

MiVoice Office 250 6900 Handset Engineering Guidelines

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INSTALLATION GUIDE



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2 6900 Handset Engineering Guidelines

This section of the engineering guidelines provides information on the architecture and implementation of the 6900 Handsets on the MiVoice Office 250.

Overview

The 6900 handset is Mitel's new premium range of telephones. Currently there are four models of handset available; 6920, 6930, 6940. From release 5.1, the MiVoice Office Application Suite supports the 6900 handsets and acts as a configuration server to allow many of the MiVoice Office 250's features to be used on the 6900's SIP telephones.

From 5.2.20 Support was added for the 6970 IP Conference Phone

SIP Implementation

To connect to the MiVoice Office 250, 6900 handsets operate as a SIP extension. All call control functionality is provided through the SIP connection directly to the telephone system (or via a MiVoice Border Gateway).

Each 6900 handset requires a Cat-F license on the telephone system.

 Out of the box, all 6900 handsets come with MiNET firmware for a MiVoice Business. SIP firmware must be loaded onto the handsets before they will connect to a MiVoice Office 250.

Licensing

In addition to the Cat-F licensing on the telephone system, there are two levels of license on the MiVoice Office Application Suite that are used to control 6900 handsets:

- Mitel 6900 Handsets (Basic), provides access to core functionality (Firmware, Keymaps, Configuration, Paging etc)
- Mitel 6900 Handsets (Advanced), provides access to advanced functionality (Currently this includes the ability to Change Caller ID and Recording Pause/Resume toolbar buttons).

Without a license, the 6900 handset can still be connected to the telephone system but would need to be manually configured and would only support the basic SIP features.

 Licensing may change at any time. Please refer to the latest product bulletins for updates on how 6900 handsets are licensed.

Documentation

In addition to this engineering guide, the following documents contain relevant information to implement 6900 handsets on the MiVoice Office 250:

Document	Description
Mitel Communication Service -Technical Manual / Online Help	Provides information on how to configure the MCS server to support 6900 Handsets, including; Dial Plans, Configuration Profiles, Keymap Profiles Alarm Notification, Page Zones and Image Handling.
6.3 MiVoice Office 250 Features and Programming Guide	Provides information on telephone system SIP configuration and restrictions.
6800/6900 SIP 5.0 Admin Guide	Provides an exhaustive guide to all 6900 handset features including configuration options.
6920[/30/40] SIP 5.0 Quick Reference Guides	Model specific quick reference guides to introduce end-user features.
6920[/30/40] SIP Phone User Guides	Model specific phone guides to cover all model specific configuration & features.

Implementing 6900 Handsets on MiVoice Office 250

The following sections cover various aspects of 6900 handset implementation on the MiVoice Office 250.

- Connectivity, provides information on the architecture of how 6900 handsets, the MiVoice Office 250 and MiVoice Office Application Suite work together.
- Rollout, provides information on how to install 6900 series handsets including MiVoice Office 250 configuration, firmware updates and MiVoice Office Application Suite configuration.
- SIP Hot Desking, provides information on how SIP hot desking works and how it is configured.
- Teleworker/Remote worker, provides information on how 6900 handsets can be implemented in Teleworker/Remote worker scenarios.
- Multi-Node, provides information on how 6900 handsets can be implemented in multi-node environments.

69xx Power Requirements

A 69xx working with a DECT headset is still a standard class 3 POE device. As soon as a single PKM is added it requires additional power and becomes a class 4 POE device (See Notes below).

Device Config	POE Class
6940, 6970	3
6940 With DECT Headset	3
6940 With 1 M695 PKM	4 (See Notes 3 & 6)
6940 With 2 M695 PKMs	4 (See Note 3)
6940 With 3 M695 PKMs	4 (See Note 3)
6940 With 3 M695 PKMs + DECT Headset	4 (See Note 3)

An external DC power supply is available for the phones: 50006924 - 6800/6900 AC Adapter Universal.

Note 3: Attention - Devices that require more than 12.95 watts are designated as IEEE 802.3at Class 4 devices. To operate correctly, all IEEE 802.3at Class 4 devices must be connected to a L2 PoE switch that is IEEE 802.3at compliant. Some IEEE 802.3at L2 PoE switches require connected devices to send power allocation requests in excess of 12.95 W (Class 4 devices) using LLDP-MED. Therefore, Administrators must ensure that LLDP-MED is enabled on the L2 switch.

Alternatively, IEEE 802.3at Class 4 devices may be powered with an in-line IEEE 802.3at power injector, or if supported, an AC to DC adapter.

Note 6: Upon initial power up, these sets (with associated accessories) will advertise their power Class via IEEE 802.3at. Once these sets have completed power up, they will advertise modified power requirements via LLDP-MED that are less than what was reported by the IEEE 802.3at Class advertisements. However, the L2 switch must have LLDP-MED enabled to take advantage of the lower power advertisement.

2.1 6900 Handset Connectivity

Each 6900 handset connects to both the MCS and MiVO 250 during normal operation:

- A direct SIP connection between the Handset and the MiVoice Office 250 provides:
 - Audio
 - Call Control
 - Message waiting notification
- Further to this a Http/Https Configuration link between the Handset and MCS provides:
 - Firmware Updates
 - Configuration Files
 - Keymaps
 - Messages (Alarm/DND/FWD/Camp-On Notifications)
 - Status Icons (Alarm/DND/FWD status)

Connection Type	Default Ports Used	Description
SIP	TCP 5060	Used for SIP communication to the MiVoice Office 250
	UDP 20000-20500	Used for RTP Audio traffic between the handset and the MiVoice Office 250
Configuration Server	UDP 69	TFTP for initial firmware download and configuration
	TCP 8202/8203	HTTPS/HTTP requests from the handset to the MCS. This includes: • Firmware • Softkeys • Action URIs • Images • Directories
	TCP 80	HTTP traffic from the MCS to the 6900 handset
	UDP 514	Logging traffic from the 6900 handsets
	TCP 8205	LDAP Directory Queries from the 6900 handsets

SIP Connectivity

The handsets connect directly to the MiVoice Office 250 via SIP for call control and audio. Each handset has configuration options for how to connect to the telephone system:

- SIP Auth Name / Password
- SIP Registrar IP / Port

The SIP configuration, including authentication details is requested from the MCS configuration server during startup. The configuration server will then pass any updates to the phone when online if configuration changes

are made on the server.

If the 6900 handset has successfully connected to the configuration server, it will have been programmed with the necessary SIP connection information.

If the link to the configuration server is offline for any reason (server reboot, network connectivity, etc.), the 6900 handset will still be able to make/receive calls.



Remember that when connecting any SIP device to the MiVoice Office 250, the 'SIP UDP Listening Port' must be enabled in the 'Advanced IP Settings' section. Currently a reboot of the phone system is required after enabling this.



If required, Network Groups on the telephone system can be used for Peer-to-Peer Media



After a MiVoice Office 250 version upgrade, 6900 phones will not re-register with the phone system until they are rebooted or the [SIP Registration Period](#) time expires.

Configuration Server

MCS acts as a 'Configuration Server' for the 6900 handsets. In performing this role, it provides the phone with the following:

- Firmware Updates
- Configuration
- Keymaps
- Messages & Notifications
- Images
- Directories
- Call History
- MiVoice Office 250 Features:
 - ACD
 - UCD
 - IC and System Speed Dial Directories
 - Manual Forwarding
 - Paging
 - Account Codes (Optional & All Calls Following)
 - Night Mode Control
 - BLF/DSS Key Updates (Including a new User based status key)

These features are on top of the telephone system's broad range of SIP features.



Please refer to the [Feature Comparison](#) section for more information on differences between 6900 handsets and digital/Minet handsets.

Firmware Updates

MCS will provide firmware updates when they are available/configured for the handset. This can be via TFTP or HTTPS. TFTP updates are generally only used when first configuring a 6900 handset to convert it from MiNET to SIP firmware. Once this has occurred, MCS will instruct the phone to connect via HTTPS only. All future firmware requests will then be performed via HTTPS.

For more information on configuring firmware updates, please refer to the [Firmware](#) section of the Mitel Communication Service - Technical Manual.

Configuration

Each 6900 handset has a set of configuration options that control how the handset operates. These

configuration options can be set in one of two ways:

- Using the handset (through the settings application on the phone or using it's web user interface)
- MCS Configuration Server

MCS provides configuration profiles so that options can be configured centrally and then applied to multiple handsets.

In addition, MCS controls a subset of the configuration options such as SIP connection details and Dial Plan, to ensure features such as SIP Hot Desking operate correctly, and that the handset operate as closely as possible to their MiNET and Digital counterparts.

 Configuration provided by the MCS will overwrite any configuration added locally on the handset.

For more information on Configuration Profiles, please refer to the [Configuration Profile](#) section of the Mitel Communication Service - Technical Manual.

Keymaps

Each handset has a range of softkeys that can be configured to perform any one of a number of features. Keymaps can be configured locally on the handsets, however, to take advantage of any of the MCS provided features (ACD, DND, UCD etc), the keys must be programmed via a keymap profile on the MCS.

Although keys are programmed on the phone centrally, there are two types or classes of softkey that can be added to a handset:

- Phone based key
- Server based key

The features of phone based keys are provided locally by the handset, with no interaction from the MCS. The features of server based keys rely on updates and responses from the server to operate. Pressing a server based key will cause the handset to request information from the MCS server to action the required feature. Phone based keys will continue to operate if the link to the configuration server is down, server based keys will not.

For more information on Keymap Profiles and features available, please refer to the [Keymap Profile](#) section of the Mitel Communication Service - Technical Manual.

Messages & Notifications

Each handset has the ability to show messages and notifications (some will only display in idle mode) and is used by the MCS server to display the following:

- The handset DND status
- Telephone System Alarms
- Local Forward status
- Camp-On notifications
- DND notifications to calling parties where applicable

These messages are automatically sent by the MCS server to the handset and require no configuration.

In addition to the server based messages, the handset will also show the following messages to the user:

- Message waiting indication
- Missed call indication

 Calls ringing for less than 1 second will not show as a missed call.

Images

Handsets have the ability to display server provided images in a variety of scenarios:

- As an Avatar on the idle screen (6940 & 6970 only)
- Against softkeys
- During a call to show other user's avatars
- Background screen in idle mode
- Custom screen savers

The MCS server provides the handsets with the images. Where possible it will provide avatars for internal extensions that have been mapped to an MCS user and will also match external numbers to users based on external DEE numbers.

Screen savers and background images can be configured on the MCS and assigned to handsets using configuration profiles.

The handset will cache images locally and will request them again when this cache expires.

Directories

Each handset has a feature rich directory facility which allows the user to search for contacts across multiple directory sources at the same time.

The directory sources available include:

- Mobile (if Bluetooth is enabled)
- CSV Imports
- LDAP lookups

The LDAP facility can be used to configure the handsets to lookup an existing customer database.

The CSV import facility is currently used by the MCS to provide the handsets with Internal and System Speed Dial directories from the telephone system.

2.2 6900 Handset Deployment

The following list outlines all the steps involved in deploying 6900 series handsets:

 Ensure you have done all of stage 1 and 2 before you do stage 3.

- [Stage 1 - MiVoice Office 250 system / MiVoice Office Application Suite Configuration](#)
 - Configure the Encryption Password on the MiVoice Office 250
 - Configure the Encryption Password for each Node on the MiVoice Office Application Suite
 - Configure the IP Address and Port numbers for SIP against each node
 - Prepare Keymap Configuration Profiles
- [Stage 2 - Create 69xx SIP Devices on the MiVoice Office 250](#)
 - Add a '69xx/Phone Manager SIP Phone' for each 6900 handset and SIP hot desk phone using the parameters in the [MiVO Office 250 configuration](#) section
 - Add SIP credentials to each SIP device on the phone systems page (only if not using 6.3 SP1)
- [Stage 3 - Phone Deployment](#)
 - Update the handset firmware from MiNET to SIP
 - Initialize the handset with the MCS server
 - Apply keymaps and/or configuration profiles

 Phones should be deployed straight to site and not pre-staged. If pre-staged, a phone will need to be reset to factory default or the IP address of the configuration server will need to be manually updated so that they can find the MiVoice Office Application Suite server on it's new IP address.

This is because the phones use HTTP/HTTPS to communicate with the MiVoice Office Application Suite server after their initial boot and the address of the server is persistently stored on the phone.

Stage 1 - MiVoice Office 250/Application Suite Configuration

The following configurations should only need to be performed once. Stages 2 & 3 will need to be repeated for each new 69xx phone deployed.

When using release 6.3 SP1 or higher of the MiVoice Office 250, MCS has the ability to query all SIP Authorisation Credentials from the telephone system to use with Phone Manager Softphones and 6900 phones. This integration simplifies the process of installing Softphones/6900 phones and minimises the risk of mis-configuration.

To support this feature, a new configuration section within MiVO 250 Database Programming has been created:

	Name	Value
▼ MiVoice Office 250 > Maintenance Accounts - Software License ▼ System App Suite Server Configuration - CloudLink Gateway > Controller > Conference-Related Information > Devices and Feature Codes > Echo Profiles - E-mail Gateway > File-Based MOH - Flags > Hunt-Group Related Information > IP-Related Information > IP Settings	Encryption Password	

Encryption Password

On each node in the MiVO 250 network, an Encryption Password needs to be configured which will allow MCS to query and decrypt the SIP authorisation credentials.

If the password is not configured, MCS will not be able to query the credentials from the PBX and they will have to be configured manually. See the [Device Configuration](#) section for more information.

Once the encryption password has been configured on the telephone system(s), it must also be configured in the [Nodes](#) section of the MCS configuration website.

 In addition to using requiring 6.3 SP1 or higher, CT Gateway release 5.0.64 or higher is also required for the SIP authorisation credential query to work.

Node Configuration for SIP

The MCS server needs to provide each 6900 phone and Phone Manager Softphone with the IP address of a SIP server to register with (the MiVoice Office 250). The IP address required will depend on which MiVoice Office 250 node the SIP extension is configured on and whether the phone is local or a teleworker.

