidays or shutdowns that will automatically expire after a specified number of days.

Alternate Greetings: Each port can use one of eight alternate greeting sets (Open, Closed, or Temporary) to allow special greetings per port. This feature is useful in multi-tenant configurations.

Control Call Answer Time By Port: The number of rings to wait before the Auto-Attendant will answer can be controlled on a port-by-port basis, including immediate and never answer.

Flexible Mailbox Numbering (Dial plan): In addition to supporting single-digit mailboxes (1 - 8), a mailbox dial plan of 2, 3, or 4 digits can be selected.

Directory: Also known as Name Dialing. Callers may access a mailbox directory where they will be able to reach a mailbox owner by dialing the person’s first or last name rather than their mailbox number. The system can be configured for either first or last name dialing (but not both at the same time).

Caller Type-Ahead: Callers who are familiar with the system may enter their key pad selections without waiting for the system prompts.

Operator Revert: Callers may reach a live attendant at any time by dialing "0".

Fax Finder: Detects an incoming fax tone and directs it to the fax mailbox/extension.

Operator Transfer to a Mailbox: Allows an operator to transfer an outside caller to a specified mailbox where the caller will immediately hear the subscriber’s personal greeting and will be prompted to leave a message.

Transfer to Any PBX Extension: Allows the user to dial any internal extension defined in the PBX.

Quick Message Feature: Allows a caller reaching the auto-attendant to leave a message in a specific mailbox without transferring to the mailbox extension and possibly speaking live with the subscriber.

Multiple Message Capability: Allows an outside caller to leave more than one voice mail message per call, therefore saving on toll charges.

User Programmable Dial 0 Extension: Allows the user to program the dial 0 extension to any internal extension, for example, a personal or departmental secretary. The administrator can override the system default ("0" for the operator) with any valid phone number, including an external number or even a long distance number. The administrator can also override the system default on an extension by extension basis, with any valid phone number.

Voice Mail Features

Personal Greetings/Name: Subscriber name and a personal greeting can be recorded by each mailbox user.

Temporary Greeting: A personal greeting set for a specific number of days (with automatic expiration) can be recorded by each subscriber.

Password Protected Mailboxes: Access to subscriber mailboxes requires a password.

Message Envelope: Played prior to beginning of each message, containing priority type, date, and time (including caller identification for internal and external calls).

Message Length: Unlimited message length with a 5 minute continuation prompt.

Saved Messages: Messages may be saved by a subscriber. They will be automatically purged from the system after 15 days (or as reprogrammed) or you can specify that saved messages are never deleted. New messages are never purged automatically. The saved messages are played in last-in first played order.

Message Review: Allows immediate replay of a message, including message envelope (timestamp, calling party information). The timestamp states the time only for messages less than a day old. Messages less than a week old include the day; messages older than a week include the date.

Message Erase: Allows immediate deletion of a message from the system. The message cannot be subsequently restored; deletion is immediate and permanent.

Message Reply: Allows immediate reply to a message received from another internal mailbox subscriber.

Message Forward: Allows messages to be forwarded to other subscribers and distribution lists with or without a pre-pended comment.

Message Rewind/Hold/Fast Forward: Allows subscribers to rewind, fast forward, or pause messages for several seconds.

Music on Hold: Provides the ability to mimic a RAD.

Urgent Messages: The message receives priority placement in the listener's mailbox.

Private Messages: The message cannot be forwarded to another subscriber's mailbox.

Certified Messages: On internal calls, the sender will be notified when the recipient has read the message.

Message Send Actions: Callers will have the ability to review, re-record, and append to a message before sending it. A message can also be cancelled prior to sending.

Memo: Subscribers will have single-digit access to send a message to their own mailbox, for future reminders and memo-type messaging.

Message Notification: The subscriber will be notified that they have received a message by the message light on their phone (MWI), and optionally by setting the notification type to one of the following options, which will cause Express Messenger to call:

- the mailbox’s associated extension number, for analog phone extensions or phones without a message light (prompts called party to log into their mailbox).
- an outside number (prompts called party to log into their mailbox).
- a message pager (plays an audio message indicating messages are waiting).
- a tone-only pager (simply hangs up after a far connection is made).
- a digital pager (plays DTMF digits corresponding to a system-wide callback number along with the specific mailbox number).

Notification options may be changed by the system administrator. They may also be modified by the mailbox owner if permission is granted by the system administrator. In addition to the notification type, the phone number and schedule are configurable. The schedule determines whether paging occurs:

- around the clock, regardless of the business schedule.
- only during open business hours.
- only during closed business hours.
- never (disabled until the schedule is changed to one of the three previous schedule options).

Finally, a mailbox may be configured to do non-MWI notification only in response to urgent messages (as opposed to all messages).

By default, a busy or no answer condition detected on a notification call will result in two additional retries occurring at 15 minute intervals. All notification results are posted to the system log file.

Outside Message Notification Calls: The administrator will configure a trunk access code for use in all outside notification calls. The trunk access code will control the lines to be used for notification.

Distribution Lists, Broadcast Message: Allows four system-wide and five (per mailbox) personal distribution lists as well as a broadcast message facility to deliver a message to all mailboxes.

New mailbox Tutorial: The system will guide the user through the steps required for initial configuration of mailbox, including specification of a (non-default) passcode and recording of a personal greeting and name.

Mailbox Types: Six mailbox types are available:

- Extension - the auto-attendant will transfer a caller to the mailbox's associated extension. If the called party is busy or does not answer, the caller will be prompted to leave a message in the mailbox. The extension mailbox may be linked to other mailboxes for transfer only (dual mailboxes). This permits the caller to transfer to other mailboxes in the same department.
- Message-Only - the auto-attendant will not attempt a transfer but will immediately prompt the caller to leave a message in the mailbox.
- Transfer-Only - the auto-attendant will transfer a caller to the mailbox's associated extension but will not take a message if the called party is busy or does not answer.
- Information-Only - the auto-attendant will only play the mailbox greeting; no transfer or prompt to leave a message will occur.
- Guest mailbox - the guest can use basic voice mail functionality. Guests can play messages, create their own greetings, and set up their own wake-up calls.
- Front desk mailbox - the front desk attendant can administer the guest mailboxes.

Hospitality Option for Hotel or Motel Applications

In Mitel Express Messenger Release 2.1 systems or later, option provides two new types of mailboxes: guest mailbox and front desk mailbox.

A guest mailbox provides a guest with basic voice mail functionality. Guests can

- play messages
- create their own greetings
- set up their own wake-up calls.

A front desk mailbox allows the front desk attendant to administer the guest mailboxes. The front desk attendant can

- help guests access their messages
- select the language that the guest prefers to access the mail system
- set the status of a guest mailbox to checked-in

- set the status of a guest mailbox to checked-out
- move a guest's messages to another mailbox
- access a guest's mailbox
- use the standard mailbox features (the Front Desk Menu provides access to the Main Menu of the voice mailbox).

Multiple Languages

Prompts in English, French, or Spanish

Mitel Express Messenger supports English (North American), French (North American), and Spanish (North American).

The system is always shipped with the voice mail prompts in English. To enable Spanish or French prompts, you configure this option through Express Manager and then restart the system.

Spanish prompts are supported with MEM Release 2.1 systems or greater. You cannot upgrade Release 1.0 or Release 2.0 systems to support Spanish prompts.

French prompts are supported with MEM Release 3.0 or greater.

Bilingual Support

Three languages are supported:

- English (North American)
- Spanish (North American)
- French (North American).

MEM Release 3.0 systems or later support these three languages.

The administrator chooses two languages in Express Manager (one language being the default), and then re-boots the system for the language selection to take affect.

The caller hears a welcome greeting and then a prompt instructing the caller how to change the language for the prompts that follow.

The user retrieving the message and hears the prompts in the language selected by the administrator.

Record a Call

Record a Call allows the user to record an internal or external two-party conversation at their set. The recording process may start automatically or manually. If manual, the user presses a Record a Call softkey or a hardkey (depending on the set) to initiate the recording. While the conversation is recording, the user may pause, resume, save, or erase the recording.

The recording is stored as a recorded conversation in the user's voice mail mailbox.

See Record a Call for more information.

Support for Softkey-equipped Mitel Telephones

Mitel Express Messenger systems with Release 2.1 software of later support voice mail softkeys.

Users of Mitel telephones equipped with softkeys can use them instead of dialing codes to select Mitel Express Messenger menu options. For example, to listen to message, a user can press the Play Message softkey instead of dialing the digit 7. Note that softkeys are not available for all menu options.

See Softkey Support for more information.

Express Messenger Hardware and Software

The Express Messenger card is available in four configurations

- PN 9109-080-001-NA - supports 2 voice mail ports
- PN 9109-080-002-NA - supports 4 voice mail ports
- PN 9109-080-009-NA - supports 6 voice mail ports
- PN 9109-080-008-NA - supports 8 voice mail ports.

The following software is available for the system

- PN 9109-080-007-NA - Mitel Express Manager Software
- PN 9109-540-001-NA - Mitel Express Messenger Release 2.0 Software
- PN 9109-542-002-NA - Mitel Express Messenger Release 2.1 Software.

The following optional software is available:

- PN 9109-542-100-NA - Mitel Express Messenger Hospitality Option (Note that this option requires Mitel Express Messenger Release 2.1 system software or later).

Upgrading an Express Messenger Card to Support Additional Ports

You can upgrade a card to support up to 8 ports by enabling software that is on the card with a passcode. To upgrade a card to support additional ports (for example, 4 ports to 6 ports), refer to the Mitel Express Messenger System Administration Manual (PN 9109-080-025-NA) for instructions.

Note: A Release 1.0 card stores approximately 32 hours of messages. A Release 2.0 card stores approximately 100 hours of messages. If you upgrade a Release 1.0 card with Release 2.0 software, the message storage remains at 32 hours because the message storage limit is card dependent.

Programming

Programming the SX-200 ICP

1. In CDE Form 01 (Configuration), program an Express Messenger card as a Digital Line Card into its assigned slot.
2. In CDE Form 02 (Feature Access Codes), program the following access codes as required by the SX-200 ICP:

   3 Call Forwarding - All Calls (COS Option 260, Internal/External Split Call Forwarding must be DISABLED)

   4 Call Forwarding - Internal Only (COS Option 260, Internal/External Split Call Forwarding must be ENABLED)

   5 Call Forwarding - External Only (COS Option 260, Internal/External Split Call Forwarding must be ENABLED)

   24 Abbreviated Dial Access

   32 Automatic Wake-up (if required for the Hospitality option)

   41 Send Message

   Note that you must also program the codes assigned to Feature Access Codes 41 and 32 into the Express Messenger application from Express Manager.
3. In CDE Form 03 (Class of Service Options), assign the required COS Options to the COS assigned to the Express Messenger card. Assign COS Options for the sets (ports).

Form 03 - COS Define	
COS Option Number and Name	**Status**
Voice Mail Users	
206 - Call Forwarding - Busy	ENABLED
207 - Call Forwarding - No Answer	ENABLED
208 - Call Forwarding - External	ENABLED
212 - Can Flash if Talking to an Incoming Trunk	ENABLED
213 - Can Flash if Talking to an Outgoing Trunk	ENABLED
232 - Message Waiting Setup - Lamp	ENABLED
245 - Abbreviated Dialing Access	ENABLED
246 - SMDR - Extended Record	ENABLED
259 - Message Sending	ENABLED
260 - Internal/External Split Call Forwarding	ENABLED
264 - Half Fwd NA Timer for DID Call When VM msg on	ENABLED
702 - SMDR - Overwrite Buffer	ENABLED
804 - SMDR - Drop Incomplete Outgoing Calls	ENABLED
806 - SMDR - Record Incoming Calls	ENABLED
For Softkey support	
267 - Softkey support for Voice mail	ENABLED
To set Wake-up calls	
202 - Alarm Call	ENABLED
322 - Confirm Wake-up by Offhook (required for display sets only)	ENABLED
COV/DNIC Voice Mail Ports	
212 - Can Flash if Talking to an Incoming Trunk	ENABLED
213 - Can Flash if Talking to an Outgoing Trunk	ENABLED
216 - Data Security	ENABLED
229 - Voice mail Port	ENABLED
238 - Override Security	ENABLED
245 - Abbreviated Dialing Access	ENABLED
259 - Message Sending	ENABLED
301 - Camp-on	ENABLED
606 - Telephone - Enhanced Answering Position	ENABLED
609 - Telephone - Night Service Switching	ENABLED

Form 03 - COS Define		
COS Option Number and Name	**Status**	
502 -Display ANI/DNIS/CLASS Information		ENABLED
702 - SMDR - Overwrite Buffer		ENABLED

4. In CDE Form 04 (System Options), enable the required System Options.

Form 04 - System Options		
System Options	**Status**	**Option Number**
For Softkey Support		
DTMF ON Timer	9	69
Support Softkey Access to Voice mail	ENABLED	97
To Set Wake-up Calls		
Automatic Wake-up	ENABLED	11
Automatic Wake-up Alarm	ENABLED	12
Automatic Wake-up Print	ENABLED	13
Automatic Wake-up Music	ENABLED	14

5. In CDE Form 09 (Desktop Device Assignments) program the ports as SS430 S/ATT sets. Only this set type will permit all the functions Mitel Express Messenger needs. The extension number chosen for these sets must not contain * or # characters.

   Once each port is programmed, expand it and change the line preference to manual.
6. In CDE Form 17 (Hunt Groups) program the Express Messenger ports as members of one or more hunt groups. Set the hunt group type to “terminal”. Assign an access number for each hunt group.

CAUTION: The following ports of an Express Messenger card CANNOT be programmed as phantom lines.
- ports 3 through 12 of a 2-port card
- ports 5 through 12 of a 4-port card
- ports 7 through 12 of a 6-port card
- ports 9 through 12 of an 8-port card.

7. In CDE Form 19, Call Rerouting Table, program Station Dial 0 Routing to route Dial 0 calls to the console.
8. Ensure that users forward calls to the Express Messenger voice mail hunt group access number.
9. Complete the Express Messenger installation and programming as described in the Express Messenger Installation Sheet and Express Messenger System Administration Manual.

Troubleshooting the Express Messenger Card

About the Card LEDs

There are 10 LEDs on the card faceplate:

STATUS
LINE 1
LINE 2
LINE 3
LINE 4
DIAG 1
DIAG 2
DIAG 3
DIAG 4
ALARM

The STATUS LED is a tri-color LED that indicates the overall status of the card and the status of the hard drive write activity.

LINE 1 to LINE 4 and DIAG 1 to DIAG 4 are bi-color LEDs that show port hookstate status after the voice mail application has initialized. If the power-up self-test fails, these LEDs will provide details on the failure.

The ALARM LED functions similarly to the Alarm LED on a standard Mitel peripheral interface card (PIC), indicating an alarm signal sent from the SX-200 ICP to the PIC.

System STATUS LED States

The states that can be displayed by the system status LED are listed in the following table.

STATUS LED	
STATUS LED State	**System State**
Dark (off)	No Power.
Blinking Amber	Power-on Self Test in progress.
Red	Self Test Failed, see port LEDs for failure code.
Green	Self Test has passed, system is operational.

If the Power-on Self Test fails (indicated by the system status LED being RED), the port status LEDs will display a failure code to indicate what components failed diagnostics. Each LED within the eight LEDs will show the result of one of the diagnostics. The following table lists the LED codes and the associated diagnostic that failed.

Power-on Self Test Indicators	
Port LED	**Failure**
LINE 1	DRAM
LINE 2	RTC
LINE 3	EEPROM
LINE 4	EPROM Checksum
DIAG 5	Hard Disk

Power-on Self Test Indicators	
Port LED	**Failure**
DIAG 6	Integration Processor
DIAG 7	DSP
DIAG 8	General Application Error

Port LED Status (LINE 1 to DIAG 4)

After completing the Power-on Self Test, the system is operational. Once operational, the Port LEDs display the state of each voice mail port as shown in the following table.

Port LEDs	
Port LED State	**Port State**
OFF	The port is on-hook and idle.
ON	The port is off-hook and is in use.

ALARM LED Status

The PBX alarm LED will indicate a simple binary condition corresponding to whether or not an alarm signal has been detected by the PBX main processor and sent to the voice mail card.

ALARM LED	
Alarm LED State	**PBX Alarm State**
OFF	The PBX has not detected and signalled an alarm condition on the Express Messenger card.
ON	The PBX has detected and signalled an alarm condition on the Express Messenger card.

Identifying a Faulty Mitel Express Messenger Card

If all the LEDs are on green, the card has not booted properly. Re-seat the card.

If the STATUS LED is red, re-seat the card again. Check the status of the Port LEDs for a Power-on Self Test failure (see the Power-on Self Test Indicators Table). If the card fails the Power-on Self Test, the card is faulty and you must replace it. Refer to Replacing a Mitel Express Messenger Card.

Troubleshooting the Voice mail Application

Refer to the Mitel Express Messenger System Administration Manual.

Replacing a Mitel Express Messenger Card

If the STATUS LED is red, re-seat the card. Check the status of the Port LEDs for a Power-on Self Test failure (see the Power-on Self Test Indicators Table). If the card fails the Power-on Self Test, the card is faulty and you must replace it.

If the card's hard disk is not faulty, you can maintain the Express Messenger database by transferring the hard disk to the replacement card. If the hard disk is faulty, you must restore the Express Messenger database from a backup file to a new hard disk.

The part numbers for the Mitel Express Messenger cards are

- PN 9109-080-001-NA - supports 2 voice mail ports
- PN 9109-080-002-NA - supports 4 voice mail ports.
- PN 9109-080-003-NA - supports 6 voice mail ports
- PN 9109-080-004-NA - supports 8 voice mail ports.

To replace a faulty Mitel Express Messenger Card

CAUTION: Ensure that a recent backup of the Express Messenger database is available. If the hard disk is faulty, you must restore the backup database (typically a file on the PC hard drive) to the hard disk on the replacement card. Refer to the Express Messenger System Administration Manual for instructions.

1. Make sure the PBX ground is connected.
2. Put on the antistatic wrist strap when removing and repacking cards.

   The antistatic wrist strap must be connected to the PBX chassis, which must be connected to an approved ground to provide protection from static discharges.
3. Ensure that all LINE LEDs are off (circuits are idle).
4. Park the hard disk drive as described in the System Administration Manual (from Express Manager select the Maintenance Menu and then select Shutdown).
5. Remove the card by using the extractor as a lever and pulling the card towards you.

   Each card has one card extractor. The extractor helps seat the card firmly in the backplane.

   The extractor provides leverage to pull the card free of the backplane connector.
6. If the hard disk on the card is not faulty, remove it from the card. The hard disk contains your voice mail database. If the hard disk is faulty, do not remove it from the card.

   To remove the hard disk (see the Express Messenger card perspective Figure), remove the four screws that fasten the hard disk to the card. Then, disconnect the ribbon cable from the card.
7. Unpack the replacement card. If you removed the hard disk from the faulty card, transfer this hard disk to the replacement card.

   The replacement card is shipped with a hard disk. Keep this disk as a spare.
8. Install the replacement card into the card slot.
9. Check the STATUS LED to ensure that the card boots up correctly (see Table).
10. If hard disk on the card is new (blank), you must restore a backup Express Messenger database file to the hard disk. Refer to the Express Messenger System Administration Manual for instructions.
11. Pack the faulty card in the package that the replacement card was shipped in. Packaging is shown in the Printed Circuit Card Figure. When handling a printed circuit card:
    - Handle the cards by its edges only, except when seating connectors.
    - Do not touch the gold edge connectors.
    - Avoid contact with any exposed electrical connections.
    - Ground the antistatic packaging before putting a card in it.

- As soon as you remove a card from a slot, place it in antistatic packaging to prevent further possible damage
- When you are finished, replace the anti-static wrist strap in the cabinet.

Program CDE

Introduction

After the successful mechanical installation of the SX-200 ICP (refer to the Installation Information section), the system is ready for programming. This section describes the Customer Data Entry (CDE) package and outlines the procedures for entering customer data entry mode.

The system is programmed in groups. Classes of service group together users with the same COS restrictions. Pickup groups assemble users within a group. Hunt groups classify users together with a common knowledge about how to handle certain calls. Trunks are grouped together with common outgoing characteristics.

Each device type should be assigned to its own Class of Service (COS).

Tenant groups also relate to this grouping theme. Tenant groups facilitate separate attendant services for different areas of a corporation. These services include handling “DIAL 0” calls locally, routing and recalling incoming trunk calls as required and locally switching to night service mode.

One variant of this multi-tenant operation occurs when each group are separate customers that cannot access each other’s trunks. Another variant occurs when DID trunk service handles incoming calls into a number of different customers. For this plan, each customer requires identification of its incoming calls.

Equipment Used For Customer Data Entry

The programming of the database is supported by three devices: a PC, the attendant console, and an ASCII CRT terminal (VT100 compatible).

PC Requirements

You can use a desktop or laptop computer to program the SX-200 ICP on-site via a serial connection or from a remote location via a web browser or secure Telnet. The computer must meet the following requirements:

- Windows NT/98/2000/ME/XP PC or laptop (Windows 98 does not support secure Telnet)
- for serial connections, a VT-100 emulator or other communications program
- a serial port and serial cable
- a Network Interface Card (NIC) and Ethernet cable
- for remote connections, either
  - a secure Telnet client that supports SSL/TLS (the Mitel Telnet Client is recommended)
  - a web browser (Internet Explorer 6.x and Mozilla Firefox are supported)

For local connections, the management computer connects to the Maintenance port of the SX-200 ICP. For remote connections, the management computer connects to port 2000 of the SX-200 ICP.

CDE and MTCE Access Levels

There are five levels of access to the CDE and to the MTCE. Each level has its own username, password and privileges. For CDE programming steps, see Form 28, Form Access Restriction Definition. For MTCE programming steps, see Setting Password.

Serial Connection to the Controller

A desktop or laptop computer can be connected directly to the Maintenance port of the SX-200 ICP with an RS-232 serial cable. A VT-100 emulator or other communications program is required to access CDE/MTCE.

To connect to the controller using serial cable:

1. Connect an RS-232 straight DTE serial cable between the PC's serial port and the controller's Maintenance port.
2. Program the PC's serial port (from the communication program) with the following settings:

   Baud Rate: 9600 (to change, see Setting the Maintenance Port Baud Rate)
   Stop Bits: 1
   Data Bits: 8
   Flow Control: None
   Parity: None
3. Verify the connection as follows:

   a) In the VT-100 emulator or other communications program, press RETURN.

   b) If the connection is functioning, a prompt to "Press Return Four Times to Login" displays.

While the maintenance session is active, do not disconnect the serial cable or attempt to open another maintenance session with a Telnet connection. Doing so will cause an error message stating that CDE is currently in use.

Secure Telnet Connection to the Controller

Telnet is a terminal emulation program for TCP/IP networks such as the Internet. To enable a Telnet connection between a PC and the controller, the PC must be equipped with a secure Telnet client that supports SSL/TLS. The Mitel Telnet client is recommended.

To connect to the controller using a secure Telnet client:

Note: Two logins are required to open a session. The first connects to the secure telnet port of the controller. The second connects to CDE or MTCE.

1. Connect the management PC and the controller to the LAN.
2. Launch the secure Telnet client. Use the Mitel Telnet client (included on the software CD-ROM), or any other Telnet client that supports SSL/TLS.
3. Enter the hostname or IP address of the Controller, and port number 2000.

   For example, to open a connection with the Mitel Telnet client, enter: open 192.168.2.25 2000
4. Enter a username and password to log in to the secure telnet port. (Default is *installer/1000*.)
5. Specify the terminal type:

   1- VT100 COMPATIBLE
   2- TTY TYPE
   3- IBM PC (See Note below)
6. Specify the application:

   1- Maintenance
   2- CDE
   3- Quit

You must specify the terminal type and application within ten seconds after logging in to SSL/TLS. Otherwise, the session will timeout and the Mitel secure telnet client will close.

7. Enter a username and password to log in to CDE or MTCE. (Default is *installer/1000*.)

   Note: While the maintenance session is active, do not disconnect the Ethernet cable or attempt to open another maintenance session with a serial connection. Doing so will cause an error message stating that CDE is currently in use.

After log in, the terminal displays the top level CDE form. To select a form, enter <form-number>, then press Enter. To select a softkey, enter Esc-<softkey-number>.

SX-200 ICP Web Interface Connection to the Controller

A web browser can be used to connect to the SX-200 ICP from the LAN and manage the system using the SX-200 Web Interface.

The SX-200 ICP Web Interface provides the following functionality:

- The interface is divided into three panels:
  - the left panel contains the CDE/MTCE form menu
  - the center panel contains the main CDE/MTCE form interface
  - the upper panel contains an information display (alarm status and software version) and buttons to configure user preferences (font settings and contrast mode), display online help, and exit the application
- Users can select forms and softkeys by clicking on them, or by entering shortcut key commands.
- The online help system is context sensitive. When a user clicks the help button, a page displays containing information pertaining to the CDE form or MTCE command currently in use.

Figure: SX-200 ICP Web Interface

Before you begin, configure the management PC to support the connection:

- disable your web browser's pop-up blocker software
- install Sun Java plugin version 1.5 or later

To connect to the controller using the SX-200 Web Interface:

1. Connect the management PC and the controller to the LAN.
2. Launch a web browser (IE 6.x or Mozilla Firefox).
3. Enter http://<controller IP or hostname>

   For example, http://192.168.1.2
4. Enter a login name and password to log in to the SX-200 Web Interface. (Default is *installer/1000*.)

   A CDE session will open.

Note: While the session is active, do not disconnect the Ethernet cable or attempt to open another maintenance session with a serial connection. Doing so will cause an error message stating that CDE is currently in use.

Establishing a Management Connection from the Internet

To establish a management connection to the system from a PC located on the Internet, you need to open a hole in the firewall that protects the LAN from the outside world. For systems that connect to the Internet through their LAN interface (some CX/CXi and all MX Controllers), program the dedicated firewall on your network to permit remote access. For a CXi controller that connects to the Internet through its Internet Gateway (WAN interface), program the internal firewall to perform either:

- PPTP Remote Access (recommended method)
- Firewall Forwarding (NAT Redirect)

IP Port Usage

If the SX-200 ICP is operating behind a firewall, you may need to open the following ports.

Table: SX-200 ICP IP Port Usage

Port Range	Transport	Purpose
22	TCP	AMC Communications
53	UDP	DNS
67	UDP	DHCP Server
68	UDP	DHCP Client
69	UDP	TFTP
80	TCP	HTTP
443	TCP	HTTPS access to: - SX-200 ICP Web Interface - MAS Server Manager
1066	TCP	IP trunk signalling (IP networking)
1067	TCP	Secure IP networking
2000	TCP	Telnet to CDE/MTCE
2005	TCP	Telnet to MAS Server
6543	TCP	IMAT
6800	TCP	MiNet Server
6830	TCP	VM CMPS Server
6900	TCP	MiNet Client
8000	TCP	MiTAI
8001	TCP	MiTAI (SSL)
9000	UDP	IP Phones - Rx B1
9002	UDP	IP Phones - Rx B2
50000 to 50127	UDP	E2T IP voice stream
61320 to 61328	TCP	User Defined (Hotel PMS/Call Log)

ASCII Terminal

An ASCII Terminal can be used for local programming only; remote programming requires a PC. The terminal connects via an RS-232 connection to the Maintenance connector on the SX-200 ICP controller.

A VT100 compatible terminal displays the full screen version of the CDE forms. Forms consist of a header line, 12 lines of data, the command line, and two rows of softkeys which are selected by pressing ESC and the softkey number (1 to 0).

This Figure shows the 4 main areas of a typical CDE form on a terminal interface or PC:

1. The column title area is used to title the columns of information in the CDE form.
2. The display area is used to display up to 12 lines of information. The cursor (the line of data between the 2 angle brackets **>** and <) points to the line of information which may be modified.
3. The cursor line area repeats the line of data marked by the cursor, and contains data which may be modified.
4. The softkey area is usually 2 lines, and contains the softkeys used to perform actions within the form.

Figure: CDE Terminal/PC Display Areas

Attendant Console

On-site customer data entry can be performed via the attendant console. The console's softkeys and display facilitate this task. The display has four lines of 80 characters. These lines are: the header line, the command line (which displays the data that can be edited) and two lines for the 10 softkeys. Note that there are some forms which have two header lines and only one line for the softkey display.

This Figure shows the 3 main screen areas of a typical CDE form on a terminal interface; they are:

1. The column title area is used to title the columns of information in the CDE form.
2. The display area contains data which may be modified.
3. The softkey area is usually two lines, and contains the softkeys used to perform actions within the form.

Levels of Access to Customer Data Entry

The system provides five password protected levels of CDE access. These levels are, in descending order of priority:

- Installer
- Maint1
- Maint2
- Supervisor
- Attendant

The access for any of these levels (except Installer) can be set to "read/write access", "no access" or "read only access" for each CDE form. Program the levels of access on Form 28, Form Access Restriction Definition.

An attendant may be restricted, for example, to station number moves and review of pickup groups only. Similarly, a maintenance person may be given access to class-of-service modifications and station/Mitel telephone additions but not to ARS programming. Installers must be able to access the entire database.

When programming from a console, the user can exit CDE mode (for Call Handling) by pressing any hardkey on the attendant console. Pressing the FUNCTION key and then the APPLICATION softkey automatically returns the console to CDE mode.

Operation

CDE Access from a Terminal

The login procedure for initial CDE access (from a terminal) consists of four basic steps. These steps are:

1. Specify the terminal type:

   1- VT100 COMPATIBLE
   2- TTY TYPE (suppresses graphic characters)
   3- IBM PC

2. Select the function:

   1- MAINTENANCE
   2- CDE
   6- QUIT

3. Enter the level of access:

   INSTALLER,
   MAINT1,
   MAINT2,
   SUPERVISOR or
   ATTENDANT

4. Enter the password (the default password is *1000*).

After log in, the terminal displays the top level CDE form - a list of the names and numbers of the available forms (see example in this Figure). Forms and system options that are not available are marked RESERVED, and cannot be accessed.

Note: Programming can be done in any order, however, Form 4 must be completed to enable purchased software options.

Figure: CDE Top Level Form

The command line displays ENTER FORM NUMBER:. To select a form, enter <form-number>, then press Enter. To select a softkey, enter Esc-<softkey-number>.

CDE Access from the Attendant Console

The login procedure for initial CDE access from the attendant console consists of the following steps:

1. Press the FUNCTION key.
2. Press the APPLICATION softkey.
3. Press the CDE softkey.
4. Select a level of access.
5. Enter the password (default is *1000*).
6. Press the ENTER softkey.

When the CDE application has been selected, the console LCD displays the top level CDE form. See the figure below.

Figure: Available Forms: Attendant Console Display

The lower command line displays ENTER FORM NUM:. Select a form by entering a valid form number. It is not necessary to have the desired form number displayed on the upper command line. Press the ENTER softkey.

Form Editing

General

The forms in the CDE package have several columns and lines of information. On the attendant console, or on a terminal that has cursor control keys, the left and right arrow keys (← and →) move the cursor from field to field on the command line. On a terminal, the **TAB** and **DEL** keys perform an equivalent function. Note that both the **DEL** and left arrow keys delete edited data as the cursor moves left. The up and down arrow keys move the cursor up and down the form. On a terminal, the **LINE FEED** key also moves the cursor down the form. Note also that the **RETURN** key on a terminal performs the same function as the ENTER softkey. On the attendant console, cursor movement is indicated by the underscore character (_). On the terminal, cursor movement is indicated by a flashing solid block and by a line pointer (represented by **> <** characters at the ends of the screen line). To select a softkey on the terminal, press Esc followed by the softkey number.

From the Top Level CDE Form

On the terminal interface, the line at the cursor position is displayed on the command line. Press the cursor control keys to move the cursor through the list a line at a time. When the cursor reaches the bottom (or top) data line, the list will scroll up (or down) if there are more items on the list to display. Press the **TOP** or **BOTTOM** softkeys to move immediately to the top or bottom of the list. Press the **TO MTCE** softkey to go to a Maintenance session (requires matching access levels for the CDE and MTCE).

On the console interface, the word FORMS is on the header line. Under this are two command lines and one row of softkeys as shown in this Figure. The upper command line displays the names and numbers of the first two available forms. Press the cursor control keys to display the names and numbers of subsequent forms, two at a time. Press the TOP or BOTTOM softkeys to move immediately to the top or bottom of the list. Refer to the Customer Data Entry Forms table below for the complete list of available forms for each software generic.

Exit from CDE

To exit from CDE, press the QUIT softkey at the forms level. The terminal returns to the application level; the system is now ready for another application (such as Maintenance).

Customer Data Entry Forms

Table: Customer Data Entry Forms

Form Number	Form Name
01	Feature Access Codes
02	COS Define
03	System Options/System Timers
04	Tenant Interconnection Table
05	Tenant Night Switching Control

Table: Customer Data Entry Forms

Form Number	Form Name
06	Console Assignments
07	Attendant LDN Assignments
08	Desktop Device Assignments
09	Pickup Groups
10	Data Circuit Descriptor
11	Data Assignment
12	Trunk Circuit Descriptors
13	Non-Dial-In Trunks
14	Dial-In Trunks
15	Trunk Groups
16	Hunt Groups
17	Miscellaneous System Ports
18	Call Rerouting Table
19	ARS: COR Group Definition
20	ARS: Day Zone Definition
21	ARS: Modified Digit Table
22	ARS: Route Definition
23	ARS: Route Lists
24	ARS: Route Plans
25	ARS: Digit Strings
26	ARS: Maximum Dialed Digits
27	Form Access Restriction Def’n.
28	DTE Profile
29	Device Interconnection Table
30	System Abbreviated Dial Entry
31	CDE Data Print
32	Account Code Entry
33	Directed IO
34	Global Find Access Code
35	Modem Assignment
36	Guest Room Key Template
37	ACD Keys Template
38	ACD Agent Groups
39	ACD Supervisors
40	ACD Paths
41	T1 Link Descriptors
42	T1 Link Assignment
43	System Configuration

Table: Customer Data Entry Forms

Form Number	Form Name
44	Network Synchronization
45	Not Used
46	Key System Toll Control
47	IP Networking
48	Voice Networking
49	Voice mail
50	Mailboxes
51	Voice mail Distribution Lists
52	Email
53	Bay Location Assignment
54	Calling Party Number
55	Digit Translation Table

Softkeys Available in Most CDE Forms

The following softkeys appear in most forms. They have the same purpose, regardless of which form they appear in.

QUIT: Pressing the QUIT softkey exits from the current form and returns the display to the previous - another form, or the level where the forms are selected. Also, if another softkey was activated, pressing the QUIT softkey returns the display to the previous state.

CANCEL: This softkey appears after a programming error has occurred. Pressing the CANCEL softkey returns the display to the level where the programming error was made. The CANCEL softkey appears with an error message. Refer to Programming Error Messages Appendix for a list of these error messages.

BAY/SLT/CCT: Instead of moving the line pointer to the desired line of the form, the programmer can call it up directly by specifying the bay, slot and circuit. Pressing the BAY/SLT/CCT softkey displays Bay: Slot: Circuit: on the command line. The cursor appears to the right of the Bay: prompt. A single digit specifies the bay location. When a valid digit has been entered, the TAB or → cursor key can move the cursor to the Slot field. If the programmer enters an invalid number, the system inhibits subsequent cursor movement. Use the DEL or ← cursor key to delete the incorrect entry. When the slot number has been entered, the ENTER softkey appears.

DELETE: This softkey appears when the command line is displaying data. Pressing the DELETE softkey followed by the ENTER softkey removes the selected entry from the form.

ENTER: This softkey appears only after data for an entry has been modified. Pressing the ENTER softkey stores the change in the database. **Note:** in some forms, it is necessary to press ENTER for every change. Form 01 is one exception to this rule.

TOP: Pressing the TOP softkey moves the line pointer to the first line of the form. The command line displays the first line.

BOTTOM: Pressing the BOTTOM softkey moves the line pointer to the last line of the form. The command line displays this line.

**** MORE ****: When the MORE softkey is pressed, a new set of softkeys are displayed. Most forms with this softkey have two sets of softkeys; some forms have three.

Programming Aids

After you have installed the circuit cards, and before you begin CDE programming, plan the CDE programming requirements using the blank CDE forms.

Remote Printing of CDE Reports

CDE reports are generated and captured on the remote terminal so the user can obtain a softcopy of the reports and print them. This functionality is dependant on System Option 102, Feature level 2. The technician uses Form 32, CDE Data Print and the tools provided on the PC to capture and print the file.

The CDE reports are generated on the PC screen with the use of CDE Form 32, CDE Data Print. The softkeys: DISPLAY and DISPLAY ALL. are equivalent to the PRINT and PRINT ALL softkeys; causing the selected print option to be executed but with the data reflected back to the terminal. Once the CDE report is generated on the screen, the screen refreshes to show Form 32 in the state before making the request. The file is captured within the terminal emulation program and the captured file can be opened with the word processor.

CDE Form Descriptions

Note: Programming can be done in any order, however, Form 4 must be completed to enable purchased software options.

Default Database Configuration

The CDE Forms are factory-set with default values that make it easier and faster to program the system. The defaults allow you to install the SX-200 ICP in a square KTS (key telephone system) configuration with up to 20 IP phones and two analog terminals (phone, fax, or modem) and make extension-to-extension calls without doing any programming. You will also be able to receive fax and modem calls, but will have to program ARS to make external calls from ONS devices.

Note: The controller is identified as the “IP Bay” in some CDE forms. By default, the IP bay is bay number 1. It becomes bay 8 when an SX-200 EL/ML database is installed to migrate the system to an SX-200 ICP. All references in documentation are to the default number.

The default database includes the following:

Telephone Related

- 3 digit extension numbers that start at extension 100

  Note: The default numbering plan uses three-digit extension numbers. If you require four-digit extension numbers, you will have to reprogram Forms 9, 17, and 50. Or, you can use the blank database provided on the SX-200 ICP software CD. For more information, see Installing the Blank Database.
- IP phone extensions that start at extension 100
- Ports on the Controller (MX and CX/CXi only)

  - 6 LS CLASS

  - 2 ONS on MX (extensions 201 to 202) and 4 on CX/CXi (extensions 200 to 203)

  - 2 DNIC on MX only (extension 198 is the SUPERCONSOLE 1000 and 199 is the subattendant.)

- 7 default classes of service (COS 1 – 7). They are for IP Phones, ONS, Subattendant, Attendant Console, LS/CLASS, Voice Mail, and IP trunks.
- default key programming on the sets for a 6-line square system (MX and CX/CXi only)
- default ring cadences
- all phones assigned to paging group 1
- the handsfree microphone is not automatically turned on when receiving a page (auto-latched).

Voice Mail Related

- (MX and CX/CXi only) - 4 Voice mail ports (Business 1 Option with Dual DSP); 8 ports (Business 2 and Hospitality Options with Dual DSP)
- For AX - 20 Voice mail ports (with Onboard Quad DSP)
- 20 Voice mail mailboxes are assigned with the same extension numbers as the
  - first 16 IP phones (extension 100 to 119)
  - Attendant Console (SUPERCONSOLE 1000; extension 198); MX only
  - Subattendant (extension 199); MX only
- 2 ONS ports (extensions 200 and 201)
- Hunt Group for Voice mail ports with pilot number 300
  - COS 6 and
  - reserve extension 301 to 316 for voice mail port extensions
- system-wide Call Forward No Answer to voice mail, for internal and external calls.
- users are permitted to store 10 messages, which will be automatically deleted after 15 days.

Trunk Related

- trunks in Form 14 are non-dial-in to the CO line keys (MX and CX/CXi only)
- LS trunk circuit descriptor defaulted as CLASS
- six LS trunks programmed to keys on IP Phones (MX and CX/CXi only)
- no ARS, no dial 9 for trunk access

System Related

- default system options
- default feature access codes
- the default music port (located on the analog mainboard) is ON (MX and CX/CXi only)
- the default paging port (located on the analog mainboard) is ON (MX and CX/CXi only)
- the night bell extension is 340 (MX and CX/CXi only)
- SMDR/CDE Print default to ON (outputs through the serial printer port on the MX and IP socket 61328 on the AX/CX/CXi)
- default DHCP settings and a SX-200 ICP Controller default IP address (192.168.1.2) to match (factory-set).

Form 01 - System Configuration

This form specifies the intended location of each peripheral card in the controller's virtual IP bay, plus offboard ASUs, NSUs, and cards in SX-200 peripheral cabinets. Program Form 01 after assigning bay numbers to the bay connections in Form 53, Bay Location Assignment.

If peripheral cards are installed before the system is programmed, Form 01 enables the configuration of the system according to the actual installation. If the system is programmed prior to the peripheral card installation, this form acts as a guide during the installation process. If the installed card type does not match the card type in the PROGRAMMED field, then that device does not function, and the card alarm LED will flash. When the system is programmed, the CDE software uses the PROGRAMMED field of this form to generate a list of physical location (bay, slot and circuit) numbers that can be programmed in subsequent forms. These forms include:

- Form 07, Console Assignments
- Form 08, Attendant LDN Assignments
- Form 09, Desktop Device Assignments
- Form 12, Data Assignment
- Form 14, Non-Dial-In Trunks
- Form 15, Dial-In Trunks
- Form 18, Miscellaneous System Ports
- Form 43, T1 Link Assignment
- Form 44, T1 Network Synchronization
- Form 53, Bay Location Assignment

Notes:

1. By default, the IP bay is bay number 1. It becomes bay 8 when an SX-200 EL/ML database is installed to migrate the system to an SX-200 ICP.
2. When a device is assigned to a location, the system first checks the appropriate card type in this form.
3. For the AX, the slots (1-15) remain same for the IP bay (as for existing 200 ICP platforms). The onboard ASU would be visible as programmed and installed on slot 13; only data socket PLIDS, that is, 1/13/19 to 1/13/28 would be visible in form 12.

   The new embedded bay (ASU bay) will support the existing card types as supported on ASU II, 24 circuit ONSP line card and 4 ONS + 12 LS combo card. All twelve slots are available for programming the cards.

Figure: Form 01 Layout

Field Description

BAY, SLT and **CCT**: These fields specify the location of each card type. The circuit number represents the physical location of each module on the Universal Card.

PROGRAMMED: Specifies the intended location of the required card types. Data can be assigned to the PROGRAMMED field before the cards are installed. The ONS/CLASS Line card is programmed as an ONS LINE CARD. The PRI card (PRI T1 link) is programmed as a T1 Trunk Card.

INSTALLED: The data in the INSTALLED field reflects the actual installed cards installed in the system, except in the case when an ONS/CLASS Line card is installed in the system. The system updates this field; the installer cannot edit it.

COMMENTS: This field stores additional data (a maximum of 20 characters), for the programmer's reference. The system does not use this information for call processing.

Softkeys

The System Configuration form does not identify the type of card bay (peripheral-PER or ISDN) that you are programming. If you select a BAY/SLT/CCT location and the only card softkeys available are T1 ISDN and ONS, then you are programming an ISDN node. If other types of cards are available, then you are programming a PER node. Use the NODE TYPE softkey to program the node type, or to determine the node type for a location that has already been programmed. The softkeys available to each bay are provided in the following Line Cards Table and the Trunk Cards Table.

LINE CARDS: Programs the selected card slot as a line card or as an embedded voice mail card. The softkeys which are presented are dependent upon the type of bay (node): digital bay, digital node (PER), ISDN Node. The following table lists the line card softkeys which are available to each type of node.

Press the desired line card softkey and then the ENTER softkey. The selected card type is displayed in the PROGRAMMED field.

Table: Line Card Softkeys

	Bay Type			
PER Node (Digital Bay)	**IP Bay (SX-200 ICP virtual bay)**	**ISDN Node (PRI card and NSU)**	**ASU BAY**	
Sofkeys	ONS LINE	IP LINE (See Note 1)	ONS LINE (See Note 4)	ONS LINE (See Note 5)
	OPS LINE	ONBOARD ASU (See Note 2)		ONS/LS COMBO (See Note 6)
	DIGITAL LINE			
		VOICE MAIL CARD (See Note 3)		

Notes:

1. You can program IP Line cards in slots 1-5, 7, 9 and 10 only of the IP bay. Each IP Line card supports a maximum of 31 circuits.
2. The embedded voice mail card can only be programmed in the Slots 11 and 12 of the IP bay, and only if System Option 125, Licensed Embedded Voice mail Boxes (Form 04) is set to a value other than zero. The actual number of voice mail circuits (ports) available depends on the number of DSPs installed. See DSP Configuration Options.
3. ONBOARD ASU refers to the circuits in Slot 13 of the IP bay.
4. You can program an ISDN node with a phantom (not functional) ONS card. Use the phantom ONS card to store programming information for a line card that is temporarily out of use.
5. ONS LINE refers to ONS CLASS CARD. Both the 16 port and the 24 port ONS line cards are programmed using this softkey. When installed, the 16 port card is shown in the installed field as ONS CLASS CARD (16). The 24 port card is shown as ONS CLASS CARD.
6. To program a CIM-connected ASU II, program line cards in slot 1 and 2 only.

TRUNK CARDS: Programs the selected card slot as a trunk card. The softkeys which are presented are dependent upon the type of bay (node): digital bay, digital node (PER), ISDN Node, ASU BAY, or Onboard T1. The following table lists the trunk card softkeys which are available to each type of node.

Table: Trunk Card Softkeys

	Bay Type				
PER Node (Digital Bay)	**IP Bay (SX-200 ICP virtual bay)**	**ISDN Node (PRI card and NSU)**	**Onboard T1 (T1/E1 Framer or T1/E1 Combo)**	**ASU BAY**	
Softkeys	LS/GS TRUNK	IP Trunks (See Note 1)	T1 ISDN (See following paragraph for conditions)	T1 TRUNK (See Note 3)	LS/ONS COMBO (See Note 5)

T1 TRUNK (See following paragraph for conditions)			T1 ISDN	
8 CCT CLASS				
6 CCT DID (See Note 2)				

Notes:

1. You can program IP Trunks in slot 6.

2. The 6 CCT DID card is a high power card and is restricted to the upper slots of an SX-200 FD cabinet. The system generates an error message if an attempt is made to program it in any other slot. The 6 CCT DID card is not restricted in an SX-200 RM cabinet.

3. For Onboard T1 on the MX, program "T1 Trunk" in slots 2 and 4 of the module in MMC1, and slots 6 and 8 of the module in MMC2. For the CX , program "T1 Trunk" in slot 2 of the module in MMC1, or slot 6 of the module in MMC2. For the AX , program "T1 Trunk" in slot 2 and 4 of the module in MMC.

4. For embedded PRI on the MX, program "T1 ISDN" in slots 2 and 4 of the module in MMC1, and in slots 6 and 8 of the module in MMC2. For the CX, program "T1 ISDN" in slot 2 of the module in MMC1, or in slot 6 of the module in MMC2. For the AX, program "T1 ISDN" in slot 2 and 4 of the module in MMC1.

5. To program a CIM-connected ASU II, program Trunk cards in Slot 1 and 2 only.

The following conditions apply to T1 Trunk Cards/T1 ISDN:

- You can program up to two T1 Trunk Cards in a bay (slots 5 and 6 in software).
- The ISDN node has two forms, the Gateway and the PRI Card. You can program a second T1 ISDN Card in slot 8 of an ISDN node, providing that the 24th circuit of the other programmed T1 Trunk Card is not programmed as a trunk in Form 14, Non-Dial-In Trunks, or Form 15, Dial-In Trunks.

  Although T1 trunk cards are not installed in the ISDN node, the Application Fiber Controller (AFC) card in the ISDN node simulates either one or two T1 Trunk cards. (With software loads F41.0 and above, four T1 may be programmed in slots 2, 4, 6, and 8 of an ISDN node.) If you select a BAY/SLT/CCT location and the only type of card softkey available is the T1 ISDN softkey, then you are programming an ISDN node.
- Program PRI cards in bays 1 to 7 only.

UNIVERSAL: Designates that card slot as the Universal Card. Two further softkeys are presented for the programming of the Universal card modules: MUSIC PAGER, and E&M MODULE. The Universal Card is a high power card.

Note: The system automatically programs DTMF Receivers when they are installed, provided nothing is already programmed for that circuit. These circuits must be in place in order to program night bell relays and alarms in CDE Form 18.

MUSIC PAGER: Programs the selected module as a Music-on-Hold/Pager Module. The PROGRAMMED field displays MUSIC PAGER MODULE. Each MOH/Pager Module has a power rating of 1. Therefore, a Universal Card can support four of these modules.

E&M MODULE: Programs the selected module as an E&M module. The PROGRAMMED field displays E&M. Each E&M module has a power rating of 3 (a maximum of three per Universal Card).

NODE TYPE: Define the NODE TYPE as PER, ISDN, ONB T1, ASU BAY, or IP. The NSU and the PRI cards in Peripheral Cabinets are programmed as an ISDN node. Onboard T1/E1 modules (T1/E1 Dual Framer and T1/E1 Combo) are programmed as an ONB T1 node.

CONFIGURE: Configure should only be performed with new installations. Before the system can function properly, the PROGRAMMED field must match the INSTALLED field. Pressing the CONFIGURE softkey matches the PROGRAMMED field to the INSTALLED field. Note that devices can be assigned to the cards in the PROGRAMMED field but the card type cannot change. The CONFIGURE softkey cannot be pressed if any device is specified (e.g., defining a station in Form 09, (Desktop Device Assignments) When a new peripheral card is added to the system, it is necessary to manually update the PROGRAMMED field. The INSTALLED field updates upon exiting and re-entering this form.

Note: The 16 and 24 port ONS cards are programmed as ONS CLASS CARD. When installed, the 16 port card is shown as ONS CLASS CARD (16). The 24 port card is shown as ONS CLASS CARD.

VERIFY DATA: Pressing the VERIFY DATA softkey begins a series of system tests on the database. The command line displays a message as each test completes successfully. These messages are:

PLID TO SWID CONVERSION SUCCESSFUL
ALL RECEIVERS ARE IN VALID STATES
ALL HUNT GROUPS ARE VALID
ALL TRUNK GROUPS ARE VALID
ALL PICKUP GROUPS ARE VALID
ALL TELEPHONE KEYS ARE VALID
ALL TRUNK NUMBERS ARE VALID
ARS CMOS
STATIC CMOS
DWA CMOS

If a test fails, the command line displays an error message and creates a maintenance log. Refer to the Troubleshooting section. When all tests are complete, the form reverts to the original softkey display.

The standard softkeys **BAY/SLT/CCT, CANCEL, DELETE, ENTER, **MORE**** and **QUIT** are also provided. **Note:** Before a card can be deleted, any devices programmed on the card, or associated with circuits on the card must be deleted or disassociated first, using the appropriate form.

Form 02 - Feature Access Codes

This form specifies the Feature Access Codes for the system. Feature Access Codes can be a maximum of five digits (except Callback Busy and Executive Busy Override access codes which must be only one digit). Generally, the codes must be unique; they cannot match any listed directory number or other access code in the system (two exceptions: the Callback Busy and Executive Busy Override access codes). See the Feature Access Codes table for a complete list of the features which can be assigned an access code. To check assigned access codes, refer to Form 35, Global Find Access Code. The system updates this form each time a code is entered during CDE.

Figure: Form 02 Layout

Field Description

FEATURE: Lists the feature numbers in numerical order. The FEATURE field cannot be modified.

FEATURE NAME: Lists the names of the features. The FEATURE NAME field cannot be modified. When RESERVED appears in this field instead of the feature name, the feature is not available.

ACCESS CODE: Displays the access code for each feature.

Feature Access Codes

Table: Feature Access Codes

Feature Numbers	Feature Names
01	Account Code Access
02	Auto-Answer Activation
03	Call Forwarding - All Calls
04	Call Forwarding - Internal Only
05	Call Forwarding - External Only
06	Call Forwarding - I'm Here
07	Call Forwarding - Cancel I'm Here
08	Dial Call Pickup
09	Directed Call Pickup
10	Do Not Disturb
11	Extension General Attendant Access
12	Paging Access To Default Zone(s)
13	Paging Access To Specific Zones
14	TAFAS - Any
15	TAFAS - Local Tenant
16	Hold Pickup Access (Attendant Hold Slots)
17	Console Lockout Access Code
18	Not used
19	Direct Inward System Access
20	Callback Busy <<single digit>>
21	Call Hold
22	Call Hold Retrieve (Local)
23	Call Hold Retrieve (Remote)
24	Abbreviated Dial Access
25	Clear All Features
26	SUPERSET Telephone Loopback Test

Table: Feature Access Codes

Feature Numbers	Feature Names
27	Tone Demonstration
28	ADL Call Setup
29	ADL Disconnect
30	Last Number Redial
31	Executive Busy Override <<single digit>>
32	Automatic Wakeup
33	Call Park
34	Node ID
35	Maid In Room
36	SUPERSET Tel. Room Status Display
37	Direct To ARS
38	UCD Agent Login / Logout
39	Analogue Network Accept Caller’s Extension
40	SUPERSET Tel. Maid In Room Status Display
41	Send Message
42	Call Message Sender of Oldest Message
43	Callback - No Answer
44	ACD Login / Logout
45	Silent Monitoring
46	Flash Over Trunk
47	Program Feature Key
48	Key System - Direct Paging
49	Key System - Group Page Meet-Me-Answer
50	Key System - Direct CO Line Select
51	Key System - Store Personal Speed Call
52	Key System - Retrieve Personal Speed Call
53	Double Flash Over Trunk

Table: Feature Access Codes

Feature Numbers	Feature Names
54	Analogue Network Accept Call Forward Data
55	Analogue Network Accept Call Forward Reason
56	Headset Mode On/Off
57	Call Park Orbit Retrieve
58	IP Set Registration PIN
59	IP Set Replacement PIN
60	IP Set Language Selection
61	Disable Twin Phone
62	Call Park Remote
63	Call Park and Page - Telephone
64	Call Park and Page - PA
65	Phonebook
66	Open Door
67	Digit Translation Table Access
68	Secure Hot swap/unswap
69	DID Number Display

Softkeys

FEATURE NUM: Allows the user to select a Feature Access Code by number. Pressing this key clears the command line and positions the cursor after the ENTER FEATURE NUM: prompt. Entering the 1- or 2-digit feature number displays that access code with its name on the command line. The cursor moves to the start of the ACCESS CODE field on the command line ready for a new access code entry.

The standard softkeys **CANCEL**, **DELETE**, **ENTER**, **BOTTOM**, **TOP**, and **QUIT** are also provided, where applicable.

Parameters arranged alphabetically

The following are not feature descriptions; refer to the Program Features section for complete descriptions.

Abbreviated Dial Access (24): Allows users to dial pre-programmed index numbers rather than having to dial entire digit strings (which can be up to 26 digits in length). The original numbers and their corresponding index numbers must be programmed into Form 31 - System Abbreviated Dial.

Account Code Access (01): This code is dialed prior to the entry of an independent account code.

ACD Login/Logout (44): Allows an ACD position to log in and out at a Mitel telephone. Refer to the ACD TELEMARKETER Application Package section for further details.

ADL Call Setup (28): Allows an Associated Data Line (ADL) configured extension to originate a data call. Refer to the Program Features section for further information.

ADL Disconnect (29): Allows an ADL configured extension to disconnect a data call. Refer to the Program Features section for further information.

Analog Network Accept Call Forward Data (54): Allows the forwarding directory number to be forwarded to a voice mail system. Both the calling party number and the number of the called party that forwarded the call are provided to the voice mail system.

Analog Network Accept Call Forward Reason (55): Allows the reason the call was forwarded to be passed on to a voice mail system. This allows the voice mail system to provide an appropriate greeting if it has the capability.

Analog Network Access Callers Extension (39): Used by the system to display caller's extension numbers on display sets during cross-node network calls. This code is entered into the ARS: Modified Digit Table. Refer to the Program Features section "Analog Networking", for further information.

Auto-Answer Activation (02): This code is dialed prior to the activation (dial 1), or deactivation (dial 2), of the Auto Answer feature.

Automatic Wakeup (32): Allows an extension user to set up a wakeup call without talking to an attendant. The user dials the Automatic Wakeup code, followed by the time (in 24-hour format - 2 hour digits followed by 2 minute digits).

Callback Busy (20): Allows an extension to set a callback on another (busy) extension. The user then hangs up.

Callback - No Answer (43): Allows an extension to set a callback on an extension that does not answer, while listening to ringback.

Call Forwarding - All Calls (03): Allows an extension to forward internal and external calls either: ALWAYS (dial 01), BUSY (dial 02), NO ANSWER (dial 03) or BUSY/NO ANSWER (dial 04). This is available when COS Option 260 - Call Forward Internal/External Split is disabled. The NO ANSWER option allows an extension to forward calls that are not answered within a selected time-out period.

Call Forwarding - Cancel I'm Here (07): Allows a station user to cancel Call Forwarding - I'm Here, from the other station.

Call Forwarding - External Only (05): Allows an extension to forward external calls either: ALWAYS (dial 01), BUSY (dial 02), NO ANSWER (dial 03) or BUSY/NO ANSWER (dial 04). This is available when COS Option 260 - Call Forward Internal/External Split is enabled.

Call Forwarding - I'm Here (06): Allows an extension to redirect its calls to another extension from that other extension.

Call Forwarding - Internal Only (04): Allows an extension to forward internal calls either: ALWAYS (dial 01), BUSY (dial 02), NO ANSWER (dial 03) or BUSY/NO ANSWER (dial 04). This is available when COS Option 260 - Call Forward Internal/External Split is enabled.

Call Hold (21): Allows an extension to put a call on hold and go on-hook. The extension may then be used to make other calls.

Call Hold Retrieve - Local (22): Allows a user to retrieve a held call at the extension where the call was originally held.

Call Hold Retrieve - Remote (23): Allows a user to retrieve a held call from any extension. The user must dial the Call Hold Retrieve - Remote Feature Access Code, followed by the number of the extension where the call was originally held.

Call Message Sender of Oldest Message (42): Allows an extension to call the sender of the oldest message without having to dial the extension number.

Call Park (33): Allows an extension to park an active call, and go on-hook. The extension may not make other calls, but may access paging equipment.

Call Park and Page - Page (64): Allows the user to park a call and initiate a PA page with a single access code.

Call Park and Page - Telephone (63): Allows the user to park a call and initiate a direct page with a single access code.

Call Park Orbit Retrieve (57): Allows the display set user to retrieve calls from a park orbit. This access code cannot be deleted if there are any calls parked in orbit.

Call Park Remote (62): Allows the user to park a call on another set instead of his or her set.

Clear All Features (25): Allows the user to clear features currently activated at the extension with a single access code. The features affected are: all flavors of Call Forwarding, Do Not Disturb and Callbacks.

Console Lockout Access Code (17): Allows an attendant to render the console “harmless” (typically, while left unattended). The console is returned to its normal state by re-entering the code.

Dial Call Pickup (08): Allows a station to retrieve calls ringing other stations in the same pickup group. Stations using this feature must be programmed in Form 10 - Pickup Groups.

Digit Translation Table Access (67): Allows use of the Digit Translation Table for calls on any type of Dial-In trunk. The code is restricted to two digits, maximum, and can be programmed for use by Dial-In trunks in the X field of Form 15 - Dial-In Trunks.

Directed Call Pickup (09): Allows a station to retrieve calls ringing other stations. The user must dial the access code, followed by the extension number of the ringing station.

Direct Inward System Access (19): This is the DISA security code. The DISA access code used with DID and Tie trunks must not lead with an asterisk *, if ANI/DNIS is to be collected. Refer to the Program Features section, Trunk Operation - DISA, Trunk Support - Direct Inward Dial, Trunk Support - Tie, and Trunk Operation - Direct Inward Dial for further information.

Direct To ARS (37): Allows an extension to access ARS immediately, without dialing ARS leading digits. The system automatically dials the Direct To ARS code upon the set going off-hook, or after an account code.

Disable Twinning (61): Allows a user of a phone that is twinned with another phone to turn off/on the ringer and enable/disable the SUPERKEY on the twinned phone. See Phone Twinning for more information.

Do Not Disturb (10): Allows a station to prevent any incoming calls from ringing. This code is dialed prior to the activation (dial 1), or deactivation (dial 2) code.

Double Flash Over Trunk (53): Allows a Centrex extension to be reconnected to the CENTREX extension that it put on softhold while attempting to reach a second CENTREX extension.

Executive Busy Override (31): Allows an extension to override busy extensions by dialing a single code while listening to busy tone.

Extension General Attendant Access (11): This code (usually 0) allows a station to access an attendant directly, without knowing the specific extension number or an LDN number of a specific attendant as defined in CDE Form 19.

Flash Over Trunk (46): Allows an extension to access Central Office CENTREX™ features (sends a flash out over a trunk).

Headset Mode On/Off (56): Allows a set user to control whether the set will be used in headset mode or handset mode by use of a feature key (default is Off).

Hold Pickup Access (16): Allows a station to connect to a held call in an attendant console hold slot. Typically, the attendant will relay the digits to dial: the Hold Pickup Access code, the console/subattendant ID number, followed by the hold slot number, through the system's zone paging equipment.

IP Set Registration PIN: Allows you to assign PIN numbers to register an IP telephone. The Registration PIN is comprised of the PIN access code plus the extension number of the IP set being registered.

IP Set Replacement PIN: Allows you to assign PIN numbers from an IP telephone when replacing it. The Replacement PIN is comprised of the PIN access code plus the extension number of the IP set being replaced.

Key System - Direct CO Line Select (50): Allows an extension to access a specific trunk. The user dials the code, followed by the trunk number. **Note**:This feature can be accessed by multi-line sets only - not by ONS sets.

Key System - Direct Paging (48): Allows a set to directly page another (idle) set through the set's speaker. The user can page a specific set, the entire paging group, or perform an all set page. Page groups are programmed in Form 09 - (Desktop Device Assignments).

Key System - Group Page - Meet Me Answer (49): Allows an extension to respond to a group page by dialing a single access code.

Key System - Store Personal Speed Call (51): Allows an extension to store up to 5 dial access personal speed call numbers.

Key System - Retrieve Personal Speed Call (52): Allows an extension to make a call using previously stored dial access Personal Speed Call numbers.

Language Selection (60): Provides an access code that allows language selection from a PDA phone.

Last Number Redial (30): Allows an extension to use the Last Number Redial feature.

Maid In Room (35): Allows a maid to change the status of the room using the telephone in the room. The maid dials the Maid In Room code, followed by one of: 1-maid in room, 2-maid not in room, 3-room clean, 4-room to be inspected.

Maintenance Functions (18): Not used.

Node ID (34): Allows a uniform numbering plan in a network of systems. Refer to "Analog Networking" in the Program Features section for further information.

Open Door (66): Allows a station to activate relays for a door entry system. The relays are programmed in Form 09 - Desktop Device Assignments by default at bay/slot/circuit 1/13/2.

Paging Access to Default Zones (12): Allows a station to access the default paging zone equipment. The paging equipment must be programmed in Form 18 - Miscellaneous System Ports, and COS Option 312 must be set to a default value.

Paging Access to Specific Zones (13): Allows a station to access specific paging zones. The access code is dialed, followed by the number corresponding to the paging zone number. The station must have access to the zone(s) via COS Options 303 through 311 (paging zone 1 through 9).

Phonebook (65): Allows users of with single-line displays (IP or DNIC) to access the dial-by-name (Phonebook) feature.

Program Feature Key (47): Allows users of Mitel 5010 IP, 5207 IP, SUPERSET 4015, SUPERSET 410, and SUPERSET 3DN sets to program feature keys on their own sets, from their own sets. This eliminates the need to access CDE through the console or terminal interface to make these changes.

Secure Hot swap/unswap (68): Allows a user to swap programming between two devices. The originating device must be a physical device (IP Phone, DNIC set, or station); the destination can be a physical or phantom device. Both devices require a Secure Hot Swap PIN code and must have COS Option 692 (Secure Hot Swapping) enabled. See Secure Hot Swap for more information.

Send Message (41): Allows an extension user to send a message to another extension. The message is in the form of one of: a flashing lamp, a display indication, or a distinctive ringing pattern. Refer to the Program Features section, “Messaging - Call Me Back”, for further information.

Silent Monitoring (45): Allows calls to be monitored without the callers hearing noise from the monitoring set. Refer to the ACD TELEMARKETER Application Package section for further details.

SUPERSET Maid In Room Status Display (40): Allows a Mitel 5020 IP, 5220 IP, 5224 IP, SUPERSET 4025, SUPERSET 4125, SUPERSET 4150, SUPERSET 420, SUPERSET 430, or SUPERSET 4DN station to view room status information.

SUPERSET Room Status Display (36): Allows a Mitel 5020 IP, 5220 IP, 5224 IP, SUPERSET 4025, SUPERSET 4125, SUPERSET 4150, SUPERSET 420, SUPERSET 430, and SUPERSET 4DN user to view room status information. Refer to the Hotel/Motel Feature Package Description section for further information.

SUPERSET Telephone Loopback Test (26): No longer available.

TAFAS - Any (14): Allows a station to answer incoming calls ringing at common alerting devices (night bells) in any tenant group, provided COS option 248 has been enabled.

TAFAS - Local Tenant (15): Allows a station to answer incoming calls ringing at common alerting devices (night bells) within the station’s tenant group, provided COS option 249 has been enabled.

Tone Demonstration (27): Allows a user to listen to all of the possible tones available on the system. Going on-hook terminates the demonstration.

UCD Agent Login/Logout (38): Allows a UCD agent to log in and out of a UCD agent hunt group, to control the arrival of calls from the hunt group. This code is dialed prior to the login (dial 1), or logout (dial 2) code.

Form 03 - COS Define

This form defines the Classes of Service for the system. Classes of Service group together stations with common feature operations and restrictions. The system accommodates a maximum of 50 Classes of Service. Each device (including attendants, data devices and all trunks) are supplied with a Class of Service. COS options are listed in groups. Refer to the Class Of Service Options table for the complete list of COS options.

Figure: Form 03 Layout

Field Descriptions

The header line indicates the Class of Service being programmed and which set of options are selected for either the enabled or disabled options list. The command line displays the current indexed option. When Form 03 - COS Define is selected, the command line displays the first enabled option of the first Class of Service.

OPTION: This field lists the option titles. The actual option names cannot be modified. The option names are classified into two groups: enabled options and disabled options. When RESERVED appears as the option name, the option is not available.

STATUS: This field displays the status of each option; either DISABLED, ENABLED or a timer value.

OPTION NUM: This field displays the number of each Class of Service option. The actual option number cannot be modified.

Softkeys

DISABLE/ENABLE: This softkey enables and disables COS options. The DISABLE softkey appears when the form shows the enabled options list. Pressing the DISABLE softkey followed by pressing the ENTER softkey twice disables the selected option. The ENABLE softkey appears when the form shows the disabled options list. Pressing the ENABLE softkey followed by pressing the ENTER softkey twice enables the selected option.

Notes:

1. For those COS options which have a status other than 'ENABLED' or 'DISABLED', this softkey has no function or indication.
2. If the ENTER softkey is not pressed twice after each selection, softkeys 2 and 3 are not available until the data is completely entered.

COPY COS: Pressing the COPY COS softkey copies the contents of one Class of Service to another. This is useful when two Classes of Service are similar. The command line displays the COPY FROM COS prompt; the user enters the 1- or 2-digit COS number. The command line then displays the TO COS prompt; the user enters the second 1- or 2-digit COS number. Pressing the ENTER softkey twice completes the copy process.

COS NUMBER: Pressing the COS NUMBER softkey prompts the user for a COS number (one or two digits) which selects a specific COS number. The header line displays the new COS number.

OPTION NUM: This softkey selects a specific COS option. Pressing the OPTION NUM softkey displays the ENTER OPTION NUM: prompt on the command line. The selection is completed by entering a valid option number (100 to 908). The command line displays that COS option name, status and number.

SHOW DISABLE/SHOW ENABLE: This softkey has two functions: it displays the disabled or enabled COS options for the selected COS. Pressing the SHOW DISABLE softkey displays the currently disabled COS options. This softkey now shows the SHOW ENABLE prompt and softkey 1 displays the ENABLE prompt. Pressing the SHOW ENABLE softkey shows those COS options that are enabled. This softkey returns to the SHOW DISABLE prompt and softkey 1 displays the DISABLE prompt. Note: Those options with a status other than "ENABLED" or "DISABLED" are listed when the SHOW ENABLE softkey is pressed.

COS NAME: When the programmer presses the COS NAME softkey, the system requests a name for the Class of Service. The COS name can be up to 8 characters in length.

The standard softkeys **CANCEL**, **ENTER**, **BOTTOM**, **TOP**, and **QUIT** are also provided, where applicable.

Class of Service Options

Table: Class Of Service Options

COS Option Numbers	Class Of Service Option Name
100	Attendant Bell Off
101	O/G Restriction/Room Status Setup
102	Attendant Display of System Alarms
103	Attendant DISA Code Setup

Table: Class Of Service Options

COS Option Numbers	Class Of Service Option Name
104	Attendant Flexible Night Service Setup
105	Guest Room Key
106	Attendant New Call Tone
107	Attendant Automatic Call Forward - No Answer
108	Attendant Audible Alarm
109	Attendant Serial Call
110	Attendant Abbr. Dial Confidential Number Display
111	Attendant Abbreviated Dial Programming
112	Attendant Station Busy-Out
113	Attendant Call Block Key
114	Attendant Trunk Busy-Out
115	Attendant-Timed Recall (No Ans) 5 -240 s; 0=Disable
116	Attendant-Timed Recall (Hold) 5 - 240 s; 0=Disable
117	Attendant-Timed Recall (CampOn) 5 - 240 s; 0=Disable
118	Attendant Call Forward - No Answer Timer 10 - 240 s.
119	Attendant Tone Signaling
120	Attendant Conference Disable
121	Station Do Not Disturb
122	Setup Time/Date
123	Call Forward Setup and Cancel
124	Attendant Hold Position Security
125	Attendant Multi-New Call Tone
126	Apply Key Line Conference Warning Tone
150	Subattendant Station Setup Advisory Messages
200	Account Code, Forced Entry - External Calls
201	Account Code, Forced Entry - Long Distance Calls
202	Alarm Call
203	Broker’s Call

Table: Class Of Service Options

COS Option Numbers	Class Of Service Option Name
204	Call Block Applies (Room To Room)
205	Flash For Waiting Call
206	Call Forwarding - Busy
207	Call Forwarding - No Answer
208	Call Forwarding - External
209	Call Forwarding - Always
210	Call Forwarding Inhibit on Dial-In Trunks
211	Call Hold and Retrieve Access
212	Can Flash If Talking to an Incoming Trunk
213	Can Flash If Talking to an Outgoing Trunk
214	Cannot Dial a Trunk after Flashing
215	Cannot Dial a Trunk if Holding or in Conference with One
216	Data Security
217	Direct To ARS
218	Directed Call Pickup
219	Discriminating Dial Tone
220	Do Not Disturb
221	Clear All Features
222	Call Forward Inhibit on Hold Timeout
223	Flash Disable
224	Flash for Attendant
225	Hold Pickup (Attendant Paged Access)
226	Inward Restriction (DID)
227	Lockout Alarm Applies
228	Manual Line (Dial 0 Hotline)
229	Voice Mail Port
230	Message Register Overflow Alarm
231	Message Waiting Setup - Bell

Table: Class Of Service Options

COS Option Numbers	Class Of Service Option Name
232	Message Waiting Setup - Lamp
233	Never a Consultee
234	Never a Forwardee
235	Originate Only
236	Outgoing Trunk Callback
237	Outgoing Trunk Camp-On
238	Override Security
239	Priority Dial 0
240	Line Privacy
241	Receive Only
242	Repeated Camp-On Beep
243	Non-Busy Extension
244	Room Status Applies
245	Abbreviated Dialing Access
246	SMDR - Extended Record
247	SMDR - Record Meter Pulses
248	TAFAS Any Access
249	TAFAS Access Tenant
250	TAFAS Access During Day Service
251	Transfer Dial Tone
252	Broker's Call with Transfer
253	Call Forward - Don't Answer Timer (2 - 6 Rings)
254	Call Hold Recall Timer (PBX Telephones) 0 - 10 Minutes
255	Repeated Camp-On Beeps Timer (5 - 15 Seconds)
256	UCD Music-on-Hold Timer (0 - 50 Minutes)
257	Flash Over Trunk
258	Display Prime as Forwarder
259	Message Sending

Table: Class Of Service Options

COS Option Numbers	Class Of Service Option Name
260	Internal / External Split Call Forwarding
261	ONS Voice mail Port
262	Ignore Forward Busy with Free Appearance
263	Delay Ring Timer (2 - 6 Rings)
264	Half Fwd NA timer for DID call when VM msg on
265	Voice mail System Speed Dial Index (0-255)
266	Camp-on Before Forward on Busy
267	Softkey Support for Voice mail
268	Record a Call in Voice mail
269	Record a Call: Start Recording Automatically
270	Record a Call: Save Recording on Hangup
271	Privacy Released at Start of Call
272	Guest Suite Extension
273	Display Held Caller ID to ONS/CLASS
274	ONS Ring Group Member
275	Enable Voicemail Single Key Access
276	Twin Phone
277	Automatic Mailbox Creation
278	Intercom Mode
279	Display Account Codes on Phone
280	PC (2nd) Port on IP Phone
281	TAFAS Access During Day/Night Service
282	Standard Hard hold
300	Automatic Callback
301	Camp-On
302	Flash-in Conference
303	Paging Zone 1 Access
304	Paging Zone 2 Access

Table: Class Of Service Options

COS Option Numbers	Class Of Service Option Name
305	Paging Zone 3 Access
306	Paging Zone 4 Access
307	Paging Zone 5 Access
308	Paging Zone 6 Access
309	Paging Zone 7 Access
310	Paging Zone 8 Access
311	Paging Zone 9 Access
312	Paging Default (0 - 9) (0 Gives All Enabled Zones)
313	CO Trunk to CO Trunk Connect
314	CO Trunk to TIE Trunk Connect
315	CO Trunk to DID Trunk Connect
316	TIE Trunk to TIE Trunk Connect
317	TIE Trunk to DID Trunk Connect
318	DID Trunk to DID Trunk Connect
319	Extension Non-CO Trunk to Trunk Connect
320	Transparent Multi-Console Operation
321	Ignore Call Forward After Transfer
322	Confirm Wakeup by Offhook
326	Account Code, Forced Entry - Data Internal Calls
327	Account Code, Forced Entry - Data External Calls
328	Account Code, Forced Entry - Data Long Distance Calls
400	Contact Monitor
401	Call Park
402	Long Loop (Off-Premise Extensions Only)
403	Trunk Recall Partial Inhibit
404	Recording Failure to Hangup Timer (1 - 255 Seconds)
405	Can Initiate Call Monitor
406	Allow To Be Monitored (0=dis, 1=no beep, 2=beeps)

Table: Class Of Service Options

COS Option Numbers	Class Of Service Option Name
500	Override
501	Override Announce
502	Display ANI/DNIS/CLASS Information
503	Display CLASS Name
504	SS420 Optional CLASS/ANI Display
505	ONS Stations Support CLASS
506	ONS Positive Disconnect < 0-5 secs > < 0=disabled >
507	Station/Set: Allow My Number to be Displayed
508	Station/Set: Show Internal Number on My Phone
509	Display Caller ID for Non-Prime Lines
510	ONS Stations Support CLASS Visual Call Waiting
511	COV Voice Mail Displays Calling
600	Telephone - Auto-Answer
601	Telephone - Auto-Hold Disable
602	Telephone - Background Music
603	Telephone. - Disconnect Alarm
604	Telephone - Automatic Outgoing Line
605	Telephone - Message Program
606	Telephone - Enhanced Answering Position
607	Telephone - Associated Modem Line
608	Telephone - Room Status Display
609	Telephone - Night Service Switching (DAY/NIGHT1/NIGHT2)
610	Telephone - Guest Room Template (0 - 3) (DN)
611	Telephone - Limited New Call Ring
612	Telephone - Headset Operation
613	Display ANI Information Only
614	Telephone - Handset Volume Saved
615	Telephone - Offhook Voice Announce

Table: Class Of Service Options

COS Option Numbers	Class Of Service Option Name
616	Alarm Monitor Point
617	Immediate Off Hook Alarm
618	Alarm Audio Level for Sets
619	Direct Speak@Ease Access
620	Telephone - Max Call logs Allowed (0-20)
621	Call Logging - Multiple CO/DTS
622	Ring Group Call Logging
623	Automatic DID Number Assignment
624	Wakeup Alarm Notification
650	ACD - Agent Template (0 - 3; 0 = Disable)
651	ACD - Supervisor Template (0 - 3; 0 = Disable)
652	ACD - Senior Supervisor Template (0 - 3; 0 = Disable)
653	ACD - Agent Always Auto-Answer
654	ACD - Display Path Always
655	Allow Continuous Monitor
680	Key System - Direct CO Access
681	Key Set/Sub Att. - Call Hold Notify Timer (0 -600 s)
682	Key System - Auto Answer - Internal Calls
683	Key System - Direct Paging Handsfree Answerback
684	Key System - Can make All Set Page
685	Key System - Can receive All Set Page
686	Group Page Include Overhead Paging
687	All Set Page Include Overhead Paging
688	IP Set Requires Compression
689	DTS/CO Line Transfer Call Handling
690	Hold And Page
691	Telephone - Day/Night1 Switching
692	Secure Hot Swapping

Table: Class Of Service Options

COS Option Numbers	Class Of Service Option Name
700	SMDR - Does Not Apply
701	No Dial Tone
702	SMDR - Overwrite Buffer
703	Message Register Applies
704	Incoming / Internal Modem Pooling Access
705	Automatic Overflow From Attendant
709	Follow External Call Forward
800	ANI Applies
801	Incoming Trunk Call Rotary
802	Limited Wait for Dial Tone
803	SMDR - Drop Calls < n Digits (0 ... 11, 0 = disable)
804	SMDR - Drop Incomplete Outgoing Calls
805	Trunk No Dial Tone Alarm
806	SMDR - Record Incoming Calls
807	SMDR - Display Private Speedcall
808	Special DISA
809	Standard Ring Applies
810	DISA During Night Service Only
811	ANI/DNIS Trunk
812	Loop Start Trunk to ACD Path Connect
813	Delay ONS Ring - Wait for Network Name (0-6 s)
814	SMDR - Record ANI/DNIS/CLASS
815	DTS/CO Line Key Honors Forwarding
816	CENTREX Flash Over Trunk
900	Data Station Queuing
901	DTRX Herald
902	DTRX Message Code
903	DTRX Message Code Text

Table: Class Of Service Options

COS Option Numbers	Class Of Service Option Name
904	DTRX Complete Message Text
905	DTRX Herald Text Select (1- 4)
906	DATA SMDR - Does Not Apply
907	DATA SMDR - Extended Record
908	DATA SMDR - Overwrite Buffer

Parameters arranged alphabetically

The following are not feature descriptions; refer to the Program Features section for complete descriptions.

Abbreviated Dialing Access (Option 245): Allows access to the system Abbreviated Dialing tables. You cannot access abbreviated dialing for the purpose of making external or long distance calls if you have COS option 200: Account Code, Forced Entry - External Calls, or 201: Account Code, Forced Entry - Long Distance Calls.

Account Code, Forced Entry - Data External Calls (Option 327): Requires an extension to enter an account code before making external data calls.

Account Code, Forced Entry - Data Internal Calls (Option 326): Requires an extension to enter an account code before making internal data calls.

Account Code, Forced Entry - Data Long Dist Calls (Option 328): Requires an extension to enter an account code before making long distance data calls.

Account Code Forced Entry - External Calls (Option 200): Forces an extension to enter an account code before allowing external calls. If you have this option you cannot access Abbreviated Dialing (Option 245) for the purpose of making external calls.

Account Code Forced Entry - Long Distance Calls (Option 201): Forces an extension to enter an account code before allowing long distance calls. If you have this option you cannot access Abbreviated Dialing (Option 245) for the purpose of making long distance calls.

ACD - Agent Always Auto-Answer (Option 653): Allows an ACD agent to be in auto answer mode at all times. See the ACD TELEMARKETER Application Package section.

ACD - Agent Template (Option 650): Allows certain configurations of Speed Dial keys and Feature Keys. See the ACD TELEMARKETER Application Package section.

ACD - Display Path Always (Option 654): When enabled, this option displays the ACD path name, programmed in Form 41, on the Mitel telephone display. If disabled, the ACD access code is presented.

ACD - Senior Supervisor Template (Option 652): Allows certain configurations of Speed Dial keys and Feature Keys. See the ACD TELEMARKETER Application Package section.

ACD - Supervisor Template (Option 651): Allows certain configurations of Speed Dial keys and Feature Keys. See the ACD TELEMARKETER Application Package section.

Alarm Audio Level for Sets (Option 618): Allows the display set to detect and report alarms for 911 calls and lockouts (off-hook states) for sets. Defines the volume level on the display set for 911 calls and offhooks on sets. Three choices exist: HIGH, MEDIUM, and RINGER LEVEL.The MEDIUM level is approximately 70% of the volume of the HIGH level. The RINGER LEVEL is the same volume that is set for normal ringing on the display set.

Alarm Call (Option 202): Allows an extension to be programmed to ring at a specific time.

All Set Page Includes Overhead Paging (Option 687): Allows the user of a Mitel telephone to initiate a page to all the telephones and include the overhead paging as well. This option functions with Feature Level 3 or greater in System Option 102.

Allow Continuous Monitor (Option 655): Allows a set to monitor another set continuously, regardless of call state. This COS option is enabled in the class of service of the monitoring set.

Allow To Be Monitored (Option 406): Allows a device to be monitored using the Call Monitor feature.

ANI Applies (Option 800): Provides for ANI information being transmitted on outgoing trunks.

Note: This option is obsolete and must be disabled in EVERY class of service in the system.

ANI/DNIS Trunk (Option 811): Allows an incoming trunk to receive ANI/DNIS information. ANI/DNIS trunks must use DTMF signaling, have a wink timer and one of the following trunk circuit descriptors: “T1 E&M”, “T1 E&M DISA”, “T1 DID/TIE” or “T1 TIE DISA”.

Apply Key Line Conference Warning Tone (Option 126): Allows or prevents a warning beep to occur when a new party joins a multi-party conference call via a key line appearance (with privacy released). The default setting is ENABLED. This option is programmed for the owner of the line, that means, the set that has the key line number as the prime line or the first set that was programmed with a key line appearance with this number. The functionality provided with this option is dependant on Feature Level 3 in System Option 102.

Attendant Abbr. Dial Confidential Number Display (Option 110): Allows the attendant to view abbreviated dial numbers which have been programmed as “confidential”.

Attendant Abbreviated Dial Programming (Option 111): Allows the attendant to program numbers into the system’s abbreviated dial table.

Attendant Audible Alarm (Option 108): provides audible alarms to ring at the console when a suitably programmed station goes into the lock-out state or when a station dials an emergency 911 call.

Attendant Automatic Call Forward - No Answer (Option107): Allows unanswered calls to be rerouted to a secondary answer point. The call is automatically rerouted after it rings for a pre-determined length of time. See COS Option 118 - Attendant Automatic Call Forward - No Answer Timer.

Attendant Bell Off (Option 100): Allows the attendant to mute the console ringer.

Attendant Call Block Key (Option 113): Allows the attendant to block station to station (room to room) calls. This feature is part of the Hotel/Motel feature package.

Attendant Call Forward - No Answer Timer (Option 118): This sets the Call Forward - No Answer timer for the Attendant Automatic Call Forward Feature.

Attendant Call Forward Setup/Cancel (Option 123): Allows attendant or subattendant to change the call forward status of a station in its tenant group.

Attendant Conference Disable (Option 120): Disallows the console from making conference calls.

Attendant DISA Code Setup (Option 103): Allows the attendant to change the DISA access code, without accessing CDE.

Attendant Display of System Alarms (Option 102): Allows the attendant to receive and view alarm logs at the console without logging in to maintenance. The alarm icon is not presented for minor alarm conditions.

Attendant Flexible Night Service Setup (Option 104): Allows the attendant to change the night service assignment for non-dial-in trunks.

Attendant Guest Room Key (Option 105): Allows the attendant to access Hotel/Motel features via the *GUEST ROOM* softkey. This feature is part of the Hotel/Motel feature package.

Attendant Multi - New Call Tone (Option 125): When at least one call is waiting to be answered by an attendant or subattendant who is busy on a call, a short beep sounds at the programmed interval. (Consoles must also have COS Option 126 - Attendant New Call Tone enabled).

If COS Option 125 - Attendant Multi-New Call Tone is enabled on a SUPERSET subattendant. In the case of a subattendant, COS Option 125 - Attendant Multi-New Call Tone works independently of any other COS options.

Attendant New Call Tone (Option 106): Allows the attendant to be notified of incoming calls by a single tone burst, while engaged in a call. This COS Option does not apply to subattendants.

Attendant O/G Restriction/Room Status Setup (Option 101): Allows the attendant to restrict outgoing calls. This feature is part of the Hotel/Motel feature package.

Attendant Serial Call (Option 109): Allows the attendant to force an incoming trunk call to recall to the console after the trunk is released from the call.

Attendant Station Busy Out (Option 112): Allows the attendant to busy out (and return to service) any station in the system.

Attendant Station Do Not Disturb (Option 121): Allows attendant or subattendant to change DND status on a station in its tenant group.

Attendant Timed Recall (Hold) (Option 116): This sets the busy recall timer. If set to 0, the feature is disabled.

Attendant Timed Recall (Campon) (Option 117): This sets the campon recall timer. If set to 0, the feature is disabled.

Attendant Timed Recall (No Ans) (Option 115): This sets the no answer recall timer. If set to 0, the feature is disabled.

Attendant Tone Signaling (Option 119): Allows the console to transmit DTMF tones during an established call.

Attendant Trunk Busy Out (Option 114): Allows the attendant to busy out (and return to service) any trunk in the system.

Automatic Callback (Option 300): Allows an extension to arrange for a call to a busy extension to be completed when that extension becomes idle.

Automatic DID Number Assignment (Option 623): Applies only to devices that can be checked in or out. Allows automatic DID number assignment upon check-in.

Automatic Mailbox Creation (Option 277): Automatically attempts to create a voice mailbox for any new STATION or SET added in Form 9. If System Option 107 (Lodging), 108 (Property Management System), or 124 (Voice Mail Property Management System), are enabled in Form 04, the mailbox type is set to GUEST; otherwise, it is set to EXTEN. The new mailbox is assigned the name and number entered for the STATION or SET in Form 9 and all the other default values assigned to new mailboxes in Form 50.

Mailboxes are NOT created if

- System Option 125 (Form 4) is is 0
- the length of extension numbers is not the same as the length of mailbox numbers in Form 49
- a mailbox already exists for an extension

- an extension number starts with 9 or the language change number in Form 49

No error message is given if the attempt to create a mailbox fails.

Note: Mailbox deletion is NOT automatic when sets or stations are deleted. Mailboxes must be manually deleted from Form 50.

Automatic Overflow from Attendant (Option 705): Allows incoming calls not answered within a pre-determined period to be rerouted to a recording device while remaining in the answer queue.

Broker's Call (Option 203): Allows an extension flash to be interpreted as a swap rather than a conference attempt.

Broker's Call With Transfer (Option 252): Same as Broker's Call, except that a transfer is possible when the extension goes on-hook.

Call Block Applies (Room to Room) (Option 204): Allows the extension to be affected when the attendant applies the Call Block feature. This feature is part of the Hotel/Motel feature package.

Call Forward - Busy (Option 206): Allows the extension to set up call forwarding on busy.

Call Forward - No Answer (Option 207): Allows the extension to set up call forwarding on no answer.

Call Forward Don't Answer Timer (Option 253): Sets the number of times the set will ring before the call is forwarded.

Call Forward - External (Option 208): Allows the extension to set up call forwarding to external numbers.

Call Forward Inhibit On Dial In Trunks (Option 210) : Prevents calls from Dial-In trunks from being rerouted always/busy/no answer by Form 19, Call Rerouting Table, when they call devices with this option in their COS.

Call Forwarding Inhibit on Hold Timeout (Option 222): Causes call forward no answer to be ignored when a held party times out and begins ringing back the holding set.

Call Hold and Retrieve Access (Option 211): Allows the extension to use the Call Hold and Call Hold Retrieve access codes.

Call Hold Recall Timer (Option 254): For regular stations and Mitel telephones, this COS Option sets the length of time between when a call was placed on hold, and when the held call recalls. This COS option sets the length of time between when a call was placed on hold, and when the system provides the first Hold Recall tone. The timer can be set from 0 to 10 minutes. You select 0 minutes if the customer wishes to have a reminder instead.

Call Logging - Multiple CO/DTS (Option 621): Allows call logging for multiple appearances of DTS and CO line keys. To function, this option requires that COS Option 620, Telephone - Max Call logs Allowed (0-20), must also be greater than 0.

Call Park (Option 401): Allows an extension to park a call and go on-hook. New calls may not be received or originated, but paging equipment may be accessed.

Camp-On (Option 301): Allows an extension to notify a busy extension (via a beep) that they desire communication. The camped-on call rings when the called party goes on-hook.

Camp-On Before Forward on Busy (Option 266): Allows an extension to camp on to a busy extension without being immediately forwarded. The caller may have the call forwarded by pressing a FwdMe or Forward Me softkey. This COS option will be ignored if the user calls a busy extension that is in a recording session.

Can Flash If Talking To An Incoming Trunk (Option 212): Allows an extension to flash the switchhook when connected to an incoming trunk.

Can Flash If Talking To An Outgoing Trunk (Option 213): Allows an extension to flash the switchhook when connected to an outgoing trunk.

Can Initiate Call Monitor (Option 405): Allows the activation of the Call Monitor feature. For calls initiated from an individual telephone set, this Class Of Service option applies to the caller's COS while it will apply to the called party's COS for calls initiated from a trunk.

Cannot Dial a Trunk After Flashing (Option 214): Prevents an extension from accessing a trunk after flashing the switchhook.

Cannot Dial a Trunk if Holding or in Conf With One (Option 215): Prevents an extension from accessing a trunk if that extension already has a trunk on hold, or is in conference with a trunk.

Centrex Flash Over Trunk (Option 816): After the user performs a Flash Over Trunk, this feature allows the user to go immediately into talk state, and access Central Office CENTREX features (such as Conference, Call Waiting, Swap/Hold, and Drop Last). When disabled, this COS allows the user to perform a Flash Over Trunk and go into dialing state. CENTREX Flash Over Trunk is disabled by default.

Clear All Features (Option 221): Allows an extension to turn off all features currently active. These features include all flavors of Call Forwarding, Do Not Disturb and Callbacks.

Confirm Wakeup by Offhook (Option 322): Allows a Mitel display telephone user to acknowledge a wakeup by going offhook, instead of having to read the message and press a softkey (primarily for hotel/motel).

Contact Monitor (Option 400): Allows a line circuit port to be used as an alarm contact relay. See the Program Features section for further information.

CO Trunk to CO Trunk Connect (Option 313): Allows an extension to connect 2 CO trunks together.

CO Trunk to DID Trunk Connect (Option 315): Allows an extension to connect a CO trunk and a DID trunk together.

CO Trunk to Tie Trunk Connect (Option 314): Allows an extension to connect a CO trunk and a Tie trunk together.

COV Voice Mail Displays Calling (Option 511): Enable this option if the COV voice mail system requires the text "CALLING" to display on voice mail ports that are ringing.

Data Security (Option 216): Prevents the system from transmitting any intrusion or warning tones on an established call.

DATA SMDR - Does Not Apply (Option 906): Prevents data calls from having SMDR records.

DATA SMDR - Extended Record (Option 907): Allows for the system identifier to be included in the SMDR record.

DATA SMDR - Overwrite Buffer (Option 908): Allows older records to be overwritten by newer records when the SMDR buffer becomes full. If disabled, a full buffer will result in further calls requiring SMDR to not be allowed.

Data Station Queuing (Option 900): Allows queuing on to busy data stations.

Delay ONS Ring - Wait for Network Name (0-6 s) (Option 813) - Provides a delay in the ONS Ring for PRI trunks in order to co-ordinate the delivery time of the name and number on ONS CLASS sets. The default setting is 0.

Delay Ring Timer (Option 263): Specifies the duration of the ring delay on Mitel telephone line select keys which are line appearances with delay ring. Default value is 3 rings.

DID Trunk to DID Trunk Connect (Option 318): Allows an extension to connect 2 DID trunks together.

Direct Speak@Ease Access (Option 619): Provides direct access to the Mitel Speech Server.

Direct To ARS (Option 217): Allows an extension to be routed directly to ARS upon going off-hook. The system automatically dials the ARS leading digit string.

Directed Call Pickup (Option 218): Allows the extension to dial the Directed Call Pickup access code, and answer a call ringing at another extension.

DISA During Night Service Only (Option 810): Allows for a trunk to be a DISA trunk only when the system (or tenant group) is in night service.

Discriminating Dial Tone (Option 219): Allows an extension to have a special distinctive dial tone, informing the user that feature(s) are active. A station (an ONS or OPS telephone) with Discriminating Dial Tone uses the discriminating dial tone to indicate the presence of a message waiting at the station.

Display Account Codes on Phone (Option 279): Determines whether or not account code digits are displayed on telephones. When the option is disabled, an asterisk will appear for every account code digit entered.

Display ANI/DNIS/CLASS Information (Option 502): Allows the display of ANI and DNIS digits on Mitel telephones and consoles.

Display ANI Information Only (Option 613): Allows ANI digits only (never DNIS digits) to be displayed on Mitel telephones. Not applicable if CLASS name is available.

Display Caller ID for Non-Prime Lines (Option 509): Allows the display on Mitel telephones (SUPERSET 4DN telephones excluded) to show the caller ID for multi-call lines, key lines, direct trunk select lines, and CO lines. This option functions with Feature Level 3 or greater in System Option 102.

Display CLASS Name (Option 503): Enabled by default, this COS option gives CLASS name priority over CLASS numbers (Calling Line ID digits) whenever information about the trunk is to be displayed on Mitel telephones and consoles. If disabled, CLASS number will be displayed ahead of CLASS name.

Display Held Caller ID to ONS/CLASS (Option 273): Allows an ONS CLASS set to display the caller ID of the outside original caller if the transferring party has COS Option 273 enabled (the default setting is disabled). If COS Option 273 is disabled for the transferring party or if the held party is an internal telephone, the ONS CLASS set will show the caller ID of the transferee i.e. attendant, automated attendant, voice mail etc.

Display Prime as Forwarder (Option 258): Allows the option of displaying or toning out the forwarder's extension or logical line access code when a call is forwarded. If enabled, and a logical line of single appearance is forwarded to the ONS Voice mail, then the extension of the prime of the set will be toned out. If disabled, the extension of the line is toned out. (Calls forwarded from a single appearance DTS key that has COS option 815 enabled will always display the prime as the forwarder.)

Do Not Disturb (Option 220): Allows the extension to prevent incoming calls from ringing.

DTRX Complete Message Text (Option 904): Allows for DTRX message codes and status messages to be displayed.

DTRX Herald (Option 901): Allows a programmable message to appear when connection is made to the DTRX.

DTRX Herald Text Select (Option 905): Allows for a choice of DTRX herald messages.

DTRX Message Code (Option 902): Allows for DTRX message codes to be displayed.

DTRX Message Code Text (Option 903): Allows for DTRX status messages to be displayed.

DTS/CO Line Key Honors Forwarding (Option 815): Allows incoming Direct Trunk Select/CO Line Key calls to follow call forwarding programmed on the set if the DTS line/CO line key appears on only one set (disabled by default). If the option is disabled, the call is never forwarded.

DTS/CO Line Transfer Call Handling (689): Allows incoming calls on a DTS/CO line to stay on the DTS/CO line key when transferred unsupervised between extensions that have an appearance of the line. Without Option 689, the call will use both the DTS/CO line key and the prime line, key line, or multicall line - whichever was used to answer the call at the extension receiving the transfer. For supervised transfers, the call remains on the prime line; it moves to the DTS/CO line key only when it is placed on hold using the red Hold key. Note: This feature requires Feature Level 5 (Form 4, System Option 102).

Enable Voicemail Single Key Access (Option 275): Provides a voice mail key that transfers a caller to another user's voice mail. The voice mail key is programmed as a feature key, and in some cases can appear as a softkey.

Extension Non-CO Trunk to Trunk Connect (Option 319): Allows an extension to connect any non-CO type trunk to any other trunk.

Flash Disable (Option 223): Prevents an extension from using any feature that required the use of the switchhook flash.

Flash For Waiting Call (Option 205): Allows a user to place a call (2-party or multi-party) on consultation hold and connect to a waiting call, via a flash of the switchhook.

Flash For Attendant (Option 224): Allows an extension to ring the attendant immediately upon flashing the switchhook.

Flash-In Conference (Option 302): Allows an extension to create 4 or 5 party conferences.

Flash Over Trunk (Option 257): Allows an extension to send a switchhook flash out on a trunk.

Follow External Call Forward (Option 709): Specifies the calling device to be an external call for call forwarding (Disabled by default).

Guest Suite Extension (Option 272): Allows secondary telephones to associate with a primary telephone in a hotel/motel suite. The Suite Services feature requires this option enabled on the secondary telephones.

Group Page Include Overhead Paging (Option 686): Allows the user of a Mitel telephone to initiate a page to a group of telephones and include the overhead paging as well. This option functions with Feature Level 3 or greater in System Option 102.

Half Fwd NA timer for DID call when VM msg on (Option 264): When an incoming DID call is made to a set or station with this option, and when that set or station has Voice mail messages waiting, then the Call Forward No Answer timer is shortened by half. This is a toll saving option, allowing users who are calling in to check their messages to hang up if the cfna timer is not shortened, knowing they have no messages to retrieve.

Headset Hard Hold (Option 282): See Standard Hard Hold.

Hold And Page (Option 690): Allows an extension to page after placing a call on hold.

Hold Pickup (Option 225): Allows an extension to connect to a caller waiting in an attendant console hold slot position.

Ignore Call Forward After Transfer (Option 321): A trunk call that is transferred will ignore forwarding programmed for the set or station the call is transferred to. The call will recall to the transferring set, subattendant, or console when the recall timer expires. This COS option is enabled in the Class of Service of the transferring set, subattendant, or console. Only trunk calls are affected by this COS.

Ignore Forward Busy with Free Appearance (Option 262): When calling a station or Mitel telephone with this COS enabled, call forward busy is ignored if there are any free multi-line appearances. Sets with an appearance of this line that are in Do Not Disturb will appear busy. The lines on sets that are not in Do Not Disturb will appear busy.

Note that this COS option applies only to multi-line appearances, not key line appearances or logical line appearances.

Immediate Off Hook Alarm (Option 617): Causes a lockout alarm to a display set as soon as a set goes offhook.

Incoming/Internal Modem Pooling Access (Option 704): Allows a modem user to make a call to a Dataset.

Incoming Trunk Call Rotary (Option 801): Instructs a trunk to ignore incoming DTMF digits.

Intercom Mode (Option 278): Controls whether the set is rung or paged when called from another set or console.

Internal/External Split Call Forwarding (Option 260): Allows an extension to split call forward setup between internal calls and external calls.

Inward Restriction (Option 226): Prevents an extension from receiving incoming calls from DID trunks.

IP Set Requires Compression (Option 688): Specifies that the IP telephone is to use G.729 or G.711 compression when voice streaming.

Key Set/Sub Att - Call Hold Notify Timer (Option 681): Sets the time between when a subattendant places a call on hold on a programmable hold key and when the held call provides a burst of tone as a reminder. The timer can be disabled or set between 0 and 600 seconds. If the customer does not want a recall or a reminder, set this COS Option to 0 time. If the customer wants a reminder tone (not a recall), set this COS Option to the desired (non-zero) time. The time set in COS option 254 will be ignored. If the customer wants a recall, disable this COS Option and set the appropriate time in COS Option 254.

Key System - Auto-Answer - Internal Calls (Option 682): Allows a SUPERSET 3DN or SUPERSET 4DN telephone to respond handsfree to a directed page.

Key System - Can make All Set Page (Option 684): Allows the set to page all sets in the system that do not have Key System - Can receive All Set Page disabled. For the console to make an All Set page its COS must have this option enabled. This feature is not supported on SUPERSET 3DN or SUPERSET 4DN telephones.

Key System - Can receive All Set Page (Option 685): Allows the set to receive an All Set Page.

Key System - Direct Paging Handsfree Answerback (Option 683): Allows a a Mitel 5020 IP, 5220 IP, 5224 IP, TeleMatrix 3000IP, SUPERSET 4025, SUPERSET 4125, SUPERSET 4150, SUPERSET 410, SUPERSET 420 or a SUPERSET 430 key system set user to access the handsfree answerback feature for directed page calls.

Key System - Direct CO Access (Option 680): Allows a set to directly access a CO trunk using Feature Access Code 50.

Limited Wait For Dial Tone (Option 802): Instructs a trunk to wait a pre-determined period for dial tone from the far end before sending digits. This option must be enabled for ISDN trunks.

Line Privacy (Option 240): Prevents one Mitel telephone from overriding the key line being used by another Mitel telephone.

Lockout Alarm Applies (Option 227): Causes an alarm to be raised at the attendant console when the set goes into the *lockout* state.

Long Loop (Option 402): Designates the OPS port as having a loop length of over 2 km. This adds a compromise balance network into the circuit.

Loop Start Trunk to ACD Path Connect (Option 812): Allows loop start DISA trunks to access ACD.

Manual Line (Option 228): Causes an off-hook origination to ring the attendant. Calls are received in the normal manner.

Message Register Applies (Option 703): Causes the system to count and report (SMDR) the number of outgoing calls made by a station and the number of incoming meter pulses received. This feature is part of the Hotel/Motel Feature Package.

Message Register Overflow Alarm (Option 230) : Allows an alarm to be raised at the attendant console, and a maintenance log to be generated when the extension's message register overflows.

Message Sending (Option 259): Allows telephones to send Call Me Back messages to other extensions using the Send Message Feature access code of the **Message** key on Mitel telephones.

Message Waiting Setup - Bell (Option 231): Allows the attendant to set up a message waiting condition on a extension not equipped with a lamp. Note: this option is not supported on ASUs.

Message Waiting Setup - Lamp (Option 232): Allows the attendant to set up a message waiting condition on an extension equipped with a lamp.

Never a Consultee (Option 233): Prevents other extensions calling when they have a consultation hold in progress.

Never a Forwardee (Option 234): Prevents an extension from having calls forwarded to it by another extension.

No Dial Tone (Option 701): Inhibits dial tone. This option must be enabled for ISDN trunks.

Non-Busy Extension (Option 243): Causes calls to the busy extension to override automatically, and join the conversation.

ONS Positive Disconnect < 0-5 secs > < 0=disabled > (Option 506): Forces a disconnect on an unattended ONS device (a device that is not attended by a user to perform a manual hangup). The forced disconnect is an electrical break. The break period is programmable from 1 to 5 seconds. A value of 0 represents no electrical break.

ONS Ring Group Member (Option 274): Allows multiple secondary telephones in an ONS Ring Group to all ring when a primary telephone for that ONS Ring Group is called. This option is mutually exclusive with COS Option 607, Telephone Associated Modem Line, and COS Option 272 Guest Suite Extension, that means, only one of these options can be enabled for a telephone at any one time. The ONS Ring Group Member option functions with Feature Level 3 or greater in System Option 102.

ONS Stations Support CLASS (Option 505): Enables ONS sets to support CLASS features. The name and number of the calling party appears on the set in the ringing state and in the talk state (campon). This option also activates/deactivates the message waiting lamp and clears the call logs when a guest checks out.

ONS Stations Support CLASS Visual Call Waiting (Option 510): Enables ONS sets to support the visual call waiting CLASS feature. The name and number of the calling party appears on the set in the talk state (campon). This feature is only available on ONS display sets that support visual call waiting.

ONS Voice mail Port (Option 261) : Causes an ONS line port to be designated as an interface to an ONS Voice mail system. Refer to the Program Features section.

Originate Only (Option 235): Prevents an extension from receiving any calls, unless they are forwarded.

Outgoing Trunk Callback (Option 236): Allows Callback - Busy on outgoing trunks.

Outgoing Trunk Camp-On (Option 237): Allows Camp-On on outgoing trunks.

Override (Option 500): Allows an extension to override (intrude into the conversation of) a busy extension.

Override Security (Option 238): Prevents other extensions from overriding calls.

Paging Default (Option 312): Sets the default paging zone for the station. 0 enables all zones.

Paging Zone Access (Options 303 - 311): Allows an extension to have access to one or more paging zones. Zones 1 through 9 are programmable.

PC (2nd) Port on IP Phone (Option 280): When enabled along with System Option 131 (Form 04) a PC connected to the second IP port on the IP phone will have network access.

Priority Dial 0 (Option 239): Provides a second class of dial-0 access to the attendant. This class may then have its own LDN appearance on the attendant console(s).

Privacy Released at Start of Call (Option 271): Privacy is always released at the start of a call. The user will have to press the Make Private softkey or the Privacy Release programmable key to toggle the state back to where Line Privacy is preserved.

Receive Only (Option 241): Prevents an extension from initiating calls.

Record a Call in Voice mail (Option 268): Enables the Record a Call feature.The default has this option disabled. To enable Record a Call in Voice mail the system option (Form 04), Record a Call, Option 87 must be enabled.

Record a Call: Start Recording Automatically (Option 269) Switches the recording of the messages between the manual startup mode and the automatic startup mode.The default has this option disabled. Enables the automatic recording of external calls, incoming and outgoing.

Record a Call: Save Recording on Hangup (Option 270) Determines whether the recording will be saved or discarded when the user hangs up and does not save or erase the active recording.The default has this option disabled.

Recording Failure to Hangup Timer (Option 404): Sets the time a recording device has to hang up. If the timer expires before the device hangs up, it is placed into the *Do Not Disturb* state.

Repeated Camp-On Beeps (Option 242): Provides for repeated notification beeps, indicating that a trunk is camped on.

Repeated Campon Beeps Timer (Option 255): Sets the period for the repeated campon beeps.

Ring Group Call Logging (Option 622): Logs the following calls:

- "missed" calls for each member of a ring group if the call is not answered by any member
- missed calls for each member who does not answer a call for "ring one member" ring groups
- answered calls for the answering member (only) for "ring all members" ring groups.

Note: COS Option 620 (Telephone - Max Call Logs Allowed) must be enabled before Ring Group call logging will work.

Room Status Applies (Option 244): Allows an extension to have its room status changed - this feature is part of the Hotel/Motel feature package.

Secure Hot Swapping (Option 692): Allows a user to swap programming from a physical or phantom device onto a telephone. Both devices must be programmed to support the feature. See Secure Hot Swap for more information.

Setup Time/Date (Option 122): Allows attendant or subattendant to set the system time and/or date. Note: This feature requires Feature Level 6 (Form 4, System Option 102).

SMDR - Display Private Speedcall (Option 807): Controls the display of private speedcall numbers in SMDR.

SMDR - Does Not Apply (Option 700): Prevents calls from having SMDR records.

SMDR - Drop Calls < n Digits (Option 803): Prevents SMDR from reporting on calls which are less than a pre-determined length.

SMDR - Drop Incomplete Outgoing Calls (Option 804): Prevents incomplete outgoing calls from generating an SMDR report.

SMDR - Extended Record (Option 246): Allows for 8 additional columns in the record to accommodate 12-digit account codes and the system identifier. This increases the number of characters for an SMDR record to 88.

SMDR - Overwrite Buffer (Option 702): Allows older records to be overwritten by newer records when the SMDR buffer becomes full. If disabled, a full buffer will result in further calls requiring SMDR to not be allowed.

SMDR - Record ANI/DNIS/CLASS (Option 814): Allows ANI/DNIS information to be reported in the SMDR trunk record.

SMDR - Record Incoming Calls (Option 806): Causes incoming trunk calls to be recorded by SMDR.

SMDR - Record Meter Pulses (Option 247): Allows meter pulses received from the Central Office to be counted, and recorded.

Softkey Support for Voice mail (Option 267): Enables voice mail softkeys on a Mitel 5020 IP, 5220 IP, 5224 IP, SUPERSET 4025, SUPERSET 4125, SUPERSET 4150, SUPERSET 420, and SUPERSET 430 telephones. Note that voice mail softkeys are supported by the SX-200 ICP embedded voice mail application and the DNIC-based NuPoint Messenger system. See Softkey Support.

Special DISA (Option 808): Allows for DISA users to enter verified account codes rather than the DISA access code. (Feature Access Code 19).

SS420 Optional CLASS/ANI Display (Option 504): Allows a SUPERSET 420 telephone to receive ANI/DNIS information about a call that is forwarded to it (rather than receiving information about the telephone that forwarded the call).

Standard Hard Hold (Option 282): Allows a headset user to place a call on hard hold when the Automatic Line Selection feature is enabled (COS Option 604, Telephone - Automatic Outgoing Line). For this feature to function correctly, COS Option 690 (Hold and Page) must be disabled, and Feature Level 6 must be selected in System option 102.

Standard Ring Applies (Option 809): When enabled for trunks, this option allows for incoming trunks to provide standard ringing cadence; ignoring the Discriminating Ringing feature. When enabled for station/set, a standard ringing cadence would be provided to the station/set, regardless of any other system option or COS options.

Station/Set: Allow My Number to be Displayed (Option 507): Allows or blocks the station/set number from display on Mitel IP telephones and SUPERSET 4000 and 400 series telephones, or ONS CLASS stations. For example in a Hotel, this option can be DISABLED for guest to guest calls, giving room number confidentiality to the guests. The default for this option is ENABLED. This option works independently; for the caller and the called party. The blocking of the number is dependant on System Option 102, Feature level 2 or greater.

Station/Set: Show Internal Numbers on My Phone (Option 508): Overrides COS Option 507, Station/Set: Allow My Number to be Displayed. Allows a station or a set to always see the station/set number of the internal party. For example in a Hotel, room service would use this option.

Subattendant Station Setup Advisory Message (Option 150): Allows a subattendant to set up advisory messages on sets within the tenant group.

TAFAS Access During Day Service (Option 250): Allows the extension to answer incoming calls ringing the console during the day, regardless of the tenanting.

TAFAS Access During Day/Night Service (Option 281): Allows the extension to answer incoming calls appearing at common alerting devices (night bells) and at the console during the day or night, regardless of the tenanting.

TAFAS Any Access (Option 248): Allows an extension to answer incoming calls ringing at common alerting devices (night bells) in any tenant group.

TAFAS Tenant Access (Option 249): Allows an extension to answer incoming calls ringing at common alerting devices (night bells) within the extension's tenant group.

Telephone - Immediate Line Select: See Telephone - Automatic Outgoing Line (Option 604) below for more information.

Telephone - Associated Modem Line (Option 607): Allows a SUPERSET 4DN telephone to be associated with an ONS port connected to modem. The telephone can initiate either a data call or voice call. Simultaneous voice and data calls are permitted.

Telephone - Auto Answer (Option 600): Allows incoming calls to ring briefly, then are automatically answered in Hands Free mode.

Telephone - Automatic Outgoing Line (Option 604): If set to the auto mode, the set automatically selects the first free line it can find, starting with the Prime line, when the user begins dialing. If set to the manual mode, the user must go off-hook and select a line key. Note that this does not apply to SUPERSET 401+ telephones. Also beware that a line selected as a preferred line in the Expand Set subform of CDE Form 9 overrides this COS option. If the customer wants the system to automatically select a line (COS 604 set to auto), or if the customer wants to always manually select a line (COS 604 set to manual), do not select a preferred line in Form 9.

Telephone - Auto Hold Disable (Option 601): Disables the auto-hold feature. Auto hold allows a user to press another line appearance key, automatically placing a caller on the original line on hold.

Telephone - Background Music (Option 602): Allows a set to play music (from MOH) over the set speaker, when idle.

Telephone - Day/Night1 Switching (Option 691): Allows a telephone to be switched from Day service mode to Night1 service mode using a programmed feature key.

Telephone - Disconnect Alarm (Option 603): Allows for an alarm indication (log message is generated) when the set is unplugged.

Telephone - Enhanced Answering Position (Option 606): Allows a Mitel display telephone to be used as an enhanced answering position (see the Program Features section, Subattendant - Enhanced Functions).

Telephone - Guest Rm Template (0-3) (DN) (Option 610) : Allows certain configurations of Speed Dial keys and Feature Keys. Not available to the Mitel 5201 IP, SUPERSET 4001, or SUPERSET 401+ telephones. See the Hotel/Motel Feature Package Description section. This feature is part of the Hotel/Motel feature package.

Telephone - Handset Volume Saved (Option 614): Allows a Mitel telephone user to set the handset volume and have it saved for every call. **Note**: Volume returns to default setting when controller or telephone resets.

Telephone - Headset Operation (Option 612): Allows a Mitel telephone to be used as with a headset. Note that the feature, Headset Operation, is not supported on the Mitel 5201 IP or SUPERSET 4001 telephone.

Telephone - Limited New Call Ring (Option 611): Limits the new call ring for sets and subattendants that have a high amount of traffic on one or more line appearances or [iii]LDN indicators.

Telephone - Max Call Logs Allowed (0-20) (Option 620): Defines the maximum number of call logs assigned to the user of the Call Logging feature. If the value is 0, the Call Logging softkey will not appear in the SuperKey session for the Mitel display telephones with this COS. Reducing this number or changing this number to 0 will not delete call logs.

Telephone - Message Program (Option 605): Allows changing/programming of advisory messages from Mitel display telephones.

Telephone - Night Service Switching (Option 609): Allows a SUPERSET 4150, SUPERSET 4125, SUPERSET 4025, SUPERSET 430, SUPERSET 420, SUPERSET 4DN, 5020 IP, 5220 or 5224 IP telephone to change the night service status of the system or a tenant group.

Telephone - Offhook Voice Announce (Option 615): Allows the SUPERSET telephone to make off-hook voice announce calls.

Telephone - Room Status Display (Option 608): Allows a SUPERSET 4150, SUPERSET 4125, SUPERSET 4025, SUPERSET 430, SUPERSET 420, SUPERSET 4DN, 5020 IP, 5220 or 5224 IP telephone to check room status. This feature is part of the Hotel/Motel feature package.

Tie Trunk to DID Trunk Connect (Option 317): Allows an extension to connect a Tie trunk and a DID trunk together.

Tie Trunk to Tie Trunk Connect (Option 316): Allows an extension to connect two Tie trunks together.

Transfer Dial Tone (Option 251): Allows an extension to have a special distinctive dial tone, informing the user that there is a call on consultation hold.

Transparent Multi-Console Operation (Option 320): Allows consoles within a tenant group to read/cancel messages set by other consoles in the group. Also, any console in the group can answer recalls for any other console.

Trunk No Dial Tone Alarm (Option 805): Allows the system to raise an alarm if dial tone cannot be detected on it. The system will also take such a trunk out of service.

Trunk Recall Partial Inhibit (Option 403): Prevents a trunk from ringing the extension back while the extension is dialing, and goes on hook (phantom ringback).

Twin Phone (Option 276): Pairs a phone or a remote IP telephone with another phone to provide concurrent ringing and message waiting indication on both devices. See Phone Twinning.

UCD Music-on-Hold Timer (Option 256): Sets the length of time an incoming UCD caller hears music before being routed to an overflow answer point.

Voice Mail Port (Option 229): Allows a voice mail system to connect to this port.

Voice mail System Speed Dial Index (0-255) (Option 265): This option applies to ONS voice mail port hunt groups. When enabled, this option identifies a set/station accessing voice mail through the message waiting key to retrieve messages. The user listening to messages can also use the Messaging - Call Me Back feature to reply to messages. This COS option is Disabled by default. To enable, assign an index number from 0 to 255. This index number will point to an abbreviated dial entry in Form 31.

Wakeup Alarm Notification (Option 624): This option allows display sets to access wakeup alarm notification. When enabled, wakeup calls that are not answered after three attempts cause the LED of the Wakeup Alarm feature key to flash. Users can also access wakeup alarm details by selecting the feature key.

This index number is also used with QSIG and with centralized voice mail over IP trunks. Retrieving voice mail messages left by the QSIG message waiting supplementary feature is dependant on having COS Option 265, Voice mail System Speed Dial Index, programmed on the first QSIG trunk on the first PRI link. When centralized voice mail is used over IP Trunks, this Speed dial should be used to access the voice mail hunt group on the remote node. (Not required on the node where the voice mail resides.)

Form 04 - System Options/System Timers

This form specifies the system's options and timers that are system wide. See the System Options and Timers table for the complete list of options and timers, and the figure below for the form layout (the example used shows "BOTTOM" of the form where the new functionality is located).

Notes:

1. Attempts to enable unpurchased options causes the system to respond with PASSWORD/OPTIONS CONFLICT -- "QUIT" TO EXIT -- "ENTER" TO RE-EDIT. Conflicts are resolved by entering the correct password; a system reset is not required.
2. The system warns if changing an option requires a reset. The reset is automatic and occurs when the change is confirmed.
3. A FOR alarm message in the CDE forms header indicates that the System ID and Password do not match the database (occurs when a database from another system is installed in the controller). See Change or Enable System Options to enable the correct options.
4. For the AX, the existing system Option Maximum Devices (system Option 103) is used to control the ONS ports used. The default value for system option 133 TDM Bays is 1. For System option 132, DSP Configuration Options, the user can use Business Option 1 only; however, the system does not restrict the user to select other DSP Configuration Options.

IMPORTANT: Certain options in Form 04 could cause unexpected behaviors in system operation if changed from their factory-set (default) values. See System Options to Avoid for more information.

Figure: Form 04 Layout

Field Description

The header line indicates which set of options are selected; either the enabled or disabled options. Note that the system timers are included with the enabled options.

SYSTEM OPTIONS: This field lists the system option names. The option names cannot be modified. The option names are classified into two groups: enabled options and disabled options. When RESERVED appears as the option name, the option is not available.

STATUS: This field specifies which options are enabled, disabled or the value of the timers.

OPTION: This field lists the option number for each option or timer.

Softkeys

DISABLE/ENABLE: This softkey has two functions: it disables or enables System Options. The DISABLE softkey appears when the form shows the enabled options. Pressing the DISABLE softkey disables the selected option. The STATUS field shows the DISABLED prompt. The ENABLE softkey appears when the form displays the disabled options. Pressing the ENABLE softkey enables the selected option. The STATUS field shows the ENABLED prompt. **Note:** The enable (or disable) process is completed by pressing the ENTER softkey.

VARIABLE: This softkey appears only when System Option 55 is selected (displayed on the command line). Pressing the VARIABLE softkey sets the status of System Option 55 to Variable; the STATUS field shows the VARIABLE prompt. The selection is completed by pressing the ENTER softkey. Account Codes entered into Form 33, Account Code Entry, can be from 1 to 12 digits in length.

INTERNAL: This softkey appears only when System Options 57 and 58 are selected (displayed on the command line). Pressing the INTERNAL softkey sets the status of System Options 57 and 58 to Internal (for telephones with “Room Status” enabled) and only internal calls can be made. The selection is completed by pressing the ENTER softkey. The INTERNAL prompt disappears from the softkey display and the STATUS field now displays the INTERNAL prompt to indicate this selection.

LOCAL: This softkey appears only when System Options 57 and 58 selected (displayed on the command line). Pressing the LOCAL softkey sets the status of System Options 57 and 58 to Local (for telephones with “Room Status" enabled) and only internal and local calls can be made. The selection is completed by pressing the ENTER softkey. The LOCAL prompt disappears from the softkey display and the STATUS field now displays the LOCAL prompt.

LONG DIST: This softkey appears only when System Options 57 and 58 are selected (displayed on the command line). Pressing the LONG DIST softkey enables long distance calls. For telephones with "Room Status" enabled, internal, local and long distance calls can be made. The selection is completed by pressing the ENTER softkey. The LONG DIST prompt disappears from the softkey display and the STATUS field displays the LONG DIST prompt.

FIND OPTION: This softkey selects a specific System Option or System Timer. Pressing the OPTION softkey displays the ENTER OPTION: prompt on the command line. The selection is completed by entering a valid option number (1 to 58). The command line displays that System Option (or System Timer) name, status and number.

CHANGE: This softkey appears when System Option 138, Country Variant For Disconnect Tone Control. Pressing it displays additional softkeys labeled ARGENTINA; BOLIVIA; CHILE; ECUADOR; MX1 PA SV (Mexico, Panama and El Salvador); MEXICO 2; GUATEMALA; PERU; VENEZUELA; and DISABLED (USA and Canada). Note that MX PA SV applies to Mexico, Panama and El Salvador. Changing this option causes the system to reset.

CHANGE PLAN: This softkey appears when System Option 60, Tone Plan is selected. The only tone plan that the SX-200 ICP supports is the North American (NA) plan.

CHANGE: This softkey appears when System Option 132, DSP Configuration Options is selected. Pressing it displays additional softkeys labeled BUSINESS 1, BUSINESS 2, HOSPITALITY, ANALOG 1, ANALOG 2, and ANALOG 3: Use them to select the required DSP Configuration Options. Always use BUSINESS OPTION 1 (the default) for the CX and AX controller. If you change this option, you must reboot the system.

SHOW ALL/SHOW DISABLE/SHOW ENABLE: Use these softkeys to filter the displayed list of System Options. Pressing SHOW ALL displays the entire list of options. Pressing SHOW DISABLE displays currently disabled System Options. Pressing SHOW ENABLE displays enabled System Options. SHOW ALL is the default display. **Note:** Those options with a timer value are listed when the SHOW ENABLE softkey is pressed.

ENTER MOC: Use this softkey to enter the Mitel Options Code (MOC) that that is printed on the Feature Options Record sheet included with the software package. The MOC programs all options and timers for the system, eliminating the need to program them individually. Enter the MOC when Enabling FOR options.

The standard softkeys **CANCEL**, **ENTER**, **BOTTOM**, **TOP**, and **QUIT** are provided.

System Options and Timers

Table: System Options and Timers

Option Numbers	System Options / System Timers
01	Clock Format (example 16:00, 4:00, 4:00a)
02	Message Lamp Test Enable
03	Single Paging Amplifier
04	Message Waiting and Message Register Clear Print
05	Verified Account Codes
06	Analogue Networking SMDR
07	Cancel 24-Hour Message Waiting

Table: System Options and Timers

Option Numbers	System Options / System Timers
08	Five-Digit SMDR
09	Attendant Call Block
10	Attendant Conference Beeps
11	Automatic Wake-up
12	Automatic Wake-up Alarm
13	Automatic Wake-up Print
14	Automatic Wake-up Music
15	Data Demultiplexer
16	IP Set Voice Encryption
17	Discriminating Ringing
18	Discriminating Ringing Always
19	DID Server Application
20	Holiday Messages
21	Incoming to Outgoing Call Forward
22	Last Party Clear - Dial Tone
23	Message Register Count Additional Supervisions
24	Message Register Audit
25	Message Register Zero After Audit
26	No Overlap Outpulsing
27	Room Status Audit
28	SMDR Indicate Long Call
29	Telephone Last Number Redial
30	Mobile Extension CLID
31	Satellite PBX
32	Outgoing Call Restriction
33	Room Status
34	Auto Room Status Conversion / Auto Wake-up Print
35	DSS/BLF Call Pickup

Table: System Options and Timers

Option Numbers	System Options / System Timers
36	End Of Dial Character (#)
37	Calibrated Flash
38	Switch-Hook Flash
39	DATA SMDR Indicate Long Calls
40	Message Register Follows Talker
42	Silent Monitoring
43	ACD Silent Monitoring Beeps
44	ACD Reports
45	Disable PMS Logs
46	Rotary Dial Digit Translation Plan (0 - 3)
47	ARS Unknown Digit Length Time-out (2 - 60 Seconds)
48	Limited Wait For Dial Tone (1 - 15 Seconds)
49	Pseudo Answer Supervision Timer (10 - 60 Seconds)
50	Dialing Conflict Timer (2 - 10 Seconds)
51	Final Ring Time-out (1 - 30 Minutes)
52	Minimum Flash Timer (20 - 50 ms; in 10 ms increments)
53	Maximum Flash Timer (20 - 150 ms; in 10 ms incr)
54	DISA Answer Timer (1 - 8 Seconds)
55	Account Code Length (Variable or 2 - 12 Digits)
56	Auto Room Status Conversion / Wakeup Print Timer
57	Vacant / Reserved Room Default Call Restriction
58	Occupied Room Default Call Restriction
59	Receivers Reserved for Non-Auto-Attendant Use
60	Tone Plan
68	Not used
69	DTMF ON Timer (5-15 in 10 ms increments) default = 9 (90 ms)
70	DTMF OFF Timer (5-15 in 10 ms increments) default = 9 (90 ms)
71	Slot 10 FIM Capacity (2 or 3 Bays)

Table: System Options and Timers

Option Numbers	System Options / System Timers
72	Slot 11 FIM Capacity (2 or 3 Bays)
73	Advance to Daylight Savings Time (mm:dd:hh)
76	Go Back to Standard Time (mm:dd:hh)
79	Daylight - Standard Time Difference (30 to 240 minutes)
81	Enter offset from GMT (+/-hh:mm)
82	Use HW Echo Canceller (requires system reset)
83	Internet Gateway
84	Multiple Guest Suite Phones
85	Speak@Ease Integration
86	PRI Card: QSIG
87	Record a Call
88	Max TAPI Desktops (0...50 in increments of 5) **NOTE:** The SX-200 ICP does not support TAPI.
89	CLASS Functionality for ONS Sets
90	ACD Real Time Event
91	PRI Card: NFAS
92	PRI Card: D Channel backup
93	PRI Card: Remote LAN Access
94	PRI Card: Min/Max
95	PRI Card: Auto Min/Max
96	Number of Links (0-8)
97	Support Softkey Access to Voice mail
98	Support 3DN, 4DN and 400 Series Set Types
99	Fax Tone Detection
100	Mitel Options Password
101	System Identity Code

102	Feature Level <0-99>
103	Maximum Devices (768)
104	Maximum ACD Agents (Automatic Call Distribution)
105	Mitel Application Interface
106	Automated Attendant
107	Lodging
108	Property Management System
109	Remote Software Download
110	Maximum BNIC Cards (0 - 56)
111	Maximum BONS Cards (0 - 56)
112	SS 4000 Series Sets
113	Centralized Attendant / Voice mail
114	Maximum IP Sets
115	Maximum IP Trunks
116	Not used
117	Not used
118	Not used
119	Not used
120	

Number of Compression Resources

NOTE: Changing this option will require a system reset.	
121	Voice Mail License for Bilingual Prompts
122	Voice Mail License for Personal Contact Numbers
123	Voicemail License for DID Server (reserved for future implementation)
124	Voice Mail Property Management System
125	Licensed Embedded Voice Mail Boxes (0-748)
126	Email Messaging
127	Autoselect Park Orbits

128	Phonebook
129	Attendant Park and Page Key
130	Paging Tone
131	PC (2nd) Port on IP Phone
132	DSP Configuration Option
NOTE: Changing this option will require a system reset.	
133	TDM Bays (0-7)
134	Voicemail Recorded Announcement Device
135	Voicemail Control of MWI by DTMF Tones
136	Dual Function Key Timer (1-5)
137	Mark SMTP Forwarded Voicemails as Read
138	Country Variant For Disconnect Tone Control
NOTE: Changing this option will require a system reset.	