For each node on the MiVoice Office 250 network that MCS is connected to, it is important to configure the IP address/port number to be used for SIP registrations.

For information on configuring the IP address(es)/Ports for each node, please refer to the [Node Configuration](#) section.

Keymap & Configuration Profiles

Each handset will require a keymap and configuration profile. The MCS comes with default keymap and configuration profiles with the most commonly selected options already added. These can be customised as required or new profiles can be added to suite the customer's company wide or department needs.

 Having Time Server Settings (Time Server 1-3) in the Configuration Profile used will ensure the Handsets keep good time. Where a configuration is used by local and remote handsets, ensure at least one NTP server specified is a publicly accessible NTP server that a remote handset can access

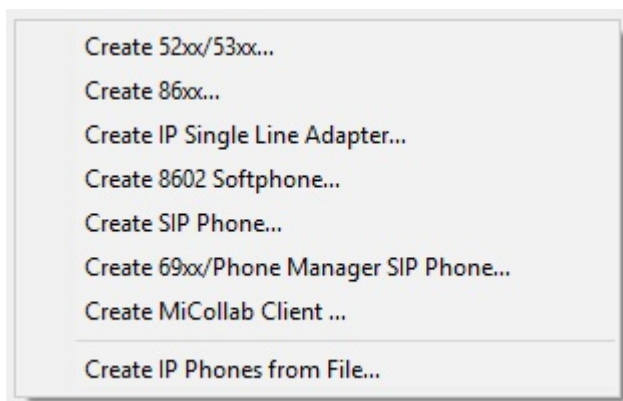
 Other items to configure on the MCS server include: Page Zones, Background & Screen Saver Images

Stage 1 is now complete - continue to [Stage 2](#) - Create 69xx SIP Devices

Stage 2 - Create 69xx/Phone Manager SIP Phones

69xx SIP Phone

From release MiVO 250 6.3 onwards, a new SIP phone type called '69xx/Phone Manager SIP Phone' (renamed from '69xx SIP Phone' in 6.3 SP2) is available for creating SIP extensions on the telephone system for use with Phone Manager softphones & 6900 phones.



When SIP extensions are created using this type, the SIP Phone Groups created will automatically be configured with the required settings and will have a default inbound authentication applied with a randomly assigned password.

 If a user is using a 6900 handset and a softphone (on either or both of Phone Manager Desktop & Phone Manager Mobile) it is important to set them up with separate SIP Endpoints on the phone system.

 For release prior to 6.3, the generic SIP Phone type should be used for Phone Manager Softphones. Please review the Phone Group settings under [Manual SIP Configuration](#) to check the required configuration.

 Remember that when connecting any SIP device to the MiVoice Office 250, the 'SIP UDP Listening Port' must be enabled in the 'Advanced IP Settings' section. Currently a reboot of the phone system is required after enabling this.

 If release 6.3 SP1 of the MiVoice Office 250 & Encryption Password feature is not being used, the SIP credentials will need to be manually configured for each new SIP device added in DB Programming. Please refer to the [Manual SIP Configuration](#) section of the Technical Manual for more information.

Stage 2 is now complete - continue to [Stage 3](#) - Phone Deployment

Stage 3 - Phone Deployment

Once all the preparation work has been completed on the telephone system and MiVoice Office Application Suite, the phones can be deployed. Out of the box, all 6900 phones are shipped with MiNET firmware. The MiVoice Office Application Suite has a built in TFTP server that will provide the phones with SIP firmware the first time they connect.

The phones must be provided with the address of the MiVoice Office Application Suite so that they can download the SIP firmware and subsequently, their SIP & softkey configuration so that they can connect to the telephone system.

 The information below describes deploying handsets local to the MiVoice Office Application Suite. We always recommend (where possible) configuring handsets locally to MiVoice Office Application Suite, prior to deploying in a remote location. For remote deployment of a 6900 Handset where the firmware has not been updated refer to [6900 Handset Diagnostics & Troubleshooting](#) section

The sections below outline the methods available for phone deployment:

1. Automatically using [Multicast DNS](#)
2. Automatically using [DHCP \(with Vendor/PBX specific options\)](#)
3. [Manually](#) (on the handset or the handset's web user interface)

1. Automatic Configuration Server discovery - using Multicast DNS

The easiest way of providing 6900 phones with the address of the MiVoice Office Application Suite is using the Multicast DNS auto discovery (by default, this is enabled on the MiVO App Suite v5.1.14 or later). If the phones are not provided with a DHCP option then they will perform an 'Auto Discovery' to find the configuration server.

 With Multicast DNS a method of providing IP address and other basic networking information to the phones is still required and will usually be DHCP. Multicast DNS is used only to supply the Configuration Server address.

If the DHCP server provides any Vendor/PBX specific options (e.g. 43, 66, 125) then the phone will not attempt to discover the configuration server by Multicast DNS. Either remove these settings from the DHCP server or use method 2 with DHCP.

 If there is more than one MiVoice Office Application Suite on the network, the phone will present a list of auto discovered servers for the user to choose between.

Once the phones have been provided with the address of the MCS server, they will contact the server and the registration process (and any firmware updates) will begin. The phone will reboot a number of times during this process. The phone will now be registered with the MCS server and display a red Setup screen shown in the next section.

Continue to the [Initialising Handset](#) section.

2. Automatic Configuration Server discovery - using DHCP (with Vendor/PBX specific options)

If the phones are on a different subnet to the server, DHCP can be used instead. Please review the section below if DHCP is being used paying specific attention to the information on removing options 43 & 125.

DHCP Configuration

If using DHCP to provide the phones with the address of the MCS server, it is recommended that the following DHCP options are used:

- Option 66 for the 6900 SIP phones
- Option 128, 129 & 130 for the 5300 Series MiNET phones

Using these DHCP options will allow 5300 & 6900 phones to work alongside each other on the same network using the same DHCP server.

Configure Option 66 in your DHCP server with the IP address of your MCS server.

For specific information on configuring the MiVoice Office 250's built-in DHCP server, please refer to the ['MiVoice Office 250 DHCP Configuration'](#) section.

 When using option 66 on a DHCP server, it should be configured as an ASCII parameter containing an IP Address (not an IP Address parameter). This includes the built in DHCP server of the MiVoice Office 250.

 The 6900 Handsets will also pick up the following DHCP options and use them as the configuration server address (in the order listed):

- 43, 159, 160, 66

If any of these options are already in use on the DHCP server then it will overwrite the configuration server used on the handset and could stop it working. One possible source for causing a conflict is DHCP options for 5300 phones using option 43/125. These can be changed to use the preferred options 128, 129 & 130 to avoid conflict.

Remove options 43 & 125 from the DHCP if they are there so that the 6900 phones do not pick up the phone system's address.

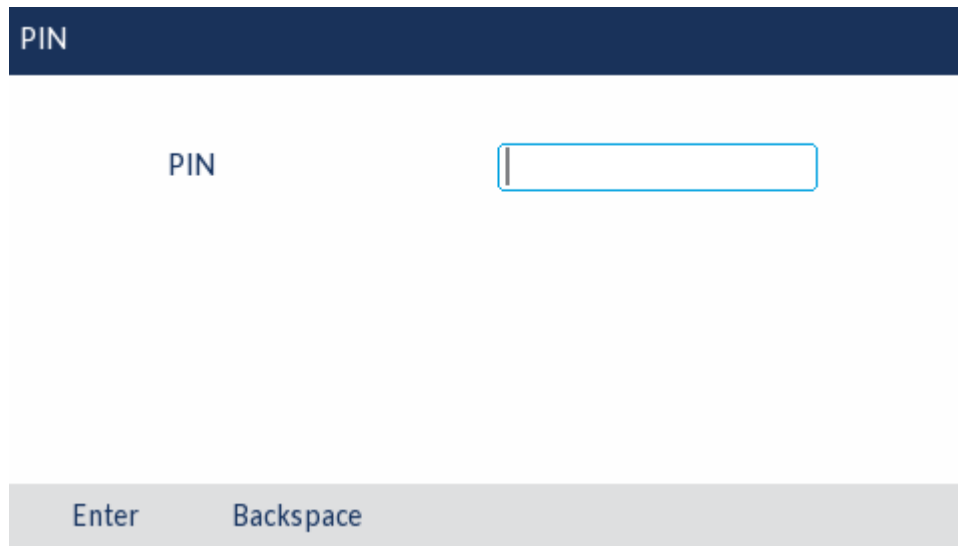
Once the phones have been provided with the address of the MCS server, they will contact the server and the

registration process (and any firmware updates) will begin. The phone will reboot a number of times during this process. The phone will now be registered with the MCS server and display a red Setup screen shown in the next section

If you have this continue to the [Initialising Handset](#) section.

Most common Incorrect Configuration

If the phone displays the following screen instead of the red setup screen, it has connected to the MiVoice Office 250 via MiNET.



If this happens, the phone has not been passed the IP address of the MiVoice Office Application Suite and the DHCP configuration is incorrect. Once the DHCP configuration has been fixed, restart the phone. DO NOT enter a PIN at this point.

Now bypass Manual Configuration section below and move on to the [Initialising Handsets](#) section.

3. Manual Configuration Server discovery

 DHCP is the preferred method to automate Firmware update and Configuration setup as it reduces the number of steps involved.

If you cannot use DHCP then follow the steps for manual setup below

Step 1

Out of the box, all 6900 handsets come with MiNET firmware installed. The first step in deploying the handsets is to change this to SIP firmware so that they can talk to the telephone system and the configuration server (MCS).

- Press the settings button (⚙️) on the handset, then press the 'Advanced' key along the bottom of the screen.
- At this point you will be prompted to enter the administrator password. The default MiNET password is '73738'
- Navigate to 'Network' and then 'Static Settings' using the navigation keys (D-pad) or the touch screen if using a 6940 or 6970.
- In the 'TFTP server IP Address' box, enter the IP address for the MCS server then 'Close' to exit the menu, and the handset will restart connecting to MCS, download and update to SIP firmware.

Step 2

Once the phone is running the correct firmware version, the next step is to tell the phone where the

configuration server is.

- Add the configuration server details on the phone manually, press the settings button (⚙️) on the handset, then press the 'Advanced' key along the bottom of the screen.
- At this point you will be prompted to enter the administrator password. The default SIP password is '22222'.
- Once the password has been accepted, use the navigation keys (D-pad) or touch screen on 6940 or 6970 to navigate to the 'Configuration Server' section.
- Populate the 'Primary Server' box with the IP address or DNS name of the MCS server. Press 'Save' and then reboot the handset.

After a reboot, the phone will connect to the MCS server. As the phone is connecting over TFTP, the MCS server will provide HTTPS connection information and will then tell the phone to reboot.

 It is possible that there will be more than one reboot at this stage as the firmware update is completed

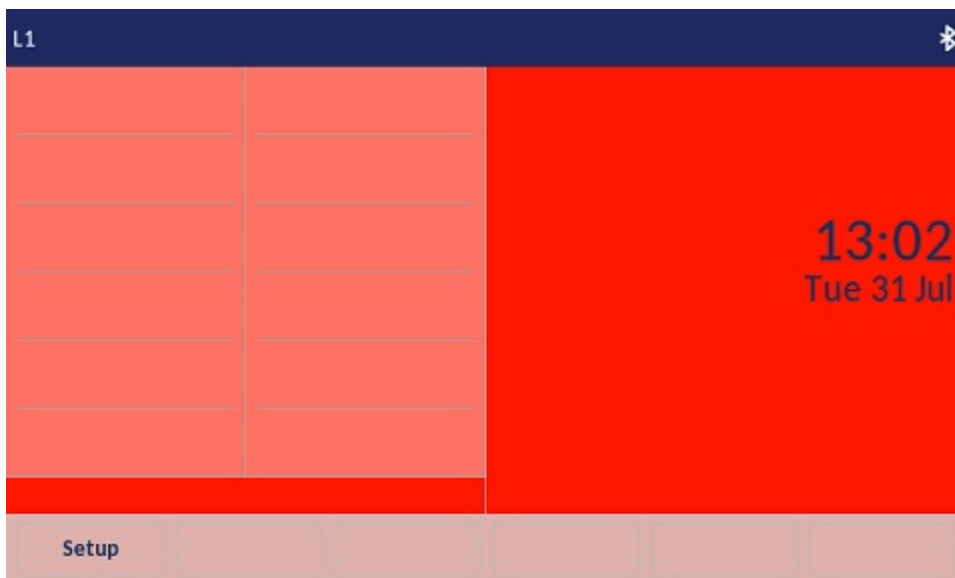
The phone will now be registered with the MCS server and display a red Setup screen shown in the next section.

continue to the [Initialising Handset](#) section below.

Initialising Handsets

Once a handset makes it's first connection request to the MCS, it will be in 'Unassigned' mode. This means the MCS server is aware of the handset, but does not know which SIP extension on the telephone system it should be configured as.

The screen of the phone should match the screen shot below:



Handsets can be initialised by using either (a) the 'Setup' button on the phone, or (b) editing the handset on the 'Phones' page of the MCS website:

 Remote phones do not have the Setup button and require a SIP extension to be assigned by option (b)

(a) To use the setup process on the phone, press the 'Setup' button and enter the setup PIN (default ***). The handset will then prompt for the extension number. Once the extension number has been correctly entered, the MCS server will pass the handset all SIP, keymap and configuration information it requires.

(b) To use the configuration in MCS, browse to MCS Configuration -> Features -> 6900 Handsets -> Phones. Locate the MAC address of the phone in question and Edit the record to add the SIP extension number.

2.3 MiVoice Office 250 DHCP Configuration

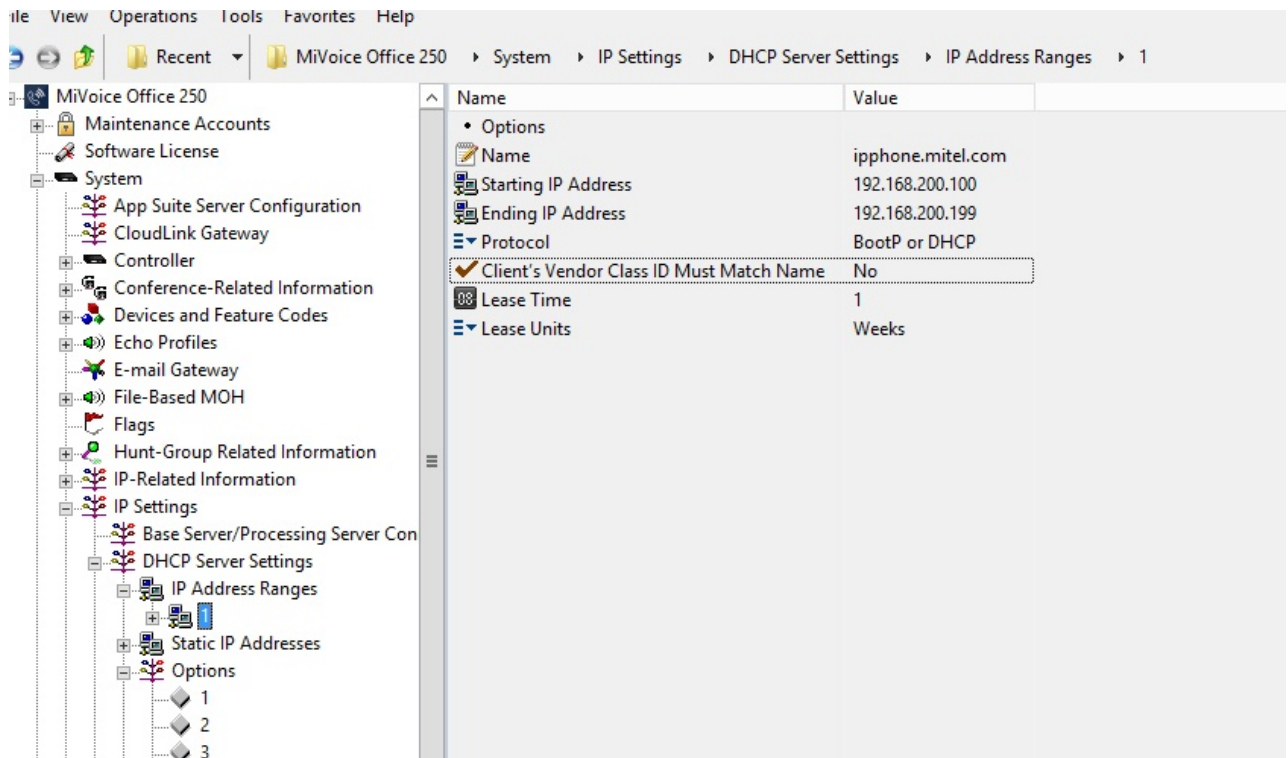
If using the MiVoice Office 250 built in DHCP server, it is important to review the follow these configuration guidelines to ensure the DHCP performs as required.

DHCP Server Configuration

The DHCP server should be configured to support both 6900 and 5300 series phones.

Disable Vendor Class ID Matching

By default, the IP Address range provided will only provide addresses to 5300 series phones. To open the range up to 6900 phones, uncheck the 'Client's Vendor Class ID Must Match Name' option.



Removing the Vendor Class ID matching will cause all devices (including computers) to receive DHCP addresses from the MiVoice Office 250.

6900 Vender Class IDs: AastraIPPhone6920, AastraIPPhone6930 & AastraIPPhone6940

Update the DHCP Options

The DHCP options must now be updated. The image below shows the minimum required configuration to support 6900 and 5300 series phones:

Office 250 > System > IP Settings > DHCP Server Settings > Options

Option Definition ID	Option ID	Name
1	66	6900
2	3	Router
3	128	mitel-option-128
4	129	mitel-option-129
5	130	mitel-option-130

The following step outline how to configure the DHCP options:

- Remove Options 43 & 125
- Add Option 66

With the DHCP server configured in this way, option 66 will provide the 6900 series phones with the information they need to connection to the MiVoice Office Application Suite and options 128, 129 & 130 will provide the 5300 series phones with the information they need to connect to the MiVoice Office 250.

 There is a known issue with the MiVoice Office 250 DHCP server that requires all Option Definition IDs to run contiguous (sequential starting from 1). If there is a break in the Option Definition IDs then options configured after the break may fail to be presented correctly to the phone. This issue is fixed in MiVoice Office 250 6.3 SP2 and higher.

Option 66

The image below shows how option 66 should be configured (replace the option value with the IP address of the MiVoice Office Application Suite). The format MUST be set to 'ASCII String' in order for this option to work.

cent > MiVoice Office 250 > System > IP Settings > DHCP Server Settings > Options > 1

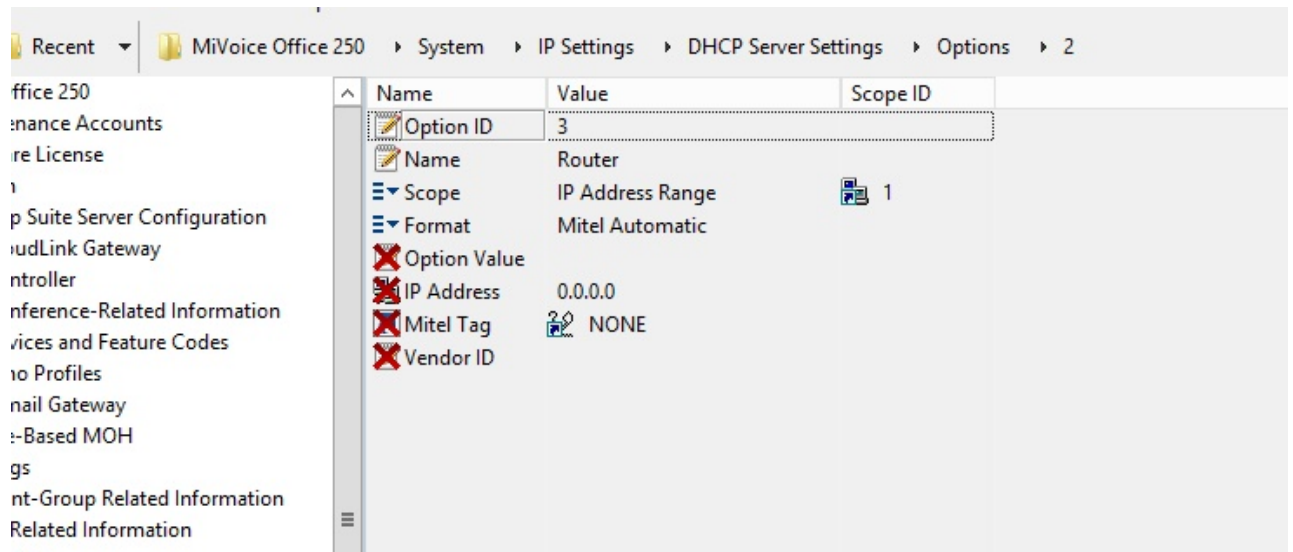
Name	Value	Scope ID
Option ID	66	
Name	6900	
Scope	IP Address Range	1
Format	ASCII String	
Option Value	192.168.1.4	
IP Address	0.0.0.0	
Mitel Tag	NONE	
Vendor ID		

 If Option 66 is already used on-site for some other purpose, it can be removed from the option list. The phones will use Multi-Cast DNS to find the server. If Option 66 is removed, ensure the options left are re-configured so that the Option Definition IDs are still contiguous.

Option 3

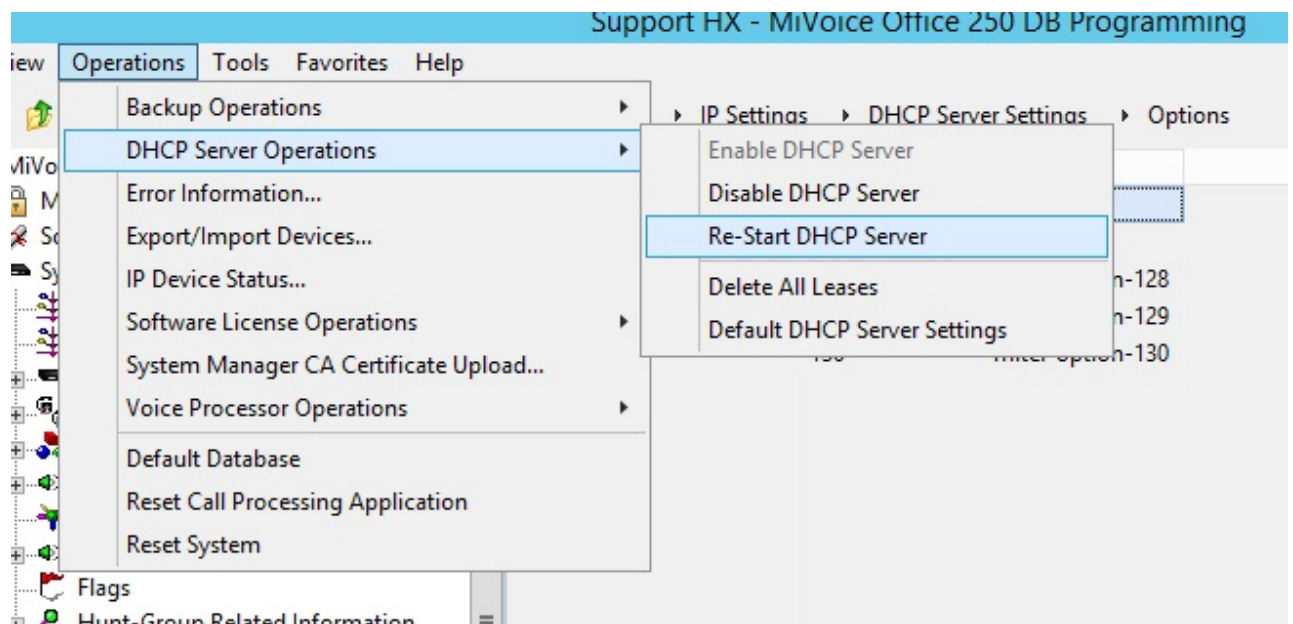
The image below shows how option 3 should be configured. The 'Mitel Automatic' setting will pass through the

router/gateway that has been configured for the MiVoice Office 250's network adaptor. If an alternative IP address is required, select 'IP Address' as the format and enter the IP address in the option field.



Restart the DHCP Server

Once all DHCP configuration has been completed, the DHCP server must be restarted for the changes to take effect. The image below shows how to restart the DHCP server without having to restart the MiVoice Office 250.



2.4 6900 Handset SIP Hot Deskling

SIP Hot Deskling is a feature that has been designed specifically for 6900 handsets. Any '69xx/Phone Manager SIP Phone' extension on the telephone system can be configured on the MCS as a SIP Hot Desk Phone. Once this has been done, it can be used to log onto any 6900 handset connected to the MCS server that has been enabled to accept hot desk logons.

 SIP hot deskling is different from both agent hot deskling and MiVoice Office 250 hot deskling. The different types of hot deskling cannot be used together.

SIP Hot Desk Features

When logging into a 6900 handset with a SIP hot desk login, the handset will be reconfigured so that it becomes the hot desk extension. The following list shows what configuration gets updated on the handset:

- SIP credentials (Extension No, Username etc)
- Keymap
- Configuration Options

When a SIP hot desk logs in to a handset, the handset will cease to be its base SIP extension and will become the SIP hot desk extension. If the SIP hot desk that has logged in resides on a different node than the handset's fixed extension, the handset will un-register as a SIP device on one node and will re-register on the new node where the handset resides.

When the SIP hot desk logs out, the handset will revert back to its fixed extension.

SIP Hot Desk phones can be configured on the MCS for all features that a fixed 6900 handset can:

- Configuration Profiles
- Keymaps
- Alarm Notification
- Page Zones

All of the above features will be applied when a SIP hot desk phone logs on.

If a SIP hot desk phone has not been assigned a Keymap Profile or Configuration Profile directly, the keymap and configuration profiles for the base extension will be used.

Adding a SIP Hot Desk Extension

 A SIP extension to use for Hot Deskling is no different in PBX configuration from a SIP Extension used as a base handset extension. Any SIP extension not assigned to a base extension is available for use as a SIP Hot Desk Extension

On the MCS server, Configuration -> Features -> 6900 Handsets

Enable Base extensions for Hot Deskling

In the Phones section edit all the base extensions you will want to be able to SIP Hot Desk into and Tick the 'Enable Hot Deskling check box'

Enable an SIP Extension to be a Hot Deskling device

In the SIP Hot Deskling section, press the 'Add' button. In the form that appears, enter the number of the SIP device on the telephone system to configure as a hot desk phone (the device picker will filter out any SIP device which already have a fixed mapping to a handset).

Once a hot desk phone has been created, it can be used to logon to any handset that has been enabled for

accepting hot desk logons.

 It is important to set authentication against each SIP extension and ensure the password is complex. For example, *Mitel*Server1!*. If connecting externally through and MBG, a complex password is a requirement.

 If a user is using a 6900 handset and a softphone (on either or both of Phone Manager Desktop & Phone Manager Mobile) it is important to set them up with separate SIP Endpoints on the phone system.

2.5 6900 Handset Teleworker

69xx phones are supported as Teleworkers on the MiVoice Border Gateway (MBG) using a SIP User. The following section outlines the configuration required to deploy a teleworker on MBG.

- [Stage 1 Configure AppSuite for MBG IP details](#)
- [Stage 2 Configure Port Forwarding on MBG for 6900 Data Connection](#)
- [Stage 3 Configure MBG ICP connection for PBX](#)
- [Stage 4 Configure API integration for automatic provisioning](#)

(All the above are 'one off' configuration items unless the infrastructure changes)

- [Stage 5 Configure each device for remote access](#) (this is required for each device that needs remote access)
- [Stage 6 Manually add the Configuration Server address for each remote phone](#)

MiVoice Border Gateway Requirements

To use MiVoice Office Application Suite with a MiVoice Border Gateway (MBG) for remote connections, the MBG must be running v10 or higher and be configured in Gateway mode.