System Options to Avoid

Certain options in Form 04 could cause unexpected behaviours in system operation if changed from their factory-set (default) values.

Table: System Options to Avoid

Option #	Option Name	Default
60	Tone Plan	NA
69	DTMF ON Timer (5-15 in 10 ms increments)	9 (90 ms)
70	DTMF OFF Timer (5-15 in 10 ms increments)	9 (90 ms)
71	Slot 10 FIM Capacity (2 or 3 Bays)	2
72	Slot 11 FIM Capacity (2 or 3 Bays)	2
99	Fax Tone Detection	Disabled

Parameters arranged alphabetically

The following are not feature descriptions. Refer to the Features section for complete descriptions.

Account Code Length (Option 55: Sets the length of account codes (VARIABLE, or any fixed length between 2 and 12).

ACD Real Time Event (Option 90): Allows a PC to report real time events of ACD activities. The system transmits call status messages to the host computer reflecting the changes of state on the line or on the device.

ACD Reports (Option 44): Changes SMDR reports into the format required by the ACD TELEMARKETER Reporting Package.

ACD Silent Monitoring Beeps (Option 43): Enables the ACD agent to be notified when being monitored by the ACD supervisor.

Advance to Daylight Savings Time (Option 73): Specifies when the system should advance the system clock for daylight savings time (mm:dd:hh). If the month is set to 00, this option is disabled. By default, this option is disabled. See Daylight Savings Time Adjustment for more information.

Analog Networking SMDR (Option 06): Enable if SMDR records are to be kept for calls made via analog networking.

ARS Unknown Digit Length Time-out (Option 47): Sets the time ARS will wait for more digits when dialing a digit string which includes a variable-length account code (2 - 60 sec).

Attendant Call Block (Option 09): Allows call blocking on the system (Hotel/Motel Feature).

Attendant Conference Beeps (Option 10): Allows for warning beeps to be heard by the source and destination parties of a call, before the attendant enters into a conference with them, and during the conference.

Attendant Park and Page Key (Option 12): This option controls whether the PARK & PAGE or SYSTEM PARK softkey appears on the attendant console. Enabling the option reduces the number of keystrokes otherwise required for certain park and page operations. For more information, see Park and Page.

Auto Room Status Conversion/Auto Wakeup Print (Option 34): Allows an automatic change of room status and wakeup audit daily.

Auto Room Status Conversion/Auto Wakeup Print Timer (Option 56): Sets the time when the conversion/audit takes place (the default time is 00:00).

Automated Attendant (Option 106): Enables the Automated Attendant Feature Package - refer to the Automated Attendant Application Package section for further information.

Automatic Wakeup (Option 11): Enables the Automatic Wakeup feature.

Automatic Wakeup Alarm (Option 12): Allows an alarm to be raised at the attendant console when a wakeup alarm is unanswered three times.

Automatic Wakeup Music (Option 14): Allows music (MOH) to be heard upon answering a wakeup call.

Automatic Wakeup Print (Option 13): Allows a message to be printed on the default printer whenever a wakeup call is set up, canceled, answered or honored.

Autoselect Park Orbits (Option 127): If disabled, the user is required to dial an orbit number in the range 01 to 25 to park a call; otherwise, the system selects the orbit.

Calibrated Flash (Option 37): Allows the system to create the proper flash time to prevent confusion between a flash and a hang-up attempt. This applies to rotary-dial telephones, and DTMF sets with flash buttons.

Cancel 24-Hour Message Waiting (Option 07): Allows the system to automatically cancel message waiting indicators after 24 hours.

Centralized Attendant / Voice mail (Option 113): Allows the system to program and access centralized Attendant or Voice mail facilities. Refer to the Centralized Voice mail section for further information.

CLASS Functionality for ONS Sets (Option 89): Allows the system to deliver CLASS information to analog sets.

Clock Format (Option 01): Sets one of three clock formats. The three clock format options appear on softkeys: 24 HOUR, 12 HOUR, and 12 HOUR AM. The 24 HOUR enables a 24-hour clock (example 16:00). The 12 HOUR enables a 12-hour clock (example 4:00). The 12 HOUR AM enables a 12-hour clock with an "a" to represent a.m. (example 4:00a). There will be no letter to represent the p.m. It is recommended that you select the 12 HOUR AM option.

Country Variant For Disconnect Tone Control (Option 138): Specifies the tone that indicates the far-end has disconnected. There are 10 disconnect tone options: ARGENTINA; BOLIVIA; CHILE; ECUADOR; GUATEMALA; MX1 PA SV (Mexico, Panama, and El Salvador); MEXICO 2; PERU; VENEZUELA; and DISABLED (USA and Canada).

NOTE: Changing this option will result in a system reset.

Data Demultiplexer (Option 15): Enable if a Data Demultiplexer is connected to the system's printer port.

Data SMDR Indicate Long Calls (Option 39): Provides an identifying character in column 1 of the SMDR report to indicate the approximate length of a call.

Daylight - Standard Time Difference (Option 79): Specifies the time change between Daylight Savings Time and Standard Time (30 to 240 minutes). Default is 60 minutes. See Daylight Savings Time Adjustment for more information.

Dialing Conflict Timer (Option 50): Sets the time that the system will wait for conflict dialing (2 - 10 sec).

DID Server Application (Option 19): Enable automatic DID number assignment. A new AMC password must be entered to enable this option. **Note**: This option is mutually exclusive to System Option 123 Voicemail License for DID Server.

DISA Answer Timer (Option 54): Sets the time between seizure of the incoming DISA trunk and the provision of dial tone.

Disable PMS Logs (Option 45): Prevents the system from writing PMS log messages to the maintenance log.

Discriminating Ringing (Option 17): Enables the Discriminating Ringing feature.

Discriminating Ringing Always (Option 18): Causes discriminating ringing to be the normal ringing pattern for all calls on the system.

DSP Configuration Option (Option 132): Determines the number of DSP modules to configure on boot up. The options are BUSINESS 1, BUSINESS 2, HOSPITALITY, ANALOG 1, ANALOG 2, and ANALOG 3. The default is BUSINESS 1. For more information about the options, see DSP Configuration Options.

DSS/BLF Call Pickup (Option 35): Allows you to retrieve a call with a DSS (Direct Station Select) key on a multi-line Mitel telephone or on a PKM 48 or PKM 12 programmed in Form 09. The BLF lamp reflects the state of a directory number (DN). When the BLF lamp flashes, the DSS key can retrieve the held or ringing call.

DTMF OFF Timer (5-15) (Option 70): The value programmed, when multiplied by 10, represents how many milliseconds of silence there is between each tone. Valid values are 5 to 15, meaning 50 to 150 milliseconds. Default = 9, meaning 90 milliseconds silence between tones.

DTMF ON Timer (5-15) (Option 69): The value programmed, when multiplied by 10, represents how many milliseconds the DTMF tone stays on. Valid values are 5 to 15, meaning 50 to 150 milliseconds. Default = 9, meaning each tone is ON for 90 milliseconds.

If your system supports voice mail softkeys for the NuPoint Messenger feature, ensure that you set this timer for a minimum of 90 milliseconds (value 9).

Dual Function Key Timer (1-5) (Option 136): This system timer defines the amount of time users are given to page through secondary key functions (such as Hold And Page or Call Park - System Orbit). Valid values are 1 to 5 seconds with the default value being 2 seconds.

Email Messaging (Option 126): This purchasable FOR option enables the SX-200 ICP to forward e-mail using IMAP and/or SMTP. IMAP allows users to manage their voice messages with one or more e-mail clients, and to synchronize the status of messages throughout the system. SMTP allows users to download their voice messages to an e-mail client. It also allows administrators to receive automatic notification of alarms and E911 calls, and to manually send system logs to an e-mail address.

End of Dial Character (Option 36): Allows the user to dial a special character (#) to indicate to the system that there are no more characters coming. This eliminates the end of dial timer.

Enter offset from GMT (+/-hh:mm) Option 81: Specifies the difference in hours and minutes between the time zone that the SX-200 ICP is in and Greenwich Mean Time (GMT). The default is -0500 hours, which corresponds to the Eastern Time Zone in the U.S. and Canada. The offset appears in the timestamp on e-mail messages sent by the SX-200 ICP in response to 911 emergency calls, alarms and other events. See Emergency Calls (911) - Reporting by Email for more information. **Note:** Manual time changes (as opposed to automatic changes that the system makes to adjust for daylight savings) will require a manual update to the GMT offset.

Fax Tone Detection (Option 99): If enabled, allows the system to recognize Fax tone on incoming calls to the Automated Attendant. System Option 106, Automated Attendant, must be enabled first.

Feature Level <0-99> (Option 102): Provides access to selected functionality offered in the new release of software. The option is a purchasable FOR option and is incremented for each Feature Level. The level is enabled in all new installs and in software upgrades. The level is not enabled when sites receive software fixes. The fixes give sites new software, but not the new features. Click here for a list of feature-level dependent functionality.

Final Ring Time-Out (Option 51): Sets the time the system will allow an unanswered station to ring (1 - 30 min) before dropping the call. The default is one minute.

Five-Digit SMDR (Option 08): Allows SMDR to record 5 digit telephone extension numbers rather than the usual 4. Used in networking and hotel/motel applications.

Go Back to Standard Time (Option 76): Specifies when the system should set the system clock back to standard time (mm:dd:hh). If the month is set to 00, this option is disabled. By default, this option is disabled. See Daylight Savings Time Adjustment for more information.

Holiday Messages (Option 20): If enabled, the system automatically sets up holiday messages on Mitel display telephones at Christmas and the New Year (2 days before and after).

Incoming to Outgoing Call Forward (Option 21): Allows call forwarding to external numbers by incoming trunks.

Internet Gateway (Option 83): Allows an Ethernet connection to the Internet through the WAN interface of the CX controller. Disabled by default.

IP Set Voice Encryption (Option 16): If enabled, voice packets transmitted over IP are encrypted. This is in addition to encryption for signaling messages, which are always encrypted provided that the devices engaged in the call support the available encryption algorithms. Available algorithms include SSL and Secure MiNET for protection of signaling messages, and AES for protection of the voice stream. Most Mitel IP Phones support encryption. Enabled by default.

Note: The following devices do *not* support encryption: SpectraLink sets; Symbol MiNET Wireless IP sets, Embedded Voice Mail; NuPoint (6510); Speech Server; Your Assistant.

Last Party Clear - Dial Tone (Option 22): Allows the last party remaining in a call to receive dial tone rather than silence.

Licensed Embedded Voice Mail Boxes (0-748) (Option 125): Determines the number of voice mailboxes available for programming in Form 50, Mailboxes. The SX-200 ICP system includes 20 mailboxes; licenses for more can be purchased. The number does not include the Administrator's Mailbox and the Operator Mailbox which are provided at no charge. Once the value is set it cannot be lowered below the number of mailboxes programmed, nor can it be reset to zero if the voice mail cards are programmed in Form 01, System Configuration.

Limited Wait For Dial Tone (Option 48): Sets the time a trunk will wait before outpulsing digits (1 - 15 sec). A value of 1 second is recommended if ISDN and T1 trunks are present.

Lodging (Hotel/Motel) (Option 107): Enables the Hotel/Motel Feature Package.

Mark SMTP Forwarded Voicemails as Read (Option 137): Controls the state of voice mail messages forwarded to e-mail with SMTP. If the option is enabled, messages will change from "new" to "read" state when they are forwarded. If the option is disabled, forwarded messages will remain in their original state (for example, a "new" message will remain "new" after forwarding). Only "new" messages cause the user's MWI lamp to flash. Enabled by default.

Note: This option does not apply to voice mail messages that are deleted after forwarding.

Max TAPI Desktops (0...50 in increments of 5) (Option 88): Allows a PC to control first-party calls. The PC runs a TAPI-compliant software application that will communicate with the PBX via a SUPERSET 4150 or SUPERSET 4125 telephone.

NOTE: TAPI is not supported in the current release of the SX-200 ICP.

Maximum ACD Agents (Option 104): Displays the maximum number of ACD agents enabled, from 0 through 100, in increments of 5. This is the maximum that can be logged in concurrently.

Maximum BNIC Cards (0- 56) (Option 110): Specifies the quantity of BNIC cards that have been purchased as part of a package. No additional BNIC cards can be purchased separately. The quantity entered must exactly match the quantity on the FOR sheet.

Note: To confirm the number of BNIC cards installed in the system, start a maintenance session and enter REPORTS > SHOW > CONFIG > ALL. Each BNIC card displays as a DIGITAL LINE CARD * in the INSTALLED column.

Maximum BONS Cards (0 - 56) (Option 111): Specifies the quantity of BONS cards that have been purchased as part of a package. No additional BONS cards can be purchased separately. The quantity entered must exactly match the quantity on the FOR sheet.

Note: To confirm the number of BONS cards installed in the system, start a maintenance session and enter REPORTS > SHOW > CONFIG > ALL. Each BONS card displays as an ONS LINE CARD * in the INSTALLED column.

Maximum Devices (Option 103): Displays the maximum number (up to 768) of user devices enabled. The count excludes IP devices.

Maximum Flash Timer (Option 53): Sets the maximum time a set can be on-hook to be recognized as a flash. If this time is exceeded, it is recognized as an on-hook hang-up (20 - 150 ms, in 10 ms increments).

Maximum IP Sets (Option 114): Sets the maximum number of IP phones per SX-200 ICP. The MX supports a maximum of 192 IP phones (248 with Hospitality option) and the CX/CXi a maximum of 100. The AX supports a maximum of 248 IP devices or a maximum of 288 ONS devices.

Maximum IP Trunks (Option 115): Sets the maximum number of IP Trunks per SX-200 ICP. The maximum number is 16 on the CX and 30 on the MX and the AX.

Message Lamp Test Enable (Option 02): When enabled, allows testing of telephone message lamps by the system.

Message Register Audit (Option 24): Allows the attendant to print the message register count for all rooms that have a count greater than zero.

Message Reg. Count Additional Supervisions (Option 23): Allows meter pulses to be counted after supervision has been received from an outgoing trunk.

Message Register Follows Talker (Option 40): Allows the last party connected to a trunk to be charged with the register count.

Message Register Zero After Audit (Option 25): Allows the system to automatically clear all message registers after an audit.

Message Waiting & Message Register Clear Print (Option 04): If enabled, a message is printed on the default printer whenever a message register is cleared or a message waiting is canceled.

Minimum Flash Timer (Option 52): Sets the minimum time a set must be on-hook before it is recognized as a flash (20 - 50 ms, in 10 ms increments).

Mitel Application Interface (Option 105): Enable if the Mitel Application Interface Package (MAI) was purchased (listed on FOR sheet).

Mitel Options Password (Option 100): The 8 digit numeric code provided by Mitel and entered by the user. It is required to decode and enable the options selected. If the numeric code provided by Mitel exceeds 8 digits, enter only the last 8 digits.

Enable only the purchased options listed on the FOR sheet that is included with the system software package. An error results when a purchased option is not enabled, when an enabled option is not purchased, or when the system id does not match.

Users must purchase the option and its associated password through their usual ordering channels prior to attempting to install or implement the option.

Mobile Extension CLID (Option 30): Mobile Extension provides ability for a twin device to receive the caller's name/number in the same way it's received on the desk phone.

Multiple Guest Suite Phones (Option 84): Provides the ability to group a number of telephone lines through interconnected hotel or motel rooms (suites), for the purposes of billing and shared telephone services.

No Overlap Outpulsing (Option 26): Forces ARS to collect all dialed digits before outpulsing them on a trunk.

Number of Compression Resources (Option 120): This purchasable FOR option controls the number of G.729 codecs available to IP devices in the system. Compression enables more devices to share available bandwidth. The option is purchasable in multiples of 1 to a maximum of 24. The default value is 0, which is displayed as a blank field. The quantity entered must match the quantity on the FOR sheet.

NOTE: Changing this option will require a system reset.

Number of Links (0-8) (Option 96): Limits the number of T1 type links from the NSU or from a PRI card in a Peripheral cabinet, or for the onboard Dual T1/E1 Framer module(s). This option is not required for links on the T1 trunk card. The total number of links available in the system is 8 (combination of purchasable links and non-purchasable links).

Occupied Room Default Call Restriction (Option 58): Sets default call restriction for occupied rooms; one of INTERNAL, LOCAL or LONG DISTANCE.

Outgoing Call Restriction (Option 32): Prevent unauthorized trunk calls after a guest has checked out of the room.

Note: Enable Option 32 only if a property management system (PMS) is being used.

Paging Tone (Option 130): When enabled, both paging and paged parties hear paging tone instead of a one second burst of ringback tone when a directed (set-to-set) page, group page or all set page is performed. Paging tone is the same tone a user hears when accessing PA paging.

PC (2nd) Port on IP Phone (Option 131): This purchasable FOR option activates the second network port on IP phones which allows an attached PC to access the network. The phone also requires COS Option 280 (Form 03).

Phonebook (Option 128): This purchasable FOR option enables the Dial-by-Name (Phonebook) feature. Access to the feature also requires an entry in the "Phonebook Number For This Tenant" field in Form 19.

Note: Phonebook and Speak@Ease (Option 85) cannot both be enabled.

PRI Card: Auto Min/Max (Option 95) Controls the number of incoming and outgoing calls for each type of service within a single link or multiple links. The level of control ranges from minimums and/or maximums for an entire range of links down to control for a specific directory number. The Min/Max automatically changes to different Min/Max databases depending on the time of day programming. Min/Max is required in order to program Auto Min/Max.

PRI Card: D-Channel backup (Option 92) Used for signaling to establish and maintain the circuit, and to send user data. D-Channel backup is used in case the primary D- Channel fails. PRI Card: NFAS is required in order to program PRI Card: D-channel Backup.

PRI Card: Min/Max (Option 94) Allows a customer to control incoming and outgoing call traffic. Minimums are assigned to ensure that a particular type of call (such as INWATS) always has a set amount of lines available. Maximums are assigned to limit certain types of calls, i.e., OUTWATS. This ensures that resources are not used up by a single type of call. Different Min/Max databases can be created for different times of the day or for special occasions such as telethons or infomercials.

PRI Card: NFAS (Option 91) (Non-facility associated signaling) Allows the customer to purchase a single D-channel that will provide all signaling for multiple groups of B-channels.

PRI Card: QSIG (Option 86) Allows you to connect PBXs from different vendors together to form a private network. QSIG currently supports incoming calls and incoming Calling Name.

PRI Card: Remote LAN Access (Option 93) Provides LAN access to the wide area network (WAN) for both incoming and outgoing calls through LAN servers (for example, routers, bridges) using Mitel's PRI Gateway interface.

Property Management System (Option 108): Enables the Property Management System (PMS) feature package.

Pseudo Answer Supervision Timer (Option 49): Sets the time a trunk will wait while providing pseudo answer supervision (10 - 60 sec).

Receivers Reserved for Non-Auto-Attendant Use (Option 59): Reserves receivers for standard call processing to safeguard against the auto-attendant feature using all of the available receivers. The default value is UNKNOWN. The system prevents the user from programming any Auto-Attendant groups until this value is changed to a number from 1 to 99 or ALL. For more information, see Automated Attendant - Programming.

Record a Call (Option 87): Allows the user to record both ends of a two-party conversation as a message in the user's mailbox.The state of this option is DISABLED by default.

Remote Software Download (Option 109): Allows the user to upgrade the system from a remote site. Refer to Mitel On-Line for further information.

Room Status (Option 33): Allows the status of a room to be displayed and changed. Provides the three call privileges: internal, local and long distance. Set the default call privileges using Options 57 and 58.

Note: Enable Option 33 only for the hotel/motel feature package.

Room Status Audit (Option 27): Allows room status printouts.

Rotary Dial Digit Translation Plan (Option 46): Chooses one of four available digit translation plans for rotary dial signaling. See Rotary Dial Digit Translation in the Program Features section.

Satellite PBX (Option xx): Enable if the system is to serve as a satellite PBX.

Silent Monitoring (Option 42): Allows silent monitoring of sets.

Single Paging Amplifier (Option 03): Enable if system has only one paging amplifier output. Allows one user at a time to access the paging feature.

Slot 10 FIM Capacity (2 or 3 Bays) (Option 71): Specifies whether a Control Dual FIM Carrier (2) or a Control Triple FIM Carrier (3) is installed in slot 10 (default is 2).

Slot 11 FIM Capacity (2 or 3 Bays) (Option 72): Specifies whether a Control Dual FIM Carrier (2) or a Control Triple FIM Carrier (3) is installed in slot 11 (default is 2).

SMDR Indicate Long Calls (Option 28): Allows the SMDR record to flag calls which are greater in duration than 5 minutes.

Speak@Ease Integration (Option 85): This purchasable FOR option is responsible for the appearance of the Speak@Ease softkey on the display set. This key provides direct access to the Mitel Speech Server.

SS 4000 Series Sets (Option 112): Allows programming of SUPERSET 4000-series telephones.

Support 3DN, 4DN and 400 Series Set Types (Option 98): This purchasable FOR option provides support for SUPERSET 3DN, SUPERSET 4DN and 400 Series telephones.

Support Softkey Access to Voice mail (Option 97): Enables voice mail softkeys on all Mitel telephones that have softkeys. Note that voice mail softkeys are supported on SX-200 ICP embedded voice mail, on Mitel Express Messenger, and on the DNIC-based NuPoint Messenger system. See Softkey Support.

Switch-hook Flash (Option 38): Allows stations to flash the switch-hook to access system features.

System Identity Code (Option 101): Displays the system identity code from the System ID Module (MX) or System i-Button (AX/CX) installed on the main board of the controller. It must match the number printed on your FOR sheet, and is either five numeric digits (MX) or twelve hexadecimal characters (AX/CX).

TDM Bays (Option 133): This purchasable FOR option controls the number of digital bays that boot up -- from 0 to 7. Note that this option applies only to digital bays and NOT to PRI bays. PRI bays (either PRI cards or NSUs) are controlled by System Option 96, Number of Links.

Telephone Last Number Redial (Option 29): Allows telephone users and attendants to redial the last manually-dialed number with a single key or using an access code.

Tone Plan (Option 60): Shows NA for North America, the only tone plan the SX-200 ICP supports.

Use DSP Echo Canceller (Option 82): Use this option to specify which system resources are used for echo cancellation. If enabled, hardware resources are used. If disabled (the default), DSP resources are used.

Note: This option requires a system reset. Enable this option (use DSP resources for echo cancellation) only when instructed to do so by Mitel Product Support.

Vacant/Reserved Room Default Call Restriction (Option 57): Sets default call restriction for vacant/reserved rooms; one of INTERNAL, LOCAL or LONG DISTANCE.

Verified Account Codes (Option 05): Enable if verified account codes are to be used.

Voicemail Control of MWI by DTMF Tones (Option 135): Allows a Centralized Voice mail system to control the Message Waiting Indicator (MWI) lamp on remote PBXs by transmitting DTMF digits over tie trunks.

Voicemail License for Bilingual Prompts (Option 121): This purchasable FOR option provides simultaneous prompts in two of the three available languages: English, French, and Spanish. Callers reaching the auto attendant or a subscriber's mailbox can dial the Language Change Mailbox number (defined in Form 49, Voice mail Options) to hear subsequent prompts in the alternate language.

Voicemail License for Personal Contact Numbers (Option 122): If the Personal Contacts Numbers option has been purchased (listed on FOR sheet) then mailbox owners can program alternate numbers (cell phone, pager, fax etc.) where callers can contact them instead of leaving a message. Callers reaching the owner's mailbox will hear the owner's greeting followed by prompts such as "...to reach this person's cellular phone, press C, the 2 key." Use of this feature can be enabled or disabled for the entire system in Form 49, Voice mail Options).

Voicemail Property Management System (Option 124): This purchasable FOR option adds PMS Integration with Voice mail and voice mail-related Hospitality Features to the standard SX-200 ICP system features. Enabling or disabling this option requires a system reset for the change to take effect. See Resetting the System for more information.

Voicemail License for DID Server (Option 123): Reserved for future implementation.

Voicemail Recorded Announcement Device (Option 134): This purchasable FOR option enables embedded voice mail ports to be used to provide recorded announcement device (RAD) service, eliminating the need for external tape machines or other audio-playing devices. Enabling this option allows the RAD ports to be placed in RECORDING type hunt groups. For more information see, Recorded Announcement Devices.

Form 05 - Tenant Interconnection Table

This form specifies which tenant groups may be connected together. The system allows for a maximum of 25 tenant groups. Each group specifies its own trunk answering points, attendant answering points and night service status.

Figure: Form 05 Layout

Field Description

Initially, the system interconnects all tenant groups. The asterisk (*) character indicates this condition. When the system inhibits tenant group interconnection, it is indicated by the period (.) character. The tenant group numbers are listed in the header line and the first column. The letter (O) functions as a marker and cannot be modified.

Softkeys

TENANT NUM: The TENANT NUM softkey allows a user to select a tenant group by number. Press the softkey to display the ENTER TENANT GROUP NUM: prompt. Enter the 2-digit tenant number (1 to 25) to display the tenant group with a series of ‘*’ characters (allow interconnection) and ‘.’ characters (disallow interconnection). Control cursor movement on the command line with right and left cursor control keys.

ALLOW/DISALLOW: This softkey has two functions: it enables or disables interconnection between tenant groups. Pressing the DISALLOW softkey disables the interconnection between the Row Tenant (displayed by the command line) and the Column Tenant (highlighted by the cursor) in one direction only. For example, when modifying connections for tenant group 5 (the command line displays line 5) and the DISALLOW softkey is pressed when the cursor is under the sixth column, then tenant group 5 cannot call tenant group 6. However, tenant group 6 can still call tenant group 5. Total interconnection is inhibited only when a '.' (disallow) character is inserted at row 6 (tenant group 6) under the fifth column (tenant group 5). The softkey now displays the ALLOW prompt. Pressing the ALLOW softkey enables the unidirectional interconnection between the selected tenant groups; the '*' character replaces the '.' character.

Standard softkeys **CANCEL**, **ENTER**, **BOTTOM**, **TOP**, and **QUIT** are also provided.

Form 06 - Tenant Night Switching Control

In some systems it is necessary for one attendant to control the night service switching of more than one tenant group. This form specifies which tenant groups are switched to night service simultaneously and which tenant has control.

Note: The system defaults to tenant groups switching to night service independently of each other.

Figure: Form 06 Layout

Field Description

Initially, the system inhibits tenant groups from switching each other into night service. This condition is indicated by the period (.) character. When the system permits tenant groups to switch each other into night service, it is indicated by the asterisk (*) character. The tenant group numbers are listed in the header line and the first column. The letter (O) functions as a marker and cannot be modified.

Softkeys

TENANT NUM: The TENANT NUM softkey allows a user to select a tenant group by number. Pressing this softkey displays the ENTER TENANT GROUP NUM: prompt. Entering the 1- or 2-digit tenant number displays that tenant group with a series of '.' characters (single tenant group entry into night service) and '*' characters (multiple tenant group entry into night service). Cursor movement on the command line is controlled by the right and left cursor control keys.

SWITCHED/NOT SWITCHED: This softkey has two functions: it enables or disables multiple tenant group switching into night service. This softkey displays the SWITCHED prompt when the cursor is at a period (.) character. Pressing the SWITCHED softkey specifies that the tenant group being programmed (as indicated by the row number) can switch the other tenant group (as indicated by the column number) into night service. The system reflects this modification by replacing the '.' character with the '*' character. The softkey now displays the NOT SWITCHED prompt. Pressing the NOT SWITCHED softkey specifies that the tenant group being programmed cannot switch the other tenant group into night service. The '.' character replaces the '*' character and the softkey now displays the SWITCHED prompt.

The standard softkeys **CANCEL**, **ENTER**, **BOTTOM**, **TOP**, and **QUIT** are also provided.

Form 07 - Console Assignments

This form specifies the physical location and the CESID number for each SC1000 attendant console. The system provides an entry line in Form 07 for each digital line circuit in a high power (upper) slot not assigned to a Mitel telephone, DATASET, or Music-on-Hold/Paging Unit. The system can support a maximum of 11 attendant consoles. There can be no more than four Consoles (applicable to the SC1000) per Digital Line Card.

For the 5540 IP Console, this form specifies an IP PLID for the IP Console. The IP PLIDS can be assigned in Form 07 (for IP Consoles) and in Form 09 (for IP sets). Once the PLID is occupied in one of the forms, it will not be available in the other form. The SX-200 can support a maximum number of eleven 5540 IP Consoles. The IP line card can support any number of of 5540 IP Consoles (no restrictions).

A new field MAC ADDRESS is now available to enter the MAC (Media Access Control) Address of the 5540 IP Console.

Each attendant console has a Class of Service (COS), a Class of Restriction (COR), a tenant group number and an extension number. The extension number enables calls between attendant consoles. These calls appear on the INTERNAL softkey. Note that the provision of a COR means that attendant consoles are not necessarily toll-allowed on all calls.

Note: If a console is a member of a ring group, it can not be deleted using this form.

Figure: Form 07 Layout

Field Description

EXT NUM: This field displays the extension number of each attendant console. Calls directed to the console's extension number route to softkey 2 on the attendant console.

COS: This field lists the Class-of-Service number specification for each console (1 to 50).

COR: This field lists the Class-of-Restriction number specification for each console (1 to 25).

TENANT: The tenant group for each attendant console is specified in this field (1 to 25).

COMMENTS: This field is reserved for notes about each console. It contains a maximum of 15 alphanumeric characters.

For detection and reporting 911 calls to attendant consoles, enter the physical location of each console in the Comments field. Type an "@" followed by the physical location of the console. All consoles are notified of a 911 call regardless of their tenant group. Only the text following the @ is displayed on the consoles. A maximum of 14 characters can be displayed on the console.

CESID: This field is reserved for a maximum of 10 digits that identifies the voice device. The string of digits provide the index that accesses the ALI database. The ALI database provides the emergency administration center the name, address, postal code, and general location of the user of the voice device that originates a 911 call. Each voice device is assigned a unique CESID.

Users that have a unique network number (residential and DID users) should have the same number assigned for their CESID. Users that share access to network connections and that do not have DID numbers must request a CESID for each user from the public network authority.

MAC ADDRESS: This field lists the MAC (Media Access Control) Address of the IP Console. The MAC address is a unique 48-bit address that is assigned to each IP device at the time of manufacture. It identifies specific hardware devices on a TCP/IP network.

The CESID and MAC ADDRESS field overlay each other. To toggle between them press the number nine softkey (SHOW MAC/SHOW CESID).

Softkeys

The standard softkeys **BAY/SLT/CCT**, **CANCEL**, **DELETE**, **ENTER**, and **QUIT** are provided. The **EXPAND PKM** softkey only appears if the FOR Option 102, Feature Level is set to level 1 or greater.

EXPAND PKM : Allows you to program up to two PKM 48 devices directly attached to a SUPERCONSOLE 1000 (5448 PKM for the 5540 IP Console). You can only program DSS/BLF keys on the PKM 48 (5448 PKM for the 5540 IP Console). The appearance of the EXPAND PKM softkey is dependant on the FOR Option 102, Feature Level being set to level 1 or greater.

SHOW CESID: Allows the MAC ADDRESS field to switch to a CESID field in order to show the CESID number (10 digit maximum).

SHOW MAC: Displays the MAC ADDRESS field instead of the CESID field.

Subform 07- Expand PKM

The SUPERCONSOLE 1000 can support up to two PKM 48 devices. Each PKM 48 has 48 programmable keys that may be programmed as DSS/BLF keys only. The appearance of this form is dependant on the FOR Option 102, Feature Level being set to level 1 or greater.

The Form 07 subform provides the number (1 or 2) to identify each PKM 48.

PKM 48 number 1 programming precedes the key number with a 1.

PKM 48 number 2 programming precedes the key number with a 2.

Figure: Form 07 Expand PKM Subform Layout

Field Description

KEY: This field lists the PKM 48 key numbers and cannot be modified. The key numbers are preceded by a 1 or a 2. The 1 or 2 indicate the PKM address. The key numbers on a PKM 48 correspond to those in this Figure.

TYPE: This field lists the key function. The default assignment is “Speed Dial." If the line is programmed as a BLF appearance, the key type is “Busy Lamp." If the line is programmed as a Mailbox appearance, the key type is "Mailbox."

DIR: If the key is a line appearance, the directional variant of the line (In/Out or Incoming) is shown.

RING: If the key is a line appearance, the ringing variant of the line (Immed, Delay, or None) is shown.

SEC: If the key is a line appearance, this field indicates (Yes or No) whether the secretarial variant is enabled. For a Busy Lamp key, setting this field to YES causes an immediate release when the DSS key is pressed.

DSS: NO in this field indicates that the DSS key cannot call the selected station. CALL in this field allows the DSS key to call the selected station. PAGE in this field allows the DSS key to page the selected station.

EXT NUM: This field contains the extension number of the line. This applies to the prime line, key lines, multiple call lines, personal outgoing lines, Mailbox lines, or BLF/DSS lines. When this field is filled for a specific key, no entry is allowed in the corresponding TRK ACC field. The EXT NUM field is blank if the key directly selects a trunk.

TRK ACC: This is either a trunk number (CO Line) or an ARS leading digit string for a CO line group. These come from Form 14, Non-Dial-In Trunks, or Form 26, ARS: Digit Strings. Only non-dial-in trunks may be entered here. Note that when this field is filled for a specific key, no entry is allowed in the corresponding EXT NUM field.

LABEL: Only used if the key type is defined as LDN.

Softkeys

BLF/DSS: Programs the selected key as a busy lamp field appearance (for the associated extension number). When a PKM 48 is associated with a console, this and MAILBOX are the only key types supported.

MAILBOX: Program the selected key as a Mailbox appearance (for the associated extension number). When a PKM 48 is associated with a console, this and BLF/DSS are the only key types supported.

KEY: This softkey selects a key by number. Pressing the KEY softkey displays the ENTER KEY NUM: prompt on the command line. Any key may be selected except for Key 01 (Prime Line).

The standard softkey **QUIT** is also provided.

Form 08 - Console LDN Assignments

This form specifies the LDN assignments for the consoles. A maximum of nine LDN assignments can be programmed for each attendant console. The attendant LDNs are assigned to the softkeys. Console softkey 1 is reserved for the RECALL function. Each LDN assignment is identified by a directory number. The directory numbers are subject to the same constraints as all Listed Directory Numbers (i.e., number conflicts are not allowed). If there are many attendant consoles in one tenant group and if “DIAL 0” calls are shared, then a common Listed Directory Number must be specified for the consoles. Note that this form is related to the Form 09 Expand Set Subforms, where LDN keys are programmed for subattendants.

Figure: Form 08 Layout

Field Description

BAY/SLT/CCT: This field specifies the physical location of the attendant console being programmed. This form relates to the programmed consoles in Form 07.

KEY: This field displays the console softkeys 2 through 10 (10 is displayed as 0). The KEY field cannot be modified.

DIR NUMBER: This field is reserved for assigning a directory number for console softkeys 2 to 10. This number (a maximum of five digits) links this form to Form 19 (Call Rerouting Table), and to Form 14 (Non-Dial-In Trunks), where the call type is defined for the directory number. An LDN directory number can only appear once per console. If it is required that two consoles share the same LDN, then both consoles must be in the same tenant group.

LABEL: This field specifies the actual text that the console LCD displays as softkey prompts. The LABEL field provides for a maximum of 12 characters. The label for console softkey 1 defaults to RECALL and cannot be modified. The label for console softkey 2 defaults to INTERNAL. Softkey 2 is shared between the extension number programmed in the DIR NUMBER field and calls directed to the attendant console’s extension number (as defined in the EXT NUM field of Form 07, Console Assignments). Console softkeys 2 to 10 can be edited. **Note**: Label display on 5330 and 5340 IP Phones may be truncated to fit in allowable space.

COMMENTS: This field further specifies the attendant LDN assignments with text. The COMMENTS field has a maximum of 15 characters. It is stored by the system but not used.

Softkeys

NEXT: Pressing the NEXT softkey displays the physical location (bay, slot and circuit numbers) of the next programmed attendant console. If the physical location of the last programmed console is displayed, then pressing this softkey again displays the bay, slot and circuit numbers of the first programmed console.

The standard softkeys **BAY/SLT/CCT**, **CANCEL**, **DELETE**, **ENTER**, **BOTTOM**, **TOP**, and **QUIT** are also provided.

Form 09 - Desktop Device Assignments

This form assigns telephones to the system.

Figure: Form 09 Layout

Field Description

BAY, SLOT and CCT: These fields list the physical location number of each station, telephone or embedded voice mail port. They are generated by the system based on what was entered in the PROGRAMMED field of Form 01, System Configuration. These fields cannot be modified.

TEN: This field lists the tenant group number for each station or telephone. Default tenant number is one.

EXTN: This field lists the extension number of each station or the prime line extension number of each telephone.

COS: This field lists the Class-of-Service number for each station or telephone. Default COS number is 1.

COR: This field lists the Class-of-Restriction number for each station or telephone. Default COR number is 1.

TYP: The TYP field will display **Wrlss** for a Symbol MiNET Wireless Phone, **Spctr** for a SpectraLink NetLink Wireless Phone, **TMX3K** for a TeleMatrix 3000IP Phone, **5201** for a 5201 IP Phone, **5010** for a 5010 IP Phone, **5020** for a 5020 IP Phone, **5207** for a 5207 IP Phone, **5212d** for 5212 Dual Mode IP Phone, **5215** for a 5215 IP Phone, **5215d** for a 5215 Dual Mode IP Phone, **5220** for a 5220 IP Phone, **5220d** for a 5220 Dual Mode IP Phone, **5224d** for 5224 Dual Mode IP Phone, **5304** for 5304 Dual Mode IP Phone, **5312** for 5312 Dual Mode IP Phone, **5324** for 5324 Dual Mode IP Phone, **5330** for a 5330 IP Phone, **5340** for a 5340 IP Phone, **4001** for a SUPERSET 4001; **4015** for a SUPERSET 4015; **4025** for a SUPERSET 4025; **4125** for a SUPERSET 4125; and **4150** for a SUPERSET 4150; **401** for a SUPERSET 401+; **410** for a SUPERSET 410 and for a blank DNIC port; **420** for a SUPERSET 420; and **430** for a SUPERSET 430; For a SUPERSET 4150, SUPERSET 430 or SUPERSET 4DN programmed as an enhanced subattendant, **SUB** is displayed. For subattendants, a SUPERSET 4150 displays **SUB,** a SUPERSET 430 displays **430S,** a SUPERSET 4DN displays **4DNS**. For a Mitel 5020, 5220, 5220 Dual Mode IP Phone programmed as an enhanced subattendant, **SS5220S** is displayed. For a Mitel 5224 Dual Mode IP Phone programmed as an enhanced subattendant, **SS5224S** is displayed. For a Mitel 5324 Dual Mode IP Phone programmed as an enhanced subattendant, **5324S** is displayed. For Mitel 5330 and 5340 IP Phones programmed as enhanced subattendants, **5330 S/ATT** and **5340 S/ATT** is displayed. For a programmed DMP, **DMP** is displayed. For a Your Assistant Softphone, **YAPRO** is displayed. For a DSS/BLF Interface Module, **HOST** is displayed. For SUPERSET display telephones or DSS/BLF Interface Modules that are programmed with a PKM, an asterisk (*) precedes the set type (e.g., ***420, *HOST**). The TYP field displays **Stn** for analog device connected to an ONS circuit in the controller, peripheral cabinet or ASU. For an ONS music source, **Mus** is displayed. **Door** is the relay used to operate a remote door opener. Ports for the embedded voice mail system display as **VMAIL**.

PAGE: Shows the paging group that the telephone is in (the default is a blank field - no paging group). Valid entry is a number between 1 and 50.

NAME: This field is reserved for a set name up to 10 characters long. The name's first character must NOT be an asterisk (*). To enable callers to dial by name with the auto attendant, enter the last name before the first name (e.g., Jones Mark or Jones M, not Mark Jones or M Jones). Do enter periods, commas, or other punctuation as part of the name.

Note: When COS 277 is enabled on a new device created in Form 9, an attempt is made to automatically create a mailbox for it. No error message is given if the attempt fails. Common reasons for failure include the device extension number and the default mailbox number are unequal lengths, the maximum devices are already programmed, or a mailbox already exists with this number. The name entered in Form 9 is also used to create the voice mail Phonebook. Care should be taken to enter the name as it will be used for Phonebook searches. For more information, see Phonebook.

Note: For calls routed over IP trunks, the @ character displays as a pound sign (#) on the called party's phone.

Note: Changing the information in this field will automatically set display for

ASSOC: Associates the device in the TYP field with another device. A Mitel telephone can be associated with a single modem (enter the ONS port extension number). More than one telephone can be associated with the same modem. This field is also used to associate a PKM with an attendant console (the PKM HOST). If the TYP field is HOST then only the extension of a console can be entered in the ASSOC field.

The device in the TYP field device may be disassociated with the device in the ASSOC field simply by entering another device extension number, or by deleting the existing extension number (enter a space).

Hotel/motel suite services and the Phone Twinning feature use the ASSOC field to associate a secondary telephone to a primary telephone (the Primary DN Number is entered in the ASSOC field). CDE allows you to associate up to four telephones to a single Primary DN.

Door Relays programmed in Form 18 use the ASSOC field to associate the general-use relays to ONS door phones.

COMMENTS: This field is reserved for additional data (a maximum of 15 characters). It is stored by the system.

The E911 feature alerts the attendant if an extension user places a 911 call and identifies the extension that placed the 911 call. If the E911 feature is enabled, you can have the console display the physical location of the telephone that placed the E911 call. The location information can also be sent by e-mail to addresses specified in Form 52. You must enter the physical location of the set in the Comments field by typing an "@" followed by the physical location of the telephone. Only text following the @ is displayed. A maximum of 14 characters can be displayed on the console. For an example, see the Comments field in this Figure. Note that if you program the physical location, the tenant information will not be displayed in the E911 alarm for the extension. If E911 notifications are sent to e-mail, the contents of the Comments field will be added to the Subject line of e-mail messages.

The COMMENTS field overlays the CESID field. Pressing the SHOW CESID softkey switches the COMMENTS field to read CESID. Pressing the COMMENTS softkey switches the CESID field to read COMMENTS.

CESID: This field is reserved for a maximum of 10 digits that identifies the voice device. The string of digits is the index for access to a network PSAP ALI database. The ALI database provides the emergency administration center information such as the name, address, postal code, and general location of the user of the voice device that originates a 911 call. In general, each voice device will have a unique CESID.

If the device type is a BLF, Host, or DMP, Form 9 will block entries in the CESID field.

Users that have a unique network number (residential and DID users) should have the same number assigned as their CESID. Users that share access to network connections and that do not have DID numbers must request a CESID for each user from the public network authority.

Note: To send CESID data to a Release 6 (or later) PRI card, the *03 digit sequence must be programmed in the Digits To Be Inserted Field in Form 22 (ARS: Modified Digit Table) for all ARS routes used for outbound calls.

The CESID, MAC ADDRESS, PIN and COMMENTS field overlay each other. To cycle between them, press the number nine softkey.

MAC ADDRESS: This field is used to enter the MAC (Media Access Control) Address of IP telephones. The MAC Address is a unique 48-bit address that is assigned to each IP device at the time of manufacture. It is used to identify specific hardware devices on a TCP/IP network.

The CESID, MAC ADDRESS, PIN and COMMENTS field overlay each other. To cycle between them, press the number nine softkey.

PIN: This field is used to enter a four-digit Personal Identification Number (PIN) to a Symbol MiNET Wireless phone, or to a physical or phantom device that is programmed to support the Secure Hot Swap feature. Devices that have been hot swapped display an asterisk (*) in the PIN field.

The CESID, MAC ADDRESS, PIN and COMMENTS field overlay each other. To cycle between them, press the number nine softkey.

Softkeys

AUTO PROGRAM: This softkey automatically programs all unprogrammed circuits on installed cards. The softkey does not support Analog Interface Modules, or consoles.

All unprogrammed circuits on installed cards are automatically programmed as follows:

- ONS and OPS devices are assigned as
  - COS = 2
  - COR = 1
  - Tenant = 1
- DNIC and IP sets are assigned as
  - COS = 1
  - COR = 4 for all set types except SUPERSET 430 and SUPERSET 4150
  - COR = 5 for the SUPERSET 430 and SUPERSET 4150
  - Tenant = 1

Note: Before an extension number can be automatically assigned, a DNIC set or DMP must be physically connected to the circuit. If no set or console is detected on a circuit, that circuit is skipped and no extension number is assigned.

To automatically program all connected stations, DNIC sets and DMPs

1. Install all ONS cards, OPS cards, and DNIC cards. Do not program stations, DNIC, sets or DMPs.
2. Connect the DNIC sets or DMPs to the circuits.
3. Access Form 09, Desktop Device Assignments.
4. Press Auto Program.

   The system prompts for an extension number to assign to the first auto-programmed extension. The number is incremented by one for each subsequent extension -- for example, 100, 101, 102.
5. Enter a starting extension number.

   Conflicts with existing extension numbers result in an error message and a prompt to press the CONTINUE or ABORT softkey. Auto-programming will not proceed to the next BAY/SLT/CCT until a valid extension number has been found for the current position.

   There are no restrictions on the length of the starting extension number. However, it is recommended that the length be equal to the “Length of Mailbox Numbers” in Form 49, to take advantage of automatic mailbox creation.

   After verifying that the unprogrammed sets are physically connected, the system automatically programs all unprogrammed circuits. The system displays all of the devices that have been programmed.

   If a circuit cannot be programmed because an extension number has already been assigned to that circuit, a warning is displayed. Press ABORT to cancel the procedure or press CONTINUE to skip the device and resume programming the other devices in the range.

DELETE: This softkey deletes a device and all of its dependent resources including

- All line keys (including DLN) that are programmed on a deleted set, standalone BLF, or PKM module
- All line and feature keys that are programmed on associated BLF or PKM modules
- The associated BLF and PKM modules
- Any other line appearances and BLF's of the deleted station or set.

A warning message appears if any of the above conditions exists. To continue, press CONFIRM. To abort, press CANCEL.

RANGE DELETE: This softkey allows block deletion of devices and all dependent resources.

To delete a range of devices and all dependent resources

1. Press RANGE DELETE.

   The system displays: FROM BAY: SLOT: CIRCUIT:.
2. Enter valid Bay, Slot and Circuit numbers for the first device and press the ENTER softkey.

   The system displays: TO BAY: SLOT: CIRCUIT:.
3. Enter valid Bay, Slot and Circuit numbers for the last device and press the ENTER softkey.

   The system prompts you to choose whether or not you want to receive warnings as devices are deleted.
4. To receive a warning prior to deleting each device and its dependent resources, press YES. To delete all devices and dependent resources without warning, press NO.

   The system prompts you once again to confirm the deletion of the range.
5. To proceed with the deletion of the range of devices and all dependent resources, press CONFIRM. Press CANCEL to abort.

   If you chose to receive warnings each time a device is deleted, the system displays "deleting all keys, ..." for each device in the range.
6. To proceed, press CONFIRM. Press CANCEL to abort the deletion of the specific device and continue with the next device.

To abort the deletion of a range of devices at any time during the procedure, press ABORT. Any devices that were deleted prior to pressing the ABORT key will remain deleted.

MOVE: This softkey relocates a device via its bay slot, and circuit numbers. When the MOVE softkey is pressed, the command line requests the FROM location (BAY: SLOT: CCT:). When the location is specified and the ENTER softkey is pressed, the command line requests the TO location. The new location is designated and the ENTER softkey is pressed. Note that entering invalid numbers inhibits cursor movement.

Note: You can move a device from one card to another, but the cards must be of the same type. For example, you can move a DNIC device from one digital line card to another but not from a digital line card to the Onboard ASU card (embedded AMB/AOB).

FIND EXT: This softkey selects a device by its extension number. Pressing the FIND EXT softkey displays the ENTER EXTENSION NUM: prompt on the command line. The selection is completed by entering an extension number of a station or the Prime Line number of a Mitel telephone. The selected device information appears on the command line.

Note: If an SX-200 ICP CX database with ONS PLIDs 1/13/19, 1/13/20, 1/13/21, and 1/13/22 programmed as Door Relays is restored to an SX-200 ICP MX system, the PLIDs will not normally be listed in Form 09. Performing a search using the FIND EXT feature for extensions assigned to those PLIDs will, however, display them in Form 09.

EXPAND SET: Pressing this softkey displays the Expand Set subform. Refer to Expand Set Subform. **Note**: This softkey is valid only when reviewing or programming a Mitel telephone, and appears only when the TYP field on the command line says 4015, 4025, 4150, 410, 420, 430, HOST, SUB, DN, or BLF.

EXPAND PKM: Pressing this softkey displays the Expand PKM (Programmable Key Module) subform. This form is used to associate up to three PKMs with SUPERSET 410, SUPERSET 420, and SUPERSET 430 telephones, and to program the keys on the PKM. This form is also used to associate one PKM 12 or one or two PKM 48 devices to a SUPERSET 4125, SUPERSET 4025, and SUPERSET 4150 telephone. Likewise, it is used to associate 5412 and 5448 PKMs to a Mitel 5220 or 5224 IP telephone and 5410 and 5415 PKMs to a Mitel 5020 IP telephone. Refer to Expand PKM Subform. The EXPAND PKM softkey appears only when you are reviewing or programming a Mitel IP or SUPERSET telephone that can have an associated PKM, PKM 12, or PKM 48.

MORE: Pressing the MORE softkey provides the RANGE, RANGE W/KEYS, PROGRAM LIKE, REVIEW softkeys.

RANGE: This softkey allows block programming for consecutive stations and Mitel telephones. Pressing this softkey displays the prompt: FROM BAY: SLOT: CIRCUIT:. Enter valid Bay, Slot and Circuit numbers for the first device and press the ENTER softkey. The system then prompts TO BAY: SLOT: CIRCUIT:. Enter valid Bay, Slot and Circuit numbers for the last device and press the ENTER softkey. Enter Tenant Group, Extension Number, COS and COR for the first device, if this has not been done already. Press the ENTER softkey. The system automatically assigns incremented extension numbers, the same COS, COR and tenant group numbers to the rest of the devices in the block. When range programming on DNIC circuits, the DEVICE TYPE softkey will appear when the extension number for the first circuit is assigned. The following device types are offered: SS4001, SS4015, SS4025, SS4090, SS4150, SS401, SS410, SS420, SS430, and 3/4DN.

RANGE W/KEYS: This softkey operates the same as the RANGE softkey, but includes key programming (with the exception of the Recall, LDN, and Hold position keys). Pressing this softkey displays the prompt: FROM BAY: SLOT: CIRCUIT:. Enter valid Bay, Slot and Circuit numbers for the first device and press the ENTER softkey. The system then prompts TO BAY: SLOT: CIRCUIT:. Enter valid Bay, Slot and Circuit numbers for the last device and press the ENTER softkey. Enter Tenant Group, Extension Number, COS, COR, line appearances and feature keys for the first device, if this has not been done already. Press the ENTER softkey. The system automatically assigns incremented extension numbers, the same COS, COR and tenant group numbers to the rest of the devices in the block. When range programming on DNIC circuits, the DEVICE TYPE softkey will appear when the extension number for the first circuit is assigned. The following device types are offered: SS4001, SS4015, SS4025, SS4090, SS4150, SS401, SS410, SS420, SS430, and 3/4DN.

Note: After upgrading to Release 4.0 or later, when attempting to program RANGE W/KEYS for sets that were previously programmed using RANGE W'/KEYS, you must first perform a range delete.

PROGRAM LIKE: Use this softkey to program one device like another. All programming is carried forward (with the exception of the Recall, LDN, and Hold position keys). Press the PROGRAM LIKE softkey and then respond to the command line prompts: ENTER THE EXTENSION TO COPY FROM and ENTER THE EXTENSION FOR THE NEW SET. Only unprogrammed circuits can be programmed in this manner.

REVIEW: Presents three more softkeys: SET APP, PAGING GROUP, and CALL FORWARD. SET APP displays a new form - refer to Review (Set App) Subform. This form lists all programmed appearances of the selected extension number on other Mitel telephones. The SET APP softkey appears only when a device has been defined.
PAGING GROUP displays a new form - refer to Review (Paging Group) Subform. This form lists all of the sets in the paging group. The PAGING GROUP softkey appears only if the extension is a member of a paging group.
CALL FORWARD displays a new form - refer to Review (Call Forward) Subform. This form lists all of the Call Forwarding programming.

SHOW CESID: Allows the COMMENTS field to switch to a CESID field in order to show the CESID number (10 digit maximum) for each voice set.

COMMENTS: Allows you to change a SHOW CESID or SHOW MAC field to a COMMENTS field.

SHOW MAC: Displays the MAC ADDRESS field instead of the COMMENTS field.

DEVICE TYPE: Allows the line circuit to be programmed as one of: SUB ATT, DMP, SUPERSET, IP Phone, Symbol Wireless Phone, SpectraLink NetLink Wireless Phone, YA Softphone, or ONS door phone. This softkey appears only when a circuit is not programmed and is NOT an embedded voice mail port (VMAIL). Programming any other fields “freezes” the device type at its current value. After this, only the fields valid for that device type are available. The device type can be changed by deleting all of the programmed entries - the DEVICE TYPE softkey will then reappear. The DEVICE TYPE softkey allows the line circuit to be programmed as one of the following:

DMP: Sets the TYP field to **DMP** for the DNIC MOH/Pager Unit.

SUPERSET: Sets the TYP field to **SET**, and designates the device type as a normal SUPERSET telephone.

SS4150 S/ATT: Sets the TYP field to SUB and designates the device type as an enhanced subattendant.

SS4001: Sets the TYP field to 4001 and designates device type as SUPERSET 4001 telephone.

SS4015: Sets the TYP field to 4015 and designates device type as SUPERSET 4015 telephone.

SS4025: Sets the TYP field to 4025 and designates device type as SUPERSET 4025 telephone.

SS4125: Sets the TYP field to 4125 and designates device type as SUPERSET 4125 telephone.

SS4150: Sets the TYP field to 4150 and designates device type as SUPERSET 4150 telephone.

SS430 S/ATT: Sets the TYP field to 430S and designates device type as SUPERSET 430 telephone.

SS4DN S/ATT: Sets the TYP field to 4DNS and designates device type as SUPERSET 4DN telephone.

SS401: Sets the TYP field to **401** and designates device type as SUPERSET 401+ telephone.

SS410: Sets the TYP field to **410** and designates device type as SUPERSET 410 telephone.

SS420: Sets the TYP field to **420** and designates device type as SUPERSET 420 telephone.

SS430: Sets the TYP field to **430** and designates device type as SUPERSET 430 telephone.

3/4DN: Sets the TYP field to **DN** and designates device type as either a SUPERSET 3DN or a SUPERSET 4DN.

PKM HOST: Sets the TYP field to **HOST** and designates device type as a DSS/BLF Interface Module.

5201: Sets the TYP field to **5201** and designates device type as a Mitel 5201 IP telephone.

5207: Sets the TYP field to **5207** and designates device type as a Mitel 5207 IP telephone.

5010: Sets the TYP field to **5010** and designates device type as a Mitel 5010 IP telephone.

5212d: Sets the TYP field to **5212d** and designates device type as a Mitel Dual Mode 5212 IP telephone.

5215: Sets the TYP field to **5215** and designates device type as a Mitel 5215 IP telephone.

5215d: Sets the TYP field to **5215d** and designates device type as a Mitel Dual Mode 5215 IP telephone.

5020: Sets the TYP field to **5020** and designates device type as a Mitel 5020 IP telephone.

5220: Sets the TYP field to **5220** and designates device type as a Mitel 5220 IP telephone.

5220d: Sets the TYP field to **5220d** and designates device type as a Mitel Dual Mode 5220 IP telephone.

SS5220 S/ATT: Sets the TYP field to **5220S** and designates the device type as an enhanced subattendant.

5224d: Sets the TYP field to **5224d** and designates device type as a Mitel Dual Mode 5224 IP telephone.

SS5224 S/ATT: Sets the TYP field to **5224S** and designates the device type as an enhanced subattendant.

5304: Sets the TYP field to **5304** and designates device type as a Mitel Dual Mode 5304 IP telephone.

5312: Sets the TYP field to **5312** and designates device type as a Mitel Dual Mode 5312 IP telephone.

5324: Sets the TYP field to **5324** and designates device type as a Mitel Dual Mode 5324 IP telephone.

5324 S/ATT: Sets the TYP field to **5324S** and designates the device type as an enhanced subattendant.

5330: Sets the TYP field to **5330** and designates device type as a Mitel 5330 IP telephone.

5340: Sets the TYP field to **5340** and designates device type as a Mitel 5340 IP telephone.

5330 S/ATT and **5340 S/ATT**: sets type field and designates type as an enhanced subattendant.

WRLSS: Sets the TYP field to **WRLSS** and designates device type as a Symbol MiNET Wireless Phone.

SPECTRALINK: Sets the TYP field to **Spectr** and designates device type as a SpectraLink NetLink Wireless Phone.

TMX 3000: Sets the TYP field to **TMX3K** and designates device type as a TeleMatrix 3000IP Phone.

DOOR: Sets the TYP field to **DOOR** and designates device type as a door opener extension used to unlock a door.

Note: To program the circuit as a YA Softphone, select 5020 IP Phone, 5220 IP Phone, or 5224 IP Phone as the DEVICE TYPE.

SHOW PIN: This softkey allows you to assign a four-digit Personal Identification Number (PIN) to a Symbol MiNET Wireless phone, or to a physical or phantom device that is programmed to support the Secure Hot Swap feature. To log in to a Symbol MiNET Wireless phone, the user enter the DN followed by the PIN and then presses ENTER. In response, the system generates a "virtual" MAC address for the phone. To initiate a Secure Hot Swap, the user enters the DN and PIN of the "destination" device on the "originating" device. In response, the system swaps programming between the two devices. Devices that have been hot swapped display an asterisk (*) in the PIN field.

UNSWAP: This softkey allows you to unswap programming for two devices that are currently hot swapped.

The standard softkeys **BAY/SLT/CCT**, **CANCEL**, **ENTER**, ****MORE**** and **QUIT** are also provided.

If you attempt to MOVE a DMP from one plid to another you will receive the following error message: "Cannot move DMP". You must delete the DMP first and then reprogram it at another plid. In addition, if you try to delete a device that is programmed in form 18 (for example as Music On Hold), you will get an error message. You must delete the DMP first from Form 18, and then from Form 9.

The same plid cannot be programmed as a PKM host and a dataset.

If you install a SUPERSET telephone on a DNIC circuit that is not the same device type, in many cases, the programmed device type for the circuit will change automatically to the device type of the installed set. For example, if a circuit is programmed with a 410 device type and you install a SUPERSET 420 telephone on that circuit, the system will automatically change the programmed device type from 410 to 420. The Programmed SUPERSET vs Installed Device Type table below shows which SUPERSET telephones will operate when they are installed on the programmable device types.

Programmed SUPERSET vs Installed Device Type Table

Table: Programmed SUPERSET vs Installed Device Type

SUPERSET	Installed Device Type								
401	410	420	430	3DN/4DN	4001	4015	4025/4125	4150	
401+	Yes	No	No	No	No	Yes	No	No	No
410	No	Yes	Yes[1]	Yes[1]	Yes[1]	No	Yes[2]	Yes[2]	Yes[2]
420	No	Yes[1]	Yes	Yes	Yes[2]	No	Yes[3]	Yes[2]	Yes[2]
430	No	Yes[1]	Yes	Yes	Yes[2]	No	Yes[3]	Yes[2]	Yes[2]
3DN/4DN	No	Yes[1]	Yes	Yes	Yes	No	Yes[3]	Yes[2]	Yes[2]

SUB ATT (4DN)	No	No	No	Yes[2]	Yes	No	No	No	Yes[2]
SUB ATT (430)	No	No	No	Yes	Yes	No	No	No	Yes[2]
4001	Yes	No	No	No	No	Yes	No	No	No
4015	No	Yes1	Yes	Yes	Yes	No	Yes	Yes	Yes
4025/4125	No	Yes[3]	Yes[3]	Yes[3]	Yes[3]	No	Yes[1]	Yes	Yes
4150	No	Yes[3]	Yes[3]	Yes[3]	Yes[3]	No	Yes[1]	Yes	Yes
4150 S/ATT	No	No	No	Yes[3]	Yes[3]	No	No	No	Yes

Notes:
1. Yes - Set will come up unconditionally.
2. Yes[1] - Set will come up if additional keys on the programmed set are not line or feature keys,
3. Yes[2] - Set will come up only if there are no programmed PKMs
4. Yes[3] - Both conditions Yes1 and Yes2 must be satisfied.
5. No - Installed set will not operate.

Subform 09 - Expand Set and Expand PKM

Expand Set Subform

This form appears when you press the EXPAND SET softkey in the Desktop Device Assignments form. This form programs SUPERSET telephone and IP Phone line appearances and feature keys. This form is not available to SUPERSET 4001 or SUPERSET 401+ telephones.

Figure: Form 09 Expand Set Subform Layout

Expand PKM Subform

This form appears when you press the EXPAND PKM softkey in the Desktop Device Assignments form. You can use this form to program PKM line appearances and feature keys. An * indicates a PKM is associated with a telephone. Refer to this Figure for the form layout. This form is available to all Mitel phones except the SUPERSET 4001, and 401 and the 5201 IP and the Symbol MiNET Wireless telephones. This form is also available to a PKM HOST that is associated with an attendant console.

With SUPERSET 400 series sets, a PKM module is used which has 30 programmable keys. A maximum of three PKM modules may be programmed with a 400-series set.

With SUPERSET 4000 series sets, a PKM 12 or up to two PKM 48 devices are used. A PKM 12 has 12 programmable keys. A PKM 48 has 48 programmable keys. A maximum of two PKM 48 devices may be programmed with a 4000-series set.

With Mitel 5020 IP telephones, a 5410 PKM or up to two 5415 PKMs are used. The 5410 has 12 programmable keys and the 5415 has 48.

With Mitel 5220 IP and 5224 IP telephones, a 5412 and up to two 5448 PKMs are used. The 5412 has 12 programmable keys and the 5448 has 48.

The Form 09 subform selects the correct number of keys and modules according to the set that the PKM or PKM 48 is programmed against.

PKM 48 number 1 programming precedes the key number with a 1.

PKM 48 number 2 programming precedes the key number with a 2.

Figure: Form 09 Expand PKM Subform Layout

Field Description

KEY: This field lists the Mitel telephone Line Select key numbers and cannot be modified. If the keys are on SUPERSET DSS modules or Programmable Key Modules (PKMs), the key numbers are preceded by a 1, 2, or 3. The 1, 2, or 3 to indicate the PKM address. If the keys are on PKM 48 devices, the key numbers are preceded by a 1 or 2 to indicate the PKM address. The key numbers on a PKM correspond to those shown in this Figure. The key numbers on a PKM 48 correspond to those in this Figure.

The SUPERSET 410 has 6 programmable keys; SUPERSET 420, SUPERSET 430, and SUBATT each have 12 programmable keys. The Mitel 5010 IP, 5215 IP, SUPERSET 4015 sets have 7 programmable keys. The Mitel 5212 and TeleMatrix 3000IP sets have 11 programmable keys. The Mitel 5207 IP, 5020 IP, 5220 IP, SUPERSET 4025, SUPERSET 4125, SUPERSET 4150, and SpectraLink sets have 13 programmable keys. The Mitel 5224 IP set has 23 programmable keys. The key numbers on SUPERSET 410, SUPERSET 420, SUPERSET 430, SUBATT, SUPERSET4015, SUPERSET 4025, SUPERSET 4125, SUPERSET 4150, or SUBATT telephones correspond to those in this **Figure**.

TYPE: This field lists the key function. If it is a Speed Dial key, the default assignment, the words “Speed Dial” are shown. If it is a line appearance, the line type is shown. If it is a feature key, the feature name is shown. If the line is a BLF Appearance, the key type is “Busy Lamp." If the line is a Mailbox appearance, the key type is "Mailbox."

DIR: If the key is a line appearance, the directional variant of the line (In/Out or Incoming) is shown.

RING: If the key is a line appearance, the ringing variant of the line (Immed, Delay, or None) is shown.

SEC: If the key is a line appearance, this field indicates (Yes or No) whether the secretarial variant is enabled. For a Busy Lamp key, setting this field to YES causes an immediate release when the DSS key is pressed.

DSS: NO in this field indicates that the DSS key cannot call the selected station. CALL in this field allows the DSS key to call the selected station.

EXT NUM: This field contains the extension number of the line. This applies to the Prime Line, Key lines, Multiple Call lines, Personal Outgoing lines, Mailbox lines, or BLF/DSS lines. When this field is filled for a specific key, no entry is allowed in the corresponding TRUNK NUMBER field. The EXT NUM field is blank if the key directly selects a trunk.

TRK NUM: If the key is assigned as DTS or private trunk, this field contains the trunk number. Trunk numbers are defined in Form 14, Non-Dial-In Trunks and Form 15, Dial-In Trunks. Note that when this field is filled for a specific key, no entry is allowed in the corresponding EXT NUM field.

LABEL: Only appears if the softkey type is LDN, and the device type is a subattendant. The only exception is, if the softkey type is Recall, the LABEL field is automatically RECALL. For all other LDN softkeys, any character string may be entered up to 12 characters in length.

> **Note**: Each subattendant can have its own LDN key labels independent of other subattendants, but they must be programmed by the Administrator for each extension. If no label is programmed, the key will display the extension number.
>
> **Note**: Label display on 5330 and 5340 IP Phones may be truncated to fit in allowable space.

R#: Applies to Key Line (but not multi-call) keys, DTS keys and CO line keys. Enter a number from 1-16 to assign the line a ringing tone. For more information about tones, see Distinctive Ringing Tones.

Ringing Number	Tones (Hz)	Pitch (Hz)
1	400 + 348	5
2	400 + 348	10

Ringing Number	Tones (Hz)	Pitch (Hz)
3	604 + 385	5
4	604 + 385	10
5	620 + 426	5
6	620 + 426	10
7	668 + 470	5
8	668 + 470	10
9	1040 + 800	5
10	1040 + 800	10
11	1430 + 1100	5
12	1430 + 1100	10
13	1560 + 1200	5
14	1560 + 1200	10
15	1690 + 1300	5
16	1690 + 1300	10

Softkeys

BLF/DSS: Programs the selected key as a busy lamp field appearance (for the associated extension number). When a PKM 48 is associated with a telephone, this and MAILBOX are the only key types supported.

Mailbox: Programs the selected key as a Mailbox appearance (for the associated extension number). When a PKM 48 is associated with a telephone, this and BLF/DSS are the only key types supported.

BOTH WAY: Pressing this softkey enables the selected Mitel telephone key (line appearance) to originate and receive calls. The DIR / DIRECTION field displays the In/Out indication.

CO LINE: Assigns the selected Mitel telephone key as a CO Line appearance. This gives the telephone direct access to a trunk. Only non-dial-in trunks can be specified. Up to 64 telephones (64 IP sets, or 32 IP and 32 DNIC sets) can have an appearance of the same trunk. The SEC, DSS, EXT NUM and LABEL fields are not accessible.

Note: Line appearances programmed to key positions three and above on SpectraLink NetLink telephones fail to display unless the trunk name assigned in Form 14 is "Line n", where "n" is the trunk number.

CO LINE GRP: Assigns the selected key as a CO line group key. This gives the telephone direct access to a group of trunks through ARS. It is the responsibility of the installer to program access limitations via CORs (Form 20 and Form 26). The TRK ACC field must match an ARS leading digit string from Form 26.

DELAY RING: Pressing this softkey causes incoming calls to flash the selected key (line appearance) for a programmable period of time and then ring the Mitel telephone for incoming calls. The "Delay" indication appears in the RING field. The duration of the ring delay is controlled by COS Option 263, Delay Ring Timer.

DELETE PKM: This softkey appears when a PKM with unassigned keys has been selected (i.e., all keys are speed call keys). Pressing this softkey deletes the PKM, and exits the PKM subform.

DIR TRK ACC: Pressing this softkey programs the selected Mitel telephone key as a Direct Trunk Access line. “DTS” appears in the TYPE field. Then use the TAB or → key to move the cursor to the DIRECTION field.

DSS CALL: Appears only in the DSS field. Enables the DSS key associated with a BLF appearance to call the selected extension. The CALL indication appears in the DSS field.

DSS PAGE: Appears only in the DSS field. Enables the DSS key associated with a BLF appearance to direct page a selected station. The PAGE indication appears in the DSS field.

FEATURE: Pressing this softkey assigns the selected line appearance key on Mitel telephones as a feature access key. This softkey appears only if the set is a Mitel display telephone. The following softkeys appear when the cursor is at the TYPE field and the FEATURE softkey is pressed:

- ACCOUNT CODE
- ALARM
- AUTO ANSWER
- FWD ALWAYS
- FWD BUSY
- FWD NO ANS
- FWD BUSY/NA
- FWD ALL
- CALL PARK/RETRIEVE
- CALL PICKUP
- CALL/ATTN (Data Call Connect)
- CALLBACK
- CALL BLOCK
- CALLERS
- CAMPON
- DATA DISC (Data Call Disconnect)
- DAY/NIGHT
- DIRECT PAGE
- DO NOT DIST (Do Not Disturb)
- DOUBLE FLASH
- FORWARD CALL
- GROUP LISTEN
- GUEST ROOM
- HANDSET MUTE
- HEADSET MODE
- MUSIC
- NIGHT ANSWER
- OPEN DOOR
- ORBIT
- OVERRIDE
- PA PAGING
- PARK&PG SETS
- PARK&PG GRP
- PARK & PAGE
- PARK & PA PAGE

- PHONEBOOK
- PRIVACY REL (Privacy Release)
- RECORD CALL
- RESPOND
- RELEASE
- SINGLE FLASH
- SWAP (Trade Calls)
- SYSTEM PARK
- VOICE MAIL
- VM PROMPTS
- WKUP ALARM

Note: In additional to the features mentioned above, the 5304 IP Phone has six additional features: Superkey, Hold, Cancel, Transfer/Conference, Message and Redial.

Note: Refer to the Program Features section, for a description of the feature keys available for Mitel telephones.

HOLD POS: Programs the selected key as an enhanced subattendant hold slot. Hold Slot is displayed in the TYPE field. No other fields may be programmed. An enhanced subattendant may have up to 6 hold slots.

IMMED RING: Pressing this softkey programs the selected Mitel telephone key (line appearance) to ring the Mitel telephone immediately for incoming calls. The form displays Immed in the RING field to indicate this condition.

IN ONLY: Pressing this softkey restricts the selected Mitel telephone key (line appearance) to receiving incoming calls only. No call originations are permitted. The DIRECTION field displays “Incoming”.

KEY: This softkey selects a Mitel telephone key by number. Pressing the KEY softkey displays the ENTER KEY NUM: prompt on the command line. Any Mitel telephone key may be selected except for Key 01 (Prime Line).

KEY LINE: Pressing this softkey assigns the selected Mitel telephone key as a Key Line appearance. “Key” appears in the TYPE field. Then use the TAB or → key to move the cursor to the DIRECTION field.

LDN: Programs the selected key as an enhanced subattendant LDN key. LDN is displayed in the TYPE field. At this point, a ring type can be selected, an extension number entered, and a label entered. The label is displayed on the SUPERSET 4DN and the SUPERSET 430, and on Mitel 5020, 5220, and 5224 IP Phones programmed as enhanced subattendants. An enhanced subattendant may have up to 6 LDN positions

LINE PREF: Allows selection of the preferred line. This can be one of: PRIME key, CO LINE key, PRIVATE TRUNK key, CO LINE GRP key, PERSONAL O/G key or MANUAL (must press a line key) - no other line types are permitted to be the preferred origination line. The line programmed as the preferred line will have an asterisk (*) beside its number in the KEY field.

The selected preferred line will take priority over COS option 604 (Telephone - Automatic Outgoing Line); the system will use the preferred line if selected.

If the customer wants the system to automatically select a line (COS 604 set to auto), or if the customer wants to always manually select a line (COS 604 set to manual), do not select a preferred line.

****MORE**:** Pressing this softkey displays the next group of softkeys available. You may continue to press the MORE softkey to view all softkeys, and return to the first groups presented.

MULTI-CALL: Pressing this softkey assigns the selected Mitel telephone key as a Multiple Call Line appearance. “Multiple” appears in the TYPE field. Then use the TAB or → key to move the cursor to the DIRECTION field.

NON DSS: Appears only in the DSS field. Disables the DSS key associated with a BLF appearance. The “No” indication appears in the DSS field.

NON SECR: Pressing this softkey disables the secretarial function for the selected Mitel telephone key (line appearance). The “No” indication appears in the SECRETARIAL field.

PERSONAL O/G: Pressing the PERSONAL O/G softkey designates the selected Mitel telephone key as a personal outgoing line. No other fields can be edited. Press the ENTER softkey. “Personal” appears in the TYPE field. “Outgoing” appears in the DIRECTION field. “None” appears in the RING field.

PRIVATE TRK: Pressing PRIVATE TRK designates the selected Mitel telephone key as a private line.

RECALL: Programs the selected key as an enhanced subattendant Recall key. Recall is displayed in the TYPE field. The only other field that may be programmed is the RING field. An enhanced subattendant may have only one Recall Key.

REVIEW: Presents the set application review subform (refer to Review Subform - SET APP).

SAVE PKM: This softkey is presented the first time a PKM is created. It allows the user to save the PKM without entering any key assignments, thereby designating all keys as speed call keys. Pressing this softkey saves the PKM and exits the PKM subform.

SECRETARIAL: Pressing this softkey enables the secretarial function for the selected Mitel telephone key (line appearance). The ‘Yes’ indication appears in the SECRETARIAL field. When a Line Select key is set as a secretarial key, then the user can override the DO NOT DISTURB feature on the SUPERSET telephone corresponding to that line appearance. For a DSS key, this enables the secretarial option.

The standard softkeys **CANCEL**, **DELETE**, **ENTER**, **** MORE **** and **QUIT** are also provided.

Figure: PKM Key Numbers

Figure: SUPERSET Telephone Key Numbers

Figure: Mitel 5010 and 5215 IP Telephone Key Numbers

Figure: Mitel 5212 IP Telephone Key Numbers

Figure: Mitel 5020, 5207 and 5220 IP Telephone Key Numbers

Figure: Mitel 5224 IP Telephone Key Numbers

Figure: 5448 PKM Key Numbers

Figure: 5412 PKM Key Numbers

Figure: PKM 48 Key Numbers

Figure: PKM 12 Key Numbers

Subform 09 - Review (SET APP)

This form appears when the REVIEW and then the SET APP softkey are pressed in the Desktop Device Assignment form, or when the REVIEW softkey is pressed in the Expand Set Subform for Form 09. This form lists all programmed appearances of the selected extension number on other Mitel telephones. When entered from the Desktop Device Assignment form, this form displays a list of all programmed appearances of the selected extension number. When entered from the Expand Set Subform or Expand PKM Subform, the form displays all appearances of the selected key’s extension number, trunk number, or subattendant LDN number. Refer to the figure below for the form layout.

Figure: Review (Set App) Subform Layout

Field Description

The header line displays the selected line appearance access number. For telephones, the type (set, station, subattendant) along with the extension number is shown. For trunks, the type (trunk number, CO line, CO line group) is shown along with the trunk number, or leading ARS digits. Extension numbers are also shown for LDNs and logical lines.

BAY, SLT and CCT: These fields list the bay, slot and circuit numbers of each extension or Mitel telephone that has an appearance of the selected line. These fields cannot be modified. The system generates them based on the PROGRAMMED field of Form 01, System Configuration.

SET TYPE: This field displays the listed device type; STATION indicates an industry standard telephone, SET indicates a Mitel telephone and SUB-ATT indicates a subattendant.

KEY: For listed Mitel telephones, the KEY field displays the key number where the line appears.

EXT NUM: This field displays the extension number assigned to a particular bay/slot/circuit.

Softkeys

TRUNK NUMBER: This softkey selects a trunk and shows all sets which have appearances of it. Pressing this softkey displays ENTER TRUNK NUM: prompt on the command line. The trunk selection is completed by entering a valid trunk number (1 - 200), and pressing the ENTER softkey.

EXT NUM: This softkey selects an extension number and shows all sets which have appearances of it. Pressing this softkey displays the ENTER EXTENSION NUM: prompt. The selection is completed by entering a valid Station Number or Mitel telephone Number and then pressing the ENTER softkey.

LDN NUM: This softkey selects an LDN number and shows all sets which have appearances of it. Pressing this softkey displays the ENTER LDN NUM: prompt. The selection is completed by entering a valid LDN number and then pressing the ENTER softkey.

MAILBOX: Applies only to key system telephones. Selects an extension and shows all Mailbox keys programmed for it. Pressing this softkey displays the ENTER MAILBOX EXTN NUM: prompt. The selection is completed by entering an extension number and then pressing the ENTER softkey.

CO LINE GRP: Applies only to key system telephones. Selects a CO line group access code and shows all sets which have appearances of it. Pressing this softkey displays the ENTER ARS LEADING DIGIT STRING: prompt. The selection is completed by entering a valid access code and then pressing the ENTER softkey.

The standard softkeys CANCEL, ENTER, and QUIT are also provided.

Subform 09 - Review (PAGING GROUP)

This form appears when the REVIEW and PAGING GRP softkeys are pressed in the Desktop Device Assignments form. This review form lists all of the sets in the paging group.

Field Descriptions

The header line displays the page group, and the page group name (if programmed).

EXTN: This field displays the extension number of the page group members.

BAY, SLT and CCT: These fields list the bay, slot and circuit numbers of the page group members.

COMMENTS: This field displays the COMMENTS field from the corresponding lines in Form 09.

Figure: Review (Paging Group) Subform Layout

Softkeys

NAME: Specifies a character name for the selected paging group. The “ENTER PAGING GROUP NAME:” prompt is displayed. Selection is completed by entering a character string (maximum eight characters), followed by the ENTER softkey. The page group name is then displayed in the header line.

PAGING GROUP: Selects a paging group. The “ENTER PAGING GROUP:” prompt is displayed. Selection is completed by entering a valid page group number (1 - 50), followed by the ENTER softkey.

EXT NUM: This softkey selects a device by its extension number. Pressing this softkey displays the “ENTER EXTENSION NUM:” prompt. The selection is completed by entering a valid Key System Mitel telephone number and then pressing the ENTER softkey.

The standard softkeys CANCEL, ENTER, and QUIT are also provided.

Subform 9 - Review (CALL FORWARD)

This form appears when the REVIEW and CALL FORWARD softkeys are pressed in the Desktop Device Assignments form. This review form lists all Call Forwarding programming.

Field Descriptions

The header line displays the page group, and the page group name (if programmed).

SET: This field displays the Call Forwarding settings for the selected extension.

DESTINATION: These fields list the call forwarding destinations that are available for the selected device. Destinations that are stored as speed dials or buttons are expanded. Speed Call destinations appear with designation KEY; System Abbreviated dial destinations appear with designation SSC; Personal Speed Call destinations appear with designation PSC; Private numbers appear as PRIVATE NUMBER.

Note: Destinations can only be valid internal numbers. External numbers can not be programmed in this form. To use external destinations, program the devices to be forwarded with system abbreviated dial numbers and change the destination using Form 31.

STATUS: This field lists the option selected for each parameter.

Figure: Review (Call Forward) Subform Layout

Softkeys

ENABLE/DISABLE: Used to enable or disable the Call Forwarding setting. The current status of the settings for the selected extension are displayed.

SPEED DIAL: Used to enter a speed dial number to use as a forwarding destination. Destinations can be any valid internal number. **Note**: If you want to change a Call Forward profile via CDE to use external destinations, you must program the devices to be forwarded to system Abbreviated Dial numbers and change the destination in CDE Form 31.

The standard softkeys **DELETE** and **QUIT** are also provided.

Form 10 - Pickup Groups

This form specifies the members of each pickup group. Memberships are specified by the extension number of an industry standard telephone or the prime line number of a Mitel telephone. Attendant consoles are not allowed. The system supports a maximum of 50 pickup groups; each group supports a maximum of 50 members.

Figure: Form 10 Layout

Field Descriptions

EXT NUM: This field displays the pickup group member extension number.

BAY/SLT/CCT and **COMMENTS**: These fields cannot be modified. The form displays the BAY/SLT/CCT and COMMENTS fields from the corresponding lines of Form 09, Desktop Device Assignments.

Softkeys

INSERT: This softkey adds a new member to the pickup group on a new line just above the current line pointer. Pressing the INSERT softkey clears the command line and moves the cursor to the EXT NUM field. Enter a valid extension number and press the ENTER softkey.

Note: This softkey only appears if there is data present in this form.

PICKUP GROUP: This softkey selects the pickup group to be displayed. Pressing the PICKUP GROUP softkey displays the ENTER PICKUP GROUP NUM: prompt on the command line. Enter the pickup group number and press the ENTER softkey.

EXT NUM: This softkey selects a Pickup Group member by its extension number (or Prime Line number). Pressing the EXT NUM softkey displays the ENTER EXTENSION NUM: prompt on the command line. Entering the extension number displays that member with its bay, slot and circuit location, and (if any) comments. Note that if the selected extension number is not in the current Pickup Group, then the system automatically displays the Pickup Group where the selected device is located.

NAME: This softkey provides a prompt for you to enter a Pickup Group name. The entered name for the Pickup Group will then appear in the header of the form.

The standard softkeys **CANCEL**, **DELETE**, **ENTER**, and **QUIT** are also provided.

Form 11 - Data Circuit Descriptor

A data circuit descriptor specifies the parameters the data processing software and attached DTE (Data Terminal Equipment) require. This form provides 25 programmable descriptors. The main form displays the descriptor numbers and the number of circuits associated with each descriptor. The system generates the data in this form based on the entries in Form 12, Data Assignment. The user can modify only the COMMENTS field. A softkey provides access to the individual parameters of each descriptor via a sub-form.

Figure: Form 11 Layout

Field Descriptions

DESCRIPTOR: This field lists the circuit descriptors, numbered 01 to 25.

NUMBER OF DATA CIRCUITS ASSIGNED: This field records the number of devices assigned to each descriptor.

COMMENTS: This field is reserved for additional data (a maximum of 20 characters). It is stored by the system but not used.

Softkeys

DESCRIPTOR NUMBER: The DESC NUM softkey allows the user to select a DESC NUM by number. Pressing this softkey displays the ENTER DESC NUM: prompt. Entering a valid descriptor number and pressing the ENTER softkey, completes the selection.

SELECT OPTION: Pressing this softkey displays a new form. This form provides the options associated with the data circuit that is assigned to a descriptor number. Refer to the Data Circuit Descriptor Options table.

REVIEW: This softkey displays the Review List Subform, a read-only form containing the BAY, SLOT, CIRCUIT and SUBCIRCUIT location of all devices assigned that descriptor, and the comments which were entered in Form 12, Data Assignment. The REVIEW softkey appears only if at least one circuit has been assigned the descriptor displayed on the command line.

The standard softkey **QUIT** is also provided.

Subform 11 - Data Circuit Descriptor Options

This form appears when the SEL. OPTION softkey is pressed (see this Figure). It lists the programmable parameters of the descriptor. See the Data Circuit Descriptor Options table for the complete list of options (they are described below, under Parameters).

Figure: Select Options Subform Layout

Field Descriptions

The header line displays the descriptor number.

PARAMETER NAME: This field lists the parameters. For numerical parameters, it lists the valid range of values.

VALUE: This field lists the option or numeric value selected for each parameter.

Softkeys

The standard softkeys **CANCEL**, **DELETE**, **ENTER**, and **QUIT** are also provided. The other softkeys depend on the parameter displayed on the command line (see Parameters for details). Generally, for numeric parameters, only the QUIT softkey appears. When a value is typed in, the ENTER softkey appears. Press ENTER once to terminate the entry, then again to commit it to the database. While the ENTER softkey is present and has not been pressed, the QUIT softkey cancels the new entry and restores the previous value. For parameters with YES and NO options, softkey 1 appears, marked with the option opposite to the current setting.

Data Circuit Descriptor Options

Table: Data Circuit Descriptor Options

Session Inactivity Disconnect Timer (0-255 minutes)
Guard Timer (0-99 seconds)
Minimum Baud Rate
Default Baud Rate
Maximum Baud Rate
Always Use Default Baud Rate When Called
DTR Off Disconnect Timer
DTR to CTS Delay Timer
DTR Forced High
RTS Forced High
DSR Is Held High When Device is Idle
CTS Is Held High When Device is Idle
Originate a DTRX Call With A Low->High Transition of DTR
Action Taken If The Idle DTE Has DTR Low (Auto Answer)
Pooled Modem Communication Established Indicator
First Modem Tone
Second Modem Tone
ASYNC: Keyboard Origination Allowed
ASYNC: ADL Auto Baud
ASYNC: Flow Control
ASYNC: XON Character
ASYNC: XOFF Character
ASYNC: Break Key Function
ASYNC: PBX Attention Character
ASYNC: Parity
ASYNC: Character Length
ASYNC: Number of Stop Bits
ASYNC: Autobaud To Host Character 1
ASYNC: Autobaud To Host Character 2
ASYNC: Delay Between Autobaud Characters
DS2100: Operating Mode
SYNC: Rate Adaption Scheme
SYNC: Clock Source

Data Communication Abbreviations

Table: Data Communication Abbreviations

Abbreviation	Term
CTS	Clear To Send
DCD	Data Carrier Detect
DCE	Data Communication Equipment
DSR	Data Set Ready
DTE	Data Terminal Equipment
DTR	Data Terminal Ready
DTRX	Data Transceiver
RI	Ring Indicator
ADL	Associated Data Line

Parameters

The Data Circuit Descriptor is used by ADL calls, DTRX calls, printer monitors, the PMS port and pooled modems. The parameters are ordered so that timers and baud rate options appear at the top of the form. These parameters apply to all data device types. Next are the parameters dealing with EIA leads. These also apply to all data devices but the parameter's meaning can depend on whether a modem adapter is in the RS-232 connection. The last options are device type dependent parameters, usually indicated by a prefix.

The name of each descriptor option is followed by a label in brackets ( ), defining when the parameter applies and by its default setting in brackets also (default =).

(SYSTEM) defines a parameter that is used by the system to configure the operating mode of the dataset and to define activity timers for the dataset.

(DATASET) defines parameters which apply at the dataset when powered up and remain constant when idle or active in a data call.

(DTE TO DATASET) defines a parameter which applies only to the dataset when the call flow is FROM the attached data device TO the dataset, from devices that originate data calls such as terminals

(DATASET TO DTE) defines a parameter which applies only to the dataset when the call flow is FROM the dataset TO attached data devices such as printers, which are called other devices.

(DTE TO DATASET/DATASET TO DTE) defines a parameter which applies to datasets connected to data devices that make calls, that receive calls, or that make and receive calls (PC or programmed Host computer).

(FLOW CONTROL CHARACTER) defines parameters that apply when the “XON/XOFF” option is selected for the flow control parameter.

Session Inactivity Disconnect Timer:

Options: 0-255 minutes, 1 min increments

Default: 0, option is disabled

The DATASET uses this timer to monitor when activity has ceased on the attached device (from the last key stroke). After the timer has expires, the data call is terminated and the circuits involved are returned to idle. This timer frees up resources that are not in use.

Setting the timer to “0” DISABLES the function so that the data session is NOT disconnected due to inactivity.

Guard Timer:

Options: 0-99 seconds, 1 sec increments

Default: 2

This timer specifies the amount of time the data circuit is made unavailable (make-busy state) after a session disconnect. This parameter applies to the receive end of the data call. This parameter should provide sufficient time for the far end to perform the clear-down sequence before a new call is allowed for the data device.

Minimum Baud Rate: (DTE to DATASET/DATASET to DTE)

Options: 110, 150, 300, 600, 1200, 2400, 4800, 9600 or 19.2K

Default: 110

This parameter defines the minimum baud rate at which the DATASET is expected to operate. This parameter should be set within the limits of the data device connected. The system automatically adjusts the baud rate of the parties in a data call if the Flow Control parameter limitation are not met (see Flow Control). The system will try to match the baud rate of the destination to that of the call originator. The system checks the MIN and MAX BAUD RATE settings of each DATASET to find the highest suitable match.

If the common baud rate between the source and the destination turns out not to be the initial baud rate the call was started with, the source DATASET is automatically reprogrammed to the compatible speed and a “change speed” message is sent to the user. The source device will then need to be manually set to the proper baud rate.

Default Baud Rate: (DTE to DATASET/DATASET to DTE)

Options: 110, 150, 300, 600, 1200, 2400, 4800, 9600 or 19.2K

Default: 9600

This parameter defines the initial baud rate that the DATASET is to operate with for non auto-baud data circuits. This parameter should typically be set the same as the MAX BAUD RATE parameter to benefit from the faster transmission speed.

Maximum Baud Rate: (DATASET)

Options: 110, 150, 300, 600, 1200, 2400, 4800, 9600 or 19.2K

Default: 19.2

This parameter defines the maximum baud rate the DATASET is expected to operate. It should be set within the limits of the data device connected. As described in the MIN BAUD RATE parameter, the system automatically adjusts the baud rate of the parties in a data call if the Flow Control parameter limitation are not met (see Flow Control). The system will try to match the baud rate of the destination to that of the call originator. The system checks the MIN and MAX BAUD RATE settings of each DATASET to find the highest suitable match.

Always Use Default Baud Rate When Called: (DATASET TO DTE)

Options: Yes/No

Default: No

This parameter defines the baud rate to be used when the DATASET is called by other data devices. By default, the Baud rate selected is determined by the caller's rate and the flow control parameter. By setting this parameter to “YES”, the assigned default baud rate is used by the DATASET to transmit characters to the attached data device. The calling device will be required to either have “Flow Control” or be set at the same baud rate as the default.

DTR Off Disconnect Timer: (SYSTEM)

Options: 0-99 seconds, 1 sec inc

Default: 5

This timer specifies the length of time a “low” level has to be present on DTR (RS 232C pin 20   Data Terminal Ready) before it is recognized by the DATASET as an end-of-session (disconnect) signal. Proper setting of the timer will prevent premature session disconnects if DTR is temporarily forced low when performing certain localized functions. Functions such as screen refresh, terminal reset, clear screen and others, which are not requests for session disconnect, may cause DTR “low” for a brief period.

Note: If the attached device is a pooled modem, the timer has no effect; when the pooled modem drops DSR, the data call is dropped immediately.

DTR To CTS Delay Timer: (DATASET TO DTE)

Options: 0-9900 msec, 100 msec inc

Default: 100

This timer specifies the amount time the DATASET waits before it makes CTS “high” (RS 232C pin 5 Clear To Send) after it has detected the device setting DTR “high” (RS 232C pin 20 Data Terminal Ready). This is required if the data device is the type which monitors for the transition of CTS from “low” to “high” before it allows data to be transmitted to the DATASET.

Setting the value to “0 msec” means that CTS will be asserted high directly following DTR “high”.

DTR Forced High: (DATASET)

Options: Yes/No

Default: No

Some Data Terminal Equipment (DTE) such as terminals and printers, and Data Circuit Terminating Equipment (DCE) such as modems, do not offer connections to all of the DATASET's RS 232C pins. When enabled, this parameter allows the DATASET to assume the lead is high, independent of whether the device is connected, or not.

YES: ENABLES the parameter. It causes the DATASET connected to a DTE device to assume that the DTR pin of the RS 232C connector is “high”. It causes the DATASET connected to a DCE device through a modem adapter to assume that the DSR pin of the RS 232C connector is “high”, then this parameter is translated into “DSR Forced High”.

NO: DISABLES the parameter. It causes the DATASET to monitor the data device's RS 232C pins (DTE or DCE) as defined.

RTS Forced High: (DATASET)

Options: Yes/No

Default: No

This parameter is similar to the “DTR Forced High” parameter above. When enabled, it allows the DATASET to assume the lead is high independent of whether the device is connected, or not.

YES: ENABLES the parameter. It causes the DATASET connected to a DTE device to assume that the RST pin of the RS 232C connector is “high”. It causes the DATASET connected to a DCE device through a modem adapter to assume that the DCD pin of the RS 232C connector is “high”, then this parameter is translated into “DCD Forced High”.

NO: DISABLES the parameter. It causes the DATASET to monitor the data device's RS 232C pins (DTE or DCE) as defined.

DSR Is Held High When Device Is Idle: (DATASET TO DTE)

Options: Yes/No

Default: Yes

This parameter specifies how the system will hold the state of DSR on the DATASET to the attached idle Data Device.

Note: This parameter MUST be enabled to provide “Keyboard Origination” (Auto Baud Detect).

YES: ENABLES the parameter and is the default value. It indicates that DSR pin of the dataset will be remain “high” even when the data device is idle. This is the normal setting for this parameter.

NO: DISABLES the parameter. When disabled, The DSR pin Of the dataset will be held “low” until the data call is connected. This means that the DTE cannot be used to define the destination of the call as the DATASET will be in a “not ready” state preventing the DTE to send data to the DTRX. For the data call to be connected (end to end), the destination must be dialed using the ADL or HOTLINE features. Once connected, the DATASET will raise DSR to “high”.

CTS Is Held High When Device Is Idle: (DATASET TO DTE)

Options: Yes/No

Default: Yes

This parameter specifies how the system will hold the state of CTS on the DATASET to the attached idle Data Device.

Note: This parameter MUST be enabled to provide “Keyboard Origination” (Auto Baud Detect).

YES: ENABLES the parameter and is the default value. It indicates that CTS pin of the dataset will be remain “high” even when the data device is idle. This is the normal setting for this parameter.

NO: DISABLES the parameter. When disabled, The CTS pin of the DATASET will be held “low” until the data call is connected. This means that the DTE cannot be used to define the destination of the call as the DATASET will be in a “not ready” state preventing the DTE to send data to the DTRX. For the data call to be connected, the destination must be dialed using the ADL or HOTLINE features. Once connected the DATASET will raise DSR to “high”.

Originate A DTRX Call With A Low → High Transition Of DTR: (DTE TO DATASET)

Options: Yes/No

Default: No

This parameter defines if DTR “high” will indicate the origination of a data call. This function is typically enabled for “HOTLINE services”.

YES: ENABLES the parameter. When the data device is powered “ON” the DATASET senses that DTR Is “high”. This is interpreted by the system as a request for session. “HOTLINE” applications cause the call to be completed to the defined destination. Other calls will prompt users with “*”, requesting a destination.

NO: DISABLES the parameter. It indicates that data calls will be initiated using the other means of “request for session”. Calls are started using the [CR] from the keyboard, the Call Button on the DATASET or the Call key on the Digital SUPERSET telephone.

Action To Be Taken If The Called DTE Has DTR Low (Auto Answer): (DATASET TO DTE)

Options: TOGGLE RI, RAISE DSR, RAISE DCD, REFUSE

Default: REFUSE

This parameter direct the system as to which signal to apply to the idle DTE in order for the DATASET to “call” or alert the attached device (terminal, PC, etc.) when the system is attempting the complete a data call.

A data device may keep it's DTR (RS 232C pin 20 Data Terminal Ready) “low” (not ready) until it is alerted. The data device (DTE) must respond with DTR high within one minute. Several methods are used to alert the device.

This parameter must be set to “REFUSE” for data devices which present and keep a “high” on DTR immediately upon power up.

The DATASET may signal the device in one of several ways depending on the device connected to the DATASET:

TOGGLE RI: The DATASET may be required to toggle RI (RS 232C pin 22 - Ring Indicator) to alert the device when DTR is “low”. When this is detected, the device responds to the “call” by making

DTR “high” and the call is connected. If DTR is high when the data device is called, the call will be barred.

RI is toggled with a fixed cadence of 2.5 seconds ON and 2.5 seconds OFF.

RAISE DSR: The DATASET may be required to make DSR nh4hn (RS 232C pin 6 Data Set Ready) to alert the device when DTR is “low”. The device responds to the “call” by making DTR “high” and the call is connected. If DTR is high when the data device is called, the call will be barred.

RAISE DCD: The DATASET may be required to make DCD “high” (RS 232C pin 8 - Data Carrier Detect) to alert the device when DTR is “low”. The device responds to the “call” by making DTR “high” and the call is connected. If DTR is high when the data device is called, the call will be barred.

REFUSE: The DATASET may not be required to alert the device for a data call to be completed. In this case the device has it's DTR “high” at all times. The data device will only be seized when DTR is “high”. If DTR is “low” when the data device is called, the call is barred. This is the default and is the most common setting.

Note: This parameter must be set to “REFUSE” for data devices which present and keep a “high” on DTR immediately upon power up.

Pooled Modem Communication Established Indicator: (DTE TO DATASET)

Options: DCD, DSR

Default: DCD

When data communication is being established between a Pooled Modem and a remote modem, this parameter identifies which signal lead will indicate that communication has been established. The options are raise DCD or raise DSR as required by the attached “pooled” modem.

First Modem Tone:

Options: 2025, 2100 or 2225 Hz

Default: 2025

When the system is establishing outgoing communication between a Pooled Modem and a remote modem, this parameter identifies which modem tone the pooled modem can detect. Once the system detects this tone, and recognizes it as a valid tone, it connects the call.

Second Modem Tone:

Options: 2025, 2100 or 2225 Hz

Default: 2225

When the system is establishing outgoing communication between a Pooled Modem and a remote modem, this parameter identifies a second modem tone the pooled modem can detect. Once the system detects this tone, and recognizes it as a valid tone, it connects the call.

ASYNC: Keyboard Origination Allowed (Auto Baud): (DTE TO DATASET)

Options: Yes/No

Default: Yes

The DATASET has the capability to perform auto baud detection, the ability of determining the baud rate of the attached device by detecting a carriage return [CR] character sent by the device.

With this parameter ENABLED (Set to YES), and the user types a carriage return from the data device keyboard, the DATASET detects the [CR] character and sends a baud rate report to the system before the data call is processed. The system attempts to establish the data call with the originator at this baud rate. If the originator's baud rate must be changed, the system will notify the user of the new baud rate with a message on the terminal.

This parameter has no effect when the data call is originated by using the DATASET call button. In this case the previous baud rate (of the last session) or the default baud rate is used.

In conjunction with this parameter being enabled, the following parameters must also be set to YES:

- DSR Is Held High When Device Is Idle
- CTS Is Held High When Device Is Idle.

ASYNC: ADL Auto Baud: (DTE TO DATASET)

Options: Yes/No

Default: No

This parameter will determine whether an ADL originator needs to enter a carriage return to set the baud rate. With this parameter ENABLED (set to "YES"), The ADL originator must enter a carriage return, anytime after dialing the ADL access code.

With this parameter DISABLED (set to "NO"), the system will attempt to establish the data call with the originator's last used baud rate.

ASYNC: Flow Control: (DTE TO DATASET/DATASET TO DTE)

Options: CTS, XON/XOFF, NONE, PIN 25HI/CTS or PIN 25 LO/CTS

Default: XON/XOFF

This parameter identifies the signalling method used by the attached device to control the flow of information to and from the DATASET. Flow Control is required to enable devices operating at different baud rates to communicate together (Speed Conversion). Flow control is achieved by using the RS 232C Clear to Send signal (CTS   RS 232C pin 5) or through the XON/XOFF characters sent in the data stream (description follows).

Note: Speed Conversion WILL ONLY OCCUR when both ends of a data call have Flow Control enabled. The Flow Control type is defined for the DATASET and attached data device, and does not need to be the same at both ends of a data call.

CTS: Indicates that pin 5 (CTS) of the RS 232C connector between the DATASET and the device is used to control the data transmission (CTS "low" = no data flow; CTS "high" = data flow).

Note that the CTS method of Flow Control is a one direction method of Flow Control. When the DATASET is instructed to stop the flow and is programmed with the "CTS method", it will drop CTS. The attached data device should recognize this and stop transmitting data. When the dataset is instructed to start the flow again, it will raise CTS and the attached device should begin transmitting again

XON/XOFF: Indicates that the XON and XOFF characters are used to control (turn ON and turn OFF) the data transmission. This is a common and preferred means of Flow Control.

None: Indicates that there is no "Flow Control" signalling to and from the attached device. In this case, both ends of a data call must be set to the SAME Baud Rate.

ASYNC: XON Character: (FLOW CONTROL CHARACTER)

Options: 0-127, Decimal value of ASCII code

Default: 17, [Cntrl Q]

This is the character to turn "transmit on" between the DATASET and the attached device (terminal, PC, etc.). This is often the "scroll" key on a terminal keyboard.

ASYNC: XOFF Character:(FLOW CONTROL CHARACTER)

Options: 0-127, Decimal value of ASCII code

Default: 19, [Cntrl S]

This is the character to turn “transmit off” between the DATASET and the attached device (terminal, PC, and so forth). This is often the “no scroll” key on a terminal keyboard.

ASYNC: Break Key Function: (DTE TO DATASET)

Options: SYSTEM ATTENTION or TRANSPARENT

Default: SYSTEM ATTENTION

This parameter is used to identify whether the BREAK key of the data device keyboard is to be used by the system as the “Attention” key or ignored by the system and passed through as a regular data character.

SYS ATT: Identifies that the BREAK key is to be monitored and interpreted by the system as “PBX Attention”.

For data calls initiated from the data device (stand alone data call), “PBX Attention” causes the present data call to be ABORTED (dropped) and the system to respond with the “*”, prompt at the device prompting the user to dial another data call. A new destination may be entered or the user may type [Cntrl E] to receive the “goodbye” message from the system.

For data calls initiated using the ADL feature, “PBX Attention” causes the present data call to simply drop the current call.

TRANSPARENT: Identifies that the BREAK key is not to be monitored or interpreted by the system. The BREAK character is passed transparently through the system to the destination device.

Although similar, the “disconnect” key on the DATASET or on the Digital SUPERSET telephones simply causes the data call to be dropped with no reply from the system.

ASYNC: PBX Attention Character: (DTE TO DATASET)

Options: 0-127, Decimal value of ASCII code

Default: 4, [Cntrl D]

This parameter is similar in purpose as defined for “SYSTEM ATTENTION” of the BREAK key definition. If BREAK is not to be used as the “attention” key any other character may be defined for the same purpose. By default decimal ASCII code 04, the [Cntrl D] character, is the “Attention” character.

If the current data call was made via DTRX, then the user will be prompted to dial another data call. If the current data call was via ADL, then the current call is simply dropped

Specifying ASCII code “00” as the PBX ATTENTION CHARACTER causes this parameter to be DISABLED.

ASYNC: Parity: (DTE TO DATASET/DATASET TO DTE)

Options: ODD, EVEN, NONE, SPACE or MARK

Default: ODD

This parameter is dependent on the device connected to the DATASET. It must be set the same as set for the device. Parity is simply used between the DATASET and the attached device. It is not used to communicate between the two parties of a data call.

The data device connected may send any parity to the DATASET however the DATASET will generate parity to the data device based on the programmed option for this parameter. If the programmable character length is set to 8 bits then NO PARITY is assumed.

ASYNC: Character Length: (DTE TO DATASET/DATASET TO DTE)

Options: 7 or 8, 8 implies no parity

Default: 7

This parameter is dependent on the device connected to the DATASET. It must be set the same as set for the device. The character length may be 7 or 8 bits long. Setting “8 bit words” implies that there is no parity (i.e., Parity None) independent of the “Parity” parameter setting.

ASYNC: Number Of Stop Bits: (DTE TO DATASET/DATASET TO DTE)

Options: 1 or 2

Default: 1

This parameter is dependent on the device connected to the DATASET. It must be set the same as set for the device. The number of stop bits may be 1 or 2 stop bits.

ASYNC: Autobaud To Host Character 1 And 2: (DATASET TO DTE)

Options: 0-127, Decimal value of ASCII code

Default: CHARACTER 1: 13, [CR] Carriage Return
CHARACTER 2: 0, [NUL] Null (disabled)

These parameters are designed for DATASET/data device combinations that are the destination of data calls. The decimal equivalent of the desired ASCII character is programmed.

To DISABLE this feature, program a [NUL], decimal 00, character into both Autobaud To Host Characters. If Character 1 is non zero and Character 2 is zero, then only Character 1 is sent. However, if Character 1 is zero and Character 2 is non zero, then both characters will be sent.

This feature is designed for attached data devices which can automatically set their baud rate and/or parity upon receiving specific characters. If the device does not have this capability then one of the following should be done:

- Set Minimum Baud Rate = Maximum Baud Rate = Default Baud Rate, or
- Enable the “Always Use Default Baud Rate When Called” parameter

If one of the above options is selected, the Autobaud to host characters can still be programmed for other purposes. For instance, to some data devices, a carriage return (decimal 13) will result in a prompt being returned. For dumb devices such as printers, a Form Feed (decimal 12) may be used to ensure that the document to be printed will start at the top of a fresh sheet of paper.

ASYNC: Delay Between Autobaud Characters: (DATASET TO DTE)

Options: 0-1270 msec, 10 msec inc

Default: 100

This parameter defines the time delay between sending the first AUTOBAUD TO HOST character and the second. This typically represents the scanning rate of host computer. This timer is set as specified for the computer host.

DS2100: Operating Mode: (SYSTEM) (default = ASYNCHRONOUS)

Options: ASYNCHRONOUS, SYNCHRONOUS

Default: ASYNCHRONOUS

This parameter is for DS2100 series DATASETS only. It selects the operating mode of the DATASET to be either asynchronous or synchronous.

SYNC: Rate Adaptation Scheme: (SYSTEM)

Options: MiNET or X.31

Default: MiNET

In synchronous operation of the DS2100 dataset, this parameter defines the synchronous operation to be either transparent mode or X.31 mode.

SYNC: Clock Source: (SYSTEM) (default = INTERNAL)

Options: INTERNAL, SYSTEM, TX EXT or TX & RX EXT

Default: INTERNAL

This parameter selects the clock source for synchronous operation of the DATASET 2100. The options are:

INTERNAL: This option applies when the DATASET 2100 is operated as a DCE. The DTE transmit clock signal comes from the dataset's internal baud rate generator which is not synchronized to the PBX timing. The DTE Receive clock signal comes from the dataset's receiver Phase Locked Loop (PLL). The PLL extracts the timing from the data received from the far end dataset.

SYSTEM: This option applies when the DATASET 2100 is operated as a DCE. Both the DTE transmit and receive clock signals come from the dataset's receiver Phase Locked Loop (PLL). The PLL extracts the timing from the data received from the far end data set.

TX EXT: This option applies to both DCE and DTE operation. The DTE Receive clock signal comes from the dataset's receiver PLL. The PLL extracts the timing from the data received from the far end dataset. When the DATASET 2100 is operated as a DCE, the DTE transmit clock signal comes from an external clock signal on RS-232 pin 24 from the attached device. When the DATASET 2100 is operated as a DTE, the external clock signal is derived from the DCE's received data.

TX & RX EXT: This option applies when the DATASET 2100 is operated as a DTE. Both the receive and transmit data clocks come from the attached DCE. The external transmit clock is derived from the modem's received data (Pin 17) and is remapped to pin 24 on the dataset via the DCE adapter. The external receive clock is derived from the modem's transmit data (Pin 15) and is remapped to the dataset pin 18 using the DCE adapter.

Form 12 - Data Assignment

This form is used to program IP sockets and data devices.

IP Sockets

IP sockets can be programmed to output data for the following functions:

- SMDR
- CDE Data Print
- Maintenance Logs
- Traffic Measurement
- IP Traffic Measurement
- ACD Real Time Events
- ACD Agent Print
- ACD Group Summary
- Hotel/Motel Audit
- Hotel Motel Wakeup

Telnet clients and network-ready computers can connect directly to the sockets. Serial printers can connect through the use of a 3rd-party RS232-to-IP convertor.

You can program sockets on the IP bay in slot 13, circuits 19 through 28. The actual IP socket number consists of the PLID preceded by the number 6. For example, programming PLID 1/13/22 reserves socket 61322 for data output.

After programming a socket for data output, define the printout type (Autoprint, Directed, or Monitor) on Form 34.

Data Devices

When a Digital Line Card is programmed in the System Configuration Form, the system creates an entry line for each of its circuits in Form 09 (Desktop Device Assignments) and in Form 12, Data Assignment. If the card is in an upper (high power) slot, the system also creates entry lines for its circuits in Form 07, Console Assignments.

When the programmer assigns a Digital Line Card port as a DATASET, the system removes the corresponding line from Form 09.

Digital Line Card ports which appear in Form 07 are removed if assigned as a Digital SUPERSET telephone, DATASET, or MOH/Pager Unit. Likewise, a port assigned as a console is removed from Form 09. Then the only data device type that can be selected on the corresponding line in Form 12 is DSCONS, the console printer port.

The PLID of an attendant console with a PKM 48 directly connected with it will not appear in Form 12. The attendant console can either associate with a printer or a PKM 48, not both.

Figure: Form 12 Layout

Field Descriptions

BAY/SLT/CCT/: This field specifies the physical location of each device. This list is generated by the system based on what was entered in the programmed field of Form 01, System Configuration. This field cannot be modified.

TYPE: This field identifies the type of data device programmed. The available devices are: DS1102, DS1103, DS2102, DS2103, DSCONS, 1101M and SOCKET.

TEN: The tenant group for each device is specified in this field.

EXT NUM: This field displays the assigned extension number of a data line.

COS: This field lists the Class-of-Service number specification of each device (1 to 50).

COR: This field lists the Class-of-Restriction number specification of each device (1 to 25).

CDN: This field lists the Circuit Descriptor Number assigned to a device (1 to 25).

DTE: This field lists the Data Terminal Equipment Profile number (1 to 25). A data device must have a DTE Profile number to access a DTRX. Otherwise, this field should be left blank.

AVL: This field lists the Associated Voice Line (directory number) used to associate a DATASET with a Voice set, so the ADL (Associated Data Line) can be used.

HOTLINE: This field lists the directory number of the destination DTE.

COMMENTS: This field is reserved for additional data (a maximum of 15 characters). It is stored by the system but not used.

Softkeys

AIM: This softkey appears after the DATA DEV TYP softkey has been pressed. Pressing this softkey assigns the device a SIM2, an analog interface module to an analog device (industry-standard set, modem, fax machine, etc.).

DATA DEV TYP: Pressing the DATA DEV TYP softkey displays softkeys which assign the type of data device connected to the circuit displayed on the command line.

DS1103: This softkey appears after the DATA DEV TYP softkey has been pressed. Pressing this softkey assigns the device as an Asynchronous Stand-alone dataset.

DS2103: This softkey appears after the DATA DEV TYP softkey has been pressed. Pressing this softkey assigns the device type as an asynchronous/synchronous rack-mount dataset.

DSCONS: This softkey appears after the DATA DEV TYP softkey has been pressed. Pressing this softkey assigns the device type as a Mk 2 console printer port.

SOCKET: This softkey appears after the DATA DEV TYP softkey has been pressed when the cursor is on circuits 19 through 28 of slot 13. Pressing the softkey makes the circuit available for bi-directional data transfer via a Telnet session or data output only to a printer. On the AX and CX controller, PLID 1/3/28 defaults to a printer port, directing SMDR and CDE Data Print output to socket 61328.

Note: Although not shown in the form, IP socket PLIDs include a subcircuit number. The number is 02 for all sockets. Therefore, an IP socket programmed in Form 12 at PLID 1/13/20 is actually at PLID 1/13/20/02, not 1/13/20/00. The subcircuit number is required when using Maintenance to suspend and resume data output to the printer device.

FIND EXT: Pressing the FIND EXT softkey displays the ENTER EXTENSION NUM prompt. When a valid extension number of a DATASET is entered, its physical location is displayed on the command line.

FIND AVL: Pressing this softkey prompts the user to enter an extension number. When a valid extension number of an associated voice set is entered, and the ENTER softkey is pressed, the physical location of the dataset which is associated with the selected voice set is displayed on the command line.

RANGE PRGM: This softkey facilitates block programming of data devices. Pressing this softkey displays FROM BAY: SLOT: and CIRCUIT: TO BAY: SLOT: CIRCUIT: prompts on the command line. The range of devices is then specified by entering valid bay, slot and circuit numbers for the first and last devices. The entry is completed by pressing the ENTER softkey. When the extension number for the first device is entered, the system automatically assigns incremented extension numbers, the same COS, COR, CDN, and DTE for each device in the block. A dual circuit dataset cannot be included in range programming.

DELETE FIELD: This softkey appears when the cursor is positioned in the AVL or HOTLINE field, when a value has been already programmed. Pressing the DELETE FIELD softkey removes the value which is programmed in that field. The deletion is completed by pressing the ENTER softkey.

The standard softkeys **DELETE**, **ENTER**, **BAY/SLT/CCT** and **QUIT** are also provided.

Form 13 - Trunk Circuit Descriptors

Trunk circuit descriptors are similar to Classes of Service. A trunk circuit descriptor specifies the hardware options for each type of trunk card. Trunk circuit descriptors are complemented by the switch settings on the 9105/9110 type trunk cards. Refer to the Trunk Card Switch Assignments Forms in the Engineering Information section. The trunk circuit types are as follows:

4-Circuit CLASS	T1 CO DISA	6-Circuit CO
T1 DID/TIE	E&M Module DISA	6-Circuit DID
T1 E&M	6-Circuit DISA	T1 E&M DISA
T1 LS/GS	E&M Module	
T1 TIE/DISA	BASIC RATE	8-Circuit CLASS
AMB/COMBO CLASS		

Each circuit descriptor type has its own set of parameters as detailed in the Trunk Options Table . The system supports a maximum of 25 trunk circuit descriptors.

Figure: Form 13 Layout

Field Descriptions

DESCRIPTOR: This field lists the trunk circuit descriptors, numbers 01 to 25 (maximum of 25 different descriptors in total).

TRUNK TYPE: This field lists trunk circuit type for each trunk circuit descriptor. Note that the selected trunk type (the one that appears on the command line) is not displayed on the softkeys.

Note: Trunk type cannot be changed if there are one or more trunks assigned to the descriptor.

NUMBER OF TRKS ASSIGNED: This field records the number of trunks which use each trunk circuit descriptor. The trunk circuit descriptor can be assigned a new trunk type only if this field is zero. To clear the NUMBER OF TRKS ASSIGNED field, the trunks must first be de-assigned. Refer to Form 14 (Non-Dial-In Trunks) and Form 15 (Dial-In Trunks). When a trunk type is assigned to a trunk circuit descriptor, the system prohibits any changes by clearing those softkeys that can alter the trunk types.

COMMENTS: This field is reserved for additional data (a maximum of 20 characters). It is stored by the system but not used.

Softkeys

BASIC RATE: This softkey only appears if you have Feature Level 2 or greater enabled. Selecting this softkey, enables you to select the SEL.OPTION softkey and enable the option Direct Access on CO Line keys:Bypassing Key System Toll Control.

8 CCT CLASS: Pressing this softkey specifies the selected trunk circuit descriptor as a CLASS Trunk type.

6 CCT CO: Pressing this softkey assigns the CO Trunk type to the selected trunk circuit descriptor.

CCT DID: Pressing this softkey programs the selected trunk circuit descriptor as a DID Trunk.

6 CCT DISA: Pressing this softkey assigns the DISA Trunk type to the selected trunk circuit descriptor.

E&M MODULE: Pressing this softkey programs the selected trunk circuit descriptor as an E&M Trunk Module.

EM MOD DISA: Pressing this softkey programs the selected trunk circuit descriptor as an EM MOD DISA.

T1 CO DISA: This softkey appears when the MORE softkey has been pressed for the third time. Pressing this softkey defines the selected trunk circuit descriptor as a T1 Trunk simulating a CO DISA Trunk.

T1 DID/TIE: This softkey appears when the MORE softkey has been pressed for the second time. Defines the selected trunk circuit descriptor as a T1 trunk simulating a DID/TIE trunk.

T1 E&M: This softkey appears when the MORE softkey has been pressed for the second time. Pressing this softkey defines the selected trunk circuit descriptor as a T1 Trunk simulating an E&M Trunk.

T1 E&M DISA: This softkey appears when the MORE softkey has been pressed for the second time. Pressing this softkey defines the selected trunk circuit descriptor as a T1 Trunk simulating an E&M DISA Trunk.

T1 LS/GS: This softkey appears when the MORE softkey has been pressed for the second time. Pressing this softkey defines the selected trunk circuit descriptor as a T1 Trunk simulating an LS/GS Trunk.

T1 TIE DISA: This softkey appears when the MORE softkey has been pressed for the third time. Pressing this softkey defines the selected trunk circuit descriptor as a T1 Trunk simulating a TIE/DISA Trunk.

DESC NUM: Pressing this softkey displays ENTER DESC NUM: prompt. This softkey selects a trunk circuit descriptor number. The selection is completed by entering a valid number (1 to 25).

SEL. OPTION: Pressing the SEL. OPTION softkey displays a new form. This form displays the options (parameters) associated with the trunk type that is assigned to that trunk circuit descriptor. Refer to Options Subform.

AUDIO CONFIG: This softkey appears only when the cursor is positioned on an AMB CLASS circuit. Pressing this softkey displays the Audio Configuration Table for the circuit.

AMB/COMBO CLASS: The AMB/COMBO CLASS circuit type applies only to the analog trunk circuits in the SX-200 ICP controller. Pressing this softkey specifies the selected trunk circuit descriptor as an ASU II or AMB CLASS Trunk type.

Note: The default trunks options for AMB trunks do not all apply when a trunk's descriptor is changed to AMB from something else. See Trunk Options for more information.

REVIEW: Pressing the REVIEW softkey displays a new form (Review List Subform). This form displays a list of trunks that use the selected trunk circuit descriptor. Note that this softkey appears only if “NUMBER OF TRKS ASSIGNED” on command line is greater than zero.

The standard softkeys **CANCEL**, **ENTER**, ****MORE**** and **QUIT** are also provided.

Subform 13 - Options

This form appears when the SEL. OPTION softkey is pressed and displays the options associated with each trunk type assigned to each trunk circuit descriptor. See the Trunk Options table for a list of options followed by descriptions.

Figure: Select Option Subform Layout

Field Descriptions

The header line displays the trunk type, the trunk circuit descriptor number and the type of parameter under observation.

STATUS: This field lists the option selected for each parameter.

Softkeys

YES, NO

60/40, 30/20, 66/33

IMMEDIATE, WINK, DELAY

DELAY INTEG

LOOP FLSH/**RING GND**: Toggles between an opened loop current type flash (LOOP FLSH), or a grounded ring type of flash (RING GND).

The standard softkeys **CANCEL**, **ENTER**, and **QUIT** are also provided.

Trunk Options

Table: Trunk Options

4-CIRCUIT or 8-CIRCUIT CLASS and 6-CCT AMB CLASS TRUNK
Reverse to Idle Far-End Gives Answer Supervision Inhibit Automatic Supervision No Seize Alarm No Release Alarm Save Busy-Out Status Calling Party Disconnect Timer (1 → 12 minutes) Post Call Metering (for 8-CIRCUIT CLASS TRUNK only) Ignore Remote Disconnect Disconnect Timer (100 → 9900 ms) (100 ms increments) Supervision Direction: Incoming Trunk Calls Also Guard Timer (0 → 3000 ms) (100 ms increments) CLASS Trunk (see Note below) Ring Cycle Timer (6 → 10 seconds) Ignore Line Reversal During Seizure Ringing Expected (see Note below) Ringing Debounce Timer (5 → 12 seconds) Seize Timer (10 → 60 s) (10 s increments) Flash Timer (200 → 700 ms) (100 ms increments) Flash Over Trunk Direct access on CO Line Keys: bypass Key System Toll Control (Yes or No) CO Silence telephone number CO Milliwatt telephone number **Note:** The defaults for these change from YES and NO respectively to NO and YES when a trunk's descriptor is changed to AMB from something else.
6-CCT DID
Reverse to Idle Far-End Gives Answer Supervision Inhibit Automatic Supervision No Seize Alarm No Release Alarm Line Length DTMF Save Busy-Out Status Impedance (600 Ohm or Complex) Disconnect Timer (150 → 900 ms) (50 ms increments) Release Acknowledge Timer (2-120 seconds) Guard Timer (200 → 1000 ms) (100 ms increments) Start Type (Immed, Wink, Delay) Debounce Timer (20 → 150 ms) (10 ms increments) Wink Timer (150 → 300 ms) (50 ms increments) Remote End is a Satellite Remote End is a Satellite with OPS Lines

6-CIRCUIT CO TRUNK and 6-CIRCUIT DISA
Reverse to Idle Far-End Gives Answer Supervision Inhibit Automatic Supervision No Seize Alarm No Release Alarm Line Length DTMF Save Busy-Out Status Impedance (600 Ohm or Complex) Post Call Metering (0 $\rightarrow$ 15 seconds) Calling Party Disconnect Timer (1 $\rightarrow$ 12 minutes) Dictation Trunk Ignore Remote Disconnect Disconnect Timer (100 $\rightarrow$ 9900 ms) (100 ms increments) Supervision Direction: Incoming Trunk Calls Also Guard Timer (0 $\rightarrow$ 3000 ms) (100 ms increments) Ring Cycle Timer (6 $\rightarrow$ 10 seconds) Ignore Line Reversal During Seizure Ringing Expected Ringing Debounce Timer (5 $\rightarrow$ 12 seconds) Seize Timer (10 $\rightarrow$ 60 s) (10 s increments) Flash Timer (200 $\rightarrow$ 700 ms) (100 ms increments) Flash Type (Loop Flash, Ring Ground) Flash Over Trunk Interdigit Timer (300 $\rightarrow$ 800 ms) (100 ms increments) Digit Outpulsing Ratio (60/40, 30/20, 66/33) Direct Access on CO Line Keys: bypass Key System Toll Control (Yes or No)

E&M MODULE and E&M MODULE DISA
Reverse to Idle Far-End Gives Answer Supervision Inhibit Automatic Supervision No Seize Alarm No Release Alarm Line Length DTMF Save Busy-Out Status Impedance ** use dip switch on the module to program ** (600 Ohm or Complex) E Lead Invert M Lead Invert ** required for type 5 operation ** Disconnect Timer (150 → 300 ms) (50 ms increments) Release Acknowledge Timer (2000 → 9900 ms) (100 ms inc) Guard Timer (200 → 1000 ms) (100 ms increments) Dictation Trunk Incoming Start Type (Immed, Wink, Delay) Debounce Timer (20 → 150 ms) (10 ms increments) Wink Timer (150 → 300 ms) (50 ms increments) Outgoing Start Type (Immed, Wink, Delay or Delay Integ) Digit Outpulsing Ratio (60/40, 30/20, 66/33) Outpulse Delay Timer (100 → 2000 ms) (100 ms inc) Flash Timer (200 → 700 ms) (100 ms increments) Flash Type (Loop Flash or Ring Ground) Flash Over Trunk Interdigit Timer (300 → 800 ms) (100 ms increments) Wait for Delay Timer (300 → 5000 ms) (100 ms inc) Remote End is a Satellite Remote End is a Satellite with OPS Lines Direct Access on CO Line Keys Bypass Key System Toll Control Release Link Trunk

T1 DID/TIE and T1 TIE DISA
Reverse to Idle Far-End Gives Answer Supervision Inhibit Automatic Supervision No Seize Alarm No Release Alarm Line Length DTMF Save Busy-Out Status Disconnect Timer (150 → 300 ms) (50 ms inc) Release Acknowledge Timer (2 → 120 s) Guard Timer (200 → 1000 ms) (100 ms inc) Incoming Start Type (Immed, Wink or Delay) Debounce Timer (20 → 150 ms) (10 ms inc) Wink Timer (150 → 300 ms) (50 ms inc) Outgoing Start Type (Immed, Wink, Delay or Delay Integ) Digit Outpulsing Ratio (60/40, 30/20, 66/33) Outpulse Delay Timer (100 → 2000 ms) (100 ms inc) Flash Timer (200 → 700 ms) (100 ms inc) Flash Type (Loop Flash, Ring Ground) Flash Over Trunk Interdigit Timer (300 → 800 ms) (100 ms inc) Wait for Delay Timer (300 → 5000 ms) (100 ms inc) Remote End is a Satellite Remote End is a Satellite With OPS Lines Direct Access on CO Line Keys Bypass Key System Toll Control QSIG Supplementary Services (operates for T1 DID/TIE only)

T1 E&M and T1 E&M DISA
Reverse to Idle Far-End Gives Answer Supervision Inhibit Automatic Supervision No Seize Alarm No Release Alarm Line Length DTMF Save Busy-Out Status Disconnect Timer (150 → 300 ms) (50 ms inc) Release Acknowledge Timer (2-240 s) (2 s inc) Guard Timer (200 → 1000) (100 ms inc) Incoming Start Type (Immed, Wink or Delay) Debounce Timer (20 → 150 ms) (10 ms inc) Wink Timer (150 → 300 ms) (50 ms inc) Outgoing Start Type (Immed, Wink, Delay or Delay Integ) Digit Outpulsing Ratio (60/40, 30/20, 66/33) Outpulse Delay Timer (100 → 2000 ms) (100 ms inc) Flash Timer (200 → 700 ms) (100 ms inc) Flash Type (Loop Flash, Ring Ground) Flash Over Trunk Interdigit Timer (300 → 800 ms) (100 ms inc) Wait for Delay Timer (300 → 5000 ms) (100 ms inc) Remote end is a Satellite Remote end is a Satellite with OPS Lines Direct access on CO Line Keys: bypass Keys System Toll Control (Yes or No) Release Link Trunk QSIG Supplementary Services (operates for T1 E&M only)
T1 LS/GS TRUNK and T1 CO DISA
No Seize Alarm No Release Alarm Line Length DTMF Save Busy-Out Status Loop Start or Ground Start (T1 LS, T1 GS) Calling Party Disconnect Timer (1 → 12 minutes) Disconnect Timer 100 → 9900 ms (100 ms inc) Guard Timer (0 → 3000 ms) (100 ms inc) Ring Cycle Timer (6 → 10 s) Ringing Expected Ringing Debounce Timer (5 → 12 seconds) Seize Timer (10 → 60 s) (10 s increments) Flash Timer (200 → 700 ms) (100 ms inc) Flash Type (Loop Flash, Ring Ground) Flash Over Trunk Interdigit Timer (300 800 ms) (100 ms inc) QSIG Supplementary Services (operates for T1 LS/GS Trunk only)

Parameters

Calling Party Disconnect Timer: This defines how long the system will wait for the far-end, a ground start trunk, to acknowledge a trunk release.