Stage 1 - Configure MBG IP details in MiVoice Office Application Suite

For the MCS to be able to support teleworker SIP extensions (either Phone Manager Desktop Softphones or 69xx phones), it needs to know two pieces of information:

1. The internal IP Address of the MBG
2. The external IP Address of the MBG

The internal IP address of the MBG needs to be known for two reasons. The first is so that the MCS can identify which 69xx phones it receives requests from are actually Teleworkers. The second is so that it knows how to communicate with the MBG's API for SIP User deployment.

The external IP address of the MBG needs to be known so that it can be passed to Phone Manager Desktop and 69xx phones when they are registering their SIP connections and connecting to other MCS services.

Configure the MCS with the MBG's Internal IP Address:

- 'Configuration -> Site Settings -> Phone Systems -> MiVoice Border Gateway', enter the internal IP address of the MBG

Configure the MCS with the MBG's External IP Address for each Node:

- 'Configuration -> Site Settings -> Phone Systems -> [Your PBX Name]', enter the external IP address of the MBG into the NAT IP Address property for each node. If the external registration port for SIP has been changed on the MBG, update the NAT SIP Port as well.

Configure the MCS with the MBG's External IP Address for Remote Client Connections:

- 'Configuration -> Site Settings -> Client Locations -> Remote', enter the external IP address of the MBG into the 'NAT IP Address/Fill Hostname' property. This will be used to provide the remote 69xx phones with the URIs they need to communicate with the MCS server for firmware/softkeys etc.

Once the MCS has the information above, it will be able to identify teleworker 69xx phones and will be able to pass them the information required to connect to the MBG.

 The External IP Address entered into the remote section of the [Client Locations](#) will also be used by Phone Manager when connecting to the MCS Server.

Stage 2 - Configure Port Forwarding on MBG

69xx Phone Port Forwarding on MBG

69xx phones use the following TCP port to communicate to the MCS server:

Port	Target	Description
TCP 8202	MiVO App Suite Server	Used to communicate to the MCS server to provide configuration, user data etc.

The other ports associated with 6900 phones (LDAP, TFTP, Multicast, Syslog) should not be opened up through the MBG.

 Configuration of the SIP devices associated with the 6900 is done automatically by the MiVoice Office Application Suite. Please refer to the [Automatic Teleworker Provisioning](#) section for more information.

Configuring MBG Port Forwarding for 69xx Phones

Complete the following configuration on the MBG:

- On the 'MBG Security -> Port Forwarding' page, create the port forwarding rules for TCP 8202 with the Destination Host IP Address pointing to the IP address of the MCS host.

Protocol	Source Port(s)	Destination Host IP Address	Destination Port(s)	SNAT	Action
TCP	8202	172.19.5.46	8202	Yes	Remove

 SNAT must be enabled on all the port forwarding rules added.

Stage 3 - Configure MBG ICP connection for PBX

Add one or more Nodes as ICPs on the MBG:

- On the 'MiVoice Border Gateway -> Service configuration -> ICPs' page, create an ICP instance for each MiVO 250 node that will be supporting teleworker SIP phones.

The screenshot shows the Mitel Standard Linux web interface. The top navigation bar includes 'Applications', 'ServiceLink', 'Administration', 'Security', and 'Configuration'. The left sidebar lists various services and tools. The main content area displays the 'ICP Information' table, which lists configured ICPs. Below the table is an 'Update default ICPs' button.

Default for MNet	Default for SIP	Name	Hostname or IP address	Type	Installer password	SIP capabilities	Indirect call recording capable		
☒	☒	DEV-MVO-04	192.168.106.1	MiVoice Office 250		UDP	☒	☒	☒

Ensure the 'Hostname or IP Address' setting used for the ICP matches the IP Address configured for the Node on the MCS server.

Stage 4 - Configure API integration for automatic provisioning

Automatic Teleworker Provisioning

To simplify the deployment of SIP teleworker phones, the MCS can use an API on the MBG to automatically provision SIP Users for SIP extension. This provides the following benefits:

- The default random credentials created on the MiVO 250 for each SIP extension can be passed to the MBG automatically
- The MCS can use randomly generated set-side credentials for teleworker phones
- There is no need to re-type the authorization credentials into MCS and the MBG, removing a repetitive and time consuming task and reducing the risk of mistakes
- The engineer/administrator deploying the teleworker phone does not need to know the SIP authorization credentials at any stage

For automated provisioning to work end-to-end between the MiVO 250, MiVO App Suite and MBG, the MiVO 250 must be running at least 6.3 SP1 and any CT Gateway must be running at least 5.0.64.

Enabling the Rest API for Automatic Teleworker Deployment

For the MCS to be able to communicate with the MBG and deploy teleworker SIP Users, it requires some connection information as well as a valid API token from the MBG. This section documents how to configure the MBG to accept API requests and the steps involved in setting up the token exchange. To complete this process, access to the MBG website and the MCS configuration website are required.

⚠ Once the Rest API has been enabled on MCS and it has a valid token, the MCS server will take any SIP extension that has been configured with remote authorization credentials and provision it onto the MBG. Any existing credentials configured for SIP Users on the MBG will be overwritten with those configured on the MCS.

Step 1: Create a new web service consumer on the MBG:

- On the MBG 'Administration -> Web services' page, press 'Start' to enable web services then press 'Add a new consumer' and provide the following information:
 - Active = Yes
 - Name = MiVoice Office Application Suite

- ConsumerID = MiVOAppSuite
- Permissions:
 - Base/managetoken = Read/Write
 - MBG/v1/icps = Read
 - MBG/v1/devices = Read/Write
- Make a note of the shared secret then save the new consumer.

Step 2: Complete the token request from MCS to MBG:

- On the MCS 'Configuration -> Site Settings -> Phone Systems -> MiVoice Border Gateway' page:
 1. Check the 'Enable Rest API' box.
 2. Press the 'Request Access Token' button to load the 'Request Access Token' form.
 3. Enter the Name, Consumer ID and Shared Secret to match those created on the MBG in step 1
 4. Press the 'Save & Test API Credentials' button to initiate a token request with the MBG server.
- On the MBG 'Administration -> Web services' page:
 1. Locate the 'Temporary tokens' section at the bottom of the page
 2. Press the 'approve' button against the temporary token request.
 3. Highlight and make a copy of the 'Verifier' code (you may need to refresh the page to see the temporary token)
- On the MCS 'Configuration -> Site Settings -> Phone Systems -> MiVoice Border Gateway' page:
 1. Paste or enter the verifier code into the request window
 2. Press the 'Retrieve Final Access Token' button to complete the token request with the MBG server.

If the verifier is correctly entered, the MCS should be able to successfully request an API token from the MBG. This token will allow the MCS to provision SIP Users on the MBG for a period of 12 months. To avoid having to repeat the above process every 12 months, the token's expiry date can be extended on the MBG by pressing the 'Renew' button against the token in the 'Final tokens' section of the web services page.

 The internal IP address configured in the Nodes section of the MCS website must match the IP Address configured on the corresponding ICP on the MBG website. If they do not match, the MCS will not be able to find the correct ICP when deploying a teleworker phone.

For information on how to provision SIP Users manually, please refer to the [Manual Teleworker Provisioning](#) section of the MiVoice Office Application Suite Technical Manual.

 Each Teleworker connection on the MBG requires a Teleworker licence.

Stage 5 - Configure each device for remote access

Automatic Teleworker Device Deployment

Once the Rest API has been configured, MCS will automatically provision teleworker SIP extensions on the MBG.

The MCS will provision any SIP extension that has had the 'Use remote authorisation credentials' setting configured against it:

Provisioning a SIP Teleworker Extension

To instigate the MCS provisioning of a SIP extension on the MBG, navigate to the 'Configuration -> Site Settings -> Phone Systems -> [Your PBX Name]' page. Edit the required SIP extension and then check the 'Use remote authentication credentials' check box. The MCS will pre-populate the remote authentication name and password with random values. Pressing save will update the credentials stored for the extension and will start the teleworker provisioning process.

Un-Provisioning a SIP Teleworker Extension

To un-provision a SIP extension from the MBG, follow the provisioning process but uncheck the 'Use remote authentication credentials' check box. Once save is clicked it will instigate the MCS removing the SIP User from the MBG.

This will only work for SIP extensions that were previously provisioned by the MCS. If a SIP User was manually added to the MBG it may need to be manually removed. If a SIP extension has been provisioned on the MBG by the MCS, the MBG's ID for the SIP user will be displayed on the SIP Authorisation form of the extension in MCS.

When using a MiVoice Border Gateway, the internal authentication name must match the extension number of the phone otherwise authentication with the telephone system will fail.

Any SIP extension provisioned using this method will automatically have a random remote authentication username and password assigned if they do not have them set already.

In addition to configuring SIP Users for teleworker extensions, they must also be configured for any SIP Hot Desk extensions that will be logging into a teleworker phone.

Stage 6 - Manually add the Configuration Server address for each remote phone

Remote Phone Configuration for SIP Firmware

The following steps outline how to manually configure the Configuration Server connection details for each teleworker 69xx phone.

 Each phone must be updated to SIP firmware before this configuration can be completed. It is recommended to upgrade from MiNET to SIP firmware on the local network and then manually configure the phone before sending it to the teleworker location.

- Add the configuration server details on the phone manually, press the settings button (⚙) on the handset, then press the 'Advanced' key along the bottom of the screen.
- At this point you will be prompted to enter the administrator password. The default SIP password is '22222'.
- Once the password has been accepted, use the navigation keys (D-pad) or touch screen on 6940 & 6970 to navigate to the 'Configuration Server' section.
- Populate the following entries:

Download Protocol: HTTPS

HTTPS Server: [Enter the external IP Address or external DNS name of the MCS server configured in Stage 1]

HTTPS Port: 8202

Cert Validation: false

- Press 'Save' and then reboot the handset.

After a reboot, the phone will connect to the MCS server and download firmware other configuration information.

 It is possible that there will be more than one reboot at this stage as the firmware update is completed

The handset should now be registered with the MCS server.

The Phone will have Line 1 and Line 2 showing as the Top Sofkeys.

Unlike for local deployment as 'Setup' button will not be available and the SIP extension needs to be configured in the MCS server.

The 'Phones' page within the configuration section of the MCS website can be used to view whether the handset has been identified. The MCS uses the handsets MAC Address to uniquely identify it.

2.6 6900 Handset Multi-Node Implementations

6900 handsets can be used in MiVoice Office 250 multi-node environments. When implementing multiple nodes, there are two options available:

- Single MiVoice Office Application Suite, recommended for multiple nodes at a single location
- Multiple MiVoice Office Application Suites, recommended for nodes at multiple locations

The correct implementation must be chosen depending on the infrastructure and customer requirements. The following section outlines each implementation.

 With either implementation, a CT Gateway is required for the MiVoice Office Application Suite(s) to connect to.

Single MiVoice Office Application Suite

Having a single MiVoice Office Application Suite installation acting as configuration server for all 6900 handsets provides the greatest level of functionality for the customer. Each handset will be provided configuration, keymaps and all other features from a single point. This implementation is recommended for when multiple MiVoice Office 250 nodes are at the same location or when there are resilient links between locations.

With this implementation there is no loss of functionality, all 6900 handset features will work as designed including SIP Hot Desking, which will allow users to hot desk onto any 6900 handset on any node.

Restrictions

This implementation relies on the MiVoice Office Application Suite being able to communicate with all nodes and all 6900 handsets all the time. If there are WAN links involved in the implementation then there is a risk of downtime if the link is down for any period of time.

If the link between sites is down, any 6900 handsets that cannot communicate with the MiVoice Office Application Suite will lose all server provided keys and features. The handset will continue to operate as a SIP phone and will be able to make and receive calls, but the following features will cease to be available until the link is restored:

- Server Based Keys (BLF Keys, DND, Presence Profiles, UCD, FWD, Account Codes etc)
- Screen Savers
- SIP Hot Desking
- Avatar images, local and for calls.
- Phone Status Messages/Icons
- Any configuration or keymap changes while the link is down.

Once the link is restored, the features listed above will work as designed.

Multiple MiVoice Office Application Suites

Having multiple MiVoice Office Application Suite servers provides the greatest level of resilience and is the recommended for implementations where nodes are installed at different locations and are connected using WAN links.

When using this implementation, keymap profiles, configuration profiles, screen savers and all other configurations must be programmed on each MiVoice Office Application Server individually.

However, with this implementation there are restrictions in functionality that must be outlined to the customer.

Restrictions

The following features will not be available when 6900 handsets are using different configuration servers:

- User BLFs
- Avatar images for users on different configuration servers
- Multi-Node SIP Hot Desking

 Paging between handsets will only work when handsets can receive Multicast transmissions. These packets are not routed and so paging will only work between handsets on the same network segment.

2.7 6900 Handset Feature Comparison

When using 6900 handsets in conjunction with a MiVoice Office 250, many of the features available are configured through profiles on the MCS server rather than on the telephone system. In addition, many of the features have slightly different names than those traditionally used on the MiVoice Office 250.

This section of the manual is designed to review features of the MiVoice Office 250 and provide information about what they are called and how they are configured. In addition, it highlights features of the MiVoice Office 250 that are not yet available on the 6900 handsets or have restrictions to their use.

MiVoice Office 250 Feature	Supported on 6900 Handsets	6900 Handset Equivalent
ACD Agent	Yes	ACD agents are supported on 69xx phones with softkeys for login/logout and end wrap. Auto answer is not natively supported but can be implemented using the Phone Manager 'Auto-answer hunt group calls' feature.
Account Codes - All Calls Following	Yes	N/A
Account Codes - Optional	Yes	N/A
Account Codes - Forced	No	N/A
Agent Help	Yes	N/A
Background Music	No	N/A
Calling Party Name / Number	Yes	N/A
Class of Service	Yes	N/A
Do Not Disturb	Yes	Do Not Disturb works but not DND Override
Dynamic Extension Express	Partial	Push and Pull features are available on 6900 phones but work differently to 53xx phones. Calls which are pushed/pulled are sent back to ring the DEE main extension so the user can re-answer on the required device.
Group Listen	No	N/A
Hot Desking	Yes	SIP Hot Desking can be used in place of Mitel Hot Desking but, they are not interchangeable. You cannot log onto a 6900

		handset with a Mitel Hot Desk Phantom or visa versa.
House Phone	Yes	Auto Dial - Configured using a Configuration Profile
Keymaps	Yes	Configured using Keymap Profiles
Language	Yes	Language - Configured using a Configuration Profile
Manual Forwarding	Yes	Accessible to the user via a softkey
Message Waiting Notification	Yes	N/A
Network Time Protocol (NTP)	Yes	NTP is supported but needs configuring in the Configuration Profile used by the 6900 handset (the setting is not inherited from the PBX). In Configuration Profiles set Time Server 1 to 3
Paging	Yes	Paging from 6900 handset works as normal. Paging to 6900 handsets only works from other 6900 handsets.
Record-A-Call	Partial	This feature does not currently work in conjunction with SIP devices. A softkey can be used to initiate a local conference to the Record-A-Call application which will successfully record the call but cannot be used to toggle the recording off.
Remote Feature Programming	Partial	A dedicated softkey has been provided for configuring the forward state of a remote extension, Other remote features programming is not available.
Station Monitor	Yes	N/A
Supervised Transfer	Yes	Transfer to connect is supported. Transfer to

		Hold/Ring are not supported.
System Forwarding	Yes	N/A
System Hold / Trunk Pickup	No	System hold is not available on 69xx phones and subsequently it is not possible to pickup trunk calls from another phone. Park keys should be used instead to move calls between users.
UCD Hunt Groups	Yes	N/A
Voicemail / Mailbox	Yes	N/A

 Where the '6900 Handset Equivalent' column is set to N/A, the feature works with standard MiVoice Office 250 configuration.

MiVoice Office 250 Configuration

When configuring the MiVoice Office 250 to support 6900 handsets, the following DB programming items should be checked.

DID/E&M Receive Busy Instead of Camp-On

If this setting is enabled, when transferring a call from a 6900 handset to an extension that is already busy, it will not be possible to complete the transfer.

Audio for Calls Camped-On/Ringing/On Hold

By default, this setting is configured to the MOH port for the system on each device created on the telephone system. To ensure that the 6900 handsets get audible feedback when ringing other extensions that are busy, ensure there is audio on the MOH port or change this setting to ringback for all devices.

OAI 3rd Party Call Control

When used on the MiVO 250, 69xx phones work in SIP mode and do not support the same feature set as their MiNET/Digital counterparts.

In addition to the unsupported features listed in the previous section, not all the available '3rd Party OAI Call Control' commands are supported in conjunction with 69xx phones.

Due to these 3rd Party OAI Call Control limitations, the following solutions for the MiVO 250 cannot be used in conjunction with 69xx phones:

- Attendant Console
- MiCC Office Client Go / Callviewer / Connection Assistant
- MiCC Office Intelligent Router / Reporter Pro / Realviewer (6900 phones can be reported on but cannot be used as the primary extension for a MiCC supervisor for call control, monitoring and recording)

 Attendant Console and CTI Clients such as Client Go, Callviewer and Connection Assistant will not operate with 69xx phones.

Mitel Phone Manager applications will continue to operate when paired with a 69xx phone but without the 'Merge' call feature.

The following OAI Commands can still be used are unaffected by or can be used in conjunction with SIP phones on the MiVO 250:

- Clear Call/Connection (_CC / _CX)
- Expanded Extension (_XE)
- Hunt Group Merge Out (_HM)
- Modify Call (_MD)
- Monitor Start/Stop (_MS / _MA / _MP)
- No Operation (_NO)
- Query Agent State (_QA)
- Query Device Information (_QI)
- Query DND (_QD)
- Query Event Filter (_QE)
- Query Forwarding (_QF)
- Query Hunt Group (_QH)
- Query List Extended (_QX)
- Query Message Waiting (_QM)
- Query Network Node Number (_QN)
- Query Night Mode (_QT)
- Query User Destination (_QUD)
- Query User Routing (_QUR)
- Set Agent State (_SA)
- Set Do-Not-Disturb (_SD)
- Set Forwarding (_SF)
- Set Message Waiting (_SM)
- Set Night Mode (_ST)
- Set User Destination/Information (_SUD / _SUI)
- Snapshot Device (_SP)
- Transfer Call One Step (_TO)



Commands not listed here DO NOT work with 69xx or SIP phones (e.g. Make Call (_MC), Hold Call (_HC), Retrieve Call (_RV)).

3 Mitel 6900 Handset Support Overview

The MiVoice Office Application Suite is the core configuration server for Mitel's premium 6900 handset range when used on the MiVoice Office 250 telecommunications platform.

The 6900 phones operate as a SIP phone, connecting directly with the telephone system (or via MiVoice Border Gateway for Teleworkers) for audio and call control features. The MCS server acts as configuration server for the phones, providing the following features:

Administrative Features:

- Firmware Updates
- Feature Configuration
- Keymaps
- Centralised Background Images & Screen Savers
- Page Zone Management

End-User Features

- SIP Hot Desking
- Server based feature keys including; UCD, ACD, DND, Forwarding etc
- Presence Profile visibility and control
- Alarm Notifications
- Speed Dial and Intercom Directory Access

Licensing

Each 6900 handset that uses the MiVoice Office Application Suite as a configuration server will consume a license. There are two levels of license available:

- Basic, provides access to all 6900 server based functionality apart from specific softkeys.
- Enhanced, required to access specific softkeys -> CLI Change & Recording Pause / Resume.

If there are no licenses available when a handset connects to the MCS server, a licensing message will be displayed on the screen.

Enhanced Licences

Each handset requires a 'Basic' licence in order to be provided with configuration and keymap updates from the MCS. An additional 'Enhanced' licence is required to access some of the softkey features. Any handset that is associated with a user that has been assigned a Phone Manager Professional or Team Leader licence will automatically request an 'Enhanced' licence.

6900 Features

The following sections provide information on configuring various aspects of the support for 6900 handsets:

1. [General Settings](#)
2. [Phones](#)
3. [Handset Firmware & Models](#)
4. [SIP Configuration](#)
5. [Keymap Profiles](#)
6. [Softkey Features](#)
7. [Softkey Feature Screens](#)
8. [Configuration Profiles](#)
9. [Configuration Options](#)

10. [Directories](#)
11. [Image Handling](#)
12. [Screen Savers](#)
13. [SIP Hot Desking](#)
14. [Page Zones](#)
15. [External Paging](#)

In addition, there are some engineering guidelines available for the 6900 on the following subjects:

- [Handset Rollouts](#)
- [Network Architecture & Resiliency](#)
- [SIP Hot Desking](#)
- [Teleworking with 6900 Handsets](#)
- [Multi-Node 6900 Installations](#)

3.1 6900 Handset General Settings

The following settings relate to some of the interaction between the MCS and any Mitel 6900 Handsets.

Setup PIN

The setup PIN is used to when rolling out 6900 handsets. It provides access to the setup dialog which allows an engineer to configure a newly installed 6900 handset and link it to a SIP device on the telephone system. The PIN can be configured to be any combination of keypad digits.

*Default: ****

Send DND Notifications

When 6900 handsets make calls to other extensions (MiNET, Digital, SIP or Analogue) that are in DND, they show a 'Busy' message on the screen and do not give the caller any information about the DND state.

MCS provides the DND (and Presence information if relevant) to the caller so that they can see why they are receiving a busy response to the call.

Default: Enabled

 The timeout on the Toaster style notification is not currently configurable.

Send Hunt Group Camp-on Notifications

When calls are queuing at ACD/UCD hunt groups, MCS can notify logged in members that there are calls waiting.

This is a system-wide feature currently and cannot be enabled/disabled on a per hunt group basis. This feature does not work on ACD Extension groups, only ACD Agent Groups.

Default: Enabled

 The timeout on the Toaster style notification is not currently configurable.

Show Chat Notification Preview

When enabled, a preview of the chat message received will be displayed in the notification toaster that displays on the 6900 phone.

Default: Disabled

 Chat notifications will appear on the phone even when it is locked.

Override Dial Plan

The MCS server will provide each 6900 handset with a dial plan based on the Dial Plan configured on the server. This dial plan is designed to allow the user to dial internal numbers, external numbers and service codes (including emergency numbers). The dial plan that is automatically sent to the handset can accommodate 3 & 4 digit dialling and will automatically work out the length of internal extensions based on the devices configured.

 If 3 digit dialling is required, ensure all internal extensions that begin with the same number are the same length. For example, for 3 digit dialling of extensions beginning 1, ensure all extensions beginning 1 are 3 digits long. This includes the built in modem extension.

In addition, the dial plan is designed so that the user does not need to enter the outgoing access code, it will automatically be entered for them when the dial plan detects an external number has been entered.

To manually assign a dial plan, enable the override and enter the dial plan required.

Default: Disabled

-  The outgoing access code will only be automatically entered if the external number being dialled starts with the toll digit. To dial numbers without the toll digit, the outgoing access code must still be entered.
-  The default Dial Plan for dialling any number is 'x+#{xx+*'
-  The Live Dialpad and SIP Dial Plan Terminator can be used to force the phone to evaluate the dial plan after every digit is entered and dial automatically when a match is made. Please refer to the [Configuration Options](#) section for more information.
-  Please refer to the 6900 Series Administration Guide for information on 'sip dial plans' and how to configure them.
-  By changing the Dial Plan passed to the phones, it is possible to prevent the handset from being able to make emergency calls. Ensure that all dial plan changes are thoroughly tested before rolling out.

Enable SRTP on Remote Handsets

When enabled, the MCS server will instruct any remote 6900 handsets to use SRTP for audio connections back to a MiVoice Border Gateway.

For more information on enabling SRTP, please refer to the [Manual Teleworker Provisioning](#).

-  If SRTP is not enabled on both the MBG configuration and the MCS configuration then calls will fail. The handsets will be able to make calls but not receive them.

Default: Disabled

Auto Detect Non-Internal Addresses as Remote

When enabled, the system will class any handset that does not have an IP address in the range 10.0.0.0/8, 172.16.0.0/12 or 192.168.0.0/16 as being a remote handset.

Handsets that are connecting through a MiVoice Border Gateway (MBG) will show up with internal addresses. To correctly identify these as remote, the internal address(es) of the MBG will need to be configured in the [Phone Systems](#) section.

Default: Enabled

Send Phone System Alarms to Admin Phones

When enabled, any phone system alarms will automatically be send to Mitel 6900 Handsets that have been configured as Administrator on the telephone system.

Default: Enabled

For more granular control over which alarms get sent to handsets, use the [Alarm Notification](#) feature.

Call Mode for Softkey Screens

This setting controls the behaviour of the phone if a call starts alerting when the user is in a softkey screen (changing presence or DND for example). If set to 'Notify via Toaster', the user will be left in the softkey screen but will be notified of the call via a toaster message. If set to 'Call Overrides Softkey', the softkey screen will be closed down and the call details screen will be displayed.

Default: Notify via Toaster

Device BLF Label Option

This settings controls the automatic labelling of BLF softkeys. The description or the username from the telephone system can be used as a BLF softkey label.

Default: Description

 If the device description contains a comma. It is assumed that the format is "last name, first name". This will automatically be converted to "first name last name" before being applied to a BLF label.

Display Voicemail Notification on User BLF Softkeys

When enabled, the number of new voicemails notifying a user will be displayed on any User BLF keys that are configured for the user on other phones.

Default: Disabled

Use HTTPS for Local Phone Softkeys

By default, all phones are configured to use HTTPS for softkey requests made to the MiVoice Office Application Suite server. This adds an overhead in processing time to each softkey press. Disabling HTTPS on softkey press can improve the user experience and reduce the processing overhead. This option only applies to local phones, not remote.

Default: False

 HTTPS will still be used for configuration updates if disabled for softkey requests.

Use Enhanced BLF Context Menu

When enabled, users are presented with contextual call control, queue and message options when they press a User or Extension BLF key.

Default: True

 This feature can be disabled on an extension by extension basis, please refer to the [Phones](#) section for more information.
For more information on the Enhanced BLF Context Menu, refer to the [6900 Softkey Feature Screens](#) section.

Leave a Station Message

If 'Use Enhanced BLF Context Menu' is enabled, this setting controls whether the option to leave a station message is displayed on the context menu.

Default: False

Enable Multicast DNS

With Multicast DNS enabled, the Auto Discovery feature can be used to enable out-of-the-box 6900 phones to automatically find the MiVoice Office Application Suite server without the need to configure any DHCP options.

Default: True

Call History Location

This setting controls whether the call history displayed on 6900 phones is provided by the MCS server or locally by the phone. If 'Server' is selected, a user based call history will be provided by the server to show calls for any of a user's extensions.

Default: Server

 A reboot of phones is required before a change to the call history settings will take effect.

Exclude Missed Hunt Group Calls From Call History

When excluded, only hunt group calls which are answered will appear in the call history, not missed calls.

Default: True

 This setting affects both 69xx and Phone Manager call history.

 On All-ring hunt groups (UCD & ACD), missed calls will only show in the first extension's call history.

Use the MCS as local NTP Server

When enabled, the MCS server will enable the Windows Time service on the local server as an NTP server. All local phones will then be provided the IP address of the MCS server as a backup time server (Configuration Time Server 3 is used for this).

Default: True

Directory Access

The settings below control the use of LDAP to deliver server based [Contact Directories](#) to 6900 phones. By default, users will automatically have access to the phone system directory and their own Phone Manager personal directory through the built in CSV directories of the phone.