CO Silence Telephone Number: Enter the silence (balance) termination number (up to 12 digits) provided by the Central Office. The number is used when the Line Quality Test is run to determine the trunk's impulse response and provide a recommended IMPEDANCE setting on Subform 13, Audio Configuration Table.

CO Milliwatt Telephone Number: Enter the milliwatt tone number (up to 12 digits) provided by the Central Office. The number is used when the Line Quality Test is run to determine the trunk's loss level and provide a recommended LENGTH on Subform 13, Audio Configuration Table.

CLASS Trunk: This option specifies that CLASS detection is required. If selected, CLASS Calling Line ID digits are expected, and acted upon when received on the trunk.

Debounce Timer: This timer specifies the period for which an incoming seizure is to be debounced before being recognized as a valid incoming seizure.

Dictation Trunk: If selected, this maintains trunk dialing for the duration of the call. See Dictation Trunk in the Features Descriptions section.

Digit Outpulsing Ratio: This field specifies the break/make ratio during outpulsing. It can be set to 60/40, 66/33, or 30/20.

Direct Access on CO Line Keys: Bypass Key System Toll Control: If set to yes, allows a user to press a CO Line Key to bypass Key System Toll Control and seize a CO line. Default is off.

Disconnect Timer: This defines the time a release signal must be continuously present before a call is disconnected. Note: For digital DID trunks, the range is 150-300 ms. For E&M Release Link Trunks, the range is 150-900 ms.

DTMF: If selected, forces DTMF digits to be transmitted on the trunk when dialing. If not selected, digits are pulsed onto the trunk.

E Lead Invert/ M Lead Invert: These two fields provide the flexibility to specify the polarity of the E and M leads to match the far end connection. M Lead Invert must be enabled for type 5 operation.

Far-End Gives Answer Supervision: If selected, answer signals are expected, and acted upon when received on the trunk; answer signals are not generated internally. If not selected, answer signals received are ignored.

Flash Over Trunk: Enables the Flash Over Trunk feature for the descriptor number.

Flash Timer: This defines the duration of a flash transmitted onto a trunk in a digital bay. Note: this is not programmable for some analog trunks; in these cases, the flash timer is always 200 ms.

Flash Type: Specifies whether the flash will be done by opening the current loop (Loop Flash) or grounding the ring (Ring Ground).

Guard Timer: This defines how long the system will wait after releasing the trunk before seizing it again for an outgoing call.

Ignore Line Reversal During Seizure: If selected, line reversal is not recognized as an incoming seizure.

Ignore Remote Disconnect: If selected, release signals from the far-end are ignored. If not selected, release signals cause disconnection of the call.

Incoming Start Type: This field specifies the incoming type of the trunk, which can be set to immediate, wink start, delay dial or delay dial with integrity. IP trunks use Wink as the incoming and outgoing start type.

Inhibit Automatic Supervision: If selected, the system waits for the far-end to provide answer supervision before providing answer supervision to an incoming Tie or DID trunk.

Interdigit Timer: This defines the time gap inserted between outpulsed digits.

Loop Start or Ground Start: This defines trunk signaling type (T1 only).

No Release Alarm: If selected, a trunk failing to release is removed from service and maintenance is notified. The trunk can only be returned to service manually. If not selected, the trunk remains in service.

No Seize Alarm: If selected, a trunk failing to return a seize acknowledgment on three successive occasions is removed from service; maintenance is notified. If not selected, the trunk remains in service. If the trunk originates, the no seize count is reset. As well, if the trunk had been removed from service due to a no seize count, the trunk is returned to service and the incoming call on the trunk is processed.

Outgoing Start Type: This field specifies the outgoing type of the trunk, which can be set to immediate outgoing, wink start outgoing, delay dial outgoing or delay dial with integrity. IP trunks use Wink as the incoming and outgoing start type.

Outpulse Delay Timer: This timer specifies the pause between seizing and the start of dialing, applicable to immediate outgoing trunks only. This value should be specified after determining the far end characteristics.

Present Node ID over IP Trunking: If selected, the Node ID is sent over IP Trunks. For more information on IP trunking, see Networking Mitel IP-PBXs.

Post Call Metering: This defines how long the system will wait for and record meter pulses after the release signal is received.

QSIG Supplementary Services: This supports the QSIG features other than Basic Call and Calling Name. For a list and description of the QSIG features, see QSIG. IP Trunks require QSIG supplementary services.

Release Acknowledge Timer: This specifies the time-out period to wait for a release acknowledge signal from the far end.

Release Link Trunk: The release link trunk (RLT) allows a flash to be sent over it when setting up a call between stations at a different PBX from the attendant. If a device receiving a RLT call has a line key appearance, a flash will be ignored. Do not program Direct Trunk Select or CO Line keys that access the RLT.

Remote End is a Satellite: Not applicable to the SX-200 ICP.

Remote End is a Satellite with OPS Lines: Not applicable to the SX-200 ICP.

Reverse to Idle: In some central offices, upon seizure, the CO reverses the polarity on the trunk. When the call ends, the CO again reverses the polarity, returning the trunk to its normal idle state. When the Reverse to Idle option is enabled, the PBX treats the reversal to idle condition as a disconnect signal from the CO. The Far-End Gives Answer Supervision option has no effect.

Ring Cycle Timer: This defines a period during which a minimum ring burst (250 ms) must be present before the system will recognize it as an incoming call.

Ringing Debounce Timer: This defines the duration during which the system tries to detect the minimum ring burst, indicating the persistence of an incoming call.

Ringing Expected: If selected, incoming calls are not reported unless ringing is recognized. If other seize signals are received before ringing, the trunk is busied-out for outgoing calls, but the incoming call is not reported until ringing is received.

Save Busy Out Status: If selected, all trunks in this descriptor type which were in the busy-out state before a system reset will be in that state after the reset.

Seize Timer: This defines the time the system will wait for a seize acknowledge from a ground start trunk. Also see Option "No Seize Alarm" in this list.

Start Type: This field specifies the start type of the trunk, which can be set to immediate, wink start, or delay dial.

Supervision Direction: Incoming Trunk Calls Also: If this option is selected, the system will record meter pulses for incoming trunk calls. If the CO provides a reversal as a disconnect signal, this option must be set to YES for the PBX to respond to the reversal.

Wait For Delay Timer: This timer specifies the period to wait for the delay signal from the far end. It is only applicable if the trunk is of type delay dial outgoing (without integrity).

Wink Timer: This timer specifies the duration of the wink signal sent to the far end if the trunk is programmed as a Wink Start Incoming or Delay Start incoming.

Subform 13 - Audio Configuration Table

This form appears when the AUDIO CONFIG softkey is pressed while the cursor is positioned on an AMB CLASS circuit in Form 13 - Trunk Circuit Descriptors. Use this form to program audio configuration settings (line length and impedance) for the trunks, and to program a loopback milliwatt test if the milliwatt tone number is unknown. Refer to the figure below for the form layout.

Loopback Milliwatt Test

If the milliwatt tone number is unknown, you are required to program the trunks so that one trunk provides milliwatt tone for all other trunks in a loopback setup when the Line Quality Test is performed.

Sample setup:

TRK	TEL NO:	MWT:
1	2331001	6
2	2331002	6
3	2331003	6
4	2331004	6
5	2331005	6
6	2331006	1

In the above example, trunk 6 provides milliwatt tone for trunks 1 to 5.

Figure: Audio Configuration Table Subform Layout

Field Descriptions

TRK: This field lists the trunk numbers assigned to the selected trunk circuit descriptor number. If this trunk provides milliwatt tone to one or more other trunks in a loopback setup, then its trunk number must be entered in the MWT field for these trunks.

BAY, **SLT** and **CCT**: These fields list the physical location of each trunk number according to their bay, slot and circuit numbers. The onboard ASU circuits always use slot 13 of the IP bay.

LENGTH: Depending on which line length has been selected, this field will display SHORT, LONG, XLONG, or AUTO. If AUTO is selected, the line length is programmed automatically by the Line Quality Test. By default, prior to running the test, this field is set to AUTO and functions as LONG.

IMPEDANCE: Depending on which impendence has been selected, this field will display COMPLEX, CENTREX, DSL, 600 OHM, IMP A, IMP B, IMP C, IMP D, or AUTO. If AUTO is selected, the impedance is programmed automatically by the Line Quality Test. By default, prior to running the test, this field is set to AUTO and functions as COMPLEX.

TEL NO: The incoming telephone number of the trunk at the CO.

MWT: The number of the trunk that provides milliwatt tone in a loopback setup. The trunk that provides milliwatt tone must be an AMB trunk, and must be a different trunk than the trunk being tested (for example, MWT 1 cannot be entered for Trunk 1).

COMMENT: This field stores additional data (a maximum of 14 characters) for the programmer's reference. The system does not use this information for call processing.

Softkeys

LONG LOOP, SHORT LOOP, X LONG LOOP, and **AUTO**: These softkeys display when the cursor is in LENGTH column. Use them to define the optimum line length setting for the trunk. Run the Line Quality test to determine which setting to use. If AUTO is selected prior to running the test, the recommended setting is programmed automatically. The test results (and the actual setting for AUTO) are available for review in the maintenance logs.

COMPLEX, CENTREX, DSL, 600 OHM, IMPEDANCE A, IMPEDANCE B, IMPEDANCE C, IMPEDANCE D, and **AUTO**: These softkeys display when the cursor is in IMPEDANCE column. Use them to define the optimum impedance setting for the trunk. Run the Line Quality test to determine which setting to use. If AUTO is selected prior to running the test, the recommended setting is programmed automatically. The test results (and the actual setting for AUTO) are available for review in the maintenance logs.

Notes:

1. Centrex business lines are provided in some areas as an alternative to the standard complex business lines. Some features such as CLASS/CLIP may only be available on centrex based lines
2. Most modern aDSL lines work with either complex or centrex line terminations. However, some older installation may need the DSL termination, identifiable by a large filter unit fitted to the line.
3. For longer lines the complex balance may give better performance than a 600 Ohm termination.
4. Impedance B, C, and D use the 600 Ohm setting.
5. If Impedance and Line Length are correctly set but the site is experiencing echo, forward the logs containing the Line Quality Test results to Mitel Product Support for analysis.
6. Most causes for echo between a LS trunk and IP Phones are caused by Impedance mismatches. Impedance mismatches have two components: line length and impedance settings. By default the AMB Trunk Circuit descriptor for the LS trunks on the Analog Main Board (AMB) is set to Long and Complex, which will match 80% of the installations.

The standard softkeys **CANCEL**, **ENTER**, and **QUIT** are also provided.

Subform 13 - Review List

This form appears when the REVIEW softkey is pressed in Form 13 - Trunk Circuit Descriptors. The form lists the trunks that use the selected trunk circuit descriptor. Refer to the figure below for the form layout. The data in this form cannot be modified.

Figure: Form 13 Review List Subform Layout

Field Descriptions

The header line displays the descriptor number and the trunk type.

TRK NUM: This field lists the trunk numbers assigned to the selected trunk circuit descriptor number. Trunk numbers are arbitrarily assigned to the trunks in Form 14 (Non-Dial-In Trunks) and Form 15 (Dial-In Trunks).

BAY, **SLT** and **CCT**: These fields list the physical location of each trunk number according to their bay, slot and circuit numbers.

COMMENTS: This field displays any additional information about each trunk as it was entered on Form 14 (Non-Dial-In Trunks) and Form 15 (Dial-In Trunks). The COMMENTS field stores a maximum of 15 characters. The data in this field is stored by the system but not used.

Softkeys

DESC NUMBER: This softkey selects a trunk circuit descriptor number. Pressing the DESC NUMBER softkey displays the ENTER DESC. NUM: prompt. The selection is completed by entering a valid number (1 to 25).

The standard softkeys **CANCEL**, **ENTER**, and **QUIT** are also provided.

Form 14 - Non-Dial-In Trunks

This form specifies the characteristics of the system's Non-Dial-In Trunks. These trunks cannot dial any digits into the PBX and are usually CO trunks. The Day, N1, N2 answer points are assigned in this form. Alternate recall points are assigned in Form 19 (Call Rerouting Table). Refer to the figure below for the form layout.

Note: Bay 1, Slot 13 is not applicable to the AX.

Figure: Form 14 Layout

Field Descriptions

BAY, **SLT**, and **CCT**: These fields list the physical location of each Non-Dial-In Trunk. They are generated by the system based on what was entered in the PROGRAMMED field in Form 01, System Configuration. This field cannot be modified.

COS: This field specifies the Class of Service of each Non-Dial-In Trunk. The default COS is 1.

TEN: This field specifies the Tenant Group number of each Non-Dial-In Trunk. The default Tenant Group number is 1.

DAY, **N1** and **N2**: These fields are reserved for the Day, Night1 and Night2 answer points. The answer points may be specified as an LDN on the attendant console, an index number for an abbreviated dial number, a console extension number, a station, a hunt group, a night bell, a logical line, a data station, or an ACD path access code. Note that an LDN and night bell cannot be rung simultaneously. These fields may be left blank.

CDN: The CDN (Circuit Descriptor Number) field links this form to Form 13, Trunk Circuit Descriptors, which defines the trunk hardware parameters. This field must be filled in before any changes for the selected physical location are stored in the database.

TK NUM: This field lists the trunk identification numbers. This field must be filled in before any changes for the selected physical location are stored in the database. In this field, trunks are listed according to their trunk number (1 to 200). This method of identifying trunks is used for the following:

- SMDR records of a trunk call (only three digits are allocated for trunk identification),
- Identification of a trunk in a call on the attendant console or a Mitel display telephone,
- Attendant Direct Trunk Select (DTS) capability,
- Form 09 for Mitel telephone line appearance programming (DTS or Private Trunk), and CO Line appearance programming,
- Form 16 for listing members of trunk groups.

TK NAME: This field lists the trunk names. Names can be up to eight characters long.

Note: Line appearances programmed to key positions three and above on SpectraLink NetLink telephones fail to display unless the trunk name is "Line n" where "n" is the trunk number.

COMMENTS: This field is reserved for additional data (a maximum of 15 characters).

For detection and reporting 911 calls to attendant consoles or by e-mail, enter the physical location of each non-dial-in trunk in the Comments field. Type an "@" followed by the physical location of the non-dial-in trunk. All consoles are notified of a 911 call regardless of their tenant group. Only the text following the @ is displayed on the consoles. A maximum of 14 characters can be displayed on the console.

Softkeys

RANGE PRGRM: This softkey allows block programming of trunks on the same card. You must program the first trunk in the range and then press RANGE PRGM to copy the values to the range of trunks that you specify. The following fields are copied from the first trunk in the range:

- COS, TEN, DAY, N1, N2, CDN

The TK NAME and COMMENTS fields are left blank.

The trunk numbers are assigned in sequence, beginning with the first trunk in the range. If there are trunks that are already programmed within that range, those trunks are skipped. The trunk number sequence is incremented each time a trunk is skipped. If the trunk number is incremented to a value that is already used, the sequence continues to increment until a free number is found. The trunk numbers restart at 1 if necessary. An error message is displayed if there are no more free trunk numbers.

To program a range of trunks

1. Program the first trunk in the range. Enter values for the **COS**, **TEN**, **DAY**, **N1**, **N2**, and **CDN** fields.
2. Press **RANGE PRGRM**.
   The system displays: FROM BAY: SLOT: CIRCUIT:.
3. Enter valid Bay, Slot, and Circuit numbers for the first trunk, and press the ENTER softkey.
   The system displays: TO BAY: SLOT: CIRCUIT:.
4. Enter valid Bay, Slot, and Circuit numbers for the last trunk, and press the ENTER softkey.
   The system copies the values from the first trunk to all the trunks specified in the range.

TRUNK NUMBER: This softkey selects a trunk by its trunk number. Trunk numbers are assigned in Form 14 and Form 15. Pressing the TRUNK NUMBER softkey displays the ENTER TRUNK NUM: prompt on the command line. Entering a valid trunk number (1 to 200) selects that Non-Dial-In Trunk and displays it on the command line.

DELETE FIELD: This softkey appears when the cursor is on a programmed DAY, N1 or N2 field. It allows the field to be deleted, without affecting the other two fields.

SPEED DIAL: Allows a speed dial number to be used to route a call via ARS to a Centralized Attendant at another PBX (this is the only intended use for this speed dial softkey). The Day, N1, and N2 fields are preceded by the letters sc when SPEED DIAL has been selected.

The standard softkeys **CANCEL**, **BAY/SLT/CCT**, **DELETE**, **ENTER**, and **QUIT** are also provided.

Form 15 - Dial-In Trunks

This form specifies the characteristics of the system’s Dial-In Trunks. This form also designates where incoming calls on Dial-In Trunks are routed by modifying the incoming digits. See the figure below for the form layout.

Note: Bay 1, Slot 13 is not applicable to the AX.

Figure: Form 15 Layout

Field Descriptions

BAY, **SLT** and **CCT**: These fields list the physical location identification of each Dial-In Trunk. They are generated by the system based on what was entered in the PROGRAMMED field of Form 01, System Configuration. This field cannot be modified.

COS: This field specifies the Class of Service for each entry. The COS defaults to 1.

COR: This field lists the Class of Restriction for each entry. The COR defaults to 1.

TEN: Tenant group specifications are listed in this field. The tenant group number defaults to 1.

N: This field lists the number of expected digits. The range is 0 to 10. If a value is specified in this field, then digit translation on incoming calls does not commence until the system receives the specified number of digits. This field must be filled in for a DID Trunk. Otherwise, this field defaults to 0, and the trunk is treated as a TIE trunk (regardless of the circuit descriptor).

For a DISA trunk, N must be 0 or blank.

The digit in the N column has no impact once a DISA access code is introduced.

M: This field specifies the number of digits (0 to 8) that must be absorbed after the incoming trunk is seized. The M field defaults to 0. This field is applicable for any type of Dial-In Trunk.

For a DISA trunk, M must be 0 or blank.

The digit in the M column has no impact once a DISA access code is introduced.

X: This field specifies a maximum of two digits that may be inserted before the digit string. This field is applicable for any type of Dial-In Trunk.

For a DISA trunk, X must be 0 or blank.

The digit in the X column has no impact once a DISA access code is introduced.

To apply the Digit Translation Table to calls received on a Dial-In trunk, program X with Feature Access Code 67 (Digit Translation Table Access) from Form 02 - Feature Access Codes. The code is restricted to two digits, maximum.

CDN: The CDN (Circuit Descriptor Number) field lists the circuit descriptor numbers for each Dial-In Trunk. This field links this form to Form 13 (Trunk Circuit Descriptors), which defines the trunk hardware parameters. Note that this field must be filled in before any changes for the selected physical location are stored in the database.

TK NUM: This field displays the Dial-In Trunks according to their trunk number. Note that this field must be filled in before any changes for the selected physical location are stored in the database. Trunk numbers range from 1 to 200 and are used for the following:

- SMDR records of a trunk call (only three digits are allocated for trunk identification),
- Identification of a trunk in a call on the attendant console or a Mitel display telephone,
- Attendant Direct Trunk Select (DTS) capability,
- Form 09 for Mitel telephone line appearance programming (DTS or Private Trunk) and
- Form 16 for listing members of trunk groups.

TK NAME: This field lists the trunk names. Names can be up to eight characters long.

COMMENTS: This field is reserved for additional data (a maximum of 15 characters).

For detection and reporting 911 calls to attendant consoles or by e-mail, enter the physical location of each dial-in trunk in the Comments field. Type an "@" followed by the physical location of the dial-in trunk. All consoles are notified of a 911 call regardless of their tenant group. Only the text following the @ is displayed on the consoles. A maximum of 14 characters can be displayed on the console.

Softkeys

RANGE PRGRM: This softkey allows block programming of trunks on the same card. You must program the first trunk in the range and then press RANGE PRGM to copy the values to the range of trunks that you specify. The following fields are copied from the first trunk in the range:

- COS, COR, TEN, N1, M, X, and CDN

The TK NAME and COMMENTS fields are left blank.

The trunk numbers are assigned in sequence, beginning with the first trunk in the range. If there are trunks that are already programmed within that range, those trunks are skipped. The trunk number sequence is incremented each time a trunk is skipped. If the trunk number is incremented to a value that is already used, the sequence continues to increment until a free number is found. The trunk numbers restart at 1, if necessary. An error message is displayed if there are no more free trunk numbers.

To program a range of trunks

1. Program the first trunk in the range. Enter values for the COS, COR, TEN, N1, M, X, and CDN fields.
2. Press RANGE PRGRM.

   The system displays: FROM BAY: SLOT: CIRCUIT:.
3. Enter valid Bay, Slot, and Circuit numbers for the first trunk, and press the ENTER softkey.

   The system displays: TO BAY: SLOT: CIRCUIT:.
4. Enter valid Bay, Slot, and Circuit numbers for the last trunk, and press the ENTER softkey.

   The system copies the values from the first trunk to all the trunks specified in the range.

TRUNK NUMBER: This softkey selects a trunk by its trunk number. Pressing the TRUNK NUMBER softkey displays the ENTER TRUNK NUM: prompt on the command line. Entering a valid trunk number (1 to 200) selects that Dial-In Trunk and displays it on the command line.

The standard softkeys **CANCEL**, **BAY/SLT/CCT**, **DELETE**, **ENTER**, and **QUIT** are also provided.

Form 16 - Trunk Groups

This form specifies the members of each trunk group by trunk numbers. The trunk number is assigned in Form 14 (Non-Dial-In Trunks) and Form 15 (Dial-In Trunks). The system supports a maximum of 50 trunk groups and each group supports a maximum of 50 members. See the figure below for the form layout.

Figure: Form 16 Layout

Field Descriptions

The header line indicates the trunk group being programmed, via a number and a name (maximum of eight characters). This line also indicates the presence of the SMDR option and whether the trunk group is subjected to terminal hunting or circular hunting.

TK NUM: This field lists the members of each trunk group according to their trunk number. Members are added by entering a valid trunk number (1 to 200) when the cursor is at the TK NUM field on the command line. The trunk group is displayed on the header line.

BAY, **SLT**, **CCT** and **COMMENTS**: These fields are informational fields only. They cannot be modified in this form. When a trunk number is added to the trunk group, the physical identification (BAY, SLT and CCT) and the COMMENTS fields from Form 14 (Non-Dial-In Trunks) or Form 15 (Dial-In Trunks) are automatically displayed.

Softkeys

SMDR/NO SMDR: This softkey enables and disables the outgoing Station Message Detail Recording (SMDR) feature for the Trunk Group. When the SMDR feature is enabled, the header line displays [SMDR] and the softkey displays NO SMDR. Pressing the NO SMDR softkey disables the SMDR feature for that trunk group. The softkey now displays SMDR and the header line displays [NO SMDR].

CIRCULAR/TERMINAL: This softkey selects circular or terminal hunting. When the trunk group is defined as a terminal type, the header line displays [TERM] and the softkey displays CIRCULAR. Pressing the CIRCULAR softkey programs the selected trunk group as a circular type. The header line now displays [CIRC] and the softkey displays TERMINAL. Refer to the Hunt Groups section in Program Features for details on circular and terminal trunk groups.

INSERT: This softkey adds new members to the trunk group. Pressing the INSERT softkey clears the command line and moves the cursor to the TK NUM field. The addition is completed by entering a valid trunk number. The system inserts the addition before the line previously displayed on the command line. Note that this softkey only appears if there is data present in this form.

TK GRP NAME: This softkey specifies a character name for the selected trunk group. Pressing the TK GRP NAME softkey displays the following prompt on the command line: ENTER TRUNK GROUP NAME:. The name specification is completed by entering a character name (a maximum of eight characters). The trunk group name is displayed on the header line beside the trunk group number.

TRUNK GROUP: This softkey selects a trunk group. Pressing the TRUNK GROUP softkey displays the ENTER TRUNK GROUP NUM: prompt on the command line. The selection is completed by entering a valid trunk group number (1 to 50).

The standard softkeys **CANCEL**, **DELETE**, **ENTER**, and **QUIT** are also provided.

Form 17 - Hunt Groups

This form specifies the members of each hunt group by extension or dataset numbers. The system supports a maximum of 99 hunt groups. Each group supports a maximum of 50 members. Only modems can be added to a modem hunt group and modem hunt groups can only contain modems.

Figure: Form 17 Layout

Figure: Recording Hunt Group Options Subform Layout

Figure: Auto Attendant Hunt Group Options Subform Layout

Figure: Station / Set Hunt Group Options Subform Layout

Figure: RABTOS Hunt Group Options Subform Layout

Figure: Ring Group Options Subform Layout

Figure: Ring Groups Review Subform Layout

Field Descriptions

The header line indicates which hunt group is being programmed, its access code, the type of hunting used (terminal or circular) and the type of hunt group (Stn/set, Agent, Recording, Auto Att., RABTOS). Select the OPTIONS softkey to program the hunt group name.

EXT NUM: This field lists the members of each hunt group according to their extension numbers. Valid numbers include extension numbers of rotary dial or DTMF sets and Mitel telephone prime line and dataset numbers.

BAY, **SLT**, **CCT** and **COMMENTS**: These fields are informational fields only. They cannot be modified in this form. When an extension or dataset number is added to a hunt group, the corresponding physical location number (BAY, SLT and CCT) and the COMMENTS fields from Form 09 (Desktop Device Assignments) and Form 12 (Data Assignment) are automatically displayed.

Softkeys

GROUP TYPE: Pressing this softkey displays softkeys which represent the alternative hunt group types. (Softkeys GROUP TYPE and CIRCULAR/TERMINAL do not appear until the hunt group exists. A hunt group is not created until the first member is defined.) Note that the selected group type does not appear on the softkey display.

STN/SET: Pressing this softkey programs the selected hunt group as a Station/Mitel telephone type. A bay/slot/circuit number of any type of line card (including the embedded voice mail card) can be used with this type of hunt group. The header line displays the [STN/SET] prompt to indicate this type of hunt group.

DEFAULT/DEFAULT OFF: Applies to modem pooling, a feature the SX-200 ICP does not support.

RECORDING: Pressing this softkey programs the selected hunt group as a recording type. Only those bay/slot/circuit numbers referring to the ONS/LASS Line Card and Voice mail Card can be used with this type of hunt group. The header line displays [RECORD]. This appears for the ONS/CLASS Line Card only if the hunt group contains industry standard telephones. SUPERSET 401+ telephones cannot be programmed as a member of this hunt group.

Note: Voice mail card numbers cannot be entered unless System Option 134, Recorded Announcement Device is enabled in Form 04.

AGENT: Pressing this softkey programs the selected hunt group as a UCD agent type. The header line displays [AGENT].

CIRCULAR/TERMINAL: This softkey has two functions. It specifies the selected hunt group as a circular or terminal type. Refer to the Hunt Groups section in Program Features for details on circular and terminal hunt groups. When the hunt group is defined as a terminal type, the header line displays [TERM] and the softkey displays CIRCULAR. Pressing the CIRCULAR softkey programs the hunt group as a circular type. The header line now displays [CIRC] and the softkey displays TERMINAL.

AUTO ATT.: Programs the selected hunt group as an automatic attendant hunt group. The header displays [AUTO ATT]. Only ONS line circuits may be entered in this type of hunt group (these would be connected to RADs - see the Automated Attendant Application Package section for further information).

Notes:

1. SUPERSET 401+ telephones cannot be programmed as a member of this hunt group.
2. The system Automated Attendant cannot detect DTMF tones for calls transmitted over IP trunks. As a result, the system will not respond to key presses entered by remote callers.

RABTOS.: Programs the selected hunt group as a Recall Appearance Back To Originating Set (RABTOS) hunt group. The header displays [RABTOS]. In a RABTOS hunt group if all the hunt group members are busy, including the overflow point, the call will recall the Mitel telephone multi-call line appearance that the call originally came in to. Normally, you program Call Center Manager attendants into this type of hunt group.

You must program hunt groups that are used with Mitel Applications Interface (MAI) applications as RABTOS hunt groups. Note that System Option 105 (Mitel Application Interface) must be enabled or the system will not allow you to program a RABTOS hunt group. Also, before you can disable System Option 19, you must first delete all RABTOS hunt groups.

RING ONE/RING ALL:This softkey toggles the setting for ring groups between ringing one member and ringing all members. (Default setting for ring groups is RING ALL.)

INSERT: This softkey adds new members to the selected hunt group. Pressing the softkey clears the command line and moves the cursor to the EXT NUM field. The addition is completed by entering a valid extension number. The system inserts the addition on the line preceding the current line. Note that this softkey only appears if there is data present in this form.

OPTIONS: This softkey allows access to the various Options Subforms, which allows selection of options for the different hunt group types. Value inputs are entered on the command line in the usual manner.

HUNT GROUP: This softkey selects a hunt group. Pressing the HUNT GROUP softkey displays the ENTER HUNT GROUP NUM: prompt on the command line. The selection is completed by entering a valid hunt group number from 1 to 99.

REVIEW: This softkey displays the number of ring groups to which a member belongs. This key only appears for hunt groups of the ring group type.

ACCESS CODE: This softkey assigns an access code for each hunt group. Pressing the ACCESS CODE softkey displays the ENTER NEW ACCESS CODE: prompt on the command line. The access code specification is completed by entering a valid number which must be unique in the database. The access code is displayed on the header line and can be a maximum of five digits.

EXT NUM: Pressing this softkey displays ENTER EXTENSION NUM on the command line. When a valid number is entered, followed by pressing the ENTER softkey, the screen displays the appropriate hunt group and the command line displays the number and its location BAY/SLT/CCT.

DELETE FIELD: Appears in any of the Options Subforms. Deletes programmed data on the command line. Will not appear if there is no data on the command line.

The standard softkeys **CANCEL**, **DELETE**, **ENTER**, and **QUIT** are also provided.

Option Parameters

Overflow: Allows the programmer to assign a destination where calls can be answered when all members of the hunt group are busy.

Record a Call: Maximum Port Usage <1-20> Limits the number of voice mail ports allowed (simultaneously) for the Record a Call feature. A single DSP module provides 16 permanently assigned Record a Call conference bridges. For information about DSP modules and configuration options, see DSP Resources.

Name: Allows the programmer to assign a name to the hunt group. The name can be up to 10 characters long. The first character must not be *, # or a number; the name cannot contain blanks or dashes.

DTRX Enable/Disable: Allows for immediate DTRX access. DTRX DISABLED also provides DTRX access, but only after the interdigit time-out has expired (about 15 seconds). This applies to modem type hunt groups only.

Message Length: Applies only to recording or auto attendant type hunt groups. The length is entered in minutes and seconds. The default value is 10 seconds; the maximum is 4 minutes. The timer should be set at least 3 seconds longer than the actual message length.

Default Destination: This is the destination for incoming calls - if the caller does not dial digits, the call connects to this destination.

Dialing Over Recording: If enabled, allows for dialing digits while listening to a recording.

Fax Destination: This is the destination for incoming Fax calls - when the call is recognized as incoming Fax, the call connects to this destination.

Prefix Digits: These are added to the digit the caller dials in response to (or while listening to) a recorded announcement.

Wait For Resources: This is the time that an incoming caller will wait for connection to a resource. When this times out the call is routed to the default destination.

System Greeting (1-8): Applies to voice mail hunt groups (type: STN/SET). This is the auto attendant greeting that callers to the hunt group hear. Edit the Name to identify the greetings for easier management. For more information, see Recording System Greetings.

RAD Greeting Set (10-39): Applies to voice mail ports that are programmed for RAD use. Defines which set of RAD greetings callers to this Recording-type hunt group hear. For more information, see Recorded Announcement Devices.

Access Code: The pilot number or access code of the hunt group.

Ring All Members Timer: Applies to ring groups that are set to Ring All members; sets the length of time that calls will ring all free members.

Ring One Member Timer: Applies to ring groups that are set to Ring One member; sets the length of time that calls will ring a free member.

Queue Call timer: Used to determine how long a caller is queued to a ring group. When this timer expires, caller is re-routed to overflow point.

Ring All Members: Applies to ring groups that are set to Ring One member; if a call is not answered after it reaches the last member of a ring group, this option controls whether the call will then ring all members. (Default setting is No.)

Tenant: the tenant of the ring group.

Form 18 - Miscellaneous System Ports

This form assigns a physical location (bay, slot, circuit) to four types of devices and three alarms. The devices are a Music-on-Hold (MOH) source, paging equipment, night bells and door opener equipment. The alarms are minor, major and critical. The system supports 25 night bells, nine paging zones, three door openers (four on the CX), and one Music-on-Hold source.

Note: Devices connected to the MOH and Paging ports on the back panel of the SX-200 ICP MX controller are assigned to bay/slot/circuit numbers 1/13/29 and 1/13/30 respectively and cannot be reassigned elsewhere.

Note: If a night bell is a member of a ring group, it can not be deleted using this form.

Figure: Form 18 Layout

Field Descriptions

ENTRY NUMBER: This field lists the entry numbers for the miscellaneous ports. The field cannot be modified.

DESCRIPTION: This field lists the titles assigned to the entry numbers. The field cannot be modified.

BAY, **SLT** and **CCT**: These fields list the bay, slot and circuit numbers of the device being programmed.

SCT: The SCT (subcircuit) field specifies the relay location on each module for each night bell and each alarm circuit. This Figure shows the circuit and subcircuit layout of modules on the Universal Card.

DIR: Enter the direction of the paging device as "1" (one way) or "2" (both way).

PAGER: This field associates a pager number with each night bell, providing integrated paging. Integrated paging causes a warble tone to be produced on the paging system whenever selected night bells are active. The PLID used to program Pager 1 through Pager 9 must be a PLID programmed as a DMP on a DNIC circuit. Once a Pager is programmed, it may be integrated with night bells.

EXT: This field applies only to the night bell entries. The EXTENSION NUMBER field lists the extension numbers (a maximum of five digits) assigned to the night bells. This field links the night bell designations to Form 19 (Call Rerouting Table). This field also links the incoming trunks of Form 14 (Non-Dial-In Trunks) to the night bell designations.

DOOR RELAY: This field associates general-use relays with door phones connected to ONS circuits. The relays connect to solenoids that unlock the door in response key presses at an extension.

Softkeys

ENTRY NUM: This softkey selects an entry number. Pressing the ENTRY NUM softkey displays the ENTER ENTRY NUM: prompt on the command line. The selection is completed by entering a valid entry number (1 to 38). This entry number can now be specified by its bay, slot, circuit and, if applicable, subcircuit numbers.

The standard softkeys **CANCEL**, **DELETE**, **ENTER**, **TOP**, **BOTTOM** and **QUIT** are also provided.

Miscellaneous System Ports Entry Number Designations

Table: Miscellaneous System Ports Entry Number Designations

Entry Number	Title	Entry Number	Title
01	Music-on-Hold	22	Night Bell 9
02	Pager 1	23	Night Bell 10
03	Pager 2	24	Night Bell 11
04	Pager 3	25	Night Bell 12
05	Pager 4	26	Night Bell 13
06	Pager 5	27	Night Bell 14
07	Pager 6	28	Night Bell 15
08	Pager 7	29	Night Bell 16
09	Pager 8	30	Night Bell 17
10	Pager 9	31	Night Bell 18
11	Minor Alarm	32	Night Bell 19
12	Major Alarm	33	Night Bell 20
13	Critical Alarm	34	Night Bell 21
14	Night Bell 1	35	Night Bell 22
15	Night Bell 2	36	Night Bell 23
16	Night Bell 3	37	Night Bell 24
17	Night Bell 4	38	Night Bell 25
18	Night Bell 5	39	Door Relay 1
19	Night Bell 6	40	Door Relay 2
20	Night Bell 7	41	Door Relay 3
21	Night Bell 8	42	Door Relay 4

Form 19 - Call Rerouting Table

This form designates where intercepted and attendant-directed calls will route based on Day Service, Night1 Service and Night2 Service. Dial-in trunks to ACD Paths may also be rerouted as defined in this form. The ACD Path is assigned a tenant number in Form 41.

Refer to the Call Rerouting Options table for Call Rerouting Options. Each Tenant Group requires a Call Rerouting Table. “DIAL 0” calls can be directed to an LDN, Rotary Dial or DTMF set, Mitel telephone or Night Bell extension number, or to an individual console. For further information, refer to the Call Rerouting section under Program Features.

Figure: Form 19 Layout

Field Descriptions

The header line displays the tenant group number being programmed.

DAY: This field designates a directory number for each type of call in day service mode. The directory number is defined in one of the following forms:

Form 07, Console Assignments
Form 08, Attendant LDN Assignments
Form 09, Desktop Device Assignments
Form 17, Hunt Groups
Form 18, Miscellaneous System Ports
Form 41, ACD Paths

If this field is blank, calls do not reroute for features such as no answer or busy forwarding, and reorder tone is heard for features such as vacant number intercept or Do Not Disturb intercept. An ONS Voice mail port with a caller on soft hold will receive dial tone.

N1: This field specifies the extension that calls are routed to during Night1 Service Mode. If this field is blank, the call reroutes to the extension specified in the DAY field.

N2: This field specifies the extension number where calls are routed to during Night2 Service Mode. If this field is blank, the call does not reroute.

Call Rerouting Options

Table: Call Rerouting Options
Station Dial 0 Routing - Any extension which dials the extension general attendant access code (0) is routed here. This rerouting option is based on the caller's tenant.
Priority Dial 0 Routing - Any extension which dials the extension general attendant access code (0) and has COS Option 239 (Priority Dial 0) enabled in its COS is routed here. This rerouting option is based on the caller's tenant.
DID Recall Points on Busy - A DID call reaching a busy extension or hunt group is routed here. This rerouting option is based on the destination tenant.
DID Recall Points on No Answer - A DID call reaching an extension which does not answer is routed here. This rerouting option is based on the destination tenant.
DID Routing for Calls into this Tenant - All DID calls normally routed to extensions are routed here to allow screening of DID calls. This rerouting option is based on the destination tenant.
DID Illegal # Intercept for this Tenant - A DID call to an illegal number is routed here.   This rerouting option is based on the DID's tenant.
DID Vacant Number Routing for this Tenant - A DID call to a vacant number is routed here. This rerouting option is based on the DID's tenant.
DID Attendant Access Night Points - A DID call to the attendant while the system is in Night Service is routed here. This rerouting option is based on the DID's tenant.
Non-Dial-In Trunks Alternate Recall Points - Non-Dial-In trunks that have waited for a busy or non-answering extension for the pre-determined recall time are routed here. This rerouting option is based on the destination tenant.
Dial-In Tie Recall Points on Busy - A Dial-In Tie call reaching a busy extension is routed here. This rerouting option is based on the destination tenant.

Table: Call Rerouting Options
Dial-In Tie Recall Points on No Answer - A Dial-In Tie call reaching an extension which does not answer is routed here. This rerouting option is based on the destination tenant.
Dial-In Tie Routing for Calls into this Tenant - All Dial-In Tie calls normally routed to extensions are routed here to allow screening of Dial-In Tie calls. This rerouting option is based on the destination tenant.
Dial-In Tie Illegal # Intercept for this Tenant - A Dial-In Tie call to an illegal number is routed here. This rerouting option is based on the Tie trunk's tenant.
Dial-In Tie Vacant Number Routing for this Tenant - A Dial-In Tie call to a vacant number is routed here. This rerouting option is based on the Tie trunk's tenant.
Dial-In Tie Attendant Access Night Points - A Dial-In Tie call to the attendant while the system is in Night Service is routed here. This rerouting option is based on the Tie trunk's tenant.
DND Intercept Routing for this Tenant - An extension with Do Not Disturb activated has its incoming calls routed here. This rerouting option is based in the extension's tenant.
Automatic Wake-up Routing for this Tenant - Extensions answering a wakeup call are routed here. Normally the routing point is a recording group. This rerouting option is based on the extension's tenant. Ensure there are sufficient RADs to provide good wakeup performance.
Personal Wake-up Routing for this Tenant - Extensions with personal wakeup calls are routed here. When a personal wakeup timer expires, the attendant console or subattendant telephone receives a callback and the attendant personally provides the wakeup call to the guest. This type of wakeup call can be repeated daily until canceled. Note that speed calls cannot be programmed in this form for the personal wakeup LDN.
UCD/Attendant Recording for this Tenant - Incoming calls destined for UCD agents are routed here when all of the agents are busy. Calls are normally routed to a recording group. This rerouting option is based on the agent/attendant receiving the call.
UCD on Hold Time-out for this Tenant - Incoming calls to busy UCD hunt groups which are not answered after a pre-determined time-out period are routed here.
DISA Day Service Routing for this Tenant - Direct Inward System Access (DISA) calls are routed here. This rerouting option is based on the DISA trunk's tenant.
Station Vacant Number Routing for this Tenant - Any station dialing a vacant number is routed here. This rerouting option is based on the extension's tenant.
CO Line Routing Points on No Answer - Incoming calls to a CO line that are not answered within a programmed time-out period are routed here. The time-out period is specified by the Attendant-Times Recall (No Ans) timer in the class of service of the trunk. This rerouting option is based on the CO line's tenant.
Music Source for This Tenant - Any station put on hold during the programmed times receives its music from this music source. The music source is based on the holder's tenant. If Tenant 1 has a music source but Tenant 2 does not, a caller from Tenant 1 will hear Tenant 1 music while on hold from a set in Tenant 2.
Record a Call Voice mail Destination For This Tenant - This feature (Record a Call Option 87) enables the Record a Call key. The option defines a vm or separate record a call hunt group that stores the Record a Call recordings. If the user decides to share a single VM port for the Record a Call messages

Table: Call Rerouting Options
and voice mail messages, the voice mail hunt group must be programmed with this option in Form 17 (Hunt Groups). Form 19 (Call Rerouting table) only allows programming of valid station/set hunt group numbers (VM group or separate Record a Call hunt group).The first member of the hunt group must be a VM port that has COS 229 (Voice Mail Port) enabled. An invalid entry, in the hunt group, receives a "Access code is not a valid answer point" error message.
Station Illegal Number Routing for this Tenant - Any station or ONS Voice mail port with a caller on soft hold that dials an illegal number is routed here. This rerouting option is based on the extension's tenant.
Speak@Ease Number For This Tenant - This option must be programmed in order for the Speak@Ease softkey to appear on the display set and for the user to have direct access to the Mitel Speech Server. The hunt group number or the speed dial number associated with the hunt group of the DNIC ports connecting to the Speech Server must be programmed. ONS ports may be integrated with the Speech Server, if a speed dial number is programmed for the Speech Server number. N1 and N2 columns are automatically filled with the same number that is in the DAY column. If different numbers are needed for N1 and N2, program these manually.
Phonebook Number For This Tenant: This option provides users with direct access to the voice mailbox directory where they will be able to reach an extension by dialing the person's first or last name rather than their extension number. The hunt group number or the speed dial number associated with the hunt group of the voice mail ports must be programmed in the DAY column. The N1 and N2 columns are automatically filled with the same number the DAY column number. If different numbers are needed for N1 and N2, enter them manually.
Call Forward Busy Number For This Tenant - This option identifies a forwarding destination number for the tenant group with Call Forwarding - Busy. The number programmed in the DAY column will automatically appear in the N1 and N2 columns.
Call Forward No Answer Number For This Tenant - This option identifies a forwarding destination number for the tenant group with Call Forwarding - No Answer.The number programmed in the DAY column will automatically appear in the N1 and N2 columns.
Voice mail Number For This Tenant - This option identifies the number that the system uses for voice mail for this tenant. This number is typically the Master hunt group number for voice mail. The entries for Day, Night 1, and Night 2 must be programmed if voice mail access is required for these service modes. The Single Button Transfer to Voice mail feature is dependant on this CDE Entry.
ONS Notification Number for 911 Calls and Lockouts - This option identifies the number that the system uses for notification of 911 calls and lockouts for this tenant. Only one number is allowed per tenant group. This number must also be an ONS Ring Group master number. The ONS phones that are to be notified must be included in an ONS Ring Group that is associated with this master number using CDE Form 09, Desktop Device Assignments.

Softkeys

SPEED DIAL: Allows a dial 0 call to be routed via speed call and ARS to a Centralized Attendant at another PBX (this is the only intended use for this speed dial softkey). This softkey does not appear when you are programming the voice mail field.

TENANT: This softkey selects a tenant group. Pressing the TENANT softkey displays the ENTER TENANT GROUP NUM: prompt on the command line. The selection is completed by entering a valid number (1 to 25). The system displays the selected tenant group number on the header line.

TENANT NAME: Allows a name to be programmed for the selected tenant group. The name may have a maximum of eight characters.

TENANT CPN: Allows a default Calling Party Number (CPN) to be programmed for the selected tenant group. Once added, the default CPN displays in the form header for the selected tenant group. You can program the same CPN for all extensions in the group, or you can program leading digits that are added to each extension number to create a unique CPN for each extension in the group.

- To program a single CPN for all extensions in the group, enter a number between 10 and 26 digits that does not include * or # characters.
- To program a unique CPN for each extension in the group, enter the leading digits of the CPN followed from three to five * characters in place of the extension number. For example, after programming 6135925*** as the Tenant CPN for group 1, if extension 201 belongs to group 1 but does not have a CPN programmed in Form 54 (Calling Party Number), then CPN 6135925201 will be generated when extension 201 places a call.

Note: If more * characters are entered than the length of the extension, the extra space(s) are padded with zeroes; for example, programming 613592**** as the Tenant CPN for group 1 will yield 6135920201 for extension 201. If fewer * characters are entered than the length of the extension, then only the tail of the extension is used; for example, programming 6135925*** as the Tenant CPN for group 1 will yield 6135925201 for extension 2201.

DELETE CPN: Deletes the default Calling Party Number for the selected tenant group.

The standard softkeys **CANCEL**, **DELETE**, **ENTER**, **TOP**, **BOTTOM**, and **QUIT** are also provided.

Form 20 - ARS: Class of Restriction Groups

Class of Restriction groups together users with the same outside call capabilities. A COR is assigned to each attendant console, station, Mitel telephone and Dial-In Trunk. This form specifies the Class-of-Restriction (COR) Group members. Refer to the figure below for the form layout. The system supports a maximum of 50 COR Groups with up to 25 CORs per group. Each COR Group specifies by COR number those extensions which are restricted from accessing the route.

Figure: Form 20 Layout

Field Descriptions

COR GROUP: This field lists the COR Group numbers from 1 to 50. The COR GROUP field cannot be modified.

COR GROUP MEMBERS: This field lists the separate members of each COR group. The COR Group members must be separated by a space (the → key, TAB key or space bar on the terminal or the → key on the console). Consecutively numbered CORs can be separated by a dash (by pressing the "-" key on the terminal or the ninth softkey on the console).

COMMENTS: This field is reserved for additional data (a maximum of 20 characters). It is stored by the system but not used.

Softkeys

COR GROUP: This softkey selects a Class-of-Restriction (COR) group. Pressing the COR GROUP softkey displays the ENTER COR GROUP NUM: prompt on the command line. The selection is completed by entering a valid number (1 to 50). The selected COR group number is displayed on the command line.

'-': This softkey is available only while a COR group is being edited. Pressing this softkey inserts a dash between a set of consecutive COR group members. It is valid only when it is inserted between consecutive COR group members. For example, 1 2 3 4 5 is equivalent to 1-5. The softkeys **CANCEL**, **DELETE** (this softkey appears after data has been entered), **ENTER**, **TOP**, **BOTTOM**, and **QUIT** are also provided.

Form 21 - ARS: Day Zone Definition

This form defines the day zones for each day of the week. There is a maximum of three day zones. All days of the week must have a zone specification before this form can be saved. Refer to the figure below for the form layout.

Figure: Form 21 Layout

Field Descriptions

DAY ZONE: This field lists the day zones for the system. Pressing the up and down arrow keys select different day zones. This field cannot be modified.

SUN., **MON**., **TUE**., **WED**., **THU**., **FRI**., and **SAT**.: These fields list the zone specification. Pressing the right and left arrow keys (or the TAB key on the CRT terminal) selects separate days. Only one day zone number can be assigned to each day of the week. When a day is assigned to a day zone, the form displays an asterisk (*) in that day field.

Softkeys

ENABLE: This softkey enables the day zone specification for each day. Pressing the ENABLE softkey sets the selected day to the day zone that is displayed on the command line. The form displays an asterisk (*) in the selected day field opposite the chosen day zone.

When you are allowed to assign a day to a day zone, the "enabled" softkey appears. Each day can only be assigned to one day zone.

The standard softkeys **CANCEL**, **ENTER**, and **QUIT** are also provided.

Form 22 - ARS: Modified Digit Table

This form specifies those digits that modify the user-dialed digits. The digit modification prepares the dialed digits for dialing out on certain trunks such as FX, TIE and WATS. Refer to the Automatic Route Selection and Toll Control section for details. Refer to the figure below for the form layout.

Figure: Form 22 Layout

Field Descriptions

ENTRY NUM: This field lists the entry numbers. There is a maximum of 100 entry numbers. The ENTRY NUM field links this form to Form 23, ARS: Route Definition. This field cannot be modified.

QTY TO DELETE: This field lists the quantity of digits that the system removes from the user-dialed digits before outpulsing on a trunk. A maximum of 25 digits can be deleted from each entry number. The digit "1" instructs the system to delete the first dialed digit.

DIGITS TO BE INSERTED: This field lists those digits that the system adds to the user-dialed digits for each entry number. A maximum of 38 digits can be inserted; including any pauses and wait for dial tone symbols. Special number sequences are:

*1 = Pause for 5 Seconds

*2 = Wait for Dial Tone

*3 = Switch to DTMF for Subsequent Digits

*4 = Stop or start displaying modified digits. Modified digits are displayed on the sets and in the SMDR records (SMDR must be enabled in CDE Form 16, Trunk Groups). The first time *4 appears in a digit string, the system stops displaying the following modified digits. The next time *4 appears the system starts displaying the following modified digits. You can repeat *4 in a digit string to stop or start the displaying of digits.

*5 = Pause 10 Seconds

*6 = Insert caller's ID (for analog networking)

*7 = Insert caller's dialed account code (for analog networking)

*8 = Insert PBX node ID number (for analog networking)

*9 = Pause for 1 second.

*01= Identifies call forwarder extension number (for analog networking)

*02= Identifies call forwarding reason (for analog networking)

*03= Inserts CESID (for 911 calls over ISDN Trunks and outbound calls from a DID calling extension--for example, a phone in a retirement residence.)

*04= Inserts Caller's Name (for calls over ISDN Trunks)

Note: The sending of the name of the caller requires the caller to use a set, station, or a trunk with a name. The name of the trunk can be a calling line ID, trunk group name, or trunk name. The trunk being dialed on must be on an ISDN bay.

*05= Inserts Billed Extension (provides the CO with the identity of the calling extension number). If the calling party is a PBX extension, the identity of the PBX extension number is sent. If the call is forwarded (i.e. voice mail), the identity of the last PBX extension number from which the call was forwarded is sent (includes original calls from trunks or extensions). If the call is from a multi-hop call forward scenario, the Advance Analog Networking Billing Number Identifier will be the last forwarding extension number between SX-200 systems or E & M devices.

CPN: This fields specifies which number is assigned to outgoing calls—the calling party's extension number or the calling party's CPN. Press the NO softkey to select the extension number, or YES to select the CPN. Program Calling Party Numbers (CPNs) for extensions on Form 54 - Calling Party Number.

Notes:

1. The sending of the Billed Extension Number is dependant on the purchasable System Option 113 (Centralized/Attendant Voice mail) being enabled.
2. The DISA access code is delivered when the calling party is a DISA DID/Tie Trunk and dials ARS with a *05.

The asterisk (*) character is generated on the Attendant Console and the terminal by pressing the * key. If however, the asterisk character is required in a string of characters, the * key must be pressed twice.

COMMENTS: This field is reserved for additional data (a maximum of 20 characters). It is stored by the system but not used.

Softkeys

ENTRY NUM: This softkey selects an entry number and displays it on the command line. Pressing the ENTRY NUM softkey displays the ENTER ENTRY NUM: prompt on the command line. The selection is completed by entering a valid number (1 to 100).

The standard softkeys **CANCEL**, **DELETE**, **ENTER**, **TOP**, **BOTTOM**, and **QUIT** are also provided.

Form 23 - ARS: Route Definition

This form defines each ARS Route by specifying the trunk group number, COR group number and the modified digit entry. Refer to the figure below for the form layout.

Figure: Form 23 Layout

Field Descriptions

ROUTE NUM: This field lists the route numbers. Note that the ROUTE NUM field cannot be modified. There is a maximum of 200 route numbers.

TRUNK GROUP: This field displays the trunk group number specification (1 to 50) for each route number.

COR GROUP: This field lists the COR group number specification (1 to 50) for each route number. This field links this form to Form 20 (refer to Form 20, ARS: COR Group Definition). Those users with CORs in the listed COR Group number are restricted from using this route. If no COR group number is specified, then all users can use this route.

MOD DIGIT ENTRY: This field lists the entry number specification (1 to 100) for each route number. This field links this form to Form 22, ARS: Modified Digit Table.

COMMENTS: This field is reserved for additional data which is stored by the system but not used (a maximum of 20 characters).

Softkeys

ROUTE NUM: This softkey selects a route number and displays it on the command line. Pressing the ROUTE NUM softkey displays the ENTER ROUTE NUM: prompt on the command line. The selection is completed by entering a valid number from 1 to 200.

SHOW IP: This softkey appears if a trunk group in the form has IP trunking. It displays the routes of IP trunk groups in the Show IP Subform.

The standard softkeys **CANCEL**, **DELETE**, **ENTER**, **TOP**, **BOTTOM**, and **QUIT** are also provided.

Subform 23 - Show IP

This form specifies the routes of IP Trunk Groups in the previous form.

Figure: Show IP Subform Layout

Field Descriptions

ROUTE NUM: This field lists the route numbers. Note that the ROUTE NUM field cannot be modified. There is a maximum of 200 route numbers.

CONCURRENT CALLS: This field defines the maximum number of calls originating from a specified route. Enter a value between 0 and 100.

IP NODE NUM: This field defines the destination IP Node number (as defined in form 48) for this route. Enter a value from 1-255.

COMPRESSION: This field selects G.711 or G.729 compression. Enter "y" to specify G.729 (8 Kbps or enter "n" to to specify G.711 (64kbps) uncompressed voice.

REMOTE PROFILE: This field specifies the profile programming on a remote 3300 ICP. If you are routing to a 3300 ICP, enter the remote profile number here that matches the local profile number. If you are routing to an SX-200 ICP or SX-200 IP Node this information is not used. However the value must still be valid, so use a value of 1 when routing to an SX-200 ICP or SX-200 IP Node.

Softkeys

ROUTE NUM: This softkey selects a route number and displays it on the command line. Pressing the ROUTE NUM softkey displays the ENTER ROUTE NUM: prompt on the command line. The selection is completed by entering a valid number from 1 to 200.

The standard softkeys **DELETE**, **TOP**, **BOTTOM**, and **QUIT** are also provided.

Form 24 - ARS: Route Lists

This form specifies the order that the route numbers are selected. There are 100 route list numbers; each route list number accommodates a maximum of six route numbers. Refer to the figure below for the form layout.

Figure: Form 24 Layout

Field Descriptions

LIST NUM: This field displays the route list numbers. Note that the LIST NUM field cannot be modified. There is a maximum of 100 list numbers.

FIRST, **SECOND**, **THIRD**, **FOURTH**, **FIFTH** and **SIXTH**: These fields specify the route numbers for each route list number. Valid entries are 01 to 200. These fields link this form to Form 23 (ARS: Route Definition).

WT: There are five of these; one for each of the SECOND, THIRD, FOURTH, FIFTH and SIXTH fields. ON in this field indicates that the chosen route number is an expensive route. The system indicates this by providing an audible warning tone when that route is selected.

Softkeys

LIST NUM: This softkey selects a route list number and displays it on the command line. Pressing the LIST NUM softkey displays the ENTER ROUTE LIST NUM: prompt on the command line. The selection is completed by entering a valid route list number (01 to 100).

DELETE/ADD: This softkey appears when the pointer is pointing to data (i.e., data on the command line). This softkey has two functions; it deletes and adds data to the form. Pressing the DELETE softkey removes the data from the selected field. The deletion is completed by pressing the ENTER softkey. If the delete key is pressed while the cursor is in the [FIRST] field, data on that line is deleted. The field is ready for new data and the softkey blanks. When the cursor is at a blank WT field on the command line, this softkey displays ADD. Pressing the ADD softkey enables the warning tone. The WT field displays ON and the softkey now displays DELETE.

The standard softkeys **CANCEL**, **ENTER**, **TOP**, **BOTTOM**, and **QUIT** are also provided.

Form 25 - ARS: Route Plans

This form assigns the route lists to the time and day zones. There are 18 time zones (six for each day zone). Each time zone has a start time that can be set by the installer. The last specified start time creates a time period from its start time to the first specified start time of that day zone. If a time zone has no assigned route list number, then all calls during that time period are restricted from this route plan. There are a maximum of 50 route plans. Refer to the figure below for the form layout.

Figure: Form 25 Layout

Field Descriptions

TIME ZONE: This field lists the six time zones for each day zone. The TIME ZONE field cannot be modified.

START HOUR: There are three of these fields (one for each day zone). The START HOUR field specifies the starting time of each time zone. The time is represented by two digits in 24 hour format. For example, 18 represents 18:00.

ROUTE LIST: There are three of these fields (one for each day zone). The ROUTE LIST field displays the route list numbers and links this form to Form 24, ARS: Route Lists. Valid entries are 01 to 100.

Softkeys

ROUTE PLAN: This softkey has two functions. It displays the selected route plan number and enables the user to select an alternate route plan. Pressing the ROUTE PLAN softkey displays the following on the command line: CURRENT ROUTE PLAN: XX ENTER ROUTE PLAN NUM: ROUTE PLAN: where XX is route plan number 01 to 50. The route plan selection is completed by entering a new route plan number.

The standard softkeys **CANCEL, DELETE**, **ENTER**, and **QUIT** are also provided.

Form 26 - ARS: Digit Strings

This form (and its nested form) link the digits dialed by the user to the appropriate route, route list or route plan. It selects the relevant route (if there is only one route), or route list (if there is more than one route and the time of day is not important), or route plan (if the choice of routes vary with the time of day) by the user-dialed digits. Maximum number of entries in this form is 350. Refer to the figure below for the form layout. Refer to the Automatic Route Selection and Toll Control section for details.

Figure: Form 26 Layout

Field Descriptions

LEADING DIGITS: This field displays the first digits of each digit string for digit analysis. The maximum number of digits in this field is five.

RETURN DIAL TONE: If this field displays YES, the system provides a dial tone after the leading digits have been dialed. Alternately, the system does not provide a temporary dial tone when NO is displayed in this field. Refer to the Automatic Route Selection and Toll Control section for details.

RESTRICTED COR GROUP: This field lists the COR group which cannot dial the specified leading digit(s). If this field is left blank, then every COR group can access the specified leading digit(s).

Softkeys

YES/NO: This softkey has two functions; it enables or disables system dial tone for each entry. Pressing the YES softkey enables the system dial tone when that leading digit is accessed. The RETURN DIAL TONE field displays YES and the softkey now displays NO. Pressing the NO softkey disables the system dial tone when that leading digit is accessed. The RETURN DIAL TONE field displays NO and the softkey displays YES again. Note that this softkey only appears when the cursor is in the "RETURN DIAL TONE" field.

INSERT: This softkey adds new entries to the form. Pressing the INSERT softkey clears the command line and moves the cursor to the LEADING DIGITS field. The addition is completed by entering the new data for each field and pressing the ENTER softkey. Note that the system inserts the addition after the line that was displayed on the command line.

LEADING DIG: This softkey selects an entry in the LEADING DIGITS field. Pressing the LEADING DIG softkey displays the ENTER LEADING DIGITS: prompt on the command line. The selection is completed by entering a valid number. Valid entries are the digits 0 through 9 or an asterisk (*).

SHOW STRINGS: Pressing this softkey accesses the nested form for any defined leading digits entry. Refer to the ARS: Nested Digit Strings subform.

The standard softkeys **CANCEL**, **DELETE**, **ENTER**, ****MORE**** and **QUIT** are also provided.

Programming Tip

When programming ARS, it is recommended that Quantity to Follow always be zero. This is accomplished by padding Digits to Analyze with Xs. For example, instead of entering digits as shown in Example A, enter them as shown in Example B:

Example A:

DIGITS TO ANALYZE	QUANTITY TO FOLLOW
613	7
1800	7
1	10
592	4

Example B:

DIGITS TO ANALYZE	QUANTITY TO FOLLOW
613XXXXXXX	0
1800XXXXXXX	0
1XXXXXXXXXX	0
592XXXX	0

Subform 26 - Digit Strings

This form is reserved for specifying subsequent digits for each entry in the LEADING DIGITS field of the previous form. It can only be accessed from the previous form. Refer to the figure below for the form layout. Refer to the Automatic Route Selection and Toll Control section for details.

Figure: Digit Strings Subform Layout

Field Descriptions

DIGITS TO BE ANALYZED: This field displays those digits for digit analysis. Digit analysis is required so that the appropriate route, route list or route plan can be selected. The total number of digits in this field, the number of digits in the QTY TO FOLLOW field plus the digits in the LEADING DIGITS field in the previous form (refer to the top level Digit Strings form) cannot exceed 26.

QTY TO FOLLOW: This field lists the number of digits that the user dials AFTER the analyzed digits. The 'Unknown' prompt in this field indicates that the number of subsequent digits is unknown to the system.

DESIGNATION: This field specifies which digit string entries require LONG DISTANCE, LOCAL, or EMERGENCY management. Several features (account codes, Hotel/Motel, etc.) use this field to control access to ARS. The default condition is LOCAL. The EMERGENCY designation allows the call to proceed regardless of the restrictions applied to the originating station (see Conditions in Emergency Call Handling for exceptions). Emergency call digits do not have to be 911 calls. This field serves areas that do and do not have 911 service available or areas that have another set of digits for their local emergency code. The EMERGENCY designation will not cause 911 alarms to be generated to the attendant console and will not send CESID information to the network.

TERM TYPE AND NUM: This field specifies where the digit string terminates. If there is only one route, then ROUTE is selected. If there is more than one route, but the time of day is not important, then LIST is selected. If the choice of routes vary with the time of day, then PLAN is selected.

Softkeys

LOCAL: This softkey only appears when the cursor is at the DESIGNATION field. Pressing the LOCAL softkey indicates to the system that this entry does not require “long distance” management. The default condition is not long distance as indicated by LOCAL in the DESIGNATION field.

LONG DIST: This softkey only appears when the cursor is at the DESIGNATION field. Pressing the LONG DIST softkey provides "long distance" management.

EMERGENCY: This softkey only appears when the cursor is at the DESIGNATION field. Pressing the EMERGENCY softkey provides emergency call management. The EMERGENCY designation allows the call to proceed regardless of the restrictions applied to the originating station. (see Conditions in Emergency Call Handling for exceptions). Emergency call digits do not have to be 911 calls. This designation serves areas that do and do not have 911 service available or areas that have another set of digits for their local emergency code.The EMERGENCY designation will not cause 911 alarms to be generated to the attendant console and will not send CESID information to the network.

UNKNOWN: This softkey appears only when the cursor is at the QTY TO FOLLOW field. Pressing the UNKNOWN softkey indicates to the system that the quantity of dialed digits AFTER the analyzed digits is unknown.

INSERT: This softkey adds new entries to the form. Pressing the INSERT softkey clears the command line and moves the cursor to the DIGITS TO BE ANALYZED field. The addition is completed by entering the new data for each field and pressing the ENTER softkey. Note that the system inserts the addition one line after the line that was displayed on the command line. The system automatically places all inserted or added strings in numerical ascending order with relation to existing strings.

N0X: This softkey functions as a wild card sequence, where N is any digit from 2 to 9. It represents half of the area codes in North America. Pressing this softkey displays N0X in the DIGITS TO BE ANALYZED field. Note that this softkey can only be pressed at the beginning of a digit string.

ROUTE: This softkey only appears when the cursor is at the TERM TYPE AND NUM field. Pressing the ROUTE softkey terminates that entry at a route (the route number must still be defined). ROUTE appears in the TERM TYPE AND NUM field.

X: This softkey functions as a wild card digit; it represents any digit from 0 to 9. Pressing this softkey displays X in the DIGITS TO BE ANALYZED field. Note that this softkey can only be pressed at the end of a digit string.

PLAN: This softkey only appears when the cursor is at the TERM TYPE AND NUM field. Pressing the PLAN softkey terminates that entry at a route plan (the number must still be defined). PLAN appears in the TERM TYPE AND NUM field.

FIND STRING: This softkey selects an entry in the DIGITS TO BE ANALYZED field. Pressing the FIND STRING softkey displays the ENTER DIGIT STRING: prompt on the command line. The selection is completed by entering a valid digit string. Note: The entered digit string does not have to be an exact match; the system accepts subsets of digit strings and moves the cursor to the closest entry.

N1X: This softkey functions as a wild card sequence, where N is any digit from 2 to 9. Pressing this softkey displays N1X in the DIGITS TO BE ANALYZED field. Note that this softkey can only be pressed at the beginning of a digit string.

LIST: This softkey appears only when the cursor is at the TERM TYPE AND NUM field. Pressing the LIST softkey terminates that entry at a Route List (the number must still be defined). LIST appears in the TERM TYPE AND NUM field.

NXX, **1N0X**, **0N1X**, **0N0X:** These strings are used for area codes created under the old North American Numbering Plan, and where dialing is preceded by a 1 or 0 (long distance access code). These softkeys can only be pressed at the beginning of a digit string. They are accessed by pressing the ** MORE ** softkey once.

1N1X, **0NXX**, 1**NXX:** These strings are used for area codes and office codes that follow the new North American numbering plan, where dialing is preceded by a 1 or 0 (long distance access code) and the second digit can be any value from 0 to 9. These wildcard sequences determine whether a 0 or a 1 as well as a valid area or office code has been dialed. They are accessed by pressing the ** MORE ** softkey twice.

10XXX0N0X, 10XXX0N1X, 10XXX1N0X, 10XXX1N1X, 10XXX0, 10XXX1: These wildcard sequences, designed for the call aggregator market (i.e. hotels, motels, hospitals, universities), prevent unauthorized calls from being billed to the originating line, while allowing customers access to the long distance carriers of their choice. These softkeys can only be pressed at the beginning of a digit string. Press the ** MORE ** softkey again to access 10XXX0N0X, 10XXX0N1X, 10XXX1N0X and 10XXX1N1X. Press ** MORE ** softkey additional times to access 10XXX0 and 10XXX1.

101XXXX0N0X, 101XXXX0N1X, 101XXXX1N0X, 101XXXX1N1X, 101XXXX0, 10XXXX1: These wildcard sequences, designed for the call aggregator market (i.e. hotels, motels, hospitals, universities), prevent unauthorized calls from being billed to the originating line, while allowing customers access to the long distance carriers of their choice. These softkeys can only be pressed at the beginning of a digit string. Press the ** MORE ** softkey additional times to access 101XXXX0N0X, 101XXXX0N1X, 101XXXX1N0X and 101XXXX1N1X. Press ** MORE ** softkey additional times to access 101XXXX0 and 101XXXX1.

Either wildcard sequences 10XXX or wildcard sequences 101XXXX should be programmed on a PBX, but not both.

The standard softkeys **CANCEL**, **DELETE**, **ENTER**, and **QUIT** are also provided.

Form 27 - ARS: Maximum Dialed Digits

This form specifies the maximum number of dialed digits allowed for each Class of Restriction. The purpose of this form is to accommodate countries with open numbering plans, where it is generally not possible to determine from the leading digits the number of digits to follow. For North America, the specified default value of Unlimited applies. Refer to the Automatic Route Selection and Toll Control section for detailed information and additional CDE considerations if a value other than 'Unlimited' is chosen. See the figure below for the form layout.

Figure: Form 27 Layout

Field Descriptions

COR: This field lists the COR (1 → 25). **Note**: The COR field cannot be modified.

Maximum Number of Dialed Digits: This field lists the allowed number of dialed digits, 1 to 25, plus the default value of Unlimited. Twenty-six is equivalent to UNLIMITED. Therefore, when 26 is entered the value UNLIMITED is displayed.

Softkeys

UNLIMITED: Pressing the UNLIMITED softkey enters the default value of 'Unlimited' in the Maximum Number of Dialed Digits field. If the cursor is positioned at the default value, 'Unlimited' is not displayed.

COR: When this softkey is pressed the user is prompted with ENTER COR NUMBER. After entering the COR number and pressing the ENTER softkey, the cursor is positioned to the COR specified. If the COR selected is out of range, the message "The value xx is outside valid range for COR (1 → 25)" is displayed.

The standard softkeys **CANCEL**, **ENTER**, **TOP**, **BOTTOM**, and **QUIT** are also provided.

Form 28 - Form Access Restriction Definition

This form specifies the level of access for the various CDE forms. There are five levels of access; the installer level has the highest degree of access and the attendant level has the lowest. Each form is defined as read only, read/write or no access. When the system is first initialized, the installer level and MAINT1 level have read/write access for each form; the rest of the levels default to no access. Note that at each level of access the user can only modify those forms plus the forms at the lower levels of access. For example, the user at the supervisor level can only modify those forms at SUPERVISOR and ATTENDANT levels. Refer to the figure below for the form layout.

Figure: Form 28 Layout

Field Descriptions

FORM NAME: This field lists all the form numbers and names of the CDE package. The FORM NAME field cannot be modified. If the FORM NAME is listed as RESERVED, the form is not available.

INST, **MAINT1**, **MAINT2**, **SUPER** and **ATT**: These fields represent the five levels of access and list the access type (read only, read/write or no access) for each form. The selected level of access is not displayed on the softkeys. The INST field cannot be modified.

Softkeys

READ ONLY: Pressing the READ ONLY softkey restricts the user to viewing the selected form; no modifications can be made. The R prompt appears beside the selected form and under the selected access level.

READ/WRITE: Pressing the READ/WRITE softkey enables the user to view and modify the selected form. The form displays the R/W prompt to indicate this state.

NO ACCESS: Pressing the NO ACCESS softkey restricts form access. The form displays the 'none' prompt to indicate this state.

SET PASSWORD: This softkey changes the password for Customer Data Entry. Pressing the SET PASSWORD softkey shows a new softkey display (ATTENDANT, SUPERVISOR, MAINT2, MAINT1, and INSTALLER) and the following prompt: SELECT LEVEL OF ACCESS:. The user cannot change the password for a level of access higher than the current one.

For example, a user logged on as MAINT2 cannot change the password for MAINT1. After the user selects a level of access, the system prompts ENTER XXXXXXXX NEW PASSWORD:. When the new password is entered, the system prompts ENTER XXXXXXXX NEW PASSWORD TO VERIFY:. When changing the current level's password, the system first prompts ENTER XXXXXXXX OLD PASSWORD:, where XXXXXXXX is the selected level of access.

ALL FORMS: This softkey allows a user to change the level of access to all forms for a lower level user. For example, INSTALLER may use this to change the access for MAINT1, and MAINT2 may use this to change the access for SUPERVISOR.

FORM NUM: This softkey selects a form by number. Pressing the FORM NUM softkey displays the ENTER FORM NUMBER: prompt on the command line. The selection is completed by entering a valid form number (1 to 35).

The softkeys **CANCEL**, **ENTER**, **TOP**, **BOTTOM**, and **QUIT** are also provided.

Form 29 - DTE Profile

In order for the data transceiver (DTRX) to communicate with the attached data terminal equipment (DTE), it is necessary to specify the programmable options which define the characteristics of the terminal. The data transceiver circuit provides data devices with the ability to dial a destination via a keyboard. The DTE profile form provides 25 programmable profiles.

Figure: Form 29 Layout

Field Descriptions

PROFILE NUMBER: This field lists all the profile numbers 01 to 25. The PROFILE NUMBER field cannot be modified.

NUMBER OF DATA SETS ASSIGNED: This field lists the number of data sets assigned to each profile number.

COMMENTS: This field is reserved for additional data (a maximum of 20 characters). It is stored by the system but not used.

Softkeys

PROFILE NUM: The PROFILE NUM softkey allows a user to select a device by number. Pressing this softkey displays the ENTER PROFILE NUM: prompt. When the number has been entered, the command line updates and the line pointer moves to profile number. The selection is completed by pressing the ENTER softkey.

SEL. OPTION: A new form is displayed when the SEL. OPTION softkey is pressed. Refer to the Options Subform for Form 29.

REVIEW: A new form is displayed when the REVIEW softkey is pressed. Refer to the Review Subform for Form 29. This softkey appears when the number of datasets is greater than 0.

The softkeys **CANCEL**, **ENTER**, **TOP**, **BOTTOM**, and **QUIT** are also provided.

Subform 29 - Options

The system displays this form when the programmer presses the SEL. OPTION softkey in Form 29, Data Terminal Equipment Profile. The options displayed are for the profile number that was on the command line of Form 29.

Figure: Options Subform Layout

Field Descriptions

The header displays the profile number.

VALUE: The VALUE field displays the current setting for each option.

Softkeys

Some of the softkeys displayed are the alternative settings for the option displayed on the command line. Refer to *Parameters*, for details. The following softkeys are continuously present: **TOP**, **BOTTOM**, and **QUIT**.

Parameters

Terminal Type: This parameter is used to determine how the delete character is defined on a video or teleprinter terminal. A teleprinter terminal displays a "/" when the delete character is used and the video terminal transmits a <backspace>, <space>, <backspace>.

Language: Commands and responses for the DTRX can be English or French.

DTRX Echoplex: This parameter determines whether the DTRX (data transceiver), will echo back transmitted characters to the originating station. The exception is programmed keys in the DTE and data circuit descriptor forms.

Editing: Enabled, this option provides the user with editing function DTRX delete character, and DTRX display line.

Editing Character Delete: An ASCII code in decimal form is used to define a character as a delete key. If echoplex and DTRX editing options are enabled, a "/" appears as the delete character on a teleprinter and a <backspace>, <space>, <backspace>, is transmitted back to a video terminal.

Editing Line Display: An ASCII code in decimal form is used to define a character as a display line key. When DTRX editing is enabled and the ASCII code for display line is transmitted, the current DTRX command line and the input digits are displayed on a new line.

Inject <LF> After <CR>: This field is used to accommodate terminal variations in the handling of carriage returns (<CR>). Some terminals automatically insert line feed (<LF>) after carriage return. While connected to the data transceiver the following options are available:

NEVER: No linefeed insertions after <CR> detected.

FROM DTE: Insert <LF> after <CR> from DTE if the echoplex feature is enabled. The <CR> and <LF> is returned to the Data Terminal Equipment by the Data Transceiver after a <CR> was received from the DTE. All messages originated by the DTRX (data transceiver) would only have a <CR>. The DTE would typically provide this.

FROM SYSTEM (DTRX): Insert <LF> after <CR> from DTRX. All messages originated by the PBX that have a <CR> will have a <LF> injected. This does not include <CR> which are echoed back to the DTE. This option would be used if the DTE provided local echoing of characters transmitted.

ALWAYS: The <LF> will be injected if the <CR> is originated from the DTE or DTRX.

Number of Pads after <CR>: This field is used for terminals that require delays after a carriage return before receiving printable characters (printers with small or no buffers). Values entered range from 0 to 7. This option is valid only if echoplex is enabled.

Number of Pads after <LF>: This field is used for terminals that require delays after a line feed return before receiving printable characters (printers with small or no buffers). Values entered range from 0 to 7. This option is valid only if echoplex is enabled.

DTRX Inactivity Timer: This field specifies the length of time between the last character received or transmitted from a data device and the DTRX being dropped. Values range between 1 to 60 seconds; default is 10 seconds.

Subform 29 - Review List

This form appears when the REVIEW softkey is pressed in Form 29, DTE Profile. The form provides a list of users of a particular profile identified by their physical location.

Figure: Review List Subform Layout

Field Descriptions

The header line displays the profile number.

BAY, **SLT**, **CCT**, **SCT**: These fields list the bay, slot, circuit and subcircuit of the device programmed. These fields cannot be modified.

COMMENTS: This field displays the comments for each device from the Data Assignment form. It cannot be modified.

Softkeys

PROFILE NUM: When this softkey is pressed the command line displays ‘ENTER PROFILE NUM:’. After entering a valid number and pressing the ENTER softkey, the screen will display the list of users of that DTE profile, if any.

The standard softkeys **CANCEL**, **ENTER** and **QUIT** are also provided.

Form 30 - Device Interconnection Table

This form specifies which devices may be connected. See the figure below for the form layout. The system allows a maximum of 25 devices.

Figure: Form 30 Layout

Field Descriptions

Initially, the system interconnects all devices except trunks. The asterisk (*) character indicates the device the row represents is allowed to connect to the device the column represents. When the system inhibits device interconnection, it is indicated by the period (.) character. The device numbers are listed in the header line and the first column. The devices these numbers represent are listed in the nested REVIEW form.

Softkeys

INTERCON NUM: The INTERCON NUM softkey allows a user to select a device by number. Pressing this softkey displays the ENTER INTERCONNECT NUM: prompt. Entering the interconnect number (1 to 25) displays that device with a series of ‘*’ characters (allow interconnection) and ‘.’ characters (disallow interconnection). Cursor movement on the command line is controlled by the right and left cursor control keys.

DISALLOW/ALLOW: This softkey has two functions; it enables and disables interconnection between devices. Pressing the DISALLOW softkey disables the interconnection between those two devices unidirectionally. For example, when modifying connections for device 5 (the command line displays line 5) and the DISALLOW softkey is pressed when the cursor is under the sixth column, then device 5 cannot communicate with device 6. However, device 6 can still communicate with device 5. Total interconnection is inhibited only when a ‘.’ (disallow) character is inserted at row 6 (device 6) under the fifth column (device 5). The softkey now displays the ALLOW prompt. Pressing the ALLOW softkey enables the unidirectional interconnection between the selected devices; the ‘*’ character replaces the ‘.’ character.

REVIEW: Pressing the REVIEW softkey displays a new form (refer to Review List Subform for Form 30). This form lists all the device types.

The standard softkeys **CANCEL**, **ENTER**, **TOP**, **BOTTOM**, and **QUIT** are provided.

Subform 30 - Review List

This form appears when the REVIEW softkey is pressed in the Device Interconnection Table Form. When entered from Form 30, this form displays a list of the device types. Refer to the figure below for the form layout. Note that the data in this form cannot be modified.

Figure: Review List Subform Layout

Field Descriptions

ENTRY NUM: This field lists the entry numbers for the device types. There is a total of 25 entry numbers. The ENTRY NUM field cannot be modified.

DEVICE TYPE DESCRIPTION: This field lists the titles assigned to the entry numbers. The DEVICE TYPE DESCRIPTION field cannot be modified. The last 13 entries are reserved for future use.

INTERCONNECT NUM: This field lists all the interconnect numbers; it links this form to the previous form (refer to Form 30, Device Interconnection Table). The interconnect numbers range from 1 to 25. The INTERCONNECT NUM field cannot be modified.

ENTRY NUM: This softkey selects an entry number. Pressing the ENTRY NUM softkey displays the ENTER ENTRY NUM: on the command line. The selection is completed by entering a valid entry number (1 to 25).

INTERCON NUM: This softkey selects a device type by its interconnect number. Pressing the INTERCON NUM softkey displays ENTER INTERCONNECT NUM: on the command line. The selection is completed by entering a valid interconnect number (1 to 25).

The standard softkeys **CANCEL**, **ENTER**, **TOP**, **BOTTOM**, and **QUIT** are provided.

Form 31 - System Abbreviated Dial Entry

This form specifies System Abbreviated Dial numbers. Refer to Abbreviated Dial for details. See the figure below for the form layout.

Figure: Form 31 Layout

Field Descriptions

INDEX NUMBER: This field lists the index numbers: there is one for each entry in the form. The System Abbreviated Dial Access Code followed by the index number forms the abbreviated dial number. Each index number can be a maximum of three digits.

DIGIT STRING: This field lists the digit strings: there is one for each Index number. The digit string can be a maximum of 26 digits for a non-private number, 25 digits for a private number. The following special characters may be entered:

*3 =	Wait for user to manually insert digits (2 digits expected)
*4 =	Send call forward condition (for ONS Voice mail feature)
*6 =	Insert caller's ID (for ONS Voice mail feature)
*8 =	Send digits of calling party (for ONS Voice mail feature)
*9 =	Pause for 1 second
**=	DTMF digit *
# =	DTMF digit #

PAUSES: Pauses may be entered in speed dial numbers by pressing a red HOLD key while programming the speed dial number. Each pause is of one second duration. After a pause, further digits are outpulsed in DTMF.

PRIVATE: This field specifies which entries are private (as indicated by the PRIVATE prompt) and which are non-private (as indicated by a blank).

Softkeys

FIND INDEX: This softkey selects an index number. Pressing the FIND INDEX softkey displays ENTER INDEX NUM: on the command line. The selection is completed by entering a valid index number (a maximum of three digits).

INSERT: This softkey adds new digit strings to this form. Pressing the INSERT softkey clears the command line and moves the cursor to the INDEX NUMBER field. The addition is completed by entering a valid index number and digit string. The system inserts the addition in ascending numerical order according to the INDEX NUMBER field. Note that if there is no data in this form or if the line pointer is pointing to the last line of data, then this softkey does not appear.

PRIVATE/NON-PRIVATE: This softkey has two functions. Pressing the PRIVATE softkey sets the selected digit string entry to private; the PRIVATE prompt appears in the PRIVATE field and the softkey now displays the NON-PRIVATE prompt. Pressing the NON-PRIVATE softkey sets the selected digit string to non-private; the PRIVATE field blanks and the softkey displays the PRIVATE prompt once again.

The softkeys **CANCEL**, **ENTER**, **DELETE**, **TOP**, **BOTTOM**, and **QUIT** are provided.

Form 32 - CDE Data Print

This form lists all the customer data entry options that can be sent to an output device, such as a printer or a terminal. The options are selected by Print Option Number. Note that this form cannot be modified. Refer to the Customer Data Entry Forms table for a complete list of the form numbers and to the figure below for the form layout. Print Options with "*" are subforms of the preceding form.

Figure: Form 32 Layout

Field Descriptions

PRINT OPTION: This field lists the print option numbers for the CDE options that can be sent to an output device (a terminal or a printer). The PRINT OPTION field cannot be modified.

CDE DATA PRINT: This field lists the form names associated with the print option numbers. Print options with the form name RESERVED are not available. The CDE DATA PRINT field cannot be modified.

Softkeys

PRINT ALL: This softkey transmits the contents of all the CDE Data Print option ranges to an output device (printer or terminal). Pressing the PRINT ALL softkey blanks the softkey display (with the exception of the QUIT softkey).

Note: The CDE Form printouts do not include system IP information. To view this information, go to Form 47 (IP Networking), Subform 01 (System IP).

Further CDE is prohibited while PRINT ALL is in effect. The command line sequentially prompts the user to enter the desired ranges for options. The softkey ENTER must be pressed after each entry. The Data Print options are listed below:

1. PRINT FROM COS START: TO COS END
2. PRINT FROM TRK CCT DESC START: TO TRK CCT DESC END
3. PRINT FROM DATA CCT DESC START: TO DATA CCT DESC END
4. PRINT FROM DTE PROFILE START: TO DTE PROFILE NUM. END
5. PRINT FROM T1 LINK DESC. START: TO T1 LINK DESC. END.

When printing starts, the command line displays: CDE DATA PRINT IN PROGRESS and the only softkey available is ABORT.

PRINT OPTION 75 (Automatic Route Selection) is not included with the PRINT ALL command.

Note: The system does not generate an error message if the specified printer is not operational.

DISPLAY ALL: This softkey is used at a remote terminal, when the user wants to print the contents of all the CDE Data Print options. The DISPLAY softkey is equivalent to the PRINT ALL softkey; causing the selected print option to be executed but with the data reflected back to the terminal. This softkey is dependant on Feature Level 2.

ABORT: This softkey appears whenever a print is in progress. Pressing this softkey cancels the current active printout.

PRINT OPTION: This softkey selects a print option number. Pressing the PRINT OPTION softkey displays ENTER PRINT OPTION: on the command line. The selection is completed by entering a valid print option number. The command line displays the selected option.

PRINT: This softkey transmits the contents of the selected print option to an output device (printer or terminal). Pressing the PRINT softkey blanks the softkey display (except for the ABORT softkey) for the duration of the print operation. When completed, the system displays CDE DATA PRINT OPTION XX HAS COMPLETED PRINTING where XX is the print option number. The softkey display returns to the original format.

DISPLAY: This softkey is used at a remote terminal, when the user wants to print the contents of a selected print option. The DISPLAY softkey is equivalent to the PRINT softkey; causing the selected print option to be executed but with the data reflected back to the terminal. This softkey is dependant on Feature Level 2.

The standard softkeys **CANCEL**, **ENTER**, **DELETE**, **TOP**, **BOTTOM**, and **QUIT** are also provided.

Form 33 - Account Code Entry

This form specifies the account codes in the database. If the account codes are verified, they are stored in the account code database. Refer to the figure below for the form layout. Refer also to the Station Message Detail Recording (SMDR) section. This form can only be accessed if System Option 05, Verified Account Codes is enabled.

Figure: Form 33 Layout

Field Descriptions

ACCOUNT CODE: This field lists the stored account code entries. The length of the account code is specified in Form 04, System Options/System Timers (System Option 55, Account Code Length). Depending on the status of System Option 55, account codes can be variable from 1 to 12 digits or fixed to a set value from 4 to 12 digits.

COS: This field specifies a COS number for each account code entry. This COS number replaces the caller's COS number for the duration of the call. When the call is finished, the system restores the caller's original COS number. If this field remains blank, then the caller's original COS number is used for the call.

COR: This field specifies a COR number for each account code entry. The system replaces the caller's COR number with this COR number for the tenure of the call. Upon call completion, the caller's original COR number is restored. If there is no specified COR number, then the caller's original COR number is used for the call.

ACTIVE: This field specifies which account codes can be accessed (as marked by the ACTIVE prompt) and which account codes are denied access (as marked by the INACTIVE prompt).

Softkeys

ACTIVE/INACTIVE: This softkey has two functions. Pressing the ACTIVE softkey activates the selected account code entry so that the account code can be used. The ACTIVE prompt appears in the ACTIVE field and the softkey now displays the INACTIVE prompt. Pressing the INACTIVE softkey sets the selected Account Code entry to 'Inactive' and the account can no longer be accessed. The ACTIVE field now displays the INACTIVE prompt and the softkey displays the ACTIVE prompt once again. Note that this softkey only appears when the cursor is in the ACTIVE field.

NULL COS/NULL COR: This softkey has two functions; it deletes selected COS and COR number entries. When the cursor is in a COS field which has a COS number, the NULL COS prompt appears on the softkey display. Pressing the NULL COS softkey erases the data in the COS field; the NULL COS prompt disappears only when the cursor moves to the next field. Similarly, when the cursor is in a COR field which has a COR number, the NULL COR prompt appears on the softkey display. Pressing the NULL COR softkey erases the data in the COR field; the NULL COR prompt disappears only when the cursor moves to the next field.

INSERT: The INSERT softkey adds a new account code to this form. Pressing this softkey opens a window with a clear command line and moves the cursor to the ACCOUNT CODE field. The new account code is programmed by entering an account code value, a COS number and a COR number (if required). The system inserts the new account code in its appropriate sequential position. The line pointer now points to the new account code. Note that if there is no data in this form or if the line pointer is pointing to the last line of data, then this softkey does not appear.

DELETE RANGE: Pressing the DELETE RANGE softkey displays the FIRST ACCOUNT CODE TO DELETE: prompt on the command line. After an Account Code has been entered, the display returns the LAST ACCOUNT CODE TO DELETE: prompt. The deletion is completed by entering an Account Code. All Account Code entries between and including these specified Account Codes are removed from the form. If an invalid account code is entered the system will display 'Non-existent account code value has been entered'. Pressing CANCEL followed by QUIT returns the display without any modifications.

FIND: This softkey selects an account code and appears only when there is an account code in the form. Pressing the FIND softkey displays the ENTER ACCOUNT CODE: prompt on the command line. The selection is completed by entering a valid account code.

The standard softkeys **CANCEL**, **ENTER**, **DELETE**, **TOP**, **BOTTOM**, and **QUIT** are also provided.

Form 34 - Directed IO

This form allows the user to specify the location of the system printer ports, designate the type of printout for each printer, and define whether the printout is guaranteed or not (will or will not print). Data outputs such as Traffic Measurement, SMDR, Hotel/Motel can be routed to any data port with an asynchronous dataset or to an IP socket which a computer can connect to via telnet. If no new point is specified, printouts continue to default to the system printer RS-232 port (MX only). If the printer specified is currently active, then any request to print is queued. The system can support 7 different printers.

Note: When the Printout type of an IP socket extension is changed, the Telnet session needs to be disconnected, and then reconnected to restore functionality.

Figure: Form 34 Layout

Field Descriptions

EXT NUM: This field lists printer ports and extension numbers of programmed datasets. This form is linked to Form 12, Data Assignment. On the AX and CX controller, the printer port extension is reconfigured for SMDR and CDE Data Print output to IP socket 61328.

PRINTOUT: This field lists the specified data outputs, such as Traffic Measurement, SMDR, Hotel/Motel, DATA SMDR, etc., for each programmed printer. Values in this field are entered through softkey commands. Note that the PMS (Property Management System) softkey is displayed only if the PMS system option is enabled.

Program printer ports only if there is a dedicated printer.

PRINTOUT TYPE: This field lists the type of printout provided for each data output such as Autoprint, Directed and Monitor. Values in this field are entered through softkeys. The softkeys displayed will depend upon the data output programmed in the PRINTOUT field. The Available Softkeys table below lists available softkeys as determined by the PRINTOUT field.

Table: Available Softkeys

Application	Printout Type Options
Maintenance Logs	AUTOPRINT, DIRECTED, MONITOR
Traffic Measurement	AUTOPRINT, DIRECTED
IP Traffic Measurement	DIRECTED
SMDR	AUTOPRINT
CDE Data Print	DIRECTED
Hotel/Motel Wakeup	AUTOPRINT
Hotel/Motel Audit	DIRECTED
ACD Agent Summary	AUTOPRINT
ACD Monitor Print	DIRECTED
ACD EVENTS	AUTOPRINT
ACD Group Summary	MONITOR
MAI*	AUTOPRINT

* If software option is enabled.

GUARANTEED: This field is only modifiable for SMDR and Data SMDR printouts. The field defaults to NO in all other cases. If this field is set to YES, SMDR records are guaranteed to print without losing records.

Softkeys

DELETE: Pressing the DELETE softkey deletes from the form the printout listed at the cursor position. The ENTER softkey must then be pressed to change the database. The user is alerted if the printout is currently active (printing or queued to print) before the delete is performed. If the printout is active, the user can cause the immediate deletion of the printout by pressing the CONTINUE softkey or cancel the delete operation by pressing the QUIT softkey. When the deletion is completed the message ‘DELETION COMPLETED’ is returned. The printer is deleted when the last printout directed to it is deleted.

ADD: This softkey appears upon entry to the form and is used to insert additional printer locations and/or extension numbers of datasets. When the ADD softkey is pressed, the form is in the ADD mode, and the following softkeys are presented.

PRINTER PORT: This softkey appears when the cursor is positioned on the EXT NUM field. The programmer can enter a valid dataset number or IP socket or press the PRINTER PORT softkey to enter printer port in the EXT NUM field.

SMDR: This softkey defines the printout as an SMDR printout.

CDE DATA: This softkey appears after the MORE softkey has been pressed. It defines the selected printout as a CDE DATA record, allowing the user to print the CDE programmable data to a specified printer.

HM DID: This softkey selects the dedicated DID printer for DID audits. **Note**: If the DID Server Application and ACD are both enabled, the HM DID softkey does not appear on the main screen. Press the MORE softkey to access the HM DID softkey in this instance. (Prior to Release 3.2.0.15, the HM DID type printer is not supported if ACD is enabled.)

DATA SMDR: This softkey defines the printout as a DATA SMDR. DATA SMDR is a record of internal data calls.

ACD AGT SUM: This softkey appears after the MORE softkey has been pressed. Available only if system software contains ACD Option. Refer to the ACD TELEMARKETER Application Package section for further information.

ACD MONITORS: This softkey appears after the MORE softkey has been pressed. Available only if system software contains ACD Option. Refer to the ACD TELEMARKETER Application Package section for further information.

ACD GRP SUMMARY: This softkey appears after the MORE softkey has been pressed. Available only if system software contains ACD Option. Refer to the ACD TELEMARKETER Application Package section for further information.

ACD EVENTS: This softkey appears after the MORE softkey has been pressed. Available only if system software contains ACD Option. Refer to the ACD TELEMARKETER Application Package section for further information.

TRAFFIC: This softkey defines the printout as a traffic measurement report.

IP TRAFFIC: This softkey defines the printout as an IP voice networking measurement.

MAINT LOGS: This softkey defines the printout as a maintenance log printout.

HM WAKEUP: This softkey defines the printout as a report of wakeup calls.

HM AUDIT: This softkey defines the printout as an audit report.

PMS: This softkey appears only if Property Management System is enabled in the CDE System Options/System Timers form. When the programmer presses the PMS softkey, “PMS Port” appears in the PRINTOUT field the AUTOPRINT softkey appears. Pressing the AUTOPRINT softkey completes programming of the PMS port.

MAI: This softkey appears only if Mitel Application Interface is enabled in the CDE System Options/System Timers form. When the programmer presses the MAI softkey, “MAI” appears in the PRINTOUT field and the AUTOPRINT softkey appears. Pressing the AUTOPRINT softkey completes programming of the MAI port.

You can only program one MAI (HCI) port. If you try to program more than one MAI port, you will receive the following error message "Attempting to program more than 1 HCI port(s)."

AUTOPRINT: The printout occurs automatically when a certain condition in the system is met.

DIRECTED: The printout occurs at the user’s request.

MONITOR: This softkey is available only if the PRINTOUT field is set to maintenance logs. The logs are printed as they occur.

YES: Pressing YES guarantees the printing of the record. By default, the GUARANTEED field is set to NO.

NO: This softkey appears if the GUARANTEED field has been set to YES. Pressing this softkey restores it to NO.

The standard softkeys **CANCEL**, **ENTER**, **DELETE**, **TOP**, **BOTTOM**, ****MORE****, and **QUIT** are also provided.

Form 35 - Global Find Access Code

This form lists access codes in the CDE database, including ARS Leading Digits. Callback Busy and Executive Busy Override are excluded, since they can only be dialed when receiving busy tone. This form exists to provide access code information conveniently to the user. Codes can be assigned, modified or deleted only in the appropriate forms.

Figure: Form 35 Layout

Field Descriptions

ACCESS CODE: This field lists all programmed access codes as assigned in numerical order by the first digits, for example 10, 111, 1210, 132, 20.

Note: 200-203 and 340 and 502 are not applicable to the AX.

DEFINED: This field lists the areas where access codes have been assigned.

BAY, **SLOT**, **CCT** and **SCT**: These fields list the physical location of devices, and LDNs.

MULTIPLE APP.: This indicates if there are multiple appearances of number (LDNs or extension numbers) - this will be one of YES, NO or N/A.

Softkeys

ACCESS CODE: Pressing the ACCESS CODE softkey displays ENTER ACCESS CODE prompt on the command line. After the user enters the access code and presses the ENTER softkey, the code is then verified by the system. If the number entered is not presently used as an access code, the following message is displayed: The access code xxx does not exist.

NEXT: This softkey is displayed when an unassigned number is entered at the ENTER ACCESS CODE prompt, and is cancelled. Pressing this softkey displays the next access code that follows the one requested.

The standard softkeys **CANCEL**, **ENTER**, **TOP**, **BOTTOM**, and **QUIT** are also provided.

Form 36 - Modem Assignment

This form is not used.

Form 37 - Guest RM Key Template

Form 37 provides 3 preprogrammed sets (templates) of speed dial and feature keys for hotel/motel guest room phones. In each COS, the programmer can enable one template which will apply to all Mitel display telephones in that COS that have Room Status Applies enabled.

Figure: Form 37 Layout

Field Descriptions

TYPE: This field lists the function of each key, either speed dial (the default) or a feature (e.g. Do Not Disturb).

SPEED DIAL NUMBER: If the key is a speed dial key, the programmer enters the number here. In order to transfer calls to voice mail, you can program * 9 for a speed dial pause. Pauses are required to allow the voice mail to answer.

PRIVATE: If the key is a speed dial key, the programmer can prevent the user from displaying the number by selecting the PRIVATE feature for this field.

Softkeys

FEATURE: Pressing this softkey makes the set key a feature access key. The following softkeys appear:

ACCOUNT CODE	GROUP LISTEN
AUTO ANSWER	CALLERS
CALL/ATTN	RECORD CALL
CALLBACK	MUSIC
FWD ALWAYS	FWD BUSY
FWD NO ANS	FWD BUSY/NA
FWD ALL	NIGHT ANSWER
CALL PARK/RETRIEVE	CALL PICKUP
OVERRIDE	CAMPON
PA PAGING	DATA DISC
PRIVACY REL	DOUBLE FLASH
RELEASE	FORWARD CALL
SINGLE FLASH	DIRECT PAGE
SWAP	DO NOT DIST
VM PROMPTS	SYSTEM PARK
ALARM	CALL BLOCK
DAY/NIGHT	ORBIT
GUEST ROOM	WKUP ALARM

These softkeys are the same ones that appear in the nested Expand Set form of the Desktop Device Assignments form. Refer to Form 09 for more information.

TEMPLATE NO: Pressing the TEMPLATE NO softkey prompts the user to “ENTER TEMPLATE NUMBER:”. Valid template numbers are 1, 2 and 3. When a valid template number is entered, the new template form is displayed.

KEY: Pressing this key prompts the user to “ENTER KEY NUM:”. The valid range for the key number is 2 - 24 (2 - 16 on the 5340 IP Phone).

PRIVATE: This softkey appears when a speed dial number has been entered and the cursor is on the PRIVATE field. Pressing this key makes a non-private speed dial number private. When a speed dial number is entered under the SPEED DIAL NUMBER column, just tab over to the PRIVATE column and press the PRIVATE softkey to make the number private. The word PRIVATE will appear under the PRIVATE column.

NON PRIVATE: This softkey appears when the cursor is on the PRIVATE field of a line containing a private speed dial number. Pressing this key makes a private speed dial number non-private.

The standard softkeys **CANCEL**, **DELETE**, **ENTER** and **QUIT** are also provided.

Form 38 - ACD Keys Template

The ACD Keys Template form provides up to three different function key configurations for each ACD position: agent, supervisor and senior supervisor. In each COS, however, only one template for one position type can be enabled. Line appearance keys assigned in the Desktop Device Assignments form have priority over ACD feature keys.

Figure: Form 38 Layout

NOTE: Applying an ACD template to 53xx series phones will not override keys that have been programmed using the Superkey or Applications key.

Field Descriptions

The header lists the ACD position to which the template applies and the template number (1, 2 or 3).

KEY: This field lists the line select keys. Key 01, the prime line key, is not shown because it cannot be reprogrammed.

NOTE: Even though the ACD phone may have more than 16 keys, programming is supported on keys 02 to 15 only. Keys programmed on Page 2 and Page 3 of the 5330 and 5340 IP Phones will not function.

TYPE: This field lists the function of each key. The default is speed dial.

SPEED DIAL NUMBER: This field lists the speed dial number for each speed dial key. In order to transfer calls to voice mail, you can program * 9 for a speed dial pause. Pauses are required to allow the voice mail to answer.

PRIV: The word ‘Yes’ in this field indicates that the speed dial number is private and cannot be displayed at the set.

SEC: If the key is a line appearance, this field indicates (Yes or No) whether the secretarial variant is enabled. For a Busy Lamp key, setting this field to YES causes an immediate release when the DSS key is pressed.

DSS: NO in this field indicates that the DSS key cannot call the selected station. CALL in this field allows the DSS key to call the selected station.

DN: The extension of a phone programmed in Form 09 (Desktop Device Assignment), or of an analog interface module (AIM) programmed in Form 12 (Data Assignment).

Softkeys

ACD KEYS: Pressing the ACD KEYS softkey provides a set of softkeys used to assign ACD feature keys to the set’s line select keys. The feature keys presented depend on the type of keys template being programmed. All templates have a QUEUE STATUS key available. In addition, an agent template can have a MAKE BUSY key and a supervisor template can have an AGENT STATUS key. Both the SUPERVISOR and SENIOR SUPERVISOR can be provided with a SHIFT key.

AGENT: Pressing the AGENT softkey prompts the user to enter an agent template number "ENTER AGENT TEMPLATE NUMBER:". When a valid template number (1-3) is entered, the selected agent keys template is displayed. The title line is updated with the agent template number.

DOUBLE FLASH: Pressing the DOUBLE FLASH softkey provides the ability to send TRANS/CONF then dial the Double Flash Over Trunk access code.

DSS CALL: Appears only in the DSS field. Enables the DSS key associated with a BLF appearance to call the selected extension. The CALL indication appears in the DSS field.

DSS PAGE: Appears only in the DSS field. Enables the DSS key associated with a BLF appearance to direct page a selected station. The PAGE indication appears in the DSS field.

KEY: Pressing the KEY softkey prompts the user to enter a key number (2-24) (2 - 16 on the 5340 IP phone). When a valid line key number is entered, the cursor points to that line. The command line displays the line, ready for editing.

NON-ACD KEYS: Pressing the NON-ACD KEYS softkey provides a sub-level of softkeys through which the user can select NON-ACD feature keys:

ACCOUNT CODE	GROUP LISTEN
AUTO ANSWER	CALLERS
CALL/ATTN	RECORD CALL
CALLBACK	MUSIC
FWD ALWAYS	FWD BUSY
FWD NO ANS	FWD BUSY/NA
FWD ALL	NIGHT ANSWER
CALL PARK/RETRIEVE	CALL PICKUP
OVERRIDE	CAMPON
PA PAGING	DATA DISC
PRIVACY REL	DOUBLE FLASH
RELEASE	FORWARD CALL
HEADSET MODE	HANDSET MUTE
SINGLE FLASH	DIRECT PAGE
SWAP	DO NOT DIST
VM PROMPTS	SYSTEM PARK
ALARM	CALL BLOCK
DAY/NIGHT	ORBIT
GUEST ROOM	WKUP ALARM

NON DSS: Appears only in the DSS field. Disables the DSS key associated with a BLF appearance. The "No" indication appears in the DSS field.

NON PRIVATE: Pressing the NON PRIVATE softkey makes a private speed dial number non-private.

NON SECR: Pressing this softkey disables the secretarial function for the selected Mitel telephone key (line appearance). The “No” indication appears in the SECRETARIAL field.

PRIVATE: Pressing the PRIVATE softkey makes the programmed speed dial number private.

SECRETARIAL: Pressing this softkey enables the secretarial function for the selected Mitel telephone key (line appearance). The ‘Yes’ indication appears in the SECRETARIAL field. When a Line Select key is set as a secretarial key, then the user can override the DO NOT DISTURB feature on the SUPERSET telephone corresponding to that line appearance. For a DSS key, this enables the secretarial option.

SENIOR: Pressing the SENIOR softkey prompts the user to enter a senior supervisor template number “ENTER SENIOR SUPERVISOR TEMPLATE NUMBER:”. When a valid template number (1-3) is entered, the selected senior supervisor keys template is displayed. The title line is updated with the senior supervisor template number.

SINGLE FLASH: Pressing the SINGLE FLASH softkey provides the ability to send TRANS/CONF, dial the Flash Over Trunk access code, then go into either dial state or talk state (if the CENTREX Flash Over Trunk option is enabled in the COS of the trunk).

SUPERVISOR: Pressing the SUPERVISOR softkey prompts the user to enter a supervisor template number “ENTER SUPERVISOR TEMPLATE NUMBER:”. When a valid template number (1-3) is entered, the selected supervisor keys template is displayed. The title line is updated with the supervisor template number.

The standard softkeys **QUIT** and **DELETE** are also provided.

Form 39 - ACD Agent Groups

The ACD AGENT GROUPS CDE form lists the agents in each ACD group. It cannot be accessed unless a “Maximum ACD Agents” system option is selected. The title line contains the agent group number and name. Entries in this form are sorted by ID.

Figure: Form 39 Layout

Field Descriptions

The header shows the ACD group number, and ACD group name.

AGENT ID: This field lists the agent ID. This is an access code that identifies the agent to the system. The form lists the agents in numerical order by agent ID.

AGENT NAME: This field lists the agent name. Use of this field is optional, but recommended.

COS: This field lists the agent’s class of service.

Softkeys

ACD GRP NAME: Pressing the ACD GRP NAME key displays ENTER ACD GROUP NAME: on the command line. This softkey appears only if there is at least one agent in the group, because a group cannot exist without members. The name can be up to eight characters long.

FIND ID: Pressing the FIND ID key displays ENTER AGENT ID: on the command line. The system searches the data base for the requested agent. If found, the group containing this agent is displayed with the cursor pointing at the agent.

ADD: Pressing the ADD key permits the programmer to enter the information needed to add an agent to the displayed group. When the Agent ID has been entered, the ENTER softkey appears. Optionally, the programmer can fill in the agent name and COS fields prior to pressing the ENTER softkey. If nothing is entered, the name defaults to blanks and the COS to 1. The added agent will appear in the correct position in the sorted list. If the agent is already assigned to another agent group, the system warns the programmer on the command line and asks the programmer to CONFIRM or CANCEL the entry via softkeys. If the programmer confirms it, any previous group assignment for that agent is deleted. Agents can be reassigned in this way at any time, even while the agent is active. When reassigning an agent, only the ID field should be filled in. The NAME and COS fields will be filled automatically when the ENTER key is pressed.

AGENT GROUP: Pressing the AGENT GROUP softkey displays ENTER ACD GROUP NUM: on the command line. When the ACD Group number is entered, the requested group is displayed.

OPTIONS: Pressing the OPTIONS key displays the options sub-form. This softkey is not provided if the group is empty.

The standard softkeys **DELETE**, **TOP**, **BOTTOM** and **QUIT** are also provided.

Subform 39 - Options

For each agent group a set of options can be set to control the ACD group environment (see the figure below).

Figure: Option Subform Layout

Field Descriptions

OPTIONS: This field displays the option name. For options with a numerical value, the format and units of measure are given in brackets (e.g. MM:SS for minutes:seconds).

STATUS: This field displays the current setting of the option.

Softkeys

The standard softkey **QUIT** is provided.

Parameters

Afterwork Timer: After an ACD call has ended, this timer provides a time period for the agent to complete paperwork. The agent will receive no ACD calls during this period. The timer has a range of 00:00 to 15:00. The default is 00:00. It is recommended that a time be programmed for the after work timer.

Overflow Timer: This timer specifies the maximum time a call can wait for answer in this ACD group. When the time period has elapsed, the call is sent to the overflow destination. The timer has a range of 00:00 to 54:00 (Minutes:Seconds). The default is 00:00.

First Status Threshold: This threshold time provides a visual indication to the supervisor that the system has reached a defined level of activity. When any call has waited for the defined time period, the supervisor's Queue Status LCD shows a light circle in a dark square. The threshold has a range of 00:00 to 54:00 (Minutes:Seconds). The default is 03:00. The first threshold time must be less than the second threshold time.

Second Status Threshold: This threshold time provides a visual indication to the supervisor that the system has reached a defined level of activity. This is a higher level of activity than the first status threshold represents. When any call has waited for the defined time period, the supervisor's Queue Status LCD shows a dark square. The threshold has a range of 00:00 to 54:00 (Minutes:Seconds). The time must exceed that defined for the first status threshold.

Form 40 - ACD Supervisors

The ACD Supervisors form shows the ID numbers and names of ACD senior supervisors. A subform lists the supervisors. These forms cannot be accessed unless a “Maximum ACD Agents” system option is selected. Refer to the figure below.

Figure: Form 40 Layout

Field Descriptions

SENIOR SUPERVISOR IDS: This field lists the senior supervisor ID codes. The form is sorted by ID code.

NAME: Senior supervisor and supervisor names are carried to the set where they log on. Their ID’s are used for logging on.

COS: This field specifies the class of service of each senior supervisor. The range is 1 to 50.

Softkeys

FIND GROUP: The FIND GROUP softkey displays ENTER ACD GROUP NUM: on the command line. When the group number is entered, the requested group is displayed. If the requested group is assigned to a senior supervisor, that senior supervisor’s sub-form is shown, with the cursor pointing at the requested group.

FIND SUPER: The FIND SUPER softkey displays ENTER SUPERVISOR OR SENIOR SUPERVISOR ID: on the command line. The data base is searched for this senior supervisor or supervisor ID. If the requested senior supervisor exists, the top level form is shown, with the cursor pointing at the requested ID. If the requested supervisor exists, the sub-form is shown, with the cursor pointing at the requested ID.

ADD: This softkey appears upon entry to the form and is used to add additional senior supervisors to the form. When assigning an agent as a senior supervisor, only the ID field needs to be filled in. The NAME field will be filled automatically when the ENTER key is pressed. When the ADD softkey is pressed, the form is in the ADD mode. To leave ADD mode, use the QUIT softkey.

EXPAND: The EXPAND softkey displays the expand sub-form, which lists all the groups of the supervisors under the current senior supervisor. The sub-form is sorted by supervisor ID code. All groups reporting directly to the senior supervisor are listed at the end of the form in order of group number.

The standard softkeys **DELETE**, **TOP**, **BOTTOM** and **QUIT** are also provided.

Subform 40 - Expand

For each senior supervisor, this form, shown in the figure below, lists the assigned ACD Groups and their supervisors. The programmer positions the line pointer at the desired senior supervisor in the ACD senior supervisors form and presses the EXPAND softkey.

Figure: Expand Subform Layout

Field Descriptions

The subform header shows the senior supervisor’s name and ID. If the senior supervisor has no name programmed, the header will show the senior supervisor’s number.

GRPS OF: This field lists the agent groups reporting to the senior supervisor. All groups reporting directly to the senior supervisor (with no supervisor) are listed at the bottom of the form, sorted by agent group number. Groups that have supervisors are sorted by supervisor ID. Groups with the same supervisor are sorted by agent group number.

SUPER ID: This field lists the supervisor ID number for each supervisor under the senior supervisor. If a group has no supervisor, the field displays NO SUPER.

SUPER NAME: The supervisor's name can be programmed in this column. It can be up to 10 characters long; it must not begin with the character "*".

COS: This field lists the class of service of each supervisor.

Softkeys

FIND GROUP: The FIND GROUP softkey displays ENTER ACD GROUP NUM: on the command line. When the group number is entered, the requested group is displayed. If the requested group is assigned to a senior supervisor, that senior supervisor's sub-form is shown, with the cursor pointing at the requested group.

FIND SUPER: The FIND SUPER softkey displays ENTER SUPERVISOR OR SENIOR SUPERVISOR ID: on the command line. If the requested senior supervisor exists, the top level form is shown, with the cursor pointing at the requested ID. If the requested supervisor exists, the sub-form is shown, with the cursor pointing at the requested ID.

ADD: Pressing the ADD key assigns a group and its supervisor to the displayed senior supervisor. Any previous assignment for that group is deleted. Groups can be reassigned at any time, even while active.

NO SUPER: The NO SUPER key assigns the agent group directly to the senior supervisor; the group has no supervisor.

The standard softkeys **DELETE**, **TOP**, **BOTTOM** and **QUIT** are also provided.

Form 41 - ACD Path

This CDE form defines routing for ACD calls. It cannot be accessed unless a "Maximum ACD Agents" system option is selected. Each path has its own form.

Figure: Form 41 Layout

Field Descriptions

The header displays the ACD path number and name.

OPTIONS: The options field lists programmable timers and options for the ACD path.

STATUS: The status field is the only field that can be edited; however, no fields on an ACD path can be edited without first assigning the "Access Code For This ACD Path", and the "Primary ACD Agent Group" (the first two lines on the form).

PATH NAME: The PATH NAME softkey displays ENTER PATH NAME: on the command line. When the name (up to 8 characters) is entered, it appears on the form top line, beside the path number. This softkey appears only after the first two lines of the form are both filled.

ACD PATH: Pressing the ACD PATH softkey displays ENTER ACD PATH NUMBER: on the command line. This softkey appears whenever the first two lines of the form are either both filled or both empty.

DELETE PATH: Pressing the DELETE PATH softkey displays the CANCEL and CONFIRM softkeys. Pressing CONFIRM deletes the ACD path. Pressing CANCEL restores the softkeys without deleting the ACD path.

DELETE FIELD: Blanks the current status field.

DROP CALL: Appears only when the cursor is in the "Interflow Point Access Code" line - the STATUS will be set to "DROP CALL".

YES/**NO**: Enables / disables the “Allow Overflow to Interflow Point Before Timeout” or “Interflow Enabled” parameters.

Parameters

ACD PATH PARAMETERS
Access Code For This ACD Path Primary ACD Agent Group Delay For Ringback (MM:SS) Recording 1 : Start Time (MM:SS) Access Code Music Source Following Recording 2 : Start Time (MM:SS) Access Code Music Source Following Recording 3 : Start Time (MM:SS) Access Code Music Source Following Recording 4 : Start Time (MM:SS) Access Code Music Source Following Overflow 1 Agent Group Overflow 2 Agent Group Overflow 3 Agent Group Interflow Enabled Interflow Timeout (MM:SS) Interflow Point Access Code (Default = DROP CALL) Allow Overflow to Interflow Point Before Timeout Priority Service Time Immediately Interflow when no Agents Logged In Tenant

Access Code For This ACD Path and **Primary ACD Agent Group** must be defined to establish an ACD path. Until this is done, no other options can be edited.

Delay For Ringback (MM:SS): Allows the caller to hear ringback before being connected to an agent.

Recording n: Start Time (MM:SS) is counted from when a caller has finished the Delay For Ringback period. The range of this timer is 00:00 to 54:00. Recording 1 must be defined before Recording 2, Recording 2 before Recording 3, Recording 3 before Recording 4. For example, the programmer cannot edit “Recording 3: Start Time”, unless “Recording 2” and “Recording 1” are both defined. If <cursor down> is pressed from a blank “Recording n: Start Time” field, the cursor moves to “Overflow 1 Agent Group”. Each recording’s start time must be later than the preceding recording’s start time.

Recording n: Access Code must be an access code for a recording hunt group. The programmer cannot make an entry for this option until a start time is defined for the recording. Once a recording start time is entered, the programmer can only move back and forth between the Start Time and Access Code fields for that recording, until both fields are filled or deleted (by the DELETE FIELD key). PATH NAME, ACD PATH, and DELETE PATH keys are removed during this time too.

Music Source Following defines, for each recording, an ONS port to which the caller will be connected when the recording ends. The ONS port is permanently off-hook with a music source connected. The caller stays connected until the call is answered or another scheduled recording plays. If no music source is defined, the Music-on-Hold source is used. If this is not provided, the caller receives silence.

Overflow 1 Agent Group must be defined before **Overflow 2 Agent Group**, Overflow 2 Agent Group before **Overflow 3 Agent Group**. If <cursor down> is pressed from a blank "Overflow X Agent Group", the user is positioned at the "Interflow Timeout" field.

Interflow Enabled, when set to YES, allows the waiting ACD call to exit ACD and be answered at a defined interflow point. Default is NO.

Interflow Timeout (MM:SS) specifies when a waiting ACD call should be directed to the interflow point. The timer has a range of 00:01 to 54:00. The default is 54 minutes.

Interflow Point Access Code can be an extension number, another ACD path, a hunt group (including a UCD agent hunt group), a system abbreviated dial number, night bells or DROP CALL.

Priority for the ACD path has a range of 1 (highest priority) to 99. Calls are answered in order of priority. Default priority is 99.

Service Time: Defines a standard time to answer (from 1 second to 54 minutes), used in the measurement of ACD path performance. Statistics can be seen from the ACD Path Monitors and Path Summary Reports (see the ACD TELEMARKETER Application Package section).

Immediately Interflow when no Agents Logged In: When set to YES, and "Interflow Enabled" is YES, then any callers dialing in to an ACD path will interflow immediately when no agents are logged in. This interflow takes place regardless of the status of the "Interflow Timeout," or the option "Allow Overflow to Interflow Point Before Timeout," or the "Interflow Point Access Code" having a value of DROP CALL. The field default is NO.

Tenant: When a tenant number is assigned to the ACD Path, DID and TIE trunks which dial into the ACD path directly will follow the routing for this tenant as defined in Form 19 - Call Rerouting Table. Enter a valid number (1 to 25). The default is blank (no tenant).

Form 42 - T1 Link Descriptors

This form defines the parameters that control the behavior of each T1 link. A link descriptor may be shared with several links provided that they have similar operating characteristics, or each link may be assigned a separate link descriptor. Ten (10) descriptors may be defined although only a maximum of seven (7) links (672 port system) may be programmed.

The descriptor parameters define four characteristics for the link:

- Which conditions require the link to be put out of service
- Which conditions are required for the link to return to service
- The encoding used for the transmission of consecutive zeros
- Which condition ensures an accurate sync source for the system.

The main form displays the number of users of each descriptor. A 20-character comment field is provided for each descriptor.

CAUTION: Calls in progress on T1 links that are affected by changes in this form will be lost. Restrict changes to off hours if possible.

Figure: Form 42 Layout

Field Descriptions

DESCRIPTOR: This field lists the descriptor numbers. It cannot be edited.

LINK TYPE: This field lists the type of link for each descriptor. You may program the T1 module as T1 DS1 or T1 CSU. The default is T1 DS1. You may change this field when there are links assigned to the descriptor.

NUMBER OF LINKS ASSIGNED: This field lists the number of links assigned this descriptor in the T1 Link Assignment form.

COMMENTS: This 20-character field is provided for the programmer's notes. The system does not use this information.

Softkeys

SEL. OPTION: Pressing the SELECT OPTION softkey displays the Link Descriptor Options subform for the descriptor on the command line.

REVIEW: Displays the Review List subform for the descriptor on the command line, a read-only form that provides a list of users for each descriptor. This softkey appears only if the number of links is greater than 0.

The standard softkey **QUIT** is also provided.

Subform 42 - Link Descriptor Options

Note: Altering this form may have side effects on T1 links in operation.

Figure: Link Descriptor Options Subform Layout

Field Descriptions

The header line displays the link descriptor number and the direction of the link.

Value: This field lists the selected value for each parameter.

Softkeys

AMI/AMI&ZCS/B8ZS: The Line Coding entry has three possible values: AMI (Alternate Mark Inversion), AMI&ZCS (AMI with Zero Code Suppression), and B8ZS (Bipolar with 8 Zeros Inversion). AMI only applies to links connected to T1/E1 modules. AMI&ZCS applies to links connected to T1 cards and is the default setting.

CONFIRM: This softkey appears when an option value has been changed. Press the CONFIRM softkey to enter the change in the database.

The standard softkeys **CANCEL**, **ENTER** and **QUIT** are also provided.

Parameters

Alarm Debounce Timer

The alarm debounce timer defines the length of time a fault condition must be present before it is reported to the main controller, and the link is removed from service. The range of the timer is 300 to 3200 ms. The reportable fault conditions are

- Failure of power supply
- Loss of incoming signals at 1544 Kbps
- Loss of frame alignment (synchronization)
- Alarm indication received from the remote end.

Alarm Debounce Timer Parameter
Options: 300 to 3200 ms, Default: 2500

- Defines the length of time that a fault must be present before it is reported. The following conditions apply:

  - Fault is detected on incoming (Rx) pair of the link
  - Fault may be due to internal problem (T1 card) or external problem on the Rx pair (wiring or far end)
  - Fault raises a RED Alarm (that is, if red LED on the card is lit, link is out of service).
- Reports loss of frame synchronization. The following conditions apply:
  - Causes a Yellow alarm to be transmitted out to the far end the is, bit 2 of all channels on the link is forced “low”)
  - Generates a “Log Report”.
- Reports reception of alarm from remote end. The following conditions apply:
  - Indicates that link is receiving a YELLOW alarm from far end
  - Generates a “Log Report”.

Line Coding

This parameter defines the special coding used for consecutive “zeros” transmitted in the link in order to satisfy the AT&T T1 transmission specifications for the proper density of “1's” to keep the circuit “synchronized”.

There are two standards presently used to represent the data byte “zero” in the network, AMI bit jamming, and B8ZS.

There are three possible values: AMI (Alternate Mark Inversion), AMI&ZCS (AMI with Zero Code Suppression), and B8ZS (Bipolar with 8 Zeros Inversion). AMI only applies to links connected to T1/E1 modules. AMI&ZCS applies to links connected to T1 cards

The B8ZS Zero Code Suppression parameter setting must match the “zero code suppression” coding used on the specific link. A mismatch will result in excessive number of “bipolar errors” which could cause to link to be busied out.

To better understand how these apply, a definition of the T1 transmission signal is required. The T1 signal transmits bits through the link by representing the “1 bits” as a voltage and the “0 bits” as no voltage. “1 bits” are commonly called “MARK” (dates back to the days of the telegraph).

In order to detect interference errors in the transmission of the “mark” bits, the bits are sent alternately as a positive (+) voltage then negative (-) voltage, specifically +3 Volts then -3 Volts. This is known as “ALTERNATE MARK INVERSION”, “AMI” or “BIPOLAR” transmission.

Figure: AMI/Bipolar Transmission

Setting the Line Coding Parameter
Options: B8ZS / AMI&ZCS, Default is AMI&ZCS

- Default is “AMI&ZCS” which selects AMI bit jamming with Zero Code suppression.
  - AMI&ZCS bit jamming is the most common T1 voice channel bit coding. AMI&ZCS applies to links on T1 cards.
- To select coding type for consecutive “0s” in channel
  - Select one of two Alternate Mark Inversion (AMI) zero coding rules.
  - Enter “B8ZS” to select special AMI rule, Bipolar 8 Zero Substitution (B8ZS).
  - Enter “AMI&ZCS” to select standard AMI rule, Bit Jamming.
- To meet AT&T T1 transmission specifications, coding of consecutive zeros is required
  - Note that “1's” are required for clock extraction for proper transmission.

“AMI&ZCS” Line Coding

- Selects AMI, Bit Jamming with zero code suppression. The following conditions apply:
  - Selects the standard AMI rule for consecutive “0's”.

- AMI rule forces the second bit to “1” if channel bits are all “0”.
- Receiver cannot distinguish between a byte with the jammed bit and the quantized voice data byte of “2” because the resulting bit pattern is the same.
- Effects of bit jamming does not affect quantized voice transmission noticeably.
- Bit jamming is not suitable for data transmission as it corrupts the data value.
- AMI&ZCS applies to links on T1 cards.

Figure: Bit Jamming (AMI/Bipolar Format)

"AMI" Line Coding

- AMI applies to links on T1/E1 modules.