In addition to the telephone number fields, the following server directory fields are passed to the phone:

Directory Field Name	Phone Field Name
Name	First Name
Field 1	Company
Field 2	Job Title
Field 3	Work Address Street
Field 4	Work Address City
Field 5	Work Address State/Province
Field 6	Work Address Zip/Postal Code
Field 7	Work Address Country
Field 8	Email 1
Field 9	Email 2
Field 10	Email 3

 The directories on phones will get updated once every 24 hours, new entries will not appear until the phone updates. Check the '[Auto Resync Time](#)' configuration for information on when this occurs.

Make Global Directories available to 6900 handsets

When enabled, 6900 phones will be given access to the global contact directories of the server using an LDAP query. The phones will download a copy of the directories on startup and will use them for contact matches on call or directory searches by the user. Use the Directory Access setting below to restrict which directory the phones have access to.

Default: Enabled

Directory Access for Remote Phones

Control how directories are made available to remote phones; No Access, CSV or LDAP. CSV is recommended for remote phone connecting over an unsecured network due to the way LDAP is implemented. Instead the CSV option can be used to merge the server contacts with the local user's contacts which are then provided in CSV format over a secure channel.

Default: None



The LDAP query used to provide server contacts to the phone is not secure. This interface should not be enabled for phones connecting over an unsecured network.

Directory Access (& Specified Directory)

This setting provides a way to select which directories are available. By default, all directories are available. When this mode is selected, the system will automatically filter the directories searched based on user permissions. If a specific directory is selected, all phones will be given access to it. Each phone is restricted to only accessing 5,000 directory items. By choosing a specific directory, the number of contacts the phones have access to can be restricted to ensure the 5,000 item limit is not breached.

Default: All Directories

Enable Asynchronous LDAP Queries

6900 phones can query the contact directories in one of two ways:

- Cached Download (default)
- Asynchronous Query

When in cached download mode, the phone will download all contacts from the server and make the local copy available to users when searching the directory or performing contact matches on telephone calls. This method provides the best user experience but is not suitable when there is a large number of contact records (specifically 5,000 records or more). When this threshold is reached, asynchronous queries should be enabled.

When asynchronous mode is enabled, users will still be able to search for contacts using but contact matches on telephone calls will no longer occur.

Default: False (cached mode)



If asynchronous mode is not enabled when there are large numbers of contact records, delays will be experienced by the user while the phone is performing contact match searches.

3.2 6900 Handset Phones

The 6900 Handset Phones section displays each physical handset that has been configured to use the MCS server as a configuration server.

If required, handsets can be added manually by pressing the 'New' button and filling out the required parameters. Otherwise, the MCS will automatically add a new handset to this section the first time it sees a request from it.

Handset Information/Configuration

The following is a list of information and configurable options for each physical handset:

Parameter	Description
Model Number	This is the model number of the handset. Models 6920, 6930, 6940 & 6970 are currently supported.
MAC Address	This hardware address for the handset. It is used by the MCS to uniquely identify the handset. Note: To swap out a handset with another, the MAC Address of the handset can be updated.
IP Address	This displays the current or last known IP Address of the handset. If the handset is remote (Teleworker), this will display the address of the internal address of the MiVoice Border Gateway.
Status	This displays the current status of the handset. Possible states are: • Uninitialised, the phone has not yet been mapped to a SIP extension. • Offline, the phone is not currently in contact with the MCS Server • Online (No SIP), the phone is successfully connected to the MCS Server but is not connected to the telephone system. • Online (Not Licensed), the phone is successfully communicating but there are no licences left on the MiVoice Office Application Suite to support it. • Online, the phone is operating normally. Both SIP and Configuration connections are online.
Firmware	This displays the current the version of firmware that the handset is currently running.
Extension	This shows the Extension number of the SIP device on the telephone system that this handset is mapped to.
Keymap	This displays the name of the keymap profile assigned to the phone (if any).
Configuration	This displays the name of the configuration profile assigned to the phone (if any)
Mode	This displays the mode the phone is configured to run in. Possible modes include:

	• Not Initialised, the phone has not yet been assigned a SIP extension • Fixed, the phone has a fixed SIP extension assign. • Mixed, the phone has a fixed SIP extension but also allows Hot Desking
Last seen	Indicates when the server last had communication with the phone.
Programmable Key Modules (PKMs)	Each 6900 handset can have up to three programmable key modules connected. If a handset has PKMs connected, they will be displayed here.
Remote	Indicates whether the phone is currently remote.
Diagnostics	Displays whether or not the handset has diagnostics enabled or not
Phone Diagnostics	Controls whether the Syslog based diagnostics are enabled on the 6900 handset itself.
Effective Keymap	Select this option to see the combined keymap that will be sent to the phone (System, Model, Phone & User

Editing a Handset

To edit the individual settings of handset, select the handset in the grid and then press the 'Edit' button or select 'Edit' from the right-click grid menu.

A form providing access to the following setting will be displayed:

Enable Hot Desking

This setting controls whether [SIP Hot Desk](#) users are allowed to log onto this phone or not. Once enabled, the system will ensure that the phone always has a Hot Desk softkey available on it's keymap.

Configuration & Keymap Selection

These settings optionally allow a specific Configuration or Keymap to be assigned to the phone. Remember, any keymap assigned will be merged with the System, Model and [User](#) keymaps.

Diagnostics

By default, diagnostics are disabled on all phones. Enabling diagnostics will cause the MiVoice Office Application Suite server to log traffic and diagnostic information for the phone on the server. When diagnostics is enabled, the option to set the phone's local diagnostic options is also enabled.

Server based logs are stored in the following location:

"c:\ProgramData\Mitel\Mitel Communication Service\Logs\Handset\logs\handsets\allhandsets\logs"

The phone's MAC address is included in the name of logs files associated with the phone.



Only enable the phone's local diagnostics when advised to do so by a Mitel engineer. Enabling a phone's local diagnostics can affect the performance of the phone.

Override Firmware

This setting provides the ability to override the default firmware for the phone's model with a new one. This is useful for testing new firmware versions before applying them to all phones or for apply individual patches to phones.

Enable BLF Context Menu

This setting provides the ability to disable the context menu on User and Extension BLF keys on specific extensions. This setting is only available if the 'Use Enhanced BLF Context Menu' has been enabled on the [General Settings](#) page.

Enable Auto Update NAT IP

For future use.

Rebooting Handsets

If a handset is not offline, the reboot button can be used to force it to restart. Restarting a handset is useful for forcing it to request new configuration files and firmware.



If any calls are in progress on the handset when it is rebooted, they will be cut off.

User Keymaps

In addition to any keymap assigned to a phone, each user has their own keymap which allows them to configure any softkeys that haven't been configured by a keymap profile.

An administrator can get access to view/configure a user's keymap by right-click on an extension in the 'Phones' grid and selecting the 'Edit User Keymap' option.

For more information, please refer to the [User Keymaps](#) section.

3.3 6900 Handset Models

The MCS server supports three different models of 6900 handset. The models supported are listed in the table below with a brief description of the difference in features:

Model No	Screen Size	Softkeys	Bluetooth
6920	320 x 240	• Softkeys - 18 • Top Softkeys - 20	No
6930	480 x 272	• Softkeys - 24 • Top Softkeys - 44	Yes
6940,6970	800 x 480	• Softkeys - 30 • Top Softkeys - 48	Yes

Configuring 6900 Handset Models

The Model configuration page shows a list of models support and provides an edit form to configure the following properties:

- Firmware
- Keymap
- Configuration

Firmware

When a handset requests new firmware, the MCS will check the model to see which firmware should be provided. This allows a different firmware to be run on different models of handset if required.

 Firmware can be overridden at handset level if required. Please refer to the [Firmware](#) section for more information.

Keymap

Optionally, each model can have a specific keymap applied. When a handset requests a keymap, the *[Default]*, model and phone assigned keymaps are applied. For more information, please refer to the [Keymap Profiles](#) section.

Configuration

Optionally, each model can have a specific configuration applied. When a handset requests configuration, the *[Default]*, model and phone assigned configurations are applied. For more information, please refer to

the [Configuration Profiles](#) section.

3.4 6900 Handset Firmware

The MCS server provides 6900 series handsets with firmware updates as part of its configuration server role. The firmware configuration page provides an interface to see which firmware files are currently loaded onto the system and provides a utility to upload firmware updates.

MCS can manage multiple versions of firmware at the same time. Having multiple versions of firmware allows new firmware to be evaluated on a subset of handsets and/or models before it is rolled out system wide.

The table on the 6900 Firmware configuration page shows the firmware currently loaded. The table shows the following information:

- Version, the version number of the firmware in the format 'x.x.x.x'
- Phone Types, a list of models which the system has this firmware version for

Managing Firmware

New Firmware

New firmware is automatically added to new releases of MiVoice Office Application Suite and are automatically installed when upgrading. If firmware is released outside of a MiVoice Office Application Suite release then it may be necessary to upload new firmware.

New firmware will be provided in the form of a ZIP file. The name of the ZIP will correspond to the version number of the firmware and there will be a firmware file inside the ZIP for each model of handset.



Please speak to your system maintainer about getting access to firmware updates.

To upload a new zip file, press the 'Upload Firmware' button. Press the 'Choose Files' button and browse for the new firmware ZIP file. When the upload has completed, the new firmware files will be copied to the following location:

"[ProgramData]\Mitel\Mitel Communication Service\Net Store\Firmware"

The table should automatically refresh and show the new firmware has been updated.

Deleting Firmware

When old firmware is no longer required, it can be deleted from the system. To do this, select a firmware from the table, then press the 'Delete' button. Firmware can only be deleted if it is not currently assigned to a model of handset.

Rolling Out Firmware

Firmware can be assigned at two levels:

- Model
- Phone

By default, all phones will be supplied with the firmware that matches the model number unless the firmware version has been overridden at phone level. Overriding firmware at phone level can be useful for testing a new release on one or a subset of handsets before rolling it out system wide. To override firmware at phone level, edit the required phone from the [6900 Phones](#) configuration page.

Handsets will request new firmware at the following times:

- During the start-up sequence
- Upon resync (if configured)

The resync process is controlled by the 'Auto Resync Mode' and 'Auto Resync Time' configuration options. If set, the handsets will request new configuration files and firmware at the configured time.

If auto resync is disabled, handsets will need to be rebooted to force them to request new firmware.



Firmware is provided over https to all handsets except in initial configuration when a 6900 handset with MiNET firmware requests and update over TFTP. For more information on the initial configuration and rollout of 6900 series handsets, please refer to the [6900 Engineering Guidelines](#).

3.5 6900 Handset Keymap Profiles

Keymap profiles are used to control the softkeys that are displayed on each handset. Once a key has been configured centrally via a keymap profile, it cannot be configured by the user.

Keymap profiles can be assigned to handsets in three different ways:

- Default Keymap, assigned to all handsets by default
- Model Keymap, assigned to all handsets of a specific model
- Phone Keymap, assigned directly to a phone or Hot Desk Device

When a handset is provided a keymap, the Default (system-wide) keymap and any model or phone keymaps are merged together. Keys set at model level will override any set at Default level and keys set at handset level will override those at model level.

Keymaps are not model specific, they can be assigned to any model of handset. This simplifies the process of having departmental keymaps for users with different model handsets. It also allows users to hot desk into any 6900 Handset and get their assigned keymap.

If there are more keys configured in the keymap than the handset allows, it will just show as many as is possible.



Any softkeys that have not been configured using a Keymap Profile can be configured by the user, all other softkeys will be read-only to them. Users can program softkeys themselves from the [User Keymaps](#) section.

SoftKey Types & Locations

Each handset has three different types of key available:

- Softkeys, Run along the bottom of the screen
- Top Softkeys, Display in the main area of the screen
- Expansion Module Softkeys, If fitted with one or more programmable key modules (PKMs), like top softkeys but do not support images.

The softkey features available can differ depending on location. For example, Line keys cannot be programmed as Softkeys.

Depending on the handset model, there will be a different number of Softkeys and Top Softkeys available. Please refer to the [6900 Handset Models](#) section for more information.

If more softkeys are configured than can fit on the handset's display, they will be paged out by the phone and one of the softkeys will become a screen-page button (not applicable to the 6940 & 6970 where the touch screen allows navigation between pages without a screen-page button).

By default, the handset will compress any configured softkeys so that any blank entries are removed. For example, if a Top Softkeys 3, 6 & 8 are programmed in a keymap. The handset will render these in locations 3,4,5 (Top Softkeys 1 & 2 are reserved for Line keys by the handset) so that no spaces are displayed. This behaviour can be changed if required. Please refer to the [Configuration Options](#) section for more information.

Editing Keymap Profiles

Each keymap profile has the following sections:

- General, Provide a description for the keymap to make it easily identifiable
- Softkeys, Configure any softkeys along the bottom of the handset
- Top Softkeys, Configure any softkeys in the main display area.
- PKM (1,2 & 3), Configure softkeys for any programmable key modules.

For each location tab (Softkeys, Top Softkeys, PKMs) there is a list of softkeys which are numerically ordered. To add a softkey, configure the following properties:

- Type, select the type of key to add to the handset
- Label, select the label that will display against the key (some labels are controlled by the software/handset)
- Value, this will be different depending on the key type

For information on the different softkey types available, please refer to the [Softkey Features](#) section.

 Use the 'Clone' button to quickly create a copy of a keymap profile that can then be edited rather than creating a new one from scratch.

 Ensure you are using the latest version of your browser to take advantage of keymap editing speed improvements.

Call States

For softkeys that run along the bottom of the phone's screen, a 'Call State' option can be selected. This controls when the softkey is displayed.

By default, when a softkey is added, it will always be displayed. If required, one or more of the following call states can be selected:

- Idle - The phone is not being used
- Connected - The current line is in an active call (or is on hold)
- Incoming - The phone is ringing
- Outgoing - The user is dialling a number, or the far-end is ringing
- Busy - The current line is busy because the line is in use.

For example, the 'Queue' softkey could be set to only display when the phone state is busy.

 The call states can be used in conjunction with the 'Collapsed Context User Softkey Screen' & 'Collapsed More Softkey Screen' configuration options to move softkey buttons onto the main screen when on a call.

Assigning Keymap Profiles

Once a keymap has been configured, it can be assigned to a Model or Phone/Hot Desk Device. To assign the keymap to a certain model, use the Models configuration area.

To apply a keymap to multiple phones and/or hot desk devices at the same time, press the 'Assign' button in the keymaps grid next to the keymap. A device selection form will then be loaded allowing multiple phones and/or hot desk devices to be assigned or removed from a keymap.

Any changes to a keymap or assignment will have an immediate effect on handsets that are online.

Enforced Softkeys

Some of the softkeys are essential to the operation of the handset. Where softkeys are essential, the MCS will automatically add them to a handset's keymap if they have not been added anywhere else.

Currently there are two types of key which are essential to handset operation:

Hot Desk Softkey

For handsets that allow hot desking or have hot desk devices currently logged in, the Hot Desk softkey is an essential key. If the Hot Desk key does not appear anywhere where it is required, one will automatically be added to softkey position 1. If there is another key programmed in this position, it will be overwritten.

Line 3 & 4 Softkeys

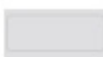
To control up to 4 calls on a 6900 handset from Phone Manager Desktop, softkeys for lines 1 to 4 are required. By default, all handsets are configured with softkeys for line 1 and 2. If there are no softkeys configured for lines 3 and 4, these will automatically be added to the last 2 topsoftkey positions available. If there are no topsoftkey locations available, the last two topsoftkeys will be overridden.

3.5.1 6900 Softkey Features

The following table lists all the different softkeys that are available to add to a 6900 handset. The features behind each softkey are either provided by the phone itself or by the MCS server.

For information about the user interaction with on screen dialogues associated to softkeys, please refer to the [Softkey Feature Screens](#) section.

 Phone based softkey features are processed locally on the phone, Server based softkey features are processed by the MCS server and will communicate with the server when pressed.

 Softkeys configured along the bottom of the phone show status with a blue outline   instead of the  /  displayed on the top softkeys.

Softkey Type	Feature Description	Softkey Examples	Phone or Server
Account Code	Used to set optional accounts on any external call that is in progress at the handset. Parameter: Enter an account code or leave blank to prompt the user when pressed. Enter a comma separated list to give the user a choice when pressed.	 No call, no account code or account code does not match call  Call in progress, the account code matches the parameter	Server
Account Code Following	Used to set account codes on all external calls made from the handset until the feature is toggled off. Parameter: Enter an account code or leave blank to prompt the user when pressed. Enter a comma separated list to give the user a choice when pressed.	 No account code following or account code following set does not match parameter  Account code will be associated with every valid call until disabled	Server
ACD End Wrap	If there is an Agent ID logged in to the handset, this softkey will allow the user to end wrap-up status early if required. If pressed when in the free state, users can place themselves in wrap-up manually. If placed in wrap-up manually, it must also be manually removed. No parameters.	 Not in Wrap-up state  Agent in Wrap-up state	Server
ACD Toggle	Provides support to log ACD agents into and out of the handset. Parameter: Enter a hunt group or comma separated list of hunt groups. Leave blank to provide the user a list of hunt groups the agent is a member of when pressed. Use * as the parameter to log in/out of all hunt groups without prompting the user.  If an Agent ID has been assigned to the MCS user the phone is associated with, it will be pre-populated into the Agent ID dialog when the user presses the ACD Toggle key.	 Logged Out  Logged In/Free  Busy  Wrap-up	Server

Agent Help	Allows the user to invoke the Agent Help feature on the telephone system. No parameters.	 Agent help not in progress  Agent help in progress	Server
BLF - Extension	Shows the status of an extension on the telephone system and provides one-click dialling. Parameter: Select an extension from the telephone system.	 Idle  Busy  Do-not-disturb  Wrap-up  Offline	Server
BLF - Hunt Group	Shows the status of a hunt group on the telephone system and provides one-click dialling. Parameter: Select a hunt group from the telephone system.	 Idle/Calls Ringing  Calls Queuing  No free agents  Offline	Server
BLF - Trunk	Shows the status of a trunk on the telephone system and provides one-click access. Parameter: Select a trunk from the telephone system.  when using the BLF - Trunk key to dial will not seize the trunk until the outgoing number has been dialled.	 Idle  Busy  Offline	Server
BLF - User	Shows the unified status of a Phone Manager User based on all their associated devices. User BLF softkeys will display a user's avatar image where possible, if not it will show the user's initials.  Maria Garcia  David Cole	 Idle  Busy  Wrap-up  Do-not-disturb	Server

	 Paul Clerk Parameter: Select a MCS user.	 If the ' Display Voicemail Notification on User BLF Softkeys ' settings is enabled, the number of unread voicemail messages the user has will be displayed in the top right of the icon.	
Call History	Provides access to the Call History page on the local handset. No parameters.	Refer to the 6900 User Guide/Quick Reference Guide.	Phone
Caller's List	Provides access to the inbound page of the phone's call history screen. No parameters.	Refer to the 6900 User Guide/Quick Reference Guide.	Phone
CLI Change	Provides the ability to change the calling party number programmed against the handset on the telephone system. Parameter: Enter a CLI or comma separated list of CLIs for the user to choose from when pressed. Leave blank to allow the user to type in the CLI manually.  This softkey type requires an enhanced 6900 license for the phone.  If using the CPN Substitution feature, changing the CLI via a softkey will only affect calls made via the handset.	☐ Caller ID on the phone does not match the parameter ☒ Caller ID on the phone matches the parameter	Server
Conference	Start a conference using the built in features of the handset. No parameters.	Refer to the 6900 User Guide/Quick Reference Guide.	Phone
DEE On/Off	Toggles the Dynamic Extension Express feature of the extension in the telephone system No parameters.	 Dynamic Extension Express is disabled  Dynamic Extension Express is enabled	Server
Directory	Provides access to the built in directory features of the handset. This includes accessing the System Speed Dials & Intercom directory from the telephone	Refer to the 6900 User Guide/Quick Reference Guide.	Phone

	system. No parameters.		
Discreet Ringing	Enables discreet ringing on the local handset. No parameters.	Refer to the 6900 User Guide/Quick Reference Guide.	Phone
Do-not-disturb	Shows status and provides access to control the DND status of the handset. Parameters: Select a DND message from the list or let the user select when pressed. If no DND Text is provided, the user will be prompted when pressed.  If enabled, it is advised that DND is controlled using Presence Profiles.	 Do-not-disturb is disabled  Do-not-disturb is enabled	Server
Door Relay	Activate the door relay on the telephone system. No parameters.	☐ No status displayed	Server
Empty	Programs an empty key on the keymap. This is useful when the configuration option for collapsing the keys is enabled. No parameters.	Refer to the 6900 User Guide/Quick Reference Guide.	Phone
Feature Code	Provides the ability to enter supported feature codes on the telephone system Parameter: Select the feature code to apply when the key is pressed. Only a limited number of feature codes are supported at this time.  Please refer to the MiVoice Office 250 Features & Programming Guide for a supported list.	☐ No status displayed	Server
Flash	Provides access to invoke a flash on an active SIP call. No parameters.	Refer to the 6900 User Guide/Quick Reference Guide.	Phone
Forward	Provides control of manual forwarding on the telephone system Parameter: Select a manual forward type or let the user select when pressed. If no destination is provided, the user will be prompted when pressed.	 or  Manual forward is disabled or does not match the softkey parameters  or  Manual forward is	Server

		enabled and matches the softkey parameters	
Group Mailbox	Shows the status of a mailbox on the telephone system. Parameter: Select a valid mailbox on the telephone system  The mailbox must be configured with a notification station  Available from release 5.1.13	 No messages  Messages waiting (Button will show yellow on non-touch screen phones)	Server
Group Missed Calls	Shows the last 10 missed calls for a hunt group on the system Parameter: Enter a valid hunt group on the telephone system	 No missed call alerts (there may be calls that have not been redialled)  Missed calls alerting (Button will show yellow on non-touch screen phones)	Server
Group Pickup	Shows the status of a hunt group on the telephone system and provides one-click pickup. Parameter: Select a hunt group from the telephone system.	 Idle (When LED flashes, calls are ringing at the group)  Calls Queuing  No agents free  Offline	Server
Hand Off/Pull	When using Dynamic Extension Express (DEE), this softkey can be used to push active calls from an internal extension and pull back calls currently active on an external DEE device. Calls that have been 'Pushed/Pulled' will divert back to alert the primary DEE device so that they can be answered again. No parameters.  This feature requires the handset to be associated to an MCS user, either as a primary, secondary or DEE device. 'No User' will appear on the label of the button if the phone is not associated to a user.	 Feature Inactive  Call is available to push/pull to the primary DEE extension	Server

	 Available from release 5.1.13		
Handsfree	Provides access to toggle on/off the handsfree for intercom calls (called SIP Allow Auto Answer in the phone configuration). No parameters.  Handsfree Intercom calls will not work if 'DEE On/Off' is in the 'On' state.  Handsfree Intercom calls will not work if the SIP device was created on the PBX as a generic SIP device - only if it was created as a 69xx SIP Device  Available from release 5.1.15	 Handsfree Disabled  Handsfree Enabled	Server
Hold	Provides access to put the call on hold. This is advised to be added to the keymap used by any 6970 as it has no physical Hold hardkey No parameters.	 Hold	Phone
Hot Desk	Provides access to SIP hot desking features and the ability to log into or log out off a handset. No parameters.	 No hot desk user logged in  Hot desk user is logged in	Server
Line	Displays call activity on the handset and provides outgoing access. Parameter: Enter a line number from 1 to 24.	Refer to the 6900 User Guide/Quick Reference Guide.	Phone
Mobile	Displays mobile call activity for any mobile phone connected via Bluetooth to the handset. No parameters.	Refer to the 6900 User Guide/Quick Reference Guide.	Phone
Night Mode	Toggle night mode on or off on the selected node(s). Parameter: Select a telephone system node from the list or select 'All Nodes'.	 or  Night mode is off for the configured node  or  Night mode is on for the configured node  If 'All Nodes' has been selected, the Night mode softkey will only	Server

		display 'On' if all nodes are in night mode. If only some of the nodes are in night mode, it will display 'Off'.	
Outgoing Access	Dials the outgoing access digit on the keypad to initiate an outgoing call. Parameter: Leave blank to dial the outgoing access digit or enter the number of a trunk/trunk group number.  When using the Outgoing key to dial it will not seize the trunk until the outgoing number has been dialled.	☐ No status displayed	Server
Paging (PBX)	Provides access to the page zones on the telephone system to page non-6900 handsets. Parameter: Select a specific page zone ID or 'User Choice' to allow page zone select when the softkey is pressed.	☐ No status displayed	Server
Paging (Phone)	Provides access to the SIP paging features of the handset to page other 6900 handsets. Parameter: Select a SIP Page Zone from the list or select 'User Choice' to allow page zone selection when the softkey is pressed	☐ No status displayed	Phone
Park/Pickup	Park or pickup calls from designated hunt groups or phantoms on the telephone system. Parameter: Enter the number of a hunt group or phantom on the telephone system.  If using a hunt group, ensure it has members configured and has camp-ons enabled.  The 'Park Recall' timer does not work for calls parked at a phantom using a 6900 phone. Use the 'Transfer Available Timer', 'Transfer Attendant Timer' or 'Transfer Vice Processor Timer' instead. Calls parked at a hunt group will follow the hunt	 No call parked  Call parked	Server

	group's 'Recall Timer' & 'Recall Destination'.		
Phone Lock	Lock or unlock the phone. To unlock the phone, the user will need to know the PIN that has been locally configured. No parameters.	Refer to the 6900 User Guide/Quick Reference Guide.	Phone
Presence Profile	Shows the current active profile for the user and provides the ability to switch between profiles. No parameters.  This feature requires the handset to be associated to an MCS user, either as a primary, secondary or DEE device. 'No User' will appear on the label of the button if the phone is not associated to a user.	Label will display current profile selected.	Server
Queue	Toggle queue requests on and off. Queue requests can be requested when dialling someone. Once in place they can be cancelled at any time. No parameters.	 No queue requested  Queue requested	Server
Record-A-Call	This softkey can be used to Record-A-Call feature of the telephone system by invoking a local conference on the phone. Parameter: Select the Record A Call application to use.  An associated mailbox is required for this feature to work. The recorded part of the call will appear as a second call at the extension in both Phone Manager and any call reports.  This feature cannot be used as a MiVoice Office Call Recorder method.	No status is shown. To disable the recording, the user must hang up the conference leg to the Record-A-Call application.	Phone
Recording Pause / Resume	Pause or resume an a call recording that is in progress on a MiVoice Office Call Recorder or linked Xarios Call Recorder. No parameters.	 No call  Active call, not recorded  Active call, recording paused	Server

	 This softkey type requires an enhanced 6900 license on the MCS server.	 Active call, recorded	
Redial	Provides access to redial functionality. This is mainly for use on 6970 that has no hardkey for redial No Parameters	 Redial	Phone
Remote Forward	Provides remote control of manual forwarding of a different extension on the telephone system Parameter: Select the extension on which to control the manual forwarding. Select a manual forward type or let the user select when pressed. If no destination is provided, the user will be prompted when pressed.  To use this softkey, the extension must be an Administrator on the telephone system.	 or  Manual forward on remote extension is disabled or does not match the softkey parameters  or  Manual forward on remote extension is enabled and matches the softkey parameters	Server
Reverse Transfer	When pressed, this softkey dials the 'Reverse Transfer' feature code. The user can then enter an extension number to pickup from or press a configured BLF key. No parameters.	 No status displayed	Server
Ring Intercom Always	Toggle the ring intercom always feature on the telephone system for the handset. When 'Off', any internal calls will go through handsfree if the dialled extension has the 'Handsfree' feature enabled. No parameters.  Available from release 5.1.15  Requires MiVoice Office 250 6.3 SP2 or Higher	 Outgoing internal calls will be handsfree if the dialled extension supports it.  Outgoing internal calls will always ring.	Server
Secondary Extension	Provides status and secondary extension audible alerts for an extension or hunt group on the telephone system. Parameter: Enter an extension or hunt group	See 'BLF - Extension' or 'BLF - Hunt Group' for softkey status information.	Server