"B8ZS" Line Coding

- Selects AMI, Bipolar 8-Zero Substitution. The following conditions apply:
  - Selects the B8ZS rule for consecutive “0's”.
  - B8ZS inserts a specific bipolar pattern for “0's”.
  - If last valid “1” was “+” then “0 0 0 + - 0- +” is inserted.
  - If last valid “1” was “-” then “0 0 0 - + 0 + -” is inserted.
  - Uses Bipolar Violation.
  - Receiver detects the known pattern and substitutes it with the byte 0.
  - Suitable for both quantized voice and DATA transmission.

Figure: B8ZS (AMI/Bipolar Format)

Line Build Out

Line Build Out appears when the link is connected to a T1/E1 module and the link type is T1 CSU.The four possible values are 0, -7.5, -15, -22.5 dB. The default is 0.

Line Length

Line Length appears when the link is connected to a T1/E1 module and the link type is T1 DS1. The five possible values are max 132, 265, 398, 533, or 655. The default is 266-399.

Framing

Framing enables the ESF functionality for the link connected to a T1/E1 module. This option applies to both link types, T1 CSU and T1 DSU. The two settings are D4 or ESF. The default is D4.

Slip Rate Limits

A “slip” is the deletion or repetition of a single frame of information in a digital bit stream due to a mismatch in the synchronization of the system.

The “Slip Rate Limit” refers to the unacceptable number of lost or added frames due to having a system clock that is too fast or too slow compared to the network.

The slip rate is measured over 24 hour window (That is, 24 one hour segments where the newest hour count deletes the oldest count). If the sum of the accumulated slips over the 24 hour segments exceeds the set limit the related action is taken.

There are 3 slip rate limits: maintenance, service and network. The slip rate limit is between 0 and 9000 slips over a 24 hour period.

Slip Rate - Maintenance Limit
Options: 0 to 9000, Default: 255

- Defines the number of slips required for a “Log Report” to be generated. The following conditions apply:
  - AT&T uses 255 slips as offering acceptable grade of service.
  - There are no rules as to what is “acceptable”.
- The “Log Report” indicates to the maintainer that there may be a problem with the quality of transmission.

Slip Rate - Service Limit
Options: 0 to 9000, Default: 7000

- Defines the number of slips required for the links to be put out of service, that is, the number of slips which renders the link unusable
- A slip rate fault causes a “YELLOW” alarm to be sent out on the link. The far end is notified that the link is not available for transmission
- The RTS timer “Service Limit Exceeded” timer starts
  - Allows the system to check that the link is back within tolerance when the timer expires.
  - Allows the system to determine if the link can be returned to service to handle calls.

Slip Rate - Network Sync Limit
Options: 0 to 9000, Default: 7

- Applies only to the link being used as the system clock reference source. The link for the system clock source is programmed in Form 44.
- If the limit is exceeded, the system selects the next clock source as defined in Form 44. The system also generates a “Log Report”.
- If no alternative source is available, the system goes into freerun. Freerun means the system clock rate will not change to keep in sync with the network.
- The RTS timer “Net Slip Limit Exceeded” timer starts:
  - Allows the system to check that the link is back within tolerance when the timer is expires
  - Allows the system to determine if the link can be reestablished as the clock sync source.

BER (Bit Error Rate) Limits:

A “bit error” is considered to be any bipolar violation due to the loss or addition of a bit pulse on the link. An error is detected when consecutive bit pulses (1's) have the same polarity (a bipolar violation).

Bit errors (bipolar violations) are typically a result of external electrical interference, such as line noise, lightning inductions or crosstalk.

The Bit Errors Rate (BER) is measured over a one hour period. Exceeding the set rate causes the related action to be taken by the system.

The system keeps track of the BER for the past 24 one hour periods but uses the present hour's value for this parameter.

There are two BER rate limits: maintenance and service. The bit error rate limit is between 10^{-3} (1 error per 1000 bits) and 10^{-6} (1 error per 1,000,000 bits) over a 24-hour period.

BER - Maintenance Limit
Options: 10^{-n}/hour (n = 3, 4, 5, 6), Default: 10^{-4} (n = 4)

- Defines the number of bit errors that will generate a log report
- Default is 10^{-4}/hr = 1/10,000/hr = one bit violation per 10,000 bits per hour
- (1.544 MB/sec X 3600 sec/hr)/10,000 bit/hr = 555,840 BPV per hour.

BER - Service Limit

Options: 10^{-n}/hour (n = 3, 4, 5, 6), Default: 10^{-3} (n = 3)

- Defines the number of bit errors that will put the link out of service
- Default is 10^{-3} = 1/1,000 = one bit violation per 1,000 bits per hour
- (1.544 MB per second X 3600 seconds/hour)/1000 bit/hour = 5,558,400 BPV per hour
- If the limit is exceeded the link is taken out of service. The link transmit a Yellow alarm, the system generates a log report, and the RTS Timer “Service Limit Exceeded” timer starts. This timer allows the system
  - to check if the link is back within tolerance when the timer expires
  - to return the link to service if the link is back within tolerance.

Framing Loss Limits

A “Framing Loss” occurs when the digital trunk cannot find the proper framing bit sequence in the incoming bit stream. This framing loss results in the trunk being unable to correctly identify and decode the channels.

This parameter is used to identify the unacceptable number of times the framing bit pattern cannot be found.

The Frame Loss parameter is measured over 24 hour window (i.e., 24 one-hour segments were the newest hour count deletes the oldest count). If the sum of the accumulated frame losses over the 24 hour segments exceeds the set limit the related action is taken.

There are three framing loss limits: maintenance, service, and slip rate - network sync limit. The framing loss error rate is between 0 and 9000 losses over a 24-hour period.

Framing Losses - Maintenance Limit

Options: 0 - 9000/24 hour, Default: 255/24 hour

If the maintenance limit is exceeded, this fault

- causes the system to generate a maintenance log report
- indicates that there is some degradation in the signal quality.

Framing Losses - Service Limit

Options: 0 - 9000/24 hour, Default: 9000/24 hour

If the service limit is exceeded, the

- system takes the link out of service
- system generates a log report
- link transmits a yellow alarm
- RTS Timer - Service Limit Exceeded timer starts.

Slip Rate - Network Sync Limit

If this limit is exceeded, the system generates a maintenance log and selects a new sync source.

Return to Service Timers - RTS Timer

The “Return To Service (RTS) Timers” automatically return a link to service when the fault which caused it to fail is no longer present.

There are three RTS timers. Each of the timers start when the related condition causes the link to be put out of service. If the fault condition is still present after the timer expires, the link remains out of service and the timer is restarted.

RTS Timer - Service Limit Exceeded

Options: 1 - 255 minutes, Default: 30 minutes

- Defines the duration to wait before attempting to return link into service. Keep this timer at 30 minutes or above to prevent it from going in and out of service too frequently if the link is faulty. Then, users will be less likely to access a faulty trunk that could drop their call in the middle of a conversation.
- Applies to faults that are caused by exceeding the Framing, Slip, or BER service limits
- When the RTS timer expires, the system checks to see if link is operating at no more than 1/24th the maintenance limit (Frame, Slip, or BER). After this condition is met, the system returns the link to service. If this condition is not met, the timer is restarted.

RTS Timer - Net Slip Limit Exceeded
Options: 1 - 255 minutes, Default: 30 minutes

- Defines the duration to wait before attempting to return the links as the sync source.
- Applies to faults that are caused by exceeding the network sync slip rate limit.
- Returns the link to sync source only after slip rate is less than net sync limit.

RTS Timer - After Alarm
Options: 0 - 300 seconds, Default: 10 seconds

- Defines the duration that a link is kept unavailable after an alarm condition has cleared.
- Timer starts after the Yellow alarm is no longer being received and the Red alarm due to loss of sync is cleared (Link no longer transmitting Yellow alarm).

Termination Mode

Controls the transmit and receive settings for the Dual T/E1 Framer MMC. Select Network Termination (NT) if the system is connected to a CO. Select Line Termination (LT) if the system is connected to another system over leased lines. NT is the default.

Note: When two systems are connected to each other over leased lines, one system must use NT mode while the other uses LT mode. A system that uses LT mode requires cabling that reverses the Tx and Rx pair. For LT and NT mode pinout settings, see Tip and Ring Pinouts on the Dual T1/E1 Framer MMC.

Embedded PRI

Use these options to program embedded PRIs (Dual T1/E1 Framers or T1/E1 Combos installed in the SX-200 ICP). Use IMAT to program offboard PRIs (NSU PRIs or PRI cards installed in peripheral cabinets). For further information, see Using the ISDN Maintenance and Administration Tool (IMAT).

Protocol
Options: DMS250, NI2, 4ESS, and DMS100 (default).

- The main ISDN protocol provided by the CO.

Protocol Variant
Options: NI2-Bellcore, NI2-5Ess and NI2-GTD5, and None (default).

- If NI2 is specified as the ISDN Protocol, select Bellcore, 5Ess, or GTD5 as the Protocol Variant. If any other Protocol is specified, select None as the Protocol Variant.

Network/User
Options: Network, and User (default).

- Defines whether the PRI link is on the network or user side. Network is selectable only if FOR option 93 is enabled for Remote LAN Access.

Unknown Numbering Plan
Options: Enabled, and Disabled (default).

- Enabling the option sends out the Called Party information as "address and number unknown." Some COs will fill in this information.

Bearer Capability Voice
Options: Per-Call, 3.1 kHz, and Speech (default).

- The Per-Call option relies on ARS digit insertion, either 0 (Speech) or 1 (3.1 kHz).

CLIR Voice
Options: Per-Call, Restrict, and Allow (default).

- The Per-Call option relies on ARS digit insertion, either 0 (Allow) or 1 (Restrict).

Invert D Channel
Options: Yes, and No (default).

- This parameter allows you to invert the D channel (signaling channel) on the Dual T1/E1 Module or T1/E1 Combo Module. Select "Yes" to allow communication with other systems. For communication with other manufacturer's system equipment, you may have to set this parameter to "No" (default).

Form 43 - T1 Link Assignment

Form 43 (see the figure below) assigns one of the ten link descriptors to each link. Altering this form may have side effects on T1 links in operation.

Figure: Form 43 Layout

Field Descriptions

TRUNK TYPE: This field cannot be edited.

BAY, **SLOT**: This field lists the bay and slot location of the T1 trunk cards in the system.

LINK DESC NUM: This field lists the link descriptor that applies to each T1 trunk card.

COMMENTS: The comments field can store 15 characters.

Softkeys

CONFIRM: This softkey appears when an option value has been changed. Press the CONFIRM softkey to enter the change in the database.

The standard softkeys **ENTER** and **QUIT** are also provided.

Form 44 - Network Synchronization

This form determines the order in which the links will be used as the network synchronization clock source. When the error threshold of the first clock source is crossed, the second clock is used as the sync source, etc. Altering this form may have side effects on T1 links in operation.

Figure: Form 44 Layout

Field Descriptions

DESCRIPTION: This field lists the clock sources, first through eighth, for system synchronization to the T1 network. This field cannot be edited.

BAY, **SLOT**: These fields list the bay and slot location of the cards in the system.

CCT: This field is not used.

COMMENTS: The comments field can store 16 characters.

Softkeys

CONFIRM: This softkey appears when an option value has been changed, followed by the QUIT softkey. Press the CONFIRM softkey to enter the change in the database and exit the form.

The standard softkeys **ENTER**, **DELETE** and **QUIT** are also provided.

Form 45 - BRI Devices

The SX-200 ICP does not support BRI.

Form 46 - Key System Toll Control

This form serves to verify dialed digits for the CO line key feature. This is similar to, but independent from Form 26 (ARS: Digit Strings). See the figure below for the form layout.

Figure: Form 46 Layout

Field Descriptions

DIGITS TO BE ANALYZED: This field displays those digits for digit analysis. Digit analysis is required so that they can be verified. The total number of digits in this field, plus the number of digits in the QTY TO FOLLOW field cannot exceed 25.

QTY TO FOLLOW: This field lists the number of digits that the user dials AFTER the analyzed digits. The ‘Unknown’ prompt in this field indicates that the number of subsequent digits is unknown to the system.

DESIGNATION: This field specifies which digit string entries require LONG DISTANCE, LOCAL, or EMERGENCY management. When this field displays LONG DISTANCE, the system expects an account code for that digit string entry from users with COS Option 201 (Account Code, Forced Entry - Long Distance Calls) enabled. The default condition is LOCAL. The EMERGENCY designation allows the call to proceed regardless of the restrictions applied to the originating station. Emergency call digits do not have to be 911 calls. This field serves areas that do and do not have 911 service available or areas that have another set of digits for their local emergency code. The EMERGENCY designation will not cause 911 alarms to alert the attendant console or trigger Email Notification (if programmed in Form 52) and will not send CESID information to the network.

TERM TYPE AND NUM: Allows for selection of a termination type (TRUNK or trunk GROUP) and a number (valid trunk number: 1-200; valid trunk group number: 1-50). If TRUNK is chosen as the termination type, only non-dial-in trunks may be entered.

COR GROUP: Specifies a COR group which is restricted from accessing the digit string. If the field is left blank, all COR groups can access the digit string. This links the form to Form 20 (ARS: COR Group Definition). Valid COR group numbers: 1-50.

Softkeys

LOCAL: This softkey only appears when the cursor is at the DESIGNATION field. Pressing the LOCAL softkey indicates to the system that this entry does not require “long distance” management. The default condition is not long distance as indicated by LOCAL in the DESIGNATION field.

LONG DIST: This softkey only appears when the cursor is at the DESIGNATION field. Pressing the LONG DIST softkey provides "long distance" management.

EMERGENCY: This softkey only appears when the cursor is at the DESIGNATION field. Pressing the EMERGENCY softkey provides emergency call management. The EMERGENCY designation allows the call to proceed regardless of the restrictions applied to the originating station. Emergency call digits do not have to be 911 calls. This designation serves areas that do and do not have 911 service available or areas that have another set of digits for their local emergency code. The EMERGENCY designation will not cause 911 alarms to be generated to the attendant console and will not send CESID information to the network.

UNKNOWN: This softkey appears only when the cursor is at the QTY TO FOLLOW field. Pressing the UNKNOWN softkey indicates to the system that the quantity of dialed digits AFTER the analyzed digits is unknown.

INSERT: This softkey adds new entries to the form. Pressing the INSERT softkey clears the command line and moves the cursor to the DIGITS TO BE ANALYZED field. The addition is completed by entering the new data for each field and pressing the ENTER softkey. Note that the system inserts the addition one line after the line that was displayed on the command line. The system automatically places all inserted or added strings in numerical ascending order with relation to existing strings.

TRUNK: Appears when the cursor is in the TERM TYPE AND NUM field. Sets the termination point device as a trunk. A trunk number is then entered (1 - 200) - this must be a valid non-dial-in trunk number.

GROUP: Appears when the cursor is in the TERM TYPE AND NUM field. Sets the termination point device as a trunk group. A trunk group number is then entered (1 - 50) - this must be a valid trunk group number.

N0X: This softkey functions as a wild card sequence, where N is any digit from 2 to 9. It represents half of the area codes in North America. Pressing this softkey displays N0X in the DIGITS TO BE ANALYZED field. Note that this softkey can only be pressed at the beginning of a digit string.

X: This softkey functions as a wild card digit; it represents any digit from 0 to 9. Pressing this softkey displays X in the DIGITS TO BE ANALYZED field. Note that this softkey can only be pressed at the end of a digit string.

FIND STRING: This softkey selects an entry in the DIGITS TO BE ANALYZED field. Pressing the FIND STRING softkey displays the ENTER DIGIT STRING: prompt on the command line. The selection is completed by entering a valid digit string. The entered digit string does not have to be an exact match; the system accepts subsets of digit strings and moves the cursor to the closest entry.

N1X: This softkey functions as a wild card sequence, where N is any digit from 2 to 9. It represents half of the area codes in North America. Pressing this softkey displays N1X in the DIGITS TO BE ANALYZED field. Note that this softkey can only be pressed at the beginning of a digit string.

NXX, **1N0X**, **0N1X**, **0N0X:** These strings are for area codes created under the old North American Numbering Plan, and where dialing is preceded by a 1 or 0 (long distance access code). These softkeys can only be pressed at the beginning of a digit string. They are accessed by pressing the ** MORE ** softkey once.

1N1X, **0NXX**, 1**NXX:** These strings are used for area codes and office codes that follow the new North American numbering plan, where dialing is preceded by a 1 or 0 (long distance access code) and the second digit can be any value from 0 to 9. These wildcard sequences determine if a 0 or a 1 as well as a valid area or office code has been dialed They are accessed by pressing the ** MORE ** softkey twice.

10XXX0N0X, 10XXX0N1X, 10XXX1N0X, 10XXX1N1X, 10XXX0, 10XXX1: These wildcard sequences, designed for the call aggregator market (hotels, motels, hospitals, universities), prevent unauthorized calls from being billed to the originating line, while allowing customers access to the long distance carriers of their choice. These softkeys can only be pressed at the beginning of a digit string. Press the ** MORE ** softkey again to access 10XXX0N0X, 10XXX0N1X, 10XXX1N0X and 10XXX1N1X. Press ** MORE ** softkey additional times to access 10XXX0 and 10XXX1.

Either wildcard sequences 10XXX or wildcard sequences 101XXXX should be programmed on a PBX, but not both.

Standard softkeys **CANCEL**, **DELETE**, **ENTER**, ****MORE**** and **QUIT** are provided.

Form 47 - IP Networking Forms

This form displays a menu of IP networking subforms. The menu varies according to hardware platform, with the CXi controller having more options than the MX or AX.

IP Networking Forms options:

- 1 - System IP (AX, CX, CXi, and MX)
- 2 - DHCP Server (AX, CX, CXi and MX)
- 3 - L2 Switch (AX, CXi)
- 4 - Routing (CXi)
- 5 - WAN Interface (CXi)
- 6 - Firewall (CXi)
- 7 - PPTP Remote Access (CXi)

The command line displays ENTER FORM NUMBER:. To select a subform, enter <subform-number>, then press Enter. To exit the form and return to the CDE Main Menu, enter Esc-6.

Figure: Form 47 Layout

Form 47 Subforms

CX, CXi, AX, and MX:

- System IP subform
- DHCP Parameters subform
  - DHCP Subnet Opt / Common Opt subform
  - Mitel Options Subform
  - Vendor Options Subform
  - DHCP Expand subform
  - DHCP Lease View subform

CXi and AX:

- Layer 2 Switch Global Configuration subform
  - Ethernet Port Configuration subform

CXi

- Router subform
  - Network List subform
- Internet Gateway Configuration subform
  - PPPoE Configuration subform
  - DHCP Configuration subform
  - Static IP Configuration subform
- Firewall Configuration subform
  - Firewall Forwarding subform
- PPTP Remote Access subform

System IP Subform

Use this subform to program and verify the IP settings for the system.

Figure: System IP Subform Layout

Field Descriptions

SYSTEM IP ADDRESS: This address provides call control functionality for IP phones and management access to the system through a CDE Maintenance session. This address must be on the same subnet as the Layer 2 Switch IP Address, and must not conflict with the DHCP Server configuration (static or range address). The default is 192.168.1.2.

SYSTEM NETMASK: The subnet mask assigned to the SX-200 ICP controller during installation. The default is 255.255.255.0.

SYSTEM GATEWAY IP: Enter the IP address of the default gateway to the Internet. If there is only one subnetwork on the LAN, enter the IP address of the CXi's Layer 2 Switch, or the IP address of the APC, if installed. If there are multiple subnets, enter the IP address of the router connected to the local subnet. In all cases, DHCP server option 3 should be the same as the System Gateway IP to ensure that all DHCP clients use the proper default gateway. The default is 192.168.1.1.

L2 SWITCH IP ADDRESS: (CXi only) The IP address of the layer 2 switch on the LAN. This address must be on the same subnet as the System IP Address, and must not conflict with the DHCP Server configuration. The default is 192.168.1.1.

HOSTNAME: The system hostname of the SX-200 ICP. For the E-mail Notification feature to function, the system hostname must be a valid domain host name of 49 or fewer characters that has been registered in your DNS, or has been listed in your SMTP e-mail server's Hosts file. Otherwise, the e-mail notification messages will be rejected by SMTP e-mail servers that only accept e-mail from senders containing valid hostnames.

FTP SERVER: The IP address of the FTP server if using to install software in the SX-200 ICP controller or to upload Maintenance logs (see Sending Logs to Email Address or FTP Server).

FTP USERNAME: The name used to access folders on the FTP server.

FTP PASSWORD: The password used to access folders on the FTP server. Required entry.

DIFFSERV CODE POINT: Enter the Differentiated Services Code Point Value (0-63) for voice streaming and signaling. This value should match the value programmed for DHCP Option 134. The default is 44 (for upgrades) or 46 (for new installations).

VOICE VLAN ID: AX and CXi only. Enter the voice VLAN membership number (1-4093) of the embedded L2 switch ports. This value should match the Voice VLAN ID programmed on external L2 switches as well as the value programmed for DHCP Option 132. The default voice VLAN ID is 1.

VOICE VLAN PRIORITY: CXi only. Enter the voice VLAN priority (0-7) for expedited forwarding of traffic. The controller will apply the voice priority to traffic originating from the System IP address. This value should match the Voice VLAN priority programmed on external L2 switches as well as the value programmed for DHCP Option 133. The default voice VLAN priority is 6.

Notes:

1. The system must be reset following a change to the SYSTEM IP ADDRESS, SYSTEM NETMASK, SYSTEM GATEWAY IP, or L2 SWITCH IP ADDRESS.
2. An initial software installation restores the factory-set IP configuration data, which includes IP addresses for the controller plus the FTP and e-mail servers.
3. IP configuration data is retained when moving the internal media (CompactFlash or hard drive) and System ID or i-Button from one controller to another.

4. Expand the IP networking setup by programming e-mail forwarding of voice messages, logs, alarms, and E911 notifications. For details, see Unified Messaging.

Softkeys

QUIT: Exits the form and returns to the IP Networking Forms Menu.

DHCP Parameters

Use this subform to set up the DHCP server included with the system. The server supports the following:

- 10 subnets with two IP address ranges per subnet.
- 12 static IP addresses per subnet
- 12 Global DHCP options for Routers, DNS servers, DNS domain and other options defined in the standard created by the Internet Engineering Task Force which defines protocols for use on the Internet.
- 12 Subnet-specific options
- 12 Range-Specific Options per range
- 12 Static-Specific Options per static IP

Figure: DHCP Parameters Subform Layout

Field Descriptions

The maximum number of entries allowed in the subform is 10. When the user exits the subform, the entries are sorted in order by subnet IP address.

SUBNET NAME: The name of the subnet. This field can be edited and can contain up to 20 characters.

IP ADDRESS: The address of the subnet. This field can be edited. Use the Tab key, or the right-arrow key to move to the field. An IP address must be in the form of four three-digit numbers in the range 0 to 255, separated by periods -- e.g., 10.234.191.001.

Bit Mask: Identifies the subnet mask. Subnets are created by partitioning a larger network into smaller interconnected but independent segments or domains. The use of subnets extend the number of IP addresses otherwise available. They also improve performance by limiting the number of devices that have to compete for available bandwidth to a confined geographic area. In implementing a network to handle Voice over IP, the use of subnet is advisable to help maintain voice quality. If the SX-200 ICP is part of a network with subnets, you must enter a subnet mask (the Bit Mask) which acts as a filter to direct messages into the appropriate network segment (subnet).

Softkeys

ENABLE/DISABLE: Used to enable or disable the DHCP Server. The current status of the DHCP Server is displayed at the top of the form.

COMMON OPT: Opens the Common Options subform where the global options can be set. The following default options are provided:

Option	Format	Value
3-Router	IP Address	192.168.1.1
125-Vendor Specific	ASCII	see Notes
DHCP Options Prior to Release 4.0		
128-IP Phone TFTP Server	IP Address	192.168.1.2
129-ICP IP Address	IP Address	192.168.1.2

Option	Format	Value
130-DHCP Server Identifier	ASCII	MITEL IP PHONE
132-VLAN ID (CX only)	Hex Long	1
133-VLAN Priority (CX only)	Hex Long	6
134-DiffServ Code Point	Numeric	44 (upgrades) 46 (new installations)

Notes

- DHCP options 128-133 used to configure Mitel IP endpoints have been reclassified as public options by the Internet Engineering Task Force (see RFC 2133 and RFC 3925). To comply with the change, Mitel uses options 125 or 43 depending on the server's ability to support them and on administrator preference. (The embedded server supports both options with 125 as the default.) The old options can still be used to provide backward compatibility with IP sets that have yet to be upgraded with firmware that supports the new options. After the upgrade, the old options may be removed to prevent future conflicts with standard use, or other vendors' use of these options.
- In Release 4.0, DHCP Options 128 - 133 are replaced by a series of "tags" that form an ASCII data string in either Option 125 (default for Mitel-specific options) or Option 43 (for Mitel- and non-Mitel-specific options.) The default ASCII data string in these option looks like this:

  id:ipphone.mitel.com;sw_tftp=20.20.20.1;call_srv=20.20.20.1;vlan=2;l2p=6

SUBNET OPT: Opens the Subnet Options subform where options can be set for the selected subnet only.

DHCP LEASE: Opens the DHCP Lease subform where DHCP leases can be viewed and deleted.

EXPAND: Opens the Expand subform where address ranges and static IP addresses can be assigned to the selected subnet.

QUIT: Exits the form and returns to the IP Networking Forms Menu.

ADD: Adds a new subnet to the table by prompting you to enter a name, an IP address and a Bit Mask value.

DELETE: Deletes all information on the current line.

DHCP Subnet Opt / Common Opt Subform

This form appears when the SUBNET OPT or COMMON OPT softkey is pressed while the cursor is in a Subnet field in the DHCP Parameters Subform. Use the form to define DHCP options for a subnet. SUBNET OPTions are subnet-specific; whereas, COMMON OPTions apply to all subnets.

The minimum DHCP Options required to get IP phones working are as follows:

Option	Format	Value
3-Router	IP address	<your network Default Router IP>
125-Vendor Specific	ASCII	See Note 1
DHCP Options Prior to Release 4.0		
128-TFTP	IP address	<SX-200 ICP system IP address>

Option	Format	Value
129-ICP	IP address	<SX-200 ICP system IP address>
130-Mitel Identifier	ASCII	MITEL IP PHONE (all caps)
132-VLAN ID (CX only)	Hex Long	1
133-VLAN Priority (CX only)	Hex Long	6
134-DiffServ Code Point	Numeric	44 (upgrades) 46 (new installations)

Note 1: In Release 4.0, DHCP Options 128 - 133 are replaced by a series of "tags" that form an ASCII data string in either Option 125 (default for Mitel-specific options) or Option 43 (for Mitel- and non-Mitel-specific options.) The default ASCII data string in these option looks like this:

id:ipphone.mitel.com;sw_tftp=20.20.20.1;call_srv=20.20.20.1;vlan=2;l2p=6

Figure: Subnet Opt / Common Opt Subform Layouts

Field Descriptions

By default, all lines other than the following first four are blank. The entries are sorted by DHCP option number.

OPTION: A number in the range 1 to 255 corresponding to an IETF-compliant DHCP option. All DHCP options mentioned in RFC 2132 are predefined for you to configure and use. These include options for Routers, DNS servers and DNS domain. The predefined options have standard names and formats which automatically appear when the option number is entered and the Tab or right-arrow key is pressed. Options not defined in RFC 2132 are named "(user defined)" by default. Mitel IP phone options are listed in an ASCII string using option numbers 43 and/or 125.

FORMAT: One of four possible VALUE formats for an option:

IP ADDRESS - four three-digit numbers in the range 0 to 255, separated by periods -- e.g., 010.234.191.001.
NUMERIC **-** a decimal value in the range 0 and 65025.
ASCII **-** string of up to 40 alphanumeric characters.
OCTET STRING **-** a 12-digit hexadecimal number in the range 0 to f f f f f f f f f f f f.

VALUE: The value of the option. Enter the value in the format specified in the FORMAT field. START IP ADDRESS and END IP ADDRESS must be in the range of addresses defined by the subnet BIT MASK value. The field accepts multiple, comma-separated, values.

Softkeys

MITEL: Press this key to enter Mitel-specific values. The Mitel Specific DHCP Options Subform opens with the Vendor ID field automatically populated.

NON MITEL: Press this key to enter non Mitel-specific values.The Vendor Options Subform opens and prompts you for Vendor ID.

VENDOR ID: When entering Mitel-specific values using DHCP Option 43, the Vendor ID field is automatically populated with the string "ipphone.mitel.com". When entering non Mitel-specific values, the Vendor ID field is optional. **Note**: If a value is entered, it must be in string format.

ADD: Adds a new entry to the table, by prompting you to enter an Option Number, the Format, and then the Value.

DELETE: Deletes the selected option.

QUIT: Exits the subform and returns to the DHCP Parameters subform.

Mitel-Specific DHCP Options Subform

This form appears when the **MITEL** softkey is pressed when adding DHCP option 125.

FIGURE: Mitel-Specific DHCP Options subform

Field Descriptions

MITEL OPTIONS: Default list of DHCP options required by Mitel IP phones.

VALUE: The value of the option. Enter the value in the format specified in the FORMAT field.

Softkeys

CLEAR TAG: Clears the value corresponding to the active tag that appears in the option column. **Note**: VLAN PRIORITY and VLAN ID are linked fields. Pressing CLEAR TAG for either of these values clears both.

QUIT: Exits the subform and returns to the DHCP Parameters subform.

Vendor-Specific DHCP Options Subform

This form appears:

- when the **MITEL** softkey is pressed when adding DHCP option 43

OR

- when the **NON-MITEL** softkey is pressed when adding DHCP options 43 or 125.

FIGURE: Vendor-Specific DHCP Options Subform

Field Descriptions

VENDOR INFORMATION DATA STRING: The configuration string that takes the place of separate entries for former DHCP options 128 - 133. Data format for the string is as follows: id:<vendor_id>;<tag>=<value>;<tag>=<value>;... For example,

id:ipphone.mitel.com;sw_tftp=10.37.189.11;call_srv=10.37.189.11,10.37.190.11;vlan=1056;l2p=6;dscp=46.

The following tags are used in the vendor information data strings:

Tag	Description	Notes	Default
sw_tftp	TFTP Server Address	former option 128	RTC IP Address
call_srv	Telephony Server Address (ICP)	former option 129	RTC IP Address
ipa_srv	IP Phone Analyzer Address	former option 131	blank
vlan	VLAN ID	former option 132	1
l2p	VLAN Priority	former option 133	6

Tag	Description	Notes	Default
dscp	DiffServ CodePoint	former option 134	46
app_proxy	HTTP Proxy Server	Applies only to 5140 and 5240 IP Appliances	blank

Softkeys

VENDOR ID: When entering Mitel-specific values using DHCP Option 43, the Vendor ID field is automatically populated with the string "ipphone.mitel.com". When entering non Mitel-specific values, the Vendor ID field is optional. **Note**: If a value is entered, it must be in string format.

DHCP Expand Subform

This form appears when the EXPAND softkey is pressed while the cursor is on subnet, an IP address range, or a single static IP address in the DHCP Parameters Subform, or when the STATIC IP softkey is pressed.

Figure: Expand Subform Layout

Field Descriptions

The entries in the DHCP Expand Subform are sorted by IP address (range and static).

NAME: The user-defined name of the IP address range.

IP RANGE, IP STATIC, SUBNET NAME: The first and last IP addresses in the range or the static IP address assigned to the DHCP client. SUBNET NAME identifies the subnet to which that the IP addresses belong. **Note:** If you add a static IP address for a device that has already been assigned an IP address from a range, you must release the address (type ipconfig /release at a command prompt or turn off the device), delete the address from the DHCP Lease View, and then renew the address (type ipconfig /renew at a command prompt or turn on the device).

IMPORTANT: Ensure that the IP address of the SX-200 ICP controller is outside the range of addresses in the subnet.

LEASE TIME, MAC ADDR/CLIENT ID: LEASE TIME is the length of time the DHCP server grants the DHCP client permission to use a particular IP address. The minimum lease time is 30 minutes. MAC ADDR/CLIENT ID identifies the device either by its MAC address—the MAC ADDR—or a user-specified CLIENT ID in hexadecimal format.

Softkeys

NEW RANGE: Adds a new IP address range to the subnet. Two ranges can be entered per subnet. Pressing the softkey displays prompts for Name, Range Start, Range End, Protocol (display only), Client Class ID must match name, and Lease Time. Enter the Range Start and Range End addresses in dotted decimal format, select "Yes" or "No" for Client Class ID must match name, and specify a Lease Time greater than 30 minutes.

NEW STATIC: Adds a new static IP address to the subnet. Each subnet supports either 11 static IP addresses or 10 addresses if two Ranges are defined. Pressing the softkey displays prompts for Name, IP Address, Protocol, MAC Address, and Client ID. To add a new static IP address, enter a Name, IP Address, and MAC Address or Client ID (do not enter both values). A MAC address must be in the form of six two-digit hexadecimal numbers, separated by colons (e.g., 2C:22:22:4F:FF:FF). All fields except PROTOCOL are user definable.

DELETE: Deletes the entry at the current cursor position.

EXPAND: Opens a subform with details on the entry at the current cursor position. The subform provides the following softkeys used to modify the entry:

YES, NO: These softkeys display when the cursor is in the "Client Class ID must match name" line. Use them to define whether or not the DHCP Class ID is the same as the Client name. If set to YES, the DHCP Server will service the client only if its Class ID matches its name.

PERMANENT, MINUTES, HOURS, WEEKS, YEARS: These softkeys display when the cursor is in "Lease Time" line. Use them to define the length of time in minutes, hours, weeks or years that DHCP client has to use a particular IP address. PERMAMENT grants the client permanent use of the address. The minimum lease time is 30 minutes.

<OPTIONS>: The OPTION softkey is replaced by RANGE OPT or STATIC OPT depending on the currently selected line. Both softkeys open a subform for assigning DHCP options to the selected IP address range or static IP address.

DHCP options are numbered 1 to 255. All such options in RFC 2132 are predefined for you to configure and use. These include options for Routers, DNS servers and DNS domain. The predefined options have standard names and formats, which automatically appear when the option number is entered and the Tab or right-arrow key is pressed. Options not defined in RFC 2132 are named "(user defined)" by default. The Option numbers for Mitel IP phones are 128, 129, and 130.

Clients are granted requested options in order of precedence beginning with their RANGE Options, then heir SUBNET Options, and then finally from the COMMON Options.

QUIT: Exits the subform and returns to the DHCP Parameters subform.

DHCP Lease View Subform

This form appears when the DHCP LEASE softkey is pressed while the cursor is on subnet, an IP address range, or a single static IP address in the DHCP Parameters Subform. Use the form to view all existing leases, and to delete selected DHCP leases. Also use the form to search for DHCP leases by IP address, MAC address, or phone extension.

Figure: DHCP Lease View Subform Layout

Field Descriptions

LEASE IP: The IP address to which the lease has been assigned.

MAC ADDRESS: The MAC address of the device to which the lease has been assigned. In some cases, a message appears in place of a MAC address: IP IN USE means that the lease IP address is unavailable and cannot be assigned (for example, a device may have a static IP address within the DHCP range); RELEASED means that the device has released the lease.

EXT: The extension number of the IP phone associated with the lease. If an extension is not associated, N/A displays.

Type: Identifies the lease type:

TYPE	Name	Description
ST	Static	The lease for this IP address never expires. The static lease type applies to static DHCP IP addresses.
DY	Dynamic	When the lease duration expires, the lease for this IP address will be terminated. The device must then request another lease from the DHCP server. The dynamic lease type applies to IP addresses from DHCP ranges.

TYPE	Name	Description
SR	Server Reserved	The DHCP server has reserved this IP address because a device is already using it, or because a device has released the lease for it but the timeout period has not yet expired. Once the timeout period expires (approx. 8 hours), the IP address is made available to other devices.
CR	Configuration Reserved	The DHCP server has offered the lease for this IP address to a device and is waiting for a response.

Expiry: The date and time that the lease will expire. This field will display PERMANENT if the static IP address or address range has been programmed with a permanent Lease Time in the DHCP Expand Subform.

Softkeys

REFRESH: Causes the view to display the current DHCP leases.

TOP: Displays the first page of DHCP leases. A page can contain up to 12 leases.

BOTTOM: Displays the last page of DHCP leases. A page can contain up to 12 leases.

QUIT: Exits the subform and returns to the DHCP Parameters subform.

PREV PAGE: Displays the previous page of DHCP leases.

DELETE: Pressing the DELETE softkey provides a sub-level of softkeys which can be used to delete one or more DHCP leases. CURRENT SEL deletes the currently selected lease. RELEASED deletes all leases that have been released. IP IN USE deletes all leases for IP addresses are unavailable and cannot be assigned. ALL deletes all leases. Selecting a delete option causes a warning message to appear. To continue, press CONFIRM. To abort, press CANCEL.

SEARCH: Pressing the SEARCH softkey provides a sub-level of softkeys which can be used to search for DHCP leases. Using LEASE IP, users can search for a lease by entering an IP address in dotted decimal format (no dot is required between octets, unless entering a two-character octet). Using MAC ADDR, users can search for a lease by entering a MAC address in hexadecimal format (no colon is required between digits, unless entering a single-character digit). Using EXT., users to search for a lease by entering extension number up to five digits long; permitted characters are 0 to 9, * and #. The item being searched for will display at the top of the list.

Layer 2 Switch Global Configuration Subform (CXi and AX)

Use this subform to program the global settings for the managed Layer 2 Ethernet switch.

For CXi, the first 16 ports are the powered 10/100Base-T/TX LAN ports, and the 17th port is the unpowered 10/100/1GBase-T LAN port. The powered ports are designed for connection of IP phones but can also support PCs. The unpowered port is designed for connection to another Layer 2 switch, but it can also support an IP phone or PC.

For AX, there are 2 10/100 LAN ports.Both the ports are unpowered. These Ports are designed for connection to another Layer 2 Switch, IP Phone (through DC adapter or POE Brick) or a PC.

The internal switch uses 802.1 p/Q VLAN prioritization to ensure the quality of voice calls, routing high-priority voice packets from IP phones ahead of low-priority data packets from PCs and other IP devices. Programming is required only if an expansion switch is connected. For details, see Install a Layer 2 Switch (Voice and Data).

Note: (applicable to CXi)

1. Do not attempt to connect a switch to any of the powered ports (labeled 10/100 802.3af LAN). Doing so may cause a loop.
2. The 10/100 802.3af LAN port LEDs indicate Ethernet status, not power status.

Figure: Layer 2 Switch Global Configuration Subform Layout

Field Descriptions

Tag VLAN on Trunk Port/s: Enables/disables 802.1p/Q VLAN tagging for VLAN 1 on the 10/100/1G LAN port. By default, this parameter is disabled and all VLAN tags are removed before being forwarded by the switch. Enable this parameter only if a VLAN-capable switch is connected to the port.

The AX has two 10/100 LAN ports. The **Tag VLAN on Trunk Port/s** enables/disables VLAN tagging for VLAN 1 on these ports.Enable this parameter only if a VLAN-capable switch is connected to the port.

Note: This field applies to untagged and VLAN 1 tagged packets only; voice VLAN tagging (if enabled) is always preserved.

IGMP Snooping: The purpose of IGMP (Internet Group Multicast Protocol) snooping is to restrain multicast traffic in a switched network. When the feature is enabled (the default), multicast frames are sent only to devices that request them. When the feature is disabled, multicast frames are sent to all devices. Disabling IGMP snooping may cause problems with network or device performance.

Rapid Spanning Tree (RSTP): The Rapid Spanning Tree Protocol (RSTP) enables loop-free, redundant bridging paths by controlling the forwarding state of links between switches. Wherever two paths to a given destination exist, RSTP eliminates the possibility of a loop by forwarding on only one of the paths. The other path is used only in the event of a link or interface failure. RSTP should be enabled only if the internal L2 switch is connected to two or more expansion switches via multiple connections. Disabled by default.

STP Bridge Priority [0-15] (*4096): Enter the Rapid Spanning Tree Bridge priority. The priority is a step value of 0-15 which results in a multiple of 4096 (0x1000) from zero to fifteen (0-15). A higher value means lower priority and less chance of becoming the Root Bridge in the Spanning Tree network. The default is 15 (61440).

Notes:

1. It is recommended to leave the RSTP Bridge Priority set at the system default value of 15 (61440); if this value is to be altered, you should consult with the system administrator.
2. For additional information on RSTP, consult the Engineering Guidelines document on the Mitel Edocs website (http://edocs.mitel.com).

Softkeys

DISABLE, ENABLE: Used to enable or disable TAG VLAN 1 ON TRUNK PORT (17).

QUIT: Exits the form and returns to the IP Networking Forms Menu.

ETHER PORTS: Opens the Ethernet Port Configuration subform where the Layer 2 switch ports can be programmed. For the AX, the PoE status of these ports is disabled cannot be changed because the Layer 2 switch of the AX doesn't have PoE

Ethernet Port Configuration Subform (CX)

Use this subform to configure the Ethernet ports for the Layer 2 switch. The first 16 entries on the subform are for the powered (IEEE 802.3af) 10/100Base-T/TX LAN ports. The final entry is for port 17, the unpowered 10/100/1GBase-T LAN port.

Figure: Ethernet Port Configuration Subform Layout

Field Descriptions

PORT #: System-generated, protected field. Indicates the number of the port. The default database has 17 entries with default values. The first 16 entries are for the powered 10/100Base-T/TX LAN ports, and the 17th entry is for the unpowered 10/100/1GBase-T LAN port.

STATUS: The port state, either enabled or disabled. All ports are enabled by default.

SPEED: The operating speed of the port. Options include AUTO (auto-negotiate), 10 Mb, and 100 Mb. For 1000 Mb (gigabit) speed operation on port 17, set the speed to Auto, and it will auto-negotiate for the gigabit speed. The default is Auto. Manually select a speed only if auto-negotiation fails.

DUPLEX MODE: The duplex mode, either AUTO (auto-negotiate), FULL duplex (simultaneous transmission in both directions), or HALF duplex (transmission in both directions, but only in one direction at a time). All ports default to AUTO. Manually select the duplex mode only if auto-negotiation fails.

FLOW CONTROL: The flow control mode, either AUTO (auto-negotiate), enabled (forces the port to use flow control), or disabled (disables flow control on the port). All ports default to AUTO. Manually select flow control only if auto-negotiation fails.

Note: The Speed, Duplex Mode, and Flow Control settings must be either all AUTO or all manual for each port.

POWER OVER ETHERNET: The PoE mode, either AUTO (auto-negotiate), or disabled. Ports 1 to 16 default to AUTO. Port 17 does not support PoE. Note that PoE can be left enabled even if the connected device does not require PoE support. For diagnostic information, see the PoE Status report in Maintenance.

Note: There is a slight delay before configuration updates take effect. For example, the port will continue to receive power for 10 to 15 seconds after Power over Ethernet is disabled.

Softkeys

DISABLE, ENABLE: Used to enable or disable STATUS and POWER OVER ETHERNET.

AUTO DUPLEX, FULL DUPLEX, HALF DUPLEX: Use these softkeys to select the duplex mode: AUTO (automatically negotiated), FULL (allows simultaneous transmissions in both directions), or HALF (allows transmission in one direction at a time).

AUTO FLOW, ENABLE FLOW, DISABLE FLOW: Use these softkeys to select the flow control mode: AUTO (automatically negotiated), ENABLE (forces the port to use flow control), or DISABLE (disables flow control on the port).

AUTO, 10, 100, 1000: Use these softkeys to select the port speed: AUTO (automatically negotiated), 10, 100, or 1000 Mbps. 1000 Mbps is available only on port 17.

QUIT: Exits the form and returns to the IP Networking Forms Menu.

Router Subform (CXi)

If the LAN is divided into multiple segments, the CX can be programmed as an ICMP router discovery server. In this role, it advertises its availability by exchanging "ICMP Router Discovery" messages with hosts on the network. This process enables the hosts to dynamically discover the CX and other routers on the network, and to switch to a backup router in the event of a network failure.

Figure: Router Subform Layout

Field Descriptions

ICMP ROUTER DISCOVERY SERVER: Indicates whether the ICMP Router Discovery server is enabled for the LAN interface. When it is enabled, the server periodically sends router advertisement messages to the network. It also responds to router discovery messages sent from hosts. Enabled by default.

Softkeys

DISABLE, ENABLE: Used to enable or disable ICMP ROUTER DISCOVERY SERVER.

NETWORK LIST: Opens the Network List subform where routes can be viewed and added.

QUIT: Exits the form and returns to the IP Networking Forms Menu.

Network List Subform (CXi)

The Network List consists of routes to destination networks on the LAN. When the CX receives a packet from a host on the LAN, it checks the table. If a route is found, the CX forwards the packet to the system gateway, which is either CX itself (i.e., when the System Gateway IP = L2 Switch IP), the APC (MAS), or a router connected to the local subnet. If no route is found, the CX forwards the packet to the Internet through its WAN interface.

The administrator can add up twenty (20) routes to the Network List. The list includes three default routes to private subnets (RFC 1918 networks):

- 172.016.000.000 / 255.240.0.0
- 192.168.000.000 / 255.255.0.0
- 010.000.000.000 / 255.0.0.0

Figure: Network List Subform Layout

Field Descriptions

The header displays the gateway IP address, which is the address of the router on the local subnet. All entries on the Network List point to this gateway.

NETWORK: Destination IP address for the remote subnet. The IP address must be in the form of four three-digit numbers in the range 0 to 255 -- e.g., 10.234.191.0.

BIT MASK: Subnet mask (netmask) for the remote network. The subnet mask determines the maximum number of hosts on the subnetwork. Like an IP address, the subnet mask contains four bytes (32 bits) and is written in dotted-decimal format -- e.g., 255.255.255.0.

Softkeys

ADD: Adds a new route to the Network List. Up to 20 routes can be added. To add a new route, press the ADD softkey and enter the destination IP address and network mask for NETWORK and NETMASK, respectively.

DELETE: Deletes a route from the Network List.

QUIT: Exits the form and returns to the IP Networking Forms Menu.

Internet Gateway Subform (CXi)

The Internet Gateway subform is used to enable the WAN interface and provide Internet connectivity settings for Static IP, DHCP, or PPPoE. If the system contains an Application Processor Card (APC) with MAS software, the subform is used to designate the APC as the Internet Gateway.

Notes:

1. To use the WAN interface, you must enable the system option 83 (Internet Gateway) in CDE form 04.
2. All three WAN IP Methods (PPPoE, DHCP, and Static IP) can be programmed, but only one WAN IP Method can be enabled.
3. If PPPoE or DHCP address assignment fails, the system generates a MAJOR alarm that is recorded in the maintenance log. Assigning an IP address to the WAN interface clears the alarm.

4. IP trunking is not supported through the WAN interface, or through any other interface that performs NAT.
5. If the WAN interface is programmed as a DHCP client, it may be necessary to renew the DHCP lease to obtain an IP address. To do this, disable the WAN State for the Internet Gateway, then enable it again.
6. Speed, duplex mode, and flow control are automatically negotiated by the WAN interface. If manual settings are required, install a programmable switch between the cable modem/DSL router and the WAN interface.
7. If the CXi contains the optional APC (MAS), enable the WAN State and select APC as the the WAN IP method. Program the internet connectivity settings (static IP, PPPoE, or DHCP client) on the MAS.

Figure: Internet Gateway Subform Layout

Field Descriptions

WAN STATE: The current state of the WAN interface, either enabled or disabled. Requires system option 83. Disabled by default.

WAN IP METHOD: The method that the WAN interface is programmed to obtain its IP address, either STATIC, DHCP, PPPoE or APC. Default is DHCP client.

IP ADDRESS: If the WAN interface is programmed with a static IP address and gateway, this field displays the IP address.

SUBNET MASK: If the WAN interface is programmed with a static IP address and gateway, this field displays the subnet mask for the address.

EXTERNAL DEFAULT GATEWAY: If the WAN interface is programmed with a static IP address and gateway, this field displays the default gateway address.

Softkeys

DISABLE, ENABLE: Used to enable or disable WAN Ethernet interface of the CX.

APC: use this softkey to select the Application Processor Card (Managed Application Server) as the Internet Gateway. If selected, the user must program internet connectivity settings (static IP, PPPoE, or DHCP client) on the MAS instead of CDE.

PPPoE: Use this softkey to select PPPoE as the WAN IP Method. Then press the PPPoE_CFG softkey to program PPPoE.

DHCP CLIENT: Use this softkey to select DHCP client as the WAN IP Method. Then press the DHCP_CFG softkey to program DHCP.

Note: If the WAN interface is programmed as a DHCP client, it may be necessary to renew the DHCP lease to obtain an IP address. To do this, disable the WAN Access and then enable it.

STATIC IP: Use this softkey to select static address assignment as the WAN IP Method. Then press the STATIC_CFG softkey to program the static IP address.

PPPoE_CFG: Opens the PPPoE Configuration subform where PPPoE can be programmed.

DHCP_CFG: Opens the DHCP Configuration subform where DHCP client parameters can be programmed.

STATIC_CFG: Opens the Static IP Configuration subform where static IP address information can be programmed.

QUIT: Exits the form and returns to the IP Networking Forms Menu.

PPPoE Configuration Subform (CXi)

This subform appears when PPPoE_CFG is pressed in the Internet Gateway subform; use it to program PPPoE parameters for the WAN Interface.

Figure: PPPoE Configuration Subform Layout

Field Descriptions

USER NAME: PPPoE user name specified by your Internet Service Provider (ISP). This field can be edited and can contain up to 32 characters. The characters ', ", `, and (space) are invalid.

PASSWORD: PPPoE password specified by your ISP. This field can be edited and can contain up to 16 characters. The characters ', ", `, and (space) are invalid.

Softkeys

QUIT: Exits the subform and returns to the Internet Gateway subform.

DHCP Configuration Subform (CXi)

This subform appears when DHCP_CFG is pressed in the Internet Gateway subform; use it to program DHCP client parameters for the WAN Interface.

Figure: DHCP Configuration Subform Layout

Field Descriptions

CLIENT NAME: The DHCP client name specified by your Internet Service Provider (ISP). The name can contain up to 20 characters, and cannot contain quotes or spaces.

CLIENT ID / ETHERNET ADDRESS: Depending on which DHCP authentication method has been selected, enter either the client ID or Ethernet (MAC) address specified by your ISP. For an Ethernet address, the value is an address (11-17 characters) in hexadecimal notation separated by colons or spaces. For a Client ID, the value is an ASCII string (1-32 characters without quotes). If this field is left blank, the WAN Ethernet MAC address will be sent to the DHCP server.

AUTHENTICATE BY: The current DHCP authentication method, either Client ID or Ethernet Address.

Softkeys

ETHERNET ID: Press this softkey to select Ethernet ID as the DHCP authentication method.

CLIENT ID: Press this softkey to select Client ID as the DHCP authentication method.

QUIT: Exits the subform and returns to the Internet Gateway subform.

Static IP Configuration Subform (CXi)

This subform appears when STATIC_CFG is pressed in the Internet Gateway subform; use it to program the WAN Interface with a static IP address and default gateway.

Figure: Static IP Configuration Subform Layout

Field Descriptions

IP ADDRESS: The static IP address of WAN interface. The WAN IP Address cannot conflict with the DHCP Static IP Address and Range IP Address, and it must be on a different subnet than the System IP Address. This field can be edited.

SUBNET MASK: The subnet mask of WAN interface. This field can be edited.

DEFAULT GATEWAY: The external default gateway address. This field can be edited.

Softkeys

QUIT: Exits the subform and returns to the Internet Gateway subform.

Firewall Global Configuration Subform (CXi)

Use this subform to program and verify the firewall settings for the Internet Gateway. The firewall examines all packets attempting to access the internal network from the Internet. Unless a packet is part of an existing connection, or match specific TCP or UDP port programmed for forwarding, it is declared "unknown." All unknown packets are logged, and then either dropped or rejected. The firewall can also be programmed to allow outbound VPN tunnels with PPTP and IPSec pass-through, and inbound connections with IP Port Forwarding.

Figure: Firewall Global Configuration Subform Layout

Field Descriptions

Logging State: The current logging state of the firewall, either enabled or disabled. Enabled by default.

PPTP Pass Through: Indicates whether outbound PPTP VPN tunnels are allowed to pass through the firewall from the LAN to the Internet. Enabled by default.

IPSEC Pass Though: Indicates whether outbound IPSec VPN tunnels are allowed to pass through the firewall from the LAN to the Internet. Enabled by default.

Action for Unknown Packets: Indicates whether the firewall drops or rejects unknown packets. By default, unknown packets are dropped. Dropped and rejected packets are logged for ports that have logging enabled.

Note: The firewall sends an "ICMP port unreachable" message to the sender after rejecting a packet. No response is sent for dropped packets.

TCP do not log ports: TCP packets received on these ports do not generate logs. This field is editable. Enter a single port as <port>. Enter a range of ports as <port-port>. Use commas to separate the entries (e.g., 135,137-139). By default, TCP ports 135, 137-139 are not logged.

UDP do not log ports: TCP packets received on these ports do not generate logs. This field is editable. Enter a single port as <port>. Enter a range of ports as <port-port>. Use commas to separate the entries (e.g., 135,137-139). By default, UDP ports 137-139, 520 are not logged.

Softkeys

ENABLE/DISABLE: Used to enable or disable the Logging State, PPTP Pass Through, and IPSEC Pass Through.

DROP: Pressing this softkey causes the firewall to discard unknown packets. No reply is sent to the sender.

REJECT: Pressing this softkey causes the firewall to discard unknown packets and send an "ICMP port unreachable" response to the sender.

FIREWALLTBLE: Opens the Firewall Forwarding subform where IP Port Forwarding can be programmed.

QUIT: Exits the subform and returns to the Internet Gateway subform.

Firewall Forwarding Subform

Use this subform to program the firewall forwarding table to allow external traffic to reach resources on the internal network. The firewall forwarding table can contain up to 40 entries. Each entry consists of a protocol (TCP or UDP) and port number combination on the Internet Gateway (WAN interface), and an IP address and port number combination on the internal network. Ports can be entered a single number or as ranges of numbers. See IP Port Usage for a list of port numbers used by the SX-200 ICP.

After an entry has been added to the table, a host on the Internet can send a packet to the WAN interface of the CXi for the specified protocol/port number combination. The firewall readdresses the packet, sending it to the IP address/port number combination of the actual resource or service on the internal network. This feature is referred to as IP Port Forwarding or NAT Redirect.

Notes:

1. Each protocol/port number combination on the WAN interface must be unique.
2. Some protocols (for example, FTP) cannot be forwarded with IP Port Forwarding.
3. The port number ranges on the two interfaces must be the same size. For example, if the WAN interface has a three-number range, the LAN interface must have a three-number range.

Figure: Firewall Forwarding Subform Layout

Field Descriptions

Protocol: The WAN interface protocol, either TCP or UDP.

Start Port: The WAN interface starting port number. This field can be edited.

End Port: The WAN interface ending port number. This field can be edited.

Internal IP: The IP address on the internal network to which external packets are sent. This field can be edited.

Int Start Port: The starting port number on the internal network to which external packets are sent. This field can be edited.

Int End Port: The ending port number on the internal network to which external packets are sent. This field can be edited.

Softkeys

ADD: Press this softkey to add an entry to the firewall forwarding table. For each entry, specify the WAN interface settings (TCP or UDP protocol; single port number or range of port numbers), and LAN interface settings (IP address; single port number or range or port numbers). The table can contain a maximum of 40 entries.

DELETE: Press this softkey to delete an entry from the firewall forwarding table.

TCP, UDP: These softkeys become active after pressing the ADD softkey. Use them to specify the protocol for the WAN interface.

QUIT: Exits the subform and returns to the Internet Gateway subform.

PPTP Remote Access Subform (CXi)

Use this subform to program the Internet Gateway as a PPTP server for a remote client on the Internet.

Figure: PPTP Remote Access Subform Layout

Field Descriptions

Dial In Username: Username that the server uses to authenticate the remote client. The field can contain up to 32 characters; the characters ', ", `, and (space) are invalid. The default is blank.

Dial In Password: Password that the server uses to authenticate the remote client. This field can be edited. Use a combination of upper and lower case letters, punctuation, and numbers (for example, luv2sk!). The field can contain up to 16 characters; the characters ', ", `, and (space) are invalid. The default is blank.

PPTP LAN Client IP Address: IP address that the remote PPTP client uses on the LAN. The IP Address must be on the same subnet as the System IP address. It cannot conflict with System IP address, L2 Switch IP address, and DHCP Server configuration. If PPTP Access is enabled, the IP address cannot remain blank or zero. The default is blank.

PPTP State: Indicates whether the PPTP server is enabled or disabled. Disabled by default.

Softkeys

ENABLE/DISABLE: Use these softkeys to enable/disable the PPTP server.

QUIT: Exits the subform and returns to the Internet Gateway subform.

Form 48 - Voice Networking

Use this form to specify

- an IP node number and Bay number for each SX-200 ICP (including this one), 3300 ICP or SX-200 IP Node that is networked for voice only using IP trunks.
- the IP Node number and IP address of each remote node (other SX-200 ICPs, SX-200 IP Nodes, 3300 ICPs) in the voice network (to a maximum of 100 IP nodes)

You also use this form to manage the amount of bandwidth used by IP Trunking. You do this by specifying the maximum number of calls for each IP node in the network.

Figure: Form 48 Layout

Field Descriptions

IP NODE NUM: This field contains the numbers of each node in the IP voice network (a maximum of 100 nodes can be assigned). The number can be any number from 1-255. The numbers are entered each time an IP node is added to the system.

Note: IP Node Numbers must be consistent for all nodes in the network. Once a node number has been assigned to a particular node all nodes should refer to it in the same manner.

IP BAY: This field contains the Bay number of local IP nodes. For the local SX-200 ICP, this defaults to Bay 1 (the virtual "IP bay"). The numbers can have a value from 1-7. You cannot enter the Bay numbers of remote IP Nodes.

IP ADDRESS: This field specifies the IP address of each remote IP Node. These include other 200 ICP systems, SX-200 IP Nodes, and 3300 ICP systems. The default values are 000.000.000.000 and are displayed as a blank field. You cannot enter the IP Addresses of local IP Nodes.

MAX CALLS: This field contains the maximum number of calls allowed to each remote IP Node. You can enter a value from 1 to 60. See the description for the maximum number of calls in CDE Form 23.

Note: MAX CALLS must be greater than zero (the default); otherwise, no calls are allowed to the node.

COMMENTS: The comments field can store 20 characters.

Softkeys

INSERT: This softkey is used to add new IP Nodes to the form.

FIND IP NODE: This softkey is used to find IP Nodes in the form by using the IP Node number.

PING: This softkey displays after an IP Node has been found with the FIND IP NODE softkey. Pressing PING sends an echo request to the selected IP Node. The response indicates whether the IP Node is available on the network.

DELETE: This softkey is used to delete an IP Node from the form.

TOP: This softkey displays the top of the list of IP Nodes.

BOTTOM: This softkey displays the bottom of the list of IP Nodes.

Standard softkeys **ENTER** and **QUIT** are provided.

Form 49 - Voice Mail Options

Use this form to set up system-level options for embedded voice mail.

Note: Embedded voice mail is a purchaseable FOR option. The option is required to enable this form.

Figure: Form 49 Layout

Field Descriptions

DEFAULT LANGUAGE: Displays the language to use for voice mail prompts.

ALTERNATE LANGUAGE: Displays the second language for bilingual prompting. The language cannot be changed unless System Option 121 (Voice mail License for Bilingual Prompts) is enabled.

BILINGUAL OPTION: Enables or disables bilingual prompting (requires System Option 121).

LANGUAGE CHANGE NUMBER: Displays the digit (1-8; default 8) that users dial to hear subsequent prompts in the alternate language (requires System Option 121).

FAX DESTINATION NUMBER: The extension or hunt group to which the auto attendant directs FAX calls or the single-digit mailbox used for auto attendant transfers to the Fax machine. The extension or hunt group number must correspond to an ONS extension or hunt group programmed in Forms 9 and 17 respectively. The transfer destination--i.e., the Fax extension--for a single-digit mailbox is automatically programmed in Form 50 provided that either one-digit long, or the length specified for Length of Mailbox Numbers in Form 49.

SEND NOTIFICATION CALLS: Enable this option to have the system call a mailbox owner's pager or other device when a message is left in his or her mailbox. Notification takes place only if the option is enabled here and in mailbox owner's mailbox settings. The number that the system calls is specified in his or her mailbox settings.

DIGITAL PAGER CALLBACK NUMBER: The telephone number that will display on digital pagers when the system notifies a mailbox owner. The number can be up to seven digits long and can include 0 to 9, *, #, and , (comma).

AUTO ATT XFER TO ANY EXTN: Enable this option to allow the Auto Attendant to transfer calls to any extension in the system.

Note: Auto Attendant Transfer only works if extension numbers and mailbox numbers have the same number of digits.

AUTO ATT XFER RESTRICTIONS: A programmable field used to prevent callers from accessing system trunks via the Auto Attendant. Access is prevented by programming the system to deny transfers when the leading digit dialed matches the first digit of a trunk group access code. Up to 10 digits can be entered. Separate each digit with a comma.

DIRECTORY VOICE PROMPT: Specifies whether callers using the Phonebook or Auto Attendant to reach a mailbox owner, are prompted to dial the owner's first or last name as defined in Form 50. For special considerations regarding the naming of mailboxes, see Phonebook.

AUTOMATIC ROUTE SELECT CODE: Read-only field. Reserved for future implementation.

GENERATE ACCOUNT CODES IN SMDR: Reserved for future implementation.

PERSONAL CONTACT NUMBERS: Indicates whether callers will be prompted (after the owner's greeting), to reach a mailbox owner's alternate numbers (cell-phone, pager, fax etc) by pressing various keys. If System Option 122 (Voice mail License for Personal Contact Numbers) is disabled, then no softkey will appear when positioned at this line, the STATUS will be DISABLED, and this option cannot be modified.

DID SERVER: Reserved for future implementation.

AUTOMATIC DID NUMBER ASSIGNMENT: Reserved for future implementation.

SYNCHRONIZATION OF GREETINGS: Indicates when the Open and Closed auto attendant greetings play. If set to SYSTEM DAY/NIGHT MODE (the default), the Open and Closed greetings will play when system TENANT 1 is in Day Service and Night Service respectively. If set to BUSINESS HOURS, the times entered for OPEN TIME and CLOSE TIME will govern when the greetings play.

OPEN TIME/CLOSE TIME (HH:MM) (SUN..SAT): Indicates for each day of the week the time in 24-hour format that business hours start and end. Enter 00:00 in both fields if the business is closed all day.

LENGTH OF MAILBOX NUMBERS (2-5): Mailbox numbers (excluding single-digit mailboxes) should correspond to the length of extension numbers programmed in Form 9; otherwise, the system cannot automatically create mailboxes whenever new extensions are added. The mailboxes will have to be manually created using Form 50.

MAILBOX LOCKOUT: If this option is enabled, the system locks the mailbox after the user makes three attempts to enter an invalid password. When lockout occurs, the system plays a message that instructs the user to contact the system administrator, who can reset the password from CDE Form 50 (Mailboxes). This option is disabled by default.

LENGTH OF PASSWORDS (3-6): Specifies the number of digits required in mailbox passwords, including the Administrator's mailbox.

PMS PROTOCOL: The protocol used by the hotel/motel PMS system to communicate with the voice mail system.

Note: Voice mail integration with PMS is a purchaseable FOR option (Option 124) and is enabled in Form 04.

SMTP SERVER ADDRESS: The IP address of the SMTP server (requires System Option 126, Email Messaging). The server must be an SMTP server that is capable of forwarding emails.

SMTP SERVER USERNAME: Enter the username required for SMTP authentication.

SMTP SERVER PASSWORD: Enter the password required for SMTP authentication.

IMAP SERVER ADDRESS: The IP address of the IMAP server (requires System Option 126, Email Messaging).

Softkeys

TOP, BOTTOM: Positions the cursor at the first or last option in this form.

RAD SETUP: Opens a subform for configuring RAD (Recorded Announcement Device) greetings.

FRENCH/SPANISH/ENGLISH/ENG OVERLAID: These softkeys appear when the cursor is on the DEFAULT LANGUAGE or ALTERNATE LANGUAGE fields. Use them to change the language for voice mail prompts.

Note: Select English to provide alphabetic prompts ("Press P to play...") or English-Overlaid to provide numeric prompts ("Press 7 to play...").

DID SETUP: Reserved for future implementation.

DELETE FIELD: Restores the FAX, DIG PAGER, AUTO ATT, or ARS field to its default, blank. The softkey appears when the cursor is in the field and the field is programmed.

QUIT: Returns to the CDE main menu.

Subform 49 - RAD Setup

This form appears when the RAD SETUP softkey is pressed in Form 49. Use the subform to assign sets of messages (called "greetings") to the RAD ports and to specify how often the sets play. Up to 30 RAD greeting sets can be assigned, each containing from one to five greetings. 200 greetings can be recorded, although only 150 can be in use at any one time. Recording is done using a telephone and the Administrator's mailbox. For more information about RADs and how to implement them, see Recorded Announcement Devices.

Figure: RAD Setup Subform Layout

Field Descriptions

RAD SET: A read-only field showing the 30 available RAD sets. For each set programmed, enter its number in the Hunt Group Options subform for Recording hunt groups (Form 17).

GREETING 1 ... GREETING 5: Shows a greeting number from 1 to 200. The greetings play in order from 1 to 5. The same greeting can be assigned to multiple RAD sets. Leave the field for greetings that are not required blank.

TIMES TO PLAY: Specifies the number of times to replay the greeting set. Accepted values are 1 to 98, or FOREVER (entered via a softkey) which plays the set continuously until the caller hangs up.

Softkey

DELETE FIELD: Appears when cursor is in a non-blank GREETINGS field. Deletes the current field only.

FOREVER: Appears when the cursor is in the TIMES TO PLAY field and the field is blank or contains a number. Used to specify continuous play for the selected RAD set.

QUIT: Returns to the Voicemail Options form.

Form 50 - Mailboxes

Initial mailbox setup is done automatically for each new extension added in Form 09 if the Option 277 (Automatic Mailbox Creation) is enabled in the extension's COS. Form 50 is used to customize voice mail operation following initial setup.

If Automatic Mailbox Creation is enabled, the type of mailbox created—Extension or Guest—is determined by the state of System Options 107 (Lodging), 108 (Property Management System) or 124 (Voice Mail Property Management System). If any of those option are enabled, the system creates Guest-type mailboxes; otherwise, it creates Extension-type mailboxes.

Figure: Form 50 Layout

Field Descriptions

MBOX#: A mailbox number. The mailbox number is usually the same as the mailbox subscriber's extension number. Do not use the 0 (reserved for operator), or the Language Change mailbox number (defined in Form 49). Mailbox numbers must be one digit long or the length specified by the "Length of Mailbox Numbers" option in Form 49. Only the digits 0-9 are allowed in the mailbox number; * and # are not allowed. Mailbox numbers cannot be changed once they are entered, but they can be deleted. The maximum number of mailboxes allowed is defined by System Option 125 (Form 04).

Note: Mailboxes with a leading digit of 9 can be used.

TYPE: Mailbox type as follows:

EXTEN (Extension) - the mailbox type usually associated with an extension; it has the capability to record, send, and listen to messages.

GUEST - the mailbox type assigned to extensions if System Option 124, 107, or 108 are enabled in Form 04. Guest mailboxes are used for listening to messages.

INFO (Information) - for playing informational greetings only. For example, the message (or greeting) for this mailbox could recite the company's business hours or driving directions to a company's location. This type of mailbox does not take messages.

MESSAG (Message Only) - for recording and listening to messages. Message-only mailboxes are for people who do not have telephones or who are frequently away from the office. The mailbox may be, but does not have to be, associated with an extension number. If it is not associated with an extension number, then the system cannot turn on a message light; instead, the mailbox owner must periodically check for messages. Unlike an extension mailbox, the phone does not ring after the caller enters the extension number. Instead, the caller is immediately prompted to leave a message.

TRANSF (Transfer Only) - for playing instructions to callers to dial another number or to dial 0 to reach the company operator. This type of mailbox is associated with an extension but does not take messages. When a caller dials this extension he will hear, "You are being transferred to <mailbox name>." If the extension is busy or unanswered, the caller is returned to the company greeting. No message is taken.

MENU (Menu Tree) - allow you to set up a hierarchical menu structure for multi-level auto attendant (MLAA) operation. In an MLAA system, callers reaching the Auto Attendant are routed from the main menu through to one or more additional sub menus until their call is answered. The system plays greetings for this type of mailbox and transfers callers from the mailbox to other mailboxes or extensions.

ADMIN (Administrator) - for accessing administrative functions such as greetings recording.

F DESK (Front Desk) - Front desk staff in hotels/motels uses the front desk mailbox to administer the guest mailboxes.

For more information about the mailbox types and their uses, see Multiple Mailbox Types.

NAME: Up to 10 characters (including a single space). Identifies the user of this mailbox. Enter the name (first name first or last name first, separated by a single space) according to the option selected for Directory Voice Prompt in Form 49. For special considerations regarding the naming of mailboxes, see Phonebook.

Note: Only the first six characters in the name are used in the personnel directory (Mailbox 9).

EXTENSION: Where voice mail directs calls for this mailbox. This is normally an extension number but any valid destination, including external numbers, hunt group numbers, and system speed dial codes, up to 13 characters (* and # are allowed) is accepted. Do not use 0 (reserved for the operator). No error checking is done to validate the entry.

OPERATOR: The destination to which calls are to be directed when a caller presses"0" while listening to a greeting for this mailbox. Any valid destination from 1 to 13 characters (* and # are allowed) including extensions, external telephone numbers, hunt group numbers, and system speed dial codes can be entered. The default sends calls to Mailbox 0, the mailbox usually reserved for the company-wide operator. Specifying a destination other than the company-wide operator accommodates, for example, users who want callers to transfer to a departmental secretary, rather than a company-wide operator.

Note: For multi-tenant applications, keep the default (Mailbox 0) and program Priority Dial 0 Routing in Form 19 to route each tenant's calls to the required destination.

LANG: The Language for the voice mail prompts that the user of this mailbox hears. Cannot be changed unless System Option 121 (Voicemail License for Bilingual Prompts) in Form 04 and the Bilingual option in Form 49 are enabled. The selection of languages is whatever is specified as the Default and Alternate languages in Form 49. The choices are: ENG (English-NA), EOVL (English-Overlaid), FRE (French), SPA (Spanish), and SYST (system default).

ENV: Indicates whether or not to play the "envelope" (priority, date, time, and caller ID) for each message. If not played automatically the user can still hear the Date/Time stamp by pressing a key on the phone.

MAX: Maximum number of messages that can accumulate in the mailbox. Valid entries: 0-100. Default is 10.

DEL: Number of days saved messages are kept before they are automatically deleted by the system. Valid entries: 0-98."0" means never delete and displays as a blank field. The default is 15. **Note:** New messages can remain in the mailbox indefinitely.

NEW: A read-only field that displays the number of new messages (from 0-100) in the users mailbox.

OLD: A read-only field that displays the number of saved messages (from 0-100) in the users mailbox.

Softkeys

SHOW MORE: Displays more softkeys to set up message notification and to forward voice mail to e-mail for individual mailboxes. For more information, see Message Notification and Unified Messaging.

FIND MAILBOX: Used to search the form for a specific mailbox. Press the softkey, and then enter the mailbox number.

FRENCH/SPANISH/ENGLISH/ENG OVERLAID: Use these softkeys to change the voice mail prompt language for individual mailboxes. The languages are installed during initial installation and can be updated on Form 49 (Voice Mail Options).

Note: Select English to provide alphabetic prompts ("Press P to play...") or English-Overlaid to provide numeric prompts ("Press 7 to play...").

LANG: The Language for the voice mail prompts that the user of this mailbox hears. Cannot be changed unless System Option 121 (Voicemail License for Bilingual Prompts) in Form 04 and the Bilingual option in Form 49 are enabled. The selection of languages is whatever is specified as the Default and Alternate languages in Form 49. The choices are: ENG (English), FRE (French), SPA (Spanish), and SYST (system default).

INSERT: Used to insert a new mailbox. The new mailbox will be sorted numerically when ENTER is pressed.

TOP, BOTTOM: Positions cursor at top or bottom of form.

SET PASSWORD: Used to change the password for an existing mailbox. The password is forced to be the length specified by the "Length of Passwords" option in the Form 49. Only the digits 0-9 are allowed in the password. * and # are not allowed.

Default passwords for specific mailboxes are as follows:

PASSWORD LENGTH	**3**	**4**	**5**	**6**
OPERATOR (0)	123	1234	12340	123400
ADMIN (999)	864	8642	86420	864200
MANAGER (999)	648	6483	64830	648300
All other mailboxes (See Note)	111	1111	11111	111111

Note: Only the user of a mailbox should change the default passcode. The system automatically prompts users to record their name and a greeting the first time they access their mailbox. The prompt repeats with each successive access until the default password is changed. If a user forgets his or her passcode, it can be reset by entering the default code (all 1s) or by entering a new code in this form. Entering a new code here saves users from having to record their greeting again.

DELETE: Used to delete a mailbox. Mailbox 0 and the ADMIN mailbox cannot be deleted.

CONTACTS or MENUS: Available for Menu, Extension, Guest and Front Desk type mailboxes. Opens a subform for setting up Personal Contacts or Multi-level Auto Attendant menu trees.

QUIT: Exits to the main form.

Subform 50 - Other

This form appears after pressing SHOW MORE and OTHER in Form 50. Use the form to set up message notification for individual mailboxes.

Figure: Other Subform Layout

Field Descriptions

MBOX#: A read-only field of mailbox numbers from the main MAILBOXES form.

NOTIFICATION NUMBER: The telephone number (up to 35 digits) that is called to notify the mailbox user of waiting messages. Prefix the number with "9" or other digit required to access an outside line -- e.g., 95551234. Valid digits include 0 to 9 and *.

NOTIF TYPE: Notification Type. Specifies the device that the system calls to notify the mailbox user. The user can receive notification on any one of at any one of following devices: an internal extension (EXTENSION), digital pager (DIG PAGER), message pager (MSG PAGER), tone pager (TON PAGER), and outside telephone (TELEPHONE). For more information regarding the use of pagers, see Message Notification.

N SCHED: Notification Schedule. Indicates when the system notifies the user. Choices are

ON - notifies the user when the message arrives.
OFF- does not notify the user, effectively disabling notification
OPEN - notifies the user only during open business hours.
CLOSE - notifies the user only during closed business hours.

U ACC: User Access. Enable this option to allow the user to change his or her pager number and to specify whether notification is provided on all messages or urgent messages only If disabled, only the system administrator can change the number. Only users can specify which type of message they get notified about.

Note: This option does not work for guest mailboxes.

RESERVED: For future use.

Softkeys

FIND MAILBOX: Used to search the form for a specific mailbox. Press the softkey, and then enter the mailbox number.

TOP, BOTTOM: Positions cursor at top or bottom of form.

QUIT: Exits to the main form.

Subform 50 - Email

This form appears after pressing SHOW MORE and MAILBOX in Form 50. This form enables voice mail for individual mailboxes to be forwarded to e-mail using either SMTP or IMAP (Standard Unified Messaging). For mailboxes that use SMTP, this form enables Record a Call Messages to be forwarded to a different e-mail address than voice messages. See Unified Messaging for a feature description.

Notes:

1. For voice message forwarding to occur, System Option 126, Email Messaging must be enabled in Form 04 and the SMTP and/or IMAP server address(es) must be programmed in Form 49. For SMTP, a system Hostname must also be defined in Form 47, Subform 01. The system Hostname must be a valid domain host name of 49 or fewer characters that has been registered in your DNS, or has been listed in your SMTP e-mail server's Hosts file. For IMAP, a 6000 MAS Unified Messaging blade is required. The blade must be programmed to interact with embedded voice mail on the SX-200 ICP, and with IMAP-enabled e-mail clients.
2. For Record a Call forwarding to occur, System Option 87 (Record a Call) must be enabled in Form 04.
3. Disabling System Option 126, Email Messaging sets all mailboxes to NONE and prevents users from forwarding voice messages to e-mail.
4. System Option 137, Mark SMTP Forwarded Voicemails as Read, controls the state of forwarded, saved voice messages. When enabled (default), messages change from "new" to "read" upon forwarding. When disabled, messages remain "new" after forwarding. Only new messages cause the user's MWI lamp to flash.
5. A maximum of 350 mailboxes can have forwarding of voice mail messages enabled.
6. When the system is upgraded from Release 1.0 to 2.0, mailboxes set to FWD YES will change to FWD ALL.
7. To use the voice mail to email feature effectively, we recommend that you install an internal hard drive in the controller instead of the internal flash card. If the mailbox is not programmed to delete messages after emailing them, then two copies of each message are stored in voicemail - the initial message and the wave file that is sent to the email server. Wave files get removed during the voicemail housecleaning operation provided there are more than 50 messages in the folder. If you have a lot of mailboxes and they all use the voicemail to email feature, then a flash card could get filled.

Figure: Email Subform Layout

Field Descriptions

MBOX#: A read-only field of all EXTENSION, MESSAGE ONLY, GUEST and FRONT DESK mailbox numbers from the MAILBOXES form.

FWD: Indicates the SMTP or IMAP/Standard Unified Messaging e-mail forwarding option for the mailbox. The SMTP options are MAN (manual forwarding; voice messages saved), MN + D (manual forwarding; voice messages deleted), ALL (automatic forwarding; voice messages saved), AL + D (automatic forwarding; voice messages deleted). The IMAP option is UNI (Standard Unified Messaging; messages synchronized with IMAP server). NONE indicates that the mailbox is not programmed to forward voice mail to e-mail.

EMAIL ADDRESS: For mailboxes that use SMTP, this field specifies the e-mail address to which messages are sent.

Softkeys

FIND MAILBOX: Used to search the form for specified a mailbox number.

TOP, BOTTOM: Positions cursor at the top or bottom of the form. These softkeys appear only if the form has sufficient room.

MANUAL FWD: This softkey appears when the cursor is at FWD. Pressing MANUAL FWD allows the user to forward individual voice messages to e-mail using a telephone. The e-mail messages are sent to an SMTP server, and the original voice messages are saved in the user's mailbox.

MAN FWD DEL: This softkey appears when the cursor is at FWD. Pressing MAN FWD DEL allows the user to forward individual voice messages to e-mail using a telephone. The e-mail messages are sent to an SMTP server, and the original voice messages are deleted from the user's mailbox.

ALWAYS FWD: This softkey appears when the cursor is at FWD. Pressing ALWAYS FWD causes the system to forward all voice messages to e-mail. The e-mail messages are sent to an SMTP server, and the original voice messages are saved in the user's mailbox.

ALL FWD DEL: This softkey appears when the cursor is at FWD. Pressing DEL ALWAYS causes the system to forward all voice messages to e-mail. The e-mail messages are sent to an SMTP server, and the original voice messages are deleted from the user's mailbox.

Notes:

1. The user can update the forwarding option by entering a telephone command. The CDE displays the current forwarding option.
2. Because MAN FWD DEL and ALL FWD DEL result in the original voice messages being deleted, these options should be tested before they are implemented. Make sure that the SMTP server can reliably forward e-mails containing voice messages. Otherwise, some voice messages may be lost.

COPY ADDR: Allows an e-mail address to be copied from one mailbox to another.

UNIFIED FWD: This softkey appears when the cursor is at FWD. Pressing UNIFIED FWDenables the user's voice mail to be synchronized with the IMAP server on the 6000 MAS. Mailboxes that use IMAP do not require an e-mail address. (The e-mail address is added on the IMAP server, along with other user account information.)

SHOW MORE: For mailboxes that use SMTP, pressing this softkey changes the EMAIL ADDRESS FOR SMTP FORWARD OF VOICEMAIL column to EMAIL ADDRESS FOR RECORD A CALL. Enter an e-mail address in this column to send Record a Call messages to a different address than voice messages. Press SHOW MORE to return to the regular Email subform. Record a Call Requires option 87. **Note**: If no address is entered, Record a Call messages are sent to the e-mail address for voice mail forwarding.

DELETE: For mailboxes that use SMTP, pressing this softkey deletes the selected e-mail address.

QUIT: Returns to the CDE main menu.

Subform 50 - Contacts or Menus

Use this form to set up routing for menu-type mailboxes (used to implement Multi-Level Auto Attendant) and to set up Personal Contacts for extension-, guest- and front desk-type mailboxes.

Figure: Contacts or Menus Subform Layout

Field Descriptions

MBOX#: A read-only field that displays mailbox numbers from the MAILBOXES form.

0,1...9: Shows where calls go when a caller presses the corresponding key. For menu-type mailboxes, the destination is usually an extension number or another mailbox (including menu-type mailboxes). For extension and the other mailbox types, the destination is typically an external telephone number. Digits 2, 3 and 7 for Personal Contacts are reserved for cell-phone, fax, and pager numbers. Key 0 is usually used to transfer to the operator but it can be use for Personal Contact numbers. Use of the Language Change number for Personal Contacts is also allowed but discouraged in systems with bilingual prompting. Key 9 is typically used for the directory (phonebook), and keys 99 through 999999 are reserved for System Administrator mailbox. See About Mailbox Types for more information.

Each routing number can be a maximum of 13 digits. * and # are allowed. The form has room to display only five numbers at a time; use the SHOW MORE softkey to display the rest.

Softkeys

SHOW MORE: Used to toggle between the first five (0-4) and second five (5-9) routing numbers.

TOP, BOTTOM: Positions cursor at top or bottom of the form.

FIND MAILBOX: Used to search the form for a specific mailbox. Press the softkey, and then enter the mailbox number.

DELETE: Deletes the entry at the current cursor position.

QUIT: Returns to the main form.

Form 51 - Voice Mail Distribution Lists

This form is used to set up distribution lists for system-wide use. A distribution list is a group of mailboxes that enables users to send messages to several people at once. The system provides global lists for everyone's use and personal lists for each mailbox owner's personal use. For more information about the lists and how to set them up, see Distribution Lists.

Note: There are forty-nine programmable distribution lists for system-wide use. A fiftieth list (the Broadcast list) numbered 000 contains all programmed extension-type mailboxes. The system creates the list automatically; it cannot be modified. All mailbox owners can use the global lists but only the administrator can change the programmable ones.

Figure: Form 51 Layout

Field Descriptions

MBOX#: An existing mailbox programmed in the Mailboxes form. An entry cannot be edited once it is added to this table.

NAME, EXTENSION: Read-only fields of names and extensions from the MAILBOXES form.

Softkeys

FIND MAILBOX: Searches the form for a specific mailbox. Press the softkey, and then enter the mailbox number.

INSERT: Inserts a mailbox number from the MAILBOXES form at the current cursor position. The order of mailboxes in this form is sorted after they are entered.

DELETE: Removes the selected mailbox from a distribution list. Position the cursor in the MBOX# field to the select the mailbox.

TOP, BOTTOM: Positions cursor at top or bottom of the form.

DISTRIB NAME: Assigns a name to a list for identification purposes.

DISTRIB LIST: Selects a list for viewing or editing. After pressing the softkey, type the list number (1-49), and then press ENTER.

QUIT: Returns to the main form.

Form 52 - Email

This form is used to set up e-mail notification of appropriate personnel in response to alarm level changes and 911 emergency calls. This form also contains the e-mail address to which logs are sent.

Notes:

1. For forwarding to occur, System Option 126, Email Messaging must be enabled in Form 04 and the SMTP server IP address must be programmed in Form 49. A system hostname must also be defined in Form 47 (IP Networking), Subform 01 (System IP). The system hostname must be a valid domain host name of 49 or fewer characters that has been registered in your DNS, or has been listed in your SMTP e-mail server's Hosts file.
2. Failed delivery of SX-200 ICP generated emails registers a message in the pstlogA.txt or /db/pstlogB.txt logs files. Example error responses in the pstlog files:

   CSMTPClientMsg::ProcessSMTPErrorResponse - 553 Requested action not taken: mailbox name not allowed.
   or
   CSMTPClientMsg::ProcessSMTPErrorResponse - 501 Syntax error, parameters in command "MAIL FROM: <@P64>" unrecognized or missing

   The pstlog files also list standard SMTP error codes. Failed delivery over a store-and-forward e-mail system does not register in the logs because the SX-200 ICP is a client, not a server.

Figure: Form 52 Layout

E911 Notification

E911 notification emails are sent whenever 911 is dialed from a station in the system. Notification can be sent to as many as three addresses.

Example:

From: E911@SX200.xyz.com
Sent: Friday, May 09, 2003 10:03 AM
Subject: E911 Caller: R.Smith, 3702 4th Fl Stat A3

Location: 350 Legget Dr, Kanata, ON
6 story building, across from big parking lot

The From line of the e-mail contains "E911" followed by the hostname of the SX-200 ICP as entered in Form 47 (IP Networking), Subform 01 (System IP).

The Subject line contains "E911 Caller" followed caller's name and extension, plus the contents of the Comments field programmed in Form 9. The comments must begin with "@" to appear in the Subject line, and are intended to describe the location of a specific device (for example, "4th Fl Stat A3"). The language of the message is the DEFAULT LANGUAGE programmed in Form 49, Voice Mail Options.

The body of the e-mail comes from the Comments field programmed in this form. These comments appear in all E911 notification e-mail messages, and are intended to describe a building location (for example, "350 Legget Dr, Kanata, ON".)

The call is also recorded in the Maintenance logs and appears as follows (for example):

2003-OCT-12 10:40:42 IP LINE CARD 01 01 05 01 Ext 106 Called 911 Alarm Code = 17

Alarm Notification

Alarm notifications are sent to as many as three addresses whenever a system alarm level increases in severity -- for example, from Minor to Major. The subject line of the e-mail contains "Alarm from" followed by the SX-200 ICP hostname as defined in Form 47 (IP Networking), Subform 01 (System IP). The language of the subject line is the DEFAULT LANGUAGE programmed in Form 49. The body of the e-mail contains information saved in the Maintenance logs at the time of the alarm.

Note: If there is already a Major alarm on the system when a new alarm occurs, then no notification is sent since the alarm level did not change.

Logs

By entering a maintenance command, the administrator can send the system logs an e-mail address. For details, see Sending Logs to an E-mail Address or FTP Server.

Field Descriptions

OWNER: A read-only field containing three entries each for E911 and alarm notification.

EMAIL ADDRESS: Specifies the e-mail address (up to 65 characters in length) to which E911 and alarm notifications, and system logs are sent.

Softkeys

COMMENTS: Applies to the three E911 entries only. Used to enter information to help in the identification of the caller's location. The entire comment (three lines of 65 characters) is added to the body of all E911 notification e-mails that are sent to the specified e-mail addresses. Normally, the comment contains the address of a building. Information concerning the location of specific telephones within the building can be programmed on Form 09.

QUIT: Returns to the CDE main menu.

DELETE FIELD: Deletes the content of the Comment field at the cursor position.

Form 53 - Bay Location Assignment

This form is used to assign bay numbers to connections on the SX-200 ICP. Each connection requires a bay number before the device that it connects to the controller can be programmed in Form 01.

Connections can be physical or logical. Physical connections provide communication links to other devices such as peripheral cabinets, T1 trunks, PRI trunks, PRI cards, and NSUs. The interfaces for physical connections are supplied by the two onboard CIM ports, and by optional Quad CIM, Dual FIM, Dual T1/E1 Framer, and T1/E1 Combo modules installed in MMC slots 1 and 2. Logical connections provide communication links to the controller's virtual bay (the IP bay), and to phantom bays that enable bay numbers to be reassigned.

As a minimum, a bay number must be assigned to the onboard IP bay. The two onboard CIMs, standard with every MX controller, must be assigned bay numbers if they are connected to peripheral devices. Bay numbers must also be assigned to the following optional MMC modules, if they are installed:

- Quad CIM connected to peripheral cabinets or NSUs on the MX, or to ASU IIs on the CX/CXi
- Dual FIM for connected to peripheral cabinets or NSUs
- Dual T1/E1 Framer or T1/E1 Combo connected to T1 or PRI trunks.

A mix of MMC modules can be installed in the first and second MMC slots. For example, one system may contain two Quad CIM while another contains a Quad CIM and a Dual T1/E1 Framer.

Notes:

1. Every physical connection requires a bay number. Otherwise, the connection will not appear in Form 01 as a bay and further programming will be impossible.

2. Every physical connection requires a unique bay number, except the connection for the Dual T1/E1 Framer. If the system contains two Dual T1/E1 Framers, they must both be assigned the same bay number in order to appear in Form 01 as a single bay.

3. In addition to the IP bay, assign bay numbers to a maximum of ten physical connections. The remaining bay numbers are available for phantom bays.

4. By default, the IP bay is bay number 1. It becomes bay 8 when an SX-200 EL/ML database is installed to migrate the system to an SX-200 ICP.

5. Use bay numbers 1 to 7 for connections to PRI cards; do not use bay numbers 8 to 15.

6. If a bay is assigned a bay number, it must be connected to a device (ASU, NSU, T1 or PRI trunk, etc.); otherwise, the system will generate an alarm. To prevent this problem, move disconnected bays from physical connections (CIMs and MMCs) to phantom bays.

7. Prior to Release 3.2, programming the onboard CIMs depends on whether they are connected to ASUs or peripheral cabinets. If the CIMs are connected to ASUs, ensure that the CIMs are not programmed with bay numbers on Form 53, then assign ASUs to slots 14 and 15 of the IP bay on Form 1.

For Release 3.2 and higher, if CIMs are connected to ASUs, they must be programmed with bay numbers in Form 53 and then assigned to any slot of the IP bay on Form 1.

8. If the CIMs are connected to peripheral cabinets, ensure that ASUs are not assigned to slots 14 and 15 of the IP bay on Form 1, then program the CIMs with bay numbers on Form 53.

9. You can change the bay number assigned to a bay. Typically, this involves creating a phantom bay for the new bay number, programming phantom bay with trunks and device assignments, unprogramming the old bay, and then moving the phantom bay to a physical connection. To change the bay number for a bay:

- In Form 53, create a phantom bay for the new bay number.
- In Form 01, assign a node type to the phantom bay created in Form 53.
- In Form 09, move device assignments from the old bay to the phantom bay.
- In Form 18, delete assignments for other devices from the old bay. The old bay should now be empty. Program the other devices on the new bay.
- In Form 53, move the bay assignment from the phantom bay to an available physical connection (CIM 1 or 2, or MMC 1 or 2).

Depending on a bay's original programming and how it is being updated, the node type(s) available in Form 01 will vary. The bay type remains unchanged when a T1, PER, or ISDN node is moved between physical connections and phantom bays, or when a SPINE or IP NODE is moved from a physical connection to a phantom bay. However, when a SPINE or IP NODE is moved from a phantom bay to a physical connection (after being unprogrammed as outlined above), the bay type switches to PER.

In some circumstances, fewer steps are needed to change a bay number assignment:

- If a bay is empty (unprogrammed in Form 01, 09, 18, etc.), you can freely assign a bay number to it in Form 53.
- You can change the connection for a bay by moving it to another column on the same row in Form 53, provided that the bay number and bay type remain the same. When you make such a change, the bay is automatically reset and there is no need to update Form 01, 09, and 18. Only CIM and FIM MMC modules can be moved this way.

10 -AX will come up with IP Bay as Bay 1 and the Embedded Bay as Bay 2 on CIM1 (default Node Type for the Embedded Bay will be ASU Bay'. The Bay programming detail is transparent to the user and will be auto configured in call control.

User will not be able to delete/move the Embedded ASU bay.

Figure: Form 53 Layout

Field Descriptions

BAY: Specifies the bay number assigned to the IP bay, CIM, or MMC.

IP BAY: An asterisk (*) in this column defines the bay number for the onboard IP bay. Only one bay number can be assigned to the onboard IP bay. This is a required entry.

CIM 1, CIM 2: Asterisks (*) in these columns define the bay numbers for the two onboard CIMs. Only one bay number can be assigned to each CIM. Assign bay numbers to the onboard CIMs only when they are connected to peripheral cabinets, not ASUs.

Note: Use Form 01 to program the connection between onboard CIMs and offboard ASUs. On the IP bay, assign the CIM 1 connection to slot 14 and the CIM 2 connection to slot 15. Once programming is complete, Form 53 will display "ASU" in the CIM 1 and CIM 2 columns, on the same row as the IP bay.

MMC 1, MMC 2: These columns define the bay numbers for the two MMC slots. The values in these columns vary depending on the type of module programmed for the slot, either Dual T1/E1 Framer, T1/E1 Combo, Quad CIM, or Dual FIM.

- T1 indicates that the bay number is assigned to a T1 module (either a Dual T1/E1 Framer or a T1/E1 Combo). If two T1 modules are installed (one in each MMC slot), they must both be assigned the same bay number.
- A number (1, 2, 3, or 4) indicates that the bay number is assigned to a Quad CIM or Dual FIM module. The number represents the connector in the module.

After an MMC slot has been programmed, the column header indicates the type of module it contains, either T1, CIM, or FIM.

PHANTOM: Asterisks (*) in this column defines the bay number for phantom bays. Phantom bays appear in Form 01 and can be programmed as normal bays. There can be multiple phantom bays.

Softkeys

MODULE TYPE: This softkey appears when the cursor is moved to an MMC column. Pressing MODULE TYPE causes other softkeys to appear; use them to program the type of module installed in the MMC slot.

Note: When you select a module type, you are limited to selecting the type of card actually installed in the MMC slot.

T1 MOD: Use this softkey to set the module type to T1 (Dual T1/E1 Framer or T1/E1 Combo) for the MMC slot. This softkey appears after MODULE TYPE is pressed.

DUAL FIM: Use this softkey to set the module type to Dual FIM for the MMC slot. This softkey appears after MODULE TYPE is pressed.

QUAD CIM: Use this softkey to set the module type to Quad CIM for the MMC slot. This softkey appears after MODULE TYPE is pressed.

NONE: Use this softkey to clear the module type for the MMC slot. This softkey appears after MODULE TYPE is pressed.

Note: You can change the module type from Quad CIM to Dual FIM only if the connections 3 and 4 have not already been assigned bay numbers. To make other changes, the card must not have any bay numbers assigned to it.

CONNECTOR 1, 2, 3, 4: Use these softkeys to define bay numbers for the connections in Quad CIM and Dual FIM modules. These softkeys appear when the cursor is moved to an MMC column, the module type is either Quad CIM or Dual FIM, and the connection has not already been assigned a bay number. If the module type is Dual FIM, only CONN 1 and CONN 2 are available.

Note: If more than one Quad CIM/Dual FIM is installed, program unique connections for each module.

PROGRAM: This softkey appears when the cursor is moved to an IP BAY, CIM 1, CIM 2, or PHANTOM column, or to an MMC column programmed with a T1 module. This softkey does not appear if the connection has already been assigned a bay number. Move to the appropriate column and press this softkey to assign a bay number to an onboard IP bay, CIM, or T1 connection, or to create a phantom bay and assign a bay number to it.

ENTER: Press ENTER to store changes in the database.

DELETE: Deletes the entry at the current cursor position. The IP bay can only be deleted if there are no cards programmed on it in Form 01.

QUIT: Returns to the main form.

Form 54 - Calling Party Number

The Calling Party Number (CPN) substitution feature provides calling party identification for outgoing calls for purposes of network identification and call back. This feature is available for T1/E1 Dual Framer and T1/E1 Combo modules installed in the SX-200 ICP (embedded PRI). CPN substitution is also available for offboard PRI (PRI cards and NSU PRIs), but it must be programmed using IMAT rather than CDE. For further information, see Using the ISDN Maintenance and Administration Tool (IMAT).

Notes:

1. CPN substitution is not available to users who select an outside trunk with a line key using Direct Trunk Select (DTS).
2. If embedded PRI is added to a system that already has offboard NSU PRI with CPN substitution, then add two entries to Form 22 (ARS: Modified Digit Table). For the entry that applies to the NSU PRI, CPN is disabled. For the entry that applies to the embedded PRI, CPN is enabled. Alternatively, deprogram IMAT and use a single entry for both routes (with CPN enabled).

Figure: Form 54 Layout

Field Descriptions

EXT NUMBER: The extension that is capable of making an outgoing call. Datasets, consoles, and telephones (IP, DNIC, and ONS) can be valid extensions.

DEFINED: The device type associated with the extension number: SUB ATT, DMP, SUPERSET, SUPERCONSOLE, IP Phone, Symbol Wireless Phone, SpectraLink NetLink Wireless Phone, or ONS door phone.

CALLING PARTY NUMBER: This field is reserved for a Calling Party Number (CPN) that replaces the extension number on outgoing calls. The CPN must be 10 to 24 digits in length and cannot include * or # characters.

Softkeys

RANGE PRGRM: This softkey allows a consecutive range of extensions to be programmed with a single CPN.

To program a single CPN for a range of extensions:

1. Press RANGE PRGRM.
   The system displays FROM EXT:
2. Enter the first extension number in the range and press ENTER.
   The system displays: TO EXT:
3. Enter the last extension number in the range and press ENTER.
   The system displays PLEASE ENTER CPN:

4. Enter the CPN. You can program the same CPN for all extensions in the range, or you can program leading digits that are added to each extension number to create a unique CPN for each extension in the range.
   - To program a single CPN for all extensions in the range, enter a number between 10 and 26 digits that does not include * or # characters.
   - To program a unique CPN for each extension in the range, enter the leading digits of the CPN followed from three to five * characters in place of the extension number. For example, programming extensions 100 to 102 with 6135925*** will yield CPNs 6135925100, 6135925101, and 6135925102.

Notes:

1 If more * characters are entered than the length of the extension, the extra space(s) are padded with zeroes; for example, programming extensions 100 to 102 with 613592**** will yield CPNs 6135920100, 6135920101, and 6135920102. If fewer * characters are entered than the length of the extension, then only the tail of the extension is used; for example, programming extensions 2100 to 2102 with 6135926*** will yield CPNs 6135926100, 6135926101, and 6135926102.

2. If you attempt to overwrite a range that is already programmed, the system will give this warning: "Existing CPN will be overwritten. CONFIRM or CANCEL."

RANGE DELETE: This softkey allows block deletion of CPNs for a consecutive range of extensions.

To delete all of the CPNs for a range of extensions:

1. Press RANGE DELETE.

   The system displays FROM EXT:
2. Enter the first extension number in the range and press ENTER.

   The system displays: TO EXT:
3. Enter the last extension number in the range and press ENTER.

FIND EXT: Pressing the FIND EXT softkey displays the ENTER EXTENSION NUM prompt on the command line. When a valid extension number is entered, the selected device information is displayed on the command line.

FIND CPN: Pressing the FIND CPN softkey displays the PLEASE ENTER CPN prompt on the command line. When a valid CPN is entered, the selected device information is displayed on the command line.

The standard softkeys **CANCEL**, **DELETE**, **ENTER**, **TOP**, **BOTTOM**, and **QUIT** are also provided.

Form 55 - Digit Translation Table

This form designates where calls received by Direct Inward Dial trunks will route based on Day Service, Night1 Service, and Night2 Service. Any incoming digit string can be translated into a new number and then routed to an answer point. There is no need for the incoming and translated numbers to correspond.

To enable digit translation, program Feature Access Code 67 (Digit Translation Table Access) on Form 02 Feature Access Codes. Then select a Dial-In trunk on Form 15 - Dial-In Trunks and program it to match the expected number of incoming digits (field N), absorb a maximum of 8 leading digits (field M), and use Feature Access Code 67 (field X). Finally, program the Digit Translation Table.

Note: If field X on Form 15 contains a number rather than Feature Access Code 67, the Digit Translation Table is not consulted. Instead, the number (up to two digits) is inserted ahead of the digit string before the call is routed to its destination.

Figure: Form 55 Layout

Figure: Form 55 DID Prefix Subform

Field Descriptions

INCOMING DIGITS: The incoming digit string after modification by Form 15 - Dial-In Trunks. The string can be up to 5 digits long, and contain numbers (0-9) and the special characters * and #.

DAY: This field designates the answer point where calls are routed to in day service mode. The answer point can be any valid extension or Abbreviated Dial number, but not a Feature Access Code. Permitted values include numbers (0-9) and the special characters * and #. If this field is left blank, the call is rerouted to the Vacant Number Routing point programmed on Form 19 - Call Rerouting Table. If the intercept is not programmed, the call is rejected.

INDEX: The index number to which a DID Prefix is programmed. Assigning this index number to lines in the Digit Translation Table causes the corresponding DID prefix to be used.

DID PREFIX: In Form 55 - Digit Translation Table, this field specifies the DID Index number that corresponds to the DID Prefix programmed in the Form 55 subform (DID Prefix). In Form 55, a DID Prefix of "1" causes the prefix digits listed beside Index "1" in the subform to be appended. Prefixes are used only for display/print purposes.

N1: This field specifies the answer point that calls are routed to during Night1 Service Mode. The answer point can be any valid extension or Abbreviated Dial number. Permitted values include numbers (0-9) and the special characters * and #. If this field is blank, the call is rerouted to the Vacant Number Routing point programmed on Form 19 - Call Rerouting Table. If the intercept is not programmed, the call is rejected.

N2: This field specifies the answer point where calls are routed to during Night2 Service Mode. The answer point can be any valid extension or Abbreviated Dial number. Permitted values include numbers (0-9) and the special characters * and #. If this field is blank, the call is rerouted to the Vacant Number Routing point programmed on Form 19 - Call Rerouting Table. If the intercept is not programmed, the call is rejected.

TENANT: This field allows you to switch an individual or group of DNIS digits to night service while leaving others in day service. This field takes priority over any entry in Form 15 - Dial-In Trunks.

Softkeys

RANGE PRGRM: This softkey allows for range programming of DID numbers. After entering the DID Prefix index, you are prompted to enter starting incoming digits and then ending incoming digits. If a DID number in the specified range already exists in the table, the system skips that number and continues with the next.

DID PREFIX: This softkey opens the DID Prefix subform. DID Prefix digits are used for displaying and printing assigned DID numbers. Maximum DID number length (including prefix) is 10 digits. When creating a new Digit Translation table, the default prefix is 1.

DELETE FIELD: Deletes the content of the field at the cursor position.

FIND STRING: This softkey allows the user to find an entry by specifying its Incoming Digits string. Pressing the FIND STRING softkey displays the ENTER ACCESS CODE: prompt on the command line. The selection is completed by entering a valid, exact Incoming Digits string.

SPEED DIAL: Allows a speed dial number to be used as the answer point. The Day, N1, and N2 fields are preceded by the letters sc when SPEED DIAL has been selected.

The standard softkeys **CANCEL**, **DELETE**, **ENTER**, **TOP**, **BOTTOM**, and **QUIT** are also provided.

Program Error Messages

Table: Program Error Messages

Error Message	Meaning
XXXXX is an ACD agent ID	A supervisor with ACD ID XXXXX, cannot be displayed, as requested via the FIND SUPER key, because this ID belongs to an agent, not a supervisor.
XXXXX is an ACD supervisor	The ID entered for the FIND ID key belongs to a senior supervisor or supervisor.
XXXXX already has day zone X specified	The selected day already has a day zone specification. Only one day zone can be specified per day. Refer to Form 25, ARS: Route Plans.
XXXXX has no zone specified	All days of the week must have a day zone specification.
*3 must be followed by 01 to 14	In Form 31, System Abbreviated Dial Entry, the "*3" indicates that the subsequent digits will be manually dialed. The number of digits that will be manually dialed follows the "*3" indicator, and can be from 01 to 14.
*5 cannot be followed by further digits	In Form 31, System Abbreviated Dial Entry, the "*5" indicates that the previous number is an intercom number. Therefore, no digits can follow the "*5" terminator.
* must be followed by 1, 3, 4, 5, 6, 8, 9, or *	In Form 31, System Abbreviated Dial Entry, only the numbers "1", "3", "4", "5", "6", "8", "9" or "*" are valid entries after an "*" entry.
A recording device cannot be programmed as a keyline or multi-call line.	In CDE Form 09, you cannot program an ONS line as a key line or a multi-call line if this line is programmed in CDE Form 17 as a music source.
Access code XXXXX is already assigned	The access code is used elsewhere in the database. Select another access code. Refer to Form 35, Global Find Access Code, for a list of assigned codes.
Access Code XXXXX conflicts with the Access Code for HUNT GROUP NUM XX	The code entered has already been defined as a hunt group access code in Form 17, Hunt Groups. Select a new code or change the hunt group access code.
Access Code XXXXX conflicts with an ARS Leading Digits Entry	The system does not allow a partial match between an ARS leading digit and an access code. Leading digit strings must be unique. Refer to Form 26, ARS: Digit Strings and select a new code or change the ARS leading digit entry.
Access Code XXXXX conflicts with an Attendant LDN Access Code	The code entered has already been defined as an LDN access code in Form 08, Attendant LDN Assignments. Select a new code or change the LDN assignment.
Access Code XXXXX conflicts with a Console Access Code	The code entered has already been defined as an extension number for a console in Form 07, Console Assignments. Select a new code or change the console extension number.

Table: Program Error Messages

Error Message	Meaning
Access Code XXXXX conflicts with the Feature Access Code for FEATURE NUMBER XX	The code entered has already been defined as a feature access code. Select a new code or change the code in Form 02, Feature Access Codes.
Access Code XXXXX conflicts with a Night Bell Access Code	The code entered has already been defined as an extension number for a night bell in Form 18, Miscellaneous System Ports. Select a new code or change the night bell extension number.
Access Code XXXXX conflicts with a station number or Mitel telephone Prime Line number	The code entered has already been defined as a station number or the prime line number of a Mitel telephone. Select a new code or change the station number (or prime line number) in Form 09 (Desktop Device Assignments).
Access Code XXXXX conflicts with Mitel telephone line number	The code entered has already been defined as a Mitel telephone line extension number in Form 09 (Desktop Device Assignments). Select a new code or change the Mitel telephone line extension number.
Access code XXXXX does not correspond to a Stn, Set or logical line	Only those access codes (extension numbers) which correspond to a station, Mitel telephone key or logical line can be used.
Access code XXXXX does not exist	The selected access code has not been assigned.
Access Code XXXXX does not match with ARS Leading Digits Entry	The access code entered does not match the Direct to ARS access code assigned in Form 02, Feature Access Codes. The access code must be a defined ARS leading digit string.
Access code entered is not for a station, Mitel telephone or logical line	An invalid access code has been entered and the ENTER key is pressed.
The access code for field XXX is invalid	The extension number, hunt group access code, night bell extension number or attendant console directory number assigned to one of the DAY, N1 or N2 fields in Form 19, Call Rerouting Table is invalid. Assign a new code in that field.
Access codes for fields XXX & XXX are invalid	The extension numbers, hunt group access codes, night bell extension numbers, or attendant console directory numbers assigned to two of the DAY, N1 or N2 fields in Form 19, Call Rerouting Table are invalid. Assign new codes in the fields.
Access codes for fields XXX, XXX & XXX are invalid	The extension numbers, hunt group access codes, night bell extension numbers or attendant console directory numbers assigned to the DAY, N1 and N2 fields in Form 19, Call Rerouting Table are invalid. Assign new codes in the fields.
Access code XXXXX is not a valid answer point	

The following are valid answer points for the DAY, N1 and N2 fields in Form 14, Non-Dial-In Trunks:
- an LDN on the attendant console,
- a rotary dial or DTMF set extension number,
- a Mitel telephone directory number,

- a hunt group access code or,
- a night bell extension number.

The entry for "Record a Call Voice mail Destination for this Tenant" in Form 19 must be a station/set hunt group number. The first member of the hunt group must be a DNIC port that has COS 229 enabled.	
Account codes of unspecified lengths exist; delete these first	Ensure that all of the account codes in Form 33, Account Code Entry are equal to the specified account code length in Form 04, System Options/System Timers.
Account code value exists in the database	In Form 33, Account Code Entry, the entered account code already exists. Select a new account code.
ACD agent group XX has no members	The agent group added to the form is empty. It must have members before it can be put in this form.
ACD agent group XX already assigned to a supervisor	The ACD agent group which has been edited or inserted into the sub-form, is already programmed under some other supervisor.
ACD Agent Group XX not assigned to a supervisor	ACD group XX, requested by the FIND GROUP key, cannot be displayed, because it has not been assigned to a supervisor yet.
ACD groups under XXXXX must first be deleted	This senior supervisor cannot be deleted from the first-level form by the DELETE key, because there are ACD groups defined under this senior supervisor.
Agent XXXXX does not exist	The ID entered for the FIND ID key does not exist in the database.
Agent XXXXX is on line and cannot be deleted	The DELETE key cannot be used on an agent who is on line.
Agent group XX does not exist	The selected overflow agent group does not exist.
Agent Group XX has no members	The selected primary ACD agent group or overflow agent group is empty. It must have members before it can be put in this form.
AGENT STATUS not allowed when multiple QUEUE STATUS keys are programmed	No agent status keys are allowed on an ACD supervisor key template when multiple queue status keys are programmed.
All members must be idle before changing Hunt Group type	The user attempted to change the hunt group type. The user needs to wait for all of the devices to be free or take them out of service before changing the group type.
Alternate music sources cannot have keyline or multi-call line appearances	Music sources cannot be line appearances.
An agent's set's prime line cannot have any appearances on other sets.	An attempt was made to program a DSS/BLF key on a set logged in as an ACD Agent.

A recording device cannot be programmed as a keyline or multi-call line.	An ONS line programmed with a recording device in CDE Form 17 cannot be programmed as a keyline or a multi-call line in CDE Form 09.
ARS Leading Digits XXXX have not been assigned	The access code is valid, but it has not been assigned to any sets.
Associated device already has three SUPERSET DSS or PKM modules attached	An attempt was made to associate more than three SUPERSET DSS modules or PKM modules with a device.
Associated device has a key template established in its COS	An attempt was made to associate a PKM module with a set that has a key template enabled in its COS.
Association of SUPERSET DSS or PKM Module to a Mitel must be removed	An attempt was made to delete a telephone that has one or more SUPERSET DSS or PKM modules associated with it.
Attempting to add invalid access code	A PKM must be associated with a Mitel telephone; not with a standard telephone.
Attempting to add more than XXX members to this group	

There are a maximum of 50 members for each pickup group, trunk group and hunt groups.

Note: 100 max. for hunt groups.	
Attempting to define multiple appearances of LDN XXX on this set	An LDN may appear only once on a given set.
Attempting to define multiple KEY LINE appearances of XXXXX on this SET	An access code for a key line appearance cannot be duplicated on another Key Line appearance on that Mitel telephone. Refer to Form 09, (Expand Set Subform) Mitel Telephone Lines.
Attempting to define multiple key types for Access Code XXX	In Form 09, (Desktop Device Assignments) , only one key type can be assigned to each station number.
Attempting to define multiple key types for Trunk Number XX	In Form 09, only one key type can be assigned to each trunk number.
Attempting to delete an ACD path that is currently in use	This ACD path cannot be deleted because there are ACD calls currently being handled via this path.
Attempting to enter invalid access code	

An invalid access code was entered.

Note: a blank access code is not allowed.	
Attempting to program more than 1 HCI port(s)	You can only program one MAI (HCI) port.
Attempting to program more than 64 appearances for Mitel telephone line	

A single Mitel telephone line may have a maximum of 63 appearances on other sets.

The error message occurs when you program more than 64 appearances of the following: Direct Trunk Access, CO line appearances, key line appearances, or multi-call appearances.	
Attempting to program more than XX appearances for sub attendant LDN	There is a limited number of appearances for any line appearance on sets in the system and the user attempted to exceed this limit.
Attempting to remove an Agent Group that has calls waiting	This message will be displayed if the primary agent group or the overflow agent groups have calls waiting and the user is attempting to change or delete the agent group or delete the path.
Attempting to remove a Music Source that is currently in use	Someone is listening to the music source so it cannot be removed. This message can occur when attempting to change or delete a music source, deleting the recording that the music source is in, or deleting the path.
Attempting to remove a Recording that is currently in use	Someone is using the recording hunt group so it cannot be removed. This message can occur when attempting to change or delete the recording or when deleting the path.
Attempting update/delete of a device that is currently in use	The selected device is being used; modify this device at a later time.
Automated Attendant hunt group may not contain Mitel telephones	An auto attendant hunt group can only contain industry standard telephones.
Bay/Slot/Circuit XX/XX/XX is already assigned	The specified bay/slot/circuit is assigned elsewhere. Select a new bay/slot/circuit number or change the assignment of that bay/slot/circuit number. Refer to Form 01, System Configuration and Form 09, Desktop Device Assignments.
Bay/Slot/Circuit XX/XX/XX is already programmed elsewhere	The specified bay/slot/circuit is assigned elsewhere. Select a new bay/slot/circuit number or change the assignment of that bay/slot/circuit number. Refer to Form 07, Console Assignment, Form 09, Desktop Device Assignments, and Form 12, Data Assignment.
Bay/Slot/Circuit - XX/XX/XX cannot be programmed	Devices are assigned to the selected bay/slot/circuit. These devices must be deleted before a new card type can be programmed for the bay/slot/circuit.
Bay/Slot/Circuit - XX/XX/XX used is not appropriate.	In CDE Form 09, move the PLID to a location that is not programmed as a station/set.
Bay/Slot/Circuit - XX/XX/XX is not present	The specified circuit number is not applicable for these bay and slot numbers. Re-enter the bay and slot numbers without the circuit number or enter 0 for the circuit number.

Bay/Slot/Circuit - XX/XX/XX is not programmed as a console	The selected bay/slot/circuit is not a console. Refer to Form 01, System Configuration and reprogram this bay/slot/circuit as a console.
Bay/Slot/Circuit - XX/XX/XX is not programmed as a DIAL-IN trunk	The selected bay/slot/circuit is programmed as a Non-Dial-In Trunk. Refer to Form 14, Dial-In Trunks.
Bay/Slot/Circuit - XX/XX/XX is not programmed as a DMP DEVICE	The selected device is not a Music-on-Hold/Paging Unit.
Bay/Slot/Circuit - XX/XX/XX is not programmed as a DNIC	The selected bay/slot/circuit is not a DNIC circuit.
Bay/Slot/Circuit - XX/XX/XX is not programmed as a MUSIC/PAGER MODULE	The selected bay/slot/circuit is not a music/pager assignment. Refer to Form 01, System Configuration and reprogram this bay/slot/circuit as a Music/Pager Module.
Bay/Slot/Circuit - XX/XX/XX is not programmed as a NON-DIAL-IN trunk	The selected bay/slot/circuit is programmed as a dial-in Trunk. Refer to Form 15, Dial-In Trunks.
Bay/Slot/Circuit - XX/XX/XX is not programmed as a RECEIVER MODULE	Relays (Subcircuits 5 and 6) are located on the Receiver/Relay Module only. Ensure that there is a Receiver/Relay Module on a Universal Card at that location in Form 01, System Configuration.
Bay/Slot/Circuit - XX/XX/XX has incompatible device type programmed	The user attempted to range program over different card types, a circuit that has a BLF programmed or a circuit that has a SUB (enhanced sub attendant) programmed.
Bay/Slot/Circuit - XX/XX/XX is not programmed as a trunk	The selected trunk number corresponds to a bay/slot/circuit that is not programmed as a trunk.
Bay/Slot/Circuit - XX/XX/XX is not programmed as a UNIVERSAL CARD	The selected bay/slot/circuit is not a Universal Card assignment. Refer to Form 01, System Configuration and reprogram this bay/slot/circuit as a Universal Card.
Cannot associate a primary phone	The ONS station (primary phone) cannot be associated with another primary telephone, that is, cannot become a secondary phone.
Cannot associate to a secondary phone	The ONS station cannot be associated to an ONS station that itself is associated to a third phone.
Cannot delete last agent while callers are waiting on ACD Group XX	The user attempted to delete the last agent from ACD group XX, which would delete group XX itself. However, there are ACD calls waiting for this group, so the deletion cannot be permitted at this time.
Cannot disable option when Automated Attendant programming is present	The user attempted to disable System Option 106 - Automated Attendant when there are auto attendant groups programmed in the system.
Cannot enable DTRX due to invalid mode or DTE not assigned	The user attempted to enable the DTRX option in the hunt group options form for a modem hunt group for which DTRX does not apply. The group has to be in ANSWER or BOTH mode.

Cannot modify LEADING DIGITS until digit strings deleted	In Form 26, ARS: Digit Strings, the digit entries in the DIGITS TO BE ANALYZED field must be deleted before the digit entries in the LEADING DIGITS field can be modified.
Cannot update music/pager assignment - must delete and re-enter	To modify the selected music/pager assignment, it must first be deleted and then re-entered.
Can only use “X” wildcard at end of strings	The X softkey can only be pressed at the end of a digit string entry. Refer to Form 26, ARS: Digit Strings.
Circuit descriptor type must match programmed trunk hardware type	The selected trunk circuit descriptor type in Form 13, Trunk Circuit Descriptors must match the specified Trunk Card type in Form 01, System Configuration.
Configuration cannot be changed for the Bay/Slot/Circuit - XX/XX/XX	In Form 01, System Configuration, the data in the PROGRAMMED field cannot be matched to the data in the INSTALLED field as devices are already assigned to that physical location. Cannot change the configuration until the devices are de-assigned.
Console XXXX has PKM(s) programmed in Form 7	A console already has a PKM programmed in Form 7.
Console XXXX is already associated with a device on bb/ss/cct.	A set in the system has the console extension number programmed in its "associated" field in Form 9. The PKM can not be programmed with the console.
Console has LDN assigned - cannot delete	An attendant console can only be deleted from the system if all of its LDN assignments are removed first. Refer to Form 08, Console LDN Assignments.
Consoles that share an LDN access code must be in the same TENANT GROUP	Attendant consoles that share the same LDN access code must also share the same tenant group. Refer to Form 07, Console Assignments.
COR group out of range. Valid range is (1 - 50)	There are a maximum of 50 COR groups.
COR value is out of range	In Form 32, CDE Data Print, the COR value specified must be valid (1 -> 25 for each COR group). There are a maximum of 50 COR groups. Refer to Form 20, ARS: Class of Restriction Groups.
COS number must be 1 to 50	Valid range for COS numbers is 1 to 50.
COS value is out of range	In Form 32, CDE Data Print, the COS value specified must be valid (1 -> 50).
The current associated set is using a key on the PKM	An attempt was made to change the association of a PKM while the associated set is using a line on the set.
Data entered is incomplete. Quit to exit form	In Form 44, T1 Network Sync, the installer must complete the data entries.
Data port of the DNIC console must be deleted first	The user is attempting to delete the DNIC console in Form 07 without first deleting its data port in Form 12.

Database is out of sync	The database copy in RAM does not match the database copy on the CMOS. Delete the selected device and re-enter the device in the appropriate form.
Delete ACD Group XX from ACD PATH YY before deleting last agent	The user attempted to delete the last agent from group XX, which would have the effect of deleting group XX itself. However, group XX is referenced in the ACD PATH form, for path number YY, so it must be deleted from that form first. Then the user will be allowed to delete the last agent, and thus delete the group.
Delete ACD Group XX from ACD SUPERVISORS form before deleting last agent	The user attempted to delete the last agent from group XX, which would delete group XX itself. However, group XX is referenced in the ACD Supervisors form, so it must be deleted from that form first.
Delete in Hunt Group Form first	The user attempted to delete the last member of a hunt group programmed as an overflow point, which would delete the hunt group itself. The member must be deleted from Form 17 first.
Delete trunk at Bay/Slot/Circuit - XX/XX/XX before adding 2nd T1 ISDN Card.	The system cannot configure the peripheral bay as an ISDN node because the 24th circuit of the first T1 Trunk Card is programmed in either Form 14 or Form 15. Delete the trunk that is programmed on the 24th circuit of the first T1 Trunk Card.
Deleting all IP Route Defs using this trunk’s group. CONFIRM or CANCEL	If the trunk group that the user is attempting to delete is used in any Route Definitions in Form 23, the definition will also be deleted. This ensures that if the trunk group is re-created with IP trunks, any Route definition with this trunk group will have default IP values to start with, and will not automatically acquire previously assigned values.
Device is a member of a ring group	A ring group member cannot be a member of another type of hunt group.
Device valid for ring groups only	The user is trying to add a console or night bell to a hunt group that is not a ring group type.
DISA trunks cannot have expected, prefix, or absorb digits programmed	The user was trying to program these fields for a DISA trunk.
DNIC console cannot originate a DTRX call	The data port for a DNIC console cannot have the DTE field in Form 12 filled out since it cannot originate a DTRX.
DTE field is not programmed for this device	First member of a modem hunt group has the DTE field programmed. User attempted to add a member without the DTE field programmed - must be consistent within the hunt group.
DTE field is programmed for this device	First member of a modem hunt group does not have the DTE field programmed. User attempted to add a member with the DTE field programmed - must be consistent within the hunt group.

DTMF receivers in bay XX in use. Please try again later.	An attempt has been made to implement CLASS receivers that are currently active DTMF receivers. Wait until the receivers are idle, then reprogram or busy out the receivers in maintenance, and then implement them as CLASS receivers.
Duplicate IP address	IP Signalling Bay IP Address (Form 04) conflicts with the IP Controller Address (Form 47) or the IP Node Address (Form 48) conflict. This is not allowed. 000.000.000.000 is the only address that can be duplicated.
Duplicate mac address at BB/SS/CC	A unique MAC Address, used to register individual IP phones, has been assigned to more than one Bay/Slot/Circuit location.
Entered string is not in the system	In Form 26, ARS Nested Digit Strings, the selected digit string is not defined.
Error? Only * (01, 02, 03, 04, 05, 1, 2, 3, 4, 5, 6, 7, 8, 9, *) are valid	The asterisk sequence in the DIGITS TO BE INSERTED field in Form 22, ARS: Modified Digit Table is restricted to the following: *01 = Identifies call forwarder extension number (for analog networking) *02 = Identifies call forwarding reason (for analog networking). *03 = Inserts CESID *04 = Inserts Calling Name *05 = Inserts Billed Extension *2 = Wait for Dial Tone, *3 = Switch to DTMF for subsequent digits, *4 = Stop or start displaying modified digits. Modified digits are displayed on the sets and in the SMDR records. The first time *4 appears in a digit string the system stops displaying the following modified digits. The next time *4 appears the system starts displaying the following modified digits. *5 = Pause 10 seconds, *6 = Insert caller's ID, *7 = Insert caller's dialed account code, *8 =Insert PBX node ID number, *9 =Pause for 1 second, ** = asterisk character.
Error in COR members. Use spaces between CORs. e.g. 1 3 5-14 25	The members for each COR group must be separated by a space. For consecutive CORs, a dash is inserted.
Error in QTY TO DELETE field Can only delete up to 25	Only 25 digits can be deleted. Refer to the QTY TO DELETE field in Form 22, ARS: Modified Digit Table.
Extn. XXXX can't have a key template and a SUPERSET DSS/PKM module at the same time	A telephone may not be programmed with both a key template and an associated PKM module at the same time.

Extension XXXXX is not a Key System set	The DSS PAGE option is selected and the user enters a non-key set directory number in the EXT NUM column, or the user changes the NON DSS or DSS RING option to DSS PAGE option and the directory number in the EXT NUM column contains a non-key set directory number.
Extension number XXXXX does not exist or is not valid	The selected extension number is undefined. The extension number must first be defined in Form 09, Desktop Device Assignments. The extension number cannot be a multi-line appearance.
Extension number XXXXX has not been assigned a paging group	That extension number is not in a paging group.
EXTENSION NUMBER XXXXX is a member of HUNT GROUP XX	The selected extension number is a member of the hunt group shown. An extension number can be a member of only one hunt group at a time.
EXTENSION NUMBER XXXXX is a member of PICKUP GROUP XX	The selected extension number is a member of the pickup group shown. An extension number can be a member of only one pickup group at a time.
FEATURE NUMBER XX conflicts with system option YY in Form 4	The given feature conflicts with the specified system option in Form 4. The option and the feature access code cannot be programmed at the same time.
Feature Number out of range Valid range is (1 - 45)	Feature numbers range from 1 to 45. Refer to Form 02, Feature Access Codes.
First account code position value greater than last	When Print Option Number 14 in Form 32, CDE Data Print is selected, the first specified account code must be less than the last specified account code.
First Character in Hunt Group name must be alphabetic.	Begin the hunt group name with an alphabet letter.
First digit in Access Code must be ’*’	The Key System - Retrieve Personal Speed Call feature access code must begin with an asterisk (*).
First digit in digit string cannot be a ’*’	This will conflict with the Key System - Retrieve Personal Speed Call. If asterisks (*) are required, program them into the Modified Digit Table.
Form access disallowed; enable verified account codes system option	To access Form 33, Account Code Entry, System Option 05, Verified Account Code must first be enabled. Refer to Form 04, System Options/System Timers.
Form number XX cannot be accessed	The selected form number is reserved for future use or the selected form number has restricted user access. Refer to Form 28, Form Access Restriction Definition for the level of access for each form.
Form number out of range	The selected form number is invalid. See the Customer Data Entry FormsTable.

General failure to complete operation. Voicemail Error code: 1	The user attempted to assign a second mailbox to an extension. Extensions can only have one mailbox.
High power card cannot be programmed at this Bay/Slot/Circuit	The Universal Card can only be assigned to those card slots rated for high power consumption. These are the upper slots of any digital bay.
Hour value XX is out of range	The hour value can range from 01 to 24. Refer to Form 04, System Options/System Timers.
Hunt group must have an access code assigned before name can be programmed	The user attempted to program a name for the hunt group before an access code was programmed. The access code has to be done first.
Incorrect specification of Bay/Slot/Circuit for range programming	The specified start bay/slot/circuit numbers must be less than the end bay/slot/circuit numbers.
Inserted too many digits Up to 38 allowed, with '*n' counted as 1	The specified entry exceeds the maximum. A maximum of 38 entries are allowed in the DIGITS TO BE INSERTED field in Form 22, ARS: Modified Digit Table.
Invalid answer point, SD # XXX not programmed!	You attempted to program invalid speed dial in Form 14, Non-Dial-In Trunks. Program a valid speed dial in form 31 first.
Invalid default destination	The user attempted to program an invalid default destination for an auto attendant group. See Automated Attendant Application Package section, for a list of valid destinations.
Invalid device for a ring group	Valid ring group devices are sets, stations, consoles, and night bells.
Invalid digit to insert Use 0-9 or "* (1,2,3,4,5,*)" only	The specified entry is invalid. The entries in the DIGITS TO BE INSERTED field are restricted to the following: - 0 to 9 and - the valid asterisk sequences.
Invalid Fax destination	The user attempted to program an invalid fax destination for an auto attendant group. See Practices.
Invalid key for set in use	The user attempted to program invalid keys (7 to 12) for a SUPERSET 410 on expanded set forms for Form 09.
Invalid type change. Recording Hunt Group appears in ACD path X	The user attempted to change a recording hunt group which appears in Form 41, ACD Path.
IP Address must be in range 000 to 255	The value entered is out of range.
IP Set Registration failed: DN not found on bay.	The DN (directory number) received is not programmed to a set on that bay. Ensure that the DN is programmed in Form 09.
IP Set Registration failed: Invalid PIN.	The PIN received is invalid. Refer to Form 02 for the correct PIN.

IP Set Registration failed: IP Sets limit reached.	The maximum number of IP sets (as determined by System Option 114 in Form 04) are already registered.
IP Set Registration failed: MAC already exists.	Another set with that MAC address is already registered in the system.
IP Set Replacement failed: Other set still connected.	The received DN and MAC address belong to a set that is still connected. For security reasons, the set must be disconnected before it can be replaced.
Key number XX cannot be programmed as the Origination Line Preference key	This key can be a PRIME key, CO line key, CO line group key, MANUAL, or PERSONAL O/G key.
The key value is outside the valid range for KEY NUMBERS (2-14)	Enter a valid key number 2-14.
Link descriptor value XX out of range. Valid range (1-12).	An invalid link number has been entered for CDE Form 43, T1 Link Assignments.
Maximum ACD positions allowed are already assigned	The user attempted to insert a 1000th ACD position into the system. Only 999 are allowed. ACD positions include agents, supervisors and senior supervisors.
Maximum Afterwork Timer is 15:00	The user entered a timer out of range.
Maximum high power cards per bay exceeded	Each bay is limited to not more than 4 high power cards.
Maximum number of non-IP devices in ring all group exceeded	Maximum number of non-IP devices per ring group in a ring-all group is 32.
Maximum number of ring groups already programmed	Maximum number of ring groups in the system is 25.
Maximum T1 trunk cards are programmed.	No more T1 trunk cards can be programmed.
Maximum Time is 54:00	The user entered a timer out of range.
Minute value XX is out of range	The minute value can range from 01 to XX, where XX is option specific. Refer to Form 04, System Options/System Timers.
MODIFIED DIGIT ENTRY out of range Valid range is (1 - 50)	The selected entry number is invalid. The range is 1 to 50. Refer to Form 22, ARS: Modified Digit Table.
Modified digit table entries are from 1 to 50	The selected entry number is invalid. The range is 01 to 50. Select a new entry number. Refer to Form 22, ARS: Modified Digit Table.
MOSS (appears in the alarm status area of the form header)	MOC and password do not match. Password needs to be entered in Form 04 (Option 100) followed by a system reset.
Multiple QUEUE STATUS keys not allowed with AGENT STATUS keys	Multiple queue status keys are not allowed on an ACD supervisor key template when agent status keys are programmed.
Must change the CDN field before changing the N field	The trunk circuit descriptor cannot have prefix digits programmed.

Must delete all appearances of XXX from CALL REROUTING TABLE	This ACD path cannot be deleted, because its access code appears in the CALL REROUTING table (Form 19).
Must delete all appearances of XXXXX from CALL REROUTING TABLE	The selected device must have its access code or extension number deleted from the Call Rerouting Table before it can be removed from the system. Refer to Form 19, Call Rerouting Table.
Must delete all appearances of XXXXX from Desktop Device Assignments Form.	This digit string is used in Form 09 (Desktop Device Assignments) as CO Line Group Keys - must delete this first. Use the REVIEW form in Form 09 to review the list of Mitel telephones using this leading digit.
Must delete all appearances of XXX in NON-DIAL-IN trunks	This ACD path cannot be deleted because its access code appears in the NON-DIAL-IN TRUNKS form (Form 14).
Must delete all appearances of XXX on all sets before deleting trunk	This trunk is programmed on a set as a CO line key in Form 09 - must delete there first.
Must delete all appearances of XXXXX in Hunt Group Options form	The user attempted to delete a device which is programmed in the Hunt Group Options form as a default destination for auto attendant or an overflow point.
Must delete all Digit Strings that begin with a '*'	Before programming this access code, go to the SHOW STRINGS subform of Form 26 - ARS: Digit Strings, and delete all digit strings beginning with an asterisk (*).
Must delete all key definitions before deleting set	The selected Mitel telephone must have all its key definitions deleted before it can be removed from the system. Refer to Form 09, (Expand Set Subform) Mitel Telephone Lines.
Must delete all ONS associations	If the primary telephone is deleted, the telephone associations with the primary telephones must be deleted as well.
Must delete BLF appearances starting on device XXXX before deleting.	The user attempted to delete a device which has BLF appearances on the specified device. The appearances must be deleted first.
Must delete appearances of XXXXX from answer points in NON-DIAL-IN TRUNKS	The selected device must have its access code or extension number deleted from the answer points in Form 14, Non-Dial-In Trunks before it can be removed from the system.
Must delete BLF appearances starting on device XXXX before deleting	The user attempted to delete a device which has BLF appearances on the specified device. The appearances must be deleted first.
Must delete the appearances of XXXXX on all sets before deleting set	All appearances of the station number (or prime line number) must be deleted before that set can be removed from the system.

Must enter first choice for route list to be defined (leave no gaps)	In Form 24, ARS: Route Lists, the FIRST field must be specified with a valid route list number before the SECOND field is specified.
Must enter LEADING DIGITS	To complete the entry in Form 26, ARS: Digit Strings, must enter digits in the LEADING DIGITS field.
Must first disable Option 217 in COS (1 - 50)	COS Option 217 Direct To ARS must be disabled before the Direct To ARS feature access code can be deleted.
Must program Direct To ARS access code first	The user attempted to enable COS Option 217, Direct To ARS when there is no feature access code programmed in Form 02 for this feature. Form 02 must be programmed first.
Must program DNIC set first in Form 9.	Program the set in CDE Form 09, Desktop Device Assignments.
Must retrieve all parked calls in orbits	The user tries to delete the Call Park Orbit Retrieve feature access code while there is at least one call still parked.
Name cannot contain BLANKS or DASHES	Do not enter a blank or dash for the name.
Neither Option #248 Nor #249 are enabled yet	One of these COS Options must be enabled for correct system operation.
No CONSOLE MODULES are programmed	The console module must be assigned to a bay/slot/circuit/subcircuit in Form 01, System Configuration.
No DIAL-IN trunk cards are programmed	A card that supports dial-in trunks must be defined in Form 01, System Configuration.
No DNIC cards are programmed	To access Form 09, there must be DNIC cards programmed into Form 1.
No more space available for comments	Subsequent comments that are entered in the CDE Forms are not saved.
Non-existent account code value has been entered	During a search in Form 32, CDE Data Print, the entered account code does not exist. Select another account code.
No NON-DIAL-IN trunk cards are programmed	A card that supports non-dial-in trunks must be defined in Form 01, System Configuration.
No STATION or SUPERSET cards are programmed	A card that supports rotary dial or DTMF sets or a card that supports SUPERSET telephones must be defined in Form 01, System Configuration.
No trunks are assigned to circuit descriptor number X	The trunk circuit descriptor number entered has not been specified in Form 13, Trunk Circuit Descriptors.
Not a Valid Answer Point	The first member of the hunt group must be a DNIC port that has COS 229 (Voice Mail Port) enabled.

NUMBER OF DIGITS TO ABSORB must be in the range (0 - 8)	In the M field of Form 15, Dial-In Trunks, the maximum quantity of digits allowed is 8.
ONS limit exceeded for this part of Bay	The maximum number of ONS cards allowed has been exceeded.
ONS Port access code XXXXX does not exist	The access code entered for the Music Source Following a Recording is non-existent or illegal.
Option XX conflicts with this option	The selected option is mutually exclusive with the option number shown.
Priority must be in range 0 to 7	The value entered is out of range.
Not a Valid Option	The entry is invalid. For COS Options (Form 3), valid options are 100 to 908. For System Options (Form 4), valid options are 1 to 113.
Only 1 PER bay is allowed 2 T1 Trunk cards, all others can only have 1	This message appears if you attempt to program a second T1 Trunk Card in more than one node.
Only 4 consoles are allowed on any one Digital Line Card	You are attempting to program more than 4 consoles on a Digital Line Card in Form 7 (CONSOLE ASSIGNMENTS).The maximum number of consoles on a Digital Line Card is 4.
Only 11 consoles are allowed to be programmed in the system	You are attempting to program more than 11 consoles in Form 7. The system supports a maximum of 11 consoles.
Overflow Agent Groups must be unique	The selected overflow agent group is a duplicate of an overflow already specified for this path, or it is the primary agent group for this ACD path.
Pager X is already assigned to a Night Bell.	The pager number specified has already been assigned to a Night Bell in this form.
Pager X is not programmed	The selected pager is not a Music-on-Hold/Paging Unit.
The paging group XX has not been assigned	No members in specified group - group undefined.
The power rating of the UNIVERSAL CARD is exceeded	The power rating of the Universal Card is 10. Therefore, the number of modules that can be installed on the Universal Card depends on the individual power rating of the modules. The Receiver/Relay Module has a power rating of 2, the Music-on-Hold/Pager Module has a power rating of 1 and the E&M Trunk Module has a power rating of 3.
Print Option out of range Valid range is (1-52)	In Form 32, CDE Data Print, there are only 50 print option numbers. Select a new print option number.
This printer specification has already been programmed	System already contains printer parameters.
Recording access code XXXXX does not exist	The access code entered for a recording is non-existent or illegal. The code must be for a recording hunt group.

Recording 1 Start Time must be > = Delay For Ringback	The values for these two items must be adjusted as indicated.
Requires Higher Feature Level	The COS option requires the feature level in System Option 102, to be increased to a higher level.
Remote Profile must be in range 1 to 81	The value entered is out of range.
Ring group members cannot have keyline or MCL appearances	Stations/sets can not appear as keyline/MCL anywhere in the system
ROUTE LIST out of range. Valid range is (1 - 100)	The selected route list number is invalid. The range is 1 to 100. Refer to Form 24, ARS: Route Lists.
ROUTE out of range. Valid range is (1 - 200)	The selected route number is invalid. The range is 1 to 200. Refer to Form 23, ARS: Route Definition.
ROUTE PLAN out of range Valid range is (1 - 50)	The selected route plan number is invalid. The range is 1 to 50. Refer to Form 25, ARS: Route Plans.
Route XXX is not an IP route	The route number entered uses non-IP trunk groups. Refer to Form 16, Trunk Groups.
RTE PLN= X, DZ= X, TZ= X, Field= ROUTE LIST: Error= entry must be blank	The entered route list number does not have a corresponding entry in the START HOUR field. Therefore, the ROUTE LIST field must be cleared or a starting time must be specified in the START HOUR field. Refer to Form 25, ARS: Route Plans.
RTE PLN= X, DZ= X, TZ= X, Field= START HOUR, Error = Entry < = previous entry	The START HOUR field entry is less than or equal to the previous START HOUR field entry. The entries in the START HOUR field must be listed in ascending order. Refer to Form 25, ARS: Route Plans.
RTE PLN= X, DZ= X, TZ= X, Field= START HOUR, Error= Entry cannot be deleted	A blank entry in the START HOUR field represents 24 hours. Therefore, the subsequent entry in the START HOUR field is less than or equal to the blank entry. Starting times must be listed in ascending order. Refer to Form 25, ARS: Route Plans.
The Service Profile Identifier (SPID) needs a minimum of 9 digits	The SPID needs to be at least 9 digits long.
SD used in call reroute tables, cannot change!	You cannot rewrite over a programmed index number in CDE Form 19.
Start time of a recording must be < Interflow timeout	Adjust the start time as indicated.
Start time of a recording must be < start time of the next recording	The recording start times should be adjusted as indicated.
A sub attendant prime cannot have keyline or multi-call line appearances	The line must be a single appearance (the prime) in the system.
SUPERSET has an associated device programmed in Form 12....cannot move it.	This DNIC set in Form 09 has a dataset or AIM programmed in Form 12.

The maximum 100 IP nodes allowed are already assigned	Each 200 EL and SX-200 ICP controller supports a maximum of 100 IP nodes
The SUPERSET specified and the PKM have the same feature key	The module and the set have been programmed with the same feature key.
The SUPERSET specified and the PKM have the same key line key	An attempt was made to move a PKM to a set that already has an appearance of a DTS, PRIVATE or KEY line that is already on the module.
The value XX is outside valid range for MAX CALLS (1-60)	The value entered is out of range.
Supervisor XXXXX does not exist	A supervisor with ID XXXXX cannot be displayed as requested via the FIND SUPER key because this access code has not been assigned to a supervisor.
System Option XX conflicts with FEATURE NUMBER YY in Form 2	The given system option conflicts with the specified feature number in Form 02. The option and the feature access code cannot be programmed at the same time.
System Option 106 must be disabled	This option must be disabled before auto attendant groups can be programmed.
System Option 106 must be enabled	This option must be enabled before auto attendant groups can be programmed.
System Option 59 must be programmed before creating an Auto-Attendant group	This option must have a value programmed before an auto attendant group can be programmed.
System ram full !	Maximum devices exceeded. Check maintenance logs for alarm code 106 and 108 for detail
T1 card (for ISDN Node) in slot 2, 4, 6, 8 only	Program ISDN Node T1 cards only in slots 2, 4, 6, or 8.
Template number must be in the range (1-3).	You must enter a template value 1, 2, or 3.
T1 card (for PER Node) in 5th or 6th slot only	Program PER Node T1 cards only in slots 5 or 6.
TENANT NUMBER is out of range Valid range is 1 to 25	There are a maximum of 25 tenant groups.
This group must contain IP trunks only. Trunk xxx is not an IP trunk.	IP trunks cannot be in the same trunk group as other types of trunks.
This group must not contain IP trunks. Trunk xxx is an IP trunk	IP trunks cannot be in the same trunk group as other types of trunks.
Timer is out of range	The valid range for Ring Group timers is from 5 seconds to 5 minutes.
Too many agents programmed, must delete some	The maximum number of agents allowed by your System ID has been reached. Additional agents cannot be programmed.

Too many devices programmed, must delete some	The maximum number of devices allowed by your System ID has been reached. Additional agents cannot be programmed.
Total bay compression resources exceeded	The value entered either exceeds the maximum voice compression resources allowed
TOTAL DIGITS EXPECTED must be in the range (1-9)	In Form 15, Dial-In Trunks, the N field (total digits expected) is restricted to digits 1 to 9. (Generic 1004 and below).
TOTAL DIGITS EXPECTED must be in the range (0 - 10)	In Form 15, Dial-In Trunks, the N field (total digits expected) is restricted to digits 0 to 10.
Total string is too long Limit is 26 digits	The total number of digits in the DIGITS TO BE ANALYZED field plus the digits in the LEADING DIGITS field cannot exceed 26. Refer to Form 26, ARS: Digit Strings.
TRUNK NUMBER XX does not correspond to a CO trunk	Trunk number entered was either a dial-in trunk or a non-existent trunk. Only non-dial-in trunks may be entered here. Only CO trunks can be assigned as DTS in Form 09, (Expand Set Subform) Mitel Telephone Lines.
TRUNK NUMBER XXX is already assigned	Trunk numbers can be assigned to only one trunk. Trunk numbers are assigned in Form 14, Non-Dial-In Trunks and Form 15, Dial-In Trunks.
TRUNK NUMBER XXX has not been assigned	Entered trunk number does not correspond to a trunk. It must be defined in Form 14, Non-Dial-In Trunks or Form 15, Dial-In Trunks.
TRUNK GROUP must be entered for a route to be defined	To complete the route definition, the trunk group number must be specified. Refer to Form 23, ARS: Route Definition.
TRUNK NUMBER XXX is a member of TRUNK GROUP XX	The selected trunk number is a member of the trunk group shown. A trunk number can be a member of only one trunk group at a time.
Unable to delete - A device is programmed for the Bay/Slot/Circuit - XX/XX/XX	Circuits are assigned to the selected card in that bay and slot. Cannot delete the card in that slot until the devices are de-assigned from the forms.
Unable to locate extension XXXXX	The selected extension number is not assigned to any station or set. Refer to Form 09.
Unable to locate extension XXXXX assigned to a PICKUP GROUP	The selected extension number is not assigned to any station or set. Refer to Form 09.
Unable to stop CDE print Try again later	The CDE Data Print process cannot be halted. Try again later.
Undefined index number or digit string	In Form 31, System Abbreviated Dial Entry, either the Index Number or the Digit String is blank. Specify the required Index Number or Digit String.

Unmatched Account code length; system option account code length enabled	The length of the entered account code does not match the account code length specified in System Option 55. (See Form 04, System Options/System Timers). Enter a new account code or change the Account Code Length option.
Unprogram all non-T1 ISDN Cards from bay X before adding 2nd T1 ISDN Card	This message appears if you try to program a second T1 Card while any card, other than a T1 Trunk Card, is programmed in a 96-port bay.
User currently programming feature keys Try again later	Cannot EXPAND SET while set user is accessing Program Feature Key feature.
Valid COS range must be entered before COS print is initiated	Before a print operation can occur, a valid COS range must be entered (1 to 50). Refer to Form 03, COS Define.
Valid TRK CCT DESC range must be entered before print is initiated	This error occurs in Form 32, CDE Data Print. The valid trunk circuit descriptor range is 1 to 25.
Value XX is out of range. Valid range is 1-26 (UNLIMITED)	This error occurs in Form 27, ARS: Maximum Dialed Digits. The valid maximum number of dialed digits is 1 to 26.
Value XX is outside valid range for ACD Agent Group (1-50)	The ACD group number entered is out of range.
Value X is outside valid range for BAY (1 - 7)	The selected bay number is invalid. The range is 1 to 7.
Value XX is outside valid range for CIRCUIT (1 - XX)	The selected circuit number is invalid.
Value XX is outside valid range for CIRCUIT DESCRIPTORS (1 - 25)	There are a maximum of 25 trunk circuit descriptors.
Value XX is outside valid range for COR (1 - 25)	The selected COR group number is out of range. The range is 1 to 25.
Value XX is outside valid range for COS (1 - 50)	The selected COS number is out of range; the range is 1 to 50.
Value XX is out of range for dialed digits	Valid range for the MAXIMUM NUMBER OF DIALED DIGITS field is 1 to 26, where 1-25 is the number of dialed digits and 26 is unlimited.
Value XX is outside valid range for ENTRY NUMBER (1 - XX)	In Form 18, Miscellaneous System Ports, there are only 38 entry numbers.
The value 0 is outside valid range for HUNT GROUP (1 - 99)	The selected hunt group number is invalid. The range is 1 to 99.
Value XX is outside valid range for INTERCONNECT NUMBER (1 - XX)	In Form 30, Device Interconnection Table, there are only 25 Interconnect Numbers.

Value XX is outside the valid range for KEY NUMBERS	The selected Mitel telephone key number is invalid. Key numbers range from 2 to 15 for SUPERSET telephones; 1 to 12 for 12-button PKMs; 1 to 48 for 48-button PKMs; and, 1 to 32 for SUPERSET DSS Modules.
Value XX is outside the valid range for PAGER (1-9)	The pager number specified is invalid. The range is 1 to 9.
Value XX is outside the valid range for PAGING GROUP (1 - 50)	The selected paging group number is invalid. The range is 1 to 50.
Value XX is outside valid range for PICKUP GROUP (1 - 50)	The selected pickup group number is invalid; the range is 1 to 50.
Value XX is outside the valid range for the selected timer option	The selected timer value is invalid. Refer to the Trunk Options Table for a list of valid timer values.
Value XX is outside valid range for SLOT (1 - XX)	The selected slot number is invalid.
Value XX is outside valid range for START HOUR (0 - 23)	In Form 25, ARS: Route Plans, the START HOUR specifies the starting time for each time zone. The time is represented by two digits in 24-hour format.
Value XX is outside valid range for SUBCIRCUIT (X - X)	Subcircuits 1 to 4 refer to the DTMF Receivers; they cannot be accessed. The only subcircuits that can be accessed are the relays on the Receiver/Relay Module. These are subcircuits 5 and 6.
Value XX is outside valid range for TENANT (1 - 25)	The selected tenant group number is invalid; the range is 1 to 25. Tenant group numbers are used in the following forms: Form 05, Tenant Interconnection Table, Form 06, Tenant Night Switching Control, Form 07, Console Assignments, Form 09, Desktop Device Assignments, Form 12, Data Assignment, Form 14, Non-Dial-In Trunks, Form 15, Dial-In Trunks and Form 19, Call Rerouting Table.
Value X is outside valid range for TRUNK GROUP (1 - 50)	The selected trunk group number is invalid. The range is 1 to 50.
Value XXX is outside valid range for TRUNK NUMBERS (1 - 200)	Trunk numbers range from 1 to 200.
Value 0 is outside valid range (1-99)	The selected ACD path or priority is invalid.
Verified account codes system option must be enabled first	In Form 33, Account Code Entry, the account code can only be modified if System Option 05, Verified Account Codes is enabled. Refer to Form 04, System Options/System Timers.
VLAN ID must be in range 0 to 4095	The value entered is out of range.
Voice port of the DNIC console must be programmed first	The DNIC console must be programmed first in Form 07 before its data port can be programmed in Form 12.

Warning: Change will result in IP BAY RESET. CONFIRM or CANCEL.	The user attempted to change IP Signalling options (Form 04).
Warning: This path will be deleted unless 1st two status lines are assigned	The user attempted to QUIT this form with “Access Code for This ACD Path” or “Primary ACD Agent Group” field blank. A path is meaningless without these two pieces of information. The user is now provided with two keys: QUIT, which quits the form and deletes this path, and BACK TO FORM (softkey 0), which returns the user to the form.
Warning: Reassigning agent from group XX CONFIRM or CANCEL	The inserted ID is one that is already programmed in group XX. This agent will be reassigned now to the current ACD group. Press either the CONFIRM or CANCEL softkey.
Warning: Rec X info will be deleted unless Start Time & Access Code assigned	The user attempted to QUIT the form while only one of the indicated status fields for Recording X was assigned. Recording info is meaningless without both these pieces of info. The user is provided with two keys now: QUIT, which quits the form and deletes all entered info about Recording X, and BACK TO FORM (softkey 0) which returns the user to the form.

Blank CDE Forms

This section contains blank CDE forms that you can use to plan system programming.

Forms that cannot be edited are not shown in this section.

You begin planning the programming by defining the location of all peripheral cards and modules in the system.

Note: To edit the following FORMS you should have **MICROSOFT EXCEL Version 97** installed on your PC. Click on the form and EXCEL will be launched.

Form 01 - System Configuration

Form 02 - Feature Access Codes

Form 03 - Class of Service Define

Form 04 - System Options & Timers

Form 05 - Tenant Interconnection Table

Form 06 - Tenant Night Switching Control

Form 07 - Console Assignments

Form 07 - "Expand PKM" Subform Programmable Key Module Lines

Form 08 - Attendant LDN Assignments

Form 09 - Desktop Device Assignments

Form 09 - "Expand Set" Subform Mitel Telephone Lines

Program ARS

Introduction

This section contains a comprehensive description of the Automatic Route Selection (ARS) and Toll Control features of the PBX. The Toll Control feature will allow or deny specific telephones access to certain routes (usually long distance) that are available to the PBX. Parts 2 and 3 provide the reader with background information on the North American Numbering Plan and on the routing options offered to PBX owners by telecommunications companies. A clear understanding of these sections is essential in order to fully implement ARS. The remainder of the document is dedicated to a detailed description of ARS, which concludes with a description of how an ARS plan is prepared on paper, with a scenario centering around a fictitious company.

Within this section, references are made to the customer, the installation company, and the user. These terms are defined as follows:

- The customer is the owner of the PBX.
- The installation company is a company which is authorized by Mitel, to sell and install the PBX. This company works closely with customers to determine their requirements and then installs and programs the system accordingly.
- The user is a person who makes use of the facilities of the PBX through one of the system's peripheral devices (telephones).

When a trunk call is initiated from within a PBX, there are a number of factors which govern its routing and connection. They are:

(a) route availability, where a route is defined as a collection of similar trunks within a trunk group,

(b) cost, when more than one route exists,

(c) caller's toll restriction (i.e., whether the caller is allowed to make such a call, and if so, on what routes).

ARS is a standard feature of the PBX. The ARS feature begins automatically every time a trunk call is initiated and routes the call accordingly. The process is totally transparent to the caller, no access code is required, and the process does not depend on a fixed numbering plan.

Numbering Plans

The ARS feature is universal and is compatible with any numbering plan which may be employed by any public network. It is, however, necessary to understand the numbering plan of the public network which serves the PBX in order to make full use of the toll control application of the ARS feature.

North American Numbering Plan

The purpose of any numbering plan is to enable any subscriber in the network to be connected to any other subscriber in the network. When the North American numbering plan was originally introduced, subscribers were assigned a unique digit string comprising a maximum of ten digits, compiled as follows:

The area code defines a geographic telephone area, the office code identifies a central office (CO) within the area, and the subscriber number identifies a subscriber of the CO.

It was possible to create a distinction between area and office codes by ensuring that the second digit of the area code was 1 or 0 and the second digit of the office code was any digit in the range 2 through 9. However, as the number of COs within each area grew, it became necessary to augment the supply of office codes by allowing the second digit of the code to be in the range 0 through 9. This produced a conflict between area and office codes which was resolved by the introduction of the digit 1 as prefix to all area codes (e.g., 1-613-555-2122).

The prefix digit 1 has now been generally adopted as a toll prefix in large areas, where toll charges are incurred for calls made between offices in the same area (e.g., 1-555-2122).

In addition to the digit strings described above, there are sets of numbers which are reserved for special services; for example, 411 for directory assistance. These numbers do not conflict with area or office codes.

The North American numbering plan comprises digit strings of one, three, seven, eight, and eleven digits.

Some examples are:

Operator	0
Service Number	411
Local Call	555-1111
Toll Call Within an Area	1- home area code -2222
Toll Call to Another Area	1-416-555-3333
Toll Call Within an Area (NO 1 prefix)	557-2222

Call Routing Options

Telephone companies offer a number of different methods of routing calls over the public network (e.g., DDD, tie lines, WATS lines), with each having a different cost structure. Correct use of these trunks can provide substantial cost savings to the user.

To determine which routing options are best suited to any given PBX, a traffic survey should be completed by the installation company prior to installation. The Traffic Measurement and Station Message Detail Recording features of the PBX allow the use of these routes to be monitored once the system is installed, so that they may be modified as traffic demands change.

The PBX supports the following long distance services:

- Direct Distance Dialing (DDD)
- Tie Line
- Foreign Exchange (FX)
- Wide Area Telephone Service (WATS)
- Specialized Common Carrier (SCC)

Direct Distance Dialing

Direct Distance Dialing allows telephone users to call subscribers within the home and international networks without the assistance of the operator. Connections are completed over standard trunk routes and are charged on a usage basis at a rate which varies with distance, time of day, and day of the week. DDD rates are given in the local telephone directory or you can contact the local telephone company for rate information which is not listed.

Tie Line Service

Tie Line Service provides a “tie” between two PABXs. The charge for each tie line is a flat rate charge based on the airline mileage of the line. The figure below shows a typical tie line connection between a PBX in Ottawa and a PBX in Toronto.

IP trunks can also be used to tie multiple systems together. For details, see Networking Mitel IP-PBXs.

Figure: Typical Tie Line

Foreign Exchange Service (FX)

A Foreign Exchange (FX) Line can be thought of as a tie line between a PBX and a Central Office (CO) which is located in a telephone area other than that designated for the PBX. Via an FX Line, the PBX appears as a local subscriber to the distant CO and is billed accordingly for calls which are placed through that CO. FX lines have two applications. The first offers a method of reducing telephone costs in business situations where many toll calls are made to destinations which are within close proximity to one another. For example, a company located in Ottawa which does much of its business with companies located in and around Toronto could benefit from an FX line, as shown in this Figure. The second application allows a company to offer the use of the FX to its customers to permit them to call the company office (the PBX) without incurring toll charges.

Figure: Typical Foreign Exchange Line

Wide Area Telephone Service (WATS)

Wide Area Telephone Service (WATS) is designed to meet the needs of customers who make or receive a large number of long distance calls to or from the same geographical region(s) within the home country. Calls are originated via Outward WATS lines and received over Inward WATS lines (800 Service). Generally, each WATS line is arranged to provide either inward or outward service, but not both.

WATS divides the country into geographical regions known as zones. Zones are incremental, numbering 1 through n, from the home zone. For example, zone 4 provides a WATS subscriber in the home zone (zone 1) with access to all telephone subscribers in zones 1, 2, 3 and 4. Likewise, zone n provides a WATS subscriber in the home zone with access to all telephone subscribers in all zones. This Figure shows Canadian WATS zones 1 through 6 and the zone numbering which is unique to WATS subscribers within Area Code 613, where Mitel Corporation headquarters is located.

The rates for both Outward and Inward WATS are based on the zone and the hours of service subscribed to by the customer.

Figure: Canadian WATS Zoning (Zone 1 Being Area Code 613)

Specialized Common Carrier Service (SCC)

Specialized Common Carrier Service, offered by private companies, provides telephone service between major locations at a rate which may be less than that charged by the telephone companies. The rate is based on a monthly subscription fee plus a usage charge. Specialized common carriers must be approved by local communications regulations and may not be universally available. Currently, SCC services are not available in Canada.

The restriction of this service is that some SCC directories are limited to major locations. Therefore, to avoid additional toll charges, the SCC company office must be within a local dialing distance. A typical SCC arrangement is shown in this Figure.

When a business subscribes to an SCC, it is issued with an account code (normally seven digits). Calls can then be routed via the company's office by dialing a digit string similar to that shown in the following example:

Figure: Typical SCC Arrangement

Detailed Description

The ARS feature is part of the PBX software package. It automatically selects one of a preprogrammed (programmed during CDE) list of trunk routes every time an outgoing call is made. The routes are selected based upon the digits dialed, in order of cost (i.e., least expensive route first), and in accordance with the caller's toll restriction. The use of digit analysis and digit modification within the ARS package allows the system to recognize and modify any digit string which is dialed by the user, alleviating the need for the user to dial special trunk access codes, or to dial a different digit string for each of the various routes to the same destination. The system also recognizes designated digit strings assigned as emergency calls. The emergency designation allows calls to these digits to bypass outgoing call restrictions on all stations except Hotline and Receive Only stations (for exceptions see conditions of Emergency Call Handling).

The complete ARS package provides the following:

- Alternative Routing
- Least Cost Routing
- Toll Control
- Overlap Outpulsing
- Expensive Route Warning
- Callback Queueing
- Camp-on Queueing
- Return Dial Tone
- Emergency Call Handling.

Alternative Routing

Alternative Routing is the automatic selection of an alternate trunk route when the first choice is busy. Routes (e.g., tie trunks or WATS lines) are preprogrammed in an implied sequence of selection within the Route Lists Table.

Least Cost Routing

Least cost routing enables the customer to capitalize on the cost benefits offered by each type of trunk by allowing the installation company to define, via the Route Plans and Route Lists Tables, the order in which the trunk groups are to be selected. A number of different route lists can be defined to account for the fluctuation in rates with respect to the day and time of the week. Route lists are associated with day and time zones through the programming of the Route Plans table and Day Zone table.

Toll Control

Toll control is an integral part of the ARS feature package. It allows the customer to restrict user access to specific trunk routes and/or specific directory numbers.

Every peripheral device which is capable of accessing a trunk is assigned a class of restriction (COR). These CORs are arranged within COR groups which are associated with trunk groups through the programming of the Route Definition table. The Route Definition table defines a trunk group, how the digits dialed are to be modified, and which classes of restriction CANNOT access the route. A maximum of 50 COR Groups, each containing a maximum of 25 COR members, can be programmed. A COR group is simply a list comprised of several COR members. Once constructed, the group is assigned a number (1 to 50). This is the number used in route definition.

Toll control takes place in the following way. Each time a trunk call is initiated, the system checks that the COR of the originating device is NOT included in the COR group assigned to the selected trunk route, verifying that the call is toll allowed (that is, the user is authorized to make the call).

CORs are assigned to peripheral devices during the initial system programming, in accordance with the customer’s requirements, and can be modified at any time from an attendant workstation or CDE terminal by the proper authority (e.g., the telecommunications manager).

Proper ARS programming using COR is the only proper way to minimize toll fraud. For a detailed description on how to minimize the possibility of abuse, refer to the Technical Bulletin, SX-200 ML/EL/DIGITAL CDE Programming For Unauthorized Toll.

Overlap Outpulsing

The basic principle of overlap outpulsing is to seize a trunk and commence outpulsing as soon as sufficient digits have been received to identify the route. This is necessary in order to minimize the post-dialing delay which would otherwise be experienced due to the serialization of digit collection, trunk seizure and digit outpulsing. The number of digits collected prior to outpulsing can be programmed by the customer during customer data entry. These digits may be subject to digit modification prior to being passed to the appropriate sender (dial pulse or DTMF) for outpulsing. Subsequent digits are collected by the system and are outpulsed. At the end of dialing, indicated by an interdigit time-out, or the dialing of a complete digit string of known length, the dialing sender is disconnected. System Option 26, No Overlap Outpulsing, inhibits overlap outpulsing for all calls.

When overlap outpulsing is used, ARS destinations must not have conflicting length differences. Such conflicts will cause the first match to be used, not necessarily the best or specified match. If the following ARS example is used with overlap outpulsing, the first match on 95 will always select route 2. Route 1 will never be selected.

Leading Digits	Digits to Analyze	Route
9	56	1
9	5	2

The post-dialing delay (i.e., the time lapse between the completion of station dialing and the receipt of ringback) which would be experienced when using a DTMF trunk, is minimum (slightly more than 1 second for a 10-digit number). If no overlap outpulsing is enabled, the delay for a 10-digit number outpulsed over a dial pulse trunk would be approximately 16 seconds at 10 pulses per second (pps).

Trunk routes are seized only after the ARS process has determined the validity of the call with respect to the caller's class of restriction. In this way, false traffic will not be generated at the CO (or distant PBX) by aborted seizures.

Expensive Route Warning Tone

The Expensive Route Warning Tone is a programmable option which presents a tone to the user during call setup, and, if a Mitel display telephone is used, the message EXPENSIVE ROUTE appears on the LCD when the route selected by ARS is programmed as an expensive route. Any route but the first one may be programmed to deliver an Expensive Route Warning Tone. When alerted by the warning, the user then has the option of whether or not to continue the call.

Callback Queueing

Callback Queueing (Automatic Callback) allows a user who encounters busy tone after dialing an ARS digit string (i.e., all trunks busy) to dial a callback access code, or, if a Mitel display telephone is used, to select CALLBACK, and be placed in a queue for the first available trunk. When a trunk becomes free, it will be seized, the originating device will be rung back, and, when answered, the previously entered digits will be automatically outpulsed. When honoring a callback, expensive route choices are skipped when ARS scans for an available trunk.

Camp-on Queueing

Camp-on Queueing allows the user who encounters busy tone after dialing an ARS digit string (i.e., all trunks busy) to wait off-hook, or, if a Mitel display telephone is used, to select CAMP ON and remain off-hook until a trunk becomes free. (When a non-display telephone is used, the user remains off-hook for 10 seconds and is automatically camped on to the busy trunk group.) When a trunk becomes free, the system seizes it automatically and the previously entered digits are automatically outpulsed. Expensive route choices are skipped when ARS scans for an available trunk when honoring a camp-on.

Return Dial Tone

Return Dial Tone is a programmable option which allows the system to simulate CO dial tone for customers who consider that its absence would confuse the users of their system. For further information, refer to ARS Digit Strings Tables.

Maximum Digits Dialed

The maximum number of digits that may be dialed is 26.

Emergency Call Handling

Emergency Call Handling allows you to bypass call blocking for emergency calls by designating digit strings (for example; 911 or 8888) as emergency calls in ARS programming. See the feature Emergency Call Handling.

ARS Tables

The ARS package is a network of tables which contain data relevant to the setting up of a trunk call, such as routing options and CORs. The tables are interconnected through a series of indices and pointers. A total of nine tables make up the ARS network. They are, in order of programming:

- ARS Digit Strings
- ARS Nested Digit Strings
- ARS Maximum Dialed Digits
- ARS Route Plans
- ARS Day Zone Definition
- ARS Route Lists
- ARS Route Definition
- ARS Modified Digit
- ARS Class of Restriction Groups.

The hierarchy for the tables is shown in this Figure. The system follows this hierarchy in deciding which routes to select and which users are toll-restricted on the selected routes. The following paragraphs describe the layout and fields of each table. Refer to the Program CDE section for further information. The way in which the tables combine to form the ARS network is described in general in Part 6, and in the scenario given in Part 7.

Figure: ARS Table Hierarchy

ARS Digit Strings Tables (CDE Form 26)

The ARS Digit Strings tables consist of one primary and one nested table. The nested table is accessed from the primary table by pressing the SHOW STRINGS softkey on the attendant console or CDE terminal. (**Note:** If no leading digits are entered on the primary table, the nested table cannot be accessed.) The primary table permits the programming of leading digit information. Actual digit strings and routing information for each of the leading digit entries is programmed on the nested table.

The primary table is shown in this Figure and the nested table in is shown in this Figure.

The primary (leading digits) table is comprised of three fields:

Leading Digits: This is essentially the same as a trunk group access code (e.g., the digit 9) but it may be any digit combination that the customer desires to be analyzed. An asterisk (*) is also permitted as a leading digit. A maximum of 100 leading digit combinations may be specified.

Return Dial Tone: This field allows a simulated dial tone to be returned to the call originator, after the dial 9 access code for trunks has been received by the system, since the central office dial tone is not returned until digit analysis has been completed and a trunk seized. A YES or NO value is entered via the appropriate softkey on the attendant console or CDE terminal. The default value is NO.

Restricted COR Group: This field is optional and may be programmed with a COR group number between 1 and 50. Programming a COR group number in this field will define which group members will NOT be permitted to dial the specified leading digits. If access to specified leading digits is unrestricted, the field is left blank. For example, if all users are permitted to dial the leading digit 9, the field would be left blank. If only those peripheral devices tagged as COR 1 are permitted to dial a given leading digit combination, the COR group must contain ALL CORs EXCEPT COR 1.

Figure: FORM 26 - ARS Digit Strings Table - Leading Digits

The nested form specifies the actual digit strings which are to be analyzed. The form is comprised of four fields; the fourth field being subdivided into two. The fields are:

Digits to be Analyzed: Each line in this programmable field constitutes one entry. Digits programmed in this field are used by the system in conjunction with the leading digits to select the appropriate route. The following wildcard digits may be specified to simplify entering the digit strings:

- N0X
- N1X
- NXX
- X
- 0NXX
- 1NXX
- 1N1X
- 1N0X
- 0N1X
- 0N0X
- 10XXX0N0X
- 10XXX0N1X
- 10XXX1N0X
- 10XXX1N1X
- 10XXX0
- 10XXX1
- 101XXXX0
- 101XXXX1
- 101XXXX0N0X
- 101XXXX0N1X
- 101XXXX1N0X
- 101XXXX1N1X

where N is any digit from 2 through 9 and X is any digit from 0 through 9.

Wildcard digit sequences beginning with N, 0, and 1 may ONLY be used at the BEGINNING of the digit string; X may ONLY be used at the END of the digit string. The wildcard digits allow for the following cases:

- To provide routing for any area code NOT SPECIFICALLY ROUTED, the string NXX followed by specific digit strings will cover all unspecified area codes.

  For example, NXX-555-1212 covers routing for all directory assistance calls.
- To cover routing for any area code NOT SPECIFICALLY IDENTIFIED and where dialing is preceded by a 1 or 0 long distance access code, 1NXX and 0NXX followed by seven digits covers all unspecified area codes. This allows wild card restriction of 555 and 976 numbers.

  For example, 1NXX-976-XXXX and 0NXX-976-XXXX cover routing for all 976 calls.

  If routes are to be selected based on office codes, blocks of office codes can be specified; for example, 82X, 83X, etc.
- The wildcard sequences 10XXX0N0X, 10XXX0N1X, 10XXX1N0X, 10XXX1N1X, 10XXX0, 10XXX1, 101XXXX0, 101XXXX1, 101XXXX0N0X, 101XXXX0N1X, 101XXXX1N0X, AND 101XXXX1N1X are designed for the call aggregator market (hotels, motels, hospitals, universities) to prevent unauthorized calls from being billed to the originating line, while allowing consumers access to the long distance carrier of their choice. They are accessed by pressing the ** MORE ** softkey additional times.

1-10XXX0N0X	2- 10XXX0N1X	3- INSERT	4- 10XXX1N0X	5- **MORE**
6-QUIT	7- FIND STRING	8- DELETE	9- 10XXX1N1X	0-

The system sorts digit strings in such a way that explicitly stated digit strings will be routed to their routes, while all others will be covered by wildcards. The ordering of digit strings is performed automatically by CDE after each string is entered. If two routes are defined for 416 and 416-555-1212, CDE will ensure that the specific string will occur first in the digits to be analyzed field. The number of entries which can be made in this field is limited only by the amount of available system memory.

Wildcard sequences 10XXX0 and 10XXX1 should not be programmed in the same PBX with sequences 101XXXX0 and 101XXXX1. If both sequences are present and a digit string matches both, the 10XXX0 and 10XXX1 strings will be used for routing.

Quantity to Follow: This programmable field specifies the number of digits to be dialed AFTER the digits to be analyzed, and may be specified as UNKNOWN. The advantage of specifying the quantity to follow; i.e., 9-592 plus four digits, is that when the final digit is received, outpulsing can begin, and the DTMF receiver can be dropped; if UNKNOWN is specified, the interdigit time-out must occur before this happens, tying up PBX resources for a longer time than necessary on each call. The total number of digits in this field, and in the digits to be analyzed field, plus the leading digits (from the primary table), must be no greater than 26 digits.

Programming Tip

When programming ARS, it is recommended that Quantity to Follow always be zero. This is accomplished by padding Digits to Analyze with Xs. For example, instead of entering digits as shown in Example A, enter them as shown in Example B:

Example A:

DIGITS TO ANALYZE	QUANTITY TO FOLLOW
613	7
1800	7
1	10
592	4

Example B:

DIGITS TO ANALYZE	QUANTITY TO FOLLOW
613XXXXXXX	0
1800XXXXXXX	0
1XXXXXXXXXX	0
592XXXX	0

Designation: This field specifies which digit string entries require LONG DISTANCE, LOCAL, or EMERGENCY management. The default condition is LOCAL.
The LONG DISTANCE designation enforces COS option 201, Account Code, Forced Entry - Long Distance Calls. This field is also used for Room Status Restriction in hotel/motel applications to restrict long distance calls. A caller with this COS option must enter an account code prior to dialing one of the designated digit strings.
The EMERGENCY designation allows the call to proceed regardless of the restrictions applied to the originating station. Emergency call digits do not have to be 911 calls. This field serves areas that do and do not have 911 service available or areas that have another set of digits for their local emergency code. The EMERGENCY designation will not cause 911 alarms to be generated to the attendant console and will not send CESID information to the network. See the Emergency Call Handling feature.

Termination Type and Number: Digits dialed may terminate on a route, a route list, or a route plan. These two subfields combine to index where each valid digit string is to be found. The first subfield is programmed with one of ROUTE, LIST, or PLAN, depending on whether a route, route list, or route plan is indexed. The second subfield contains the number of the entry within the table referenced in the first subfield. For example, many destinations can be accessed only by direct distance dialing (DDD). For such a destination, ROUTE is specified as the termination type. Free calls such as 911 (in North America) always terminate directly on a route for DDD. If several route choices are available, a LIST is specified as the termination type, if the choices do not vary with time of day. A route plan, with day and time zone variations, is not required. This situation arises where an FX route is always preferred over DDD. In a situation where multiple route choices are offered, with preferences depending on time of day and day of the week, a termination type of PLAN is specified.

Figure: ARS Digit Strings - Nested Table

ARS Maximum Dialed Digits (CDE Form 27)

Countries with open numbering plans require an ARS package that restricts the user based on the number of digits dialed. Each class of restriction has a maximum number of digits which can be dialed associated with it. If the maximum is exceeded, the call follows intercept handling for Illegal number routing. The table, shown in this Figure, is comprised of two fields:

COR: The COR field cannot be modified.

Maximum Number of Dialed Digits: This field is programmable and a value must be specified for each COR. The allowable range is from 1 to 26 digits plus the default value of Unlimited. If a COR group has limited access, UNKNOWN must be entered in the Quantity to Follow column in the nested form of Form 26, ARS: Digit Strings. This ensures that ARS will not drop after analyzing the first digits. As well, System Option 47, ARS Unknown Digit Length Time-out, must be set at an appropriate value so that ARS is not terminated before the CO drops its receivers. In addition, the end-of dial key (#) which is optional via CDE should be disabled. This prevents the user from dialing undetected digits after the system's DTMF receiver has been dropped. If a COR group has unlimited access, UNKNOWN is not required in the Quantity to Follow column in the nested form of Form 26 and a quantity can be specified.

Note: The default value of Unlimited is used in North America and must be specified in this column.

Figure: FORM 27 - ARS Maximum Dialed Digits

ARS Route Plans Table (CDE Form 25)

The ARS package accommodates a maximum of 50 route plans, each of which is contained within a dedicated Route Plans table. The header of each table contains the time zone and the number of the day zone which is to be associated with the route plan. When first accessed, Route Plan 1 is displayed. By selecting the ROUTE PLAN softkey, the current route plan is identified, and the display prompts for the route plan desired: ROUTE PLAN = Entering a number (1 through 50) results in the associated plan being displayed.

This table defines which route list (see Route Lists Table) is to be used in any given time zone (1 through 6), in any given day zone (1 through 3). Up to six time zones may be defined for each day zone, creating a total of 18 possible time zones per week. A different route list may be specified for each of these. When Time Zone 1 is in effect, the Route List used at a given time is determined by specifying a START HOUR. The hour is specified as a 2-digit number (00 through 23). The last START HOUR will create a time period which extends from that time up to the first START HOUR listed which creates a time zone loop for each day zone. If no route list is specified for a given START HOUR entry, all calls accessing the route plan containing this omission WILL BE DENIED.

The Route Plans table is referenced from the ARS Digit String table. The layout of the table is shown in this Figure.

The Route Plans table contains four fields:

Time Zone: This non-programmable field lists the six time zone numbers associated with the currently accessed route plan.

Day Zone 1: This programmable field consists of two sub-fields, used to assign the Start Hour and Route List number. The Start Hour subfield indicates the time (in 24-hour format, 00 to 23) from which the route list is available. The route list is available until the start hour of the next Time Zone. (**Note:** Day zones are planned on the Day Zone Definition table. Day Zone 1 is typically Monday through Friday; Day Zone 2 is typically Saturday; and Day Zone 3 is typically Sunday.) This field on the Route Plans table informs the system as to what times each day zone is to be in effect. The Route List subfield informs the system as to what route list is to be used during these times in these day zones.

Day Zones 2 and 3: These programmable fields are identical to Day Zone 1.

Figure: FORM 25 - ARS Route Plans Table

ARS Day Zone Definition Table (CDE Form 21)

Since telephone call rates vary during the day and with the days of the week, the system must be able to select the least expensive route based on this schedule. The Day Zone Definition table does this. The week may be broken into a maximum of three zones. Typically, these are (1) Monday through Friday, (2) Saturday, and (3) Sunday. The information from this table is used in route plan definition. Refer to the FORM 21 - ARS Day Zone Definition Figure.

The system allows for one definition of three day zones. The Day Zone Definition table accommodates this feature. The table has eight fields:

Day Zone: This is a non-programmable field which lists the zone numbers 1 through 3.

Mon-Sun: These seven programmable fields either ENABLE (shown by *) or DISABLE (shown by a blank) a given day zone on a given day of the week. Positioning the cursor on the desired day of the week results in softkey 1 showing the opposite function to what is entered in that field. For example, if Day Zone 1, MON, is ENABLED (an asterisk is displayed), the softkey will show DISABLE. Pressing the softkey will disable that day zone for that day. The MON field will then be blank for Day Zone 1 and the softkey will now show ENABLE. There must be at least one day zone defined for each day of the week. A user cannot exit from this form if any day does not have one zone defined.

Figure: FORM 21 - ARS Day Zone Definition

ARS Route Lists Table (CDE Form 24)

The Route Lists table contains a maximum of 100 one-line entries, each of which defines up to six routes. The routes within each entry are listed in the order in which they are to be tried; i.e., least expensive (Route 1) to most expensive (Route 6). If there are two or more routes to a given set of locations, and the order in which they are to be tried changes with the time of day because of rate changes, two lists must be programmed to reflect this.

The table makes provision for the assignment of an expensive route warning to each of the second through sixth routes, as required by the customer.

Layout of the table is shown in this Figure. The seven fields which comprise the table are described below:

List Number: This non-programmable field lists the route list entries. Up to 100 route lists, each having up to six route choices, may be programmed.

First: This programmable field defines the first choice (least expensive) route. This route is identified by a one, two, or three-digit number in the range of one through 200. The number in this field is the route number from the Route Definition table.

Second: This programmable field contains one subfield. This field defines the second choice route. This route is identified by a one, two, or three--digit number in the range of one through 200. The number in this field is the route number from the Route Definition table. The subfield enables the Expensive Route Warning (WT), associating the warning with this route. When the warning is required, ON is entered; when the tone is not required, the WT subfield is left blank. The default value is no expensive route warning (blank subfield).

Third through Sixth: These programmable fields are identical to the second field, defining the third, fourth, fifth and sixth route choices.

Figure: FORM 24 - ARS Route Lists Table

ARS Route Definition Table (CDE Form 23)

The Route Definition table contains a maximum of 200 one-line entries. Each entry identifies a route. A route comprises a trunk group, the COR Group associated with the trunk group, and an index to the Digit Modification table. The layout of the Route Definition table is shown in this Figure.

The same trunk group may be used to create several routes; for example, a call in the U.S.A. may be routed via either the primary or secondary Specialized Common Carrier service (SCC). Also, COR groups may be varied so that different COR groups are assigned to the same trunk group. Because of these variations, more routes are required than trunk groups. The PBX therefore allows you to program up to 200 routes which allows for a worst-case situation where four routes are assigned to each of the 50 trunk groups.

The Route Definition table includes a nested table, the Show IP form; use it to program routes for trunk groups that have IP trunks. For details, see ARS Show IP Table (CDE Form 23 Subform).

The Route Definition table contains five fields:

Route Number: This non-programmable field lists the 200 route numbers.

Trunk Group Number: This programmable field identifies the trunk group associated with each route. Entries are made in the form of one or two-digit numbers in the range of one through 50. A maximum of 50 trunk groups may be programmed.

COR Group: This programmable field identifies the COR group associated with each trunk group. Entries are made in the form of one or two-digit numbers in the range of one through 25. A maximum of 25 COR groups may be programmed.

Modified Digit Entry: This programmable field is the Entry Number on the Modified Digit table. Entries are made in the form of one, two, or three-digit numbers in the range of one through 100. This entry is used by the system as an index to the Modified Digit table.

Comments: This programmable field permits the CDE programmer to enter any comment desired against each entry number. The maximum length of the comment is 20 characters.

Figure: FORM 23 - ARS Route Definition Table

ARS Show IP Table (CDE Form 23 Subform)

Routes for trunks groups with IP trunks can be programmed on the Show IP table, which is accessed by pressing the SHOW IP sofkey on the Route Definition table. The layout of the Show IP form is shown in this figure.

For details on how to program IP trunks, see Networking Mitel IP-PBXs.

The Show IP table contains five fields:

ROUTE NUM: This field lists the route numbers. Note that the ROUTE NUM field cannot be modified. There is a maximum of 200 route numbers.

CONCURRENT CALLS: This field defines the maximum number of calls originating from a specified route. Enter a value between 0 and 100.

IP NODE NUM: This field defines the destination IP Node number (as defined in form 48) for this route. Enter a value from 1-255.

COMPRESSION: This field selects G.711 or G.729 compression. Enter "y" to specify G.729 (8 Kbps or enter "n" to to specify G.711 (64kbps) uncompressed voice.

REMOTE PROFILE: This field specifies the profile programming on a remote 3300 ICP. If you are routing to a 3300 ICP, enter the remote profile number here that matches the local profile number. If you are routing to an SX-200 ICP or SX-200 IP Node this information is not used. However the value must still be valid, so use a value of 1 when routing to an SX-200 ICP or SX-200 IP Node.

Figure: FORM 23 Subform - Show IP Subform Table

ARS Modified Digit Table (CDE Form 22)

The Modified Digit table contains a maximum of one hundred 1-line entries. These are sequences indexed through a numerical index, 1 through 100, and are associated with routes through the Route Definition table. Refer to the FORM 22 - ARS Modified Digit Table Figure.

The purpose of digit modification is to allow the station user to dial calls in a consistent pattern, irrespective of the destination of the call or how it is routed. This table directs the system as to how digits are to be inserted into or deleted from the dialing sequence. For example, an FX trunk is installed between two cities, permitting calls to be placed between the two as if they were local calls. A caller in one of the cities placing a call to the other city would dial the distant area code, unaware that the system automatically selects the FX routing for the call. The Modified Digits tables instruct the system to delete the dialed area code from the dialing sequence when the FX route is chosen by the ARS package.

Note: The system will not automatically delete or insert any digits. The digits to be inserted or deleted must be programmed in CDE.

The Modified Digit table contains four fields:

Entry Number: This non-programmable field lists the entry numbers one through 100. The entry number is used in route definition.

Quantity to Delete: This programmable field defines the number of leading digits that the system must delete from a valid digit string prior to outpulsing. A maximum of 25 digits may be specified.

Digits to be Inserted: This programmable field defines the digits and dial tone markers which are to be inserted in place of the digits deleted by the previous field. These digits are prefixed to the modified valid digit string and outpulsed over the trunk. The digits may be telephony digits 0 through 9, and for DTMF trunks 0 through 9, * and #. A maximum of 38 digits can be inserted; including any pauses and wait for dial tone symbols. Special number sequences are:

*1 = Pause for five seconds

*2 = Wait for dial tone

*3 = Switch to DTMF for Subsequent Digits

*4 = Stop or start displaying modified digits. Modified digits are displayed on the sets and in the SMDR records (SMDR must be enabled in CDE Form 16, Trunk Groups). The first time *4 appears in a digit string the system stops displaying the following modified digits. The next time *4 appears, the system starts displaying the following modified digits. You can repeat *4 in a digit string to stop or start the displaying of digits.

*5 = Pause 10 seconds

*6 = Insert caller’s ID (for analog networking)

*7 = Insert caller’s dialed account code (for analog networking)

*8 = Insert PBX node ID number (for analog networking)

*9 = Pause for 1 second

*01 = Identifies call forwarder extension number (for analog networking)

*02 = Identifies call forwarding reason (for analog networking).

To generate * on a trunk, ** must be inserted.

CPN: This fields specifies which number is assigned to outgoing calls—the calling party's extension number or the calling party's CPN. Press the NO softkey to select the extension number, or YES to select the CPN. Program Calling Party Numbers (CPNs) for extensions on Form 54 - Calling Party Number.

Comments: This programmable field permits the CDE programmer to enter comments against each entry number. The maximum length for comments is 20 characters.

Figure: FORM 22 - ARS Modified Digit Table

ARS COR Group Definition Table (CDE Form 20)

The CORs of the peripheral devices are grouped within the Class of Restriction Group Definition table. These COR groups are referenced from the Route Definition table and their contents are interrogated to determine whether or not the calling device has insufficient privilege to complete the call. Absence of the calling device's COR from a COR group indicates to the system that the call CAN be completed. A maximum of 50 COR groups, each containing a maximum of 25 members, can be programmed. COR members are numbered in the range one through 25; COR groups are numbered in the range one through 50. Refer to the FORM 20 - COR Group Definition Figure.

The COR Group Definition table has three fields:

COR Group: This is a non-programmable field which lists the COR group numbers one through 50.

COR Group Members: This programmable field is used to specify which COR group members are to be associated with the specified group number. COR member numbers range from one through 25 and may be entered in any order (member numbers must be separated with spaces); where entries are consecutive, they must be entered in the format 1-13.

Comments: This programmable field is used by the programmer to enter any 20-character-long reminder against each group number.

Figure: FORM 20 - COR Group Definition Table

Key System Toll Control (CDE Form 46)

When a telephone user makes a CO trunk call, either by pressing a CO line key or by dialing a direct CO line access code followed by a trunk number, the dialed digits are subject to digit analysis according to the digit strings in CDE Form 46, Key System Toll Control.

Note: Proper ARS programming using COR is the only proper way to minimize toll fraud. For a detailed description on how to minimize the possibility of abuse, refer to the Technical Bulletin, SX-200 ML/EL/DIGITAL CDE Programming For Unauthorized Toll.

Digit analysis is performed as follows:

1. Try to match the dialed digits against the digit string entries.

   If there is a unique match, go to step 2.

   If there is a partial match, wait for the inter-digit timer to time out before deciding if the user may make a call.

   If there is no match, allow the user to make the call, without checking the termination type or COR.
2. Determine whether the user needs to dial an account code to make this call (Is the LONG DISTANCE option YES or NO?). If the option is NO, go to step 3. If the option is YES and the user has already dialed the account code, go to step 3; otherwise the call is toll denied.

3. Determine whether the termination type matches the selected trunk. If the termination type is a trunk group, the selected trunk must be a member of the trunk group to qualify for a match, otherwise the call is toll denied. If the termination type is a trunk, the selected trunk must match the termination trunk. If the termination type matches, go to step 4; otherwise, the call is toll denied.
4. Determine whether the user's COR is a member of the specified COR group. If it is, deny the call; otherwise, allow the call.

Note: The CO trunk is not seized until enough digit analysis is done to determine that the call may be allowed.

CDE Form 46, which specifies the actual digit strings to be analyzed, has five fields:

Digits to be Analyzed: Each line in this programmable field constitutes one entry. The following wildcard digits may be specified to simplify entering the digit strings (N is any digit from 2 to 9, X is any digit from 0 to 9):

- N0X
- N1X
- NXX
- X
- 0NXX
- 1NXX
- 1N1X
- 1N0X
- 0N1X
- 0N0X
- 10XXX0N0X
- 10XXX0N1X
- 10XXX1N0X
- 10XXX1N1X
- 10XXX0
- 10XXX1
- 101XXXX0
- 101XXXX1
- 101XXXX0N0X
- 101XXXX0N1X
- 101XXXX1N0X
- 101XXXX1N1X

Wildcard digit sequences which begin with N, 1, or 0 may ONLY be used at the beginning of the digit string; X may ONLY be used at the END of the digit string. The wildcard digits allow for the following cases:

To provide routing for any area code NOT SPECIFICALLY ROUTED, the string NXX followed by specific digit strings will cover all unspecified area codes.

For example, NXX-555-1212 covers routing for all directory assistance calls.

To cover routing for any area code NOT SPECIFICALLY IDENTIFIED and where dialing is preceded by a 1 or 0 long distance access code, 1NXX and 0NXX followed by seven digits covers all unspecified area codes. This allows wild card restriction of 555 and 976 numbers.

For example, 1NXX-976-XXXX and 0NXX-976-XXXX cover routing for all 976 calls.

If routes are to be selected based on office codes, blocks of office codes can be specified; for example, 82X, 83X, etc.

The wildcard sequences 10XXX0N0X, 10XXX0N1X, 10XXX1N0X, 10XXX1N1X, 10XXX0, 10XXX1, 101XXXX0, 101XXXX1, 101XXXX0N0X, 101XXXX0N1X, 101XXXX1N0X, AND 101XXXX1N1X are designed for the call aggregator market (hotels, motels, hospitals and universities) to prevent unauthorized calls from being billed to the originating line, while allowing consumers access to the long distance carrier of their choice. They are accessed by pressing the ** MORE ** softkey additional times.

1- 10XXX0N0X	2- 10XXX0N1X	3- INSERT	4- 10XXX1N0X	5- **MORE**
6-QUIT	7- FIND STRING	8- DELETE	9- 10XXX1N1X	

Wildcard sequences 10XXX0 and 10XXX1 should not be programmed in the same PBX with sequences 101XXXX0 and 101XXXX1. If both sequences are present and a digit string matches both, the 10XXX0 and 10XXX1 strings will be used for routing.

The system sorts digit strings in such a way that explicitly stated digit strings will be routed to their routes, while all others will be covered by wildcards. The ordering of digit strings is performed automatically by CDE after each string is entered. If two routes are defined for 416 and 416-555-1212, CDE ensures that the specific string occurs first in the digits to be analyzed field. The number of entries which can be made in this field is limited only by the amount of available system memory.

Figure: FORM 46 - Key System Toll Control

Quantity to Follow: This programmable field specifies the number of digits to be dialed AFTER the digits to be analyzed, and may be specified as UNKNOWN. The advantage of specifying the quantity to follow; i.e., 9-592 plus four digits, is that when the final digit is received, outpulsing can begin, and the DTMF receiver can be dropped; if UNKNOWN is specified, the interdigit time-out must occur before this happens, tying up PBX resources for a longer time than necessary on each call. The total number of digits in this field, and in the digits to be analyzed field, must not exceed 26. This field is ignored unless System Option 26, *No Overlap Outpulsing* is enabled. Entry is disabled by default.

Long Distance: This programmable field is used to specify digit strings which are to be treated as long distance in order to enforce COS option 201, *Account Code, Forced Entry - Long Distance Calls*. A YES or NO value must be specified. With a yes option, a caller must dial a specified internal number and enter an account code to receive dial tone, then select an external line, and then dial the called number.

Termination Type and Number: Digits dialed may terminate on a trunk or a trunk group. Trunks are numbered 1 to 200, and trunk groups are numbered 1 to 50; if only a number is specified, termination type defaults to trunk group.

COR Group: This programmable field identifies the COR group by a number in the range of 1 through 50. If the field is left blank, every COR group can access the digit string. Users with CORs in the listed COR group number are restricted from dialing the specific digit string.

Examples

A user selects trunk 2 and dials 416. From the FORM 46 - Key System Toll Control Figure, the first entry matches the dialed digits. The next step is to check if the user must dial an account code to make the call. For this example, the LONG DISTANCE field is NO; therefore the user is not required to dial an account code. The next step is to check if the termination type matches. The final step is to check if the user's COR is specified in COR GROUP 1; if it is not a member, the user is allowed to make the call, otherwise the call is toll denied.

If the user selects trunk 3 and dials 416, reorder tone is returned. The user is allowed to dial 555 plus seven digits, and any combination of digits other than 416 and NXX types using trunk 3.

To dial an NXX type, the user must access trunk 4.

To dial 613, the user may use trunk 2, 3, or 4 to make the call.

ARS Operation and Programming

The object of ARS is to choose one route for a call to take from one location to another (usually the least expensive) when several routes are available. The ARS package is the software program which instructs the system on how to make the choice. The choice the system eventually does make depends upon the parameters defined within it by the CDE programmer. These are not arbitrary. The parameters are determined by the needs of the PBX.

Programming the ARS features properly requires (1) an understanding of what the customer needs, and (2) what the system must know to reflect those needs. It is important for the ARS programmer to have a good understanding of the cost structure of the different routes leading from the PBX to any called destination, since it is largely on the basis of cost that route selection takes place.

Programming Process - General

In general, the ARS programming process follows this plan:

1. Determine the customer's needs. The needs of the customer will determine what types of calls will be permitted by which peripheral devices. Knowing this, the ARS programmer can assign classes of restriction to the peripheral devices on CDE Form 09, Desktop Device Assignments.
2. Determine the customer's facilities. The ARS programmer must know with what types of trunks the customer is equipped (CDE Form 14, Non-Dial-In Trunks and Form 15, Dial-In Trunks) and the relative cost of each to the customer.
3. Define CORs and COR groups on CDE Form 20, ARS: Class of Restriction Groups, and apply these to trunk groups. The COR Group tables specify which classes of restriction will be toll-denied on a given route.
4. Define day zones (i.e., when rates will vary), modified digits, routes, lists, and plans.
5. Define digit strings. The leading digits and digit strings data are most important, since these form the link between what the set user dials, and what plan, list, or route is used.
6. Specify the maximum number of digits dialed by each COR. For North America: specify UNLIMITED (no further programming required).

Note that the ARS decision hierarchy, as shown in this Figure, is essentially the inverse of the programming procedure. The first data programmed (COR members) are the last used in the ARS decision. The last data programmed (Digit Strings and Leading Digits) are the first used in the ARS decision, and point towards the required route, route list, or route plan.

The rationale for this dual approach to the ARS structure is this: From the system's viewpoint, COR groups and members define the "rank" in importance of each user to the system. When ARS is given a digit string, it will ultimately accept or reject the call on the basis of the rank of the peripheral device attempting to make the call, but in order to do this, it must first determine how the desired call is to be routed. From a programming viewpoint, it is necessary to identify who possesses what rank before access to the various routes can be allowed or denied. In this way, digit analysis programming must take place with the COR of the peripheral devices always in mind.

Programming Process - Key System Telephones

Key system toll control checks for a digit string match; if one exists, the call is denied. If there is no match, the call is allowed. In general, the ARS programming process follows these steps:

1. Determine what types of calls will be permitted by which peripheral devices. Knowing this, the ARS programmer can assign classes of restriction to the peripheral devices on CDE Form 09, Desktop Device Assignments.
2. Determine what types of trunks the customer has from CDE Form 14, Non-Dial-In Trunks, and which ones each telephone may access.

3. Define CORs and COR groups on CDE Form 20, ARS: Class of Restriction groups, that apply to these trunks and/or trunk groups. The COR group tables specify which classes of restriction will be toll-denied on a given trunk. Members of a listed COR group are restricted from dialing the specific digit string.
4. Define Digit Strings. The digit strings data is most important, since it forms the link between what the set user dials, and whether the call is allowed or denied.
5. Specify the quantity of digits still to be dialed (or unknown) for each digit string.
6. Specify whether or not long distance is allowed.

System Programming

When the paper forms are complete, the data must be entered into the system memory through the CDE terminal or attendant console. This is part of the customer data entry process, described in the Program CDE section.

Application

ARS is implemented on the PBX in a 2-stage process. First, data must be collected concerning the customer's needs and the facilities, such as trunk groups, that they possess. From this data, the ARS plan can be formulated on paper. Second, the plan must be transferred from paper to the system memory, through the attendant console or CDE terminal. Refer to the Program CDE section.

Scenario

This scenario begins with the first stage of ARS implementation, namely, the data collection and ARS plan formulation stage. For the purposes of this scenario, a fictitious company is established.

The company has two Canadian locations: its headquarters in Ottawa, and a service office in Winnipeg. The company also has an office in Herndon, Virginia, major accounts and suppliers in the Toronto area, and the company must be able to make international telephone calls. The PBX located in Ottawa is to be programmed.

Trunk Groups

In consulting the traffic studies performed by the PBX installation company, it was decided, in conjunction with the customer, that the PBX in Ottawa would be most cost-effective when connected to the public network via four trunk groups, and an SCC (specialized common carrier) link. The trunk groups were defined as follows:

Trunk Group 1: Local trunks, and specialized common carrier account, for calls to the Herndon office.

Trunk Group 2: Zone 2 WATS Trunks (covering area codes 613, 416, 705, 819 and 514).

Trunk Group 3: Ottawa-to-Toronto FX Line.

Trunk Group 4: Two-way tie line to the Winnipeg office.

The cost guidelines which apply to these groups are:

Tie Lines and FX lines are always less expensive than any other trunk group.

WATS is less expensive than direct distance dialing during the hours of 08:00 through 18:00, Monday through Friday, and 08:00 through 12:00 on Saturday.

SCC is less expensive than direct distance dialing during the hours of 8:00 through 18:00, Monday through Friday, and 08:00 through 12:00, on Saturday.

The following office codes are to be allowed:

Toronto: 471, 825, 678
Winnipeg: 786
Ottawa: All office codes
Herndon: 994

The trunking network for this scenario is shown in the Trunking Networking (Triple Play) Figure.

COR Assignments

The employees at the company's head office in Ottawa were separated into COR groups for purposes of toll control.

COR numbers were assigned to the various workers as follows:

COR 1: Executive. The executive level can access all trunk groups, including the international network.

COR 2: Upper Management. This level can access WATS, FX, tie, and local trunks, and can access the SCC office.

COR 3: Middle Management. This level can access FX, tie, and local trunks, and can access the SCC office with free calls to any area.

COR 4: Technical Staff. This level can access FX, tie, and local trunks.

COR 5: Administrative Staff. This level can access tie and local trunks.

For all other stations not previously assigned, the following COR was given:

COR 6: This level can access internal phones only.

Note: It should be remembered that toll control can be applied not only to individual digit strings, but to trunk groups as well. An example of this is described later in this scenario.

ARS Form Completion

Because ARS involves trunks and trunk groups (both incoming and outgoing), the PBX forms concerning trunks and trunk groups must first be completed before starting the ARS tables.

Figure: Trunking Network

The ARS tables in this Figure and this Figure have been completed using the raw data produced in this scenario. The order in which they were completed is the order in which they would normally be programmed. A detailed description of the contents of the tables is given in the paragraphs immediately following Steps 1 through 3.

1. Complete the COR Group Definition table, listing in each COR group the COR members to be included. The *Comments* field may include reminders concerning which level within the company is contained within each group, or comments concerning the destinations being restricted by each COR group.

   Complete the Day Zone table to provide day zones which satisfy the effect of changing rates for the trunk groups involved.
2. Complete the Modified Digits table. This table instructs the system which digits to outpulse and which to absorb. For example, if the "outside line" access code 9 is not to be outpulsed, the system should be instructed to delete the leading digit 9 from any digit string being analyzed. Similarly, if long-distance DDD calls are permitted, the system could be instructed to insert the digit 1 into the digit string, after 9 has been deleted. Since in this scenario it is known that the SCC network will be accessed, the system can be told to insert the SCC number and account code. The user would then simply dial a 7-digit telephone number (in this case, the office in Herndon). Digit modification need not consider specific user-dialed digit strings, but the various dialing possibilities MUST be considered.

Complete the Route Definition table. Determine how many routes are available for the given trunks and complete the table accordingly. For example, if Trunk Group 1 has five routes available, each route must appear on this table with its own route number.

Complete the Route List table. Assign each route defined a level of “choice”. If Trunk Group 1 has five routes available, order these according to first, second, third, fourth and fifth choices. The priority of the routes is normally dependent on cost.

Complete the Route Plan table. This will permit the system to select a route list on the basis of fluctuating costs due to time of day and day of the week. The hours at which the rates change must be entered in the *Start Hour* column.

3. List the digit strings which are to be considered valid by the system; i.e., those which satisfy the customer’s required access to the public network. The following order is recommended: (1) directory inquiry numbers and local office codes; (2) codes which provide unrestricted access to toll routes (i.e., 0 and 01); (3) specific toll route codes; (4) tie lines; (5) FX lines; (6) WATS lines; (7) calls to be completed via specialized common carriers. As each digit string is entered, specify the number of digits to follow in the *Qty To Follow* column.

   Complete the nested Digit Strings table. Digit strings index a route, route list, or route plan depending on the type of call initiated by the digit string. Analyze each digit string individually and complete the nested Digit Strings table accordingly, ensuring that each digit string terminates appropriately (as a route, list, or plan).

   Whenever possible, specify the maximum number of dialed digits by filling in the digit strings with x's and with quantity to follow as 0, otherwise the default value of unlimited applies.

ARS Digit Strings

The contents of the ARS Digit Strings tables have been composed in accordance with the requirements of the following scenario. Each entry is described below.

- The leading digit has been specified as 9, which, in the scenario, represents the trunk access code of the PBX. Return dial tone is required, therefore YES is specified in the *Return Dial Tone* field.
- Entries 1 and 2 contain digit strings which define free service calls to any area code, using the wildcard digits NXX. These calls are completed over local trunks and only one route is required (Route 9).
- Entries 3 through 5 contain the complete digit strings for emergency services, repair services, and directory assistance. As such, there are no digits to follow and a 0 is placed in the *Qty To Follow* column. These calls can only be completed over one route (i.e., local trunks), and are therefore assigned to the first available route (Route 1) in the Route Definition table.
- Entries 6 through 12 contain a cross section of office codes in the Ottawa area. In reality, it is likely that all office codes would have to be listed, followed by four x's. The number of digits to follow in each case is 0. These calls can only be completed over one route; i.e., local trunks. A route (Route 1), having the same trunk group, COR and digit modification requirements as those required for entries 4 through 10 has previously been defined. Therefore, Route 1 can be entered in the Terminator Type and Number column of these entries.
- Entry 13 provides an example of how access to an individual telephone number can be controlled. Access to this number (a local stockbroker) is restricted to the executive level by indexing it to a route (Route 2) which is associated (through COR Group 3) with COR 1. The number of digits to follow is 0.
- Entries 14 and 15 contain digits which allow unlimited access to the toll call network; i.e., digit 0 for operator assistance and 01 for access to the international network. The number of digits to follow for entry 14 is 0. The number of digits to follow for entry 15 is Unknown, since it is dependent on the call destination. The Qty To Follow entry for this string therefore contains the word Unknown. A route exists (Route 2) and satisfies the COR and digit modification requirements for entries 14 and 15. Route 2 is therefore entered in the Terminator Type and Number column of these entries.
- Entry 16 contains the complete digit string for the service office in Winnipeg. As such, the number of digits to follow is 0. This call can be completed over either of two routes: the tie trunk or DDD. Since

these routes are not time-dependent (tie trunks are always less expensive than DDD), a List (List 1) is defined in the *Terminator Type and Number* column.

- Entry 17 is an example of a toll number within the home area. This digit modification sequence is defined in entry 2 of the Modified Digits table and is referenced from a newly defined route within the Route Definition table (Route 3).
- Entry 18 contains the digit string which defines the number of the plant in Herndon. Calls to this destination can be completed using one of two long distance services: SCC or DDD. The tariff structure for SCC is similar to WATS in that SCC is less expensive than DDD during business hours. As such, the two routes for entry 18 are indexed via the Route Plan table and Route List Definition table. The number of digits to follow is 0 and the first available route plan is Plan 1.
- Entries 19 through 21 contain the digit strings which define toll routes to specific COs within area code 416. The number of digits required to complete a call to any of these offices is 4. These calls can be completed over any one of three routes: FX, WATS, or DDD. These routes are time-dependent (i.e., during some times WATS is less expensive than DDD, and at others, DDD is less expensive), and as such, they are indexed to PLAN 2 in the *Terminator Type and Number* column.
- Entry 22 contains a digit string which defines the area code 416. The digits to follow can be any combination of seven digits. This provides a user, having the required COR, with unrestricted access to any CO within the area defined by area code 416. Like entries 19 through 21, these calls can be completed over any one of three routes which are time-dependent. However, unlike entries 19 through 21, user access is not restricted to specific COs, and therefore an additional set of routes having the relevant CORs is required. Consequently, additional route lists are required to list the new routes and an additional route plan is required to associate the new route lists with day and time zones. Plan 3 is therefore entered in the *Terminator Type and Number* column.

Route Definition Table

Calling devices which are routed to Route 1 from the ARS Digit Strings tables are allowed to complete calls over the CO Trunk Group 1 if they are part of COR Group 1. The digit modification sequence for such calls is defined in Modified Digits table entry 1. Similarly, devices are routed via Routes 2 through 9 from the Route List table.

Route List Table

List number 1 is referenced from the ARS Digit Strings table. It provides alternate routing by listing two routes: Route 4 and Route 5. The routes are listed in order of cost (Route 5 is identified in the Route Definition table as the tie line between Ottawa and Winnipeg), and as such it is always less expensive than the alternative DDD route defined by Route 4.

List numbers 2 and 3 are referenced from Route Plan 1. Within that plan, they are assigned to time zones such that for any given time zone, they define the least cost routing. Each entry lists two routes: Routes 3 and 6 (Route 3 being DDD and Route 6 being identified within the Route Definition table as SCC).

List numbers 4 and 5 are referenced from Route Plan 2. They list three routes (2, 7 and 8) in order of cost for the day and time zones defined within Route Plan 2.

List numbers 6 and 7 are referenced from Route Plan 3. They are similar to entries 4 and 5 with the exception that the FX line is not included.

Route Plan Table

Route Plans 1 through 3 are referenced from the ARS Digit Strings tables. They assign route lists to the day and time zones which are defined in the associated Day Zone table. In Route Plan 1, least cost routing is provided by Route List 2 during Day Zone 1, Time Zones 1 and 2, and Day Zone 2, Time Zone 1, and by Route List 3 in the remaining day/time zones.

Figure: Table Network (Part 1)

Figure: Table Network (Part 2)

Day Zone Table

The Day Zone table defines three time zones for each day. These are combined in the Route Plan table to form nine day and time zones.

In this scenario, three trunk groups are time-dependent: the WATS lines and the CO and SCC trunks. The tariff structure for these two groups is such that WATS is less expensive than DDD on Monday through Friday from 08:00 - 18:00, and on Saturday from 08:00 - 12:00.

COR Group Definition Table

All entries in the COR Group Definition table are referenced from the Route Definition table. Group 1 contains CORs 6 through 25. Peripheral devices which have been assigned any of these CORs are restricted from completing calls via routes which reference COR Group 1. Similarly, COR Groups 2 through 5 define different levels of service.

Modified Digits Table

All entries in the Modified Digits table are referenced from the Route Definition table.

- Entry Number 1 is associated with digit strings in the ARS Digit Strings table where the digits to be outpulsed are identical to those dialed by the user. As such, only the leading digit of the string is absorbed and no digits need be inserted.
- Entry Number 2 is associated with digit strings in the ARS Digit Strings table which represent toll calls and therefore require the leading digit to be absorbed and the toll digit 1 to be inserted.
- Entry Number 3 is associated with digit strings in the ARS Digit Strings table which are outpulsed over the tie line to Winnipeg, and as such, all digits dialed by the user are deleted and a 0 for the distant attendant is inserted.
- Entry Number 4 is associated with calls which are completed via the SCC link. The SCC account code is inserted in the digits to be outpulsed.
- Entry Number 5 is associated with the digit strings in the ARS Digit Strings table which are outpulsed over the FX Line to Toronto. The first four digits are deleted and a dial tone marker is inserted in the DIGITS TO BE INSERTED column.

Scenario - Key System Toll Control

When a telephone selects a CO line to make a call, toll control is invoked before the trunk is seized. Key System Toll Control is essentially a verification to determine if the dialed digits are restricted from being dialed on the selected trunk; otherwise, the call is allowed. As digits are received, they are analyzed; if there is no digit match on the selected trunk, the call is allowed. If there is a match, the long distance and type of termination is checked to see if the call should be denied. If the call is allowed, then the COR Group is checked to see if the caller is allowed to make this call.

A typical CDE Form 46, Key System Toll Control, as shown in this Figure, has been composed for this scenario and each entry is described below.

- Entry 1 allows users who are not included in COR group 1 to make long distance calls to Area Code 416 on trunk 1.

  Area code 416 may be dialed by non-members of COR group 1.
- Entry 2 allows non-members of COR group 1 to dial 555 numbers using trunk 2.

  555 plus seven digits, and any combination of digits other than 416 and N0X types is allowed using trunk 10.
- Entry 3 allows non-members of COR group 3 to dial 976 numbers using any trunks in trunk group 4.

Denied to dial any call to 976 plus four digits.

- Entry 4 allows non-members of COR group 2 to dial N0X calls using trunk 3.

  Allowed to dial N0X long distance calls.

Figure: Typical Key System Toll Control Form

Digits to be Analyzed Strings: These digits are analyzed to find a match, or to determine that there is no match.

Quantity to Follow: The number of digits which are to follow the digit string is listed here.

Long Distance: This column allows or does not allow a long distance call.

Term Type and Number: The type of termination for this key (either trunk or trunk group) and its number are programmed in this column. After the digits are analyzed, this column is checked to verify that the selected trunk is included so that the call may be allowed.

COR Group: The COR Group Definition table lists the COR members to be included in each COR group. The COMMENTS field may include reminders concerning the destinations being restricted by each COR group. The last check that is made before allowing a call is whether or not the caller is a member of a restricted group.

Automatic Data Route Selection (ADRS)

Similar to voice calls, outgoing data calls are dependent upon the digits dialed, time of day, and restrictions set up during CDE. An additional requirement of ADRS is the grouping of trunks by their ability to carry data at a maximum baud rate. Since trunks can presently be grouped and named, no modifications are required to the existing program.

Application

Kanata. In this section, a bottom-up path through the steps taken by the programmer will be followed. This is not intended to represent the full CDE requirements.

In the following example, Mitel has three locations: Kanata, Florida, and England. The programmer needs the following information:

1. Knowledge of available trunks (number, type, and line speed)
2. The assigned class of restriction
3. Toll control requirements
4. Internal numbering plan specifics.

The following outgoing trunks are available:

1. Eight CO trunks, comprising five 600 baud lines, two 1200 baud lines and one 4800 baud line
2. A 1200 baud tie line to Florida.

Trunk Groups

The CO trunks are defined in CDE Form 14, Non Dial-In Trunks, and the tie trunk is defined in CDE Form 15, Dial-In Trunks. The following tables list trunks that are grouped according to their common characteristics in CDE Form 16, Trunk Groups. In this example, shown in the Trunk Groups table, all trunks of the same type and speed are considered to form a group.

Table: Trunk Groups

Trk Grp #	Trunk Type	Group Members
1	600 baud CO	1, 2, 3, 4, 5
2	1200 baud CO	6, 7
3	4800 baud CO	8
4	1200 baud tie	9

COR Groups

CDE Form 20, COR Group Definition, is used to create all necessary COR groupings, as shown in the following table COR Groups.

Table: COR Groups

COR Group	COR Group Members	Comments
1	1	President
2	1-3	President & Upper Management
3	1-6	President & Managers
4	1-9	All but Administration
5	1-15	All

Voice Station Requirements

The Customer Requirements table summarizes the customer's requirements and the following restrictions apply for outgoing voice calls:

1. Everyone is allowed to call Florida using the tie trunk.
2. Technical staff, all management, and the president are allowed to call anywhere in North America using any trunks.
3. Middle management is allowed to call the England office only during evening hours when the discount rate applies.
4. Upper management is allowed to call the England office any time.
5. The president is allowed to call any number (including the operator 9-0) any time.
6. Administration staff are not allowed any long distance calls.

Data Station Requirements

The following restrictions apply for outgoing data calls:

1. The president is unrestricted and can use lines of any speed during the day or night.
2. The 4800 baud CO trunk is used for data calls only.

3. The president and all management are allowed use of the 4800 baud line subject to the time of day restrictions imposed by voice station requirements.
4. Technical staff are allowed use of the 4800 baud line at night only, but for local calls only.
5. Technical staff, all management, and the president can use any trunk to Florida (subject to time of day restrictions listed above).
6. The 1200 baud tie line to Florida is only cheaper than CO trunks with lower or the same baud rates during the day.
7. Administration staff are not allowed outgoing calls originating at a data station.

Customer Requirements

Table: Customer Requirements

Digit String	Route Type	Time	COR's
System Abbreviated Dial Number (England)	600 Baud CO	Day	1..3
600 Baud CO	Night	1..6	
1200 Baud CO	Day	1..3	
1200 Baud CO	Night	1..6	
4800 Baud CO	Day	1..3	
4800 Baud CO	Night	1..6	
System Abbreviated Dial Number (Florida)	1200 Baud Tie line	-	1..15
600 Baud CO	-	1..9	
1200 Baud CO	-	1..9	
4800 Baud CO	-	1..6	
9-305-994-8500 (Florida)	1200 Baud Tie line	-	1..15
600 Baud CO	-	1..9	
1200 Baud CO	-	1..9	
4800 Baud CO	-	1..6	
9-0 (Operator Long Distance)	600 Baud CO	-	1

Table: Customer Requirements

Digit String	Route Type	Time	COR's
1200 Baud CO	-	1	
4800 Baud CO	-	1	
9-Else (Local)	600 Baud CO	-	1..9
1200 Baud CO	-	1..9	
4800 Baud CO	Day	1..6	
4800 Baud CO	Night	1..9	
9-1-Else (Long Distance)	600 Baud CO	-	1..9
1200 Baud CO	-	1..9	
4800 Baud CO	-	1..6	

Day Zones

In this example, the day zones are as follows:

- Day Zone 1, Monday to Friday
- Day Zone 2, Saturday
- Day Zone 3, Sunday

Day zones are entered on CDE Form 21, ARS: Day Zone Definition.

Modified Digits

CDE Form 22, ARS: Modified Digit Table, defines digits which are replaced before outpulsing. Four digits are deleted because there are four digits in each system abbreviated dial number, described later in this section. The digit modification table used in this example is shown below.

Table: Digit Modification

Entry	QTY to DEL	Digits to Be Inserted
1	4	011-44-62872821
2	4	None
3	4	1-305-994-8500
4	1	1
5	1	None

Route Definition

CDE Form 23, ARS: Route Definition, as shown in the table below, is derived from the Customer Requirements table. The Route Definition table finalizes the customer requirements table by specifying a trunk group number and the digit modification to use for the dialed digit string.

Table: Route Definition

Route Num	Trunk Group	COR Group	Mod Digit Entry
1	1	2	1
2	1	3	1
3	2	2	1
4	2	3	1
5	3	2	1
6	3	3	1
7	4	5	2
8	1	4	3
9	2	4	3
10	3	3	3
11	4	5	2
12	1	4	4
13	2	4	4
14	3	4	4
15	1	1	4
16	2	1	4
17	3	1	4
18	1	4	4
19	2	4	4
20	3	4	4
21	3	4	4
22	1	4	5
23	2	4	5
24	3	4	5

Route Lists

If a call can use more than one route, digit translation accesses CDE Form 24, ARS: Route Lists, instead of a route. For example, if the digit string 8822 were produced, route 7, 8, 9 or 10 can be used to complete the call. Note here that the routes in each route list are entered in order of ascending cost. The following Route List table defines the route lists for this example.

Table: Route List

List Num	First	Second	WT	Third	WT	Fourth
1	1	3		5		
2	2	4		6		
3	7	8		9		10
4	8	9		10		7
5	11	12		13		14
6	12	13		11		14
7	15	16		17		
8	18	19		20		
9	18	19		21		
10	22	23		24		

Route Plan

CDE Form 25, ARS: Route Plans, is used when time of day dependencies exist for a given digit string. Each entry in the route plan identifies either a route or a route list from which a trunk group is selected. For this example, the three time zones are:

Time Zone 1 (TZ1) = (08 to 17)
Time Zone 2 (TZ2) = (18 to 22)
Time Zone 3 (TZ3) = (23, and 0 to 7)

The following route plans were created according to the customer's requirements:

Route Plan 1

Time Zone	Day Zone 1		Day Zone 2		Day Zone 3	
Start Hr	Rte List	Start Hr	Rte List	Start Hr	Rte List	
1	08	1	08	1	08	1
2	18	2	18	2	18	2
3	23	2	23	2	23	2

Route Plan 2

Time Zone	Day Zone 1		Day Zone 2		Day Zone 3	
Start Hr	Rte List	Start Hr	Rte List	Start Hr	Rte List	
1	08	3	08	3	08	3
2	18	4	18	4	18	4
3	23	4	23	4	23	4

Route Plan 3

Time Zone	Day Zone 1		Day Zone 2		Day Zone 3	
Start Hr	Rte List	Start Hr	Rte List	Start Hr	Rte List	
1	08	5	08	5	08	5
2	18	6	18	6	18	6
3	23	6	23	6	23	6

Route Plan 4

Time Zone	Day Zone 1		Day Zone 2		Day Zone 3	
Start Hr	Rte List	Start Hr	Rte List	Start Hr	Rte List	
1	08	8	08	8	08	8
2	18	9	18	9	18	9
3	23	9	23	9	23	9

The ARS programmer defines the digit string for the Tie trunk in CDE Form 26, ARS: Digit Strings. This trunk is programmed as shown below:

Leading Digits	Return Dial Tone	Restricted COR Group
89	NO	Unrestricted

In CDE Form 31, System Abbreviated Dial Entry, the following index numbers are defined:

Index Number	Digit String
22	3059948500
23	89
30	0114462872821

These numbering plans result in the following system abbreviated dial numbers (sometimes referred to as system speed call numbers):

Location	Directory Number	Speed Call Number
Florida	305-994-8500	8822
Florida	89 (Tie Trunk)	8823
England	011-44-62872821	8830

An originator can dial either the speed call digit string or the full CO trunk directory number for Florida, but only the speed call digit string is used for any calls to England.

Scenarios

The following examples illustrate how the ADRS/ARS system uses the tables in the above application. Each example assumes a Monday to Friday day zone (DZ1).

Abbreviations used are as follows:

RP1= Route Plan 1
RL1= Route List 1
R1= Route 1
DZ1= Day Zone 1
TZ1= Time

Example 1

At 1:00 PM, a member of the administration staff dials the digit string 9-416-652-5555 from a 600 baud data terminal.

Digit translation follows the path 9-else to select RP4.

1:00 PM represents TZ1 and RP4 [DZ1, TZ1] selects RL8.

RL8 contains the routes R18, R19, R20.

Each of the routes in the route list has legal COR values of [1 to 9], barring their use since the administration staff COR is [10 to 15].

The call cannot be completed.

Example 2

A member of middle management dials the digit string 8830 from a 1200 baud data terminal at 10:00 AM.

Digit translation follows the path 8-8-3-0 to select RP1.

10:00 AM represents TZ1 and RP1 [DZ1, TZ1] selects RL1.

RL1 contains the routes R1, R3, R5.

Each of the routes in the route list allows only COR values from [1 to 3] which bars their use, since a middle manager's COR is [4 to 6].

The call cannot be completed.

Example 3

At 9:00 PM, a member of upper management dials the digit string 9-305-994-8500 from a 600 baud data terminal.

Digit translation follows the path 9-3-0-5-9-9-4-8-5-0-0 to select RP3.

9:00 PM represents TZ2 and RP3 [DZ1, TZ2] selects RL6.

RL6 contains the routes R12, R13, R11, R14.

Each of these routes will allow the call since:

1. They all allow COR values 2 to 3 (upper management)
2. The originator's baud rate of 600 is less than or equal to the baud rate of all trunk groups.

The call can be completed.

Observe the order in which the trunk groups are selected. All CO trunks of baud rates less than or equal to the tie trunk are selected before the tie trunk. Since the call originated during the night, the CO trunks are less expensive than the tie trunk and, as stipulated in the requirements, least cost route selection is demonstrated.

Example 4

At 10:00 PM, a member of the technical staff dials the digit string 8822 from a 4800 baud data terminal.

Digit translation follows the path 8-8-2-2 to select RP2.

10:00 PM is TZ2, RP2 [DZ1, TZ2] selects RL4.

RL4 contains the routes R8, R9, R10, R7.

R7 allows the technical staff's COR. R7 uses Trunk Group 4. Trunks in Trunk Group 4 are each 1200 baud. The call is barred from completion on R7 based on incompatible baud rates.

R8 allows technical staff's COR. R8 uses Trunk Group 1. Trunks in Trunk Group 1 are each 600 baud. The call is barred from completion on R8 based on incompatible baud rates.

R9 allows technical staff's COR. R9 uses Trunk Group 2. Trunks in Trunk Group 2 are each 1200 baud. The call is barred from completion on R9 based on incompatible baud rates.

R10 does not allow the technical staff COR so the call is barred on R10.

The call cannot be completed.

Example 5

At 9:00 AM, a member of the administration staff dials the digit string 8822 from a 1200 baud data terminal.

Digit translation follows the path 8-8-2-2 to select RP2.

9:00 AM is TZ1, RP2 [DZ1, TZ1] selects RL3.

RL3 contains the routes R7, R8, R9, R10.

R7 is the Tie trunk to Florida.

The call completes. If the tie trunk was already in use, the call would not complete since no other route in the route list allows administration COR [10 to 15].

Preventing Toll Fraud

This section explains how to program the ARS CDE forms to minimize the possibility of unauthorized toll access.

All SX-200 ICP systems with any combination of Direct Inward System Access (DISA), Integrated Auto Attendant or ONS interfaced Auto Attendant/Voice mail are susceptible to fraudulent use by unauthorized users.

Note: Proper ARS programming using COR is the only proper way to minimize toll fraud. For a detailed description on how to minimize the possibility of abuse, refer to the Technical Bulletin, SX-200 ML/EL/DIGITAL CDE Programming For Unauthorized Toll.

Analyze All ARS Digit Strings

ARS Digit String entries determine your toll control plan. You should analyze them closely to ensure that they provide the desired level of toll control. It's extremely important that you clearly understand which digit string entries the system will identify as the closest match to the digit string dialed. The digit string that ARS identifies as the closest match may not be the same digit string entry that you intended the dialed digits to match.

For Example:

All stations are Class of Restriction (COR) restricted from Route 9. The entry "X" is intended to pick up all local calls that begin with digits 2 through 9.

Figure: Example 1: Digit Strings Subform for Form 26

At first glance, this ARS program appears to deny all calls to 1-900. This restriction applies if a user dials 9 followed by 1-900. If however, a user dials 9 followed by 1 and then pauses, the ARS Unknown Digit Length Timer will run until it expires and the system will select Route 2. The user could then continue dialing, 9-0-0 or any other digits. This access is allowed because the entry of digit string "1" is the closest match to the digits dialed by the user.

In addition, the SMDR records for these calls may be missing or incomplete. This is a side-effect caused by users pausing during ARS digit strings for lengths of time longer than the ARS Unknown Digit Length Time setting. SMDR records are not generated for these calls because the DTMF receiver is dropped from the trunk when the ARS Unknown Digit Length Time expires if the dialed digit string is matched.

To prevent these types of calls from being routed, program the ARS as follows.

Figure: Example 2: Digit Strings Subform for Form 26

In the form shown in the Example 2: digit Strings Subform above, the entry of "1xxxxxxxxxx" forces the user to enter a full 10 digits after dialing 1 before the call routes out. Some areas may require an entry of "1xxxxxxx" for 1 plus seven digit dialing. In the above example, all calls to 1-900 are toll denied. Test all conflicting ARS entries to determine that insufficient dialing does not cause the system to route calls incorrectly.

With the recent increase in international long distance toll fraud, you should give special attention to 011 calls. Sites that do not normally require international dialing should toll control all 011 calls. These calls could be routed through the console operator for screening so that only internal employees would be transferred to an ARS route that allows 011 to be dialed.

DISA And Dial-In Trunks

DISA presents the greatest potential for abuse by external callers. Two levels of security can be provided by restricting the Class of Restriction (COR) and Class of Service (COS) of the DISA trunk from making external calls, and then allowing the trunk to have external access if a Verified Account Code is dialed after the DISA Feature access code. The Verified Account Code changes the COR and COS of the normally restricted DISA trunk, allowing external access for authorized users. The use of 12-digit account codes provides the greatest number of possible account code combinations, and therefore the highest level of protection. Using 12-digit account codes allows one trillion possible codes. The use of 4-digit account codes only allows ten thousand possible codes.

Auto Attendant

The PBX Auto Attendant feature is very similar to DISA in its operation. The only difference between DISA and Auto Attendant is that the caller is listening to a recorded announcement with Auto Attendant.

ONS Interfaces Voice mail/Auto Attendant (ONS VM/AA)

Proper considerations should be made for toll control of ONS VM/AA ports. Many peripheral systems simply perform blind transfers to any digit sequence entered on the incoming trunk. Some ONS VM/AA system use station ports looped back onto loop start trunks for message-sending setups. This loop-around setup used in conjunction with System Option 22 (Last Party Clear Dial Tone), can allow a caller who has been dropped by the VM/AA to receive dial tone, and possibly proceed to dial through ARS. All station ports used in loop backs should be properly toll controlled and should only have the minimum required COS options.

System Abbreviated Dial

It's important to note that in the SX-200 LIGHT Systems, Abbreviated Dial (speed call) is not subject to toll control. Access to system speed call is controlled through Class of Service programming for stations and Dial-In Trunks that require restriction.

A system speed call entry that consists of manually inserted digits could potentially bypass all COR restrictions. A speed call of this type would be set as follows:

∗3XX

"XX" is any digit 0 to 9 designating the quantity of user dialed digits to be inserted. If a user accesses a speed call entry which has been configured in this manner, the user could enter an ARS Leading Digit and enough digits to cause ARS to route. This type of call would not be subject to COR restrictions.

Direct to ARS

Any device with Direct to ARS enabled in its COS or the COS of an Account Code dialed, can bypass the Restricted COR Group entry programmed against the ARS leading digit programmed in Feature Access Code 37, Direct to ARS. Therefore, a trunk or station that is COR restricted from an ARS leading digit can still place a call out through this leading digit, if it is done through the Direct to ARS feature access code. COR restrictions on specific routes will still apply to all ARS or Direct to ARS calls made by all devices.

Passwords

Change all levels of passwords from the default passwords on systems that have modems connected to the maintenance port.

Troubleshoot

THE FOLLOWING POINTS SHOULD BE CAREFULLY NOTED AND THE INSTRUCTIONS HEREIN STRICTLY OBSERVED :

- Handle circuit cards by the edges only, and ensure that an anti-static strap is used. Card damage may otherwise result.
- Before replacing a card, remove the original card, check for bent or damaged connectors, inspect the backplane, and reseat the original card.
- If a problem has been eliminated through the replacement of a card, temporarily re-insert the original card to verify that the fault is located therein.
- Always provide the maximum amount of relevant data when returning a card for repair.
- Return faulty cards, etc. to Mitel for repair.
- Ensure that a system fault record is always up-to-date, and kept on site.

Introduction

Before troubleshooting is attempted, maintenance personnel should become very familiar with the PBX maintenance system. A complete description of the maintenance system is provided in the General Maintenance Information section. Further maintenance-related information may be found in the documents listed below:

Training Requirements

Only those who have successfully completed a Mitel installation and maintenance training course for the SX-200 ICP System should install or service an SX-200 ICP System.

Troubleshooting Methodology

General

Troubleshooting a malfunction in any complex electronic system is accomplished in a series of logical steps. This section assumes the following basic steps in the troubleshooting of a malfunction:

- GATHERING of information
- CLARIFICATION of the problem
- CONFIRMATION of the problem
- ISOLATION of the problem
- CORRECTION and DOCUMENTATION

When investigating a problem, the troubleshooter should continually verify each step in the isolation process so as to ensure that the system and the symptoms of the malfunction are clearly understood. This will ensure that the malfunction is accurately categorized so that appropriate diagnostics, where applicable, may be invoked.

Information Gathering and Problem Clarification

The Information Gathering and Clarification Chart below provides a list of the information which may be necessary in order to adequately categorize a fault. All relevant information should be gathered and entered into a site fault record. If the fault has resulted in total or partial shutdown of the system, much of this data will be unobtainable or irrelevant.

Chart: Information Gathering and Clarification

Step	Action	Description Follow-up
1.	Talk to station users.	

Obtain the following information:

- frequency of occurrence
- intermittent or continuous nature
- time period during which the fault occurs
- circumstances common to all occurrences (for trunk symptoms, check SMDR records)
- are faults specific to a bay/slot/circuit
- are problems associated with devices connected to a station (headset).

2.	Check Maintenance Alarm indications.

- Check maintenance log for fault / alarm reports (see Note 1).

- Check system LED and numeric display indicators for error codes.
- Ensure that maintenance diagnostics are enabled.

3.	Collect data concerning environmental conditions.

- Check if the system is located close to a heat source or a source of power radiation (see Note 2).

- Note the temperature and humidity conditions and compare with specified operating parameters.
- Check the susceptibility of the area with respect to static electricity generation.
- The following can seriously affect system performance:
  - power fluctuations or grounding
  - lightning storms
  - excessively high humidity
  - excessively high temperature
  - dust
  - rf interference.

4.	Verify system programming.

- Check the existing programming to ensure that the correct options and features have been enabled (see Note 3).

- Verify the Class Of Service assignments, trunk descriptors, and feature access codes. - Ensure all device types (phones, trunks, etc) have separate COS.		
5.	Make special checks for new installations, additions or modifications.	- Check that the procedures specified in the Installation Information section have been properly implemented.
- Verify that any changes have been made in accordance with the appropriate practices, and to the prescribed standards. - Check for possible conflicts if features have been added or deleted, or if other programming changes have been made or for separate COS.		
6.	Make random miscellaneous checks.	- Ensure all circuit cards are properly seated.
- Verify that the system fans are running. - Check the main distribution field for loose or damaged wiring, improperly seated connectors, or other signs of trouble. - Check that telephone line cords are free of defects.		
7.	Check for minor alarm indications - these assist in isolating and categorizing faults.	- Record relevant data and note the affected area of the system.

Notes:

1. Refer to the RS-232 Maintenance Terminal section, for details on procedures.

2. Refer to the Engineering Information section, for the specified operating parameters.

3. Refer to the Program Features section, and the Program CDE section.

Problem Confirmation

Many faults, particularly intermittent faults, “disappear” before the troubleshooter is able to make a positive trace. Wherever possible, attempts should be made to force the problem to recur, such that the effects may be observed and hence the cause determined. The information gathered up to this point may be used to set up conditions relating as closely as possible to those under which the fault originally manifested itself.

Problem Isolation

The aids listed in the table below, Troubleshooting Aids Table, are useful in isolating fault conditions.

Table: Troubleshooting Aids

Troubleshooting Aid	Description and Use
Maintenance Log	Provides a record of maintenance activities and causes of alarms (the primary source of troubleshooting information).
Maintenance Terminal	- Primary access to the maintenance log. - Provides ability to query alarm status, along with a variety of status reports. - Allows testing of individual functional units, using directed diagnostics. - Allows verifying of database in CDE (select Form 01 and select Verify Data). - Allows accessing traffic reports.
Circuit Card Numeric Displays	Allow system power-up testing and operation to be monitored.
Status LEDs (on peripheral cards)	Used to determine if circuit is in use, idle, or not functioning.

Correction and Documentation

Once a problem is isolated, the table of contents of this document should be consulted, and the appropriate procedure referenced. Many procedures contain instructions requiring control circuit cards to be reset, removed, powered down or replaced; in these circumstances it should be noted that these actions will cause a partial or total loss of service. If possible, these procedures should be performed during periods of little or no traffic.

All repairs or adjustments to the system should be recorded into a log book which is kept permanently at the site. Faulty equipment should be returned in the same packaging as the replacement part (FRU). For further information on FRU items, refer to the Field-Replaceable Units section.

General Troubleshooting Steps

Follow the steps below if you can’t find the problem when using the troubleshooting tables.

If you still can’t find the problem, call Mitel Technical Support.

1. Verify the status of the LEDs.
2. In the Maintenance Terminal, review the Alarm details. Identify and fix each alarm.

3. For IP Phone and physical network connectivity problems:
   - Verify that the device has power.
   - Verify the status of the port link integrity LEDs at each end of the cable.
   - Verify that each device transmits a link integrity pulse (LINK LED on).
   - If the link is down, try with another port.Verify that proper cabling is installed between the end devices.
   - Verify that a crossover cable was not installed instead of a straight-though cable.
4. For network media problems:
   - If there is excessive noise, check for cabling problems.
   - If there are excessive collisions check for duplex mismatch problems.
   - For CRC: check if there is a faulty NIC card or flow-control.
   - If there are excessive runt frames, check for bad cables, duplex mismatches or bad PC NIC.
5. For network connectivity problems, identify the path between two end devices by doing the following PING test (in order):
   - Local.
   - Local gateway.
   - Remote gateway.
   - Remote IP.

Troubleshooting System Boot Failure

During the setting of IP address or inadvertent system corruption, incorrect boot parameters may be entered and system operation is jeopardized. Correct information is essential for full functionality. If you are uncertain about current boot parameters, halting the system during boot up and entering a "c" at the [VxWorks Boot]: will allow access to the required information. In most cases a system reboot is unnecessary because the incorrect boot parameters cause the system to be in constant reboot mode. Note: All parameters must be entered in lowercase. If you make a mistake during entry, press enter for each paramater until you pass the last one, and then press “c” to start over. (There is no backspace option.)

NOTE: **The IP addresses used here are examples only and your network address should match your local environment.**

SX-200 ICP CX and MX and AX 9600 vt100 Terminal Emulation	
Parameter	**Setting**
boot device	ata=auto (see Note 1)
processor number	0
host name	bootHost
file name	/partition1/Kts8250.vx
inet on ethernet (e)	192.168.22.2:ffffff00
inet on backplane (b)	
host inet (h)	192.168.20.12
gateway inet (g)	192.168.22.1

SX-200 ICP CX and MX and AX 9600 vt100 Terminal Emulation	
Parameter	**Setting**
user (u)	ftp
ftp password (pw) (blank = use rsh)	@
flags (f)	0x0
target name (tn)	
startup script (s)	
other (o)	motfcc

Note 1: **ata=auto** means that the system will try to boot from the front flash card first (Install/Upgrade flash for AX). **ata=no auto** means that the system will bypass the front flash card (Install/Upgrade flash for AX) and boot from the internal flash card or hard drive only.

Note 2: If AX fails to boot with **ata=auto**. Use **ata=1,0** to boot the system from Install/Upgrade Flash Card.

PRI Card - 38400 vt100 Terminal Emulation	
Parameter	**Setting**
boot device	flash
processor number	0
host name	
file name	vxworks
inet on ethernet (e)	192.168.20.20:ffffff00
inet on backplane (b)	
host inet (h)	
gateway inet (g)	192.168.20.1
user (u)	
ftp password (pw) (blank = use rsh)	
flags (f)	0x0
target name (tn)	pricarda3c5
startup script (s)	

PRI Card - 38400 vt100 Terminal Emulation	
Parameter	**Setting**
other (o)	flash,c:/,xqt,sx2c-na.cmd

NSU - 38400 vt100 Terminal Emulation	
Parameter	**Setting**
boot device	flash
processor number	0
hots name	
file name	vxworks
inet on ethernet (e)	192.168.21.3:ffffff00
inet on backplane (b)	
host inet (h)	
gateway inet (g)	192.168.21.1
user (u)	
ftp password (pw) (blank = use rsh)	
flags (f)	0x0
target name (tn)	
startup script (s)	
other (o)	flash,c:/,xqt,b_loader.cmd

Bay Control Card III - Peripheral Bay	
Parameter	**Setting**
boot device	flash
processor number	0
host name	
file name	vxworks
inet on ethernet (e)	192.168.60.115:ffffff00

Bay Control Card III - Peripheral Bay	
Parameter	**Setting**
inet on backplane (b)	
host inet (h)	
gateway inet (g)	
user (u)	
ftp password (pw) (blank = use rsh)	
flags (f)	0x0
target name (tn)	
startup script (s)	
other (o)	flash,c:\,xqt,sx2c-bri.cmd

Troubleshooting Network Problems

This section contains:

- Troubleshooting IP Trunking
- Troubleshooting IP Phone Registration
- Troubleshooting Phone Connection

Troubleshooting IP Trunking

Use this section to troubleshoot the initial installation of IP trunks and when an IP trunk does not recover.

To Troubleshoot Initial Install of IP trunks:

In CDE Form 4 - SYSTEM OPTIONS/SYSTEM TIMERS :

- Option 115 - Maximum IP Trunks has some value programmed (other than 0).
- Option 86 (PRI Card: Q.sig) enabled to get name and number sent across the IP trunks. The IP trunks will work without this option enabled, but the phones will only show a trunk calling instead of the extension number or name from the remote location.

Make sure that IP trunks are programmed in Form 9 - Desktop Device Assignment and Form 16 - Trunk Group

- Make sure that IP trunks are using a COS and CDN for IP trunks (Default is 7 for both).

In CDE Form 48 - IP NODES:

- Define a unique IP Node number for each location, i.e., Main site is Node 1, Remote site is node 2.
- On main Site, program the IP Node number and IP bay number for this system. The IP address will be 000.000.000.000.
- For the Remote system, program the IP Node number (it will not have a bay assigned), IP address, and the number of calls allowed from the remote location.

- There is a PING Softkey that indicates if the destination is ALIVE or UNREACHABLE. If it is unreachable, make sure the default gateway in Form 47 is programmed properly to get the packets to this remote location.

Program the ARS to get the calls to the remote location. It is recommended to have a unique numbering plan for each location, instead of using an access code access the remote location. If using centralized Voicemail, you cannot use an access code.

- In CDE Form 26 - ARS Leading Digit String:
  - Program the leading digit as the first digit of the extension numbers at the remote site.
  - If they are 3-digit extensions, program XX as digits to analyze and 0 to follow. If they are 4-digit extensions, then program XXX as digits to analyze.
  - Select a route for these calls.
- In CDE Form 23 - Route Definition:
  - Choose the trunk group where the IP trunks are programmed and choose a modified digit table that does not add or delete any digits.
  - When you choose a route that points to a trunk group in which IP trunks are programmed, a Softkey SHOW IP is available.
  - Select SHOW IP, and in the displayed form, choose the number of outgoing calls allowed to this destination, and select a REMOTE profile. If the remote location is a SX-200 ICP, always use Remote Profile 1.
  - Program the IP Node Number for this destination to match what was programmed in CDE Form 48.
  - You can also program here if you want the calls compressed or not. If they are compressed, a call will use approximately 50K of bandwidth per call. If it is uncompressed, it will be approx 100 K per call. It is important to calculate how many calls can be made on the bandwidth available.

Once all the programming is completed, you can go into Maintenance and select REPORTS > SHOW > STATUS > IP TRUNKS. If everything is correctly programmed, it should show OPEN, and you should be able to see the type of PBX at the remote location for each Node and what software is running in that controller.

In Maintenance, go to SYSTEM > MORE > PING, and try to ping the IP address of the remote PBX as programmed in CDE Form 48.

Every IP phone in one location should be able to ping every IP phone in the remote location. If they cannot, they will not be able to talk to the phone at the other end because their voice packets will not arrive there.

If the phones are having one-way audio or you can only call in one direction, it is usually a firewall issue or the default gateway issue.

Check the the Mitel Edocs website (http://edocs.mitel.com) and the Technician's Handbook for the ports that have to be opened on any firewalls between the locations. The main ports are 1066, 1067, 50000 to 50127 and port 9000.

To troubleshoot IP Trunking Problems

1. Go through the General Troubleshooting Steps.

Note: To rule out DHCP problems, and isolate network-related issues, we recommend that you program the IP Phone with a static IP Address in Form 47 or from the phone itself. See Assigning Static IP Addresses to IP Phones.

2. If you still can’t find the problem, call Mitel Technical Support.

IMPORTANT: Make sure you have the following information on hand before calling:

- network diagram

- routeShow, ShowIPSTHLinks, ShowIPSTHRoutes, and ShowIPSTHChannels results
- PING test result between controller and IP Phone

Troubleshooting IP Phone Registration

To troubleshoot IP Phone Registration

Record the error message on the IP Phone display, then go through IP Phone Registration Troubleshooting .

Notes:

1. To rule out DHCP problems, and isolate network-related issues, we recommend that you program the IP Phone with a static IP Address in Form 47 or from the phone itself. See Assigning Static IP Addresses to IP Phones.
2. 5201, 5215, and 5010 IP phones will fail to register on a system that has a Default or Premier database (MX only) because of the line appearances programmed on keys 8 and 10—keys that exist on the 5207 but not on the 5201, 5215 or the 5010. To register these phones, first delete the line appearances in Form 09 (the 5201 must have all appearances deleted) or follow the procedure for replacing a registered phone and re-programming the circuit.

If you cannot solve the problem using the IP Phone Registration Troubleshooting Table, go through the steps in General Troubleshooting Steps.

If you still can't find the problem, call Mitel Technical Support.

IMPORTANT: Make sure you have the following information on hand before calling:

- Is problem with local or remote subnet?
- DHCP server(s) settings
- Layer 2 switch configuration and settings
- Router configuration and settings
- Network Diagram
- IP addressing scheme
- VLAN configuration and settings

IP Phone Registration Troubleshooting		
Error Message on Display	**Probable Cause**	**Corrective Action**
Invalid VLAN ID	DHCP Option not set correctly.	

1. Identify the location of DHCP server and which DHCP server is assigned IP address for the corresponding subnet.

2. For an external Microsoft DHCP server (NT server, etc.), make sure that the option type is set to LONG. **3.** For a Cisco Router DHCP server, make sure that the option type is set to hex, and padded with 0s (for example, 0x00000002 for VLAN 2). **4.** For the controller internal DHCP server, set the option type to numeric.		
Duplicated IP	Existing data device owns the IP address.	

1. Check the IP address on the phone display.

2. Disconnect the IP Phone. **3.** From a PC on the same subnet, ping the suspected IP Phone. If there is a response, identify the data device, and resolve the conflict.		
Corrupted DHCP server	On the suspected DHCP server, disable then recreate the scope. If this is a Microsoft DHCP server, reboot the server.	
DHCP discovery OR DHCP OFFER X REJ	DHCP server does not have enough IP addresses.	Create a larger scope with more IP addresses on the DHCP server.
DHCP server is acting up and cannot assign IP addresses for the corresponding subnet, even though there are enough IP addresses.		

For a Microsoft DHCP server, reboot the server.

For the controller internal DHCP server, disable DHCP and rebuilt the scope.	
L2 switch port is shut down or not configured properly.	Check the L2 switch, and ensure that the port is not shut down. Ensure that this port can access the DHCP server subnet (that is, access the port for the same VLAN, etc.).

DHCP Discovery OR DHCP OFFER X REJ (VLAN) (after releasing the first IP from the native DHCP server)	DHCP Option 125 or 43 (DHCP Option 130 -MITEL IP PHONE prior to Release 4.0) is not programmed up on the second scope of the DHCP server (or on the second DHCP server). OR VLAN ID is not assigned properly

1. Identify if there is one DHCP server for both VLANs, or if there is one DHCP server per VLAN .

2. For one DHCP server for both VLANs, ensure that option 125/43 (option 130 prior to Release 4.0) is programmed in the scope of Voice LAN as String type with value of “MITEL IP PHONE”. **3.** For one DHCP server per VLAN, ensure that option 125/43 (option 130 prior to Release 4.0) is defined in both DHCP servers properly. **4.** Verify that the 'vlan' tag in the option 125 or 43 data string (option 132 prior to Release 4.0) is set to the proper VLAN ID.

One DHCP server for two VLANs network configuration: IP helper (sometimes called DHCP Relay) address on the router interface is not set up correctly.	On the router interface (in which DHCP is not residing), enter the IP helper address and specify the IP address of the DHCP server on the other side of the subnet (that is, always set up IP helper address on the DHCP client side). Ensure the second scope is created for the corresponding VLAN.	
DHCP Discovery OR DHCP OFFER X REJ (VLAN) (after releasing the first IP from the native DHCP server)	The DHCP server is acting up and cannot assign IP addresses for the corresponding subnet even though there are enough IP addresses.	For a Microsoft DHCP server, reboot the server. For the controller internal DHCP server, disable DHCP and rebuild the scope.

The L2 switch port for the phone interface is shut down or not configured properly.	Check the L2 switch and ensure that the port is not shut down. For a Cisco L2 switch, ensure that this is a trunk port with Dot1q encapsulation, and that this trunk port allows both native and Voice LAN to pass through. For an HP L2 switch, ensure that Native Lan is untagged, and that Voice LAN is tagged
The L2 switch port for the router interface is shut down or not configured properly.	

Identify if there are two physical interfaces to the router (one per VLAN), or a router on a stick configuration (one physical with virtual sub- interfaces).
Ensure that the port(s) on both sides (L2 switch and router) are not shut down

If there is a physical interface on the router for each VLAN, make sure that the L2 switch is set to access port for the corresponding VLAN/ subnet correctly OR If there is one physical interface on the router for multiple VLANs, ensure that this is a trunk port on the L2 switch, and ensure that this trunk port allows both native and voice LAN to pass through. On the router subinterface, ensure that the proper VLAN is associated to the remote subinterface		
The DHCP server does not have enough IP addresses.	Create a larger scope with more IP addresses on the DHCP server.	
TFTP load failure	Option 125/43 (Option 128 prior to Release 4.0) is not set up to point to the right TFTP server (Controller).	Check the DHCP server, and confirm that the sw_tftp tag in option 125/43 (option 128 prior to Release 4.0) is pointing to the right TFTP server (usually the SX-200 ICP system IP address).

No network connectivity between the controller and the phone.

Confirm that the controller is connected to the network.

For a Cisco L2 switch: verify that the L2 switch is access port (Voice LAN). For an HP L2 switch: verify that the L2 switch is untagged (Voice LAN). If the router is involved, make sure that router's MTU is set to 600 or more.
Firmware on controller is missing or corrupted.

Verify that the firmware is in the sysro/tftp directory (particularly if the user has manually copied the firmware).

Confirm if TFTP on the controller is corrupted (this can be verified by connecting the IP Phone directly on the controller, or by observing the behavior of other IP Phones).		
Waiting for link	Option 125/43 (Option 129 prior to Release 4.0) is not programmed correctly.	Check 'call_Srv' tag in option 125/43 (Option 129 prior to Release 4.0 on DHCP to confirm that the IP address of the SX-200 ICP controller is correctly programmed.
The Layer 2 switch port may be running spanning tree.	Turn port fast on or disable Spanning tree (whichever is possible).	
The application server is broadcasting or multicasting on IP Phone port or on PC behind the IP Phone.	Turn off multicasting.	
The PC behind the IP Phone is changing speed 100/10.	Depending on the NIC, you may need to hardcode to 100 MB instead of auto negotiation.	
Waiting for link OR Lost link to Server	IP phone fails to receive Keepalive message in 30 seconds.	Verify if the network or the controller is down.

There is electrical interference.	Verify the power source, and change the location of the power source.
TFTP Fail (Remote IP phone (statically programmed IP address) cannot access the SX-200 ICP across the WAN).	The Layer 2 switch port for the phone interface is shut down or not configured properly.

Check the Layer 2 switch and ensure that the port is not shut down

For a Cisco Layer 2 switch: verify that this is a trunk port with Dot1q encapsulation, and ensure this trunk port allows both native and voice LAN to pass through. For an HP Layer 2 switch: verify that Native Lan is untagged and Voice Lan is tagged.
The Layer 2 switch port for router interface is shut down or not configured properly.

Verify which configuration you have

Ensure the port(s) on both sides (Layer 2 switch and router) are not shut down. If there is physical interface on the router for each VLAN, make sure that the Layer 2 switch is set to access port for the corresponding VLAN/subnet correctly; OR If this is a router on a switch, verify that this is a trunk port on L2 switch, and ensure this trunk port allows both native and voice LAN to pass through. On the router's subinterface, verify that the proper VLAN is associated to the subinterface

Typo in IP address, VLAN ID, gateway.	Delete and reprogram the static IP address. If there is no VLAN or priority, don't put 0 and leave them as blank. Verify that the gateway IP address is correct.	
Display 'stuck' at IP address while booting—i.e., fails to advance to "Enter PIN".	Failed to reset controller following installation of new system software.	Reset the controller.

Troubleshooting Phones and Peripherals Problems

If you cannot solve the problem using the following table, go through the steps in General Troubleshooting Steps. If you still can’t find the problem, call Mitel Technical Support

Phone Connection Problems Troubleshooting		
Symptoms	**Probable Causes**	**Corrective Action**
All IP Phones are not receiving power.	Controller not fully initialized.	Wait for the controller to fully initialize. The Power LED should be solid green.
IP Phone won’t boot. Display stuck at DHCP Discovery or IP address (i.e., fails to advance to "Enter PIN")	Phone cannot connect with the DHCP server.	

Check for the following:

Recycle the power to reboot the phone. Verify that a crossover cable was not used to connect the IP Phone to the controller instead of a straight-though cable. Try another phone. If it still fails, plug the phone directly into the Ethernet port on the SX-200 ICP. If it still fails call Mitel Technical Support.

IP Phone Licences exceeded.	Purchase more licences.

Failed to reset controller following installation of new system software.	Reset the controller.
Can't make calls externally (display phone may show call barred).	COR restricted or ARS incorrectly programmed.

Identify the numbers dialed by the user, then check the route used in ARS.

Remove the COR number from COR group table in COR assignment used in the route, OR Change the COR group number in the route assignment form. OR Modify ARS programming to allow call.		
COS restricted for PRI or QSIG trunk.	Enable Public network access via DPNSS in the set's COS.	
Cannot make call over analog loop trunk (intermittent problem).	System is sending the dialing digits too fast for Telco's receiver on the LOOP trunk.	Insert a one or two second delay in ARS: Modified Digit Table (Form 22) for the route used by analog loop trunk.
No dial tone on set.	Set is not programmed.	In CDE, program the extension accordingly.
Wrong wiring	Check the wiring between the phone jack and the Peripheral Cabinet, MDF, etc.	
Faulty handset wire.		

Replace the handset cord.

Replace the handset. Replace the set.
Circuit is manbusy.

Enter the Maintenance command STATE <plid of the circuit>

Enter the Maintenance command RTS <plid>.

Circuit is locked out.	Verify the wiring between the phone and the patch panel.
If NONE of the sets are responding.	Problem with controller.

Check for the following:

- No alarms - The program reboot is scheduled - There are no error logs - The database is backed up

Sets cannot receive DID calls.	Non-DID is enabled in Station Service Assignment form.

Disable non-DID in the set's COS.

Check for NMX fields in Form 15 for Valid entries.	
Sets take 10-12 seconds to receive incoming calls.	ANI/DNIS number delivery trunk option is enabled in T1 trunk's COS.

Disable the ANI/DNIS number delivery in the trunk's COS.

Tip: You can assigned an unused COS to verify if this is the problem.
Dialing digit conflict.

Check the following forms for any potential dialing conflict:

- Desktop Device Assignments - Miscellaneous System Ports - Feature Access Code - ACD Agent Groups		
IP to IP calls OK, but not IP to TDM calls (rings once, then call drops).	Problem with controller.	Call Mitel Technical Support

Phone service is lost (IP Phones display SYSTEM BUSY when they go offhook), and a MOSS alarm appears in the CDE forms header.	A database from another system has been installed in the controller.	Enable the correct options to restore phone service and clear the MOSS alarm. See Change or Enable System Options.
DNIC set displays WAITING FOR SYNC or WAITING FOR COMM at 20 second intervals.		

The set is having trouble communicating with the Peripheral Bay MCC. The likely cause is one of the following:

1. Defective Digital Line Card. 2. Defective bay controller card. 3. Defective set. 4. Loose or improperly installed wiring.	See Miscellaneous Log Reports - Excessive Messaging.

Troubleshooting Phone Audio Quality

To Troubleshoot Phone Audio Quality

1. Go to the table below.
2. If you cannot solve the problem using table 43, go through the steps in General Troubleshooting Steps.
3. Determine the recommended trunk line settings. If necessary, adjust the settings on Form 13 (Options subform).
4. If you still can't find the problem, call Mitel Technical Support

IMPORTANT: Make sure you have the following information on hand before calling:

- Setup scenario
- Is there a common denominator (router, only one side of subnet, etc.)
- Other call scenarios (IP to IP, IP to TDM, etc.)
- Layer 2 switch configuration and settings
- Router configuration and settings
- Network diagram
- IP addressing scheme
- VLAN configuration and settings
- Layer 2 switch port statistics for FCS, collision and duplex mismatch.

Table: Audio Quality Problems Troubleshooting

Symptoms	Probable Cause	Corrective Action
One-way audio between Remote IP to TDM (VM)	No gateway IP address programmed on E2T.	Make sure that the gateway IP address is programmed properly.
One-way audio between remote Teleworker phones and local phones	MAS server and controller on different VLANs (subnets)	Put the MAS and the controller on the same VLAN.
Distorted audio only on NSU	Voice encoding is not inverted.	

1. For T1, enable invert for Voice encoding in link descriptor.

2. For CEPT or API, enable ADI invert for voice encoding. (This does not apply to Embedded PRI.)		
Broken Audio, intermittent (IP trunks only)	Handsfree on the far end.	Ensure that neither device is using Handsfree. Some handsfree phones only operate at half duplex.
Packet loss, jitter via network.		

1. Identify the speech path between the two end points, including router, switch WAN in the audio stream.

2. The network administrator needs to apply QOS/TOS to minimize jitter over slow speed interface (T1, Frame Relay, etc.), and give voice traffic priority over data.
Limited bandwidth and too many calls across router, or combination data and voice.

1. Limit the number of calls to remote subnets.

2. The network administrator needs to apply QOS/TOS to give voice traffic priority over data.
Physical port error (CRC, faulty cable, duplex mismatch, HUB).

1. Identify the speech path between the two end points, including router, switch WAN in the audio stream.

2. Verify that there is no duplex mismatch in each port settings and/or faulty cable, or faulty port. **3.** Make sure that the IP Phone is not plugged into a HUB.	
Compression enabled.	Compression will save bandwidth but may cause noticeable clipping. If not sure, disable compression in Form 23 to see if it makes a difference.

Router's CPU is exhausted or congested.	Router may be running excessive filtering. The network administrator may need to monitor the performance of the router.	
	Echo between LS trunks and IP phones.	Measure the line settings for LS trunks connected to onboard ASUs. If necessary, program matching settings on Form 13 (Options subform).
Handsfree mode is used.	Check if far end is using handsfree. Switch to Handset mode to see if this corrects it, or lower the handsfree's volume.	
Echo Canceller is beyond specifications.		

1. Identify the path between the two end points, and verify if Trunk is always involved.

2. Check with Telco to see if the trunk is within specs. On a typical T1, the input signal should be -15 dB. If it is too high(-5 dB for example), echo may result. **3.** If this is the case, keep an echo log to isolate the problem.		
Network jitter issue.	If the problem only occurs between IP devices, check network jitter issue above.	
Echo between IP Phones	Layer 2 switch setting problem.	

1. Check the L2 switch for duplex mismatch and spanning tree.

2. Turn off if possible.

Troubleshooting Voice Mail

If voice mail is not working properly, refer to the chart below for suggestions on corrective action. If you are experiencing symptoms that are not listed here, call your dealer for assistance.

Symptom/Trouble	**Corrective Action**
Notification calls are configured for a mailbox but don't seem to work.	Notification is not enabled at the system level (see Voice Mail Options Form 49) -or- CO line access is restricted for voice mail port extension numbers. -or- Notification call number incorrectly programmed. Try

Symptom/Trouble	Corrective Action
	dialing the number from a telephone to see what happens.
Notification calls use the correct outside lines but the pager never beeps.	Check the notification phone number and pager type in Form 50, Mailboxes.
The date and time that a message was left is incorrect.	Check the SX-200 ICP system clock.
The system is warning that the disk space is almost full.	Delete unused mailboxes and have subscribers clean out unnecessary saved messages.
Too much silence before or after a greeting or mailbox name.	When recording greetings and names, start speaking immediately after the tone and press any key as soon as you are finished.
When outside callers reach the auto attendant and press 0, either no phones ring or the incorrect phone rings.	Check that mailbox 0 is correctly associated with the operator's extension.
When outside callers reach the auto attendant and press 0, the operator's telephone rings and never forwards to voice mail.	Set Call Forward-Busy/No Answer on the Operator's phone to forward to voice mail.
Internal callers occasionally reach the Operator (or other designated overflow point for the voice mail port hunt group) when calling the auto attendant.	All voice mail ports are busy. Try again later.
Outside callers occasionally reach the Operator (or other designated overflow point for the voice mail port hunt group) instead of the voice mail system.	All voice mail ports are busy. Try again later.
The voice mail system resets itself at times.	The voice mail system has the ability to automatically reset itself in the event of a critical error. The SX-200 ICP can also be programmed to reset following a fault or number of faults.
The message lights do not work.	

Verify that you have the proper access code to turn the light on by dialing it from a phone

Note: The ONS/CLASS circuits in the SX-200 ICP controller do not support Message Waiting lights.	
No message light exists on the phone.	Set up message notification to ring the extension number whenever messages are received.
Prompt to “please leave a message" heard when a a busy or unanswered extension is reached repeats.	Bilingual prompts enabled (System Option 121) but only one language installed. Install the missing language. See Installing Software Using an External CompactFlash Card (Optional).

Troubleshoot Cards

Digital Service Cards

PRI Card

If you suspect a problem with a PRI card, do the following:

- Verify faceplate LEDs
- Run a terminal emulator program on the PC to look at the log messages while the card boots up
- Check external equipment and connections
- Verify CDE programming
- Check for bent pins on the backplane and inside shelf
- Check the cabinet connections.

If the NFAS, QSIG, or D-Channel Backup options do not match in IMAT and the PBX, the D-Channel will not function. The error will appear in the PRI card maintenance logs and the PBX will give an alarm message to the PBX maintenance logs. Reprogram the options and reboot the PRI card.

If the Min/Max and Auto Min/Max do not match in the IMAT and the PBX, the D-Channel will function. The error will only appear in the PRI card maintenance logs.

If you are unable to establish a connection between the IMAT PC and the PRI card, check the Dial-Up Networking connection:

PRI Card Maintenance Logs

There are three categories of logs for the PRI card:	
INFORMATIONAL	Events with no or low impact on performance
WARNING	Events with an impact on performance or which may lead to an operational failure
FATAL	Events that will lead directly to an operational failure (a FATAL log results in a system reset). Usually reading a FATAL log means that there has already been a failure.

Peripheral Interface Circuit Cards

ONS / OPS / Station Line Cards

This section covers the ONS/CLASS Line Card and the OPS Line Card. The table below, ONS/CLASS Line Card Troubleshooting Summary, outlines the most likely items to cause malfunction. For ONS/CLASS Line Card problems involving Voice mail, see Voice mail - ONS Port.

Table: ONS/CLASS Line Card Troubleshooting Summary

Step	Possible Malfunction Source
1.	Faulty connections between the telephone and the cross-connect field.
2.	Faulty connections between the cross-connect field and the system.

3.	

Faulty CDE programming; likely forms:

- COS Define
- System Options/Timers
- Desktop Device Assignments
- Device Interconnection Table
- System Configuration

4.	Faulty telephone.
5.	Faulty line card.
6.	Faulty backplane cable connections.
7.	Software and Hardware requirements are not met. CLASS functionality requires the purchasable FOR options 102 (Feature Level) and 89 (CLASS Functionality for ONS Sets). The ONS/CLASS Line card must be in an SX-200 ELx cabinet with a BCC III that has a DSP module (single).
8.	Problem with DSP receivers or DTMF Receiver modules (not enough for peak traffic load).

Note: Useful maintenance commands: SHOW STATUS, TEST, SHOW ALARMS, DIAGNOSTICS ENABLED - see the RS-232 Maintenance Commands section. Also check Traffic Measurement.

Digital Line Card

General

This section covers the Digital Line Card (PN 9109-012). The table below, Digital Line Card Troubleshooting summary, outlines the most likely items to cause malfunction.

Table: Digital Line Card Troubleshooting Summary

1.	Faulty connections between the telephone and the cross-connect field.
2.	Faulty connections between the cross-connect field and the system.
3.	Bridge taps or incorrect loop lengths can cause intermittent problems. Loop length limits for devices connected to Digital Line Card or DNIC Module circuits is in the Engineering Information section.
4.	

Faulty CDE programming; likely forms:

- COS Define
- SUPERSET Telephones
- Device Interconnection Table
- System Configuration

5.	Faulty telephone.
6.	Faulty Digital Line Card.
7.	Faulty backplane cable connections.
8.	Problem with DSP receivers or DTMF Receiver modules (not enough for peak traffic load).

SHOW ERRORS DATASET Command

The SHOW ERRORS DATASET command can be a useful tool in the troubleshooting of both the Digital Line Card and the Dataset. The following table, SHOW ERRORS Command, lists the possible causes of the error types

Table: SHOW ERRORS Command

Error Type	Call State	Possible Cause				
		Dataset	Cabling	DX Chip	Line Card	DTE/DCE
CRCERR	any	yes	yes	yes	yes	no
RESETS	any	yes	yes	yes	yes	yes
LINK FAIL	Call setup/ Talk	yes	yes	yes	yes	no
LINK ABORT	Call setup/ Talk	yes	yes	yes	yes	yes
PARITY	any	yes	no	no	no	yes
OVERFLOW	Talking	no	no	no	no	yes
OVERRUN	Talking	no	no	no	no	yes
FRAMING	Talking	no	no	no	no	yes
NOSYNC	any	yes	yes	no	yes	no

Additional Troubleshooting

Digital Line Cards or DNIC Modules are used for many applications. For additional troubleshooting information, refer to procedures for the specific application:

- SUPERSET 4001 Telephones
- SUPERSET 4015 Telephones
- SUPERSET 4025 Telephones
- SUPERSET 4125 Telephones

- SUPERSET 4150 Telephones
- SUPERSET 401+ Telephones
- SUPERSET 410 Telephones
- SUPERSET 420 Telephones
- SUPERSET 430 Telephones
- SUPERSET 3DN Telephones
- SUPERSET 4DN Telephones
- DATASET 1103
- DATASET 2103
- Programmable Key Modules

Fiber Interface Modules (FIM) and FIM Carriers

Guidelines for Handling Optical Fiber Cable

When handling fiber cable:

- Never touch the tip of a fiber connector. Cleanliness of the connector ferrule (tip) is important for error free transmission.
- Always place the dust caps onto the connectors immediately after disconnecting.
- Clean the ferrule tips on the connectors with ethyl-alcohol.
- Fiber-optic cables are often more easily installed and pulled than copper because of their light weight and flexibility. However, take care not to exceed the minimum bend radius or maximum tensile strength.
- Procedures for the repairing, splicing, or assembling of fiber cables are available from fiber component manufacturers who offer relevant training courses.

WARNING: **Fiber optic sources emit infrared light invisible to the human eye which can damage the retina. Never look directly into a source or into the end of a fiber energized by a source.**

When working with raw fiber optic cable be careful of fiber ends or slivers that can puncture the skin and cause irritation.

Correcting Frame Synchronization Errors Caused by Loop Length of Fiber Optic Cable

To correct frame synchronization errors caused by loop length of fiber optic cable, increase the loop length to more than 10.6km (6.6 miles). If the existing loop length is not known, increase the loop length by 600m (1968 ft.).

To increase loop length of fiber optic cable

- add 300m (984 ft.) of two-strand or “paired” fiber optic cable to the cable run

or

- add 600m (1968 ft.) of single strand fiber optic cable to either the control node to peripheral node run or the return run.

Do not increase the run length between nodes beyond 14 km, otherwise the optical budget may be exceeded.

Peripheral Interface Module Carrier Card

The following table lists the most likely causes of a faulty Peripheral Interface Module Carrier Card.

Table: Fault Analysis

Step	Possible Malfunction Source
1.	Faulty or dirty transmit or receive connections.
2.	Card not seated properly in the cabinet.
3.	Incorrect FIM II. Verify that each end of a link has the same FIM variant (1-km, 3-km, or 14-km). Note that a 1-km FIM II can connect to a 1-km FIM.
4.	Faulty fiber cable or copper cable.
5.	Faulty connection of the FIM II or CIM.
6.	Faulty FIM II or CIM.
10.	Faulty pin out for the Cat. 5 cable exceeds 30 meters or 100 feet.
11.	Faulty power input.
12.	The S1 switch on a PRI card used as a control carrier card in a main control cabinet may be set incorrectly.
FIM II Troubleshooting	
1.	To test the FIM II, connect a fiber cable from the Tx of the FIM II to the Rx connector of a FIM II or FIM in the control bay. Both LEDs should light steady indicating that the FIM and the cable are good. You can then reverse this procedure with the Rx of the FIM II in question.
2.	Test another FIM II on the PIMCC card.
3.	Verify each fiber cable with the method in step 1.

Fiber Interface Module Types

1-km:	maximum loop length is 1 km
5-km:	maximum loop length is 5 km
14-km:	maximum loop length is 14 km

Peripheral Interface Module Carrier Card Troubleshooting	
1.	Verify that the correct type of fiber cable is being used (1-km and 5-km FIMs use 62.5/125 mm multimode cable while 14-km FIMs use 9/125 mm single mode cable).
2.	Verify that the cables from the PIMCC card are correctly connected to the control cabinet.
3.	If using copper, ensure that the correct pinout of the Cat.5 cable has been followed. See CIM Ports. Perform a loopback on the Cat. 5 cable if desired.
4.	If using copper, ensure that the Cat. 5 cable does not exceed 30 meters or 100 feet.

5.	Check the connection of the module on the card.
6.	Replace the FIM II or CIM on the PIMCC card.

Trunk Cards - General

The procedures detailed below cover the isolation and correction of faults with the various trunk cards. Supplementary procedures are provided for specific trunk types where required.The following table summarizes troubleshooting for trunk cards.

Table: Trunk Card Troubleshooting Summary

Step	Possible Malfunction Source
1.	Faulty connections between the external trunk equipment and the cross-connect field. See Note 3.
2.	Faulty connections between the cross-connect field and the system.
3.	

Faulty CDE programming; likely forms:

- COS Define
- Trunk Circuit Descriptors
- Non-Dial In Trunks
- Dial In Trunks
- Trunk Groups
- Device Interconnection Table
- Form 04, System Options
- System Configuration.

4.	Faulty external trunk equipment (see Note 4).
5.	Incorrect jumper settings (LS/GS Trunk card only) - see the LS/GS trunk card Figure.
6.	Incorrect switch settings (E&M Trunk Module).
7.	Faulty Universal Card (E&M Trunk Module only).
8.	Faulty LS/GS, LS/CLASS, or DID Trunk Card (see Note 3).
9.	Faulty backplane cable connections.
10.	Problem with DTMF Receivers (not enough for peak traffic load, see Note 2).

Notes:

1. Useful maintenance commands: SHOW STATUS, TEST, SHOW ALARMS, MONITOR SMDR, LOGS - see the RS-232 Maintenance Terminal section. Refer also to Traffic Measurement.
2. Refer to the Engineering Information section, for details on CO Busy Hour and receiver provisioning.
3. If periodic ‘opens’ in loop current are experienced during call progression through a central office, and disconnect timing has been set for a short interval, loop disconnects may cause the trunk to drop. In such cases, the timing should be increased one step at a time, until the calls are no longer dropped.
4. When the system detects a trunk protocol failure, it removes the affected trunk circuit from service. Trunk protocol failures may indicate fault(s) with the trunk circuit, in which case the card should be tested and replaced. However, they may also indicate that the external trunk is faulty. In this case, the affected trunk circuit must stay out of service until the external equipment is repaired.

Figure: LS/GS Trunk Jumper Locations

Trunk Voltage and Loop Current Readings

Table: Trunk Voltage and Loop Current Readings

Circuit	Idle	Seized
E & M (Type 1)	Tip to gnd = 0 V	Tip to gnd = 0 V
	Ring to gnd = 0 V	Ring to gnd = 0 V
	Tip to Ring = 0 V	Tip to Ring = 0 V
	E lead = -48 V	E lead = 0 V
	M lead = 0 V	M lead = -48 V
	E to M lead = 48 V	E to M lead = -48 V
	I loop = 0mA	I loop = 0mA
LS/GS or LS/CLASS (Loop Start)	Tip to gnd = 0 V	Tip to gnd = -14 V to -22 V
	Ring to gnd = -48 V	Ring to gnd = -34 V to -26 V
	Tip to Ring = -48 V	Tip to Ring = -4 V to -20 V
	I loop = 0mA	I loop = 18 mA to 90 mA
LS/GS (Ground Start)	Tip to gnd = -48 V	Tip to gnd = -14 V to -22 V
	Ring to gnd = -48 V	Ring to gnd = -34 V to -26 V
	Tip to Ring = 0 V	Tip to Ring = -4 V to -20 V
	I loop = 0mA	I loop = 18 mA to 90 mA
DID	Tip to gnd = -2 V	Tip to gnd = -33 V to -44 V
	Ring to gnd = -48 V	Ring to gnd = -17 V to -6 V

Table: Trunk Voltage and Loop Current Readings

Circuit	Idle	Seized
	Tip to Ring = -46 V	Tip to Ring = 16 V to 38 V
	I loop = 0mA	I loop = 12mA to 30mA
Loop/Tie	Tip to gnd = -2 V	Tip to gnd = -17 V to -6 V
	Ring to gnd = -48 V	Ring to gnd = -33 V to -44 V
	Tip to Ring = -46 V	Tip to Ring = -16 V to -38 V
	I loop = 0mA	I loop = 12mA to 30mA

Figure: E&M Trunk Module With Universal Card

Supplemental E&M Trunk Card Troubleshooting Procedures

Table: Supplemental E&M Trunk Troubleshooting Procedures

Step	Action	Description / Follow-Up
1.	Perform General Trunk procedures in the Trunk Card Troubleshooting Summary table.	• Go to step 2.
2.	Disconnect the affected circuits from the cross-connect field.	• Go to step 3.
3.	IDLE STATE TEST - connect voltmeter between -48V and the M lead. Reading should be -48V (see this Figure).	• If not -48 V, replace E&M card, or E&M module and/or Universal Card. Otherwise, go to step 4.
4.	INCOMING TEST - seize the trunk incoming - connect butt-set to E lead and ground. Circuit indicator should light when butt-set goes off-hook (see this Figure).	

• If not, replace card/module; if fault persists, possible control problem - go to step 11.

• Otherwise go to step 5.	
5.	Check if incoming wink is programmed.

Yes: • Go to step 6.

No: • Go to step 7.	
6.	Connect voltmeter to M lead and ground. Flash of -48V should be seen when butt-set goes off-hook (see this Figure).

• If not, replace card/module; if fault persists, possible control problem - go to step 11.

• Otherwise go to step 7.		
7.	Connect voltmeter to M lead and ground. Complete a call to an extension - when call is completed, steady -48V should be seen (see this Figure).	

• If not, replace card/module; if fault persists, possible control problem - go to step 11.

• Otherwise go to step 8.		
8.	OUTGOING TEST - repeat step 3.	• Go to step 9.
9.	Connect voltmeter to ground and the M lead.	• Go to step 10.
10.	Connect butt-set to the E lead and ground, and dial the access code for a trunk group. The reading should be a steady -48 V (see this Figure).	

• If not, replace card/module; if fault persists, possible control problem - go to step 11.

• Otherwise card is functioning.		
11.	Possible control problem - refer to Subsystem Troubleshooting Procedures.	

Supplemental DID / Loop-Tie Trunk Troubleshooting Procedures

Table: Supplemental DID / Loop-Tie Trunk Troubleshooting Procedures

Step	Action	Description / Follow-Up
1.	Perform General Trunk procedures in the Trunk Card Troubleshooting Summary Table.	• Go to step 2.
2.	Disconnect the affected circuits from the cross-connect field.	• Go to step 3.
3.	Connect the butt-set across the Tip and Ring of the trunk circuit - the circuit indicator should light when the butt-set goes off-hook (see this Figure).	

• If not, replace card; if fault persists, possible control problem - go to step 9.

• Otherwise go to step 4.	
4.	Check if circuit is a DID trunk.

Yes: • Go to step 5.

No: • Go to step 6.

5.	Use butt-set to simulate incoming digits - connection should be made to an extension/attendant, etc., depending upon call routing. The trunk circuit indicator should wink following digits pulsed.	

- If not, ensure extension/attendant console, etc. is functioning properly - replace card; if fault persists, possible control problem - go to step 9.
- Otherwise go to step 7.

6.	TIE TRUNK - dialing, or going off-hook from the butt-set should connect to an extension/attendant, etc., depending upon call routing.	

- If not, ensure extension/attendant console, etc. is functioning properly - replace card; if fault persists, possible control problem - go to step 9.
- Otherwise go to step 7.

7.	Connect voltmeter across the Tip and Ring of the trunk circuit (see this Figure).	• To check wink start or answer back supervision, go to step 8.
8.	Check the results during a simulated incoming call from the butt-set.	

- When seized, meter should read -18 to -20 V.
- For a wink start, the meter should read a 180 ms positive flash, and back to -18 to -20 V.
- For answer back supervision, deflection to +18 to +20 V should be seen.
- If these readings are not seen, retry. If this persists, replace the suspect card; if fault still persists, possible control problem - go to step 9.

9.	Replace card with a known working card.	• If problem persists, reinstall the original card and refer to Subsystem Troubleshooting Procedures

Figure: E&M Type 1 Trunk Testing

Figure: Loop Trunk Testing

T1 Trunk Card

The T1 Trunk Card provides an interface between the system and external digital trunk facilities. Each card contains one 24 channel interface. The T1 Trunk Card must be installed in a high power card slot.

Table: T1 Trunk Card Troubleshooting Summary

Step	Possible Malfunction Source
1.	Faulty external equipment or far end.
2.	T1 card not installed in proper slot to fan SX-200 FD cabinet. **Note:** with two T1 Trunk Cards in a bay, they must be in slots 5 and 6 and require a dual T1 adapter.
3.	T1 card installed in slot 10 or 11 of an SX-200 RM cabinet and slots 5 and 6 NOT vacant.
4.	Faulty connections between the T1 Channel Service Unit and the cabinet end of the T1 Adapter Cable Assembly.
5.	Faulty connection between the T1 Adapter Cable Assembly and the T1 Adapter Card.
6.	Faulty connection between the T1 Adapter Card and the shelf backplane.
7.	Faulty CDE programming; likely forms: • T1 Link Descriptor • T1 Link Assignment • T1 Network Sync • COS Define (COS Option 802 for outgoing T1 • Trunk Circuit Descriptors • Non-Dial In Trunks • Dial In Trunks • Trunk Groups • Device Interconnection Table • System Configuration.
8.	Faulty T1 Channel Service Unit (see Note 2).
9.	Incorrect switch settings on T1 Trunk Card (see Note 3).
10.	Faulty T1 Trunk Card (see Note 2).
11.	Faulty T1 Clock in the SX-200 ICP.
12.	Faulty SX-200 ICP controller.

Notes:

1. Useful maintenance commands: SHOW STATUS, TEST, SHOW ALARMS, SHOW ERRORS T1_TRUNK - see the RS-232 Maintenance Terminal section.
2. When the system detects a trunk protocol failure, it removes the affected trunk circuit/channel from service. Trunk protocol failures may indicate fault(s) with the trunk circuit/channel, in which case the card should be tested and replaced. However, they may also indicate that the external trunk is faulty. In this case, the affected trunk circuit/channel must stay out-of-service until the external equipment is repaired.
3. CDE Form 14 and 15 do not accept trunk programming changes if the T1 card or T1/E1 module is installed and the T1 link is down but still active. To make a programming change for a T1 card, unplug the T1 card, make the programming change and then re-insert the T1 card . To make a programming change for a T1/E1 module, insert a Peripheral Interface Card (PIC) into the software location (slots 5 or 6) for that T1 link, make the programming change, and then remove the PIC from the slot to reset the T1 link.
4. The switch settings are for cable length (not loop length) between the T1 card and the Channel Service Unit. The correct settings are:
   0 to 149 ft: set S1 CLOSED; S2-S8 OPEN.
   150 to 449 ft; set S2, S3, S4 CLOSED; S1, S5-S8 OPEN.
   450 to 655 ft; set S5, S6, S7 CLOSED; S1-S4, S8 OPEN.
   These switch settings all assume 22 gauge wire.

LS/CLASS Trunk Card

The following table outlines the most likely items to cause malfunction. For more information about troubleshooting trunk cards see Trunk Cards - General.

Note: The tip and ring assignments for the LS/CLASS Trunk card differ from the LS/GS card. Please refer to the SX-200 ICP Tip and Ring Assignments table.

Table: LS/CLASS Trunk Card Troubleshooting Summary

Step	Possible Malfunction Source
	Note: The LS/CLASS Trunk card installs into slots 1-8 in a SX-200 rack mount cabinet.
1.	To troubleshoot dropped calls, enable COS option 907 (DATA SMDR - Extended Record) in the COS of the trunk. Ensure that System Option 28 (SMDR Indicator - Long Calls) is disabled.
2.	Faulty connections or reversals between the trunking facilities and the cross connect field.
3.	Faulty connections between the cross connect field and the LS/CLASS Trunk card. Verify that the cable installation is correct. Refer to the Installation Information section.
4.	Problems with programming. Refer to the Program CDE section. - Trunk Circuit Descriptors - Trunk COS and Answer Point COS - Non-Dial In Trunks - Dial In Trunks - Trunk Groups
5.	LS/CLASS Trunk card requires a reset; unplug and reseat.
6.	Problem with trunking facilities (see Note 1) or CLASS not enabled at CO.
7.	SX-200 RM cabinet requires a reset.

Table: LS/CLASS Trunk Card Troubleshooting Summary

Step	Possible Malfunction Source
8.	Faulty LS/CLASS Trunk card.

Notes:

1. If periodic ‘opens’ in loop current are experienced during call progression through a central office, and disconnect timing has been set for a short interval, loop disconnects may cause the trunk to drop. In such cases, increase the timing one step at a time until the calls are no longer dropped.

2. When the system detects a trunk protocol failure, it removes the affected trunk circuit from service. Trunk protocol failures may indicate fault(s) with the trunk circuit, in which case the card should be tested and replaced. However, they may also indicate that the external trunk is faulty. In this case, the affected trunk circuit must stay out-of-service until the external equipment is repaired.

Universal Card

The Universal Card is used for several applications. Refer to procedures for the specific application; i.e.:

- DTMF Receivers / Relays
- Music-on-Hold
- Loudspeaker / Pager
- E&M Trunks (see Trunk Cards - General).

DTMF Receivers / Relay

The DTMF Receiver module is installed on the Universal Card. The module can be used as a DTMF receiver (4 circuits) and/or a relay (2 circuits) - see the Engineering Information section for further information.The following table outlines the most likely causes of DTMF failure.

Table: DTMF Receiver / Relay Troubleshooting Summary

Step	Possible Malfunction Source
1.	Insufficient receiver circuits to handle peak traffic load (see Note 2).
2.	Faulty connection between the cross-connect field and the system or the cross-connect field and the external equipment (applies to the relays only - for Night Bell control or external System Alarm indication). See Note 4.
3.	Faulty external equipment (applies to the relays only - for Night Bell control or external System Alarm indication). See Note 4.
4.	

Faulty CDE programming; likely forms:

- System Configuration
- Miscellaneous System Ports
- Call Rerouting Table.

5.	Faulty or improperly installed DTMF Receiver module.
6.	Faulty Universal Card.
7.	Faulty Universal Card modules (see Note 3).
8.	Faulty connection between Bay Controller and Main Controller.

Notes:

1. Useful maintenance commands: SHOW STATUS, TEST, SHOW ALARMS, SHOW ERRORS - see the RS-232 Maintenance Terminal section.

2. Refer to the Engineering Information section, for details on receiver provisioning.

3. Receiver / relay module malfunction could be caused by the failure of other module(s) on the Universal Card.

4. A simple test for a relay circuit is: (a) disconnect the relay from external equipment at the cross-connect field; (b) connect ohmmeter across relay leads - an open circuit should be read when the relay is open. If not, there is a problem with the module.

Music-on-Hold (MOH)

The Music-on-Hold / Pager module interfaces the system to an external music source for MOH. The module is installed on the Universal Card. See the Engineering Information section for further information. The following table outlines the most likely causes of MOH failure.

Table: Music-on-Hold Troubleshooting Summary

Step	Possible Malfunction Source
1.	Faulty music source.
2.	Faulty connection between the music source and the cross-connect field.
3.	Faulty connection between the cross-connect field and the system.
4.	Faulty CDE programming; likely forms: • System Configuration • Miscellaneous System Ports • COS Define • Call Rerouting Table.
5.	Faulty or improperly installed MOH / Pager module.
6.	Faulty Universal Card.
7.	Faulty Universal Card modules (see Note 2).
8.	Faulty connection between Bay Controller and Main Controller (if applicable).

Notes:

1. Useful maintenance commands: SHOW STATUS, TEST, SHOW ALARMS, SHOW ERRORS - see the RS-232 Maintenance Terminal section.

2. MOH / Pager module malfunction could be caused by the failure of other module(s) on the Universal card.

Pager

The Music-on-Hold / Pager module interfaces the system to an external paging amplifier. The module is installed on the Universal Card. See the Engineering Information section for further information. The following table outlines the most likely causes of paging failure.

Table: Pager Troubleshooting Summary

Step	Possible Malfunction Source
1.	Faulty paging equipment.
2.	Faulty connection between the paging equipment and the cross-connect field.
3.	Faulty connection between the cross-connect field and the system.
4.	Faulty CDE programming; likely forms: • System Configuration • Miscellaneous System Ports • COS Define • Call Rerouting Table.
5.	Faulty or improperly installed MOH / Pager module.
6.	Faulty Universal Card.
7.	Faulty Universal Card modules (see Note 2).
8.	Faulty connection between Bay Controller and Main Controller (if applicable).

Notes:

1. Useful maintenance commands: SHOW STATUS, TEST, SHOW ALARMS, SHOW ERRORS - see the RS-232 Maintenance Terminal section.

2. MOH / Pager module malfunction could be caused by the failure of other module(s) on the Universal Card.

Figure: Music, Paging, Relay and Console Connections

Troubleshoot Sets

Attendant Console - SUPERCONSOLE 1000

The following table outlines the most likely items to cause SUPERCONSOLE 1000 malfunction.

Table: SUPERCONSOLE 1000 Troubleshooting Procedures

Step	Possible Malfunction Source
1.	No connection between the SUPERCONSOLE 1000 attendant console and a DNIC circuit.
2.	Faulty connections between the console and the cross-connect field.
3.	Faulty connections between the cross-connect field and the system.
4.	Faulty CDE programming; likely forms: • COS Define • Console Assignments • Console LDN Assignments • Data Assignment • Data Circuit Descriptor • Tenant Forms 05, 06 • Device Interconnection Table • System Configuration.
5.	Environmental conditions such as static electricity.
6.	Console requires reset (disconnect, reconnect line cord).
7.	Digital Line Card or DNIC Module requires re-initialization. Reseat card or module.
8.	Faulty console, handset and cord assembly, or headset assembly (disconnect headset and test console operation).
9.	Faulty DNIC port.
10.	Faulty backplane cable connections.
11.	Bay Control Card requires reload (if applicable) - power down bay, reseat BCC, power up bay.
12.	The SX-200 ICP requires reset. Press the RESET button on the front panel of the controller. See the Attendant Console Error Indications table.
13.	Peripheral devices attached to console are faulty (headset, printer).

Notes:

1. Useful maintenance commands: SHOW STATUS, TEST, SHOW ALARMS, SHOW ERRORS CONSOLE - see the RS-232 Maintenance Terminal section.
2. Loop resistance should not exceed 200 ohms. Voltage across Tip and Ring should be between 40 and 48 volts dc.
3. Problem may be static related.

Figure: Attendant Console

Table: Attendant Console Error Indications

Indication	Meaning
HOLD 1 indicator flashing (after reset)	Console fault - replace console.
HOLD 2 indicator flashing (after reset)	Console fault - replace console.
HOLD 3 indicator flashing (after reset)	Console fault - replace console.
All indicators on always (after reset)	Console fault - replace console.
message: CONSOLE HARDWARE PROBLEM 123456789 ERROR CODE 1 PLEASE NOTE DETAILS ON REPAIR TAG	Console fault - replace console.
message: WAITING FOR SYNCHRONIZATION 123456789 PLEASE WAIT	• Wiring problem. • Problem with programming. • Problem with Digital Line Card • Problem with console.
message: WAITING FOR COMMUNICATION 123456789 PLEASE WAIT	• Problem with programming. • Problem with console.

SUPERSET 400 Series and SUPERSET 4000 Series Telephones

The following table outlines, in descending order, the most likely items to cause malfunction.

Table: SUPERSET 400 Series and SUPERSET 4000 Series Telephone Troubleshooting Procedures

Item	Possible Malfunction Source
1.	Faulty connections between the set and the cross-connect field.
2.	Faulty connections between the cross-connect field and the system.
3.	Faulty CDE programming; likely forms: • COS Define • Desktop Device Assignments • Device Interconnection Table • System Configuration.
4.	Set requires reset - disconnect, reconnect line cord.
5.	Faulty SUPERSET telephone or handset/cord assembly.
6.	Digital Line Card or DNIC Module requires re-initialization. Reseat card or module.
7.	Bay Control Card requires reset (power down system).
8.	Faulty or wrong Digital Line Card or DNIC Module.
9.	Faulty backplane cable connections.
10.	Problem with DTMF Receivers (not enough for peak traffic load).

Notes:

1. Useful maintenance commands: SHOW STATUS, TEST, SHOW ALARMS, SHOW ERRORS DIGITAL_SETS - see the RS-232 Maintenance Terminal section.
2. A synchronization or communication error with the telephone is indicated by the rapid flashing of the Message Lamp.
3. The set's LCD display will indicate synchronization problems, "NO SYNCHRONIZATION", and communication problems, "NO COMMUNICATION".
4. Loop length limit for devices connected to Digital Line Card or DNIC Module circuits is in the Engineering Information section.
5. Refer to the Program Features section, for details of features and options.

SUPERSET 3DN and SUPERSET 4DN Telephones

The following table outlines the most likely items to cause malfunction.

Table: SUPERSET 3DN and SUPERSET 4DN Telephone Troubleshooting

Step	Possible Malfunction Source
1.	FOR option required for SUPERSET 3DN and 4DN support not enabled.
2.	Faulty connections between the SUPERSET telephone and the cross-connect field.
3.	Faulty connections between the cross-connect field and the SX-200 ICP System.
4.	Faulty CDE programming; likely forms: • COS Define • System Options/System Timers • Desktop Device Assignments • Device Interconnection Table • System Configuration.
5.	Set requires reset - disconnect, reconnect line cord.
6.	Faulty SUPERSET telephone or handset/cord assembly.
7.	Digital Line Card requires re-initialization (reseat card).
8.	Bay Control Card (power down bay).
9.	Faulty Digital Line Card.
10.	Faulty backplane cable connections.
11.	Faulty switching matrices.
12.	Problem with DTMF Receivers (not enough for peak traffic load).

Notes:

1. Useful maintenance commands: SHOW STATUS, TEST, SHOW ALARMS, SHOW ERRORS DIGITAL_SETS - see the RS-232 Maintenance Terminal .
2. Loop length limit is: 3300 ft with 26 AWG wire (twisted pair - with no bridge taps).

SUPERSET Interface Module

Verify the signal connections and the power adapter connections of the SUPERSET interface module according to the Peripheral Devices section.

CAUTION: For a SUPERSET interface module (SIM1 or SIM2) to work correctly with a SUPERSET 4000-series telephone, the SUPERSET 4000-series telephone must be upgraded to the latest firmware using the system maintenance firmware upgrade command.

DATASET 1103 / DATASET 2103 (DTE Mode)

The following table outlines the most likely items to cause malfunction. Note that this table applies to stand-alone DATASET 1103 and DATASET 2103 units interfaced to DTE devices. For applications involving modems (DCE devices), refer to “DATASET 1103 / DATASET 2103 - DCE Mode”.

Table: DATASET 1103 / DATASET 2103 - DTE Mode Troubleshooting Procedures

Step	Possible Malfunction Source
1.	Faulty connections between the data station and the cross-connect field.
2.	Faulty connections between the cross-connect field and the system.
3.	Faulty connections between the DATASET and the connecting device(s) (terminal, telephone).
4.	Faulty CDE programming; likely forms: • COS Define • Desktop Device Assignments • Data Circuit Descriptor • Data Assignment • DTE Profile • Device Interconnection Table • System Configuration.
5.	Incorrect DATASET 2103 switch settings - see the DATASET 2100 Switch Settings (Mode Selection) Table through to the DATASET 2100 Switch Settings (Sync Speed) Table.
6.	Faulty far end (if applicable).
7.	Faulty external equipment (terminal, telephone).
8.	DATASET requires reset - disconnect, reconnect power cord. See the table DATASET 1103 / DATASET 2103 - DCE Mode Troubleshooting Procedures and table DATASET 1103 Indicators .
9.	Bay Control Card (power down system).
10.	Faulty DATASET unit, or power cord assembly (or no AC power).
11.	Digital Line Card or DNIC Module requires re-initialization. Reseat card or module.
12.	Faulty Digital Line Card or DNIC Module.
13.	Faulty backplane cable connections - incorrect tip and ring (tip and ring are black and yellow).

Notes:

1. Useful maintenance commands: SHOW STATUS, TEST, SHOW ALARMS, SHOW ERRORS DATASETS - see the RS-232 Maintenance Commands section.

2. See the table DATASET 1103 Indicators for proper device indicator states.
3. Loop length limit for devices connected to Digital Line Card or DNIC Module circuits is in the Engineering Information section.

Figure: DATASET 1103/2103 Stand-alone DTE Configuration (ADL)

Table: DATASET 2100 Switch Settings (Mode Selection)

Switch No.	Position	Meaning
1	Down Up	Set - PBX operation Hunt - Back to back mode
2	Up Down	Synchronous Operation Asynchronous Operation

Note: The DATASET should be in the SET mode when connected to the system.

Table: DATASET 2100 Switch Settings (Asynchronous Flow Control)

Asynchronous Flow Control Type	Switch No.	
	3	4
Flow control disabled	Down	Down
XON/XOFF flow control	Down	Up
CTS flow control	Up	Down
Flow control disabled	Up	Up

Table: DATASET 2100 Switch Settings (Async Speed)

Asynchronous Speed	Switch No. 5	6	7	8
Autobaud	Down	Down	Down	Down
110	Up	Down	Down	Down
150	Down	Up	Down	Down
200	Up	Up	Down	Down
300	Down	Down	Up	Down
600	Up	Down	Up	Down
1200	Down	Up	Up	Down
2400	Up	Up	Up	Down
4800	Down	Down	Down	Up
9600	Up	Down	Down	Up
19200	Down	Up	Down	Up

Table: DATASET 2100 Switch Settings (Synchronous Clock Source)

Asynchronous Clock Source	Switch No.	
	3	4
Internal clock	Down	Down
System clock	Down	Up
Tx external clock	Up	Down
Tx and Rx external clock	Up	Up

Table: DATASET 2100 Switch Settings (Sync Operating Mode Selection)

Switch No.	Position	Meaning
5	Down	Transparent mode
	Up	X.31 mode

Table: DATASET 2100 Switch Settings (Sync Speed)

Synchronous Speed	Switch No.		
	6	7	8
1200	Down	Down	Down
2400	Up	Down	Down
4800	Down	Up	Down
9600	Up	Up	Down
19200	x	x	Up

Note: "x" indicates up or down is acceptable.

Figure: DATASET 2103 Indicators and Connectors

Table: DATASET 2103 LED Indicators

Indicator	Meaning
DEVICE	
ON -	Indicates that the attached device is connected to the Dataset, and is supplying DSR or DTR on pin 20 of the RS-232 connector.
FLASHING -	Indicates that the Dataset is transmitting data
OFF -	Indicates that the device is not supplying DTR or DSR, or is not connected.
READY	

ON - Indicates that the Dataset is involved in an active call.

FLASHING - Indicates that the Dataset is receiving data.

OFF - Indicates that the Dataset (and the line card) are in the idle state.

ASYNC

ON - Indicates that the Dataset is operating in asynchronous mode.

OFF - Indicates that the Dataset is operating in synchronous mode.

FLASHING - Indicates that the Dataset is in an illegal connection with a DATASET 2100 in sync mode, or that the Dataset is in sync with something other than a DATASET 1103 or DATASET 2103.

POWER

ON - Indicates that the Dataset has power, and is in sync with the corresponding Digital Line Card or DNIC Module.

FLASHING - Indicates that the Dataset has power, but is not in sync with the corresponding Digital Line Card or DNIC Module.

OFF - Indicates that the Dataset is not receiving power from the 9 Vac plug-in transformer.

Note: See the Standalone DATASET 2103 - Showing Connectors and Indicators Figure .

Figure: DATASET 1103 Indicators and Connectors

Table: DATASET 1103 Indicators

Indicator	Meaning
DEVICE	
ON	- Indicates that the attached device is connected to the Dataset unit, and is supplying DSR or DTR on pin 21 of the RS-232 connector.
FLASHING	- Indicates that the Dataset is transmitting data.
OFF	- Indicates that the device is not supplying DTR or DSR, or is not connected.
READY	
ON	- Indicates that the Dataset is involved in an active call.

FLASHING -	Indicates that the Dataset is receiving data.
OFF -	Indicates that the Dataset (and the line card) are in the idle state.
POWER	
ON -	Indicates that the Dataset has power, and is in sync with the corresponding Digital Line Card or DNIC Module.
FLASHING -	Indicates that the Dataset has power, but is not in sync with the corresponding Digital Line Card or DNIC Module.
OFF -	Indicates that the Dataset is not receiving power from the 9 Vac plug-in transformer.

Note: See the DATASET 1103 Standalone,Connectors and Indicators Figure .

DATASET 1103 / 2103 (DCE Mode)

The following table outlines the most likely items to cause malfunction. Note that this table applies to stand-alone DATASET 1103 and DATASET 2103 units interfaced to DCE devices. For applications involving DTE devices, refer to “DATASET 1103 / DATASET 2103 - DTE Mode”.

Table: DATASET 1103 / DATASET 2103 - DCE Mode Troubleshooting Procedures

Step	Possible Malfunction Source
1.	Faulty connections between the data station and the cross-connect field.
2.	Faulty connections between the cross-connect field and the system.
3.	Faulty connections between the DATASET and the connecting device(s) (modem).
4.	Faulty CDE programming; likely forms: • COS Define • Desktop Device Assignments • Data Circuit Descriptor • Data Assignment • DTE Profile • Modem Assignment • Device Interconnection Table • System Configuration.
5.	Incorrect DATASET 2103 switch settings - see Table DATASET 2100 Switch Settings (Mode Selection) through Table DATASET 2100 Switch Settings (Sync Speed).
6.	Faulty far end (if applicable).

7.	Incorrect modem switch settings or modem software set-up characteristics (if applicable).
8.	Modem requires reset - refer to manufacturer's instructions.
9.	Faulty external equipment (modem).
10.	DATASET requires reset - disconnect, reconnect power cord. See the Table DATASET 2103 LED Indicators and Table DATASET 1103 Indicators.
11.	Bay Control Card (power down system).
12.	Faulty DATASET unit or power cord assembly (or no AC power).
13.	Digital Line Card or DNIC Module requires re-initialization. Reseat card or module.
14.	Faulty Digital Line Card or DNIC Module.
15.	Faulty backplane cable connections.
16.	Route compression is enabled on the IP trunk route in Form 23, Subform Show IP. Modem calls routed over IP trunks with compression enabled will always fail to negotiate with far end.

Notes:

1. Useful maintenance commands: SHOW STATUS, TEST, SHOW ALARMS, SHOW ERRORS DATASETS - see the RS-232 Maintenance Terminal section.

2. See the table DATASET 1103 Indicators for proper device indicator states.

3. Loop length limit for devices connected to Digital Line Card or DNIC Module circuits is in the Engineering Information section.

Figure: DATASET 1103 / DATASET 2103 Modem Configuration

Troubleshoot Subsystem

Peripheral Bay Troubleshooting Procedure

The following paragraphs detail the troubleshooting procedures for a Peripheral Bay that does not initialize.

Table: Peripheral Bay Troubleshooting Summary

Step	Possible Malfunction Source for Peripheral Bay Failure to Initialize
1	FOR option required for Peripheral Bay support not enabled.
2.	Software load does not support a second or subsequent bay.
3.	If FIM or CIM LEDs are flashing, reverse the cable connectors at one end only.
4.	Faulty CIM, Control cabinet FIM Carrier, or FIM.

Table: Peripheral Bay Troubleshooting Summary

Step	Possible Malfunction Source for Peripheral Bay Failure to Initialize
5.	Faulty Peripheral cabinet FIM Carrier, FIM, or CIM.
6.	Faulty Bay Control Card.

Digital Bay Control Subsystem

The following paragraphs detail the troubleshooting procedures for the Digital Bay Control subsystem.

Configuration

The Digital Bay Control subsystem consists of one card - the Bay Control Card II or the Bay Control Card III.

Bay Control Power-Up Tests

The Bay Control Card power-up tests are run automatically upon power-up of the associated bay, or a reset of the SX-200 ICP. Failure of any of the power-up tests will result in the flashing of the Bay Control Card ALARM LED soon after initialization.

First-Step Checks

Prior to replacing cards as directed by the Digital Bay Control subsystem troubleshooting procedures, carry out the following checks:

- At the maintenance terminal, SHOW ERRORS and check the DX channel links; also check for HDLC errors. Refer to the RS-232 Maintenance Terminal section, and the General Maintenance Information section.
- Reseat the suspect card and modules as appropriate.
- Check for bent pins at the backplane or module connector, as applicable.

If the above does not clear the fault, replace the modules and card.

Power-Up Sequence

The Bay Control Card power-up sequence is described in the General Maintenance Information section. The following table, Digital Bay Control Troubleshooting Summary, summarizes the troubleshooting procedures for the Bay Control Card.

Troubleshooting Procedure

The following table outlines the most likely causes of Bay Control failure.

Table: Digital Bay Control Troubleshooting Summary

Step	Possible Malfunction Source for Bay Control Failure
1.	Bay Control Card requires a reset - power down bay, reseat card, and power up bay.
2.	Note the indicators on the Bay Control Card (see the Bay Control Card II Status LEDs table or the

Table: Digital Bay Control Troubleshooting Summary

Step	Possible Malfunction Source for Bay Control Failure
	Bay Control Card III Status LEDs table.
3.	Incorrect switch settings on the BCC II (all switches (SW1-1, SW1-2, SW2-1, SW2-2) should be in the closed position). See this Figure.
4.	Faulty Bay Power Supply unit - refer to that procedure.
5.	Faulty Bay Control Card.
6.	Faulty SX-200 ICP controller.
7.	Faulty FIM Carrier, FIM, or CIM.

Note: Useful maintenance commands: SHOW STATUS, TEST, SHOW ALARMS - see the RS-232 Maintenance Terminal section.

Figure: Bay Control Card Device Locations

Table: Bay Control Card II Status LEDs

TX LED	RX LED	ALARM LED	Meaning
on	on	on	Bay Control Card is either waiting for, or has lost communication with the SX-200 ICP. If this state persists for more than a few seconds, there is no communication.
flashing	flashing	on	Bay Control Card is being downloaded by the SX-200 ICP.
off / flashing	off / flashing	off	Bay Control Card is up and running and communicating to the SX-200 ICP.
-	-	flashing	There is a failure on the Bay Control Card.

Table: Bay Control Card III Status LEDs

TX/RX LED	ETHERNET LED	ALARM LED	Meaning
on			

not applicable

See Note.	on

BCC III is either waiting for, or has lost communication with the SX-200 ICP. If this state persists for more than a few seconds, there is no communication.

The Ethernet LED and the Ethernet jack is not used by the technician in the field; only manufacturing use these.			
flashing	not applicable	on	BCC III is being downloaded by the SX-200 ICP.
off / flashing	not applicable	off	BCC III is up and running and communicating to the SX-200 ICP.
-	not applicable	flashing	There is a failure on the BCC III.

Note: The Ethernet LED and the Ethernet jack is not used by the technician in the field; only manufacturing use these. A solid green light indicates heavy Ethernet traffic, a flashing green light indicates moderate Ethernet traffic, and absence of a light represents no Ethernet activity (the present normal condition).

Power Subsystem

The power subsystem of each bay consists of one Bay Power Supply consists of a power module. The uninterruptible power supply (UPS) is optional, and is user-supplied. This Figure shows a Bay Power Supply. The following table outlines the required Bay Power Supply voltages.

Figure: Bay Power Supply and Fuse Locations

Bay Power Supply Test Point Voltages

Table: Bay Power Supply Test Point Voltages

Voltage	Minimum	Maximum
+ 5 Vdc	+ 5.07	+ 5.23
+ 12 Vdc	+ 10.8	+ 13.2
- 12 Vdc	- 13.2	- 10.8
- 5 Vdc	- 5.5	- 4.5
- 28 Vdc	- 30.8	- 23.8
-48 Vdc	- 53.76	- 40.8
90 Vac	63.0	99 .0

Uninterruptible Power Supply (UPS)

Any UPS may be used with the PBX, provided that it meets the requirements specified in the Engineering Information section. Since these are available from a number of suppliers, no troubleshooting procedures for the UPS are provided in this document. Reference should be made to the appropriate manual provided by the manufacturer of the UPS for any self-diagnostic capabilities.

Troubleshoot Peripherals

Maintenance Terminal

The device used as a maintenance terminal must satisfy the following requirements:

- Compatibility with RS-232C type interface protocol if connecting directly to the maintenance port on the controller. (Not required if using secure Telnet or the SX-200 Web Interface to access Maintenance functions.)
- 80 columns
- Compatibility with ANSI X3.64-1977 special character set for special graphics.

These procedures deal with the interface only. Reference should be made to the appropriate manual provided by the manufacturer of the terminal for any problems with the terminal itself. The following table outlines the most likely items to cause malfunction.

Table: Maintenance Terminal Troubleshooting Procedure

Step	Possible Malfunction Source
1.	Keyboard locked - reset terminal.
2.	Blown terminal fuse (if applicable).
3.	Inconsistent communication parameters between terminal and port; default values are • 8 data bits • 1 stop bit • No parity • ASCII character set • Xon/Xoff flow control.
4.	Terminal in LOCAL mode - put into ON LINE mode - see manufacturer's instructions.
5.	Terminal requires reset (or X-ON, or CONTROL-S - see manufacturer's instructions).
6.	Faulty connection between the terminal and its power source.
7.	Faulty terminal.
8.	Main Control requires reload - press *RESET* on the SX-200 ICP front panel.
9.	Faulty internal circuit board.
10.	Faulty interface between front rear panel and internal circuit board.
11.	A terminal is already logged into the CDE or maintenance application at another location.

12	Power up and power down Bay Power Supply.

Notes:

1. Refer to the RS-232 Maintenance Terminal section, for details on setting communication parameters.

2. The system does not have a maintenance panel. The maintenance port is accessed at the rear of the cabinet.

3. To connect the terminal's 25 pin RS-232 connector to the system's 9 pin mini D connector requires the use of a 9 pin to 25 pin RS-232 adapter. This is a standard off-the-shelf part found in most computer supply stores.

System Printers

System printers must satisfy the following:

1. Compatibility with RS-232C type interface protocol (and support pins 2, 3, 4, 5, and 7).
2. Width of 80 columns.

Printers can be installed with the SX-200 system in different configurations:

1. Connected to the Cabinet's 9 pin PRINTER port.
2. Connected to the SUPERCONSOLE 1000 console port.
3. Connected to the MILINK module.

Cabinet Printer Port

These procedures deal with the interface only. Reference should be made to the appropriate manual provided by the manufacturer of the printer for any problems with the printer itself. the following table outlines the most likely items to cause malfunction.

Table: Cabinet Printer Port Troubleshooting Procedures

Step	Possible Malfunction Source
1.	Blown printer fuse (if applicable).
2.	

Inconsistent communication parameters between printer and port; PBX default values are:

• 8 data bits • 1 stop bit • no parity • ASCII character set • Xon/Xoff flow control • 300 baud (baud setting may be adjusted in Maintenance from 300 to 9600 baud)	
3.	Faulty connection between the printer and its power source.
4.	Faulty CDE programming; likely form: Form 12 - Directed I/O.
5.	Faulty connection between the printer's communication port and the Cabinet printer port.

6.	Printer in LOCAL mode - put into ON LINE mode - see manufacturer's instructions.
7.	Printer requires reset (or X-ON, or CONTROL-Q).
8.	Faulty printer.
9.	Main Control requires reload - press RESET on the SX-200 ICP front panel.
10.	Faulty SX-200 ICP controller.
11.	Faulty interface between I/O panel and internal circuit board.

Notes:

1. Useful maintenance commands: LOGS PRINT, SUSPEND_PRTR, RESUME PRTR, SHOW DEVICE, SET SPEED, TEST DEVICE_TYPE MORE_KEYS PRINTER_PLID / PORT - see the RS-232 Maintenance Terminal section.

2. To connect the printer's 25 pin RS-232 connector to the system's 9 pin mini D connector requires the use of a 9 pin to 25 pin RS-232 adapter. This is a standard off-the-shelf part found in most computer supply stores

SUPERCONSOLE 1000 Printer Port

These procedures deal with the interface only. Reference should be made to the appropriate manual provided by the manufacturer of the printer for any problems with the printer itself. The following table outlines the most likely items to cause malfunction.

Table: SUPERCONSOLE 1000 Printer Port Troubleshooting Procedures

Step	Possible Malfunction Source
1.	Blown printer fuse (if applicable).
2.	

Inconsistent communication parameters (both the SUPERCONSOLE 1000 printer port and the printer should have exactly the same parameters.

Note: 2400 is the maximum baud rate for this port).	
3.	Faulty connection between the printer and its power source.
4.	Faulty connection between the printer's communication port and the SUPERCONSOLE 1000 port.
5.	

Faulty CDE programming; likely forms:

• Directed IO • Console Assignment • Data Circuit Descriptor • Data Assignment • COS Define • System Configuration.	
6.	Printer in LOCAL mode - put into ON LINE mode - see manufacturer's instructions.
7.	Printer requires reset (or X-ON, or CONTROL-Q if applicable - see manufacturer's instructions).
8.	Faulty printer.
9.	Suspect SUPERCONSOLE 1000 unit - refer to that procedure.
10.	Reset console or reseat Digital Line Card or DNIC Module.

Note: Useful maintenance commands: LOGS PRINT, SUSPEND_PRTR, RESUME PRTR, SHOW DEVICE, SET SPEED, TEST DEVICE_TYPE MORE_KEYS PRINTER_PLID / PORT - see the RS-232 Maintenance Terminal section.

System Fail Transfer

Note that these procedures are intended to deal with failure of the SFT system, not the activation of it.

A System Fail Transfer connector (an RJ11 modular jack) is located on the backplane. This connector is used to interface SFT equipment provided by external suppliers.

Table: System Fail Transfer Connector Troubleshooting Procedures

Step	Possible Malfunction Source
1.	External SFT equipment's bypass switch set incorrectly; refer to manufacturer's instructions for external SFT unit.
2.	Faulty SFT connections - refer to manufacturer's instructions for external SFT unit.
3.	Faulty connections between internal circuit board and the system's SFT connector assembly - see the System Fail Transfer (SFT) Connector Pin-Outs table.
4.	Faulty connections between the SFT connector and external SFT unit.
5.	Faulty connections at the cross-connect field - refer to manufacturer's instructions for external SFT unit.
6.	Faulty external SFT unit.
7.	Faulty power system - refer to Power Subsystem Troubleshooting procedure (if the -48 volt supply is not working).

Note: Relay contact is open during normal system operation. It is closed to indicate SFT.

Table: System Fail Transfer (SFT) Connector Pin-Outs			
Pin No.	**Wire Color**	**Signal**	**Comment**
2	Yellow	-48	Voltage source for the external SFT transfer equipment. Note, this is limited to 250 mA.
3	Green	-48 return (GND)	Ground reference (return) for the external SFT transfer equipment
4	Red	SFT Relay (A)	First side of “normally closed” SFT control relay
5	Black	SFT Relay return (B)	Second side of “normally closed” SFT control relay

Recorded Announcement Devices (RADs)

Reference should be made to the appropriate manual provided by the manufacturer of the RAD unit for any problems with the RAD unit itself. The following table outlines the most likely items to cause RAD malfunction

Table: RAD Troubleshooting Procedures	
Step	**Possible Malfunction Source**
1.	Faulty RAD. (See Note 2.)
2.	Faulty connection between the RAD and the cross-connect field.
3.	Faulty connection between the cross-connect field and the system.
4.	Faulty CDE programming; likely forms: (See Note 3.) • COS Define • Hunt Groups • Desktop Device Assignments • System Configuration.
5.	Faulty ONS port.
6.	Check items listed in the following Supplementary RAD Troubleshooting Procedures table.

Notes:

1. Useful maintenance commands: SHOW STATUS, SHOW DEVICE, TEST DEVICE_TYPE ONS, SHOW STATUS SWID SW_HUNT_GRP <swid> CP_DWA (check stagger and o_msglen), TRAFFIC_MEAS READ (check hunt group busy peg) - see the RS-232 Maintenance Terminal section.

2. Refer to the manufacturer’s instructions to repair and troubleshoot a RAD unit. For cassette type units, tape wearout or dirty heads / dirty tape / dirty capstan are the most common failures.

3. Avoid using discriminating ringing in the RAD COS; some operate only with standard ringing.

Table: Supplementary RAD Troubleshooting Procedures

Problem	Possible Cause(s)
No recordings at all.	RADs are in busy-out or DND status - use SHOW STATUS command on the corresponding ONS circuit.
RAD message is cut off.	Check that the Message Length Timer in the Hunt Groups CDE form is long enough.
RADs go into DND state.	Check if Failure To Hang Up Timer (COS 404) is long enough for proper RAD clear down.
No message heard, but RAD rings and answers.	Check that the Message Length Timer in the Hunt Groups CDE form is long enough.

Note: For further information, refer to RAD Support.

Night Bell Equipment

Night Bells are connected in 2 basic configurations:

Direct Connect Method: Night bells can be connected directly if the total current requirement does not exceed the relay contact ratings (see this Figure).

Auxiliary Relay Method: Night bells must be connected through an auxiliary relay if the total current requirement exceeds the relay contact ratings (see this Figure).

The DTMF Receiver / Relay module is installed on the Universal Card; it can be used as a night bell relay (2 circuits) - see the Engineering Information section for further information. The following table outline the most likely causes of night bell failure.

Table: Night Bell Troubleshooting Procedures

Step	Possible Malfunction Source
1.	Faulty night bell device.
2.	Faulty external relay (auxiliary relay configuration only).
3.	Faulty external relay connections (see Note 2).
4.	Faulty connection between the night bell device and the cross-connect field.
5.	Faulty connection between the cross-connect field and the system.
6.	

Faulty CDE programming; likely forms:

- System Configuration
- Miscellaneous System Ports
- Call Rerouting Table.

7.	Faulty or improperly installed DTMF Receiver module on Universal Card.
8.	Faulty Universal Card.
9.	Faulty Universal Card modules (see Note 3).

Notes:

1. Useful maintenance commands: SHOW STATUS, TEST, SHOW ALARMS, SHOW ERRORS - see the RS-232 Maintenance Commands section.

2. Refer to the Installation Information section, for details on the installation of night bells using an auxiliary relay.

3. Receiver / relay module malfunction could be caused by the failure of other module(s) on the Universal Card.

4. A simple test for a relay circuit is: (a) disconnect the relay from external equipment at the MDF; (b) connect ohmmeter across relay leads - an open circuit should be read. If not, there is a problem with the module.

Figure: Music, Paging, Relay and Console Connections

Figure: Night Bell Auxiliary Relay Configuration

Music-on-Hold/Pager Unit Troubleshooting

This section covers the Music-on-Hold/Pager Unit (PN 9401-000-010-NA). The following table outlines the most likely causes of malfunction.

Table: Music-on-Hold/Pager Unit Troubleshooting Summary

Step	Possible Malfunction Source
1.	Faulty connections between the PBX and the unit. Make sure that the status LED is either ON steadily (initialization complete) or winking (paging amplifier being used). Refer to the Installation Information section for connector pin outs.
2.	

Problems with programming. Refer to the Program CDE section. The forms to look at are:

- Form 1 System Configuration.
- Form 9 Desktop Device Assignments
- Form 18 Miscellaneous System Ports.
- Form 19 Call Rerouting Table.

3.	SX-200 bay requires a reset, unplug its power cord and plug it back in.
4.	

Faulty external equipment or connections which can include:

• External relays. • Night Bells and ringing voltage sources. • External Music Source. • External paging amplifier.	
5.	Faulty Music-On-Hold/Pager unit.
6.	Faulty Digital Line Card or DNIC Module.
7.	Faulty Control Module II or Bay Control Card.
8.	Other faulty PBX component.

Hotel/Motel Interface

The Hotel / Motel features can be accessed by 2 major interfaces - the attendant console, or the front desk interface. Some features can be accessed via SUPERSET telephones as well. Note that this section covers only the Hotel/Motel feature. The Property Management System (PMS) Interface feature is covered in the next paragraph.

See the Hotel/Motel Feature Package Description section for further information on Hotel/Motel features. The following table outlines the most likely causes of Hotel/Motel feature failure.

Table: Hotel/Motel Troubleshooting Procedures

Step	Possible Malfunction Source
1.	Faulty front desk terminal device.
2.	Blown terminal fuse (if applicable).
3.	Faulty communication parameters - terminal and RS232-to-IP serial port converter (if using) must have the same parameters.
4.	Faulty connection between the terminal and its power source.
5.	Faulty connection between the terminal's communication port and the Dataset port.
6.	Faulty connection between the Dataset and the cross-connect field.
7.	Faulty connection between the cross-connect field and the system.
8.	

Faulty CDE programming; likely forms:

• System Configuration • COS Define • all Dataset and/or console forms - refer to the appropriate procedure.	
9.	Suspect Dataset - refer to DATASET 1103/2103 - DTE MODE.

10.	Suspect Digital Line Card or DNIC Module - refer to DATASET 1103/2103 - DTE MODE.
11.	Suspect console - refer to the appropriate procedure.

Notes:

1. Useful maintenance commands: SHOW STATUS, TEST, SHOW ALARMS, SHOW ERRORS - see the RS-232 Maintenance Commands section.
2. Loop length limit for devices connected to Digital Line Card or DNIC Module circuits is in the Engineering Information section.

Property Management System Interface

The following table, PMS Troubleshooting Procedures, outlines the most likely items to cause PMS malfunction.

Table: PMS Troubleshooting Procedures

Step	Possible Malfunction Source
1.	Faulty PMS interface system.
2.	Faulty connection between the external equipment devices (if applicable) - see this Figure.
3.	Faulty connection between the PMS Interface system and the third-party RS232-to-IP serial port converter (if using)
4.	Faulty connection between Layer 2 switch and the system
5.	Faulty CDE programming; likely forms: • Hotel Options Assignment • Lodging or System Options • COS Define • Data Circuit Descriptors • Data Assignment • Console Assignment • System Configuration.
6.	Dataset / Digital Line Card or DNIC Module requires reset - refer to DATASET 1103/2103 - DTE MODE.
7.	Faulty Dataset / Digital Line Card or DNIC Module.

Note: Useful maintenance commands: SHOW STATUS, TEST, SHOW ALARMS, SHOW ERRORS DATASETS - see the RS-232 Maintenance Terminal section.

Figure: PMS Configuration (Typical)

Voice mail Interface

ONS Port Voice mail

The following table outlines the most likely items to cause ONS Voice mail malfunction.

Table: ONS Voice mail Troubleshooting Procedures

Step	Possible Malfunction Source
1.	Faulty external equipment (voice mail system) - see manufacturer's instructions.
2.	Faulty connection between the voice mail device and the cross-connect field.
3.	Faulty connection between the cross-connect field and the system.
4.	Faulty CDE programming; likely forms: • COS Define • Desktop Device Assignments • Hunt Groups • System Abbreviated Dial Entry • Feature Access Codes • System Configuration.
5.	Faulty Digital Line Card.

Notes:

1. Useful maintenance commands: SHOW STATUS, TEST, SHOW ALARMS - see the RS-232 Maintenance Terminal section.
2. Loop length limit is 1400 ft with 26 AWG wire (with no bridge taps).

DNIC Port Voice mail

The following table, DNIC Voice mail Troubleshooting Procedures outlines the most likely items to cause DNIC Voice mail malfunction.

Table: DNIC Voice mail Troubleshooting Procedures

Step	Possible Malfunction Source
1.	Faulty external equipment (voice mail system) - see manufacturer's instructions.
2.	Faulty connection between the voice mail device and the cross-connect field.
3.	Faulty connection between the cross-connect field and the system.
4.	Faulty CDE programming; likely forms:

• COS Define • Desktop Device Assignments • Hunt Groups • System Abbreviated Dial Entry • Feature Access Codes • System Configuration.	
5.	Faulty DNIC Card.

Notes:

1. Useful maintenance commands: SHOW STATUS, TEST, SHOW ALARMS - see the RS-232 Maintenance Terminal section.
2. Loop length limit is 1400 ft with 26 AWG wire (with no bridge taps).

Troubleshoot Software Options

These procedures are intended for the sole purpose of troubleshooting problems with programming and setup of applications. Fully functional hardware is assumed in all cases.

Key System Features

Table: Key System Troubleshooting

Problem	Possible Cause/Solution
KEY SYSTEM PAGING	
You dial the Direct Paging feature access code and get reorder tone and the NO ACCESS message, or rerouted to the illegal number intercept.	• Verify that the set is programmed in Form 09.
ALL SET PAGE	
After dialing the access code or pressing the Feature key you dial the * sign (end of dial character). You then get NO ACCESS and reorder tone, or are rerouted to the illegal number intercept.	Verify that All Set Page is enabled.
After dialing the access code or pressing the Feature key you dial the * sign (end of dial character). You then get 1234 BUSY and busy tone.	

Possibly due to one of the following:

• There are no members totally idle (idle or hfi) • No channels (system resource) available Try the all set page later.

GROUP PAGE

After dialing the access code or pressing the Feature key you dial the # sign (end of dial character). You then get NO ACCESS and reorder tone, or are rerouted to the illegal number intercept.	• Verify that a page group has been assigned.
After dialing the access code or pressing the Feature key you dial the # sign (end of dial character). You then get 1234 BUSY and busy tone.	Possibly due to one of the following: • There is currently a group page in progress for your page group • There are no members totally idle (idle or hfi) • An all zone page is in progress • No channels (system resource) available Try the group page later.

DIRECTED PAGE

After dialing the access code or pressing the Feature key you dial a destination. You then get NO ACCESS and reorder tone (or are rerouted to the illegal number intercept point).	• Verify that the destination is a valid set extension number. • Check if the set is restricted from accessing the destination through tenanting - dial the destination directly and see whether you get NO ACCESS again. If so, this is NOT a problem with paging but rather the set is barred from ever calling this destination.
After dialing the access code or pressing the Feature key you dial a destination. You then get xxxx BUSY and busy tone at this point.	• The destination is not idle or hands free idle. • The set's prime line is in use by another set. • No channels (system resource) available.

ALL SET PAGE OR GROUP PAGE MEET ME ANSWER

Dial the Group Page - Meet Me Answer access code and you get reorder tone and NO ACCESS on the display, or you get rerouted to the illegal number intercept.	• Verify that the set is programmed in Form 09. • Verify that the set has a page group assigned.

You get NO PAGE ACTIVE on a display set and reorder tone.	• The last page for the group was done more than 15 minutes ago (timed out). • The last telephone to page is no longer programmed in the data base. • There has not been a group page for the group since the last system reset.
STORE PERSONAL SPEED CALL	
You dial the access code and get NO ACCESS and REORDER tone.	• The set is not programmed in Form 09.
You dial the access code followed by an index number and get NO ACCESS and REORDER tone, or you get rerouted to the illegal number intercept.	• The index number dialed was not between 1 and 5.
You dial the access code followed by an index number and get INVALID # and REORDER tone.	• The index number was dialed after the interdigit timer expired.
You dial the access code followed by a VALID index and then digits and you get INVALID # and reorder tone.	• The index number was dialed after the interdigit timer expired. • Invalid number combination dialed: - * must be followed by 3, 5 or * - *3 must be followed by 01 to 14 - *5 must not be followed by further digits.
RETRIEVE PERSONAL SPEED CALL	
You dial the access code and get NO ACCESS and REORDER tone.	• Verify that the set is programmed in Form 09.
You dial the access code followed by an index number and get NO ACCESS and REORDER tone, or you get rerouted to the illegal number intercept.	• The index number dialed was not between 1 and 5.
You dial the access code, followed by a valid index number, and get INVALID # and reorder tone.	• The interdigit timer expired. • Verify that the digits stored translate to a valid destination. • The speedcall number translation looped too many (more than 4) times within speedcall translation. (The speedcall translated into another speedcall which in turn translated into another speedcall, etc - and this occurred more than 4 times).
CO LINE KEY	

You get NO ACCESS and REORDER tone when the CO line key is pressed.	• The CO trunk associated with the key is not a member of a trunk group. • The set has hotel/motel feature enabled and the user is disallowed to make external calls.
You get INVALID # and REORDER tone after dialing a digit or several digits on the CO line.	• The user is restricted to dial this specific number based on the user's COR (check the COR restriction on the dialed number in Forms 20 and 46). • The user is required to dial account code prior to dialing. • The user is restricted from dialing this specific number on this CO Line (check digit string entries in Form 46).
You get NO ACCESS and REORDER tone after dialing a few digits.	• The user has exceeded the maximum number of digits that can be dialed based on the user's COR (check Form 27).
Press a CO line key and the key selection is ignored.	• The user is barred from making calls on the CO line based on the Device/Tenant Interconnection table (check Forms 5 and 30).
LINE PREFERENCE	
You get SELECT A LINE after going offhook.	• The line preference is MANUAL. You are forced to select a line to originate calls. • The line preference is a CO Line Group Key and the Prime Line is not available to make the call. • The line preference is not manual but the line is in use.
DIRECT CO LINE SELECT	
You get NO ACCESS and REORDER tone after dialing the Direct CO Line Select access code.	• The COS option "Key System - Direct CO Access" is disabled.
You get INVALID # and REORDER tone after dialing the Direct CO Line Select access code and a trunk number.	• The trunk number dialed is not a CO trunk. • The trunk number dialed is a DTS or Private trunk. • The trunk number dialed is not a programmed trunk. • The trunk number dialed is out of range (valid range is 1 - 200). • The feature was accessed via speedcall, and the speedcall digit string contained a trunk number of greater than 3 digits in length.
PROGRAM FEATURE KEY	

You get REORDER tone on a non-display set after the Program Feature Key access code is dialed.	• Another user is currently accessing the subform of Form 9 of this set. • The feature has been accessed via speed call, and the speedcall digit string contains more digits than expected (the speedcall string cannot have any digits following the Program Feature Key access code).
You get REORDER tone after you dialed the Program Feature Key access code and a line key is pressed.	• The line key selected is not a speedcall, DSS or feature key.
You get REORDER tone was heard after a a 2-digit feature code was dialed.	• The feature selected is already programmed on another key (check CDE).• User did not have proper COS for the selected Feature Key. • The feature code is out of range.
On a display set, the CHANGE or CLEAR softkey is not shown.	• Another user is currently accessing the sub-form of Form 9 of this set.
The feature you want to program does not appear on a display set.	• User does not have proper COS option enabled for the selected Feature Key. • The feature is already programmed on another key.

Hotel/Motel Feature Package

It is assumed that the system has been properly set up to allow data calls using the DTRX facility. Please note that in most cases if a System or COS option is enabled or disabled in CDE while the front desk terminal is displaying the applicable field, the terminal will not be updated until the display is re-drawn. This can be accomplished by typing control-r or control-w while in the screen or by exiting from the screen and re-entering the screen

Table: Hotel/Motel Troubleshooting

Problem	Possible Cause/Solution
Entering MONITOR HM results in “Facility Not Available” message being displayed.	• Enable System option 33 “Room Status ”.
On entry to House Statistics screen ALL data fields are displayed as dashes (“- ”).	• COS option 244 “Room Status Applies” must be enabled in all COS’ that have guest room phones. After enabling, press the REFRESH softkey.
No guest rooms can be found search or Audits mode.	• COS option 244 “Room Status Applies” must be enabled in all COS’ that have guest room phones. After enabling, press the REFRESH softkey.

Table: Hotel/Motel Troubleshooting

Problem	Possible Cause/Solution
Data displayed on House Statistics screen is wrong: too few guest rooms	• Some guest room phones are not programmed (Form 9) or are in a COS that does not have “Room Status Applies” enabled.
Pressing the YES softkey at the “Save changes to guest room before quitting?” prompt does not save the changes.	• Names will not be stored for vacant guest rooms - check the occupancy setting.
Data displayed on House Statistics screen is wrong: too many guest rooms	• Non - guest room phones are in a COS that has “Room Status Applies” enabled.
“Wake up Set” text field on House Stats screen is not displayed.	• Enable System option 11 “Automatic Wake-up” and redraw the screen.
“Call Blocking” text field on House Stats screen is not displayed.	• Enable System option 9 “Attendant Call Block” and redraw the screen.
MSG REG AUDIT softkey not provided.	• Enable System option 24 “Message Register Audit”.
ROOM STATUS AUDIT softkey not provided.	• Enable System option 27 “Room Status Audit”.
WAKE UP AUDIT softkey not provided.	• Enable System option 13 “Automatic wake up print”.
Unable to find any rooms of a particular TYPE in room search mode (or audit mode).	• The COS associated with the guest room phone(s) in question does not have “Room Status Applies” enabled.
Room Type field is displayed as a number instead of a NAME.	• COS for room type shown does not have name field programmed.
SET MESSAGE softkey is not displayed in Room Updates Form for non-display set room phones.	• COS for the guest room phone does not have COS option 231 or 232 (mes. waiting - BELL OR LAMP) enabled.
The SET MESSAGE softkey is not displayed for any guest room telephones.	• DIAL 0 routing in Form 19 MUST be set to a console or console LDN for all settings (DAY, N1, N2).
Room has a message but it can’t be cleared by the front desk.	• Message is not from a console or console LDN (i.e., set to set message).
WAKEUP TIME text is not displayed in Room Updates form.	• Enable System option 11.

Table: Hotel/Motel Troubleshooting

Problem	Possible Cause/Solution
MESSAGE REGISTER text is not displayed in Room Updates form.	• Enable COS option 703 in guest room class of service.
DO NOT DISTURB text not displayed in Room Updates form.	• Enable COS option 220 in guest room class of service.
No ENTER key is provided after a name has been programmed in Room Updates form.	• Change the room occupancy to something other than vacant.
Unable to find guest names.	• Search string must be the starting letters of the last name of a guest room.
Non - guest names found when searching by name.	• Non guest room phones are in a COS with “Room Status Applies” enabled.
Unable to access Room Updates form.	• Logged in using the CDE attendant level password. Must logout and log back in using the CDE supervisor password.
Attendant level password allows access to Room Updates form.	• The CDE attendant and Supervisor passwords are identical. Change passwords in CDE Form 28 so they are different.

Subattendant (Enhanced Function)

This procedure deals with the Subattendant - Enhanced Function feature. It does not apply to the Subattendant - Basic Function feature.

Table: Enhanced Function Subattendant Troubleshooting

Problem	Possible Cause/Solution
PROGRAMMING: GENERAL	
The SUB-ATT softkey does nothing when pressed.	• The maximum number of enhanced-function subattendants has already been programmed.
PROGRAMMING: LINE KEYS	
The LDN softkey does not appear in the expand set form.	

- There are already the maximum number of LDN keys (6) on the set.

• The LDN key is not programmable on a PKM Module associated with a subattendant station.

Cannot alter LDN key information.	• An LDN may not be altered when it is engaged in an active call.
The RECALL softkey does not appear in the expand set form.	• There are already the maximum number of RECALL keys (1) possible on the set. • The RECALL key is not programmable on a PKM Module associated with a subattendant station.
The HOLD softkey does not appear in the expand set form.	• There are already the maximum number of HOLD keys (6) possible on the set. • The HOLD key is not programmable on a PKM Module associated with a subattendant station.
USING LDN KEYS	
Set is ringing with an LDN key LCD flashing and I can't answer the line.	• Check that the caller can connect to the sub- attendant where the LDN is programmed. Subattendants and consoles which share LDNs must be in the same tenant group. • Check that the prime line is free, since the LDN can only be answered if the set prime is available.
Calls to the LDN are not answered.	• Check that the calls are being noticed at the subattendant set. The RING option may be set to no ring for the appearance and so new calls may not be noticed. Instruct the subattendant on the Call Waiting indicator on the display or how to use the DISPLAY function on the set to see how many calls are waiting on the particular LDN or RECALL key. • Ensure that the proper LDN key is actually being called. Check the OTHER field in MAINT show status for the caller and make sure that it is referencing the correct LDN. • LDN appearances programmed as NO RING or as DELAY RING are considered lower priority calls before the delay ring timer has expired. Therefore, other calls to the set on ringing lines (other LDN appearances, RECALLs, calls to the prime line, etc.) will be answered first.
USING RECALL KEY	
Recalls go unanswered at the subattendant set.	• Check that other recall features are not taking priority over the recall to the subattendant set.

• Check that a RECALL key is programmed at the set. If not, recalls will attempt to ring the subattendant prime line which must be idle at the time. If it is not idle, the calls will attempt to recall again in a short time.	
Recall display is not there - only a normal display is shown.	• A recall feature which uses CDE Form 19 Rerouting has taken precedence - the call is routed to the subattendant in that manner rather than using the default recall handling. These calls will appear at a key other than the RECALL key.
The RING AGAIN prompt is not present.	• The prompt is only valid if the party that was answered can be put on consultation hold (i.e. they don't have a consultation hold, the party can be flashed on, etc.). The TRANS/CONF prompt is present on the Sub- attendant display if this is possible.
STATIONS SOFTKEY	
No features are available when the Stations softkey is pressed and an extension number is dialed.	

• Ensure that the extension number is valid (the back arrow prompt will be showing still if the number is invalid).

• Ensure that the appropriate COS options are enabled in the Subattendant's COS (it is not necessary in the extension's COS).	
TIME/DATE SOFTKEY	
The softkeys for these features do not appear.	• Check that COS option 122 (Setup Time/Date) is enabled.
PAGED HOLD ACCESS	
Paged Hold Access display doesn't show up when the Sub- attendant selects the pager.	

• Check that the Subattendant has a caller on hold in a hold position.

• Check that feature access code 16 (Hold Pickup Access - Attendant Hold Slots) is enabled.	
C/W INDICATOR	
The Calls Waiting indicator shows calls waiting, but no LDNs or RECALL keys have calls on them.	• A night bell may be ringing.

Automatic Call Distribution (ACD)

The following table, ACD Troubleshooting, describes troubleshooting procedures for an ACD system.

Table: ACD Troubleshooting

Problem	Possible Cause/Solution
CALLING A PATH	
Caller gets reorder tone.	• Check that the correct access code is dialed. • Verify that the caller is not interflowing immediately to DROP CALL (the path called has Overflow to Interflow enabled, Interflow enabled, DROP CALL as the interflow point and there are no agents available to take the call within the overflow time). • If a DISA or CO trunk is calling, the trunk is loop start and the trunk's COS may have the option "Loop Start Trunk to ACD Connect" disabled.
Interflow produces reorder tone.	• If the programmed interflow point is DROP CALL, and the caller is a display set, the display will indicate that the path has hung up on the caller. • If the programmed interflow point is a speedcall, keep in mind that the interflow is handled in the same way as a call forward to a speedcall. First, check that the speedcall number is a valid destination and that the caller can connect to that destination. If the caller is a display set, the display will indicate an invalid number or connection in this case. Next, verify that a receiver is available - one is needed to complete the speedcall. If there are no receivers immediately available at the time of the interflow, the ACD caller is dropped. The only indication of this will be a receiver unavailable peg in the traffic measurement report. The call will still show up in the ACD statistics as an interflow.
Trunk won't originate to a path.	• Check the party state of the trunk using SHOW STATUS. If it is not able to originate at all due to resource shortage, the state will be IDLE. In this case, check traffic measurement for channel usage. If the trunk is denied access to the routing point, or it is not programmed, the state will be ERROR. In this case, check the routing point in CDE Form 14 for the current night/day service. If correct, and it is an ACD path, and the trunk is loop start, check if the trunk's COS has the option "Loop Start Trunk to ACD Connect" enabled.

Caller runs right though the path and drops.	• Check if interflow is enabled and if overflow to interflow is enabled. If there is a shortage of agents, the system may be predictively overflowing to the interflow point.
Caller gets wrong agents.	• Check routing in Form 14 for CO trunks. • Check that the correct path is actually being dialed. • Check to see if the agents are picking up other agent's calls using the CALL PICKUP feature.
ACD RECORDINGS	
Can't get recordings.	• Check path programming. • Check RADs in RAD group using MAINT SHOW STATUS or the console STATIONS softkey. If all are in the DND state, check the programming on the RAD group and in the COS of the RADs (see section on recording malfunction) to see if timers are not set up properly. • Call RAD group directly and check that a RAD answers and gives the correct message.
Can't get first recording, but can get second recording.	• The time for the second recording is occurring while waiting for the first recording, and the first recording is being abandoned. Make more recordings available in first group and increase the start time for the second recording (and subsequent recordings).
Silence between recordings.	• System music is not programmed. • System music not connected properly. • System music does not have enough gain.
ALTERNATE MUSIC SOURCE	
Alternate music source not heard.	• Check path (see previous - *Calling a Path*).

	• Check the party state of the alternate music source - via SHOW STATUS. If state is ALTMUS, the port is plugged in and ready. Check the connections on the port. If the party state is not ALTMUS, the system has not seen the port go offhook, or the port was offhook when programmed as an alternate music source. It must be programmed first and then make the transition from IDLE to ALTMUS. Simply unplug and plug back in the alternate music source and it should go to ALTMUS state. • If the alternate music source is functioning, check the recordings. Music will not be applied, unless a caller has been given a RAD to listen to first.
ACD POSITIONS	
An ACD position cannot log in.	• A valid login feature access code and position id code must be used (check CDE). The display will indicate an invalid number dialed (all of the other errors will cause the display to indicate an invalid feature access attempt). • The position must not already be logged in. Try dialing the position code - do you complete a call to the position? • The Mitel telephone cannot have a party on consultation hold when the login is attempted. • The number of agents logged in must not exceed the maximum number. Check the number of agents logged in using the ACD MONITORS feature. • The position must have an ACD template enabled in its COS. Can the position log in at another telephone where another position of the same type was able to log in? • To log in, the set must not having any appearances of its prime line anywhere in the system. To confirm this, do a review on the Mitel in Form 9. • The Mitel telephone must not have a PKM Module associated with it. Check the ASSOC field in Form 09 or 45 for the circuit. • There must be no appearances of a multiline or key line on this set or multiline appearances from it on other sets.
An ACD position cannot log out.	• Check that the agent is not on an ACD call when this is attempted. This includes ACD calls on consultation hold and held on a line at the Mitel telephone.
Agents going make busy state.	

- Check set errors via SHOW ERRORS DEVICE-TYPE xxx command.

	• The set will be put into make busy if the set is disconnected, or loses communication with the system. • Check call forward no answer time in the COS of the set - this controls the automatic call failure timing. Increase the time if necessary.
ACD positions are unexpectedly logged out.	

- If a telephone is unplugged, and replaced with a different type of telephone while an ACD position is logged in at that position, the ACD position is automatically logged out by the system.

	• Could be caused by a system reset - check for this.
Forwarding does not work after position logs in.	

- Problems with forwarding to a speedcall key may arise with ACD position sets.

	• Forwarding to a speedcall key works by reference to the speedcall key number and not to the digits in the speedcall key. If a position logs at a set that has forwarding to a speedcall key, the forwarding on the set is not altered - the reference to the old speedcall key number remains. If the speedcall key is replaced by an ACD feature key because of the login, the forwarding will be to a key key which is now a feature key - this is illegal. When someone calls the set and the forwarding takes place, the forwarding will fail and the caller will get reorder tone. A similar problem will occur even if the speedcall key is not replaced by a feature key - it will be replaced by a blank speedcall key and so the forwarding will fail because there are no digits to dial.

Automatic Attendant Overflow (AAO)

The following table, Auto Attendant Overflow Troubleshooting, describes troubleshooting procedures for an AAO system.

Table: Auto Attendant Overflow Troubleshooting

Problem	Possible Cause/Solution
Calls come in to a console/LDN but get the wrong message.	

- Ensure that the RAD group (see CDE Form 19) for the tenant group of the called party is the correct one, and that the called party is in the correct day/night service.

- Ensure that the RADs programmed for the RAD group in Form 17 are giving the correct message. Dial them directly to verify this.

Calls come in to a console but do not get the recorded message.

- Ensure that caller has COS option 705 Automatic Overflow from Attendant enabled.

- Ensure that Day, Night1 and Night2 answer points are programmed in Form 19.
- Ensure that RADs are usable (i.e. not all DND, and have a message recorded). This can be done by calling the RAD group directly, or by calling them individually.
- Ensure that time to start recording is small enough. If it is set too high, the caller may be answered by the console before hearing the message.
- Ensure that the call is not being dropped (due to the final Ring timer) prior to hearing a message (i.e., the final Ring timer may be set too low).

The caller is cut off in the middle of a recording.

- Ensure that the message length timer for the RAD group is long enough for the RAD message.

- Ensure that the call is not being dropped due to the final Ring timer, which runs while listening to the recording. It should be at least the time to the recording plus the time to hear the recording.

Ringback followed by silence.

- Call may have been dropped, due to the final ringback timer.

- Check via MAINTENANCE SHOW STATUS, the OTHER field for the caller, to see if it actually gets connected up to the recording.
- There may not be a properly recorded message on the RAD (the recording may be silence). Call the RAD directly to verify this.

Automated Attendant

The following table, Automated Attendant Troubleshooting, describes troubleshooting procedures for an Automated Attendant system.

Table: Automated Attendant Troubleshooting

Problem	Possible Cause/Solution

Call is never answered by RAD. The call always routes to the default destination.	• Ensure that RAD is functioning (not all DND or busied-out). Use MAINTENANCE or the attendant console to check the DND status of the device. (Use SHOW STATUS command.) • Ensure that the RAD message length is not too short. If the message length for the RAD group is very short, the RAD may not even get a chance to play its message. • Check traffic measurement for skip pegs for the group. This will indicate a shortage of RADs or receivers. • Check the wait-for-resources time for the group. It may be too short, resulting in there being not enough recordings for the number of callers (given the length of time that the recording plays). The time must be at least as long as the turn around time for the RADs in the group, since this is the longest time that a waiting caller will have to wait if there are receivers available. Competition for receiver resources will increase this wait time.
Call is dropped when caller dials no digits.	• Ensure that there is a default destination for the group.
Recording does not end when digits are dialed.	• Ensure that the option “Dialing Over Recording” is ENABLED for that auto-attendant group. • In the case of an attendant calling, the console must have TONES ON in order to perform end-to-end signaling and dial over the auto-attendant recording.
Caller gets the wrong message.	• Ensure that message was recorded correctly. • Ensure that RAD is programmed in the correct group (use the EXT NUM softkey in CDE Form 17 to find the RAD). • Ensure that RAD is plugged in at correct extension by dialing the RAD extension number directly.
Dialing is slow and Mitel telephone displays SYSTEM BUSY.	• There may not be enough non-auto-attendant receivers allocated. Check traffic measurement for receiver attempt pegs and adjust the receiver count for non-auto-attendant features in the system options form if necessary.

Call Forwarding

The following table, Call Forwarding Troubleshooting, outlines the procedure for troubleshooting Call Forwarding.

Table: Call Forwarding Troubleshooting

Problem	Possible Cause/Solution
No access to call forwarding.	• If an attendant, ensure that COS option 123 - Attendant Call Forward Setup And Cancel is enabled. • If an extension user, ensure that at least one of the following COS options is enabled: - 206 - Call Forward - Busy - 207 - Call Forward - No Answer - 209 - Call Forward - Always
No access to split (internal / external) call forwarding.	• Ensure that COS option 260 Internal/External Split Call Forwarding is enabled.
Reorder tone and/or INVALID error message.	• Ensure that the forwarding destination you are entering is valid. The forwarding destination must be (directly or translate into) one of the following: - an industry standard telephone - a Mitel telephone - an attendant console - a hunt group (not data or modem) - a night bell - an ACD path - a dial 0 access code - an LDN • The extension must be allowed (i.e., device interconnection) to connect to the forwarding destination. • The forwarding destination must not have COS option 234 - Never A Forwardee enabled. • The forwarding destination cannot be the same as your extension number (i.e. you can’t forward to yourself). • If entering a personal or system abbreviated dial access code as a forwarding destination, the forwarding extension must have the following COS option settings: - 245 - Abbreviated Dialing Access - enabled - 208 - Call Forwarding - External - enabled - 200 - Account Code, Forced Entry - External Calls - disabled.

- If entering a personal speedcall button as a forwarding destination, the forwarding extension must have the following COS option settings :
  - 208 - Call Forwarding - External - enabled
  - 200 - Account Code, Forced Entry - External Calls - disabled.
- Programming call forwarding while a system copy database operation is in progress is not permitted.
- If programming “I’m Here” forwarding, the from party must be a industry standard telephone or a Mitel telephone, and cannot be a member of any type of voice mail hunt group.
- If programming forwarding using dialed forwarding access codes, verify the following:
  - the extension is permitted access to the feature,
  - For feature access codes 03, 04 and 05, the access code is entered, followed by one of: 1 (always), 2 (busy), 3 (no answer) or 4 (busy/no answer).

Calls are not being forwarded properly.

- If a display set user, use the appropriate softkeys to examine the current forwarding settings. The forwarding may have been changed by the attendant console or a sub- attendant.
- If the forwarding was set using a personal speedcall button, ensure that the contents of the button still translate into a valid forwarding destination. Ensure using Form 9 that the key is still in fact programmed as a speedcall key.
- Ensure that the COS options permit Call Forwarding. Check specifically COS options 200, 206, 207, 208, 209, 245, and 260. Consult Program Features for proper settings.
- Ensure that the forwarding destination is idle and has COS option 234 - Never A Forwardee disabled.
- Ensure that the forwarding destination has not been deprogrammed.

- Note that calls will not be forwarded if:
  - it is the attendant or a Mitel telephone that is overriding/intruding
  - it already has been forwarded via any type of speedcall
  - it already has been forwarded twice without being answered or rerouted.

- If a logical line appears on the extension and on another extension, calls to the first extension on the logical line will not be forwarded.
- If the last programmed call forwarding with COS option 260 (Internal/External Split Call Forwarding) enabled and subsequently the option was disabled, calls will be forwarded according to the external forwarding settings.
- Any changes in device interconnection rules may prohibit calls from being forwarded. Ensure that the calling party can indeed connect to the forwarding destination.
- Trunk calls to the extension will not be forwarded if calls are forwarded using any type of speedcall, with System option 21 (Incoming To Outgoing Call Forward) disabled.
- Calls will not be forwarded if the forwarding destination is the same as the calling party (i.e., calling party cannot be forwarded to itself).
- Cannot forward a caller to a busy hunt group if the caller does not have the capability of camping on to the hunt group.
- Cannot forward calls from the attendant console if the forwarding destination is a Dial 0 Access Code.
- If forwarded via a speedcall of any type, the system may be encountering difficulty acquiring the necessary resources to complete the forwarding. Ensure that there are ample available receivers and/or outgoing trunks to allow this type of forwarding.

SUPERSET Telephones

Table: SUPERSET Telephone Troubleshooting

Problem	Possible Cause/Solution
Audio always come out of the handset, even when onhook.	

- Check that COS option 612 - Headset Operation is not enabled in the set COS, the ACD position COS or the COS of a verified account code that the user dialed. This is usually the source of the problem.

- Ensure that the call is not being answered handsfree when COS option 612 (Telephone - Headset Operation is enabled in the set’s COS. The call will go to the handset even if the call is answered using the speaker key or a line key.
- Ensure that COS option 604 (Telephone - Automatic Outgoing Line) is enabled in the set’s COS.
- Run the loopback test to ensure that the hookswitch is working correctly.

Interpret Maintenance Log Messages

This part contains the complete set of information messages which are recorded in the maintenance log when a significant event occurs. Each message is self-explanatory, identifying the event and providing details about it. Information on options available to maintenance personnel is included under the Action Required heading.

There are three types of log reports:

1. **Fault Report -** a report generated when the maintenance system or Call Processing detects a fault, or an abnormal condition.

2. **Reset Report -** a report generated when a bay or the system is reset.
3. **Alarm Report -** a report generated when a change in any of the alarm levels occurs.

This part is divided into three sections which correspond to the three types of log messages. The Fault Reports section is arranged using the Alarm Code, an index number specific to the type of fault indicated. The Reset and Alarm Report sections are arranged in a logical manner.

Information on access to the maintenance log, and the use of other maintenance tools may be found in the RS-232 Maintenance Terminal Section.

Fault Reports

Table: Fault Reports

Alarm Code	Message	Action Required
06	ONS card failed at 02 01 01 00 ext 2101 inject codec test Alarm Code = 06 LS/GS trnk card failed at 02 02 01 00 inject codec test Alarm Code = 06 E&M module failed at 02 02 01 00 inject codec test Alarm Code = 06 DTMF RX module failed at 02 02 01 00 inject codec test Alarm Code = 06	Refer to the Problem Isolation part of this section. Otherwise, investigate further, using the SHOW STATUS command.
07	ONS card failed at 02 01 01 00 ext 2101 Dgl L/B codec test Alarm Code = 07 LS/GS trnk card failed at 02 02 01 00 Dgl L/B codec test Alarm Code = 07 E&M module failed at 02 02 01 00 Dgl L/B codec test Alarm Code = 07 DTMF RX module failed at 02 02 01 00 Dgl L/B codec test Alarm Code = 07	Refer to the Problem Isolation part of this section for this card or module. Otherwise, investigate further, using the SHOW STATUS command.
08	ONS card failed at 02 01 01 00 ext 2101 Ang L/B codec test Alarm Code = 08 LS/GS trnk card failed at 02 02 01 00 Ang L/B codec test Alarm Code = 08 E&M module failed at 02 02 01 00 Ang L/B codec test Alarm Code = 08 DTMF RX module failed at 02 02 01 00 Ang L/B codec test Alarm Code = 08	Refer to the appropriate part of this section. Replace as required. Otherwise, investigate further, using the SHOW STATUS command.

09	ONS card failed at 02 01 01 00 ext 2101 Message lamp test Alarm Code = 09	The specified station has been unplugged, or lamp bulb needs to be replaced.
10	ONS card failed at 02 01 01 00 ext 222 Hook test Alarm Code = 10	Refer to the ONS/CLASS Line Card part of this section. Replace as required. Otherwise, investigate further, using the SHOW STATUS command.
11		

ONS card failed at 02 01 01 00 ext 2101
Adc reference test Alarm Code = 11

LS/GS trnk card failed at 02 02 01 00 Adc reference test Alarm Code = 11	Refer to the Fault Isolation part of this section. Otherwise, investigate further, using the SHOW STATUS command.	
12	Card read test Alarm code = 12	
13	LS/GS trnk card failed at 02 02 01 00 Hybrid loopback test Alarm Code =13	Refer to the appropriate part of this section. Otherwise, investigate further, using the SHOW STATUS command.
14	DIG line card failed at 01 08 01 01 ext 111 DNIC output L/B test Alarm Code =14	DNIC chip failed at circuit 01. Problem with DIGITAL Line Card.
15	DIG line card failed at 01 08 01 01 ext 111 DNIC input L/B test Alarm code =15	DNIC chip failed at circuit 07. Problem with DIGITAL Line Card.
16	DIG line card failed at 01 08 01 01 ext 111 dn set c/s test Alarm Code = 16	Replace the specified digital SUPERSET telephone.
17	DIG line card failed at 01 08 01 01 ext 122 dn set bphone test AlarmCode = 17	Replace the specified digital SUPERSET telephone.
18	DIG line card failed at 01 08 01 01 ext 122 dn set t'ducer test Alarm Code = 18	Replace the specified digital SUPERSET telephone.

19	DIG line card failed at 01 08 01 02 ext 222 dataset L/B test Alarm Code = 19	Replace the specified dataset.
20	DTMF RX module failed at 02 02 01 00 DTMF receiver test Alarm Code = 20	Refer to the DTMF Receiver part of this section. Otherwise, investigate further, using the SHOW STATUS command.
21	PRINTER failed at 00 00 02 00 Printer test Alarm Code = 21	Check if printer is off-line or disconnected. Refer to printer troubleshooting procedures.
33	Link 00 Channel 02 Failed isol to mcc PCM loopback test Alarm Code = 33	Failure on DX module. Replace controller.
34	Link 07 Channel 19 Failed isol to mcc PCM loopback test Alarm Code = 34	Failure on DX module. Replace controller.
36	DIG line card passed at 01 08 01 01 ext 111 Alarm Code = 36	Information only. Previous problem corrected.
40	ONS card failed at 03 01 01 00 Analog 8804 test Alarm Code = 40	Card fault - replace card.
41	ONS card failed at 03 01 01 00 Analog 8804 test Alarm Code = 41	Card fault - replace card.
42	ONS card failed at 03 01 01 00 No path to junct # # Alarm Code = 42	Card fault - replace card.
43	DIG line card failed at 01 08 01 01 ext 111 dn set earpiece test Alarm Code = 43	Refer to Special Sets troubleshooting procedures.
44	DIG line card failed at 02 02 01 00 ext 111 dn set speaker test Alarm Code = 44	Refer to Special Sets troubleshooting procedures.
45	DIG line card failed at 02 02 01 00 ext 111 dn set microphe test Alarm Code = 45	Refer to Special Sets troubleshooting procedures.

46	DIG line card failed at 02 02 01 00 ext 211 dn set mouthpce test Alarm Code = 46	Refer to Special Sets troubleshooting procedures.
47	T1 trunk card failed at 01 06 01 00 Trk 19 t1 channel L/B test Alarm Code = 47	Refer to T1 Trunk Troubleshooting Procedures.
48	DIG line card failed at 05 01 01 Modem answer test Alarm Code = 48	The modem at this location is not able to answer a call. Refer to applicable Dataset Troubleshooting Procedure.
49	DIG line card failed at 05 01 01 Modem originate test Alarm Code = 49	The modem at this location is not able to originate a call. Refer to applicable Dataset Troubleshooting procedure.
52	ONS card failed at 03 01 01 00 Analog 8804 test Alarm Code = 52	Card fault - replace card.
53	ONS card failed at 03 01 01 00 No path to junct # # Alarm Code = 53	Card fault - replace card.
101	ONS card failed at 02 01 01 00 ext 2101 OFF hook too long Alarm Code = 101	Information only.
102	ONS card failed at 02 01 01 00 ext 2101 Card removed Alarm Code = 102	Verify card has been removed. If alarm is raised due to this, either replace/re-install the card, or deprogram it via CDE. Refer to the Customer Data Entry section for details.

DIG line card failed at 03 01 01 01 Card removed Alarm Code = 102 LS/GS trnk card failed at 02 02 01 00 Tk 9 Card removed Alarm Code = 102 CO trunk card failed at 02 02 01 00 Tk 09 Card removed Alarm Code = 102 E&M trunk card failed at 02 02 01 00 Tk9 Card removed Alarm Code = 102 DID trunk card failed at 02 02 01 00 Tk9 Card removed Alarm Code = 102 UNIVERSAL card failed at 02 03 01 00 Card removed Alarm Code = 102 T1 trunk card failed at 02 06 00 00 Card removed Alarm Code = 102 T1 ISDN card failed at 15 02 00 00 Card removed Alarm Code = 102	**Note:** If the controller contains a T1 ISDN card (T1/E1 module for embedded PRI or T1/D4), the following alarm codes will be generated whenever the system resets: • 102 (card removed) • 103 (card installed)
103	

ONS card passed at 02 01 01 00 ext 2101
Card installed Alarm Code =103

	DIG line card passed at 03 01 01 01 Card installed Alarm Code = 103 LS/GS trunk card passed at 02 02 01 00 Tk 9 Card installed Alarm Code = 103 CO trunk card passed at 02 02 01 00 Tk08 Card installed Alarm Code = 103 E&M trunk card passed at 02 02 01 00 Tk8 Card installed Alarm Code = 103 DID trunk card passed at 02 02 01 00 Tk 8 Card installed Alarm Code = 103 UNIVERSAL card passed at 02 03 01 00 Card installed Alarm Code = 103 T1 trunk card passed at 02 06 00 00 Card installed Alarm Code = 103 T1 ISDN card passed at 15 02 00 00 Card installed Alarm Code = 103	Information only.
104	ONS card failed at 02 01 01 00 ext 2101 Wrong card in slot Alarm Code = 104 LS/GS trk card failed at 02 02 01 00 Tk 08 Wrong card in slot Alarm Code = 104 CO trunk card failed at 02 02 01 00 Tk 8 Wrong card in slot Alarm Code = 104 E&M trunk card failed at 02 02 01 00 Tk 8 Wrong card in slot Alarm Code = 104 DID trunk card failed at 02 02 01 00 Tk 09 Wrong card in slot Alarm Code = 104 UNIVERSAL card failed at 02 03 01 00 Wrong card in slot Alarm Code = 104	The specified card type is not programmed for the specified card slot. Use the SHOW CONFIG command to observe the correct configuration. Either insert the correct card type, or reprogram the card slot (see the Customer Data Entry section).
105	Digital Line Card failed at 02 01 01 01 ext 2201 Superset unplugged Alarm Code = 105	A SUPERSET telephone is unplugged; reconnect the specified telephone. This alarm condition relates to the Disconnect Alarm feature.

106	Database failed at 00 00 xx 00 Serious ram shortage Alarm Code = 106	This is a warning. Stop CDE programming activity. Wait for a low traffic period, and reset the system. If this persists, search for an Alarm Code 108 message. Watch for further occurrences.
107	Database failed at 00 00 03 00 Serious CMOS shortage Alarm Code = 107	This is a warning. Stop CDE activity. Wait for a low-traffic period, perform a COPY DATABASE, and reset the system using the new database. If this persists, search for an Alarm Code 109 message. Watch for further occurrences.
108	RAM failed at 00 00 xx 00 No Ram space left Alarm Code = 108	No further CDE programming will be possible. There is no RAM space available. Wait for a period of low traffic, and reset the system. If further programming is required, other devices, account code numbers, ARS strings, or speed call numbers will have to be deprogrammed.
110	ONS card failed at 02 01 01 00 ext 120 Msg reg overflowAlarm Code = 110	The message registration counter for the specified extension has overflowed. The overflow threshold is 50,000. Ensure that the counters are periodically reset at the Attendant Console.
111		

LS/GS trnk card failed at 02 02 01 00 Trk 1
Can't seize trunk AlarmCode=111

CO trunk card failed at 02 02 01 00 Trk 01 Can't seize trunk Alarm Code = 111 E&M trunk card failed at 02 02 01 00 Tk2 Can't seize trunk Alarm Code = 111 DID trunk card failed at 02 02 01 00 Tk7 Can't seize trunk Alarm Code = 111	Verify the wiring from the trunk circuit to the public network. Correct as required. Refer to the Installation Information section. If this is not the problem, suspect failure or bad wiring at the Central Office.
112	

LS/GS trnk card failed at 02 02 01 00 Trk 1
Can't release trunk Alarm Code = 112

	CO trunk card failed at 02 02 01 00 Trk 1 Can't release trunk Alarm Code = 112 E&M trunk card failed at 02 02 01 00 Tk7 Can't release trunk Alarm Code = 112 DID trunk card failed at 02 02 01 00 Trk 7 Can't release trunk Alarm Code = 112	No release signal was received from the Central Office. Verify wiring. Refer to the Installation Information section for details.
113	UNIVERSAL card failed at 02 03 01 00 Exceeds power rating Alarm Code =113	The total power rating of the modules installed on the specified Universal Card exceeds the maximum permitted total power rating. Refer to the Installation Information section for details.
114	PRINTER failed at 00 00 02 00 SMDR printer down Alarm Code = 114	The printer used for SMDR printing is off-line or not working. Check printer. Refer to printer troubleshooting procedures.
117	RAM failed at 00 00 05 00 CMOS checksum failed Alarm Code = 117	This is not a serious problem if it occurs once. However, if it is persistent, refer problem to Mitel Field Service.
118	UNIVERSAL card failed at 02 03 01 00 Card in low pwr slot Alarm Code = 118 DID card failed at 01 05 01 00 Card in low pwr slot Alarm Code = 118 OPS card failed at 02 03 01 00 ext 1502 Card in low pwr slot Alarm Code = 118	The specified card is a high-power card installed in a low power slot. Use SHOW CONFIG command to obtain information on the card slot. Re-install/reprogram the card for a high-power slot.
	UNIVERSAL card failed at 02 03 01 00 Exceeds Bay card pwr Alarm Code = 118 DID card failed at 01 05 01 00 Exceeds Bay card pwr Alarm Code = 118 T1 trunk card failed at 02 06 00 00 Exceeds Bay card pwr Alarm Code = 118	More than four high-power cards are installed in this Bay. Use SHOW CONFIG command to obtain information on the card slot. Install the card into a Bay with fewer than four high-power cards.

121

PFT sense failed at 00 00 08 00
Bay has cut through Alarm Code = 121

BAY failed at 03 00 00 00 Bay has cut through Alarm Code = 121	The bay has gone into Power Fail Transfer mode. Use SHOW ALARMS command and examine logs further to find the actual cause of the cut through.

122

PFT sense passed at 00 00 08 00
Bay has cut back Alarm Code = 122

PFT sense passed at 03 00 00 00 Bay has cut back Alarm Code = 122	The bay has been cut back to normal operation. Verify that appropriate action was taken to rectify the event which caused the cut through.

123

ONS card failed at 02 01 01 00 ext 123
Recording dev failed Alarm Code = 123

ONS card failed at 02 01 01 00 ext 2101 Recording device failed - false origination Alarm Code = 123	GENERAL: The recording device attached to the specified port has malfunctioned. Check wiring. Refer to the instructions provided by the manufacturer of the recording device.

ONS card failed at 02 01 01 00 ext 2101
Recording device failed to hang up-lok
Alarm Code = 123

ONS card failed at 02 01 01 00 ext 2101 Recording device failed to hang up - sus	RAD failed to hang up - locked out.

ONS card failed at 02 01 01 00 ext 2101
Recording device failed to answer
Alarm Code = 123

ONS card failed at 02 01 01 00 ext 2101 Recording device failed - nil error Alarm Code = 123	RAD failed to hang up - suspect state.

Recording dev test failed at 01 11 07 01
failure to answer Alarm Code = 123

VOICE MAIL CARD failed at 01 11 07 01 Ext 307 Recording dev test Alarm Code = 123	The RAD for an embedded voice mail system failed to answer.	
124	PRINTER failed at 00 00 02 00 Wakeup printer down AlarmCode=124	The printer used for Hotel/Motel wakeup printing is off-line or not working. Check printer. Refer to Printer troubleshooting procedures.
125	ONS card failed at 02 01 01 00 ext 210 Wakeup not answered AlarmCode=125	Information only.
126	Any line card failed at 02 01 01 00 ext 123 Plid Restored Alarm Code = 126	Record this and watch for further occurrences. If system performance is degraded substantially, contact Mitel Field Service.
128	DTMF RX module failed at 02 02 01 00 Receiver locked out Alarm Code =128	Reseat the affected Universal Card.
	DTMF RX module failed at 00 12 01 00 Receiver locked out Alarm Code =128	Receiver in slot 12 of a Control Bay. Check and correct RCVR status. Reset the PBX. Replace the SX-200 ICP. Call Mitel Technical Support.
129		

** anything ** failed at 00 00 00 00
Group_Link_Test not a proper member
Alarm Code = 129

** anything ** failed at 00 00 00 00 Group_Link_Test Invalid Group Link Alarm Code = 129 ** anything ** failed at 00 00 00 00 Group_Link_Test many members in Group Alarm Code = 129 ** anything ** failed at 00 00 00 00 Group_Link_Test nil error Alarm Code = 129	Record this and watch for further occurrences. Perform a Verify Database operation and check for further occurrences of this Alarm Code. If system performance is degraded substantially, contact Mitel Field Service.

130	** anything ** failed at 00 00 00 00 Plid to Swid failed Alarm Code = 130	Record this and watch for further occurrences. Perform a Verify Database operation and check for further occurrences of this Alarm Code. If system performance is degraded substantially, contact Mitel Field Service.
131	RAM failed at 00 00 04 00 CMOS VS Ram Failed Alarm Code=131	Record this and watch for further occurrences. Perform a Verify Database operation and check for further occurrences of this Alarm Code. If system performance is degraded substantially, contact Mitel Field Service.
132	Key_DB_Test failed at 03 01 08 01 Set has no prime key Alarm Code = 132	Assign the phone a prime line key.
133	** anything ** failed at 00 00 00 00 Trunk Number Corrupt.	In the SX-200 LIGHT system, a CDE audit has failed. If this persists, attempt a COPY DATABASE as soon as possible. Perform a Verify Database operation and check for further occurrences of this Alarm Code. If this still persists; contact Mitel Field Service.
134	** anything ** failed at 00 00 00 00 Access Code Tbl Bad.	In the SX-200 LIGHT system, a CDE audit has failed. If this persists, attempt a COPY DATABASE as soon as possible. Perform a Verify Database operation and check for further occurrences of this Alarm Code. If this still persists; contact Mitel Field Service.
135	UPS sense failed at 00 00 09 00 UPS not available Alarm Code = 135	The Uninterruptible Power Supply is not operating. Check the relevant wiring. Refer to the UPS part of this section.

136	UPS sense failed at 00 00 09 00 UPS not available Alarm Code = 135	The Uninterruptible Power Supply is not operating. Check the relevant wiring. Refer to the UPS part of this section.
137	UPS sense failed at 00 00 09 00 ac voltage failure Alarm Code = 137	The line ac voltage has failed. Ensure UPS is functioning.
138	UPS sense failed at 00 00 09 00 Battery/charger Alarm Code = 138	Either the battery is failing, or the battery charger is not functioning. Examine battery, charger, and wiring. Also refer to the instructions provided by the manufacturer of the UPS.
139	UPS sense failed at 00 00 09 00 AC/battery/charger Alarm Code = 139	There is no line ac voltage. Also, the battery is failing, or the charger is/was not functioning properly. Examine battery, charger and wiring. Also, refer to the instructions provided by the manufacturer of the UPS.
141	Main Controller failed at 00 00 00 00 PMS is down Alarm Code = 141	The PMS system failed to respond to 20 consecutive queries from the PBX. Refer to PMS procedures.
142	Main Controller failed at 00 00 00 00 PMS is up Alarm Code = 142	After failing to respond, the PMS system has now responded to a PBX enquiry.
143	Main Controller failed at 00 00 00 00 PMS buffer is full Alarm Code = 143	The system attempted to send a message to the PMS message buffer; buffer was full. If persistent, refer to PMS procedures.
144	Main Controller failed at 00 00 00 00 No STX from PMS Alarm Code = 144	PMS has sent the PBX an invalid START-OF-TEXT message. Refer to PMS procedures.
145	Main Controller failed at 00 00 00 00 No ETX from PMS Alarm Code = 145	PMS has sent the PBX an invalid END-OF-TEXT message. Refer to PMS procedures.
146	Main Controller failed at 00 00 00 00 Bad PMS function Alarm Code = 146	PMS has sent the PBX an invalid function message. Refer to PMS procedures.

147	Main Controller failed at 00 00 00 00 Bad PMS status Alarm Code = 147	PMS has sent the PBX an invalid status message. Refer to PMS procedures.
148	Main Controller failed at 00 00 00 00 Bad PMS room number AlarmCode=148	PMS has sent the PBX an invalid room number message. Refer to PMS procedures.
149	Main Controller failed at 00 00 00 00 Cannot send PMS msg AlarmCode=149	If the PMS refuses to accept a transaction from the PBX after five tries, the PBX will generate this log.
150	LS/GS trk card failed at 01 05 01 00 trk 001 Trunk no dial tone Alarm Code = 150	This trunk was seized and after 10 seconds dial tone was not detected. The trunk has been busied out.
151	Link 07 Channel 19 Busied out Alarm Code = 151	Device busied out by maintenance personnel.
151	ONS card failed at 01 01 01 00 ext 1101 Busied out Alarm Code = 151	Device busied out by maintenance personnel.
151	BAY CCT> 0 failed at 02 00 01 00 Busied out	Return to service 02/00/01
151	BAD BAY NUM failed at 00 12 01 00 Busied out	Return to service 00/12/01
152		

ONS card passed at 01 01 01 00 ext 1101
Returned to service Alarm Code = 152

Link 07 Channel 19 Returned to service Alarm Code = 152

Device returned to service by maintenance personnel.

Device returned to service by maintenance personnel.		
	BAY CCT> 0 passed at 02 00 01 00 Returned to service	None
	BAD BAY NUM passed at 00 12 01 00 Returned to service	None

155	T1 trunk card passed at 01 06 01 00 Trk 019 has exceeded the maintenance slip threshold	The link has exceeded the specified threshold - this is a warning - watch for further occurrences. If persistent, refer to T1 Trunk troubleshooting procedures.
	T1 trunk card passed at 01 06 01 00 Trk 019 is now below the maintenance slip threshold	Link was running with errors, is now running at an acceptable error rate. Information only.
	T1 trunk card at 01 06 01 00 Trk 019 has exceeded the service slip threshold	The link has exceeded the specified threshold. A yellow alarm has been sent to the far end, and the link has been removed from service, and a new sync source selected. The RTS - Service Limit timer has been started. Refer to T1 Trunk troubleshooting procedures.
	T1 trunk card at 01 06 01 00 Trk 019 has exceeded the maint loss frame threshold	The link has exceeded the specified threshold - this is a warning - watch for further occurrences. If persistent, refer to T1 Trunk troubleshooting procedures.
	T1 trunk card at 01 06 01 00 Trk 019 is now below the maint loss frame threshold	Link was running with errors, is now running at an acceptable error rate. Information only.
	T1 trunk card at 01 06 01 00 Trk 019 is in red alarm condition due to loss of power	No power on the link; it has been removed from service. If this is the network sync source, a new source has been selected. Problem with either the local Channel Service Unit, or at the far end.
	T1 trunk card at 01 06 01 00 Trk 019 exceeded sync slip thresh as current sync source	The link is the current network sync source - a new sync source has been selected. Refer to T1 troubleshooting procedures. If this message persists, may be necessary to make changes to the network synchronization source list.

	T1 trunk card at 01 06 01 00 Trk 019 is now below the net sync slip threshold	The RTS net slip timer has expired, and the link is within an acceptable threshold allowing it to be returned to service. The link may now be used as the network sync source. Information only.
	T1 trunk card at 01 06 01 00 Trk 019 alarm condition is now cleared	A red or yellow alarm has been cleared, and the RTS after alarm timer has expired. The link has returned to service. Information only.
	T1 trunk card at 01 06 01 00 Trk 019 System in freerun mode no sync source available	There is no network synchronization source that can be used (the source list has been exhausted) - the system is therefore in freerun mode. Refer to the T1 Trunk troubleshooting procedures. May be necessary to make changes to the network synchronization source list.
	T1 trunk card at 01 06 01 00 Trk 019 is current sync source and is in auto mode	The link is the new sync source, and the system is in auto mode. Information only.
	T1 trunk card at 01 06 01 00 Trk 019 is current sync source and is in manual mode	The link is the new sync source, and the system is in manual mode. Information only.
	T1 trunk card at 01 06 01 00 Trk 019 The system’s previous sync source was freerun mode	There was a change in the network sync source. The system was previously in freerun mode. Information only.
	T1 trunk card at 01 06 01 00 Trk 019 was previous sync source in manual mode	There was a change in the network sync source. The link was the old sync source, and the system was in manual mode. Information only.
	T1 trunk card at 01 06 01 00 Trk 019 was previous sync source in auto mode	There was a change in the network sync source. The link was the old sync source, and the system was in auto mode. Information only.

	T1 trunk card at 01 06 01 00 Trk 019 is reporting unstable link (phase error)	The link is reporting phase errors. If this is the network sync source, a new source has been selected. Problem with either the local Channel Service Unit, or at the far end.
	T1 trunk card at 01 06 01 00 Trk 019 is reporting unstable link (no phase error)	The link is no longer reporting phase errors. It is now available for use as the network sync source. Information only.
	T1 trunk card at 01 06 01 00 Trk 019 sync source manual timer has expired	The network sync source manual timer has expired and the system is changing from manual to auto or freerun mode. A network sync source is picked from the network synchronization source list. Information only.
	T1 trunk card at 01 06 01 00 Trk 019 No T1 clock module, running in freerun mode	Links for network sync have been specified, but there is no T1 clock module in the system. The system is running in freerun mode as a result.
	T1 Trunk card at 01 06 01 00 Trk 019 has 24th cct programmed. Wrong bay type reported.	Delete programming for the 24th circuit from CDE Form 14 (Non-Dial-In Trunks) or Form 15 (Dial-In Trunks). Power down the bay and then power it up again to generate another bay status report.
	T1 trunk card at 01 06 01 00 Trk 019 created log for unknown reason	Non problem. If persistent, contact Mitel Field Service.
156	DIG line card and failed at 05 01 01 PST checksum failed Alarm Code =156	Checksum of dataset firmware failed, replace set. Device will be busied out.
157	DIG line card failed at 05 01 01 PST RAM failed Alarm Code = 157	RAM in dataset failed (external or 6803). Replace set. Device will be busied out.
158	DIG line card failed at 05 01 01 PST UART l/b failed Alarm Code =158	UART loopback power up test failed; replace set. Device will be busied out.
159	DIG line card failed at 05 01 01 PST DNIC l/b failed Alarm Code =159	DNIC loopback power up self test failed; replace set. Device will be busied out.

160	DIG line card failed at 05 01 01 PSC HDLC failed Alarm Code = 160	HDLC controller failed in power up test; replace set. Device will be busied out.
161	DIG line card failed at 01 02 11 02 ext 123 Firmware trap Alarm Code = 161	Firmware trap occurred in set; check power supply. The device will not be busied out.
162		

DIG line card failed at 01 02 11 02 ext 123
Flood of EIA input Alarm Code = 162

ONS line card failed at 02 07 07 00 Ext 2807 Flood of EIA input Alarm Code = 162	Call Processing received more than 125 EIA input reports within 2 minutes while in the idle state; the data device was busied out. When the device is replaced, the data device will be unbusied. Check the equipment attached to the data device for a faulty EIA output.

163	DIG line card failed at 01 02 11 02 ext 123 Flood of ASCII input Alarm Code = 163	A flood of ASCII characters (100 more than the maximum input) was received, and the data device was busied out. If the device is replaced the data device will be unbusied. Check the equipment attached to the data device for flooding of ASCII characters (such as a large file being dumped to the DTRX process).
164	NIL PLID failed at 00 00 00 00 Bad PMS Name Alarm Code = 164	The name field received from the PMS system is improperly formatted, or is inconsistent with the name operator. Refer to PMS procedures.
165	NIL PLID failed at 00 00 00 00 Bad PMS Time Alarm Code = 165	The time field received from the PMS system is improperly formatted. Refer to PMS procedures.
166	MAIN CONTROLLER failed at 00 00 00 00 PMS requires console Alarm Code= 166	Dial 0 routing for the PMS is not a console LDN or a console prime number. Problem either with PMS database or local PBX programming.

170	DIG line card failed at 05 01 01 Firmware trap Alarm Code = 170	Firmware trap occurred in set; check power supply. The device will not be busied out.
171	DIG line card failed at 05 01 01 Flood of EIA input Alarm Code = 171	Call Processing received more than 125 EIA input reports within 2 minutes while in the idle state; the data device was busied out. When the device is replaced, the data device will be unbusied. Check the equipment attached to the data device for a faulty EIA output.
172	DIG line card failed at 05 01 01 Flood of ASCII input Alarm Code = 172	A flood of ASCII characters (100 more than the maximum input) was received, and the data device was busied out. If the device is replaced the data device will be unbusied. Check the equipment attached to the data device for flooding of ASCII characters (such as a large file being dumped to the DTRX process)
173	<CARD TYPE> card 01 08 11 01 Ext 1766 Called 911 Alarm Code = 173	Information only. Extension 1766 called 911.
174	<CARD TYPE> card 02 03 12 01 Att 1988 Att cleared 911 Alarm Code = 174	Information only. Attendant cleared 911 alarm.
175	<CARD TYPE> card 01 08 11 01 Ext 1766 911 Call cleared Alarm Code = 175	Information only. 911 alarm that was set at Extension 1766 is cleared .
176	<CARD TYPE> card 07 07 01 01 Ext 7701 Ext cleared 911 Alarm Code = 176	Information only. Extension 7701 cleared the 911 alarm
177	DIG line card 01 05 06 01 Ext 1506 Call logs delete at Alarm Code = 177	Information only. The call logs on extension 1506 were deleted.

178	DIG line card 06 07 01 01 Ext 6971 Call Logs de1eted by Alarm Code = 178	Information only. The call logs were deleted by extension 6971 (sub- attendant or attendant)

Alarm Log Reports

Table: Alarm Log Reports

Alarm Code	Action Required
Tot alarm went from No Alarm to MAJOR see Alarm Reasons in Table.	Use SHOW ALARMS command for more detailed information. Also see the applicable entry in Table.
Tot alarm went from MINOR to MAJOR see Alarm Reasons in Table.	Use SHOW ALARMS command for more detailed information. Also see the applicable entry in Table.
Tot alarm went from MAJOR to CRITICAL see Alarm Reasons in Table.	Use SHOW ALARMS command for more detailed information. Also see the applicable entry in Table.
Tot alarm went from MAJOR to MINOR see Alarm Reasons in Table.	This is an improvement in service. Information only.
Tot alarm went from MINOR to No Alarm see Alarm Reasons in Table.	This is an improvement in service. Information only.

Alarm Reset Reasons

Table: Alarm Reset Reasons

Alarm Level Change Reason	Action Required
Alarm level change due to Bay XX rcvrs Alarm level change due to Bay XX trunks Alarm level change due to Bay XX lines	Check the status of the specified bay (XX) via the SHOW ALARMS and SHOW STATUS commands.
Alarm level change due to system rcvrs Alarm level change due to system trunks Alarm level change due to system lines	Check the system status via the SHOW ALARMS and SHOW STATUS commands.
Due to threshold change of Bay XX rcvrs Due to threshold change of Bay XX trunks Due to threshold change of Bay XX lines	Check the status of the specified bay (XX) via the SHOW ALARMS and SHOW STATUS commands.
Due to threshold change of system rcvrs Due to threshold change of system trunks Due to threshold change of system lines	Check the status via the SHOW ALARMS and SHOW STATUS commands.

Main Control Reset Log Reports

Table: Main Control Reset Log Reports

Reset Log Message	Action Required
Main Control was reset due to power up	Information only. SX-200 ICP is reset at power-up.
Main Control was reset due to pushbutton	The RESET button on the SX-200 ICP was pushed. Investigate.
Main Control was reset due to watch dog timer Main Control was reset due to local bay cause Main Control was reset due to msg link failure Main Control was reset due to software error Process 00 ANYTHING at address 012345 Main Control was reset due to software error Main Control was reset due to software error Process 00 has overflowed its stack Main Control was reset due to software error Exception = ANYTHING at address 012345	In all cases, check if SX-200 ICP is functioning.
Main control was reset due to memory fragmentation.	The amount of free memory has dropped below 500 KB, causing the system to reset and a critical alarm to be generated. The system reset defragments the memory and should solve the problem. See Resource Monitor for memory fragmentation thresholds.

Bay Reset Log Reports

Table: Bay Reset Log Reports

Reset Log Message	Action Required
Bay X was reset due to msg link failure	The main control was unable to communicate with the bay, and therefore reset the bay.
Bay number X reported cause: parity failure at address XXXXXX	Failure of on-board DRAM. If persistent, replace the Bay Control Card.
Bay number X reported cause: power up at address XXXXXX	Information only. Bay Control Card is reset at power-up.
Bay number X reported cause: reset at address XXXXXX	Bay was reset because SX-200 ICP was reset. Investigate possible SX-200 ICP controller problem.
Bay number X reported cause: watchdog timer at address XXXXXX	Software failure. Check if SX-200 ICP is functioning.
Bay number X reported cause: ** anything else** at address XXXXXX	Software failure. Check if SX-200 ICP is functioning.

Firmware Download Log Reports

Table: Firmware Download Log Reports

Firmware Download Log Messages	Action Required
Firmware download failed at 06 04 02 01 ext 6402 Reason: an error occurred. List aborted!	The remaining sets on the download list are removed from the list if a single download failed for any reason. Refer to the failure reason for the action required.
Firmware download failed at 06 04 02 01 ext 6402 Reason: set was disconnected during downloading.	Set was unplugged during the download. Set will reboot but will not contain valid firmware. Re-issue the download command.
Firmware download failed at 06 04 02 01 ext 6402 Reason: set reports bad firmware version.	The download completed but reported a bad firmware version. Re-issue the download command.
Firmware download failed at 06 04 02 01 ext 6402 Reason:bad code on FLASH.	The firmware code (which is stored on the system FLASH card) has been detected as bad. A remote system download without resetting could cause this situation. Ensure the FLASH is installed, reset the PBX then re-issue the download.
Firmware download failed at 06 04 02 01 ext 6402 Reason:sys FLASH busy.	Performing a remote system download while a firmware download is running will cause this. Reset the PBX after the remote download then re-issue the firmware download command.
Firmware download failed at 06 04 02 01 ext 6402 Reason:sys FLASH type, code #.	The system FLASH card has been detected as either busy, removed, or incompatible. Install the FLASH card and re-issue the download. The code number can be reported to Mitel Field Service for further action.
Firmware download failed at 06 04 02 01 ext 6402 Reason: no response from erase, check boot rev.	The set was disconnected at the beginning of the download process. Re-issue the download command. The set's firmware revision may not support downloading. Check the boot rev. which is displayed on the set briefly after it is connected and verify that it supports downloading.
Firmware download failed at 06 04 02 01 ext 6402 Reason: set did not reboot.	After the download the set did not report its set ID. Re-issue the download or refer the problem to Mitel Field Service.
Firmware download failed at 06 04 02 01 ext 6402 Reason: set was not idle.	The set was detected as in use just before the download was about to start. Re-issue the download command.
Firmware download failed at 06 04 02 01 ext 6402 Reason: could not busy out set.	The bay could not confirm the busy-out of the set required before the download could start. Re-issue the download command.
Firmware download failed at 06 04 02 01 ext 6402 Reason: Interrupted. State was: #	A time-out was detected by the download watchdog software. Re-issue the download or refer the problem to Mitel Field Service.
Firmware download failed at 06 04 02 01 ext	The download process took too long. Refer the problem to

Table: Firmware Download Log Reports

Firmware Download Log Messages	Action Required
6402 Reason: Timeout. State was #	Mitel Field Service.
Firmware download failed at 06 04 02 01 ext 6402 Reason: SW trace code: #	Software failure present. Check to make sure that the Flash Card is plugged into the SX-200 ICP controller.

Miscellaneous Log Reports

Table: Miscellaneous Log Reports

Log Message	Action Required
System Configuration/System ID module conflict. Change system configuration to clear error.	Check the installation of the System ID module. Check Form 4 option selection.
Log(s) deleted by user	A user logged in to the maintenance facility (maintenance terminal or console) deleted one or more log entries via the DELETE LOGS command.
SMDR Report Lost Records	Check Class of Service 908 (DATA SMDR - Overwrite Buffer) must be enabled. Reset the printer. Contact Mitel Product Support.
Involved in recovery is 01 04 04 04 ext 1100 01 03 03 03 Trk 001 Main Control trace back at address = XXXXXX (anything) CP Process recovered from software error # XX (anything)	

In all cases, this indicates a software error. Refer the problem to Mitel Field Service.

Have a printed copy of the logs available before you call. Be prepared to forward a copy of the logs to Mitel Field Service if requested
BB/SS/CC/ZZ Excessive messaging Remove from service (BB/SS/CC/ZZ are bay, slot, circuit, and sub-circuit numbers respectively)

Excessive messaging events apply to digital devices connected to an SX-200 Peripheral bay. The event occurs when a device experiences trouble communicating with the bay control card.

Upon detection of excessive messages, the system stops listening to the sub-circuit for 20 seconds. During this time, the set displays “WAITING FOR SYNC” or “WAITING FOR COMM”. After 20 seconds, the system listens to the sub-circuit again. Most excessive messaging is caused by defective digital line cards. If a single sub-circuit is involved, then the problem may be the set or wiring. If problems are being reported for sub-circuits in different slots in the same bay, then the cause may be the bay control card. The device should be removed from service until the problem is resolved.	
Major alarm raised upon detecting the following task suspended: <XXXX>	Investigate the status of the system task noted in the error message. There are over 100 system tasks.
CX DB successfully converted from XXXX to XXXX	

The SX-200 ICP CX/CXi controller database has been upgraded to a higher version. The numbers that display (for example, 6666 and 3333) do not equate to release numbers.

No action is required.	
BAY number XX reported cause: bus error at address = 00000000	No action is required. This error is reported when you re-connect the CIM link between the ASU or ASU-II and the controller after removing it.
Socket Failed at 01 13 22 02 Ext 31322 Message flooding detected	

A flood of messages (more than 100) was received from a data socket, and the data socket was closed.

Check the application/equipment attached to the socket for excessive messaging (such as faulty precedia, iPocket, and so forth). Replace the equipment if the problem persists. Disconnect and then reconnect the application/equipment to bring it back to service.

Some Troubleshooting Aids

The block diagrams for the SX-200 ICP MX (Release 1.x), MX (Release 2.x) and CX (Release 2.x) controllers show the major components of each system make up and how they function together to process calls. Understanding this diagram will help you to troubleshoot problems. The components that make up the controllers are:

Time Division Multiplex (TDM) Switching Matrix

The TDM switching matrix is used to provide speech paths from the Analog Options module, the NSUs and the Peripheral node devices to other devices in the system. The TDM switching matrix is not used for calls from an IP phone to another IP phone.

Ethernet-to-TDM (E2T) card

The E2T card converts IP voice packets into TDM signals and vice versa. The E2T card is only used when an IP phone talks to a non-IP device (ONS phone, trunk, etc). It is not used for calls from an IP phone to another IP phone or for calls between non-IP devices. The card has 128 channel capacity.

Real Time Complex (RTC) block

The RTC block is used to relay messages between IP phones and the Call Control in the SX-200 ICP Controller. These messages are used to set up and take down calls.

Call Control

The Call Control complex controls which speech paths are set up through the TDM switch matrix or the E2T card. Call Control receives messages from a phone making a call and using the information in the messages (such as dialed number) it sets up the correct speech path.

Digital Signalling Processor (DSP)

The DSP tone generator generates the call progress tones (dial tone, reorder tone etc.) for the non-IP devices of the system. The DSP DTMF receivers accept DTMF digits from a touch tone phone and converts these tones into the correct digits. These digits are then passed to Call Control. The DSP also provides conference circuits

Troubleshooting Kit

The following is a recommended list of items required in the Field Service Engineering Troubleshooting Kit:

- One of each circuit card as a spare, including modules, FIMs, Bay Power Supply, Control cards
- Flash Memory cards (64 Mb)
- Fiber cable
- Butt set
- Digital multimeter
- Moving coil multimeter
- Static strap
- Ground mat
- Assorted screwdrivers, wrenches
- Long nose pliers
- Spare fuses
- Wire strippers
- Fan filter
- Breakout Box (for Data).
- Network sniffer software on laptop
- Ethernet cables and crossover cables

Loop Current Measurement

There are two methods available to measure loop current:

- In-Line method
- Voltage conversion measurement

If it is not possible to break the loop (open the Tip or Ring lead) the second method may be used; however, it is not as accurate.

In-Line Method. The procedure for the in-line method of measuring loop current is shown in this Figure and is described as follows:

1. Open either the Tip or Ring lead of the trunk facility by removing one of the bridge clips on the MDF.
2. Insert an ammeter where the clip was.
3. Take current readings at the instant the trunk is seized, and after the call has been completed. Do this in both incoming and outgoing directions.
4. Release the trunk and repeat the measurements several times on the same Central Office at peak and low traffic hours.
5. Repeat the above steps using different Central Office trunks.

Voltage Conversion Method. This method of loop current measurement should be used only in such instances when it is not desirable to open the Tip or Ring leads. The procedure is as follows:

1. With the trunk seized, use a voltmeter to measure the voltage between the Tip and Ring of the trunk.
2. This voltage is then used to calculate the loop current (see below).
3. Release the trunk and repeat the measurement several times using the same Central Office trunk at peak and low traffic hours.
4. Repeat the above measurements using different Central Office trunks.
5. Typical equivalent resistance seen between Tip and Ring, when the trunk is seized, is approximately 250 ohms.
6. The following is a simple calculation used to obtain the loop current value:

$$\text{Loop Current} = \frac{\text{measured Tip to Ring voltage}}{250}$$

Figure: In-Line Current Measurement

Loop Start Versus Ground Start Check

The loop start versus ground start check should be performed as follows (also refer to this Figure):

1. Locate the PBX trunk circuit Tip and Ring on the connecting block at the MDF.
2. Remove the bridge clips on the Tip and Ring (disconnect the PBX trunk circuit from the Central Office Tip and Ring).
3. Connect the butt set across the Central Office Tip and Ring.
4. Go off-hook with the butt set. If Central Office dial tone is returned, then the Central Office trunk is loop start.
5. If no dial tone is returned, then momentarily ground the Ring lead with a butt set off-hook and attached across Tip and Ring. If Central Office dial tone is returned, then the Central Office is ground start.
6. If still no dial tone is returned, repeat the previous step and ground the Tip lead instead. If Central Office dial tone is returned, the Central Office trunk is ground start (however, the Tip and Ring are reversed).
7. If still no dial tone is returned in either of these cases, there is a problem (perhaps Tip and Ring opened or shorted, or there is a large ground differential between PBX ground and Central Office ground.) Check PBX ground. If PBX ground is good, then report the problem to telephone company.

Note: In most cases with ground start trunks, dial tone may be returned by grounding either Tip or Ring. If this should occur, repeat the 5th and 6th steps using the butt set in its “on-hook” or “monitor” state. Hold the ground lead on for a few seconds; CO should return dial tone as long as the ground lead is connected.

Figure: Loop Start Versus Ground Start Check

Mitel IP Phone Analyzer

The Mitel IP Phone Analyzer is a Windows application that collects performance information from IP Phones within the network. It allows the administrator to use one PC to monitor the status of all IP phones on the system. IP Phones within a network send debug, status, and statistical information to the IP Phone Analyzer if the application is enabled.

The application is provided on the SX-200 ICP software CD and installs on any PC or laptop running Windows 98, NT, 2000 or XP.

Installing the IP Phone Analyzer

1. Insert the SX-200 ICP software CD-ROM in the PC's CD drive.
2. Open the Tools folder, and then the IP Phone Analyzer folder.
3. Double-click Setup.exe. Follow the prompts to complete the installation.

Programming the SX-200 ICP to recognize the IP Phone Analyzer

- In Form 47, Subform 02 (DHCP Subnet Opt) add Option 131: IP Phone Debug Window IP to the list of Common Options.

  For the IP address, enter the one belonging to the PC that has the Analyzer installed on it.

Launching the IP Phone Analyzer

1. Click Start on the Windows taskbar.
2. Point to Programs.
3. Click Mitel IP Phone Analyzer.

For detailed information on using the IP Phone Analyzer, refer to its Online Help.

Enabling Tool Analysis

From the IP Phone Analyzer Tool:

1. Select Commands, and then Register Set.
2. Enter the IP address of the IP phone.

From the IP phone:

- Reboot the phone (by removing and restoring power) to add the IP address of the PC to the phone.

  The IP address will appear in the IP Phone Analyzer Status View window.

  The phone will be out of service while it resets.

Disabling Tool Analysis

From the PC hosting the Analyzer Tool:

- Access the Status View window, left-click on the IP address, right-click, and then select Delete.

From the CDE/Maintenance Terminal:

- In Form 47, Subform 02 (DHCP Subnet Opt), delete Option 131 to disable IP messaging to the Analyzer Tool.

  There will be a service outage while the phones reset.

From the IP phone:

- Restart the set to clear PC's IP address from the phone.

Maintain

Maintaining the System

Upgrading SX-200 ICP Controller Software

You can upgrade SX-200 ICP Controller software:

- On-site by upgrading using the External CompactFlash Card, or
- Remotely by loading new software from an FTP server, or

Note: An FTP server can be used on-site as well (if, for example, a CompactFlash writer is not available).

- From Release 1.x to Release 2.0 or later, or
- By migrating from an SX-200 ML/EL system to an SX-200 ICP, or
- From an SX-200 Light system to an SX-200 ML/EL (required for subsequent upgrades).

IMPORTANT: The upgrade procedure will install new software in the SX-200 ICP. The currently installed database is not changed. A typical upgrade requires approximately five minutes to complete once the new software is installed and the system rebooted. Add another five minutes if installing additional voice mail language files, another 10 minutes if the upgrade includes a new load of AMB software, and up to 30 minutes if upgrading BCC III.

Upgrading to Release 5.0 or Higher from Previous Releases

If you are planning to use English Overlaid prompts, your upgrade procedure may vary depending on your system configuration.

The following tables outline upgrade conditions.

Upgrading from Release 2.x to Release 5.0 or Higher

Your current system:	**When upgrading to Release 5.0 or higher...**
this method will install software WITH 5540 support & English Overlaid prompts	**this method will install software WITHOUT 5540 support & English Overlaid prompts**
Release 2 with internal hard drive	

- Initial install OR

• Local upgrade • **FTP upgrade not supported**	N/A
Release 2 with 512 MB Flash	

- Initial install OR

• Local upgrade (see Note) • **FTP upgrade not supported**	N/A
Release 2 with 256 MB Flash	• 5540 not available with 256 MB flash

• Initial install

• Local install • **FTP upgrade not supported**

Note: If your voicemail storage is excessive, there may not be enough room to install the 5540 software or English Overlaid prompts. The upgrade procedure provides a warning message regarding the 5540 software and will be suspended in the case of English Overlaid prompts.

Upgrading from Release 3.x/4.x to Release 5.0 or Higher

Your current system:	When upgrading to Release 5.0 or higher...
this method will install software WITH 5540 support & English Overlaid prompts	**this method will install software WITHOUT 5540 support & English Overlaid prompts**
Release 3 or 4 with internal hard drive	

• Initial install OR

• Local upgrade OR • FTP upgrade (see Note 1)	• FTP upgrade
Release 3 or 4 with 512 MB Flash	

• Initial install OR

• Local upgrade (see Note 2) OR • FTP upgrade (see Note 1)	• FTP upgrade
Release 3 or 4 with 256 MB Flash	• 5540 not available with 256 MB flash

• Initial install OR

• Local install OR • FTP upgrade

Notes:

1. The FTP upgrade must be done twice to deliver:
   - 5540 IP Console support (Release 5.0)
   - English Overlaid prompts (Release 5.0 UR1)
2. If your voicemail storage is excessive, there may not be enough room to install the 5540 software or English Overlaid prompts. The upgrade procedure provides a warning message regarding the 5540 software and will be suspended in case of English Overlaid prompts.

Upgrading to Release 4.0 from Previous Releases

If you are upgrading to Release 4.0, and plan to use Mitel 53xx IP phones, you need a 512MB internal flash or a hard drive. If you do not plan to use Mitel 53xx phones, you can still load the Release 4.0 software using a 256 MB flash. You will receive a warning stating that the 53xx software load will not be included.

Note: Your upgrade procedure may vary depending on your system configuration. The following tables outline upgrade conditions.

Upgrading from Release 2.x to Release 4.0 or Higher

Your current system:	When upgrading to Release 4.0...
this method will install Release 4.0 software WITH 53xx support	**this method will install Release 4.0 software WITHOUT 53xx software:**
Release 2 with internal hard drive	

- Initial install OR

• Local upgrade • **FTP upgrade not supported**	NA
Release 2 with 512 MB Flash	

- Initial install OR

• Local upgrade (**See Note 2**) • **FTP upgrade not supported**	NA
Release 2 with 256 MB Flash	• Not available with 256 MB Flash

- Initial install

• Local install • **FTP upgrade not supported**

Note 2: If your voicemail storage is excessive, there may not be enough room to install the 53xx software. The upgrade procedure provides a warning message in this case.

Upgrading from Release 3.x to Release 4.0 or Higher

Your current system:	When upgrading to Release 4.0...
this method will install Release 4.0 software WITH 53xx support	**this method will install Release 4.0 software WITHOUT 53xx software:**
Release 3 with internal hard drive	

- Initial install OR

• Local upgrade OR • FTP upgrade (**see Note 1**)	• FTP upgrade
Release 3 with 512 MB Flash	

- Initial install OR

• Local upgrade(s**ee Note 2**) OR • FTP upgrade (**see Note 1**)	• FTP upgrade
Release 3 with 256 MB Flash	• Not available with 256 MB Flash

- Initial install OR

• Local install OR • FTP upgrade

Note 1:The FTP upgrade must be done twice to deliver 53xx set support.

Note 2: If your voicemail storage is excessive, there may not be enough room to install the 53xx software. The upgrade procedure provides a warning message in this case.

Upgrading using the External CompactFlash Card (Release 2.0 or later)

Note: For important information about upgrading an AX Controller, see the *SX-200 CX/CXi and AX Technician's Handbook* .

Before You Begin

Installer's PC requirements

- Windows NT 4.0, Windows 98, Windows 2000, Windows XP operating system
- For serial connections, a VT100 terminal emulation program such as HyperTERM
- For remote or LAN-based connections, a secure Telnet client that supports SSL/TLS (Mitel Telnet client version 1.0.0.1 or later recommended), or a web browser (Internet Explorer 6 or Mozilla Firefox) to access the SX-200 ICP Web Interface
- A CompactFlash Card Reader with Read/Write capability
- OPTIONAL: A serial connection to the SX-200 ICP Controller Maintenance port

You also need

- SX-200 ICP software CD-ROM

To upgrade using a CompactFlash Card

Notes:

- An optional precaution before initiating this procedure is to back up the database. This is not required to perform the upgrade but may be useful in case of any failures or problems.
- This procedure is intended for systems running Release 2.0 or later software. To upgrade a system running Release 1.x software, see Upgrading from Release 1.x to Release 2.x.

1. Remove the external CompactFlash card from the controller (if installed).

WARNING: Before removing the CompactFlash card, verify the status LED next to the card slot is solid green.

2. Insert the CompactFlash card into the CompactFlash Reader/Writer on your PC.

IMPORTANT: Use only Mitel-supplied SanDisk brand CompactFlash cards. 256 MB is the minimum required card size. DO NOT partition nor copy files to the card before proceeding with the software installation.

3. Insert the Mitel SX-200 ICP software CD-ROM into the CD drive of the installer's PC or access the file from Mitel Online.
4. Run the SX-200 ICP SETUP program from the CD.
5. Press Next, then Yes to the terms of the licensing agreement.
6. Select Local Upgrade [CompactFlash Card] Installation.
7. Click Next.
8. English language voice mail prompts will be installed. If additional languages are required, select the appropriate checkbox(es). Check your FOR options for System Option 121.

   Note: Please select the same languages that are already currently installed on the system.
9. Select the CompactFlash Reader/Writer drive.
10. Select Format to format the CompactFlash card, then click Next.

    Note: When formatting the CompactFlash card, specify FAT as the file system.
11. Click Finish when prompted.
12. Wait for the flash card writer to finish writing to the card before ejecting it.

    Note: To avoid ending the writing operation too soon, DO NOT click STOP in the taskbar Hardware Removal Wizard prior to EJECT.
13. Right-click the icon for the flash card writer in Windows Explorer and select EJECT or use the software that accompanied the flash card writer (for example, SafeSwap is provided with SanDisk writers) to complete any pending writing operations.
14. Remove the CompactFlash card from the flash card writer and insert it into the card slot on the front of the controller.
15. Log into the Maintenance Terminal and begin a Maintenance session.
16. Reboot the controller by pressing

    SYSTEM
    RE_START
    RESET SYSTEM
    CONFIRM

    The system boots from the CompactFlash card and installs the new software. Installation is complete when the LED next to the card slot turns green and the phones come up.

    Note: Once the status LED for the CompactFlash card is solid green, the card can safely be removed.
17. When prompted, log into CDE the Maintenance Terminal in CDE mode and program purchased options from the FOR sheet in Form 4.

Upgrading Remotely by FTP

Note: For important information about upgrading to Release 4.0 or later with 5330/5340 phones, see the *Upgrading the System Software* in the Technician's Handbook.

Before beginning the upgrade, program your FTP server information in Form 47 (IP Networking), Subform 01 (System IP). You need the FTP server IP address, Username and Password.

Note: Immediately after restarting the system, wait until all startup processes have been completed before initiating an FTP upgrade. Initiating an FTP upgrade before all startup processes have been completed (for example, voice mail) may interfere with the system boot procedure.

Prior to updating the software, verify that the following requirements have been satisfied:

- External flash card is inserted into the SX-200 ICP (Only required when upgrading Release 1.x systems. The card should be removed following an upgrade to Release 2.0 or later. For more information, see Upgrading from SX-200 ICP Release 1.x to Release 2.x)
- existing database is backed up to the PC
- Remote Software Download option (System Option 109) has been enabled
- FTP software has been installed on the PC and a username and home directory has been specified

To upgrade a CX/CXi from a PC located on the Internet, verify the following additional requirements have been satisfied:

- PPTP Client information has been programmed on Form 47, Subform 07 (PPTP Remote Access)
- FTP Server information has been programmed on Form 47, Subform 01 (System IP)
- PPTP Client on the remote PC has been programmed to connect to the WAN interface of the CX/CXi

Note: Allocate an IP address on the LAN to the PPTP client. Program this address as the FTP Server IP on Form 47, Subform 01, and as the PPTP LAN Client IP on Form 47, Subform 07.

The upgrade takes about 3-4 minutes depending on the speed of the FTP connection. Double the time if installing additional languages for voice mail prompts.

To upgrade the SX-200 ICP Controller software from an FTP server:

1. Run the SX-200 ICP SETUP program from the software CD or on-line source.
2. Select Remote Upgrade (FTP) Installation, and then click Next.
3. Select languages if applicable. The default is English and is not selectable. (Check your FOR options for System Option 121 - Voice Mail License for Bilingual Prompts).
4. Select the FTP server home directory, then click Next twice to begin the installation.
5. Click Finish when prompted.
6. If upgrading a CX/CXi from a remote PC on the Internet, start a PPTP session between the PPTP client (the remote PC) and the server (the CX/CXi).
7. Connect to the controller using a secure Telnet client or the SX-200 ICP Web Interface.
8. Log into the system and begin a Maintenance session.
9. Press the SYSTEM softkey.
10. Press the DATABASE softkey.
11. Press the UPGRADE_SW softkey.
12. When prompted, press ENTER to begin the upgrade.
    Files from the FTP server begin transferring to the SX-200 ICP.
13. If the upgrade is unsuccessful, you see the message, "Upgrade Failure. Consult the Maintenance Logs for more information." Use the READ Maintenance command to view the logs.

Upgrading from SX-200 ICP Release 1.x to Release 2.0 or later

System software is stored on an external flash card for SX-200 ICP Release 1.x systems but this changes to the internal media (flash or hard drive) for Release 2.0 or later. As a result of this change, it is necessary to repartition the internal media by installing an "Initial" software load using a CompactFlash card (not FTP). The upgrade process does not alter the currently installed database.

For details on upgrading system software for Release 4.0 and 5.0, refer to the *Technical Handbook*.

Note:

1. Upgrades that require repartitioning cannot be performed by FTP because the FTP upgrade procedure does not support repartitioning.
2. Phantom bays programmed on a Release 1.x system should be de-programmed before migrating to Release 2.x.

To upgrade a controller from Release 1.x to Release 2.0 or later:

1. Back up the database.
2. Unplug the power cord from the controller.
3. Remove the external CompactFlash card.
4. Perform an Initial CompactFlash Card Installation.
5. Insert the CompactFlash card that contains the Initial software load into the slot on the front of the controller.

IMPORTANT: Use a Mitel-supplied SanDisk brand CompactFlash card. 256 MB is the minimum required card size. If you plan to use the full feature set of the Mitel 5330/5340 IP phones, minimum required size is 512 MB.

6. Power up the controller.
7. Wait for the CompactFlash status LED to turn GREEN, and for the phones to come up, indicating that the upgrade is complete.
8. Remove the CompactFlash card.
9. Update the IP settings to enable the database to be restored from the FTP server.
10. Restore the database.

    After the database file is restored. the system automatically resets and reboots.
11. If the system generates an alarm:
    - In Form 53, check the bay number assignments. If a bay is assigned a bay number, it must be connected to a physical device; otherwise, it will generate an alarm. To prevent this problem, move disconnected bays from physical connections (CIMs and MMCs) to phantom bays.
    - In Maintenance, revise the Alarm Thresholds to prevent unnecessary alarms.

Migrating an SX-200 EL/ML System to an SX-200 ICP MX

When migrating an SX-200 EL/ML system to an SX-200 ICP MX system, certain parts of the original system will be retained while others will no longer be used as indicated in the table below.

Table: Retained and Surplus Parts

When migrating an SX-200 EL/ML system to an SX-200 ICP MX, the following will continue to be used after the migration process is completed:	After migrating from an SX-200 EL/ML system to an SX-200 ICP MX, the following will no longer be used:
• main cabinet	• control card in main cabinet
• peripheral cabinets	• SPINE Bays
• customer database	• IP Nodes
• peripheral interface cards and modules	

- Control Dual/Triple FIM cards

• Triple CIM cards
• telephones and other peripherals (including peripheral interfaces/CIMs)

- ISDN Gateways

• COV cards/COV Vmail • SUPERSET 3 and SUPERSET 4 telephones • Datasets for PMS and ACD monitors

A typical migration will take approximately one hour to complete.

Notes:

1. This migration procedure will work only for an MX controller and not for a CX/CXi controller.
2. An SX-200 LIGHT system must upgrade to an SX-200 EL /ML (LIGHTWARE 19 Release 3.2 or later) (**Q40.0.1 UR3** or later) before migrating to an SX-200 ICP MX.
3. Mitel Express Messenger (MEM) and the embedded voice mail application in the SX-200 ICP MX are virtually identical in function. The databases, however, are not compatible; for example, an MEM database cannot be restored to the SX-200 ICP MX. This means that migrating an SX-200 EL/ML system to an SX-200 ICP will cause data stored in the MEM database (programming, greetings and messages) to be lost. If you continue to use the MEM card, the database will remain operational provided that you enable System Option 98 (Support 3DN, 4DN and 400 series Set Types).
4. Call logs will not be included in the migration and will be lost.
5. Callback requests will not be included in the migration and will be lost. If MEM is retained, Message Waiting Indicators (MWI) will not be lost.
6. The IP bay (the controller) is bay number 1 by default. It becomes bay number 8 following the migration. All references in documentation are to the default. See CDE Form 53 - Bay Location Assignment for more information.
7. The SX-200 ICP supports a maximum of 30 IP trunks. If Option 115 (Maximum IP Trunks) in CDE Form 04 - System Options/System Timers has been programmed with more than the maximum supported, only the first available 30 IP trunks will function. Previously configured settings for IP trunks will need to be removed and new settings defined for the controller during the migration process. See Trunk Support - IP for more information.
8. Migration causes phantom bays on the SX-200 EL/ML to become peripheral bays on the SX-200 ICP. You must manually re-program these bays as phantoms in CDE Form 53 - Bay Location Assignment.
9. If the SX-200 EL/ML has T1 or PRI trunks, the SX-200 ICP requires a Stratum 3 clock.

Parts Required

You require the following:

- SX-200 ICP Release 2.0 software or later
- SX-200 ICP MX
- Peripheral FIM Carrier (PFC) or Peripheral CIM Carrier (PCC) for the SX-200 cabinets
- Fiber Interface Module II (FIM II) or Cable Interface Module (CIM) with appropriate connector cables

Note: The SX-200 ICP supports 1KM, 5KM, and 14KM FIMs. Verify that the hardware revision matches on both ends of the connection between the controller and peripheral interfaces.

- Printed copies of the FOR sheets (check for IP sets and other configuration settings) for both the SX-200 EL/ML and SX-200 ICP MX.

SX-200 EL/ML Migration Procedure

Before starting the migration, survey the existing EL/ML hardware to determine any potential configuration issues such as unsupported devices. Check DSP configuration (see DSP Resources for more information) and make sure that sufficient resources are available. Triple FIMs or triple CIMs should be removed from the control cabinet before initiating this migration procedure. Ensure that you have enough FIMs or CIMs to relocate bays after the migration is completed.

CAUTION: The following procedure will take the system out of service. Perform the migration during off-hours.

Preparations

1. Compare FOR sheets for the SX-200 EL/ML system and the SX-200 ICP. Ensure that sufficient bays have been purchased as indicated under Option 133 (TDM Bays) of the SX-200 ICP's FOR sheets or the boot process may be affected. Option 102 (Feature Level) of the SX-200 ICP's FOR sheets must be set to 6 to enable SX-200 ICP 2.0 functionality. Option 98 (Support 3DN, 4DN and 400 series Set Types) must be set to Enabled if 400 series telephone sets are being used.

Note: Following the restoration of the SX-200 EL/ML database to the SX-200 ICP MX (as described in Step 6 below in this procedure), use the information on the FOR sheets (such as the Mitel Options Code Number) to resolve FOR errors.

2. Back up the database on the SX-200 EL/ML system using Kermit.

Note: This can only be carried out using the Kermit protocol. Refer to SX-200 EL/ML Technical Documentation for information on backing up a database using Kermit.

3. Print CDE Form 1 - System Configuration for reference (also print CDE Form 47 - DHCP Parameters if an IP Node is included). These can be referred to when assigning bay numbers to physical connections. Refer to CDE Form 53 - Bay Location Assignments for more information.

CAUTION: The following step takes the system out of service.

4. Power down the SX-200 EL/ML control cabinet and peripheral cabinets.

Completing the Migration to an SX-200 ICP

5. Install the new SX-200 ICP and optional modules (Stratum 3 clock, Quad CIM, Dual FIM, Dual T1/E1 Framer, etc.).

CAUTION: Ensure that the system is grounded in accordance with the installation instructions.

6. Power up the SX-200 ICP.
7. Restore the database from the SX-200 EL/ML that was backed up in Step 2 above to the SX-200 ICP using Kermit over the serial port. Another option is to select **Custom Database** during the installation of the new SX-200 ICP. Refer to Installing the System Software for more information.
8. Phantom bays on the SX-200 EL/ML become peripheral bays on the SX-200 ICP. Re-program these bays as phantoms in CDE Form 53 - Bay Location Assignments.
9. As mentioned previously, the SX-200 ICP no longer supports SPINE Bays or SX-200 IP Nodes. If the SX-200 EL/ML system includes SPINE Bays or IP Nodes, these will automatically be converted to phantom bays. Reprogram any SPINE Bays or IP Nodes as follows:
   a. In CDE Form 53 on the SX-200 ICP, create new bay number assignments for connections to which the programming data will be moved.
   b. In CDE Form 09, move each set on the SPINE bay to a PER Node and each set on the IP Node to the IP Bay.
   c. Delete all SPINE Bay or IP Node device assignments in CDE Form 18 - Miscellaneous System Ports. LS lines on SPINE Bays and IP trunks on IP Nodes should be deleted and then re-added to a new PLID.

Note: When SPINE bays and IP nodes are converted into phantom bays, any devices connected to them will no longer function unless they are moved to other peripheral bays and/or the controller.

   d. Remove ARS programming for IP Nodes - CDE Forms 26 (including subform), 22, and 23 as per the note above.
   e. In CDE Form 48, modify all IP Node programming.
10. Update CDE Form 53 - Bay Location Assignments to conform to the specifications of the new system, assigning the PER Bays to CIM or FIM ports.
11. Enter the Mitel Options Code (MOC) and Mitel Options Password (Option 100) in CDE Form 04 - System Options and Timers. Set all other options as required.

12. Reset the system (if prompted).
13. Enter the IP networking information from the IP Node into CDE Form 47 - DHCP Parameters. Ensure that all sub-forms are completed.

Note: Voice Compression in Form 47 for the IP Node is a System Option in Form 04 (Option 120, Number of Compression Resources) in the SX-200 ICP.

14. Program the embedded voice mail system with Mitel Express Messenger information (if installed in the EL/ML) or, if the MEM card is still being used, enable System Option 98 (Support 3DN, 4DN and 400 series Set Types).
15. Disconnect the fiber cables from the FIMs and/or CIMs on the SX-200 EL/ML control cabinet. Label the Tx and Rx cables, identify the Bay that the cables connect to, and label the front of the SX-200 ICP MX controller to ensure that connections are made to the proper bays and that cabling is correct.
16. Remove main control card from the SX-200 EL/ML control cabinet and replace with either a peripheral FIM or peripheral CIM card.
17. Insert peripheral CIM or FIM cards into the peripheral cabinets as appropriate for the configuration of the SX-200 ICP MX system (see Configuration Rules for restrictions that might apply).
18. Remove all CIM and FIM triple modules previously installed in Slot 10 and Slot 11 of the SX-200 EL/ML control cabinet.
19. Connect all bays to CIMs or FIMs as needed.
20. Power up what was the SX-200 EL/ML control cabinet (and is now a peripheral cabinet for the SX-200 ICP MX).

Note: A system with a BCC III may take as long as 20 minutes to come up following the upgrade. Subsequent reboots will not incur this delay.

21. Verify that the migration has been performed properly by placing calls from sets connected to different devices. Test the voicemail system by leaving, retrieving, and removing messages. Verify that features function properly and that both call forwarding and message waiting lamps are operating correctly.
22. Back up the SX-200 ICP database using FTP.
23. If the system generates an alarm:
    - In Form 53, check the bay number assignments. If a bay is assigned a bay number, it must be connected to a physical device; otherwise, it will generate an alarm. To prevent this problem, move disconnected bays from physical connections (CIMs and MMCs) to phantom bays.
    - In Maintenance, revise the Alarm Thresholds to prevent unnecessary alarms.

Upgrading an SX-200 LIGHT to an SX-200 EL System

This section provides instructions on how to upgrade an SX-200 LIGHT system to an SX-200 EL system.

Parts Required

You require the following

- LIGHTWARE 17 Release 3.1 software or later. New software is shipped on a flash memory card. Included with the card is a system ID module and your FOR sheets. You install the system ID module and flash memory card onto the Main Control IIIELx card (make sure the toggle switch on the edge opposite the connector edge of the Flash Memory Card has the Write Protect disabled).
- SX-200 EL/ML Universal Cabinet

Note: You require this control cabinet PN 9109-600-002-NA in conjunction with the MCIIIELx card to obtain three links to each peripheral bay in a 7-cabinet system.

- Main Control Card IIIELx with Stratum 3 Clock Module (9109-610-202-NA)
- Two Control Dual FIM Carrier II cards, or two Control Triple FIM Carrier cards, or one of each, depending on the number of peripheral cabinets in your system. See "Control FIM Carrier and T1 Trunk Engineering Rules" in the SX-200 EL/ML Technical Documentation for details.

Upgrade from SX-200 LIGHT

Before You Begin

If you are upgrading a system that has LIGHTWARE 16 software you can contact Mitel and arrange to have your customer database upgraded to LIGHTWARE 19 software. Note that you must purchase the CDBAR upgrade package. After you receive your upgraded customer database, you are ready to perform the software upgrade. Although the CDBAR program converts the majority of the database, you must perform some manual corrections through CDE programming (See CDBAR in the SX-200 EL/ML Technical Documentation for details).

Caution: If you do not get your customer database upgraded to LIGHTWARE 19, you will have to re-enter your database manually.

Notes:

1. Support for IP devices is defined is different ways in the SX-200 ICP and the SX-200 EL/ML. The SX-200 EL/ML supports up to 60 IP devices on an IP node. The SX-200 ICP supports up to 30 IP trunks, 192 IP sets (248 with Hospitality option), and 24 compression resources (programmed separately in FOR).
2. The SX-200 ICP may require the purchasable COS Option 98 "Support 3DN, 4DN and 400 series Set Types" to support an external DNIC voicemail system. The option is available only with the SX-200 ICP MX migration package, not with new installations.

SX-200 LIGHT Upgrade Procedure

To upgrade an SX-200 LIGHT system to an SX-200 EL system

1. Back up your customer database.

CAUTION: The following step takes the system out of service.

2. Power down the SX-200 LIGHT control node.
3. Disconnect the fiber cables from the FIMs on the rear panel of the SX-200 LIGHT control node. Ensure that you label the Tx and Rx cables. Also identify the Bay that the cables connect to.
4. Power down peripheral cabinet #1 and remove the PIC cards. Ensure that you record the slot # of each card before you remove it. See Unpack and Handle Printed Circuit Cards for guidelines on handling circuit cards.
5. Remove the Bay Power Supply and Bay Control Card from peripheral cabinet #1. Then, remove the FIM and FIM Carrier card from the Bay Control Card.
6. Ground the new control cabinet
7. Install the new control cabinet.
8. Verify the ground connection on the new control cabinet.
9. Insert the flash memory card and system ID module into the MCIIIELx card. See the SX-200 EL/ML Technical Documentation for installation instructions. (Make sure the toggle switch on the edge opposite the connector edge of the Flash Memory Card has the Write Protect disabled). The flash memory card has the new software. The system ID module and your FOR sheet was shipped with flash memory card.
10. Install the MCIIIELx card in the control cabinet.See the SX-200 EL/ML Technical Documentation for installation instructions.

11. Ensure the Bay Power Supply is seated firmly in the cabinet. Then, install the Dual or Triple FIM Carrier Cards in the control cabinet. For instructions on how to install a Dual or Triple FIM Carrier Card, see Install FIM Carrier and FIM into Cabinet.
12. Install the Bay Control Card (that you removed from peripheral cabinet #1) into slot 9 of the SX-200 EL control cabinet. Install the peripheral interface cards (from peripheral bay #1) into slots 1 to 8 of the SX-200 EL control cabinet. Install the cards in the same slots (for example, if a DNIC card was in slot 4 of the peripheral cabinet, install it in slot 4 of the new control cabinet). See Installing and Replacing Circuit Cards and Modules.
13. Complete the cabling and cross connections for the PICs in the SX-200 EL control cabinet.
14. Connect the fiber cables to the Control Triple or Control Dual FIM Carrier Cards. Ensure that you connect each FIM in the control cabinet to the corresponding FIM in the associated peripheral bay (see Triple FIM Bay Assignment or Dual FIM Bay Assignment). See Fiber Cable Installation for more information.
15. Power up the control cabinet. See the SX-200 EL/ML Technical Documentation for power-up instructions.
16. Initialize the system and enable your options. See the SX-200 EL/ML Technical Documentation for system initialization procedures.
17. Program the FIM assignment in CDE Form 4 - System Options. In Option 71 specify the number of bays that are supported by the FIM Carrier card (Control Dual or Control Triple FIM Carrier in slot 10. Confirm. In Option 72, specify the number of bays that are supported by the FIM Carrier card in slot 11. Confirm. Enter to save these options. A message will appear saying that a reset is required. Select Confirm to reset the system.
18. Re-enter your customer database.

Note: If Mitel upgraded your database to LIGHTWARE 18 you can restore your upgraded database.

19. Back up the SX-200 EL system customer database.

    The upgrade is complete.

Backing up an SX-200 LIGHT/Digital Customer Database

Back up the database during a low traffic period. Only the database is copied; the diskettes that are being written to must already have the system software stored on them.

To backup an SX-200 LIGHT customer database

1. From the Maintenance screen press
   SYSTEM
   COPY
   DATABASE
   The system verifies the data and prompts you to insert disks into drives A and B.
2. Depress the eject button on drives A and B and remove the old disks and insert the new disks (backups). Press CONTINUE.

   The system notifies you when the backup is complete.
3. Select CONTINUE.

Powering Down an SX-200 LIGHT Control Node

To power down an SX-200 LIGHT Control node:

CAUTION: The following procedure takes the system out of service.

1. End any customer data entry sessions.
2. If you don't have an up-to-date database backup, perform a database backup.

3. Reset the MCC by pressing the MCC System Reset button.
4. Depress the eject button and remove the disks.
5. Turn off the ac power switch on the rear panel of the cabinet.

Setting up an FTP Server

An FTP server is required to back up all SX-200 ICP configuration data and voice mail messages. The server is also used to download new software to the SX-200 ICP controller through its built-in FTP client and to upload Maintenance logs.

Any FTP server application designed for the Windows environment will work. Windows 2000 and XP have one built into them. Others can be downloaded for free from the Internet.

Server setup varies by vendor; the basic steps are provided below. For specific instructions, see the vendor's documentation.

The PC hosting the FTP server must connect to the SX-200 ICP through a TCP/IP (LAN) connection. Connecting through the serial Maintenance port on the controller will not work.

To set up an FTP server:

1. Create a directory (or directories) on the PC to hold the files you will transfer to and from the SX-200 ICP—example,

   C:\FTPdir\backups
   C:\FTPdir\software
   C:\FTPdir\logs

   Note: Ensure that folders are writable.
2. In the server application,
   - Create a User for password-protected logins or allow Anonymous (no password required) logins.
   - Set up the paths to the directories you created in step 1.
   - Enable read/write access to directories.
3. Restart the server.

Tip: To verify that the FTP Server works, log into it from the PC. Go to the CMD prompt (DOS) and enter ftp < IP Address of the FTP Server >. Look for the message "Anonymous user logged in" or a prompt to enter a user name.

To set up the SX-200 ICP to connect to the FTP server:

1. In Form 47 (IP Networking), Subform 01 (System IP), enter the
   - FTP Server IP address
   - FTP Server Username
   - FTP Server Password

TIP: If future attempts to connect to the FTP server fail, check the IP address of the PC to see if it has changed. Follow the steps above to reprogram the system with the new address.

Replacing IP Phones

To replace a registered IP phone, you can

- Replace it and retain all existing programming for that circuit, or
- Replace it and re-program the circuit.

Note: 5201, 5215, and 5010 IP phones will fail to register on a system that has a Default or Premier database (MX only) because of the line appearances programmed on keys 8 and 10—keys that exist on the 5207 but not on the 5201, 5215 or the 5010. To register these phones, first delete the line appearances in Form 09 (the 5201 must have all appearances deleted) or follow the procedure for replacing a registered phone and re-programming the circuit.

To replace a registered IP phone and retain all programming for that circuit

1. Access Form 09, Desktop Device Assignments.
2. Press Bay/Slt/Cct.
3. Enter the numbers for the Bay, Slot, and Circuit of the IP Line that you want to program.
4. Press Show CESID.
5. Press Show MAC.
6. Select the Circuit of the IP telephone that is to be replaced.
7. Delete the existing MAC Address.
8. Disconnect the old IP phone from the LAN.
9. Connect the new IP phone to the LAN.
10. Register the new IP phone.
11. Press Quit to exit CDE Form 09 and save your changes.

To replace a registered IP phone and re-program the circuit

1. Access Form 09, Desktop Device Assignments.
2. Press Bay/Slt/Cct.
3. Enter the numbers for the Bay, Slot, and Circuit of the circuit that you want to program.
4. Delete the Extn number of the IP phone that you want to replace.

WARNING: If you delete the Extn number of a phone, all programmed data for that circuit is deleted.

5. Assign an IP phone to the circuit in Form 01.
6. Program the features for the IP phone.
7. Register the IP phone.
8. Press Quit to exit CDE Form 09 and save your changes.

Remove/Replace the Controller Cover

MX Controller

To remove the cover

1. Unplug the power cord and disconnect all cables.
2. Remove the controller from the rack or wall and place it on a suitable work area.
3. Remove the four screws from the top of the controller.
4. Slide the cover forward until it catches, then tilt the cover upward to remove it.

5. Remove the front faceplate by clipping off from the bottom of the unit. (It may be easier to pry the end off first, and then slide your fingers along the bottom edge of the faceplate to the other end.)

To replace the cover

1. Turn the controller until the back panel is facing forward.
2. Lift the lock for the AC power cord and place the shell at an angle to hook onto the back of the unit.
3. Straighten and slide the cover forward as far as it will go.
4. Secure the shell by inserting and snugly securing the two screws on the back panel.
5. Rotate the controller until the front panel is facing forward.
6. Secure the screws on the top of the unit.
7. Clip on the front face-plate taking care not to damage the protruding FIM connectors.
8. Reinstall the controller on the wall or in the rack (if applicable).
9. Reconnect all cables.
10. Power on the unit.

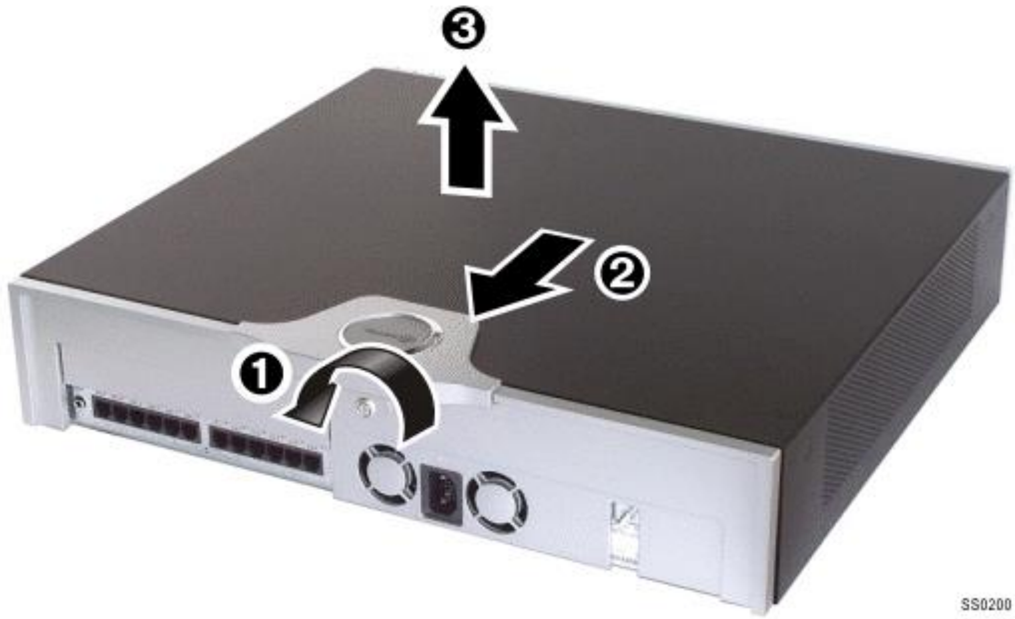
CX/CXi Controller

To remove the cover

1. Unplug the power cord and disconnect all cables.
2. Remove the controller from the rack or wall and place it on a suitable work area.
3. Loosen the screw at the back panel of the controller (1).
4. Grip the by the sides and slide it backward until it catches (2).

IMPORTANT: Once the cover of the CX has been slid back, do NOT lift the back off using the handle. Lift the cover vertically (NOT at an angle) to avoid component damage.

5. Lift the cover straight up to remove it (3).



To replace the cover

1. Turn the controller until the front panel is facing forward.
2. Align the pins inside the cover with the corresponding slots on the controller, and then lower the cover to seat it.
3. Slide the cover toward the front of the controller as far as it will go.
4. Tighten the screw on the back panel.
5. Reinstall the controller on the wall or in the rack (if applicable).
6. Reconnect all cables.
7. Power on the unit.

Replace the Hard Drive or internal CompactFlash card

See the Replace the Hard Drive or Internal CompactFlash Card procedure.

Change or Enable System Options

About the Feature Options Record Sheet

The Mitel Feature Options Record (FOR) sheet lists the options that you have purchased from the Mitel Order Desk. The FOR sheet also provides a password that you need to enable your options, a Mitel Options Code (MOC) that automatically enables all of the options and timers for the system, and a System Identity Code that matches the code programmed into the System ID Module (MX) or System i-Button (AX/CX/CXi) located on the main board of the controller.

Notes:

1. The System ID Module and System i-Button are factory-programmed by Mitel. In the event of a non-recoverable system failure, the Module or i-Button can be moved to a new controller along with the internal media (CompactFlash or hard drive). The new controller can then be started with the same settings, database, and options as the original system.

2. Attempts to enable unpurchased options causes the system to respond with PASSWORD/OPTIONS CONFLICT -- "QUIT" TO EXIT -- "ENTER" TO RE-EDIT. Conflicts are resolved by entering the correct password; a system reset is not required.
3. The system warns if changing an option requires a reset. The reset is automatic and occurs when the change is confirmed.
4. If a database from another system is installed in the controller, the System ID and Password will no longer match. Phone service will be lost (some phones may appear to be in service, but will display SYSTEM BUSY when they go offhook) and a MOSS alarm message will display in the CDE forms header. Enabling the options using the above procedure clears the alarm and restores phone service.

CAUTION: To change system options, users must purchase the option and its associated password through their usual ordering channels prior to attempting to install or implement the option.

CAUTION: A system reset occurs when options are changed.

Options are Password Protected

Options are password protected. The options password allows you to enable the options that are listed on this sheet.

Enabling the Options

1. Select Customer Data Entry mode.
2. Select Form 04, System Options.
3. Press the ENTER MOC softkey and enter the Mitel Options Code from the FOR Sheet.
4. Confirm that the System Identity Code (Option 101) matches the System Identity Code from the FOR sheet.

Note: If the System ID Module or i-Button is not installed, or if invalid software is loaded into the system, the default System Identity Code (65535 for MX) appears in Form 4.

5. Enter the Mitel Options Password (Option 100) from the FOR sheet; press ENTER twice to activate (commit) the purchased features.
6. Set all other options as required. Press ENTER only once after each entry.
7. Select Confirm if prompted to reset the controller.
8. After the system resets, go into Maintenance and revise Alarm Thresholds to prevent unnecessary alarms.

Using the Maintenance Terminal

Introduction

This section is intended to help maintenance personnel use the extensive built-in maintenance features of the SX-200 ICP. Basic maintenance functions can be performed by using either a terminal, or a PC. A PC with an FTP server installed must be used to perform the Database Backup and Restore functions. Maintenance logs can be retrieved either via FTP or a communications package, such as Hyperterminal that supports the Kermit protocol.

The maintenance information provided by this section includes how to:

- set up and use the RS-232 Maintenance Terminal, the prime maintenance tool on the system

- set up an FTP server
- use a PC and FTP server to back up and restore the database information
- use the Logs function to copy all maintenance log entries into a text file by using a PC. The saved text file can be viewed on the PC
- display the current Kermit parameters by using the Show Parameters function; change their default values by using the Set Parameters function
- use the Logs Backup function to copy all maintenance log entries into a text file by using a PC. The saved text file can then be viewed on the PC.

The maintenance terminal is also used for customer data entry (CDE) and Traffic Measurement. Maintenance terminal functions are also accessible from the attendant console. Because the scope of this section is primarily maintenance-related, refer to the Program CDE section, and the Traffic Measurement section for information on these topics.

Setting Up

Terminal Type

The SX-200 ICP maintenance subsystem is designed to interface with most 80-column terminals having an RS-232 type interface. The terminal may be either a video display terminal or a hard copy teleprinter. For ease of operation, a video display terminal capable of using the ANSI X3.64-1977 special character set for special graphics (that is, VT100 compatible) is preferred. It is recommended that you use a personal computer with a VT100 emulation package. The system prompts the user to specify the terminal type at the start of the login sequence (see paragraph following Login Procedures heading). To ensure compatibility with the maintenance subsystem, the terminal should be set up for the following data characteristics:

- 8 data bits
- 1 stop bit
- Space parity.

The terminal's baud rate must be set to 9,600.

Connecting the Terminal to the System

A standard RS-232 cable is connected to the main RS-232 communication port on the terminal. The other end of the cable is connected to the 9-pin RS-232 cable connector (labeled "Maintenance") on the front of the SX-200 ICP controller.

There are four methods of connecting a remote terminal: 1) by a secure Telnet client (Mitel Telnet client recommended), 2) by the SX-200 ICP Web Interface; 2) by dialing up to an autoanswer modem connected to a dedicated network trunk (direct access), or 3) by dialing up to an autoanswer modem connected to an ONS port (dial-up access) on an as-required basis.

Notes:

1. Do not connect more than one maintenance terminal to the system.
2. If the RS-232 cable for the maintenance terminal uses a standard 25-pin connector, you will need a 25-pin to 9-pin connector adapter.
3. A remote maintenance terminal provides a means of performing maintenance checks (logs and alarms), as well as Customer Data Entry, without visiting the customer site. It requires a modem to be permanently connected to the maintenance port.

Connecting a Printer to the System

If required, a printer (or any other ASCII output device) may be connected to the system by means of a standard 25-pin or 9-pin RS-232 cable. One end of the RS-232 cable is connected to the RS-232 port on the printer, and the other end is connected to the 9-pin connector (above the maintenance terminal connector) on the rear panel of the cabinet. Set up the printer for the following data characteristics:

- 7 data bits
- 1 stop bit
- Space parity.
- 1200 baud.

Note: The baud rate of the printer port may be changed via a command from the maintenance terminal.

Login Procedures

With the terminal powered-up, or reset, the system is ready for the user to log in. Press the RETURN key four times to initiate the log in procedure. The system will query the user for the terminal type as follows:

1 - VT100 COMPATIBLE
2 - TTY TYPE
3 - IBM PC
SELECT A TERMINAL TYPE :

If the terminal is capable of using special graphic characters, enter 1; if not, enter 2. The system will then query the user for the type of action or application intended; i.e., Maintenance or Customer Data Entry (CDE):

1 - MAINTENANCE
2 - CDE
6 - QUIT
SELECT AN APPLICATION (OR QUIT TO START OVER) :

If the maintenance system (or CDE) is being accessed by another terminal or an attendant console, one of the following messages will be returned after the RETURN key is pressed four times:

MAINTENANCE or CDE in use by Console Ext 1234.
Please Try Again Later.

OR

telneted: Maximum number of connections exceeded
Please Try Again Later.

This message is displayed because only one user can access maintenance or CDE at any one time. Assuming that there are no users currently logged in, the system will return the username prompt after an application number (1 or 2) is chosen:

ENTER USERNAME :

There are five levels of system access priority available when logging-in to the maintenance terminal. Each level has its own username and corresponding command privileges. The usernames in descending order of priority are:

- INSTALLER
- MAINT1
- MAINT2
- SUPERVISOR
- ATTENDANT.

Respond to the username prompt by entering one of these usernames. The system will then query the user for a password:

ENTER PASSWORD :

The system database contains one default password for all usernames. Passwords may be changed as required (see ”Setting Password”). The default password for all users is “1000”. Observe that for security reasons the system does not echo the password back to the terminal. If the password is accepted, the system will prepare to set up the maintenance screen.

Logout Procedures

To ensure the security of the system, use the logout procedure whenever the maintenance terminal is to be left unattended. To log out, the user presses the QUIT softkey to get out of the maintenance application. At this point, the application prompt is returned:

Enter “6” from the Main Menu to log out; or
the system will log out when the 10-second logout timeout is reached.

Figure: Maintenance Terminal Connections for the SX-200 Cabinet

Maintenance Command Input

Maintenance Terminal Display

The maintenance display screen is shown in the Top Level Maintenance Terminal Screen Layout Figure. There are five distinct and separate areas to the screen:

Status Line: -occupies a single line above the bordered area, and displays the time and date, and the system alarm status.

Header Line: -occupies the top line within the bordered area. It identifies the running software version, and its creation date. It also describes which MAINTENANCE menu is currently being displayed: one of Main Menu, System, Diagnostics, Traffic Measurement, Logs, or Reports.

Applications Area: -occupies the next 12 lines in the bordered area. Output information resulting from command input is displayed in this area.

Command Line: -occupies the line directly below the Applications Area. Commands are echoed onto this line as they are input by the user. Responses to command input (other than data; e.g., error messages) are also returned here.

Softkey Area: -changes dynamically with the MAINTENANCE mode (System, Diagnostics, To CDE, Traffic Measurement, Logs, ACD Reports, or Reports) and identifies the functions of the maintenance terminal’s 10 softkeys. The softkey area occupies the bottom two lines of the bordered area: the first line identifies the functions of softkeys 1 through 5; the second line identifies the functions of softkeys 6 through 10.

Softkey Presentation

All commands are entered using softkeys. The functions of softkeys change to suit the programming requirements of each particular application. The maintenance terminal has 10 such keys: they are the number keys (1 through 0) on the terminal keyboard. The “1” key corresponds to softkey 1 in the softkey area; similarly, all other numeric keys correspond to softkeys.

Figure: Top Level Maintenance Terminal Screen Layout

Entering Commands

Enter commands by pressing the desired softkeys in sequence, and terminate each command sequence with softkey 0, the ENTER softkey, or the conventional keyboard RETURN key. As softkeys are pressed, they are displayed on the command line. After the ENTER softkey is pressed, the command is processed by the maintenance system, and the appropriate response is returned. Press the QUIT softkey to end the current operation and return to the previous level of access.

Incorrect Command Entry

The user interface provides a comprehensive set of error messages to inform the user of incorrect command entry. While the softkey-oriented command input interface minimizes the chance of incorrect command entry, error messages provide concise descriptions of the input error. A summary of these error messages with descriptions is in the ACD Messages section.

Non-VT100 Compatible Terminal Use

When using a terminal not compatible with a VT100 terminal, softkeys are presented as described in the Top Level Maintenance Terminal Screen Layout Figure, but without the graphic bordering. Similarly, commands are entered in exactly the same manner. Instead of a Title line, the current menu is identified by the command input prompt:

- SYS> - System level menu
- DIAG> - Diagnostics level menu
- TRAFF> - Traffic Measurement menu
- LOGS> - Logs level menu
- REP> - Reports level menu.

Device Number Parameters

Some commands require the inclusion of card/circuit location numbers (referred to as physical location identification numbers: bay number, slot number, circuit number, sub-circuit number) or extension numbers as part of the input. The user is prompted for these numbers, one at a time, on the command line of the screen:

enter Bay then press RETURN:
enter Slot then press RETURN:
enter Circuit then press RETURN:
enter Sub-circuit then press RETURN:

or

enter Ext. Number then press RETURN:

When these prompts appear, the softkeys are disabled. Enter the required numbers in the conventional manner by using the keyboard number keys and pressing the RETURN key after each entry. Note that if a 2-digit number is entered, the RETURN key is not required after each digit. When all of the required device numbers have been entered, the appropriate softkeys will again be presented.

When entering circuit location numbers, the sub-circuit qualifier is often not required, and this prompt may be answered by simply pressing the RETURN key. The only devices that require sub-circuit numbers are Digital Line Card circuits and Universal Card modules such as DTMF/Receiver Modules and Music-on-Hold/Pager Modules.

Wild Card Characters

Default wild card characters may be used to perform some command-initiated functions on a range of devices, by not specifying circuit location numbers when prompted (pressing only the RETURN key). For example, entering Bay 1, but not specifying the slot, circuit, or sub-circuit would translate to “all circuits on all cards in Bay 1”. Default wild card characters do not apply to all commands; refer to the individual command descriptions.

Canceling a Command

The user may cancel any command at any point before pressing the ENTER softkey by pressing the CANCEL softkey. Any softkeys that were entered and echoed back onto the command line are now canceled, leaving the command line empty and ready for new command input.

Command Line Correction

The user may correct a current command input line before pressing the ENTER softkey, without having to cancel and enter the command over again. Press the DELETE key to delete the most recently entered softkey or device number.

System Level Functions

The system level of operation contains commands that are not necessarily maintenance applications, but that affect maintenance in some way (for example, setting of time, date, and passwords). To access system level commands, press the SYSTEM softkey. All of the following operations are done while in the system level. The table below provides a quick reference for all operations available in system level functions, except CANCEL and ENTER. To exit the current operation without committing (saving) any changes, press the CANCEL softkey at any time. To commit changes, press the ENTER softkey when it is available.

System Level Functions Table

Table: System Level Functions

LEVEL	COMMAND	PARAMETER	QUALIFIER	QUALIFIER	QUALIFIER	QUALIFIER
1-SYSTEM	1-SET	1-DATE [dd/mm/yy]				
		2-TIME [hh:mm]				
		3-PASSWORD	1-ATTENDANT			
			2-SUPERVISOR			
			3-MAINT2			
			4-MAINT1			
			7-INSTALLER			
		4-SPEED	1-MAINT-PORT [speed]			
			8-PRINTER_PORT [speed]			
		8-RESET_TIME	1-AFTER_N_FLTS			
			2-DAY/TIME	1-MONDAY	1-TIME [hh:mm]	
				2-TUESDAY	1-TIME [hh:mm]	
				3-WEDNESDAY	1-TIME [hh:mm]	
				4-THURSDAY	1-TIME [hh:mm]	
				6-FRIDAY	1-TIME [hh:mm]	
				7-SATURDAY	1-TIME [hh:mm]	
				8-SUNDAY	1-TIME [hh:mm]	
				9-DAILY	1-TIME [hh:mm]	
			3-IMMEDIATELY			
		9-ALARM_THRESH	1-LINES	1-SYSTEM	8-CONFIRM	
				2-BAY NOTE: User must	8-CONFIRM	

Table: System Level Functions

LEVEL	COMMAND	PARAMETER	QUALIFIER	QUALIFIER	QUALIFIER	QUALIFIER
				enter Bay number		
			2-TRUNKS	1-SYSTEM	8-CONFIRM	
				2-BAY NOTE: User must enter Bay number	8-CONFIRM	
			3-RECEIVERS	1-SYSTEM	8-CONFIRM	
				2-BAY NOTE: User must enter Bay number	8-CONFIRM	
	2-SHOW	1-DATE				
		2-TIME				
		4-DEVICE	1-ASU INFO			
			2-RM INFO			
			4-DATASTN_PLID	1-BAY/SLOT/CCT		
				3-EXT-NUM		
			8-PRINTER_PORT			
		7-LN_APP_COUNT				
		8-RESET-TIME				
		9-IDENTITY				
	3-DATABASE	1-BACKUP	1-KERMIT			
			3-FTP_SERVER			
		2-RESTORE	1-KERMIT			
			3-FTP_SERVER			
		4-SHOW_PARAM				
		6-TRAP_BACKUP*				
		7-UPGRADE_SW				
		8-LOGS_BACKUP				
		9-SET_PARAM	1-DEFAULT			
			2-RETRIES			
			3-BLOCK_CHECK			
			4-RECEIVE	1-EOL_CHAR		
				2-8_BIT_CHAR		
				3-CTRL_CHAR		

Table: System Level Functions

LEVEL	COMMAND	PARAMETER	QUALIFIER	QUALIFIER	QUALIFIER	QUALIFIER
				4-REPEAT_CHAR		
				6-PAD_CHAR		
				7-PACKET_SIZE		
				8-RX_TIMEOUT		
				9-NUM_PAD_CHAR		
			6-START CHAR			
			7-SEND DELAY			
	4-MONITOR	1-SMDR				
		2-DATA_SMDR				
		7-LOGS	1-MAINT_PORT			
			2-SYS_PRINTERS			
	5-SUSPEND_PRTR	7-PRINTER_PLID	1-BAY/SLOT/CCT/SCT			
			3-EXT-NUM			
		8-PRINTER_PORT				
	6-QUIT					
	7-RESUME_PRTR	7-PRINTER_PLID	1-BAY/SLOT/CCT/SCT			
			3-EXT-NUM			
		8-PRINTER_PORT				
	8-RE-START	2-RESET_SYSTEM				
		3- IP_PHONES				
		7-SHUTDOWN				
	9-STOP	7-LOGS				
	0-MORE_KEYS	1-SET_FIRMWARE	1-STATUS	1-BAY/SLOT/CCT		
				2-SET_TYPE	1-SS4150	
					2-SS4025	
				3-EXT_NUM		
				4-ALL		
			2-DOWNLOAD	1-BAY/SLOT/CCT		
				2-SET_TYPE	1-SS4150	
					2-SS4025	
				3-EXT_NUM		

Table: System Level Functions

LEVEL	COMMAND	PARAMETER	QUALIFIER	QUALIFIER	QUALIFIER	QUALIFIER
				4-ALL		
			3-ABORT_DOWNLOAD	1-BAY/SLOT/CCT		
				2-SET_TYPE	1-SS4150	
					2-SS4025	
				3-EXT_NUM		
				4-ALL		
			4-FORCE_DOWNLOAD	1-BAY/SLOT/CCT		
				2-SET_TYPE	1-SS4150	
					2-SS4025	
				3-EXT_NUM		
				4-ALL		
		2-UPGRADE_BOOT				
		3-PING				
		0-MORE_KEYS				

Note: TRAP_BACKUP and TRAP_RESTORE are diagnostic tools that only Mitel personnel use.

Setting and Showing Time

The system time-of-day may be set and verified from the maintenance terminal although the hour format used is specified during CDE. Note that the user may set the time in either 12-hour or 24-hour format by using the PM softkey as required. To set the system time from the maintenance terminal, press the following softkeys:

SET
TIME

At this point the softkeys are disabled, and the user is prompted to enter the desired time:

enter Time HH:MM

After entering a valid time, press the ENTER softkey to implement the new time-of-day, or press the CANCEL softkey to cancel the new time.

Verify the time-of-day by pressing the following softkeys:

SHOW
TIME
ENTER

Setting and Showing Date

The system date may be set and verified from the maintenance terminal. To set the system date from the maintenance terminal, press the following softkeys:

SET
DATE

At this point the softkeys are disabled, and the user is prompted to press the desired date:

enter Date DD/MM/YY

After entering a valid date, the user may implement the new date by pressing the ENTER softkey, or the user may cancel the new date by pressing the CANCEL softkey.

Verify the date by pressing the following softkeys:

SHOW
DATE
ENTER

Showing Device Status

The maintenance user can display the current data characteristics printer ports and status of the onboard ASU cards (embedded Analog Main Board and Analog Option Board). For ICPs equipped with ASU II, the command displays the IDPROM for the line cards in the ASU II slots.

For the AX, the complete information for the fan, temperature, and PSU (power supply unit) can be displayed.

To view the status of the AMB/AOB, press the following softkeys:

SHOW
DEVICE
ASU INFO
ENTER

Note: The ASU INFO command provides the part number for the AMB/AOB:
50004403 - CX AMB version 1
50004970 - CX AMB version 2
50004401 - CX AOB version 1
50004871 - CX AOB version 2
50003724 - MX AMB
50003725 - MX AOB

To view the status of the printer port, press the following softkeys:

SHOW
DEVICE
PRINTER-PORT
ENTER

To view the status of the power supply unit (PSU), fan and temperature in the AX, press the following softkeys:

SHOW
DEVICE
RM INFO
ENTER

Showing Number of Line Appearances

To show the number of line appearances programmed, press the following softkeys:

SHOW
LN_APP_COUNT
ENTER

Showing System Identity

To show the System Identity screen, press the following softkeys in Maintenance

SYSTEM
SHOW
IDENTITY
ENTER

The following display appears:

CONTROL CARD | CURRENTLY RUNNING | FLASHCARD FIRMWARE

This information includes the control card name, the load of software running on the card, and the load of firmware on the flash card.

Note: The current load is displayed only after the controller is reset following the installation or upgrading of the system software.

The control card entry shows the Stratum clock (ST3), platform type, and CCA number. Front Flash and Internal Flash entries show flash card capacity, if applicable.

An asterisk beside a load of software in the "Load on Flash Card" column shows that this software is not compatible with the software that is running. This asterisk informs the technician that another download is necessary.

Setting Password

It is recommended, for system security, that MTCE passwords be changed regularly once the SX-200 PBX has been put into service. Password changes may be made by the affected username (INSTALLER, MAINT1, MAINT2, SUPERVISOR, ATTENDANT), or any other username with a higher priority (see Login Procedures). A password may be any string of alphanumeric characters of up to 20 characters in length (any more characters are ignored).

CAUTION: Entering new passwords with alphabetic characters will inhibit login from the attendant console, because only numeric characters may be entered from the console.

To change the password, press the following softkeys:

SET
PASSWORD
(enter the required Username)
ENTER

The system then prompts the user for the old password (see Note):

Enter old password; then press RETURN/ENTER:

Enter the old password. The system prompts the user for the new password as follows:

Enter new password; then press RETURN/ENTER:

The system then prompts the user to verify the new password:

Enter new password to verify; then press RETURN/ENTER:

After the user verifies the new password, the system implements the password change; the old password is no longer valid. An incorrect entry of the old or new passwords will cause the password changing process to be aborted. Neither old nor new passwords are echoed back to the maintenance terminal display.

Note: If the Username selected is the one used when logging in, or the username selected is a higher level than the login user, the system will prompt for the old password. When the username selected is a lower level than the login user, the system will not require verification of access priority and will prompt for the new password only.

Setting Maintenance Port Baud Rate

To change the maintenance port baud rate from its default value of 9,600 baud, choose one of six available baud rates: 300; 600; 1,200; 2,400; 4,800; or 9,600 baud. To change the maintenance port baud rate, press the following softkeys:

SET
SPEED
MAINT_PORT
(select the softkey for desired baud rate)
ENTER

Note: Restoring the default database will reset the baud rate to 9,600.

Setting System Printer Baud Rate

To change the system printer port baud rate from its default value of 300 baud, choose one of six available baud rates: 300; 600; 1,200; 2,400; 4,800; or 9,600 baud. To change the printer port baud rate, press the following softkeys:

SET
SPEED
PRINTER_PORT
(select the softkey for desired baud rate)
ENTER

Before pressing the ENTER softkey, you can cancel the entry by pressing the CANCEL softkey. Note that this operation changes the baud rate of the SX-200 ICP system port only; the baud rate of the printer itself must be changed separately. Refer to the manufacturer's instructions for the particular printer being used.

Note: Restoring the default database will reset the baud rate to 300.

Assigning Printer Ports

Printouts are produced at the device specified in CDE Form 34, DIRECTED IO. See the Program CDE section for details. Setting speed for printer ports other than the system printer port is done in CDE Form 11, Data Circuit Descriptors.

Setting Kermit Parameters

To use Kermit, you must have a communications package that supports the Kermit protocol. HyperTerminal can be used for serial connections; a secure Telnet client that supports SSL/TLS and Kermit can be used for remote connections.

Note: Kermit is used to migrate a database from an SX-200 EL/ML to an SX-200 ICP MX. Backups and Restores of MX and CX/CXi databases must be done with the use of an FTP server.

To change the default values for the Kermit parameters, press the following softkeys:

DATABASE
SET_PARAM
RECEIVE

Select one of the following softkeys and modify the parameter.

EOL_CHAR
8_BIT_CHAR
CTRL_CHAR
REPEAT_CHAR
PAD_CHAR
PACKET_SIZE
RX_TIMEOUT
NUM_PAD_CHAR

Table: Default Kermit Parameters

Parameter	Softkey Default	Values
Packet Length	PACKET_SIZE	94
RX_TIMEOUT	10	
NUM_PAD_CHAR	00	
PAD_CHAR	32	
EOL_CHAR	13	
CTRL_CHAR	35	
8_BIT_CHAR	38	
Repeat Character	REPEAT_CHAR	C6
Block Check Count	---	03
Retry Count	------	25
Start Character	START_CHAR	01
Send Delay Entry	SEND_DELAY	00

Backing Up a Database

The SX-200 ICP stores information in several databases, among them are databases for CDE, voice mail (includes configuration, greetings, and user messages), and IP networking configuration. All the databases are maintained in flash memory inside the SX-200 ICP controller. To back them up, you will need to install an FTP server on your PC.

Note: Immediately after restarting the system, wait until startup processes have been completed before initiating a database backup. Initiating a database backup before all startup processes have been completed (for example, voice mail) may interfere with the system boot procedure.

It is recommended that you back up the database:

- after the system is first installed and the database is set up
- after changes have been made to the database.

Before starting the backup, ensure that the FTP host, username, and password are programmed in Form 47. Also, ensure that the FTP destination directory is writeable (i.e., not read-only).

To back up to a database:

1. Log in to the Maintenance application (serial, secure Telnet, or SX-200 Web Interface).
2. Press System, Database and then Backup.
3. Press FTP_SERVER.
4. Press FULL to back up all three databases, or MEDIUM to exclude voice mail messages from the backup. Full backups are recommended.

   Note: Voice mail is unavailable while the backup is in progress.

   Note: If voicemail storage is full, a backup can not be performed until at least some of the messages have been deleted. If the largest remaining memory fragment is more than 500KB but less than 1MB, a major log is generated and a major alarm is raised. If the largest memory fragment is less than 500KB, a critical log is generated, a critical alarm is raised, and the system resets immediately. To increase voice mail capacity on flash-based systems, install an optional internal hard drive.

The system creates a backup to the specified FTP home directory.

Restoring a Database

IMPORTANT: If the IP addressing information in the old and new (restored) database do not match, the IP Phones will reboot. Rebooting begins 10 to 15 minutes after the system resets (a reset is required following a database restore) and takes approximately two to three minutes to complete.

To restore from an FTP server:

1. Log in to the Maintenance application (serial, secure Telnet, or SX-200 Web Interface).
2. Press System, Database and then Restore.
3. Press FTP_SERVER.
4. Press CONFIRM, then enter the database file name.
5. Press CONFIRM.

   The system restores the database from the specified FTP home directory, and then reboots.

Note: The Restore fails if the voice mail language(s) in the restored database do not match those in the controller. Attempting to restore a database backed up from another system is the probable cause. The CDE, IP Networking, and other databases are unaffected by the failure. Form 50, however, may have been changed.

Installing an Alternate Database

You can replace the factory default database with one of the alternates provided on the software CD or with one that was programmed on another system and saved to a file.

The blank database on the CD has no programming except for the IP settings in Form 47, which are the same as those in the default database. The Premier database is for the MX Premier Business systems only.

A database programmed with four-digit extension numbers is available on Mitel Online.

Blank.db

The blank database is empty except for the following:

- DHCP server default settings (Form 47) as defined in the default db
- IP Set Registration and IP Set Replacement PINs (Form 02) as defined in the default db
- Voice mailboxes for the Operator (mailbox 0) and Administrator (mailbox 999) enabled thereby allowing access to voice mail Forms 49, 50, and 51.

Premier.db

The Premier Business system is available to dealers with Advanced certification. Its database includes licenses for IP trunking and has the 'PC (2nd) Port on IP Sets' option in Form 4 enabled, neither of which is included in the default database. Other differences between the two databases are as follows:

	Option Value	
Form 4 Option	**Premier Database**	**Default Database**
04 - Maximum ACD Agents	5	0
15 - Maximum IP Trunks	2	0
87 - Record a Call	Enabled	Disabled
96 - Number of Links	2	0
97 - Support Softkey Access to Voicemail	Enabled	Disabled
104 - Maximum ACD Agents	5	0
105 - Mitel Application Interface	Enabled	Disabled
114 - Maximum IP Sets	8	20
115 - Maximum IP Trunks	2	0
125 - Licensed Embedded Voicemail Boxes	8	20
131 - PC (2nd) Port on IP Sets	Enabled	Disabled

To install an alternate database, either re-install the system software and the required database to a CompactFlash card, and then use the card to re-initialize the controller (see Installing the System Software). Or, follow the procedure below to download the database to the controller via an FTP server.

To install an alternate database:

Note: The Premier database should be installed BEFORE enabling FOR options in Form 4. Doing so avoids a conflict with Option 114, which arise from the Premier database having fewer IP phones programmed than the default databases (8 versus 20). The conflict can also be resolved by deleting the excess phones from Form 9 prior to enabling the FOR options.

1. Initialize the SX-200 ICP.
2. If not using the default IP Address and FTP server information:
   - Once the system is running, use CDE Form 47 to program valid IP Address and FTP information .
   - Reset the system.
3. Establish a connection between the FTP server and the SX-200 ICP, either through the Layer 2 switch or by connecting directly to the LAN port on the controller.
4. Copy the required .db file to the home directory of the FTP server. The database files on the software CD are in the /Software directory.

5. Restore the database file.

After the Restore is complete, the system will automatically reset.

Backing Up Log Entries (Kermit)

Log entries can be saved to a file on the Maintenance PC using Kermit. The log entries, together with other files useful for troubleshooting, can also be backed up using Kermit (see Retrieving logs and other system files using Kermit), emailed or uploaded to an FTP server.

To back up log entries using Kermit, you must have a communications package that supports the Kermit protocol. HyperTerminal can be used for serial connections; a secure Telnet client that supports SSL/TLS and Kermit can be used for remote connections.

Note: The Mitel Telnet client does not support Kermit.

The Logs Backup function allows you to copy all maintenance log entries into a text file on either a directory on your PC's hard disk, or on a diskette in the PC's disk drive.

To back up log entries using Kermit, press the following softkeys:

DATABASE
LOGS_BACKUP
ENTER

The system prompts the user for the file name (see Note below).

Enter the file name.
ENTER

The system then prompts the user to exit to a local Kermit session. Ensure that the Kermit session is set to text mode, because the maintenance log entries must be copied into a text file.

In the Kermit session, issue the Receive command.

After the file transfer has completed, the Maintenance screen returns the message "Download/Upload Successful."

Note: Ensure that the file name is meaningful to you. It can include abbreviations for the name of a remote site, the purpose or function of the database, the version of the software, or the number of the database; for example, LogMit2_8.txt.

Backing up "Trap" files (Kermit)

Under severe error conditions, the SX-200 ICP flags the occurrence to itself as an exception. All exceptions are reported in a trap file, which contains a function flow of the last steps that were executed before the error occurred. Mitel Technical Support use the Trap files for troubleshooting. Trap files can be backed up and sent to an FTP server, emailed, or saved along with other system files using Kermit.

Note: Trap files are included in the files backed up using the Logs Backup function described above. The log files provide additional data about the state of the system when the exception error occurred and are more useful for troubleshooting than the trap files alone.

To back up Trap files using Kermit, you must have a communications package that supports the Kermit protocol. HyperTerminal can be used for serial connections; a secure Telnet client that supports SSL/TLS and Kermit can be used for remote connections.

Note: The Mitel Telnet client does not support Kermit.

To back up trap files using Kermit, press the following softkeys:

DATABASE
TRAP_BACKUP
ENTER

The system prompts the user for the file name (see Note below).

Enter the file name.
ENTER

The system then prompts the user to exit to a local Kermit session. Ensure that the Kermit session is set to text mode, because the maintenance log entries must be copied into a text file.

Note: Ensure that the file name is meaningful to you. It can include abbreviations for the name of a remote site, the purpose or function of the database, the version of the software, or the number of the database; for example, LogMit2_8.txt.

Showing Parameters

The Kermit protocol is used to establish and maintain communication between the PC during the backup, restore, or dump log entries processes. To use Kermit, you must have a communications package that supports the Kermit protocol. To display the current Kermit parameters, press the following softkeys:

DATABASE
SHOW_PARAM

Note: After the system is reset, the Kermit parameters return to the default values.

Monitoring Logs

The user may monitor the progress of the maintenance logs as they occur. When the monitor logs process is running, maintenance logs will be output to a device and recorded in a text file. The output device can be either the maintenance terminal or the system printer, as specified in CDE Form 34, Directed IO. If logs are monitored on the system printer port, the user can log out from the maintenance terminal without first stopping the monitor process. However, if logs are monitored on the maintenance terminal, the monitor process must first be stopped before logging out.

To monitor logs, press the following softkeys:

MONITOR
LOGS
MAINT-PORT
ENTER

To stop monitoring logs, SMDR reports, or DATA_SMDR, press the following softkeys:

STOP
LOGS
ENTER

For information on how to back up log entries, refer to Backing Up Log Entries in this section. For further information on maintenance logs, refer to the Maintenance Log Functions section, and to the General Maintenance Information section.

Monitoring SMDR

The user may monitor the progress of the system SMDR reports as they occur. Unlike the MONITOR LOGS command, monitoring of SMDR may be done only at the maintenance terminal. It is not necessary to select a print device in this case, because monitoring will be output to the maintenance terminal automatically. Spontaneous printing of SMDR data to the system printer port is not affected. To monitor SMDR reports at the maintenance terminal, press the following softkeys:

MONITOR
SMDR
ENTER

Refer to the Station Message Detail Recording section for further information on SMDR.

Monitoring DATA_SMDR

The user may monitor the progress of the system DATA_SMDR reports as they occur. Unlike the MONITOR LOGS command, monitoring of DATA_SMDR may be done only at the maintenance terminal. It is not necessary to select a print device in this case, because monitoring will be output to the maintenance terminal automatically. This does not affect the spontaneous printing of DATA_SMDR data to the system printer port. To monitor DATA_SMDR reports at the maintenance terminal, press the following softkeys:

MONITOR
DATA_SMDR
ENTER

Refer to the Station Message Detail Recording section for further information on DATA_SMDR.

Resetting the System

The RESET_SYSTEM command allows the maintenance user to reset the entire system (including the Application Processor Card, if installed). Resetting optimizes the integrity of the system software. Resetting should be done only during a period of low or no call processing traffic because the system will be totally inoperative for a period of approximately two to three minutes. To initiate a system reset, press the following softkeys:

RE_START
RESET_SYSTEM
ENTER

You can also reset the system by pressing the Reset button on the front of the controller, or by powering the controller down, then back up.

Restarting the APC

The RESET_APC commandallows the maintenance user to reset the optional Applications Processor Card. Restarting causes the module to reboot for debugging purposes if problems have been encountered loading software or accessing the module. To initiate a restart, press the following softkeys:

RE_START
RESET-APC
ENTER

Restarting the IP Phones

The IP_PHONES command allows the maintenance user to reset all IP phones on the system. Restarting will cause all IP phones to renegotiate their DHCP settings. Restarting should be done only during periods of low or no call processing because the phones will be totally inoperative for a period of approximately two to three minutes. To initiate a restart, press the following softkeys:

RE_START
IP_PHONES
ENTER

System Shutdown

The Shutdown command allows the maintenance user to shut the system down before moving it or replacing hardware. Shutdown will terminate all calls (except calls between IP phones or calls carried on IP trunks) and disable the serial and IP interfaces. While the system is shut down, new calls cannot be made, and the system cannot be pinged or accessed for management purposes. To return the system to service, unplug it and plug it in again, or press the Reset button on the front of the controller. To initiate a shutdown, press the following softkeys:

CAUTION: The following procedure takes the system out of service.

1. Log into the Maintenance Terminal (serial connection only), then begin a Maintenance session.
2. Press the following softkeys:

   RE_START
   SHUTDOWN
   ENTER
3. When prompted on the terminal, power down the controller.

Note: The SHUTDOWN command should not be used from a remote location since physical access to the controller is required to restart it.

System Reset

The maintenance user can program the system to reset after system software or resource faults occur. The reset can be scheduled as follows:

- reset occurs immediately if there is a single fault
- reset on a specific time/day if there are 50 or fewer faults, OR immediately if there are more than 50 faults.

Programming an Immediate Reset

To program the system to reset immediately if there is a single fault, press the following softkeys:

SET
RESET_TIME
IMMEDIATELY
ENTER

To program the system to reset immediately if there are more than 50 faults, press the following softkeys:

SET
RESET_TIME
AFTER_N_FLTS
ENTER

Programming a Scheduled Reset

To program the system to reset on a particular time and day if there are 50 or fewer faults, press the following keys:

SET
RESET_TIME

DAY/TIME
(press one of the seven “day” softkeys or DAILY)
(enter the hour and minutes in 12-hour or 24-hour format)
ENTER

Notes:

1. By default, the system resets on Sunday at 2:00 am if there are 50 or fewer faults, or immediately if there are more than 50 faults.
2. The system is non-operational while it resets. To minimize disruption, schedule the reset for a time/day of the week when the system is not in use. To avoid a service outage, provide an ASU with emergency phones.

Showing Reset Time

The maintenance user may obtain a report indicating when and under what conditions a system reset will occur. To obtain such a report, press the following softkeys:

SHOW
RESET_TIME
ENTER

Setting Alarm Thresholds

Alarm thresholds can be programmed by the maintenance user to facilitate the requirements of a particular system. The thresholds can be changed for system devices (Lines, Trunks, and DTMF receivers), but not for resources (tasks and components, memory consumption, and memory fragmentation). For more information about resources, including their threshold values, see Resource Monitor.

To change the alarm thresholds for lines throughout the entire system, press the following softkeys:

SET
ALARM_THRESH
LINES
ENTER
SYSTEM
(enter the desired MINOR alarm threshold percentage, or press RETURN to leave unchanged)
(enter the desired MAJOR alarm threshold percentage, or press RETURN to leave unchanged)
(enter the desired CRITICAL alarm threshold number, or press RETURN to leave unchanged)
ENTER
CONFIRM

The procedure for changing alarm thresholds for trunks and DTMF receivers is the same as for lines. The only difference is that the TRUNKS or RECEIVERS softkeys are used in place of the LINES softkey.

Alarm thresholds may be changed on a bay basis as well. The procedure is the same as that shown above; the only difference is that the BAY softkey is used in place of the SYSTEM softkey. DTMF receiver thresholds may be changed only in digital bays.

In all cases, the alarm threshold table is shown in the applications area of the screen.

Table: Default Alarm Thresholds

ALARM CATEGORY (Peripheral Devices)	ALARM THRESHOLDS	
Minor	**Major**	**Critical**

Table: Default Alarm Thresholds

Lines	0%	20%	0
Trunks	0%	20%	0
DTMF Receivers	0%	20%	0

Upgrading Set Firmware (Set Download)

Set firmware for SUPERSET 4025 sets, SUPERSET 4125, SUPERSET 4150 sets, and DSS/BLF Interface Modules can be upgraded from the main system software load. The system software now includes the two set types and their respective firmware version IDs.

Download

Set firmware downloads can be selectively performed in several methods, as follows:

- single set
- all sets of one type
- all sets on one Digital Line card
- all sets in one bay
- all sets in the PBX

To download new set firmware, press the following softkeys:

SYSTEM
MORE_KEYS
SET_FIRMWARE
DOWNLOAD
BAY/SLOT/CIRCUIT or SET_TYPE or EXT_NUM or ALL
(select one and enter/select the required parameters)

The selected set(s) should start downloading (each set download takes approximately 1 minute).

Force Download

The FORCE_DOWNLOAD command forces a download to occur unconditionally, even if the set firmware is newer than the firmware being downloaded (this option allows returning a set to a previous firmware version when required). To force download new set firmware, press the following softkeys:

SYSTEM
MORE_KEYS
SET_FIRMWARE
FORCE DOWNLOAD
BAY/SLOT/CIRCUIT or SET_TYPE or EXT_NUM or ALL
(select one and enter/select the required parameters)

The selected set(s) should start downloading (each set download takes approximately 1 minute).

Abort Download

The ABORT_DOWNLOAD command terminates a download to the specified device(s), except to a set that is currently being downloaded. To abort a download of new set firmware, press the following softkeys:

SYSTEM
MORE_KEYS
SET_FIRMWARE
ABORT DOWNLOAD
BAY/SLOT/CIRCUIT or SET_TYPE or EXT_NUM or ALL
(select one and enter/select the required parameters)

The selected set(s) which have not been downloaded are removed from the schedule.

Conditions and Restrictions

The following conditions and restrictions apply to the set firmware download feature. A message will be provided explaining why a download cannot be done.

- these commands are not available from the attendant console, even in Installer Maint1, or Maint2 mode
- these commands are not available from Supervisor or Attendant mode
- perform DOWNLOAD operations during off-peak hours to minimize service interruptions.
- if for any reason a DOWNLOAD fails, the download operation is aborted. The system generates an information maintenance log that explains the cause of the failure.
- a DOWNLOAD command cannot be issued to a set that has a firmware load equal to or newer than the system load
- a DOWNLOAD command cannot be issued to a set that is disconnected
- a FORCE_DOWNLOAD command cannot be issued to a set that is disconnected
- a DOWNLOAD or a FORCE_DOWNLOAD command will not be initiated on a set that is already scheduled for a DOWNLOAD, nor will it change its position in the sequence - the message “Already Scheduled” will be given
- a DOWNLOAD or a FORCE_DOWNLOAD command will not be initiated on an invalid set type
- disconnecting a set during a DOWNLOAD will immediately terminate the download leaving the set firmware corrupted and not usable when the set is reconnected

When the set powers up, there will be no dial tone and the message “DOWNLOAD REQUIRED” will be displayed.

- if a Bay is powered down while set downloading is scheduled, when it is powered up, all of its scheduled sets will have been removed from the upgrade schedule
- when downloading to a card, a bay, or a PBX, only SUPERSET 4025 sets, SUPERSET 4125, SUPERSET 4150 sets, and DSS/BLF Interface Modules with firmware versions older than the system version will be upgraded. All other devices are bypassed.

  The system identifies DSS/BLF Interface Modules as SUPERSET 4025 or SUPERSET 4125 sets. Therefore, if you apply the download command to SUPERSET 4025 sets or SUPERSET 4125 sets, on a card, a bay, or PBX, the DSS/BLF Interface Modules on that card, bay or PBX are also upgraded.
- a set with Do Not Disturb activated can receive a DOWNLOAD
- call forwarding settings and feature keys remain as they were after a DOWNLOAD
- if a multiple line appearance of a set’s prime line is busy at another extension, the scheduled set can still receive a DOWNLOAD
- if a line key is programmed on two sets and is active on one set, the other set can receive a DOWNLOAD
- if a CO line key is programmed on two sets and one set is receiving a DOWNLOAD, the other set can still make a call on the CO line
- if a DOWNLOAD is issued to a ringing set, the DOWNLOAD is scheduled but does not begin until the set returns to idle state

- if a DOWNLOAD is issued to a locked out set, the DOWNLOAD is scheduled but does not begin until the set returns to idle state
- if a DOWNLOAD is issued to a set that is logged in as an ACD agent, the DOWNLOAD is performed and the set returns as a logged in ACD agent when the DOWNLOAD is complete
- if a DOWNLOAD is issued to a phantom set, the DOWNLOAD will not occur
- if a DOWNLOAD is issued to a phantom bay, the DOWNLOAD will not occur
- if a set that is scheduled for a DOWNLOAD is swapped before it receives its DOWNLOAD, the DOWNLOAD does not take place
- if a DOWNLOAD is issued to a set with a busy SUPERSET interface module (SIM1 or SIM2), the DOWNLOAD is scheduled but does not begin until the set and the SUPERSET interface module return to idle state
- a database backup can be performed while a set(s) is being downloaded
- if a MOVE command is issued to a set that is scheduled for a download, the set will move to the requested plid and the DOWNLOAD will be cancelled
- if a SuperKey activity is in progress when the DOWNLOAD is scheduled to begin, the DOWNLOAD will be rescheduled
- once the DOWNLOAD starts, the system automatically turns off the telephone music.
- in CDE, a set scheduled for a DOWNLOAD can be deleted from CDE; however, a set that is actively in a DOWNLOAD cannot be deleted
- a set that is busied out can be downloaded
- a set that is actively receiving a download cannot receive a set page (the paging party receives reorder tone)
- if an all set DOWNLOAD is initiated, and then an all set page is performed, all sets except the set being downloaded receive the page, and the remaining sets do not begin to download until one of the sets that received the page goes onhook (all downloadable sets remain scheduled for a DOWNLOAD)

Upgrading Boot ROM

Upgrade the Boot ROM only if instructed in the Release Notes on the system software CD.

SYSTEM
MORE_KEYS
UPGRADE_BOOT
ENTER

After the upgrade is started, wait approximately 15 seconds, and then reset the system. There is no confirmation on screen that the upgrade has been completed.

Using the PING command

Use the ping command to send an echo request to a network host.

Note: Run the test in VT100 mode, not TTY (line interface) mode.

SYSTEM
MORE_KEYS
PING
PING Enter <IP-address>
ENTER

If the ping request is successful, the system will display round-trip times and packet loss statistics. If it is unsuccessful, "ping timeout" and "no answer from <IP-address>" will display.

Reports Level Functions

Reports level functions allow you to display maintenance information. The following types of reports are available: configuration, alarm status, circuit status, and displaying and clearing of device errors. To access the report commands, press the REPORTS softkey. All of the following operations are possible while in the reports level. All operations available in diagnostics are shown in the Reports Level Functions table, except CANCEL and ENTER. Press the CANCEL softkey at any time to exit the current operation without committing (saving) any changes, or press the ENTER softkey to commit changes.

Clear Locked Extensions

Voice mail ports that lose synchronization with the ICP may become locked (unable to take calls). Similar situations may occur with sets and stations. To clear the state of these devices:

For Specific Circuits

To clear the port for a specific circuit, press the following softkeys:

SHOW
STATUS
BAY/SLOT/CCT (enter the required bay, slot, circuit and sub-circuit numbers, pressing the RETURN key after each one)
ENTER

Check the state of the device. If it is not showing IDLE, press the following softkey (see Note 1)

CP_DEV_IDLE (See Note 2)

For Specific Extensions

To clear the port for a specific extension, press the following softkeys:

SHOW
STATUS
EXT-NUM (enter the required extension number, then press the RETURN key)
ENTER

Check the state of the device. If it is not showing IDLE, press the following softkey (see Note 1)

CP_DEV_IDLE(See Note 2)

For Specific SWID

To clear the port for a specific swid (only valid for sw_sets, sw_stations and sw_line), press the following softkeys:

SHOW
STATUS
SWID
SW_SET (or SW_STATION / SW_LINE)
(enter the required Index number, then press the RETURN key)
ENTER

Check the state of the device. If it is not showing IDLE, press the following softkey (see Note 1)

CP_DEV_IDLE(See Note 2)

Note1: It is not necessary to make the device IDLE in each and every scenario. (For example, you cannot make devices IDLE from the DND state.)

Note 2: We strongly recommend that you use these commands in problem scenarios only, and not in active calls.

Note 3: Only Installer and Maint1 users can clear locked extensions.

Show Configuration

The configuration report provides the maintenance user with information on the hardware that is currently installed in the system. The user may request a configuration report on a specific card slot, a specific extension number, or the entire system. The information provided includes:

- The physical location(s), in terms of bay number, slot number, and circuit (module) number
- The type of card / module installed in a location
- The type of card / module programmed for that location.

The SX-200 ICP Controller (Bay 0) shows

- Circuits 0 as the main controller programmed and installed.
- Circuits 1 and 2 as DTMF Receivers programmed and Pseudo Receivers installed
- Circuit 3 blank
- Circuit 4 as a DTMF Generator programmed and installed.

The DSP module on the IP Bay 1 shows

- Circuit 2 as a Dual DSP module programmed and installed
- Circuit 4 as a DTMF receiver programmed and installed.

The absence or presence of an asterisk (*) next to ONBOARD ASU and OFFBOARD ASU indicates the number of circuits present. The number varies depending on the slot. For slot 13, the asterisk indicates the presence of four circuits (the maximum); the absence of an asterisk indicates the presence of two circuits. For Slots 14 and 15, the number of circuits is 24 (asterisk present) or 16 (asterisk absent).

Specific Card Slot

To obtain a configuration report on a specific card slot, press the following softkeys:

SHOW
CONFIG
BAY/SLOT/CCT
(enter the required bay, slot, and circuit numbers; press the RETURN key after each one)
ENTER

Note: Both BNIC and BONS cards display with an asterisk (*) to distinguish them from regular digital line cards/ONS line cards.

Specific Extension

To obtain a configuration report on a specific extension number, press the following softkeys:

SHOW
CONFIG
EXT-NUM
(enter the required extension number; then press the RETURN key)
ENTER

The system displays the Bay and Slot numbers at which this extension is terminated.

Entire System

To obtain a configuration report on the entire system, press the following softkeys:

SHOW
CONFIG
ALL
ENTER (or MORE or CANCEL)

In all cases, the system outputs the configuration data in the applications area of the screen. In cases where the data requires more space than is available on the screen, the user is prompted to request more data via the MORE softkey, or to cancel the output via the CANCEL softkey.

Table: Reports Level Functions

LEVEL	COMMAND	PARAMETER	QUALIFIER	QUALIFIER	QUALIFIER	QUALIFIER
9-REPORTS	2-SHOW	1-CONFIG	1-BAY/SLOT/CCT			
			3-EXT-NUM [number]			
			4-ALL			
		2-ALARMS	2-DEVICE TYPE	1-LINES		
				2-TRUNKS		
				3-RECEIVERS		
				7-BAY		
			4-ALL	0-MORE		
		3-STATUS	1-BAY/SLOT/CCT	DEVICE STATUS	See Note	
			2-SWID	1-SW_STATION	See Note	
				2-SW_RECEIVER	See Note	
				3-SW_CONSOLE	See Note	
				4-SW_LINE	See Note	
				6-SW_DTMF_GEN	See Note	
				7-SW_SET	See Note	
				8-SW_DATA_STN	See Note	
				0-MORE_KEYS	1-SW_CO_TRUNK	TRUNKS See Note
					2-SW_DID_TRUNK	TRUNKS See Note
					3-SW_TIE_TRUNK	TRUNKS See Note
					4-SW_DISA_TRUNK	TRUNKS See Note
					7-SW_TRUNK_GRP	TRUNK GROUPS See Note
					8-SW_HUNT_GRP	HUNT GROUPS See Note

Table: Reports Level Functions

LEVEL	COMMAND	PARAMETER	QUALIFIER	QUALIFIER	QUALIFIER	QUALIFIER
					0-MORE_KEYS	
			3-EXT-NUM [number]			
			4-ALL	1-CP_DWA		
				2-CP_DWA_MEM		
				3-LINK STATUS		
				6-MT_DWA		
				7-MT_DWA_MEM		
				8-UP_1_PAGE		
				9-DOWN_1_PAGE		
			7-POE STATUS (CXi only)			
			8-VOICEMAIL			
			9-IP_TRUNKS			
		6-CHANNEL-MAP	1-LOGICAL	1-CHANNEL NUM		
			2-PHYSICAL	1-BAY_NUM		
				2-LINK_NUMBER		
		7-ERRORS	2-DEVICE_TYPE	1-SS3_SS4		
				3-DIGITAL_SETS		
				4-HDLC		
				6-DATASETS		
				7-CONSOLE		
				9-T1_TRUNK		
	3-CLEAR	7-ERRORS	1-BAY/SLOT/CCT			
			2-DEVICE_TYPE	1-SS3_SS4		
				3-DIGITAL_SETS		
				4-HDLC		
				6-DATASETS		
				7-CONSOLE		
				9-T1_TRUNK		
			3-EXT-NUM			
			4-ALL	8-CONFIRM		
	6-QUIT					

Table: Reports Level Functions

LEVEL	COMMAND	PARAMETER	QUALIFIER	QUALIFIER	QUALIFIER	QUALIFIER

Note: The following softkeys are presented when a SWID selection is made: 1-CP_DWA, 2-CP_DWA_MEM, 3-LINK_STATUS (or CP_DEV_IDLE based on the device type), 6-MT_DWA, 7-MT_DWA_MEM, 8-UP_1_PAGE, 9-DOWN_1_PAGE. Press CP_DWA to view the device work area for the selected device. For CP_DEV_IDLE, See Clear Locked Extensions.

Show Alarms Report

The alarms manager is a software program that monitors the performance of the system, compares it to a set of default thresholds and, if the system performance is below the specified level, causes an alarm to be raised.

Alarm categories:

1. Lines
2. Trunks
3. DTMF Receivers
4. Bays

There are four alarm levels:

1. NO ALARM
2. MINOR
3. MAJOR
4. CRITICAL

There are three alarm types:

1. **Bay Alarms** - these are the alarm levels of the categories specific to each separate bay in the system.
2. **System Alarms** - these are the alarm levels of the categories on a system-wide basis.
3. **Overall Alarm** - this is the overall system alarm level, derived from all bay alarms and system alarms in all categories. It is displayed at all times above the upper right corner of the enclosed area of the maintenance display.

For more information on alarms, refer to the General Maintenance Information section.

Show Alarms All

The user can obtain an alarm report on the entire system (i.e., all device types, in all bays of the system) by pressing the following softkeys:

SHOW
ALARMS
ALL
ENTER (or MORE or CANCEL)

Enter MORE to step through displays of specific device type alarms (different device types are not summed together on one display).

Note: Although the report shows fifteen bays, the SX-200 ICP supports a maximum of eight physical bays (seven peripheral cabinets and the IP bay).

Show Alarm - Device Type

The user may obtain an alarm report on a specific device type (category) by pressing the following softkeys:

SHOW
ALARMS
DEVICE TYPE (LINES or TRUNKS or RECEIVERS or BAY)
ENTER

In all cases, the system will output the alarm status data in the applications area of the screen. In cases where the data requires more space than is available on the screen, the user is prompted to request more data via the MORE softkey, or to cancel the output via the CANCEL softkey. An example of an alarm report is shown in the figure below.

Figure: Example of LINE ALARM Status Display

The following table summarizes the terms used in the alarm status report:

Table: Terms Used In The Alarm Status Report

Term	Meaning
BAY# OR SYSTEM	The range of the specified alarm category is Bay 1.
NUMBER OF DEVICES	Total number of devices programmed in the specified category in the specified range; a total of 13 lines in Bay 1. Category is displayed on the command line.
TOTAL DEVICES UNAVAIL	Total number of devices unavailable to Call Processing in the corresponding TOTAL. Note: If a bay is programmed but not yet installed, it will appear in this column and generate a Major Alarm. Phantom bays programmed in Form 53 are not subject to this limitation.
PERCENT UNAVAILABLE	The percentage of devices unavailable to Call Processing in the corresponding TOTAL.
ALARM LEVEL	The current alarm level in the specified range.
MINOR	The Minor Alarm threshold - a percentage of the total number of devices in the specified range.
MAJOR	The Major Alarm threshold - a percentage of the total number of devices in the specified range.
CRITICAL	The Critical Alarm threshold - the actual minimum number of devices in the specified range allowed before the system will reset.
MOSS	This alarm appears if you attempt to enable an option in Form 04 -- System Options/ System Timers that has not been purchased

Show Status

Equipment Status Report

The equipment status report provides the maintenance user with information on current call processing and maintenance states of any device or range of devices. The information provided includes:

- physical location(s), in terms of bay number, slot number, circuit number, and sub-circuit number
- Software Identification (SWID) of device (where applicable)
- extension or trunk number (where applicable)
- programmed type of circuit
- maintenance status of the circuit
- software status of the circuit (where applicable)
- hardware status of the circuit (where applicable)
- background diagnostics status
- power-up diagnostics status
- Channel number used (where applicable).

An example of an equipment status report is shown in this Figure. Sixteen DTMF receivers is shown for each bay with a DSP module under slot 12. Eight CLASS generators are shown for each bay with a DSP module under slot 0.The CP_DWA and CP_DWA_MEM softkeys only appear if the device has such a work area.

Status - Bay/Slot/Circuit/Subcircuit

To obtain an equipment status report starting at a specific bay, slot, circuit, and subcircuit, press the following softkeys:

SHOW
STATUS
BAY/SLOT/CCT (enter the required bay, slot, and circuit number, followed by 00 for the subcircuit number. For any single-digit entries, press RETURN to receive a prompt for the next entry)
ENTER

Use the Up and Down arrows to scroll the display.

Status - Specific Extension

To obtain an equipment status report on a specific extension number, press the following softkeys:

SHOW
STATUS
EXT-NUM (enter the required extension number, then press the RETURN key)
ENTER

Figure: Example of an EQUIPMENT STATUS Report

The following table summarizes the terms used in the equipment status report:

Table: Terms Used In the Equipment Status Report

Term	Meaning
BB	The bay in which the device is located
SS	The card slot in which the device is located
CC	The circuit number of the device
SC	

The sub-circuit number of the device.

Note: For devices, such as LS trunks, that are located at the circuit level, enter 00 for the sub-circuit number.	
SOFTWARE_ID	Type of device installed and its software identification number (SWID)
EX/TK	The extension or trunk number of the device (where applicable); a number up to five digits in length.
TYPE	The circuit type; one of the following:
	ons - ONS line circuit
	ops - OPS line circuit
	dnic - DNIC line circuit
	rcvr - DTMF Receiver module
	moh - Music-on-Hold module
	lsgs - CO trunk circuit
	T1 - T1 trunk circuit
	dncon - DNIC console
	cutvr - cutover sensor
	ups - UPS sensor
	lamp - ONS lamp test
	did - DID trunk
	e&m - E&M trunk
	pcm - pcm channel
	dsp - digital signal processor
BG	Background diagnostics enabled; either “on” or “off”
PWR	Boot diagnostics enabled; either “on” or “off”
MTSTAT	The current maintenance status; one of the following:
	avail - available to CP and maintenance
	progr - programmed in CDE but not installed
	unprog - installed but not programmed in CDE
	suspt - suspect - failed diagnostic test once

	bsout - busied-out by maintenance - failed diagnostic test at least twice, or busied-out by maintenance user
	flty0 faulty with no passes
	flty1 faulty with one pass
	flty2 faulty with two passes
	flty3 faulty with three passes
	flty4 faulty with four passes
	flty5 faulty with five passes
	flty6 faulty with six passes
SWSTAT	The current call processing (CP) software status; for lines and trunks, one of the following:
	altms - alternate music
	acdwt - ACD wait
	bsout - busied-out
	bst - receiving busy tone
	cwait - using Auto Attendant overflow
	dlgrd - using Auto Attendant
	kspag - directed or broadcast paging
	idlks - receiving broadcast page
	bsgks - receiving directed page
	onstk - using ONS Voice mail feature
	colre - CO line is reserved
	campd - camped on
	on dialg - dialing
	dnd - do not disturb
	error - receiving reorder tone
	hfi - handsfree idle
	hfree - handsfree ringing
	hfs - handsfree suspended
	hold - consultation hold

	idle - idle
	lockd - locked-out
	pagng - paging
	parkd - parked (held by attendant)
	rngbk - ringback
	rngng - ringing
	rs232 - data station is establishing RS-232 protocol
	stowd - stowed (hard or call hold)
	suspd - suspended
	talkg - talking
	tkd - trunk dial
	wdtrx - data station waiting for dtrx response
	wfjct - waiting for call resources (receiver, channel)
	wflin - waiting for line
	The current call processing (CP) software status, for receivers; one of the following:
	free - ready for use by CP
	busy - currently in use by CP
	down - currently unavailable to CP
HWSTAT	The current hardware status; one of the following:
	idle - available to CP
	busy - busy
	down - card not present - unavailable to CP
	dwnld - downloading prompts to a SUPERSET 430 or SUPERSET 4DN telephone

Show Status - SWID

This operation enables a user to show software status by selecting a device type. To obtain an equipment status report on a software identifier, press the following softkeys:

SHOW
STATUS
SWID
(press one of the softkeys shown in the Software Identification (SWID) Types table below
ENTER

The system outputs the equipment status data in the applications area of the screen. Where the data requires more space than is available on the screen, the user is prompted to request more data via the 8-UP_1_PAGE or 9-DOWN_1_PAGE softkeys, or to cancel the output via the CANCEL softkey.

Table: Software Identification (SWID) Types

Softkey	Device Types
SW_STATION	single line port
SW_CO_TRUNK	CO trunk - 6 cct LS/GS, 4-cct CO, 4 -cct CLASS or 8-cct CLASS
SW_DID_TRUNK	DID trunks - 6 cct digital
SW_CONSOLE	console
SW_TIE_TRUNK	TIE trunk - E&M module
SW_LINE	SUPERSET line work areas
SW_DISA_TRUNK	6 cct LS/GS
SW_DTMF_GEN	DTMF generator
SW_SET	SUPERSET telephones
SW_TRUNK_GRP	trunk group
SW_DATA_STN	dataset - standalone
SW_RECEIVER	DTMF receiver
SW_HUNT_GRP	hunt group

Show Status PoE

To display information about Power over Ethernet (PoE) for the CXi controller's 10/100 802.3af LAN ports, press the following softkeys:

SHOW
STATUS
POE STATUS
ENTER

Figure: Example of a PoE Status report

PoE Module Status Fields

The following fields provide information about the PoE module:

- **Current Total POE Power Consumption** - The power currently being sourced by the 16 PoE ports.
- **Maximum POE Power Allowed** - The total power available from the 16 PoE ports (120 Watts). When this limit is exceeded, power will be turned off to the ports, starting with port 16 to and ending with port 1, until less than 120 Watts is being consumed.
- **RX timeout errors** - The total number of messages that were received and then rejected because of timeout conditions (for example, the messages were incomplete).
- **RX read errors** - The total number of messages that were received and then rejected because of read error conditions.

- **RX other errors** - The total number of messages that were received and then rejected because of other error conditions.
- **TX write errors** - The total number of messages that were not transmitted because of write error conditions.
- **Checksum errors** - The total number of messages that were received and then rejected because of checksum error conditions.
- **Protocol errors** - The total number of messages that were received and then rejected because of protocol error conditions.

PoE Port Status

The following fields provide about the 16 PoE ports:

Table: PoE Port Status Fields

Field	Meaning
pt	The port numbers from 1 to 16. (Port 17 does not supply PoE.)
cfg	The port PoE configuration, either enabled (yes) or disabled (no).
ena	The port enabled status. Displays ON if the port is configured for PoE and there is enough power available in the power budget (120 Watts, total). Otherwise, displays OFF. When a low-priority port loses power, this field displays OFF.
mode	The power source mode for the port. Can be used for debugging. • **RESET** - The port has been reset (usually because it has been disabled in CDE). • **DISC_R** - The PoE subsystem is discovering the port's resistative signature. The result is placed in the Oms field. If the resistative signature for a powered device is discovered, the port switches to CLASSI mode. Otherwise, it remains in DISC_R mode. • **DISC_C** - The PoE subsystem is discovering the port's capacitive signature. • **CLASSI** - If the resistative signature for a powered device is discovered, the PoE subsystem will then attempt to classify the power demands of the device. The result is placed in the CI field. • **RAMP_U** - After the port has been classified (CLASSI), the port PoE subsystem ramps up power to the device. • **I-SAMP / V-SAMP** - While the PoE subsystem is providing power to a device, it periodically samples the current (I) and voltage (V) consumption. • **RAMP_D** - A device has been unplugged and the port is powering down. After power down, the port switches DISC_R mode.
CI	The discovered Class type of the connected device. Each Class type has an allowable power consumption range (see below). The PoE module allocates the maximum amount for every range; for example, it allocates 6.49 Watts to all Class 2 devices.

Class **Power Consumption Range** 0 0.44 to 12.95 Watts 1 0.44 to 3.84 Watts 2 3.84 to 6.49 Watts 3 6.49 to 12.95 Watts **Note:** Unclassified (Class 0) devices are budgeted 7.5 Watts by the PoE subsystem, but can receive up to 13 Watts depending on need.	
Ohms	The discovered resistance of the connected device, in Ohms.
mA	The discovered current of the connected device, in milliamps.
Volts	The voltage on the port.
Watts	The power being consumed by the connected device, in Watts. If this field displays 0, then the device is not currently consuming power. A port may be enabled for PoE yet not provide power to the connected device.
Bdgt	The power budgeted for the port, in Watts.
ps	Port paused bit. Displays 1 if the port is paused; otherwise, displays 0. Used for debugging purposes only.
dny	Port power denied count.
OL	Port power overload fault count.
sht	Port short circuit fault count.
flt	

Fault condition. Displays a message only if the port is out of service. Otherwise, displays NONE.

- **NONE** - No fault, port in service
- **UV_OV** - Under voltage / over voltage failure
- **UV_OV_SPIKE** - UV / OV failure, plus a spike
- **PEAK_OC** - Peak overcurrent
- **OVERLOAD** - Overloaded port
- **DISCOVERY_R_FAIL** - Classification failed
- **CLASS_VIOLATION** - Power class type violation
- **DISCONNECT** - Port has been disconnected
- **DISCO_R_A2D_FAIL** - Analog to digital (A2D) failure
- **CLASSIFY_A2D_FAIL** - Classify A2D failure
- **DEVICE_FAULT_A2D_FAIL** - A2D failure
- **SAMPLE_DISC_A2D_FAIL** - Current and voltage sample A2D failure

Show Status Voicemail

To display information about voice mail disk usage and voice mail port status, press the following softkeys:

SHOW
STATUS
VOICEMAIL
ENTER

Figure: Example of a Voicemail Status report

Ports 1-12 correspond to circuits 1-12 on the first programmed voice mail card (slot 11) and voice mail ports 13-24 correspond to circuits 1-12 on the second programmed voice mail card (slot 12). "Blank" denotes ports that have not been programmed.

The information displayed is current as of the time the screen was entered. To refresh the information, press the UPDATE softkey to refresh.

Status - IP Trunks

To show the status of the IP trunks, press the following softkeys:

SHOW
STATUS
IP_TRUNKS
ENTER

Figure: Example of an IP TRUNKS Report

The following table summarizes the terms used in the IP Trunks status report:

Table: Terms Used In the IP Trunks Status Report

Term	Meaning
NODE	IP tunking node number
IP ADDRESS	IP address of the remote node
STATUS	The state of the link; one of the following:
UNASSIGNED - initial state ASSIGNED - incoming connection is being initiated OUT_OF_SERVICE - system is shut down EST_INCOMING - receiving "verify node" message EST_OUTGOING - initiating connection to remote node	

VER_OUTGOING - sending "verify node" message OPEN - remote node has been verified CLOSED - connection to remote node has failed (periodically, an attempt will be made to reestablish the connection) RELEASING - connection is being disconnected INVALID CONFIG - a programming error, most likely ARS needs to be configured to route to an ICP in Form 48. UNKNOWN - the remote node is an SX-200 IP Node or a Release 1.x SX-200 ICP.	
ICP	Type of remote node: 3300 ICP or SX-200 ICP
ICP VERSION	ICP software version Note: Early releases do not include version information.
V	IP Trunking Version; one of the following:
	0 - 3300 ICP Rel. 3.3 or earlier; 200 IP node; SX-200 ICP Rel. 1.x
	1 - 3300 ICP Rel. 4.0 - 4.1
	2 - 3300 ICP Rel. 5.x; SX-200 ICP Rel. 2.x
LL	Local link number assigned by the local switch
RL	Remote link number assigned by the remote switch
SID	System ID assigned by the local switch

Device Status Reports

Device status reports display call processing information for devices (sets, stations, consoles, trunks, receivers, etc.) that are programmed into the system.

A device status report displays the call processing work area (CP_DWA) of a device. The figure below shows an example of a CP_DWA of a tie trunk.

Figure: Device Status Report

To display the CP_DWA for a device, press the following softkeys:

REPORTS
SHOW
STATUS
BAY/SLOT/CIRCUIT
(enter the PLID.... of the desired device. Note that you can also specify a device by its extension/trunk number, or by its software identification (SWID))
CP_DWA

The device status report for the device appears on the terminal screen. Softkeys, available at the bottom of the report, allow you to display the device status reports of other devices that are connected to, or interacting with, the currently displayed device.

Note that pressing a softkey will have no effect if the softkey does not apply to the current state of the device. For example, pressing the OTHER softkey when a device is not connected (in conversation) with another device, has no effect on the display.

Monitor T1 Trunk Activity

The user can monitor the activity on the ST bus to the T1 Trunk Module by displaying a Monitor T1 Trunk Activity report on the maintenance terminal screen. A Monitor T1 Trunk Activity report provides data for the DSTi, DSTo, CSTi, and CSTo buses.

This feature also allows the user to perform the following functions:

- Send a yellow alarm to the receiving end
- Put the T1 Trunk card into analog loopback mode
- Manually select the synchronization source for the T1 link
- Send data values down the selected trunk
- Change the transmit and receive gains
- Change the A B signaling bits
- Put selected trunks into digital loopback.

The figure below shows a sample Monitor T1 Trunk Activity report.

Figure: Monitor T1 Trunk Activity Report

To display a Monitor T1 Trunk Activity report, press the following softkeys when in maintenance mode:

REPORTS
SHOW
STATUS
BAY/SLOT/CIRCUIT
(enter PLID.....)
LINK_STATUS (This softkey only appears if the affected circuit is a T1 circuit)
TRUNK-NUM
Enter the circuit location on the T1 Trunk Card (24 T1 circuits per T1 Trunk Card)
ENTER

Data from the DSTi, DSTo, CSTi, and CSTo buses of the selected trunk is presented on the screen.

To display a Monitor T1 Framer Status Report, press the following softkeys when in maintenance mode:

REPORTS
SHOW
STATUS
BAY/SLOT/CIRCUIT
(enter PLID.....)
FRAMER_STAT (This softkey will provide a full dump of the 10 pages of the T1 framer chip.)
UPDATE
RETURN

To send test data, press the following softkeys:

TRUNK-NUM
Enter the circuit location on the T1 Trunk Card (24 T1 circuits per T1 Trunk Card)
ENTER
MORE_KEYS
Enter the test data - hex 00 to FF (Note: if 00 is entered, no test data is sent; if nothing is entered, test data 00 is sent)
ENTER

To manually change the clock source while monitoring, press the following softkeys:

MANUAL
Enter the bay number of new source and press RETURN *
Enter the slot number of new source and press RETURN *
ENTER
* This number must be the locations of a functioning T1 Trunk Card.

To return to the original clock source:

AUTO
ENTER

To start a loopback test:

LOOPBACK
INTERNAL/EXTERNAL/CLEAR (See notes)
ENTER

Notes:

1. INTERNAL LOOPBACK causes the DX to route transmit data signals back into the receive channels.
2. EXTERNAL LOOPBACK causes the DX to route receive data signals back out to the transmit channels.
3. CLEAR clears the loopback test that is currently in progress.
4. If the T1 link is not in synchronization, or is transmitting a yellow alarm, or if the TX or RX pads are not set to zero, the data received is altered. Bit 0 changes because of the transmission of AB bits.

Softkey Definitions

The following table provides definitions for the softkey commands that you can enter from the maintenance terminal to test the T1 link.

Table: Softkey Definitions

Softkey	Definition
TRUNK-NUM	Prompts for target trunk. Defaults to 01. Valid is 01-24.
CANCEL	Goes back to REPORTS LEVEL form. The default settings for all control signals will be set.

Table: Softkey Definitions

Softkey	Definition
YELLOW	Toggles bit that forces the T1 module to send a yellow alarm to the far end or removes the yellow alarm condition (bipolar).
RETURN	Goes back to the SHOW STATUS summary form.
MANUAL	This softkey will allow the user to change the sync source. The user will be prompted for Bay/Slot.
AUTO	This softkey will restore the original synchronization mode.
SEND	Prompts user for data to be transmitted. Data is sent continuously down the channel currently selected. Valid input is 00-FF (hex).
TxPadz	Increments transmit attenuation control bits.
RxPad	Increments receive attenuation control bits.
Tx/AB_00	Transmits A = 0 , B = 0
Tx/AB_01	Transmits A = 0 , B = 1
Tx/AB_10	Transmits A = 1 , B = 0
Tx/AB_11	Transmits A = 1 , B = 1
LOOPBACK - INTERNAL	Analog loopback. Toggles the loopback relay for the card. Loops the signals back through the hardware in the card.
LOOPBACK - EXTERNAL	External loopback causes the DX to route receive data signals back out to the transmit channels. External loopback function sends PBX “A” “B” signaling bits, not incoming “A” “B” bits.
LOOPBACK_D	Digital loopback. Toggles the loopback bit for the selected channel. The DS1 channel is looped internally to replace the corresponding receive channel.
UPDATE	This softkey updates the values on the screen.

Definition Of Maintenance Terminal Display

The information in the following table is displayed for information only; it cannot be altered from the terminal.

Table: Definition Of Maintenance Terminal Display

Display	Meaning
DSTi	Data ST bus Input (32 channels 24 active).
DSTo	Data ST bus Output (32 channels 24 active).
CSTi	Control ST bus Input (32 channels 25 active).

Table: Definition Of Maintenance Terminal Display

Display	Meaning
CSTo	Control ST bus Output (32 channels 25 active).
SYNC	SYNC indicates whether there is synchronization to the RECEIVED DS1 link.
SLIP	This bit changes state once a slip condition occurs between the RECEIVED DS1 data and the ST-BUS data.
BPV	This bit changes state after 256 bipolar violations, other than the B8ZS code, within a sample period of 200 ms.
XS1	This EXTERNAL SCAN POINT bit contains the data sampled at the XS1 pin once per frame.
RxYLW	RxYLW indicates that a yellow alarm is being received on the RECEIVED DS1 link.
XCTL	XCTL indicates whether the link is active or not (act, inact).
B8ZS	B8ZS bit value in control register.
8KHZ	8KHz bit value in control register. If 1, then the 8khz pin is low for received channels 1 to 15 and high for channel 16 to the S-bit.
TxYlw	TxYlw indicates whether a yellow alarm is being sent down the link.
DAC	The current value being written to the DAC. The DAC is on the T1 Clock Interface Module. Its purpose is to provide the ability to adjust the system clock. This is accomplished by writing a 12-bit word to the DAC.
SRCE	This field gives the current synchronization source (Bay/slot/circuit).
MODE	This is the current mode of operation. There are three modes: AUTO (AUTO), FREERUN (FREE), or MANUAL (MANU).
	AUTO - the T1 process is adjusting the system clock to lock on to the incoming 1.544 Mhz signal. The link that the process looks at for an external source is based on the order of the links in the network synchronization form.
	FREERUN - the system clock is not being adjusted to lock on to the incoming 1.544 Mhz signal. The reason is that there is nothing programmed in the network sync form or the links all exceed the error threshold.
	MANUAL - the T1 process is forced via maintenance to look at a particular link as an external source. The system clock is adjusted to this link and locks to this source. The T1 process will automatically switch external sources after 24 hours or if the error threshold has been exceeded.
Tx	This is the current data sample for the transmit side of the channel that is being monitored.
Rx	This is the current data sample for the receive side of the channel that is being monitored.
TxA, TxB	TxA and TxB are the transmit A and B bits used for controlling channels on the DS1

Table: Definition Of Maintenance Terminal Display

Display	Meaning
	link.
TxPD	The per channel transmit attenuation control bits.
RxA, RxB	RxA and RxB are the receive A and B bits used for monitoring channels on the DS1 link.
RxPD	The per channel receive attenuation control bits.
LPBK	LPBK indicates whether any loopbacks have been activated: no - there are none Ext - External loopback has been activated for the card Int - Internal loopback has been activated for this card dig - digital loopback has been activated for this channel (is not seen if the card is in analog loopback).

Data Fault Analysis Procedures

Generic information and procedures for analyzing datasets and data-related problems is provided here; refer to it for troubleshooting problems before referring to specific data device troubleshooting charts.

Table: Possible Causes Of Data-related Errors

ERROR	CALL STATE	POSSIBLE CAUSE				
DATASET	**CABLING**	**DX**	**DLC**	**DTE / DCE**		
CRCERR	ANY	Yes (DNIC/HDLC)	Yes	Yes	Yes (DNIC/HDLC)	No
RESETS	ANY	Yes	Yes	Yes	Yes	Yes
LINK FAILURES	CALL SETUP OR TALKING	Yes	Yes	Yes	Yes	No
LINK ABORTS	CALL SETUP OR TALKING	Yes	Yes	Yes	Yes	Yes
PARITY	ANY	Yes (UART)	No	No	No	Yes (Connectors)
OVERFLOW	TALKING	No	No	No	No	Yes
OVERRUN	TALKING	No	No	No	No	Yes
FRAMING	TALKING	No	No	No	No	Yes
NOSYNC	ANY	Yes	Yes	No	Yes	No

CRC Error: A CRC (Cyclic Redundancy Check) error will be logged whenever the HDLC chip reports a CRC ERROR, FIFO OVERFLOW, or FRAME ABORT. They are recorded on both the B and the D channel. The probable cause is a hit on the transmission line; the protocol usually recovers gracefully. If the errors become so bad that the protocol cannot continue to run, then a link reset will occur. If the link resets, the link reset may not be successful - if the dataset is on the B-channel, it will return to the D-channel with a disconnect reason of link abort.

CRC errors happen on one end of a call if the dataset at the other end is unplugged. The connected dataset records a large number of CRC errors, followed by a link reset, and then a link abort.

Link Resets: A link reset occurs when the dataset sends a message, does not receive a response, retransmits the message more times, and still cannot get a response. At this point, the dataset will log a link reset, and then try to re-establish communication by sending SABMs (Set Asynchronous Balance Mode) to the far end. Link resets can occur on both B and D channels.

Link Aborts: A link abort occurs when the dataset, after sending a SABM 8 times, cannot get a valid response from the far end. Thus, a link abort often follows a link reset (specifically when the link reset happened because of many transmission line errors, set unplugged, or circuit switch path broken).

Link aborts can also occur just after the dataset is sent to the B-channel. The dataset will send up to 64 SABMs in an attempt to achieve communication; if it does not receive a valid response, a link abort will occur, and the dataset will return to the D channel. There will NOT be an associated link reset (because the link was never in a “normal” state).

Link Failures: There are two events which occur on the B-channel that can cause a link failure:

- If the dataset is connected on the B-channel, in “normal” mode (NOT go-ahead mode), and the dataset receives an idle “1”s pattern rather than flags for more than 0.5 second, the dataset will disconnect from the B-channel, with a B-to-D reason of link failure
- If the dataset is connected on the B-channel in go-ahead mode, and the dataset does NOT receive a go-ahead after transmitting flags, the dataset will return to the D-channel, and report a link failure.

Overflows: Overflows occur when the device attached to the dataset sends data to the dataset faster than the dataset can send it off to the far end. Two scenarios are:

- Dataset A is at a high baud rate, dataset B is at a lower baud rate, and flow control is NOT enabled on the datasets. In this case, overflows will occur in dataset A. This should NEVER happen; software should not allow two datasets at different baud rates to communicate unless flow control is enabled.
- Dataset A and dataset B both have flow control enabled. The device attached to dataset A is transmitting a large amount of information. The device attached to dataset B flow-controls dataset B. Dataset B stores up as much data from A as it can, then tells A to stop transmitting. Dataset A sends a flow control character to the attached device, but the device ignores it and continues to transmit. Overflows occur. Overflows can happen as a result of a defective attached device, an attached device not having flow control enabled, or the dataset using different flow control characters or kind of flow control (CTS) than the device is expecting

I/F Framing Errors: Framing errors occur because the device is set at a different baud rate from the dataset. A common cause is that the user changes the terminal baud rate during a session (or while idle, if the dataset is not programmed for autobaud operation). One possible scenario is:

- A user establishes a call at 1,200 baud.
- The user decides that is too slow.
- The user then sets his terminal baud rate to 9,600.

Framing errors occur and the user cannot communicate, because the dataset is still at 1,200 baud. The only recovery is to disconnect the call and start over.

NOSYNC Errors: NOSYNC errors occur when the dataset has lost synchronization with the SX-200 system; loss of synchronization usually occurs when a dataset has been powered off or a data connection has been broken.

Power Up Self Test Causing the Dataset to be Busied Out: If a dataset fails its power up self test, it will be busied out. Maintenance logs will have two log entries: the busied out log and the power up self test failure reason. If a new dataset is installed that passes the power up self test, the device will be returned to service, without an installer using a maintenance command.

Error Reports

Show Errors

The Error Reports provide the maintenance user with an up-to-date record of all the transmission checksum errors that have occurred since the system was initialized, or since they were last cleared (see Clearing Error).

To obtain an error report, press the following softkeys:

SHOW
ERRORS
DEVICE_TYPE

At this point the following softkeys are presented for device selection:

DIGITAL_SETS
HDLC
DATASETS
CONSOLE
T1_TRUNK
ENTER

The system outputs the error data in the applications area of the screen. In cases where the data requires more space than is available on the screen, the user is prompted as follows:

- Select the MORE softkey for more data.
  OR
- Select the CANCEL softkey to cancel the output.

Examples of error reports follow.

Digital Set Errors

Figure: Example of Digital Set Error Report

Explanation of error types:

No Sync: A synchronization signal is sent between the digital circuit and the set. The digital circuit monitors the physical line for the presence of this signal. Synchronization is lost when a set is unplugged or when a line is very noisy (external interference or bad connection). A loss of synchronization increments the ‘no sync’ counter; however, maintenance reports are updated only after every 25 occurrences.

In normal operation, this counter will not exceed about 50 for the operating life of a set. A set will get about 25 occurrences each time the set is unplugged or the system is reset.

Investigate a set that gets 50 or more ‘no syncs’ a day:

- check tip/ring connection (remove bridge taps, proximity to noise sources, loop length)
- swap set
- swap circuit card.

Link Reset: A link reset will occur when communications between a digital set and circuit is torn down. In most cases, the reset is because a set was unplugged or the system was reset; however, it can also occur because of a protocol violation.

In most cases, a set will get only one link reset during its life. In a 24-hour period, 20 or more link resets may affect set operation and the problem should be investigated.

Retransmits: The retransmit counter increments each time a digital circuit has to resend just sent information to a set. If the set has not responded to having received the just sent information, the circuit resends.

This problem can be caused by:

- a noisy line
- bad set
- bad circuit
- software error (protocol violation).

In most cases, a set will have no retransmits in a 24-hour period; however this counter can be influenced by the amount of traffic to the set. The more messages sent to a set, the greater the possibility that a message may not make it to that set and will have to be retransmitted. Investigate more than 50 retransmits in a 24 hour period although the user will probably not observe any problem at this rate.

Checksum: Bad checksum in set ROM will report error to maintenance.

HDLC Link Errors

Figure: Example of HDLC Link Error Statistics Report

Explanation of HDLC Link Error types:

HDLC Link errors are explained following:

ERROR	BAY TO MC	MC TO BAY	Error Rate Expected
TX_ERR	Transmitter underrun.	Transmitter underrun.	none
RX_ERR	Undefined error.	Undefined error.	none
OVRFLW	Bay HDLC receiver fifo overflowed.	Main HDLC receiver fifo overflowed.	none
ABORTS	Bay received HDLC ABORT sequence from main.	Main received HDLC ABORT sequence from main.	see note 1
ODDPKT	Bay received packet with odd number bytes.	Main received packet with odd number bytes.	see note 2
RETRAN	Bay retransmitted packet.	Main retransmitted packet.	see note 2

Notes:

1. Dependent upon message traffic on a main-bay link, but should be low, less than 10.
2. Dependent upon message traffic on a main-bay link, but should be low, generally less than 30 per 24 hours.

Console Errors

Figure: Example of Console Error Report

Explanation of error types:

No Sync: A synchronization signal is sent between a digital circuit and a console. The digital circuit monitors the physical line for the presence of this signal. Synchronization is lost when a console is unplugged or when a line is very noisy (external interference or bad connection). A loss of synchronization increments the 'no sync' counter; however, maintenance reports are updated only after 25 occurrences.

In normal operation, this counter will not exceed about 50 for the operating life of a console. A console will get about 25 occurrences each time the console is unplugged or the system is reset.

Investigate a console that gets 50 or more 'no syncs' a day.

- check tip/ring connection (remove bridge taps, proximity to noise sources, loop length)
- swap console
- swap circuit card.

Link Reset: A link reset will occur when communications between a digital console and circuit is torn down. In most cases, the reset is because a console was unplugged or the system was reset; however, it can also occur due to a protocol violation.

In most cases, a console will get only one link reset during its life. In a 24-hour period, 20 or more link resets may affect console operation; investigate the problem.

Retransmits: The retransmit counter increments each time a digital circuit has to resend just sent information to a console. If the console has not responded to having received the just sent information, the circuit resends.

This problem can be caused by:

- a noisy line
- bad console
- bad circuit
- software error (protocol violation).

In most cases, a console will have no retransmits in a 24 hour period; however this counter can be influenced by the amount of traffic to the console. The more messages sent to a console, the greater the possibility that a message may not make it to that console and will have to be retransmitted. Investigate more than 50 retransmits in a 24-hour period although the user will probably not observe any problem at this rate.

Checksum: Bad checksum in CONSOLE ROM will report error to maintenance.

DATASET Errors

Figure: Example of a DATASET Error Statistics Report

Explanation of error types:

FAILRS: The dataset sends flags requesting communication but does not receive acknowledgment from the Digital Line Card; the HDLC on the DLC is time-shared with up to 12 datasets, and sends "go-aheads" to each dataset when it is ready to communicate; if the HDLC is busy with one dataset for too long, other datasets will not receive "go-ahead".

This error can also apply to situations which involve link layer errors such as failing to achieve link reset after 64 tries.

Check the dataset and the Digital Line Card.

ABORTS: The link is up but the command - response exchange (Set Asynchronous Balance Mode - Unnumbered Acknowledgment) does not succeed. The SABM-UA could happen between the dataset and the SX-200 system when they are programmed as DTRX, or between the two datasets.

This error can also apply to situations such as received idle HDLC link when flags were expected.

Check the dataset and the Digital Line Card.

CRCERR: number of retransmissions on the link; this value is set to zero when DNIC synchronization is lost.

The line is noisy or of poor quality.

RESETS: number of times the link initiated link reset; this value is set to zero when DNIC synchronization is lost. This number is also incremented by link aborts and link failures.

Check the dataset and the Digital Line Card.

PARITY: number of bytes received from the attached device with parity errors; this value is set to zero when DNIC synchronization is lost.

Check that the DTE device and the dataset have the same parity settings.

OVRFLW: number of buffer overflows in the following cases; overflows can be caused because the dataset cannot flow control the DTE device.

DATASET 2103: (Sync_mode) number of overflows of PLL buffer

DATASET 2103: (Async_mode) overflows on receive information from the locally attached device.

DATASET 1103: (Async_mode) overflows on receive information from the locally attached device.

Note: DATASET Error values are set to zero when DNIC synchronization is lost.

NOSYNC: This occurs when there has been loss of synchronization between the dataset and the system.

The most common cause is a disconnected dataset.

T1 Trunk Errors

Figure: Example of a T1 Trunk Error Statistics Report

Explanation of error types:

HOUR - data is accumulated hourly

SLIPS - number of data slips due to internal and external timing clocks

FRAME - number of framing errors

BIPOLAR - number of bipolar violations

STATUS - appears only for the current hour and shows current link status. Valid values are:
clear - when there is no alarm condition on the link
active - not currently used
yellow - receiving a yellow alarm
red - link is in a red alarm condition
shrt term - this link is the current sync source and is using the short term formula to adjust the system clock
long term - this link is the current sync source and is using the long term formula to adjust the system clock

STATE - appears for the current hour only and shows the current link state. Valid values are:
no sync - the status of the link is red because it is not in sync
no power - the status of the link is red because it has a power fault
active - there is no alarm on the link
inactive - there is an alarm condition on the link.

Clear Error Counter

For Specific Devices

To clear the Error Counter for a specific device, press the following softkeys:

CLEAR
ERRORS
DEVICE_TYPE

The following softkeys are presented for device selection:

DIGITAL_SETS
HDLC
CONSOLE
DATASETS
T1_TRUNK
ENTER

Note: T1 Trunk errors are tracked on a 24-hour basis. Every hour that the T1 Trunk operates, it generates a new report. The error count is a series of 24 one-hour reports; the error count is updated every hour; the oldest entry is deleted.

If CDE Form 42, T1 Link Descriptors, is changed or a new card is plugged in, the counter is cleared.

For Specific Circuits

To clear the Error Counter for a specific circuit, press the following softkeys:

CLEAR
ERRORS
BAY/SLOT/CCT (enter the required bay, slot, circuit and sub-circuit numbers, pressing the RETURN key after each one)
ENTER

For Specific Extensions

To clear the counter for a specific extension, press the following softkeys:

CLEAR
ERRORS
EXT-NUM
(enter the required extension number, then press the RETURN key)
ENTER

For All Devices

To clear all error counters, press the following softkeys:

CLEAR
ERRORS
ALL
ENTER
CONFIRM
ENTER

The user may verify the error counter clearing via the “SHOW ERRORS” command, which operates in the Reports Level.

Show Channel Map

The channel map report provides the maintenance user with the current status of the system's PCM links. The user may choose between either PHYSICAL or LOGICAL channels. Physical links will show what bay the link is connected to if it is used for voice connection. To obtain a channel map report, press the following softkeys:

SHOW
CHANNEL-MAP
PHYSICAL
LINK-NUMBER or BAY NUMBER
(enter the desired LINK or BAY number, then press the RETURN key)
ENTER

or

SHOW
CHANNEL-MAP
LOGICAL
CHANNEL-NUM
(enter the desired CHANNEL number, followed by the RETURN key or simply press the RETURN key to view all busy channels)
ENTER

The system outputs the channel map report in the applications area of the screen. In cases where the data requires more space than is available on the screen, the user is prompted to request more data by pressing the MORE softkey, or to cancel output by pressing the CANCEL softkey. An example of a physical channel map report is shown in this Figure. The following table gives a summary of the terms used in the channel map report.

Figure: Example of a PHYSICAL CHANNEL MAP Report

Channel Map Terms

Table: Terms Used In The Channel Map Report

Term	Meaning
channel	- Channel number
Rx	- Receive channel
Tx	- Transmit channel
free	- ready for use by CP
cp_busy	- currently in use by CP
down	- currently unavailable to CP
mt_busy	- being tested by maintenance
b_syout	- busied out by maintenance
music	- music on hold (MOH module and DMP unit)
ringbk	- ringback

Table: Terms Used In The Channel Map Report

Term	Meaning
tone a	- channel connected to tone a
tone b	- channel connected to tone b
tone c	- channel connected to tone c
misc	- channel connected to misc tone
faulty	- failed test: unavailable to CP
os_busy	- channel used for messaging between the bay and main control cabinet
os_msg	- channel used by operating system

Maintenance Log Functions

The maintenance log records all maintenance-related information, including anything that affects the functioning or the capacity of the system. Typical maintenance log entries would be circuits that fail diagnostics, cards that have been unplugged, and alarm level changes. The user may read, delete and print log entries, as well as send them to an e-mail address or upload them to an FTP server. For the READ, PRINT and DELETE commands, the following qualifiers apply:

ALL - causes all log entries to be read, printed, or deleted.

NEWEST - causes the most recent user-defined number of log entries to be read, printed, or deleted.

OLDEST - causes the oldest user-defined number of log entries to be read, printed, or deleted.

For further information on the maintenance log, refer to the General Maintenance Information section, and to the Troubleshooting section. To access the logs level commands, press the LOGS softkey. All of the following operations are possible while in the logs level. The Logs Level Functions table offers a quick reference for all log operations except CANCEL and ENTER. The user can press the CANCEL softkey at any time to exit the current operation without committing (saving) any changes, or press the ENTER softkey, when it is available, to commit changes.

Reading Log Entries

Note: Logs cannot be read and printed concurrently.

All Log Entries

To read all maintenance log entries, press the following softkeys:

READ
ALL
ENTER

Newest Log Entries

To read the newest user-defined number of maintenance log entries, press the following softkeys:

READAT
NEWEST
(enter the number of log entries to be read)
ENTER

Oldest Log Entries

To read the oldest user-defined number of maintenance log entries, press the following softkeys:

READ
OLDEST
(enter the number of log entries to be read)
ENTER

In all cases, the system will output the requested number of log entries into the Applications Area of the screen. In cases where the log data requires more space than is available on the screen, the user is prompted to request more log data via the MORE softkey, or to cancel the output via the CANCEL softkey. An example of reading logs is shown in this Figure.

Logs Level Functions

Table: Logs Level Functions

LEVEL	COMMAND	PARAMETER	QUALIFIER	QUALIFIER	QUALIFIER	QUALIFIER
7-LOGS	1-SET	4-AUTOPRINT	1-ON			
			2-OFF			
	2-READ	1-NEWEST [number]				
		2-OLDEST [number]				
		4-ALL				
		8-LOGS_TEXT*				
		9-TRACE_INFO*				
	3-PRINT	1-NEWEST [number]				
		2-OLDEST [number]				
		4-ALL				
		8-LOGS_TEXT*				
	4-DELETE	1-NEWEST [number]				
		2-OLDEST [number]				
		4-ALL				
		9-TRACE_INFO*				
	6-GET_SYS_LOGS					
	7-TO_EMAIL					
	8-STOP					

Table: Logs Level Functions

LEVEL	COMMAND	PARAMETER	QUALIFIER	QUALIFIER	QUALIFIER	QUALIFIER
	9-TO SERVER					

Note: TRACE_INFO and LOGS_TEXT are diagnostic tools that only Mitel personnel use.

Figure: Example of a LOGS READ Display

Deleting Log Entries

All Log Entries

To delete all of the maintenance log entries, press the following softkeys:

DELETE
ALL
ENTER
CONFIRM
ENTER

Newest Log Entries

To delete the newest user-defined number of maintenance log entries, press the following softkeys:

DELETE
NEWEST
(enter the number of log entries to be deleted)
ENTER

Oldest Log Entries

To delete the oldest user-defined number of maintenance log entries, press the following softkeys:

DELETE
OLDEST
(enter the number of log entries to be deleted)
ENTER

In all cases, the system will echo the command into the applications area of the screen. The user may verify that the particular log entries have been deleted by using the READ command.

Printing Logs on System Printer

Note: Logs cannot be read and printed concurrently.

All Log Entries

To print all of the maintenance log entries onto the printer that was defined during Customer Data Entry, press the following softkeys:

PRINT
ALL
ENTER

Newest Log Entries

To print the newest user-defined number of maintenance log entries, press the following softkeys:

PRINT
NEWEST
(enter the number of log entries to be printed)
ENTER

Oldest Log Entries

To print the oldest user-defined number of maintenance log entries, press the following softkeys:

PRINT
OLDEST
(enter the number of log entries to be printed)
ENTER

In all cases, the system echoes the command into the applications area of the screen.

Setting Print Device

Log entries are produced at the device named in CDE Form 34, DIRECTED IO. See the Program CDE section for details.

Setting and Stopping Automatic Printing

Maintenance log entries may be printed without the need of a maintenance user to explicitly request printing using the “PRINT” command. Automatic printing eliminates the risk of losing maintenance log information because of overflow. When the maintenance log contains 75% new (unprinted) log entries, the new entries are automatically printed. Once this initial 75% is printed, logs are printed frequently thereafter (usually four at a time). When logs are deleted, the system accumulates 75% of entries, then prints them out followed by groups of four, until the logs are deleted again. The maintenance log contains a maximum of 96 log entries.

To initiate the automatic printing of logs, press the following softkeys:

SET
AUTOPRINT
ON
ENTER

The system echoes the command into the applications area of the screen.

To turn off automatic printing, press the following softkeys:

SET
AUTOPRINT
ON
ENTER

To stop printing of logs in progress, press the following softkeys:

STOP
ENTER

Sending Logs to an E-mail Address or FTP Server

To send logs to an e-mail address, program the SMTP server IP address in Form 49, the e-mail address in Form 52, and the system hostname in Form 47, Subform 01.

To send logs to an FTP server, program the server IP Address, Username, and Password in Form 47, Subform 01, System IP.

Also, System Option 126, Email Messaging must be enabled in Form 04, and System Option 81 should be set to the time zone that the SX-200 ICP is in.

Logs e-mailed or uploaded an to an FTP server are compressed in a .tar file. The estimated size of compressed file is no more than 1MB. Use WinZip® or other compression utility to open the file.

To e-mail logs, press the following softkeys:

LOGS
TO_EMAIL
ENTER
CONFIRM

To send logs to an FTP server:

LOGS
TO_SERVER
ENTER
CONFIRM

Retrieving Logs and other System Files using Kermit

This procedure is an alternative to emailing the logs and system files or uploading them to an FTP server. Use it to retrieve the files via a modem connection to an SX-200 ICP that is installed for voice only.

You must have a communications package that supports the Kermit protocol to use this procedure. HyperTerminal can be used for serial connections; a secure Telnet client that supports SSL/TLS and Kermit can be used for remote connections.

Note: The Mitel Telnet client does not support Kermit.

The retrieved files are compressed in a .tar.gz file. Use WinZip® or other compression utility to open the file.

To retrieve logs and other system files using Kermit:

Establish a local serial connection to the Maintenance Terminal and begin a Maintenance session.

Press,
LOGS
GET_SYS_LOGS
ENTER

The system prompts the user for the file name (see Note below).

Enter the file name.
ENTER

The system then prompts you to exit to a local Kermit session.

Note: Ensure that the file name is meaningful to you. It can include abbreviations for the name of a remote site, the purpose or function of the database, the version of the software, or the number of the database; for example, LogMit2_8.tar.gz. Be sure to include the .tar.gz extension.

ACD Reports Function

The ACD Reports Level commands allow the user to set up reporting parameters for the ACD TELELMARKETER package. Refer to Report Commands for descriptions of the parameters and how they are selected.

Diagnostic Functions

The Diagnostics Level commands allow the user to take equipment out of service and return it to service, and to measure the line settings of trunks. All operations available in diagnostics, except CANCEL and ENTER, are shown in the table below. The user can press the CANCEL softkey at any time to exit the current operation without committing (saving) any changes, or press the ENTER softkey to commit changes.

Diagnostic Level Functions Table

Table: Diagnostics Level Functions

LEVEL	COMMAND	SUBCOMMAND	PARAMETER	QUALIFIER	QUALIFIER	QUALIFIER
3-DIAGNOSTICS	1-RANGE_B_OUT			BAY/SLOT/ CCT/SUBCCT		
	2-CLR_FEATURE	1-FORWARD	1-BAY/SLOT/CCT			
			3-EXT_NUM			
		2-DO_N_DISTURB	1-BAY/SLOT/CCT			
			3-EXT_NUM			
		3-CALL_BACK	1-BAY/SLOT/CCT			
			3-EXT_NUM			
		4-ALL	1-BAY/SLOT/CCT			
			3-EXT_NUM			
	3-BUSY_OUT	1-BAY/SLOT/CCT				
		3-EXT-NUM				
	4-LS_MEASURE	1-LOSS_LEVEL	BAY/SLOT/ CCT/SUBCCT	mW TONE NUMBER		
		2-IMPULSE_RESP	BAY/SLOT/ CCT/SUBCCT	SILENCE TEST NUMBER		
		3-DISTORTION	BAY/SLOT/ CCT/SUBCCT			
		4-ECHOTEST	BAY/SLOT/ CCT/SUBCCT			
	5-RANGE_RTS					
	6-QUIT					
	8-RET-TO-SVC	1-BAY/SLOT/CCT				

Table: Diagnostics Level Functions

LEVEL	COMMAND	SUBCOMMAND	PARAMETER	QUALIFIER	QUALIFIER	QUALIFIER
		3-EXT-NUM				
	9-DISC_TRUNK	1-BAY/SLOT/CCT				

Busying Out Equipment

Peripheral circuits may be placed in a state such that they are accessible only through maintenance. While in this state, the device will appear busy when requested by call processing.

Specific Circuits

To busy out a specific circuit, press the following softkeys:

BUSY-OUT
BAY/SLOT/CCT
(enter the required bay, slot, circuit and sub-circuit numbers; pressing the RETURN key after each)
ENTER

Specific Extensions

To busy out a specific extension number, press the following softkeys:

BUSY-OUT
EXT-NUM
(enter the required extension number; then press the RETURN key)
ENTER

Note: On System Reset as replacement of a card, any circuit that had been busied out, will remain busied out. The exception is DTMF receiver modules, which reboot or power up to the idle state.

Range of Specific Circuits

To busy out a range of trunks on the same card

1. From the Maintenance screen, press the DIAGNOSTICS softkey.
2. Press RANGE_B_OUT.

   The system prompts you to enter the number of the first Bay in the range.
4. Enter the number of the first Bay in the range, and press RETURN.
5. Continue to enter the Slot, Cct, and Sub-cct numbers as prompted, and press RETURN at each prompt.
6. After you enter the range, press RETURN to continue or CANCEL to exit.

   The message "Busy out sequence initiated - check device status for success/failure" is displayed.

Returning Busy Equipment to Service

Specific Circuits

To return a specific circuit to service, press the following softkeys:

RET-TO-SVC
BAY/SLOT/CCT
(enter the required bay, slot, circuit and sub-circuit numbers; pressing the RETURN key after each)
ENTER

Specific Extensions

To return a specific extension number to service, press the following softkeys:

RET-TO-SVC
EXT-NUM
(enter the required extension number; then press the RETURN key)
ENTER

Range of Specific Circuits to Service

To return a range of trunks on the same card to service

1. From the Maintenance screen, press the DIAGNOSTICS softkey.
2. Press MORE_KEYS.
3. Press RANGE_RTS.
   The system prompts you to enter the number of the first Bay in the range.
4. Enter the number of the first Bay in the range, and press RETURN.
5. Continue to enter the Slot, Cct, and Sub-cct numbers as prompted and press RETURN at each prompt.
6. After you enter the range, press RETURN to continue or CANCEL to exit.

Disconnect Trunk

To force release a locked-up trunk, press the following softkeys:

<PS:"List indent">MORE_KEYS
DISC_TRUNK
BAY/SLOT/CCT
(enter the required bay, slot, and circuit numbers; pressing the RETURN key after each)
ENTER

Clear Extension Features - From Remote Terminal

You can use the maintenance terminal to clear CALL FORWARDING, DO NOT DISTURB, and CALL BACK features that are active on an extension. The associated feature keys will also be turned off. You can perform this function from a remote maintenance terminal without requiring a local Attendant console. Only Mitel telephones and industry standard sets may be cleared from the maintenance terminal.

The extension to be cleared may be identified either by extension number or by Bay/Slot/Circuit; standard error messages are returned if invalid values are entered. If the identification or extension number entered is not that of an extension, the following message is displayed: Device type must be a SUPERSET telephone or industry standard telephone.

Enter this application from the maintenance diagnostic menu, as follows:

PRESS SOFTKEY 2 CLR-FEATURE

The screen displays 4 softkey prompts:

1-FORWARD
2-DO-N-DISTURB

3-CALL-BACK
4-ALL

Select the feature (or ALL features) to be cleared, by pressing the softkey.

The screen displays 2 softkey prompts:

1-BAY/SLOT/CCT
3-EXT-NUM

Enter bay, slot, circuit, and sub-circuit numbers (sub-circuit is not used, but is part of standard prompt format) or enter the extension number, when prompted. When you have entered all the information, the screen displays the 0-ENTER softkey prompt and all the entered parameters.

Press 0-ENTER to clear the specified telephone.

Note: Softkey 5-CANCEL is also available with these prompts.

Measuring LS Trunk Line Settings

These tests allow you to obtain optimum circuit descriptor settings for Loop Start (LS) trunks connected to the Analog Main Board and Analog Option Board (onboard ASUs) in the controller. The tests are based on the signals received from the CO. Correct programming reduces the possibility of echo between the trunks and IP phones.

The test results are posted to the maintenance logs and sent to the email address specified on Form 52 - Email (if programmed).

The tests should be performed when the system is first installed, and then, if a setting change is required, during periods of low system activity (early Sunday morning, for example).

The following tests are available:

- **Line Quality:** Measures the loss level and impulse response of the trunk and provides recommended settings for LENGTH and IMPEDANCE on Subform 13, Audio Configuration Table. If AUTO is selected for LENGTH and IMPEDANCE prior to running the test, the settings will be programmed automatically. Otherwise, the settings must be programmed manually. The test results (and the programmed settings if AUTO is selected) are available for review in the maintenance logs.
- **Distortion:** Measures distortion on the line using an external silence (balance) termination number. Run this test only when instructed to do so by Mitel Product Support.
- **Echo Test:** Measures the echo characteristics of the line using a silence (balance) termination number. Run this test only when instructed to do so by Mitel Product Support.

Before you Begin

Before running these tests:

- Program the LS trunks.
- Establish a connection between the LS trunk lines on the AMB/AOB and the local CO.
- Obtain the milliwatt tone and silence (balance) termination numbers from the CO. If the silence (balance) termination number is not available, no action is required; the termination will be provided automatically (Line Quality test only). If the milliwatt tone number is not available, you can program another LS trunk (not the test trunk) to carry the tone in a loopback setup.

Notes:

1. Problems with echo and signal loss may occur on LS trunk lines where the maximum loss level exceeds 8 dB. Refer to your telephone service provider to ensure trunk specifications are met. For more information on trunk length, see Attenuation Levels for Short and Long CO Trunks.
2. Run the tests in VT100 mode, not TTY (line interface) mode.

3. If running the Line Quality test in a loopback setup, the two trunks (the trunk under test and the trunk providing the milliwatt tone) should be symmetrical. Problems may occur when one trunk is physically longer than the other.
4. For additional information concerning analog local loop characteristics, refer to the "Analog Local Loop Characteristics" section of the Engineering Guidelines document which is provided in PDF format on the Mitel Edocs website (http://edocs.mitel.com).
4. While a trunk is being tested, it is out of service.
5. The Line quality test takes approximately two minutes to complete for each trunk being tested. The Distortion test takes 40 minutes per trunk. The Echo test takes 5 minutes per trunk.

Line Quality Test

The Line Quality Test measures the loss level on the trunk with a milliwatt tone, and the impulse response with a silence tone. These tones can be generated with test numbers provided by the CO, or with workarounds.

Specific LS Trunk

To run the Line Quality test for a specific trunk:

1. Enter the milliwatt tone and silence (balance) termination numbers in Subform 13, Circuit Descriptor Options. If the milliwatt tone number is unavailable, program another trunk (not the test trunk) to provide milliwatt tone in Subform 13, Audio Configuration Table. If the silence (balance) termination number is unavailable, take no action; the termination will be provided automatically.
2. The system will automatically busy out the trunk and will return it to service when the test is complete.
3. In Maintenance, enter
   DIAGNOSTICS
   LS_MEASURE
   LINE_QUALITY
   BAY/SLOT/CCT/SUBCCT
   (Enter the bay, slot, circuit and sub-circuit numbers for the trunk being tested.)
   ENTER
   Select IMMEDIATELY to run the test now, or SCHEDULE to run it later in the week.
   (If SCHEDULE is selected, enter the day and time when prompted.)
   ENTER

   The test results are sent to the maintenance logs and emailed to the address specified on Form 52 - Email. The test provides recommended settings for LENGTH and IMPEDANCE on Subform 13, Audio Configuration Table. Note that if LENGTH and IMPEDANCE are set to AUTO, the settings will be programmed automatically when the test is run.

All LS Trunks

To run the Batch Line Quality test for three or more trunks:

1. Enter the milliwatt tone and silence (balance) termination numbers in Subform 13, Circuit Descriptor Options. If the milliwatt tone number is unavailable, program another trunk (not the test trunk) to provide milliwatt tone in Subform 13, Audio Configuration Table. If the silence (balance) termination number is unavailable, take no action; the termination will be provided automatically.
2. In Maintenance, enter
   DIAGNOSTICS
   LS_MEASURE
   BATCH_QUAL
   ENTER
   Select IMMEDIATELY to run the test now, or SCHEDULE to run it later in the week.
   (If SCHEDULE is selected, enter the day and time when prompted.)
   ENTER

The test results are sent to the maintenance logs and emailed to the address specified on Form 52 - Email. The test provides recommended settings for LENGTH and IMPEDANCE on Subform 13, Audio Configuration Table. Note that if LENGTH and IMPEDANCE are set to AUTO, the settings will be programmed automatically when the test is run.

Distortion

Notes:

1. This test requires an external silence (balance) termination number programmed in Subform 13, Circuit Descriptor Options.
2. This test takes approximately 40 minutes to complete for each trunk being tested.
3. Run this test only when instructed to do so by Mitel Product Support.

Specific LS Trunk

To run the Distortion test for a trunk:

1. Program LENGTH and IMPEDANCE for the trunk in Subform 13, Audio Configuration Table.
2. The system will automatically busy out the trunk and will return it to service when the test is complete.
3. In Maintenance, enter
   DIAGNOSTICS
   LS_MEASURE
   DISTORTION
   BAY/SLOT/CCT/SUBCCT
   (Enter the bay, slot, circuit and sub-circuit numbers for the trunk being tested.)
   ENTER
   Select IMMEDIATELY to run the test now, or SCHEDULE to run it later in the week.
   (If SCHEDULE is selected, enter the day and time when prompted.)
   ENTER

   The test results are sent to the logs and emailed to the address specified on Form 52 - Email. Forward the results to Mitel Product Support. No user reviewable output is provided.

All LS Trunks

To run the Batch Distortion test for three or more trunks:

1. Program LENGTH and IMPEDANCE for all trunks in Subform 13, Audio Configuration Table.
2. In Maintenance, enter
   DIAGNOSTICS
   LS_MEASURE
   BATCH_DIST
   ENTER
   Select IMMEDIATELY to run the test now, or SCHEDULE to run it later in the week.
   (If SCHEDULE is selected, enter the day and time when prompted.)
   ENTER

   The test results are sent to the logs and emailed to the address specified on Form 52 - Email. Forward the results to Mitel Product Support. No user reviewable output is provided.

Echo Test

Notes:

1. This test requires an external silence (balance) termination number programmed in Subform 13, Circuit Descriptor Options.
2. This test takes approximately five minutes to complete for each trunk being tested.
3. Run this test only when instructed to do so by Mitel Product Support.

To run the Echo test for a trunk:

1. Program LENGTH and IMPEDANCE for the trunk in Subform 13, Audio Configuration Table.
2. The system will automatically busy out the trunk and will return it to service when the test is complete.
3. In Maintenance, enter
   DIAGNOSTICS
   LS_MEASURE
   ECHO_TEST
   BAY/SLOT/CCT/SUBCCT
   (Enter the bay, slot, circuit and sub-circuit numbers for the trunk being tested.)
   ENTER
   Select IMMEDIATELY to run the test now, or SCHEDULE to run it later in the week.
   (If SCHEDULE is selected, enter the day and time when prompted.)
   ENTER

   The test results are sent to the logs and emailed to the address specified on Form 52 - Email. Forward the results to Mitel Product Support. No user reviewable output is provided.

All LS Trunks

To run the Batch Echo test for three or more trunks:

1. Program LENGTH and IMPEDANCE for all trunks in Subform 13, Audio Configuration Table.
2. In Maintenance, enter
   DIAGNOSTICS
   LS_MEASURE
   BATCH_ECHO
   ENTER
   Select IMMEDIATELY to run the test now, or SCHEDULE to run it later in the week.
   (If SCHEDULE is selected, enter the day and time when prompted.)
   ENTER

   The test results are sent to the logs and emailed to the address specified on Form 52 - Email. Forward the results to Mitel Product Support. No user reviewable output is provided.

Cancelling a Test

You can cancel tests that have not yet started (either scheduled or batch). You cannot cancel a test that is currently being run on a trunk.

To cancel a test:

In Maintenance, enter
DIAGNOSTICS
LS_MEASURE
CANCEL TEST
ENTER

Note: If you press this command while a batch test is running, the test will run to completion for the trunk that is currently being tested. Testing on the remaining trunks in the batch will then stop.

Determining the Amount of Time Remaining for a Test

To determine the amount of time remaining for a test that is scheduled or in progress:

In Maintenance, enter
DIAGNOSTICS
LS_MEASURE
TIME REMAIN
ENTER

Approximate test times are:

- Line Quality: 2 min / trunk
- Distortion: 40 min / trunk
- Echo: 5 min / trunk.

Line Quality Test Log Messages

The Line Quality Test results are written to the maintenance logs, and e-mailed to the address programmed Form 52 - Email.

To view output for the Line Quality Test:

In Maintenance, enter
DIAGNOSTICS
LS_MEASURE
TEST_RESULTS
ENTER

The first line contains the date and time of the test, and indicates whether the test passed or failed. The second line provides the test results in seven components:

<IMPEDANCE> <ERL> <COMMENT1> <LEVEL> <LENGTH> <COMMENT> COLOR>

Table: Line Quality Test message components

Message Component	Description
Impedance	The recommended impedance setting for optimum voice quality: - COMPLEX - CENTREX - DSL - 600 OHM - IMP A - IMP B, C, or D (same as 600 OHM)
ERL	The minimum ERL for the line.
Comment1	Describes anomalies detected (if any) when optimizing impedance. Possible comments include:

- LOADED LOOP: A loaded loop condition can cause echo problems and may require special impedance settings. - CAN'T SEIZE: Unable to seize the test trunk. It may be busy. - NO DIAL TONE: Dialtone was not detected on the trunk. The user should verify the trunk connections. - ERR DIALING: Unable to pulse the appropriate digits. The user should validate the milliwatt tone and silence (balance) termination test numbers. - ERR RECORD: The milliwatt tone and silence (balance) termination was not detected. Validate the configuration.	
Level	The loss measured on the line.
Length	

The recommended length setting for optimum voice quality:

- SHORT - LONG - XLONG	
Comment2	Describes anomalies detected (if any) when optimizing length. See Comment1 for a list of possible comments.
Color	

Describes overall line characteristics:

- GREEN: Recommended.
- YELLOW: Borderline; voice quality problems are possible.
- RED: Not recommended; voice quality problems are likely.

Other Maintenance Terminal Applications

Customer Data Entry (CDE)

The Maintenance terminal is also the main device used for customer data entry (CDE). At the start of the login procedure, the user is queried to start either a maintenance session or a CDE session (see Login Procedures). A user currently in a maintenance session can go to CDE by pressing the **TO CDE** softkey (requires matching access levels for the CDE and MTCE). For further information on CDE, refer to the Program CDE section.

Verification of the Database

The Maintenance terminal is used to check a system for database errors.

- Access CDE
- Select Form 01
- Select Verify Data.

Traffic Measurement

Traffic measurement is a separate level in maintenance. All of the information in the Command Input section applies to Traffic Measurement as well. Refer to Traffic Measurement, for command descriptions and further information. The following table, Traffic Measurement Functions, contains the operations available in traffic measurement functions, except CANCEL and ENTER. The user can press the CANCEL softkey at any time to exit the current operation without committing (saving) any changes, or press the ENTER softkey, when it is available, to commit changes.

Table: Traffic Measurement Functions

LEVEL	COMMAND	PARAMETER	QUALIFIER	QUALIFIER	QUALIFIER	QUALIFIER
5-TRAFFIC_MEAS	1-SET	1-UNITS	1-CCS			
			2-ERLANGS			
		2-PERIOD [number]				
		3-DURATION [number]				
		4-AUTOPRINT	1-ON			
			2-OFF			
		7-START_TIME [hh:mm]	1-PM			
		8-CONDENSED	1-ON			
			2-OFF			
	2-SHOW	3-STATUS				
	3-PRINT					
	4-READ					
	5-IP_MEAS	1-SET	1-SAMPLE			
			2-PERIOD			
			3-DURATION			
			4-MAX_DELAY			
		2-SHOW	3-STATUS			
		3-PRINT				
		4-READ				
		5-TRAFFIC_MEA				
		7-START				

Table: Traffic Measurement Functions

LEVEL	COMMAND	PARAMETER	QUALIFIER	QUALIFIER	QUALIFIER	QUALIFIER
		8-TO_SERVER				
		9-STOP				
	9-STOP	1-TRAFFRPT				
		3-PRINT				

Username Command Privileges

The following table lists the command privileges of all of the valid username levels:

Table: Username Command Privileges

COMMAND	Installer	Maint1	Maint2	Supervisor	Attendant
DATABASE RESTORE	X				
DATABASE BACKUP	X				
LOGS_BACKUP	X				
SET TIME	X	X	X	X	X
SET DATE	X	X	X	X	X
SET PASSWORD	X	X	X	X	X
SET SPEED	X	X			
SHOW DATE	X	X	X	X	X
SHOW TIME	X	X	X	X	X
SHOW DEVICE	X	X	X	X	X
SHOW IDENTITY	X	X	X	X	X
MONITOR DATA SMDR	X	X	X	X	
MONITOR SMDR	X	X	X	X	
MONITOR LOGS	X	X	X	X	
STOP MONITOR	X	X	X	X	
SHOW ERRORS	X	X			
CLEAR LOCKED EXTS	X	X			
CLEAR ERRORS	X	X			

Table: Username Command Privileges

COMMAND	Installer	Maint1	Maint2	Supervisor	Attendant
ENABLE/DISABLE BG DIAGS	X	X			
ENABLE/DISABLE PWR UP DIAGS	X	X			
BUSY-OUT RETURN TO SVC	X	X			
TRAFFIC SET	X	X	X	X	
RESTART	X				
SET RESET_TIME	X				
SET ALARM_THRESH	X				
TRAFFIC SHOW	X	X	X	X	
TRAFFIC PRINT	X	X	X	X	X
TRAFFIC READ	X	X	X	X	X
TRAFFIC STOP	X	X	X	X	X
LOGS SET AUTO	X	X			
LOGS READ	X	X			
LOGS PRINT	X	X			
STOP PRINT LOGS	X	X			
LOGS DELETE	X	X			
SHOW ALARMS	X	X			
SHOW CONFIG	X	X	X		
SHOW STATUS	X	X	X		
SHOW CHANNEL_MAP	X	X	X		

Maintenance Terminal Error Messages

The following table lists status and error messages which may appear at the Maintenance Terminal during a maintenance session:

Table: Maintenance Terminal Error Messages

Message	Meaning
ACD reports already in progress.	In traffic measurement, parameters cannot be changed without

Table: Maintenance Terminal Error Messages

Message	Meaning
STOP first.	stopping the data collection procedure first.
A specific value cannot follow the default value ‘XX’.	If the user has entered a default value for the BAY/SLOT/CIRCUIT prompt, a specific value cannot follow.
An invalid or incorrect password was entered.	Displayed in SET PASSWORD when the old password does not match the one stored in the system, or when the new passwords contain invalid characters (characters other than [‘A’..‘Z’,‘a’..‘z’,0..9]).
Card not installed.	No card is installed at the specified location. Use SHOW CONFIG command to check the state of the card.
Change terminal speed and press RETURN when ready.	An instruction message rather than an error message; appears when changing the speed of the maintenance port while on the maintenance terminal.
Circuit “XX” does not exist on this card OR card not programmed.	Use SHOW CONFIG command to verify installed cards.
Clearing of T1 Trunk errors is not permitted.	The user attempted to clear T1 errors; this is not permitted.
Data value % outside of valid range (0-255).	Enter new value within range of 0 - 255.
Database Corrupt in: templates.	The verification of the database has failed. The template section may be corrupted.
Note: If more than one section is corrupted, one of these may be displayed instead:	- Database Corrupt in: templates and static section. - Database Corrupt in: templates and b-tree. - Database Corrupt in: static section and b-tree. - Database Corrupt in: templates, static section, and b-tree.
Device is currently in use.	Wait until device is idle (device is locked up).
Device not programmed.	An attempt was made to RET-TO-SVC or BUSY-OUT a device by specifying a PLID which is not programmed.
Error with voicemail backup or restore	An attempt was made to restore a database containing voice mail prompts in two or more languages. The attempt failed because the system has fewer languages installed.
Flash write error?	`A download to a 2-megabyte flash memory card was attempted. Card must be 4-megabyte.
Invalid data station specified.	The user has tried to SHOW DEVICE DATASTN_PLID and specifies a PLID which is not a data station.
Invalid day for the month specified. Date not set.	Valid month entries are dependent upon the Roman calendar. For example, an attempt may have been made to set the date to February 30.

Table: Maintenance Terminal Error Messages

Message	Meaning
Invalid parameter “XX”. TIME (or DATE) not set.	Time may be set to 00:00-23:59, date may be set to 01-31 for days, 01-12 for months, 00-99 for years. Valid month entries are dependent upon the Roman calendar.
Invalid qualifier.	The qualifier specified is invalid for this device type.
IP Set Registration failed: DN not found on bay.	The DN received is not programmed on a set on that bay.
IP Set Registration failed: Invalid PIN.	The PIN received is invalid.
IP Set Registration failed: IP Sets limit reached.	The maximum number of IP sets (as determined by System Option 114 in Form 04) are already registered
IP Set Registration failed: MAC already exists.	Another set with that MAC address is already registered in the system.
IP Set Replacement failed: Other set still connected.	The received DN and MAC address belong to a set that is still connected. For security reasons, the set must be disconnected before it can be replaced.
Logical channel specified is not in use.	The user has tried to SHOW CHANNEL-MAP LOGICAL CHANNEL NUM ### which is idle. This function shows only those channels which are in use.
MONITOR LOGS already in use.	A second attempt was made to monitor logs. Monitor logs was already in progress.
Must specify at least a link number.	The user has tried to BUSY-OUT or RET-TO-SVC a pcm channel and uses a wild card for the link parameter.
New System ID Module Installed	The system ID module was replaced by a new and different one.
NO CONSOLE programmed.	Program a console first.
No Data Sets programmed.	Program a data set first.
No Digital Sets programmed.	Program a digital telephone first.
No errors found within specified range.	The user attempted to SHOW ERRORS and the devices specified (DIGITAL_SETS, HDLC, DATASETS, CONSOLE, T1_TRUNK) have no errors. Note: This function always gives the T1 Trunk error form, even though there are no errors.
No loopbacks are presently set.	An analog loopback test has been started after clear loopbacks (internal or external).
No programmed devices within specified range.	There are no programmed devices within the specified range. Use SHOW CONFIG command to check range.
No wild cards allowed for this function.	The user has specified wild cards in a PLID parameter when trying to BUSY-OUT or RET-TO-SVC.

Table: Maintenance Terminal Error Messages

Message	Meaning
Original system ID module installed	The original system ID module was reinstalled in this system.
PRINTER_PLID is invalid as a printer device.	The user has tried to SUSPEND_PRTR or RESUME_PRTR. The specified PLID is not a printer PLID. **Note:** PLIDs for IP sockets used to output printer data include a subcircuit number, although the number is not shown in Form 12 where the PLIDs are programmed. The number is the same for all sockets: 02. Therefore, an IP socket programmed at PLID 1/13/20 is actually at PLID 1/13/20/02, not 1/13/20/00.
Stop of print pending or no print currently in progress.	There was no PRINT process running when STOP PRINT was entered.
System busy, please try again later.	Copy database is inhibited for several minutes after a system reset; maximum time is approximately eight minutes.
System ID Module was removed	The system ID module was removed from the SX-200 ICP.
The access code “XXX” does not exist.	The specified extension number does not exist. Use SHOW STATUS command to check the status code.
The extension number “XXXX” is not a SUPERSET telephone, CONSOLE, or DATASET.	The user has specified an extension number when he tries to CLEAR ERRORS. Errors are only compiled for these types of devices.
The hour value “XX” is out of range. Start time not set.	Used in traffic measurement for invalid values in the SET START TIME command.
The minor threshold may not be greater than the major threshold.	Assign a higher value to the major threshold or else assign a lower threshold to the minor threshold - SET Alarm threshold.
The minute value “XX” is not a multiple of ten. Start time not set.	Used in ACD Report traffic measurement for an invalid minute value in the SET START TIME command.
The minute value “XX” is out of range. Start time not set.	Used in traffic measurement for invalid values in the SET START TIME command.
The value “XX” is invalid for PERIOD	Enter a new value between 1 and 60. Appears when setting a period in traffic measurement or ACD Report traffic measurement
The value “XX” is outside the valid range for SUBCIRCUIT.	The specified sub-circuit number is invalid for this particular device type. Use SHOW STATUS command to verify card type and number of programmed circuits.
There are no logs currently in the database	There are no logs in the database to READ or PRINT.

This function is not available for this device.	The feature requested is not available for this device.
This function is not available for this console.	The feature requested is not available for this console.
This function is not available for this device.	The user has tried to BUSY-OUT or RET-TO-SVC the console.
This function is not available on the console.	The user has tried to perform a command which is not available when logged into maintenance from the console.
TIMEOUT PERIOD EXPIRED. Press Return to login.	After being prompted for the username, the user has ten seconds to begin entering characters.
Traffic measurement already in progress. STOP first.	In traffic measurement, the following parameters cannot be changed while traffic measurement is collecting data: PERIOD, DURATION, and START TIME.
Trunk value “XX” outside of valid trunk range (1-24).	The user requested to monitor the LINK_STATUS of a trunk which was outside the valid range of 24 circuits for the T1_Trunk card.
Unable to print. Maintenance print already in progress.	There can be only one PRINT or READ occurring at one time. If required, enter STOP PRINT command to initiate a second PRINT.
Universal Card. Module “X” in Bay Slot Module is out of range (1-4).	The user has chosen a Universal Card, but the module number is out of range (not between 1 and 4).
Universal Card. Sub_circuit “X” does not exist on module.	The user has chosen a Universal Card; the module number is valid and exists but the sub-circuit number specified does not.
Unrecognized qualifier %s. All others ignored.	The qualifier in “Test” command is unrecognized.
Value cannot be greater than 100.	Assign a new value that is less than 100 for alarm threshold.
Warning. Invalid qualifier found. %s will be ignored.	The qualifier specified during set time will be ignored.
Warning. Sub_circuit field ignored for this card.	Sub-circuit parameter was entered, but was not required. For information only.
Wrong bay type for specified device.	An incompatible bay number is given for the device selected (i.e., Bay 0 is selected for receivers).

Reset Codes

The system banner on the maintenance console is displayed in the following format:

MITEL SYSTEM ROM	**R2.0/1**	**Feb 9 2004**	**(3)**
Standard banner introduction	BootROM version	Build Date	Reset cause (if any)

If a code number is displayed for a reset cause, it will indicate one of the following:

Table: Reset Causes

Code	Definition	Notes
3	Hard reset	Cause of reset is unknown.
13	Checkstop reset	A double bus fault has occurred forcing a reset.
43	Watchdog reset	A software problem has occurred with either a high-priority task being running at a higher priority than the watchdog task or interrupts have been disabled preventing the operating system from performing a context switch.
83	Power-on reset	The system has lost AC power forcing a reset. This normally indicates that the AC plug was removed and re-inserted.
103	Programmed reset	The system software has intentionally restarted the system.
203	Push-button reset	The user has depressed the RESET button on the front of the controller.

General Maintenance Information

Maintenance Tools

The Maintenance Terminal

The RS-232 ASCII Maintenance Terminal is the primary maintenance tool for the SX-200 ICP. There can be only one maintenance session active at any time. The maintenance terminal connects by means of a standard 9-pin RS-232 cable.

A PC can be used as a maintenance terminal to back up a customer database to the PC storage medium by using the Database Backup function.

One end of the cable is plugged into the main RS-232 communication port of the terminal; the other end is plugged into the Maintenance Connector port on the cabinet backplane.

For further information on connecting the Maintenance Terminal, refer to the RS-232 Maintenance Terminal section.

Note: The maintenance terminal user must “LOGOUT” after every session.

The attendant console can be used as an alternate maintenance workstation. All commands available at the maintenance terminal are available at the attendant console.

Maintenance Port

The maintenance port provides a connection to the system for an RS-232 ASCII terminal for maintenance or programming purposes. The terminal is connected to the system through the Maintenance Connector port either directly, or indirectly through a modem and a null modem adapter. The Maintenance

Connector port is located on the rear panel of the Control cabinet; refer to the RS-232 Maintenance Terminal section.

System Reset Return-to-Service Type

Whenever the system resets, it returns to service in the type that it was in prior to the reset (Day, Night 1, or Night 2). Previously, it always returned to service in Night 1.

System Maintenance Log

The system maintenance log is a text file-based record of all maintenance-related information. Any event which potentially affects the functioning of the system is entered into this log. Typical maintenance log entries include circuits that fail diagnostics, cards that have been unplugged, and alarm levels that change. The user may read, delete, and print log entries, as well as set a variety of printing options. For additional information, see Maintenance Log Functions.

Three types of log reports are generated:

- **Fault report** - Call Processing or the maintenance system has detected an error or an abnormal condition.
- **Reset report** - A bay or the SX-200 ICP has reset.
- **Alarm Level Change report** - The overall system alarm level has changed.

Alarms

Alarms allow the SX-200 ICP to determine its own functional state. The Alarm Manager software program monitors the performance of all peripheral devices and resources in the system, and compiles up-to-date statistics on anomalies. The alarm level is determined by the actual or potential effect on service that the anomalies cause. The alarm status level displays in the status line (top right corner) of the maintenance terminal.

Whenever the alarm level worsens, notification can be sent to appropriate personnel at the e-mail addresses entered in Form 52.

Alarm Levels

There are four distinct levels of alarm defined for the maintenance system. These levels provide maintenance personnel with up-to-date information on the severity of existing anomalies. The four alarm levels are:

- **NO ALARM** - indicates that the system is functioning properly.
- **MINOR** - indicates that there are problems affecting the system in small proportion.
- **MAJOR** - indicates that there are problems causing a serious degradation of service.
- **CRITICAL** - indicates that there has been a very serious loss of call processing capability. It automatically invokes system fail transfer (SFT) and resets the system.
- **MOSS** - indicates that the Mitel Options password entered in Form 04 (System Options/System Timers) does not match the selected system options. To clear this condition, enter the correct password and reset the system.

Alarm Categories

Three basic alarm categories relate to peripheral equipment. All problems affecting system performance will fall into one or more of these categories. Failure of other system components will indirectly cause failure of peripheral equipment. The categories are:

- Lines
- Trunks
- DTMF Receivers.

Alarm Types

Because the SX-200 ICP is modular in design, the Alarm Manager keeps alarm statistics in three categories:

- Bay Alarms - alarm levels of the categories specific to each bay in the system.
- System Alarms - alarm levels of the categories on a system-wide basis.
- Overall Alarm - overall system alarm level, derived from all bay alarms and system alarms in all categories. This is the alarm that is displayed on the upper right corner of the console.

Alarm Thresholds

For each alarm category, the thresholds represent the alarm level trip points; that is, the precise divisions between the alarm levels. The Minor and Major Alarm thresholds are simple percentages, indicating availability: the number of working devices is compared to the number of programmed devices. The Critical Alarm threshold is not a percentage, but is a precise numerical value. When the number of available devices falls below this number, a critical alarm is raised.

The administrator can update Alarm thresholds for system devices (Line, Trunks, and DTMF receivers), but not for resources (tasks and components, memory consumption, and memory fragmentation). See Setting Alarm Thresholds and Resource Monitor for more information.

Resource Monitor

The system continually monitors the following resources:

- memory consumption
- memory fragmentation
- stale tasks/components

If a resource passes its threshold, the system generates an alarm and a log record. Critical alarms cause the system to be reset.

Notes:

1. The system will monitor resources and raise alarms regardless of the System Reset schedule.
2. The resource thresholds are set in the system software. You cannot configure them.

Stale Tasks and Components Threshold

The system performs the following steps while monitoring tasks and components:

1. If a non-critical task is suspended, a log is generated and a Major alarm is raised on the system. The system will be reset according to the System Reset schedule.
2. If a critical task is suspended, a log is generated and a critical alarm is raised on the system. The system will reset immediately.

Memory Consumption Threshold

If the amount of free memory becomes less than

- 20 MB, a minor log is generated
- 10 MB, a major log is generated and a major alarm is raised on the system. The alarm is generated to notify the system maintainer that the system should be rebooted. Under normal circumstances, the system should remain up for 2 to 4 days after the alarm is generated. The system will be reset according to the System Reset schedule.
- 2 MB, a critical log is generated and a critical alarm is raised on the system. The system resets immediately.

Memory Fragmentation Thresholds

If the largest memory fragment becomes less than

- 1 MB, a major log is generated and a major alarm is raised on the system. The alarm is generated to notify the system maintainer that the system should be rebooted. Under normal circumstances, the system should remain up for 2 to 4 days after the alarm is generated. The system will be reset according to the System Reset schedule.

- 500 KB, a critical log is generated and a critical alarm is raised on the system. The system will reset immediately.

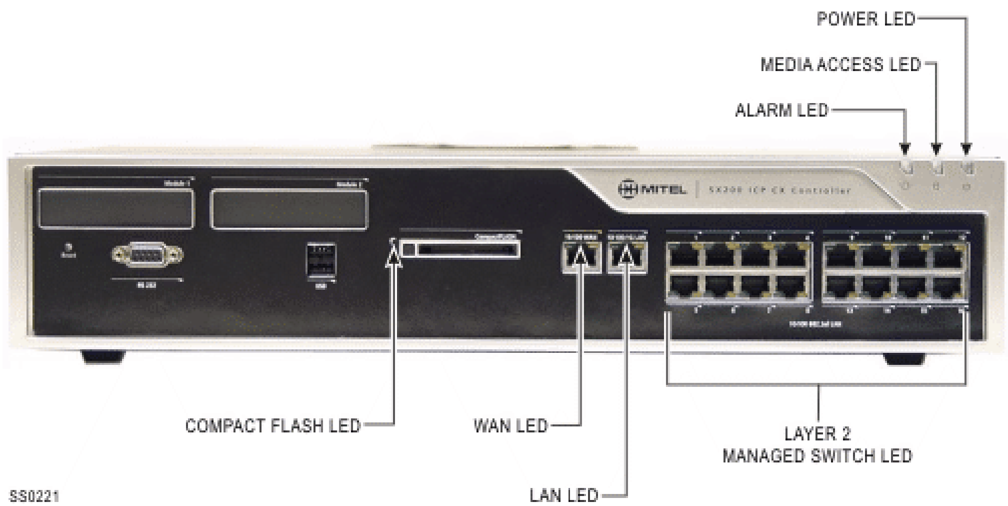
Alarm Totals

The Alarm Manager keeps a record of the total numbers of the various devices that should be available to Call Processing, as well as the actual number that are available. Alarm totals are maintained for each of the alarm categories in each bay, as well as for the entire system. These totals are compared to the alarm thresholds, to determine the level of alarm that is raised.

Maintenance Controls and Indicators

The following paragraphs describe the maintenance controls and indicators found on the components of the SX-200 ICP system. Most of the indicators are software-controlled, and provide maintenance personnel with information on the current status of the system.

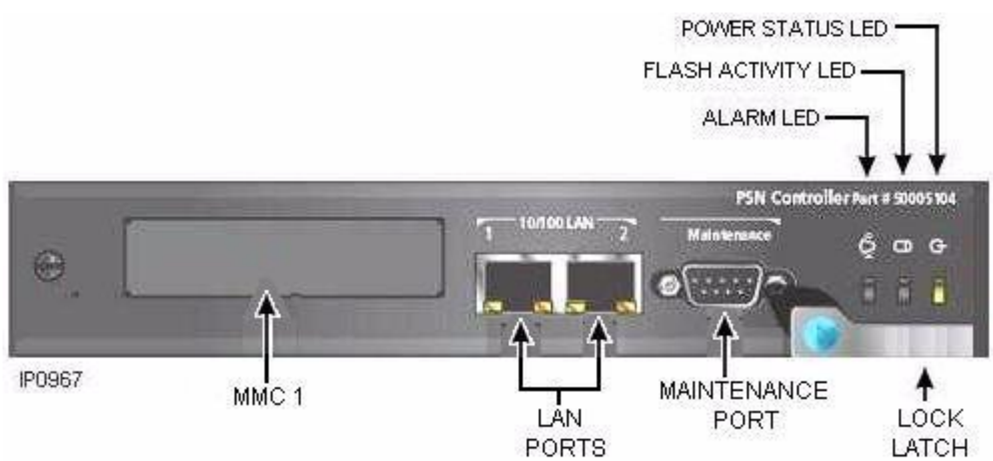
CX/CXi Controller and AX Controller



Status LEDs

Note: The CXi controller is shown above. LED status is the same for the CX.

AX Controller



Power

The Power LED indicates the following by color and activity:

Table: Power LED Status - SX-200 ICP CX/CXi and AX	
LED Status	**Meaning (All LEDs)**
GREEN (on - solid)	Unit is plugged in and operating normally.
RED (two successive flashes)	The unit is starting up properly (seen only during the boot process).
RED (on - two seconds)	The unit is starting up properly (seen only during the boot process).
RED (on - solid)	The unit has detected an error and is being held in reset mode until the problem is corrected.
RED (blinking)	The unit has detected an error during the power-on process and will attempt to reset itself.
OFF	Unit is not plugged in. OR Unit is faulty.

External CompactFlash

The external CompactFlash Card LED indicates the following by color and activity:

Table: External CompactFlash Card LED Status (CX/CXi)	
LED Status	**Meaning (All LEDs)**
ORANGE (on - solid)	Flash card in use, do not remove.
ORANGE (blinking)	Flash card has not been properly formatted and the platform is having difficulty accessing it.
GREEN (on - solid)	Safe to remove Flash card.
OFF	No Flash card detected

CompactFlash/Hard Drive Activity

The CompactFlash/Hard Drive LED indicates the following by color and activity:

Table: CompactFlash Card/Hard Drive (CX)/Flash Activity (AX) LED Status	
LED Status	**Meaning (All LEDs)**
ON	Internal hard disk or CompactFlash is being accessed.
OFF	Internal hard disk or CompactFlash is inactive.
ON/OFF (blinking)	When the System Reset Button is depressed, the CompactFlash/Hard Drive LED will flash on and off.

Alarm

The Alarm LED indicates the following by color and activity:

Table: Alarm LED Status

LED Status	Meaning (All LEDs)
RED (blinking)	Critical Alarm
ORANGE (blinking)	Major Alarm
YELLOW (blinking)	Minor Alarm
RED/ORANGE/YELLOW	When the System Reset Button is depressed, the CompactFlash/Hard Drive LED will flash between red, orange and yellow.
OFF	Normal Operation

10/100 WAN, 10/100/1G LAN, 10/100 802.3af LAN Ports (CXi/AX)

WAN and LAN Port behavior is indicated using two LEDs, one green and one yellow. The WAN/LAN LEDs indicate the following by color and activity:

Table: WAN/LAN Port LED Status

LED Status	Meaning (All LEDs)
GREEN (on)	Link is active.
GREEN (blinking)	Link is active and transmitting or receiving.
GREEN (off)	Link is inactive.
YELLOW (on)	Data transmission/reception is at 100 Mbps (the port speed for the 10/100/1GigE LAN Port can be up to 1 Gbps).
YELLOW (blinking)	Port Error.
YELLOW (off)	Data transmission/reception is at 10 Mbps.

Notes:

1. When the System Reset Button is depressed, the WAN/LAN LEDs will all be on.
2. The 10/100 802.3af LAN port LEDs indicate Ethernet status, not power status.
3. The AX has two 10/100 LAN Ports and there isn't a WAN Port on the AX.

Controls

System Reset Button (RESET): When this manual system reset button is pressed, the system will cease all activity, run all initialization tests, and reload the software.

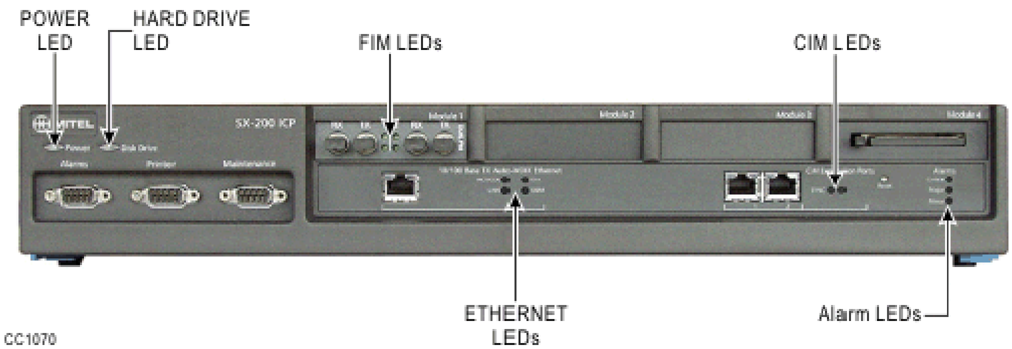
CAUTION: **Never press the System Reset button while the controller is handling traffic because all calls will drop, and the system will switch to System Fail Transfer mode.**

Flash Memory Card Ejector Bottom: The flash memory card containing the system software is inserted into the slot on the controller only while power is OFF.

MX Controller

The MX Controller contains the following status LEDs and controls:

Status LEDS



Power LED

Table: Power LED Status	
LED Status	**Meaning (All LEDs)**
ON	Unit plugged in and operating normally.
OFF	Unit not plugged in. OR Unit is faulty.

Hard Drive LED

Table: Hard Drive LED Status	
LED Status	**Meaning (All LEDs)**
On or Flashing	Reading/writing data to disk/card.
OFF	No flash card or hard disk activity.

FIM LEDs

The top LED indicates the status of local FIM. The bottom LED indicates the status of the remote FIM. The controller FIM monitors the synchronization of the clock appearing on the fiber link from the SX-200 Peripheral cabinet. The FIM in the Peripheral cabinet monitors the synchronization of the clock appearing on the fiber link from the controller (this synchronization state is encoded with other information and sent back to the controller).

Table below shows the meaning of the FIM LEDs.

Table: FIM LED Status

LED Status	Meaning (Both LEDs)
ON	In frame synchronization.
Flashing	Out of synchronization. OR Tx and Rx cables reversed.
OFF	Power off. OR Held in reset.

T1/E1 LEDs (MX, AX and CX/CXi)

Each interface has two LEDs. The upper LED is green; it indicates the status of the interface. The lower interface is red/yellow; it indicates the alarm state of the interface.

The LED indications are the same for the Dual T1/E1 Framer module and the T1/E1 Combo module.

The table below shows the meaning of the T1/E1 LEDs.

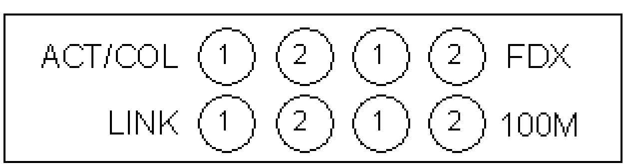
Table: T1/E1 LED Status

Status LED (green)	Alarm LED (red/yellow)	Meaning
Off	Off	Link not programmed or link descriptor not assigned.
Off	Solid Red	Red alarm. Loss of signal; check link connection.
Off	Solid Yellow	Yellow alarm. No signal from remote end; check link with analyzer. (This state is normal during startup.)
Solid Green	Solid Yellow	Blue alarm. Check link with analyzer.
Solid Green	Off	Layer 1 synchronized. Good link state; no alarms.
Flashing Green	Solid Yellow	Alarm indication from remote end.

Ethernet LEDs

The table below shows the meaning of the LAN Ethernet port LEDs.

Table: Ethernet Ports LED Status



	Normal	On Reset
ACT/COL	GREEN: Activity RED: Collision	RED

LINK	GREEN: Link	GREEN
FDX	GREEN: Full Duplex	GREEN
100M	GREEN: 100Mbs	GREEN

CIM LEDs

The table below shows the meaning of the CIM LEDs.

Table: CIM LED Status

LED Status	Meaning (All LEDs)
ON	Communication link established and configured.
Flashing	Link established but not configured.
OFF	No power.

Alarm LEDs

The alarm status LED indicates the overall system alarm level. Refer to the Alarms section for further information on alarms.

Table: Alarm LED Status

LED	Status	Meaning (All LEDs)
Minor	ON	There are problems affecting the system in small proportion.
OFF	No problem; system system is functioning properly.	
Major	ON	There are problems causing a serious degradation of service.
OFF	No problem; system system is functioning properly.	
Critical	ON	There has been a very serious loss of call processing capability; an automatic system fail transfer (SFT) is invoked and the system is reset.
OFF	No problem; system system is functioning properly.	

Flash Card LED

Table below shows the meaning of the Flash Card LED

Table: Flash Card LED Status

LED Status	Meaning (All LEDs)
Orange	Flash card in use, do not remove.
Green	Safe to remove Flash card.
OFF	No Flash card detected

Controls

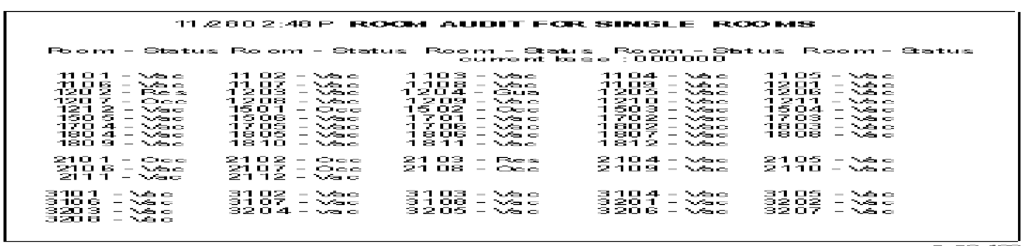
System Reset Button (RESET): When this manual system reset button is pressed, the system will cease all activity, run all initialization tests, and reload the software.

CAUTION: **Never press the System Reset button while the controller is handling traffic because all calls will drop, and the system will switch to System Fail Transfer mode.**

Flash Memory Card Ejector Bottom: The flash memory card containing the system software is inserted into the slot on the controller only while power is OFF.

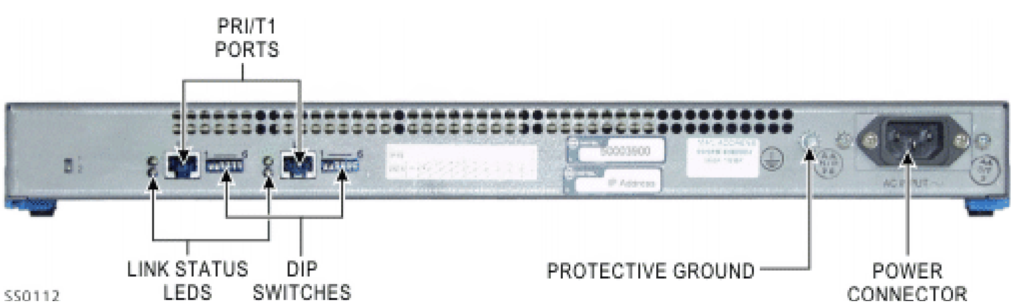
Network Services Unit (NSU)

The NSU contains the following status LEDs


11/28 02:48 P ROOM AUDIT FOR SINGLE ROOMS

current base : 000000

Room - Status	Room - Status	Room - Status	Room - Status	Room - Status
1101 - Vac	1102 - Vac	1103 - Vac	1104 - Vac	1105 - Vac
1106 - Vac	1107 - Vac	1108 - Vac	1109 - Vac	1201 - Vac
1202 - Res	1203 - Vac	1204 - Gua	1205 - Vac	1206 - Vac
1207 - Occ	1208 - Vac	1209 - Vac	1210 - Vac	1211 - Vac
1212 - Vac	1501 - Occ	1502 - Occ	1503 - Vac	1504 - Vac
1505 - Vac	1506 - Vac	1701 - Vac	1702 - Vac	1703 - Vac
1704 - Vac	1705 - Vac	1706 - Vac	1802 - Vac	1803 - Vac
1804 - Vac	1805 - Vac	1806 - Vac	1807 - Vac	1808 - Vac
1809 - Vac	1810 - Vac	1811 - Vac	1812 - Vac	
2101 - Occ	2102 - Occ	2103 - Res	2104 - Vac	2105 - Vac
2106 - Vac	2107 - Occ	2108 - Occ	2109 - Vac	2110 - Vac
2111 - Vac	2112 - Vac			
3101 - Vac	3102 - Vac	3103 - Vac	3104 - Vac	3105 - Vac
3106 - Vac	3107 - Vac	3108 - Vac	3201 - Vac	3202 - Vac
3203 - Vac	3204 - Vac	3205 - Vac	3206 - Vac	3207 - Vac
3208 - Vac				



NSU Ethernet Status LEDS

The table below shows the meaning of the LAN LEDs

Table: NSU Ethernet Status LEDs

LED	Status	Color	Meaning
LAN	Flashing		LAN activity
MS (Message Status)	ON	Green	Message link open to the system.
OFF	—	Message link not open to the system. OR With L0/L1 LEDs marching: downloading	
ST (Status)	ON		Card is booting. *OR* Card is not operating
Flashing		Operational (flashes in 0.5 seconds intervals).	
OFF		Not operational.	

NSU L0/L1 Status LEDs

The table below shows the meaning of the LAN LEDs

Table: NSU L0/L1 Status LEDs

LEDs		Meaning
Left	**Right**	
—	ON (Red)	No Layer 1.
ON (Green)	—	

D-channel established (PRI). *OR*

Layer 1 established (T1).		
Flashing (Green)	ON (Yellow)	Alarm indication from far end.
OFF	ON (Yellow)	Blue alarm from card (this is normal during link startup; PRI NA or response to yellow)
Flashing (Green)	—	Layer 1 established (PRI).
Flashing (Green)	ON (Yellow)	Alarm indication from far end.
Flashing (alternating yellow and green)		Downloading (15-20 minutes).
Flashing (alternating yellow)		Decompressing and copying files (2-4 minutes).
—	OFF	No error
OFF	—	No Link.

NSU FIM LEDs

The table below shows the meaning of the NSU FIM LEDs.

When a remote MFC status LED is off, check the local FIM status LED. If it is ON, then the fiber optic cable may be faulty.

Table: NSU FIM Status LEDs

Top LED Status	Meaning for local, upper and remote, lower FIM
ON	In frame synchronization.
Flashing	Out of synchronization. OR Tx and Rx cables reversed.
OFF	Power off. OR Held in reset.

NSU CIM LEDs

The table below shows the meaning of the NSU CIM LEDs (CIM 1 is on the left, and CIM 2 is on the right).

Table: NSU CIM Status LEDs

LED Status	Meaning (all LEDs)
ON	Communication link synchronized with controller.
Flashing	BSP running.
Marching	When top and bottom LEDs are alternating, NSU is powering up. Do NOT interrupt.
OFF	No power.

Analog Services Unit (ASU)

The ASU contains the following status LEDs

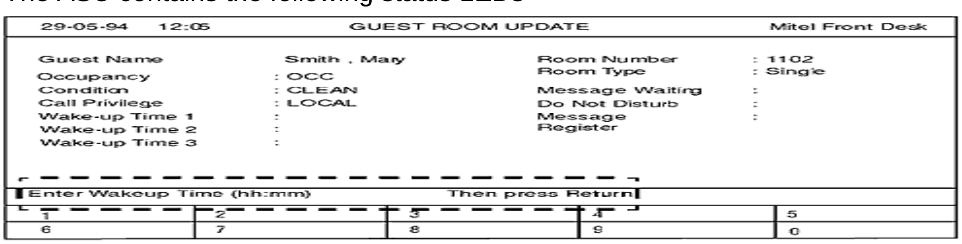


Table: ONS Circuit Status LEDs

LED Status	Meaning (all LEDs)
ON	Circuit is in use.
Flashing (slow)	Circuit is not available (undergoing maintenance).
Flashing (fast)	A fault has occurred with the circuit.
OFF	Circuit is idle.

Table: CIM Status LEDs

LED Status	Meaning (all LEDs)
ON	Communication link synchronized with controller.
Flashing	Link not functioning (due to cable not properly plugged in, incorrect cable, or broken cable).
OFF	No power.

Analog Services Unit II (ASU II)

The ASU II contains the following status LEDs:

```
 7:47 AM   16-JUL-04                                          alarm status = NO ALARM
              79  -                      Sub form           - 79
 [ ASU        TRUNK:   1  ]          | TRK NUM | BAY | SLT | CCT | COMMENTS
>                                    |    1    |  1  | 13  |  7  |            <
                                     |    2    |  1  | 13  |  8  |
                                     |    3    |  1  | 13  |  9  |
                                     |    4    |  1  | 13  | 10  |
                                     |    5    |  1  | 13  | 11  |
                                     |    6    |  1  | 13  | 12  |

                                     | _  1    |  1  | 13  |  7  |
 1-              | 2-             | 3-          | 4-          | 5-
 6-QUIT          | 7-DESC NUMBER  | 8-          | 9-          | 0-
```

Table: ASU II ONS and Combo Card Alarm LED:

LED Status	Meaning
Red ON	System error.
Red OFF	No error.

Table: ASU II ONS Card Activity LED	
LED Status	**Meaning**
Green ON	System error.
Green OFF	No error.

Table: ASU II Combo Card Activity LED	
LED Status	**Meaning**
Red ON	Out of service and power applied. as SFT call can be made.
Green OFF	No error.
Red OFF	No error.
Green ON	There is an established SFT or normal call.

Bay Control Card

The Bay Control Card II has status LEDs labelled Tx, Rx and ALARM. The Tx and Rx LEDs indicate communication (transmit and receive) with the SX-200 ICP controller. The ALARM LED has two functions. A flashing ALARM LED indicates a failure on the Bay Control Card. A permanently lit ALARM LED indicates that the Bay Control Card is waiting for, or has lost, communication with the SX-200 ICP controller. See the Bay Control Card II Status LEDs table.

The Bay Control Card III has three status LEDs in the middle of the faceplate; Ethernet, Tx/Rx, and Alarm. See the Bay Control Card III Status LEDs table. Also see the Bay Control Card III description.

Fiber Interface Module/Copper Interface Module

Two types of fiber interface modules exist, the FIM and the FIM II. The following paragraph and table applies to the FIM. For information on the maintenance controls and indicators for the FIM II and the CIM (Copper Interface Module) see the circuit card descriptions for the PRI card, and the Bay Control Card III. The FIM II and CIM can reside on all of those cards.

The Fiber Interface Module (FIM) has two status LEDs, labelled Local and Remote, located on the card's front panel as shown on this Figure, or on an onboard FIM. The two LED indicators show the status of the local and remote clocks. The SX-200 Bay Control Card FIM Status LED table below summarizes what it means when the LED is flashing, on, or off.

Table: SX-200 Bay Control Card FIM Status LED		
FIM LEDS	**LED STATUS**	**MEANING**
LOCAL	on	Local Receiver in sync (normal operation)
off	No power to FIM	
flashing	Local Receiver not in sync	

Table: SX-200 Bay Control Card FIM Status LED

FIM LEDS	LED STATUS	MEANING
	(not connected / wrong connection)	
REMOTE	on	Remote Receiver in sync (normal operation)
off	No power to FIM	
flashing	Remote Receiver not in sync (not connected / wrong connection)	

Note: FIM status LEDs are located on the Fiber Interface Modules. The status indicated applies to both FIMs.

Figure: Fiber Interface Module

Music-on-Hold/Pager Unit

The Music-on-Hold/Pager Unit has a single status LED which indicates the following states:

Table: Music-On-Hold/Pager Unit LED Indicator

Indication	MEANING
OFF	No power from DNIC circuit
Flashing (2 Hz - On 250ms, Off 250ms.)	Loss of synchronization. The Music-on-Hold/Pager Unit may be faulty. Refer to Troubleshooting section.
On Solid	Music-On-Hold/Pager Unit is operating normally
Winking (2 Hz - On 50 ms, Off 1 sec.)	Music-on-Hold/Pager Unit is operating and the paging amplifier is being accessed

Peripheral Circuit Cards

Digital Cards. On a digital card, each circuit has one associated LED which indicates the busy (on) or idle (off) status of the circuit. When a digital circuit is removed from service by the maintenance system, the associated LED will flash with a period of 0.5 second on and 0.5 second off. Each digital peripheral card has an alarm LED located at the bottom of the card's front panel. If any circuit on the card has a known fault or if a card is installed in an unprogrammed or incorrect card slot, the alarm LED will flash with a period of 0.5 second on and 0.5 second off (See Figure A and Figure B below.

Figure A: Typical Peripheral Card Front Panels

Figure B: Typical Peripheral Card Front Panels

Attendant Console

The attendant console function LEDs shown in this Figure are used as maintenance indicators during the console initialization sequence. The following table Attendant Console Maintenance Led Indicators and the Attendant Console LCD Display Messages table describe the console maintenance indicators.

Table: Attendant Console Maintenance Led Indicators

Test Name	Test Fail Indication
Console RAM Test	HOLD 1 LED flashes.
Console EPROM Test	HOLD 2 LED flashes.
Console IRQ Test	HOLD 3 LED flashes.
Console LED Test	Any of the console LEDs fail to go on, and stay on for 2 seconds.

Figure: Attendant Console Keyboard

Table: Attendant Console LCD Display Messages

Message	Meaning
CONSOLE HARDWARE FAILURE 123456789 ERROR CODE 1 PLEASE NOTE DETAILS ON REPAIR TAG	Console power-up tests failed.
WAITING FOR SYNCHRONIZATION 123456789 PLEASE WAIT	Power is available, but there is no synchronization to the line.
WAITING FOR COMMUNICATION 123456789 PLEASE WAIT	Console is synchronized, but no messages are being received from the system.

Note: In all cases, refer to the Troubleshooting section

Power Supply

Bay Power Supply. Each digital bay requires one Bay Power Supply (see the Bay Power Supply Figure). There are two LED indicators located on the front panel of the Bay Power Supply; the top LED is the power ON indicator, and the bottom LED is the ring generator indicator. The ring generator indicator will flash on only when ringing signal is applied.

Figure: Bay Power Supply

System Initialization

The system software and a default database are installed in the controller at the factory. Initialization re-installs the software and database from the external flash. You may need to initialize the system to recover from a failed attempt to upgrade or to restore a unstable system to a know state. For initialization details, refer to the following chart, Initial Power-up Procedure.

IMPORTANT: Re-initializing a working system forces the IP Phones to reboot. When the system is running again, the phones will return to service approximately 10 to 15 minutes later.

Chart: Initial Power-up Procedures

Step	Action
1.	Verify that the options identified on the Feature Options Record have been purchased.
2.	Insert a CompactFlash card containing the system software and a database into the slot on the front of the controller.
3.	Power the controller off and on, and wait for the power-up sequence to complete.
4.	Establish a serial connection to the controller.
5.	Press the RETURN key four times; the terminal display returns: 1 - VT100 COMPATIBLE 2 - TTY TYPE 3 - IBM PC SELECT A TERMINAL TYPE :
6.	Select the terminal type by entering “1”. The display returns: 1 - MAINTENANCE 2 - CDE 6 - QUIT SELECT AN APPLICATION (OR QUIT TO START OVER) :
7.	Select the Customer Data Entry application by entering “2”. The display returns: ENTER USERNAME :
8.	Enter the INSTALLER level of access. The display returns: ENTER PASSWORD:
9.	Enter the required password to access Customer Data Entry. The default password is 1000. A list of the system’s forms is displayed.
10.	

Select Form 04 and enable all the software options that were purchased. (Requires the Mitel Feature Options Record (FOR) sheet.)

Press the ENTER MOC softkey and enter the Mitel Options Code from the FOR sheet to enable the options and timers for the system.

Confirm that the System Identity Code (Option 101) matches the System Identity Code from the FOR sheet.

Enter the Mitel Options Password (Option 100) from the FOR sheet to activate the enabled features.

Press the ENTER softkey.

Enable only the purchased options listed on the FOR sheet for this system.

Users must purchase the option and its associated password through their usual ordering channels prior to attempting to install or implement the option.

A message appears that states a reset is required. Select Confirm to reset the system.

Note: If a database from another system is installed in the controller, the System ID and Password will not match. Phone service will be lost (some phones may appear to be in service, but will display SYSTEM BUSY when they go offhook) and a MOSS alarm message will display in the CDE forms header. Enabling the options using the above procedure clears the alarm and restores phone service.

11.

To prevent unnecessary alarms:

- In Form 53, check the bay number assignments. If a bay is assigned a bay number, it must be connected to a device (ASU, NSU, T1 trunk, etc.); otherwise the system will generate an alarm. To prevent this problem, move disconnected bays from physical connections (CIMs and MMCs) to phantom bays.
- In Maintenance, program the Alarm Thresholds for lines, trunks, and DTMF receivers to suit system requirements. Increase threshold values to prevent unnecessary alarms.

12.	Remove the CompactFlash card from the controller.

Glossary

3G

3G is an ITU specification for the third generation (analog cellular was the first generation, digital PCS the second) of mobile communications technology. 3G promises increased bandwidth, up to 384 Kbps when a device is stationary or moving at pedestrian speed, 128 Kbps in a car, and 2 Mbps in fixed applications. 3G will work over wireless air interfaces such as GSM, TDMA, and CDMA.

10 Base-T

10 Base-T or 10 BaseT. Unshielded twisted-pair, minimum Category 3, Ethernet cabling standard. The transmission rate is 10Mbps with a cabling limit of 100m from the Ethernet switch or hub. It uses two pairs of a 4 pair cable with RJ-45 connectors.

2500

A single-line, standard DTMF telephone. This terminology is used in the United States and Canada.

4ESS

A digital central office switching system that is typically used for "tandem switching office" for long distance calls. It is made by Lucent.

5ESS

A switching system made by Lucent. It is typically used as an "end office".

802.11a

A faster protocol than 802.11b, operating at 54Mb/s. 11a devices run in the 5GHz spectrum range. Until recently in the UK this band was not available for free use. The Radio Authority has just made it a licence-exempt band for limited use.

802.11b

The first published standard for wireless computer networks. 802.11b devices run at 11Mb/s (11 mega-bits per second). Devices from different manufacturers using this protocol will be compatible. 802.11b operates in the 2.4GHz spectrum range and works in 11 different channels in the UK, 13 or 3 elsewhere.

802.11b+

Referred to as Enhanced 802.11b this is a non-standard protocol being offered by some manufacturers that works at a nominal speed of 22Mb/s.

802.1D

An IEEE standard for bridging LANs (specifically 802.3, 802.4, and 802.5 networks). 802.1D operates at the MAC layer.

802.11g

A faster protocol than 802.11b, also operating at 54Mb/s. 11g devices run in the 2.4GHz spectrum range. 11g is likely to take over from 11b as the standard for the foreseeable future.

802.1p

This is the prioritization standard on your switch. This standard allows prioritized data such as voice packets to pass through the switch first. In the case of VoIP equipment, it will mark the data as high-priority. 802.1p supports eight classes of prioritization, each of which has to be mapped onto a hardware queue in the switch. If you are implementing VoIP then it's best to make sure that your switches support all eight classes.

802.1p/Q

The 802.1p/Q standard was introduced to standardize the various methods of priority and VLAN tagging that were introduced by different manufacturers. An extra field for VLAN identification, which includes 3 bits for priority, is inserted into the Ethernet frame.

802.1Q

This is the standard for virtual LANs. It inserts an extra 4-bytes in the header for designating the VLAN. The extended header also includes the 802.1p priority scheme. See VLAN.

802.3

The official IEEE Ethernet Standard. Ethernet is the leading LAN architecture that originally used BUS topology and CSMA/CD at 10 Mbps. Although 802.3 is not technically the same as Ethernet, the name is commonly used.

802.3ac

A new extended frame length for Ethernet, which includes the 802.1p/Q priority and VLAN tag.

802.3af

802.3af, or Power over Ethernet, defines a way to deliver 48 volts of AC power over unshielded twisted-pair Ethernet wiring. The internal Layer 2 switch on the CXi controller supports 802.3af.

802.5

An IEEE physical layer standard for a token-ring LAN access method at 4 or 16 Mbps over unshielded twisted-pair cable.

A-wire

See Tip (lead).

ABH

See Average Busy Hour.

ABSBH

See Average Busy Season Busy Hour.

ACD

See Automatic Call Distribution.

ACD Call

An ACD call is an incoming call destined to an ACD path of agent groups.

ACD Caller

An ACD caller is a caller that has successfully entered an ACD path and remains in the path until the call interflows out of the path, and is transferred to an non-path destination, or to an extension that has no logged in agent.

ACD Path

An ACD path is a directory number based service that guides an incoming call through a list of ACD agent groups.

ACO

Analog Central Office

A/D

See Analog/Digital.

ADL

See Associated Data Line.

ADPCM

See Adoptive Differential Pulse Code Modulation.

ADSL

See Asymmetric Digital Subscriber Line.

ALI

See Automatic Location Identification.

AMB

Analog Main Board (MX Controller with embedded analog)

ANI

See Automatic Number Identification.

ANSI

See American National Standards Institute.

API

See Application Program Interface.

APNSS

Analogue Private Network Signalling System

Application Layer

See OSI Model Layer 7

ARP

See Address Resolution Protocol.

ARS

See Automatic Route Selection.

ASCII

American Standard Code for Information Interchange. It was developed by the American Standards Association for both synchronous and asynchronous data transmission between DTEs. Characters consist of an 8-bit binary code and incorporated parity bits.

ASU

Analog Services Unit: A component of Mitel 3300 ICP that provides analog connectivity to the system.

ATO

Analog Toll Office trunk

ATT

Analog Tie Trunk

AU

Australia

Abbreviated Dialing

Abbreviated dial gives users the ability to dial abbreviated speed call codes, which substitute for a system wide list of frequently-called numbers. These numbers can be displayed or programmed at the attendant console or the maintenance terminal.

Absorbed Digits

In certain call processing functions performed by the SX-200 EL/ML system, it may be necessary to suppress the onward transmission of certain digits received in a dialed sequence of digits. This digit absorption is required for applications such as DID calls and ARS purposes. See also digit modification.

Access Point

An Access Point provides an interface between the wireless network and a wired network. It provides a bridge between Ethernet wired LANs and the wireless network. Access Points are the connectivity point between Ethernet wired networks and devices equipped with a wireless LAN adapter card.

Active Agent

Agent that can take ACD calls. An agent is considered "active" when: logged in, not in make busy, not in DND, work time is inactive, idle.

Active Loop

A fault condition on an IEEE 802.3 LAN in which there is more than one active route between source and destination nodes; active loops can occur when more than one bridge is attached to the same LAN.

Address Ageing

The process that automatically removes infrequently used dynamic MAC addresses after a configurable length of time. Separate ageing times can be configured for addresses associated with extensions on the central LAN and remote LANs.

Address Filtering

A method of deciding which data packets are allowed through a device; the decision is based on the source and destination addresses on the data packet.

Address Resolution Protocol

Protocol used for mapping an Internet Protocol address (IP address) to a physical machine address that is recognized in the local network. The physical machine address is also known as a Media Access Control (MAC) address.

Adoptive Differential Pulse Code Modulation

A voice encoding compression scheme providing digitally encoded voice at 32Kbps.

Agent

Agents answer incoming ACD calls. Agents are specially trained to deal with the caller’s requests.

Agent Groups

A call center term, also known as Split, Gate, Queue or Skills Group. An agent group is a collection of agents who share a common set of skills, such as being able to handle customer complaints.

Alarms

There are three categories of urgency of fault condition on the SX-200 EL/ML system: minor, major, and critical.

Minor alarms indicate problems affecting a portion of the system, such as failure of a line or trunk circuit. Major alarms indicate problems causing a system-wide degradation of service. Critical alarms indicate serious problems that cause automatic activation of system fail transfer.

American National Standards Institute

This organization develops and publishes voluntary standards for a wide range of industries for companies based in the US.

American Wire Gauge

Standard measuring gauge for conductors such as copper & aluminum. The gauge measures the thickness of the cable.

Amphenol Connector

Amphenol is a manufacturer of connectors. An Amphenol connector in this documentation refers to the 25-pair connector.

Analog

Analog technology refers to an electronic transmission whereby the voice wave-form of a person having a conversation on a telephone is converted by the telephone of the calling party into an analogue electrical signal with almost exactly the same wave-form. Until recently, telecommunications systems were invariably analog.

Analog/Digital (A/D)

Implies the transformation of analog signals (such as normal telephone speech signals) into their equivalent digital data signals. The device in general use that performs this transformation is called an A/D converter. The device that converts digital signals into their analog form (if required), is called a D/A Converter.

Analog to Digital Conversion (ADC)

The conversion of an analog signal into a digital signal.

Analog Loopback

A method of testing modems and data terminals in which the device is disconnected from the telephone line, and a signal is looped out through the transmit side of the device and in through the receive side. This test indicates if the problem is with the telephone line or with the modem.

Analogue Multiplexing

In analogue transmission systems, multiplexing can be achieved by a process known as frequency division multiplexing whereby information is transmitted simultaneously within different, and unique, frequency ranges over the same transmission medium.

Analog Transmission

The transmission of a continuously varying signal. For example, in the transmission of speech the magnitude of the signal at any instant in the transmission path is proportional to the magnitude of the original input. This type of transmission is distinct from digital transmission in which the original input is encoded (for example, CODEC) and the resulting line signal is in digital form.

Anonymous FTP

By using the word anonymous as your user ID and your email address as the password when you log in to an FTP site, you gain limited access to public files on the remote computer. This type of access is available on many FTP sites but not all.

Answering Point

A device to which an incoming call is directed. It usually consists of an industry-standard telephone or an attendant console. Under certain conditions an answering point may be a hunt group, a trunk, an ACD path or a device such as a night bell, an answering machine or a recorder/announcer machine.

Application Processor

A processor containing one or more application programs which meet a customer’s particular needs; for example, a hospital, a governmental agency or a university environment. The processor is usually arranged to be accessed directly by an input/output device. However, it can be connected to the SX-200 EL/ML system by means of suitable interface arrangements, and therefore be capable of access by suitable input/output devices which are also connected to the SX-200 EL/ML system.

Application Program

See Application Processor.

Application Program Interface

A collection of protocols, routines or tools used by programmers to develop software application programs.

Associated Data Line

A DTE connection to an SX-200 EL/ML system by means of a dataset which has an associated telephone set. The user sets up a data call by dialing an access code and destination dataset number.

Asymmetric

When the downstream/upstream flow of data-speed is unequal.

Asymmetric Digital Subscriber Line

ADSL is a switching technology used to increase transmission speed over copper cable. ADSL allows higher speed transmission downstream (1.5 Mbps - 9 Mbps), but has significantly lower speed (16 Kbps - 800 Kbps) upstream. ADSL is permanently on. ADSL allows the multiplexing of voice data over the same cable. ADSL allows a service provider to transmit data services including internet services by squeezing more data capacity through the copper wire used for telephony.

Asynchronous Mode

In asynchronous data transmission the time between bytes (characters) is indeterminate and depends upon external factors. The transmitted data has its own start and stop elements, and thus controls the receiving device. See also Synchronous Mode.

Asynchronous Transfer Mode (ATM)

ATM is a LAN protocol that uses cell-based packet-switching protocol for enterprise LANs. It is dedicated-connection switching technology that can support multimedia traffic (voice, video, text, data) at multi-gigabit speeds in the same transmission. ATM runs on fibre-optic or twisted pair cable. It offers universal, service-independent switching and multiplexing capabilities. ATM transports data through fixed length packets called cells. Because ATM is designed to be easily implemented by hardware (rather than software), faster processing and switching speeds are possible. These cells are easier to store and quicker to pass through the network than the variable length packets used by other LAN formats. With the capability of providing a virtual connection with bandwidth allocation as needed, ATM can deliver multi-gigabit speeds. ATM has the ability to support integrated audio, video-conferencing, three dimensional imaging, and high-definition movies in the same transmission and switching fabric that performs the routing function. Frequently called cell relay.

ATM has not been widely successful in the enterprise due to the high cost of installing and maintaining it. Gigabit Ethernet has overtaken the market, Gigabit Ethernet's 1 Gbps bandwidth is sufficient to meet the needs of most enterprise network application requirements today, development is underway for multi-Gigabit link aggregation and10 Gbps Ethernet for greater bandwidth resiliency and scalability in the future.

Gigabit Ethernet is inexpensive compared to ATM, with ATM currently costing 2 to 3 times as much per port. ATM not only costs more for the initial hardware and software acquisition but also results in much higher costs for testing, deployment, operational support, problem resolution, and skills training. There are fewer ATM skilled personnel available, meaning increased HR costs., As Ethernet is widely deployed it means the network administration and management skills can be leveraged for the deployment and management of Gigabit Ethernet without additional costs.

Attendant

An attendant is also known as an operator. The person who answers a call using a telephone system console.

Attendant Busy Override

Attendant Busy Override is also known as Operator Intrusion. This is a feature where the attendant can, if required forcibly enter an existing telephone conversation.

Attendant Status Busy-out

A feature that allows an attendant to busy-out a specific station by using the attendant console. An alternative name for this features is Extension Busy-out by operator.

Authentication

Any process that ensures that users are who they say they are. When you type your name and password, you are authenticated and allowed access.

Authorized Access Codes

The SX-200 EL/ML system can only be accessed for programming, maintenance or administration purposes by first entering an authorized access code (user name and password).

Auto Attendant

An Auto Attendant is a telephony device that uses automated prompts to ask questions of callers and then asks the caller to respond by pressing a number. The system prompts callers to respond to choices (e.g., press one for this, two for that"). This enables the system to direct the call to the appropriate department or service area.

Auto Wrap-up

Auto Wrap-up is a feature of the ACD whereby the ACD will automatically put agents into After-Call Work once they have completed a call. When they have completed the required After-Call Work, they return their status to Available.

Autobaud Detection

Some data communication equipment can determine, on receipt of one or more characters, the baud rate of the transmitting source. It then sets its own receive circuits to accommodate this baud rate. In the SX-200 EL/ML system this feature is applicable to datasets and to the maintenance/CDE port which automatically adjusts its baud rate to match that of the terminal during the initial setting up procedure.

Automated Software Distribution

An OPS Manager Feature that automatically distributes upgrades from the OPS Manager Station to the network of PBXs.

Automatic Call Distribution

Automated Call Distribution (ACD) is a function of a telephone system that manages incoming calls and handles them based on the number called and provides associated handling instructions. An ACD is used to validate callers, make outgoing responses or calls, forward calls to the right party, allow callers to record messages, gather usage statistics and balance the use of phone lines

Automatic Callback

Also known as Callback.

Automatic Location Identification (ALI)

The use of a database to associate a telephone number with a physical location.

Automatic Number Identification (ANI)

ANI is a method that is used by telephone companies to identify the billing account for a toll call but is not the same as Call Line ID.

Automatic Route Selection

Automatic route selection software automatically selects the optimum trunk route when a user makes a call. This selection is based on many factors, including cost, user priority, the day, and time of day.

Average Busy Hour (ABH)

The clock hour that has the highest average business day traffic (see Busy Hour).

Average Busy Season Busy Hour (ABSBH)

This is the hour calculated to have the highest average business day traffic load during the three highest traffic months of the year.

Average Hold Time

The average time a caller must wait before their call is answered by an agent.

AWG

American Wire Gauge

B Channel

The B channel is the 64-K bit channel of a DNIC device. It can carry digitized voice or ASCII characters at a maximum rate of 19.2 Kb/s.

BAP

Bandwidth Allocation Protocol. Layer of the Point-to-Point Protocol (PPP) which simplifies the management of multiple links in data networks.

BGP

Border Gateway Protocol. A routing protocol that interconnects organizational networks and evaluates each of the possible route to find the best one.

Back-up Battery

Also known as Standby Battery.

Backbone

The backbone is the main cabling of a network that all of the segments of that network connect to. Typically, the backbone is capable of carrying more information than the individual segments. For example, each segment may have a transfer rate of 10 Mbps (megabits per second), while the backbone may operate at 100 Mbps.

A Backbone is also the network that interconnects other networks, employing high-speed transmission paths and often spanning a large geographic area.

Backbone Network

A network that interconnects other networks.

Backplane

A high-speed communications line to which individual components are connected.

Bandwidth

Bandwidth determines the rate at which information can be transmitted over a computer channel, communications line, or bus. The greater the bandwidth, the greater number of megabits can be transmitted per second.

Bandwidth on demand

A cost saving feature that allows a device to establish a communications link to the central LAN only when bandwidth is required to transfer data. Bandwidth on demand is suitable for high speed connection of voice, data, and video services.

Basic Rate Interface

ISDN standards and specifications for provision of low-speed ISDN services. Supports two B-channels of 64Kbps each and one D-channel of 16Kbps on a single wire pair.

Battery and Ground Pulsing

A method of signaling used on long lines, in which both wires use battery and ground at each end of the circuit. When signaling to the remote end of the trunk, the battery and ground connections are reversed, opposing the potentials at the remote end and increasing the current supply to the trunk.

Baud Rate

A measure of transmission speed over an analog phone line. Often, incorrectly used as interchangeably with bit rate.

Blocking

The condition existing in a switching system when the immediate establishment of a call is impossible due to insufficient switching connections being available in the system at that time.

BRI Card

The Basic Rate Interface (BRI) card allows the SX-200 to communicate to Central Offices and devices that support BRI. The SX-200 supports the termination and the origination of ISDN voice and data calls for trunk side and line side U interfaces. The trunk side provides Basic Call and Incoming Calling Name.The line side provides data and voice calls from BRI devices, and voice calls from sets. The BRI card provides up to 12 ISDN BRI U interfaces.

Busy Hour

The hour when a system carries the most traffic (the busiest hour of the busiest day of a normal week).

CCS

A unit used in traffic analysis to denote the traffic occupancy of a switched circuit in a PBX exchange. One CCS represents 100 call-seconds.

CDE

See Customer Data Entry (CDE).

CIM

The CIM (Copper Interface Module) provides a copper communications link between the control cabinet and a peripheral cabinet or a PRI card bay.

COV

See Control Over Voice.

CP

See Call Processing.

CLASS

See Custom Local Area Signaling Service

CRC

CRC or Cyclical Redundancy Check is an error-checking measure used to ensure the accuracy of transmitting data. The transmitted message is a small integer value computed from a sequence of octets. This value is used to detect errors that result when the sequence of octets is transmitted from one machine to another.

CSMA/CD

CSMA/CD or Carrier Sense Multiple Access with Collision Detection is a method of getting onto and off of a LAN. Multiple workstations access a common transmission medium by listening until no signals are detected, then transmitting and checking to see if more than one signal is present. IEEE 802.3 and Ethernet LANs are based on the CSMA/CD standard transmission method.

Call Processing

The software package which handles all aspects of the setting up of connections within the system.

Circuit Switch (CS)

The SX-200 EL/ML system circuit switch provides a matrix of bidirectional switch links. Each circuit switch link accommodates 32 channels. Each channel can be used for a voice or data transmission. Through the circuit switch, any device can be connected to any other device in the system. It is located on the DX Module on the main control card.

The number of links in the matrix depends upon the system configuration.

Circular Hunting

Circular hunting starts at the extension after the last extension in the hunt group to which a call was completed (the extension rung), and hunts overall extensions in the hunt group in the sequence programmed. Hunting stops at the first idle extension found.

Class of Restriction (COR)

A class of restriction controls station and trunk access to trunk circuits. It performs functions similar to toll control and is programmable on a station (or trunk) basis.

Class of Service (COS)

A class of service has a number of different feature options assigned to it. A class of service can be allotted to one or many stations, and enables these stations to have, or be denied, features which are available within the SX-200 EL/ML system. Up to 50 classes of service are available which allow a large number of different groups of station users to be programmed, each with differing feature characteristics.

Client-Server

A computer on a LAN that provides information or applications. The client-server splits the workload between desktop PCs (workstations) and one or larger computers (servers) connected on a LAN.

CODEC

The COder-DECoder is a device used in digital switching and transmission systems, for coding analog signals (voice signals) into a digital format for onward transmission, and decoding a digital transmission to recover the original analog signal.

CODEC/Filter

The CODEC/Filter chip used in the SX-200 EL/ML system consists of a CODEC, a filter and other elements. It forms part of the peripheral card, with the CODEC portion performing the necessary A/D and D/A functions and the filter portion providing low–pass filtering for the line transmission.

Consultation Hold (Soft Hold)

A form of temporary hold. It is used to put a second party on hold while the first party is speaking (consulting) with a third party, or wants to temporarily isolate the second party from conversation.

Control Dual FIM Carrier Card

The card supports up to two Fiber Interfaces Module and connects them to the backplane. It has a 1 km FIM onboard and a connector to plug in a second optional FIM (either a 1 km FIM or an extended FIM).

Control Triple FIM Carrier Card

The card supports up to three Fiber Interface Modules and connects them to the backplane. It has two 1 km FIMs onboard and a connector to plug in a third optional FIM (either a 1 km FIM or an extended FIM).

Control Over Voice

Control over voice is used by the voice mail system to perform most of its signaling functions. A 32 kHz carrier signal is modulated according to the control function, and is transmitted to or from the voice mail system on the same pair of wires used for the audio connection. The carrier frequency lies above the normal audio range of the voice mail system and is therefore inaudible to the user.

Critical Alarm

For each alarm category, the thresholds represent the alarm level trip points; that is, the precise divisions between the alarm levels. The thresholds are simple percentages, indicating availability: the number of working devices is compared to the number of programmed devices. The Critical Alarm threshold is not a percentage, but is a precise numerical value. When the number of available devices falls below this number, a critical alarm is raised.

Cross-Connect Field

See Main Distribution Frame.

Custom Local Area Signaling Service

An SX-200 EL/ML system permits advanced voice features such as Calling Line ID digits and CLASS Name to accompany or precede the telephone call across multiple switches.

Customer Data Entry (CDE)

Customer data entry (CDE) is the process employed when data particular to a specific customer installation is entered into the SX-200 EL/ML system. This data includes such things as numbering plan, ARS routings, and trunk descriptors. CDE is entered into the SX-200 EL/ML system by the maintenance/CDE terminal.

D Channel

The D Channel is the 16 K bit/s control channel of a DNIC device.

DCE

See Data Communication Equipment.

DHCP

DHCP or Dynamic Host Configuration Protocol is a mechanism that allows systems to obtain configuration information from a central server.

DID

Direct inward dialing.

DIL

Direct-in line.

DNS

DNS or Domain Name System (or Server) is used to map names to IP addresses and vice versa.

DOD

Direct outward dialing.

DSP

Digital Signal Processor.

DSR

DSR or Data Set Ready is a hardware signal defined by the RS-232C standard to indicate that the device is ready to operate.

DTE

See Data Terminal Equipment.

DTMF

Dual tone multifrequency signaling.

DTR

Data terminal ready RS-232 pin.

DTRX

Data transceiver.

DX

An abbreviation of the term digital crosspoint, the fundamental switching element of the SX-200 EL/ML circuit switch. The circuit switch is composed of a large number of digital crosspoint switch elements in the form of DX chips.

Data Communication Equipment

Data communication equipment (DCE) interfaces a communications line or data device to data terminal equipment (DTE) over an RS-232 line. A modem and the local maintenance port on the SX-200 EL/ML system are examples of a DCE.

Data Encryption

A security procedure in which messages are enciphered in secret code so that only the intended receiver can read them.

Data Terminal Equipment

Data terminal equipment (DTE) is terminal equipment, usually consisting of a keyboard and video screen or printer, which is used to communicate with a variety of other equipment (i.e. another DTE or a computer).

Default

The value assigned to a particular function which most nearly represents the normal or standard value of the function. A typical default value used in the SX-200 EL/ML system, for example, is a value of 2 minutes allowed before an unanswered ringing extension times out. However this value can be changed in CDE programming from the default value to a value which lies between 1 and 30 minutes.

Digit Modification

The process of restructuring a dialed sequence of digits received by the SX-200 EL/ML system to effectively result in a different sequence of digits. The revised sequence can have new digits added and/or digits deleted (absorbed); or certain digits in the original sequence can be repeated. This process is performed automatically by the SX-200 EL/ML system and is transparent to the user. Digit modification is used in speed calling, tandeming of trunk circuits, processing incoming DID calls, processing calls in ARS, and other applications.

Digital/Analog

A term used in connection with the conversion of digital signals to equivalent analog signals. The original signals are usually in analog form and are converted from analog to digital signals for transmission (see also Analog/Digital).

DTRX Messages

DTRX message code used to standardize messages between Mitel PBXs as well as different language text. Intelligent devices can interpret these codes.

Dynamic Address

An Ethernet address that can be automatically deleted by the bridge if the bridge has not seen a packet from that address within a configurable time period (aging).

E and M

A type of tie trunk. Also the signaling method used for this and for other types of trunks. The term is derived from the use of the E and M leads forming part of the trunk equipment, and taken respectively to denote the receive and transmit leads. The two leads are used to pass supervisory conditions over the trunk.

End Node

A network node that supplies information to the network but does not receive information from the network.

Ethernet

A local network design, characterized by 10 Mbps baseband transmission over a variety of cable media, and employing CSMA/CD as an access control mechanism.

FCS

FCS or Frame Check Sequence refers to the special bits appended to the end of a data frame for error checking. An FCS calculation involves a polynomial equation being performed on the data and address fields. FCS is typically accurate to one packet in 4 billion.

Fiber Interface Module (FIM)

The Fiber Interface Module supports the transmission of voice and data signals over fiber optic cables. The FIM plugs into the Control FIM Carrier.

Filtering

The process used to ensure that only selected frames are forwarded.

Forced Account Code

When the forced account code feature is entered in a station's class of service, the user at that station must dial a valid account code each time an outgoing trunk call is made. If it is not entered in a user's COS, the user is denied access to the trunk. The account code appears as part of the SMDR record.

Frame

A fixed block of data bits with a flag at each end to indicate the beginning and end of the frame. The defined format enables network equipment to recognize the meaning and purpose of specific bits. Also called a packet, in LAN terminology.

FTP

FTP or File Transfer Protocol is a TCP/IP protocol in LAN technology that is used to log on to the network, list directories, and copy files. FTP can also perform ASCII to EBCIDIC conversions.

Full Duplex

A method of operation which allows simultaneous transmission from both ends of a communications link.

Gateway

A port that connects multiple networks or subnetworks. Incoming and outgoing signals from and to dissimilar networks are converted from one form to another in terms of speed, protocols, and interchange codes.

Ground Button

See Recall Button.

Ground Start

A particular type of trunk circuit on which a ground condition is applied to the ring lead of the trunk when an outgoing call seizes the trunk.

HDLC

HDLC or High Level Data link Control is an ISO communications protocol used in X.25 packet-switching networks. HDLC provides error correction at the DCE.

Half Duplex

The transmission of data in both directions, but only in one direction at a time.

Hard Hold

A station user, or an attendant, puts another party on hold, and can perform any of the functions which are normally available at the station (as opposed to consultation hold, which restricts the functions which can be performed).

Host

Any computer connected to the network that serves as a source or is the recipient of information.

Hot Repair

A system can allow maintenance or repair action to be performed without power being removed from the system first.

Hub

A LAN device that receives data from one link and rebroadcasts it to other links.

Hunt Group

A hunt group is a group of stations to which incoming calls are directed by dialing a master number. Two types of hunting are provided by the system, circular and terminal:

- Circular hunting starts at the extension after the last extension in the hunt group to which a call was completed (the extension rung), and hunts overall extensions in the hunt group in the sequence programmed. Hunting stops at the first idle extension found.
- Terminal hunting starts at the first extension in the hunt group and terminates at the first idle extension found. Hunting takes place in the order in which the extensions were programmed into the hunt group.

IMAP

IMAP or Internet Message Access Protocol is a mail protocol that provides management of received messages on a remote server. The user can review headers, create or delete folders/mailboxes and messages, and search contents remotely without downloading.

ICMP

ICMP or Internet Control Message Protocol is a protocol that generates error messages, test packets, and informational messages related to IP.

Interconnection Restrictions

Certain interconnections between stations and trunks, and between trunk and trunk circuits, are not allowed for various reasons. These interconnections are prevented by setting appropriate parameters in the device interconnection table which is programmed as part of the SX-200 EL/ML CDE procedures. Calls made with trunk circuits are subject to the parameters in the table.

Interconnectivity Gateway

A gateway that ensures connectivity across attached networks of different natures without ensuring compatibility between applications.

IP

IP or Internetworking Protocol is the network layer protocol from the TCP/IP suite of protocols. A standard describing software that keeps track of the internetwork address for different nodes, routes outgoing messages, and recognizes incoming messages.

IPSec

IPSec or Internet Protocol SECurity is a set of protocols for encryption of IP traffic over the Internet through virtual private networks (VPNs).

IP Address

A unique 32-bit address that identifies LAN devices on a TCP/IP network or the Internet. An IP address consists of four groups of digits; each group has a maximum value of 255 (for example, 204.113.49.5) and contains a network (subnet) portion and a host portion. This partition makes routing efficient.

IP Routing

The action of directing network traffic according to the rules of the TCP/IP protocol.

IP Subnetworking And Masking

A mask is a value associated with each IP address. A mask is configured by the network manager and allows the nodes and routers to read the portion of the IP address that represents the network (subnetwork) number.

ISO

ISO or International Standards Organization is a voluntary, non-treaty organization founded in 1946. The ISO is responsible for creating international standards in areas including computers and communications.

ISP

ISP or Internet Service Provider is an organization that provides users with an Internet connection.

ISDN

The Integrated Services Digital Network (ISDN) accurately transmits voice, data, and video at high speeds without a modem. The SX-200 system with LIGHTWARE 17 Release 4.0 or greater software supports voice with 3.1 kHz audio.

kbps

kbps or kilobits per second is a data transmission rate measured in thousands of bits per second.

LAN

LAN or Local Area Network is a short-distance (usually within a single building complex) data communications network used to link together computers and peripheral devices.

LCP

LCP or Link Control Protocol is a protocol that provides a method of establishing, configuring, maintaining, and terminating a point-to-point connection.

Leased Line

A telecommunications channel leased between two or more service points within one exchange or between different exchanges, usually at a monthly rate. Also known as a dedicated circuit or a private line.

LED

LED or Light Emitting Diode is an indicator that emits light when an electrical current is passed through it.

LLC

LLC or Logical Link Control is the upper portion of the data link layer, as defined in IEEE 802.2. The LLC sublayer presents a uniform interface to the user of the data link service, usually the network layer.

Least Cost Routing

One of the functions of automatic route selection and refers to the economical aspects of the ARS facility. In least cost routing, the trunk circuits are programmed with regard to the effects of the costs of the possible alternative trunk routings. In practice the customer may require the economical aspects to be subordinate to the overall traffic efficiency requirements of the System. For example, less costly trunk routes may be available, but offer too low a traffic grade of service for the customer’s needs. Actual requirements may be subject to traffic analysis of the customer’s needs.

Link

A link is the system connection that contains 32 channels. Each channel is a digital "speech path". In the SX-200 EL/ML system, there are three links per fibre connection or Bay (Node). Therefore each peripheral node has the equivalent of 3 x 32 or 96 "speech paths" available.

Load Balancing

A technique used by Ethernet bridges to evenly divide traffic between two channels, thus maximizing throughput.

Loop Start

A form of signaling used by a certain type of CO trunk which designates that type of trunk. It denotes an outgoing trunk circuit which is seized by the system placing a loop condition on the trunk.

Loop Tie Trunk

A tie trunk between PBXs which is seized by the application of a loop condition on the trunk. Subsequent supervisory conditions can be determined by the presence/absence of the loop or by battery-reversal conditions.

MAC Address

A factory-defined 48-bit number programmed into the device that is unique to each LAN device. Destination and source MAC names are contained in the LAN packet and are used by bridges to filter and forward packets.

Mbps

Mbps or Megabits per second is a data transmission rate measured in millions of bits per second.

MCC

Main Control Card.

MDF

See Main Distribution Frame.

MIB

MIB or Management Information Base is a structured database listing the names of all information resources residing in a network that can be used to manage the network.

MIS

MIS or Management Information System is a computer-based information system that integrates data from all the departments that it serves in order to provide operations and management with the information they need.

MODEM

A MOdulator-DEModulator is a piece of Data Communications Equipment (DCE) that accepts data signals from a piece of Data Terminal Equipment (DTE) and converts them into modulated tone signals suitable for transmission over telephone lines. The DCE at the far end converts the tone signals back into data signals and sends them to its DTE. The data circuit is commonly a duplex circuit; it is capable of operation in both directions simultaneously.

MP

MP or Multilink PPP is a protocol that bundles multiple links (logical or physical) into one logical link with greater bandwidth than any of the original links. Data fragmentation in the PPP Multilink algorithm ensures optimal utilization and load- sharing of the bandwidth of both links by splitting a data frame into multiple tiny subframes and spreading them out over the physical links.

MPU

Main processing unit. In the SX-200 EL/ML system, this refers to the 68000 CPU on the main control card.

Main Distribution Frame

The main distribution frame (MDF) forms the interconnection point between the in-house PBX (for example the SX-200 EL/ML system, and the internal and external cabling to the PBX. The MDF provides a convenient and flexible means of interfacing the cabling to the system.The MDF is also known as the cross-connect field.

Message Subsystem

The message subsystem is one of the subsystem blocks of the SX-200 EL/ML system. Its function is to act as the main message collection and distribution facility for the system, and links the main controller with the intelligent entities at the peripheral level or below. In effect it is the “nervous system” of the SX-200 EL/ML system, in that it passes messages and commands between the lowest and highest levels of the system.

Mixed Station Dialing

The SX-200 EL/ML system caters to the use of both rotary dial and DTMF types of industry-standard telephones installed on the system.

Multicast

A technique that transmits a message to multiple recipients at the same time. Multicasting is used in teleconferencing and data communications networks.

Multiple Consoles

More than one attendant console can be installed on an SX-200 EL/ML system. The trunk groups can be arranged to terminate such that they can be accessed from all of the consoles, and any call can be answered from any console.

NAT

NAT or Network Address Translation translates the IP address used within one network to a different IP address in another network. Typically, numerous IP addresses on the internal network are translated to a single IP address on the external network.

NIC

NIC or Network Interface Card is also known as a network adapter, a LAN Adapter, or a LAN card. Usually, a PC expansion board executes the code needed by the connected device to share a cable or some other media with other stations.

NMS

NMS or Network Management System is a combination of hardware and software that allows the network manager to configure the network, perform traffic analysis, detect and correct faults, and perform regular network maintenance. NMS systems typically manage large networks containing combinations of LAN servers, concentrators, bridges, routers, and repeaters.

node

A connection point for two or more communications links. The node serves as the control location for forwarding data among the elements of a network or multiple networks. In some cases, it performs local processing functions.

octet

An 8-bit byte.

ONS

See On-Premises Stations.

OPS

See Off-Premises Stations.

OSI

OSI or Open System Interconnection is a set of international standards governing interconnection of multiple vendors' architecture. OSI standards are enforced by the International Organization for Standardization (ISO) and the Telecommunication Standardization Sector (TSS).

OSI Reference Model

An architectural framework used to describe the structure and function of data communications protocols. The model consists of the following seven layers: the physical layer, the data link layer, the network layer, the transport layer, the session layer, the presentation layer, and the application layer. Each layer represents a function performed when data is transferred between applications on a network.

Off-Premises Stations

Stations which are located at a considerable distance from the parent communication system, and require special circuit terminating arrangements at the system.

On-Premises Stations

Stations which are installed on the same premises as the PBX, or which can operate satisfactorily with the system when installed in adjacent premises without special circuit arrangements

Overlap Outpulsing

A feature used in the SX-200 EL/ML system when making trunk calls. It results in dial pulses (or tones) being outpulsed prior to the receipt of all required digits from the user, the purpose being to reduce the time needed to process the call.

PAP

PAP or Password Authentication Protocol is an authentication phase that PPP calls must go through. In PAP, the ID and the Password pair is sent repeatedly by the user to the authenticator until authentication is acknowledged or the connection is terminated.

PCB

Printed Circuit Board.

PCM

Pulse Code Modulation.

PING

PING or Packet INternet Gopher is a useful internetworking debug and diagnostic program. When trying to determine the accessibility of a device (a router or even a PC), a PING is sent to the IP address of that device. The sender then waits for a response and reports the success or failure of the operation.

PKM

Programmable Key Module

PPP

PPP or Point-to-Point Protocol is a communications protocol that allows a computer to use TCP/IP with a standard telephone line and a high-speed modem. PPP is a new standard that replaces SLIP. It can also run on any full-duplex link from dial-up to high-speed DS1 and DS3 lines.

PPPoE

PPPoE or Point-to-Point Protocol over Ethernet is an access control method that allows remote hosts to log on and off using a simulated dial-up connection. PPPoE is typically offered by cable and DSL Internet service providers.

PPTP

PPTP or Point to Point Tunneling Protocol encapsulates data sent over the Internet within a virtual private network (VPN).

packet

A set of data with address and control signals organized in a specified format, transferred as a whole through the network.

packet filtering

The recognition and selective transmission or blocking of packets based on destination addresses or other packet contents.

Partitioning

The division of a LAN into two or more distinct segments joined by a bridge. Partitioning can reduce LAN media loading by localizing traffic and increase reliability by isolating LAN segments.

Peripheral Equipment

All external equipment connected to the SX-200 EL/ML system, such as stations, telephones, trunks, attendant consoles, and maintenance terminals.

Peripheral Interface Card

A card which provides the interface facilities between the external peripheral equipment, such as stations, trunks and attendant consoles. A prime function is to convert the external analog inputs to the internal digital PCM signals (and conversely convert digital PCM to analog output).

Power Fail Transfer (PFT)

See System Fail Transfer.

PRI Card

The PRI (Primary Rate Interface) card provides SX-200 systems with two links of Primary Rate Access (PRA) to the ISDN service provider.This PRI connection supports the simultaneous transmission of voice and data. Each channel is capable of transmitting voice or data calls at 64 kbps. By interfacing the SX-200 switch with the ISDN service provider, it allows SX-200 users to access ISDN services, such as Direct Dial In (DDI), Calling Line Identification (CLID), and Call By Call Service Selection (CBC).

PRI Gateway

The PRI Gateway is an interface that can include the ISDN Network Gateway, the PRI cards and the Ipera 2000 interfaces.

Protocol Filtering

A feature available in some network bridges in which the bridge can be programmed to automatically forward or reject transmissions associated with specified protocols.

Protocol Prioritization

A process in which the bridge orders a WAN output queue in order of priorities, based on protocol types.

Proxy ARP

Proxy ARP or Proxy Address Resolution Protocol is a simulation of the ARP protocol used by hosts that do not support IP subnet routing, but have an interface on a subnetted network.

RS-232C

A North American data interchange standard, issued by the Electronics Industries Association (EIA). The equivalent European standard is the V.24 specification.

RTS

RTS or Request To Send is a hardware signal defined by the RS-232C standard to request permission to transmit.

RXD

RXD or Receiving data is a hardware signal defined by the RS-232C standard to transport information from one device to another.

Recall Button

This refers to the push-button installed on certain types of industry-standard telephones, for the purpose of providing a ground condition to the line when the button is pressed. When used in conjunction with the SX-200 EL/ML system, pressing the Recall button corresponds to a switch hook flash; for example, when a party is being placed on hold. The button is sometimes referred to as the ground button.

Remote Destination

A location for a DCE device that is not at the central or control site. A typical application would have a terminal at the remote site and the host computer at the central or control site.

Remote Digital Loopback

A test that checks the telephone link and the transmitter and receiver of a remote modem.

Remote LAN

A communications networks that lies outside the geographical area of a central LAN but is connected to it through a communications network.

Ring Lead

The second wire of a telephone pair, originally named because it was connected to the “ring” of a telephone plug. The first wire (called the tip lead) was connected to the tip of the plug.

Router

A device that intelligently connects two or more networks, selecting the best travel path for data and directing the information accordingly.

SMDR

See Station Message Detail Recording.

SMTP

SMTP or Simple Mail Transfer Protocol is a protocol used to transfer e-mail between computers, usually over Ethernet. SMTP is a server-to-server protocol, so other protocols are used to access the messages.

SNAP

SNAP or Subnetwork Access Protocol is an extension of the IEEE 802.2 LLC header used to encapsulate IP datagrams and ARP requests and replies within an 802.X (802.3, 802.4, or 802.5) frame.

SNMP

SNMP or Simple Network Management Protocol is a multi-vendor transport layer network management protocol used to manage bridges, routers, and concentrators.

SSL/TLS

Secure Sockets Layer (SSL) and Transport Layer Security (TLS) are protocols that ensures privacy and data integrity between client/server applications communicating over the Internet. Although their differences are minor, SSL and TLS do not interoperate

Static Route

An entry defined in a routing table that never changes or expires. A static route entry will contain the target address, the next hop router address on the way to that target, and a value assigned to the entry indicating how it rates compared to other routes to the same target.

STP

STP or Spanning Tree Protocol is an algorithm specified in the IEEE 802.3 standard to manage multiple links within a LAN. The Spanning Tree Algorithm allows the use of redundant links within the same network without creating active loops.

Second Dial Tone

A user making a trunk call through a system usually receives dial tone after the handset is removed, and then dial tone from the CO after the trunk access code has been dialed. However, the ARS feature of the SX-200 EL/ML system would mask the CO dial tone, because the outpulsing sequences are isolated from the user. To prevent confusion a second dial tone can be provided to the user by the SX-200 EL/ML system (as a programmable option) at the appropriate point in the outpulsing sequence.

Soft Hold

See Consultation Hold.

Station Message Detail Recording (SMDR)

Station message detail recording (SMDR) records and prints out the details of incoming and outgoing trunk calls in the SX-200 EL/ML system. Such details include the numbers of all parties involved in the call, the time and duration of each call, account codes and other pertinent details.

Store and Forward Dialing

See Overlap Outpulsing.

Subnet Address

An extension of the IP addressing scheme that allows a site to use a single IP address for multiple physical addresses. Outside of the site, routing continues as usual by dividing the destination address into an Internet portion and a local portion. Gateways and hosts inside the site interpret the local portion of the address by dividing it into a physical network portion and a host portion.

Subnet Mask

A bit mask used to identify which bits in an IP address correspond to the network address and subnet portions of the address. This mask is referred to as the subnet mask because the network portion of the address can be determined by the class inherent to an IP address. The subnet mask has the digit '1' in positions corresponding to the network and subnet numbers and the digit '0' in the host number positions.

SX-200 EL/ML System

The SX-200 EL or SX-200 ML system is a microprocessor-controlled telephone system that handles both voice and data switching. The system hardware is electrically compatible with most single line telephones, Mitel proprietary sets, key telephone systems, telephone systems, and central office exchanges.

Synchronous Mode

This term is associated with data which is transmitted in a continuous stream at a fixed rate, with the receiving terminal synchronized to the transmitting terminal by means of elements transmitted on a regular basis. See also Asynchronous Mode.

System Configuration

The particular hardware and software initially installed for the system. Any subsequent additions, deletions and any other changes which occur result in a new system configuration being created. The listing of hardware and software items which comprise the current system configuration can be obtained on command from the maintenance terminal.

System Fail Transfer (SFT)

The system fail transfer feature allows selected stations of the system (or portions of the system, according to the type of outage), to be transferred to certain trunks. Such transfer action is automatic in the event of a failure of the main power supply.

T1

A digital carrier facility used to transmit a DS-1 formatted digital signal at 1.544 Mbps.

TCP

For details on TCP or Transmission Control Protocol, see TCP/IP below.

TCP/IP

TCP/IP or Transmission Control Protocol/Internet Protocol is a connection-oriented protocol that provides transport facilities to dissimilar processors within a single or multiple network environment.

TDM

TDM or Time Division Multiplex is a mechanism for simultaneously transmitting multiple voice, data, and/or video signals by dividing the bandwidth of a link into separate channels or time slots.

TFTP

TFTP or Trivial File Transfer Protocol is a simple TCP/IP file transfer protocol used for downloading boot code to diskless workstations.

Tandem Trunking

Tandem trunking describes the facility of transparently switching co-located trunks together at the SX-200 ICP system. This type of switching is subject to digit modification, and the parameters programmed during CDE for the interconnection restrictions table.

Telco

An abbreviation of tel(ephone) co(mpany).

Telnet

Telnet or Teletype Network is an Internet standard protocol for remote terminal connection service that allows a user to interact with a system at another site as if the terminal were connected directly to the remote machine.

Terminal Emulation

A type of software that allows a computer connected to a piece of equipment to appear to be a traditional terminal with a standard interface-often a DEC VT100 (asynchronous) terminal.

Terminal Hunting

Terminal hunting starts at the first extension in the hunt group and terminates at the first idle extension found. Hunting takes place in the order in which the extensions were programmed into the hunt group.

Throughput

The end result of a data call. Throughput also refers to the actual amount of useful and non-redundant information that is transmitted or processed.

Tie Trunks

Tie trunks directly interconnect two systems together. This enables a station, terminated on one of the systems, to be interconnected to any other station, terminated on the other system.With tandem trunking the calling party can be extended through more than one node of the network.

Tip Lead

The first wire of a telephone pair, originally named because it was the lead connected to the tip of a telephone plug. The second wire of the pair (called the ring lead) was connected to the ring of the plug.

Toll Control

Toll control restricts the users to certain trunk routes and denies the use of specific directory numbers. It is part of the ARS feature. Each user is assigned a COR which is associated with the trunk route tables in ARS and determines what degree of access the particular station has to the trunk network.

Topology

The layout of nodes and links that constitute a LAN.

Transparent Bridging

A bridging protocol in which the bridge learns the addresses and location of the network devices (the PCs) while those devices remain unaware of the presence of the bridge.

Traveling Class Marks

In a private network, the caller’s class of service can be passed to the destination node to control access to services.

Trunk

A communications channel between two points. A trunk generally refers to a high-bandwidth, fibre-optic line between telephone switching centers (central offices). Telephone "trunks" handle thousands of simultaneous voice and data signals, whereas telephone "lines" are the wires from the telephone company to your home or office. Telephone lines are low bandwidth, usually carrying a signal voice call at a time.

UART

UART or Universal Asynchronous Receiver/Transmitter is an integrated circuit used for serial communications, containing a transmitter (parallel-to- serial converter) and a receiver (serial-to-parallel converter).

UDP

UDP or User Datagram Protocol is a series of internet standard network layer, transport layer, and session layer protocols that provide simple but unreliable datagram services.

UPS

Uninterruptible power supply.

V.25 bis

An automatic calling and answering protocol and command set for modems that need auto-dial capabilities.

VT100 (Asynchronous) Terminal

An industry-standard, asynchronous terminal originally created by Digital Equipment Corporation.

VLAN

VLAN or Virtual Local Area Network enables network resources to be grouped together into logical, rather than physical, network segments. Enhance the Quality of Service (QoS) of voice calls by assigning IP phones and PCs to separate VLANs.

WAN

WAN or Wide Area Network is a communications network that serves geographically separate areas.

Wink Start

The wink start feature applies generally to tie trunk circuit operation. When an incoming trunk is seized it may be necessary to prevent the transmission of any digit sequences, until the incoming trunk equipment is ready to receive these digits. When the incoming trunk equipment is ready to receive the digits, a wink start condition is sent from the incoming end to the originating end of the trunk. The distant termination can now send digit sequences over the trunk.

X.25

A CCITT standard that defines the packet format for data transfer in a public data network.

XON/XOFF

A method of flow control in asynchronous transmissions between two PCs. The receiving PC sends an XOFF control character to pause the transmission of data when the receive buffer is full, and then sends an XON character when it is ready to continue the transmission.

[i] Listed Directory Number
[ii] Listed Directory Numbers
[iii] Listed Directory Number