	Ring When: Set the threshold for the number of calls ringing/queuing which will cause the Secondary Extension to generate an alert. If set to 0 the Secondary Extension softkey will never generate an alert.  Available from release 5.1.13		
Station Message	Provides the ability to leave a station message with another extension when they are busy. The button can be pressed on an outgoing call to another extension. No parameters  Available from release 5.2.12	 No status displayed	Server
Station Monitor	Provides the ability to initiate a station monitor session of another phone. If a station monitor is in progress, this key can also be used to perform barge-in and steal operations. Parameter: Enter an extension to monitor	 Not currently station monitoring  station monitor is in progress	Server
Speed Dial	Dial a number pre-programmed on a button. Parameter: Enter the number to be dialled by the handset when the key is pressed. A # can be used to indicate the end of the number to dial, commas and digits can then be used to dial DTMF on the call after it has connected. For example, 1300#,,1000# could be used to dial the conference bridge on extension 1300, pause and then enter 1000# as DTMF to log directly into a specific conference.	Refer to the 6900 User Guide/Quick Reference Guide.	Phone
Speed Dial Conference	When on a call, conference in another number directly. Parameter: Enter the number to be dialled by the handset when the key is pressed.	Refer to the 6900 User Guide/Quick Reference Guide.	Phone
Speed Dial Transfer	Transfer a call straight to another number Parameter: Enter the number to be dialled by the handset when the key is	Refer to the 6900 User Guide/Quick Reference Guide.	Phone

	pressed.		
System Speed Dial	Dials a speed dial bin configured on the telephone system. Parameter: Enter a speed dial bin number or leave blank to let the user choose when pressed.  This softkey can also be used to deflect ringing calls to a specific destination. For example, Send to Voicemail can be implemented using this key and entering the voicemail application as the parameter.	☐ No status displayed	Server
Transfer	Places a local call on hold to begin a transfer. No parameters.	Refer to the 6900 User Guide/Quick Reference Guide.	Phone
Transfer Immediate	Transfers a currently active call at the extension to a specific device on the telephone system. Parameter: Enter a target device or leave blank to be prompted for a device when pressed.  Available from release 5.2.12	☐ No status displayed	Server
Transfer to Voicemail	Transfers a currently active call at the extension to a specific mailbox on the telephone system. Parameter: Enter a target mailbox or leave blank to be prompted for a mailbox when pressed.  Available from release 5.1.13	☐ No status displayed	Server
UCD	Toggle the handset's availability in any UCD hunt groups on the telephone system. No parameters.	 UCD calls disabled  UCD calls enabled	Server
Voicemail	Provides access to the Voicemail. This is mainly for use on a 6970 that does not have a hardkey for voicemail No parameters.	 Voicemail	Server

 Any softkey implemented by the phone will continue to operate if the link to the MCS server is down. Any softkey implemented by the server will only work when the MCS server is online.

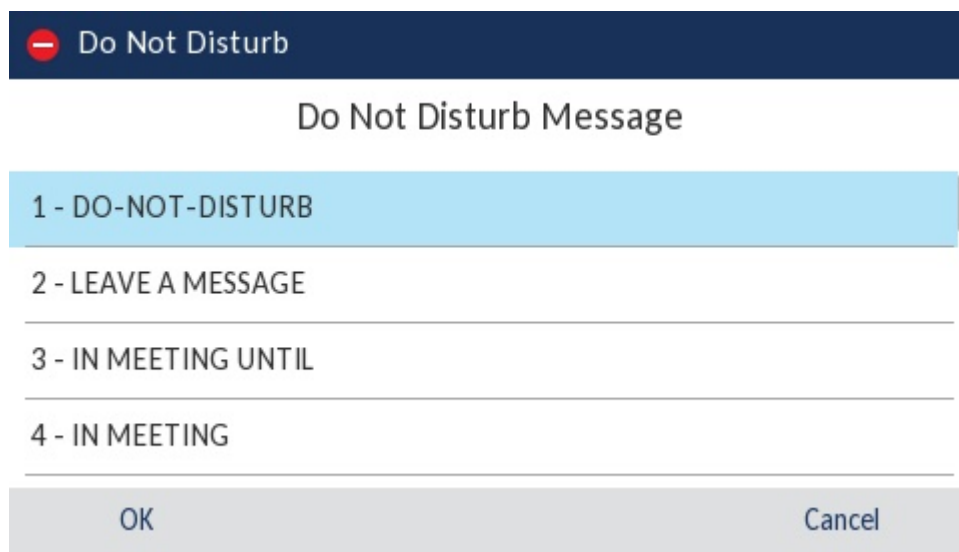
3.5.2 6900 Softkey Feature Screens

The following section outlines the behaviour of softkeys which present screens to the user. All softkeys listed in this section interact with the user through a series of screens and responses. If a softkey is not listed here, it does not require additional information from the user to perform its action.

Softkey Selection Screens

Many of the softkeys available provide a 'Selection Screen' as part of the feature interaction. When a selection screen is displayed, one of the available options must be selected.

The image below shows an example of the do-not-disturb selection screen:



There are multiple ways in which select the required option:

- Use the D-Pad/Cursor to highlight and select the required option
- Press a keypad digit corresponding with the required option (e.g. pressing '2' on the key pad would select 'Leave a message')
- Use the touch screen (6940, 6970 models) to scroll/select the required option.

When using the keypad to select the required option, 0 selects the 10th item in the list. Any options beyond 10 cannot be selected in this manner and one of the other selection methods must be used.

Selection screens are presented on the following softkey screens:

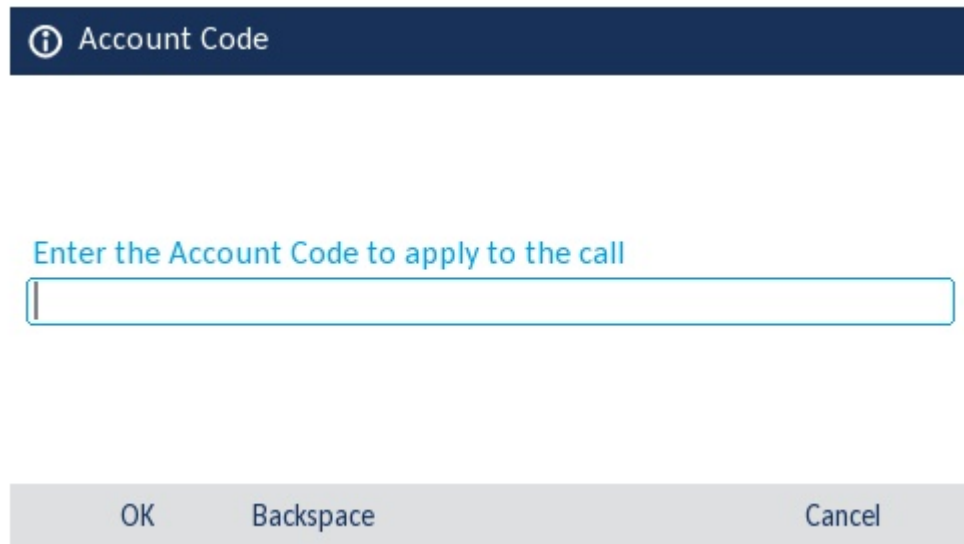
- [Account Code/Account Code Following](#) (Multiple Parameter Selection)
- [ACD Toggle](#) (Hunt Group Selection)
- [BLF Softkeys](#)
- [Caller ID Change](#)
- [Do-Not-Disturb](#)
- [Forwarding](#)
- [Paging](#) (Page Zone selection)
- [Presence](#)
- [Silent Monitor](#)
- [System Speed Dial](#)
- [Voicemail](#)

Account Code

The 'Account Code' softkey can be used to enter an account code on a call which is in progress at the phone.

No Code

If no codes have been provided in the softkey's parameter, the following dialogue will display prompting for an account code to be entered:



Codes of up to 12 digits in length are supported*.

 * Check with your administrator

Single Code

If a single code has been added as a parameter to the softkey, no dialogue will show when the key is pressed. The code defined in the parameter will be immediately applied to the call.

Multiple Codes

Multiple codes can be added to the softkey's parameter by using a comma separated list (e.g. 5,12,100). When multiple codes are configured against a softkey, the following dialogue will be displayed when the key is pressed, prompting for one to be selected:

 Account Code Following

Account Code Selection

1 - 1111

2 - 2222

OK

Cancel

To select and apply a code, use the '>' chevron or select a code then press the 'OK' button.

Account Code Following

The 'Account Code Following' softkey can be used to enter an account code on all the following calls made from the phone. The sections below outline the user interaction when pressing an account code following softkey when in the 'Off' position. When 'On', pressing the key will turn off the account code following feature for the phone.

No Code

If no codes have been provided in the softkey's parameter, the following dialogue will display prompting for an account code to be entered:

 Account Code Following

Enter the Account Code you wish to use

OK

Backspace

Cancel

Codes of up to 12 digits in length are supported.

Single Code

If a single code has been added as a parameter to the softkey, no dialogue will show when the key is pressed. The code defined in the parameter will be immediately applied to the call.

Multiple Codes

Multiple codes can be added to the softkey's parameter by using a comma separated list (e.g. 5,12,100). When multiple codes are configured against a softkey, the following dialogue will be displayed when the key is pressed, prompting for one to be selected:

Account Code Following

Account Code Selection

1 - 1111

2 - 2222

OK Cancel

To select and apply a code, use the '>' chevron or select a code then press the 'OK' button.

ACD Toggle

The 'ACD Toggle' softkey can be used to log an ACD agent in or out of the phone. The sections below outline the user interaction when pressing the key in the 'logged out' state. Pressing the softkey when logged in will result in an immediate logout of the agent ID.

Agent ID Prompt

When the ACD toggle softkey is pressed, the dialogue below will appear prompting for an Agent ID to be entered. If the user associated with the phone has an agent ID assigned then the dialogue will be pre-populated with that agent ID to save the user time.

ACD Login

Agent ID

Please enter the Agent ID you wish to log in with

OK
Backspace
Cancel

No Hunt Group

If no hunt group is configured against the parameter of the softkey, the following dialogue will appear prompting the user to select which hunt group(s) to log into (the system will query which hunt groups the provided agent ID is a member of and will only display these):

ACD Login

Hunt Groups

1 - Ann_OF (24005) (Selected)

2 - Ann_OF_R (24006)

3 - HG 24050 (24050)

4 - No Message (24003)

Login
Cancel

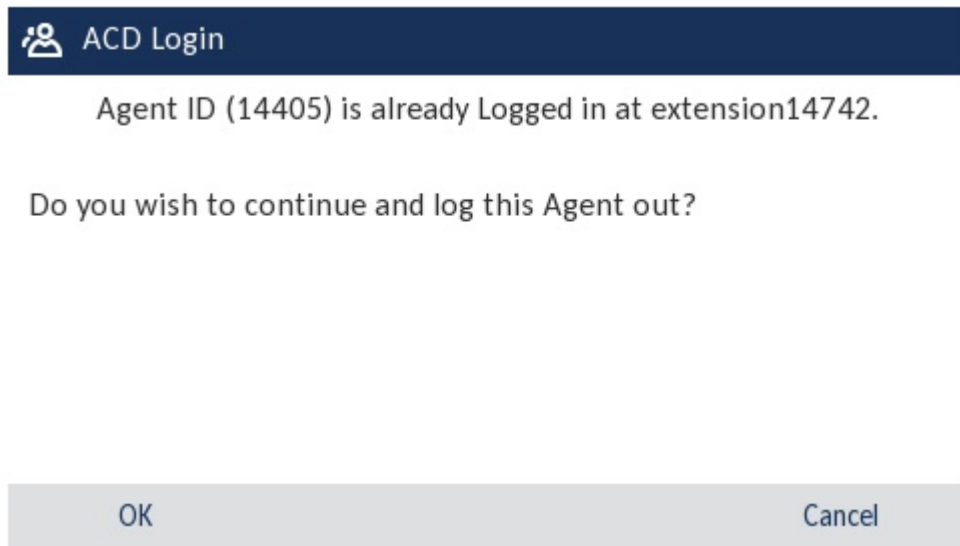
Pressing the 'Login All' button will log the agent into all hunt groups listed. To select specific hunt groups to log into, press the '>' chevron next to each group to log into then press the 'Login' button.

One or More Hunt Groups

If one or more hunt groups have been configured against the softkey's parameter (e.g. 2000, 2001, 2002), when the key is pressed the hunt groups selection dialogue will not be displayed. Instead, the agent will be immediately logged into or out of the configured hunt groups.

Forced Logout

If when logging in the agent ID provided is already logged into another phone, the dialogue below will be displayed prompting whether to continue and force a logout of the agent from the other extension first before completing the login operation.



BLF Softkeys

By default, when pressing a BLF softkey (Extension, Hunt Group or User, not Trunk) a call will be made to the target device. If however the local extension is on a call or the target device is not idle (in do-not-disturb, on a call or has a call on hold/ringing) a context sensitive selection screen will display offering various options. The options offer will depend on the status of the local extension as well as the status of the extension/user/trunk of the BLF softkey.

 The 'Enhanced Context BLF' softkey screen can be disabled if required. When disabled, pressing a BLF softkey will result in a call being immediately dialled to the target device without the BLF screen showing.

The image below shows an example of the BLF softkey screen. The target device's avatar is displayed top left along with the device's description and current status.



1001 Sandra Walker - Idle

Options



1 - Dial



2 - Transfer



3 - Transfer Immediate



4 - Transfer to Voicemail



5 - Conference



6 - Cancel Message

OK

Cancel

Icon	Action
	Dial - Make a call to the target device. If the local extension is on a call, this call will be put on hold.
	Transfer Announced - Setup a announced transfer to the target device.
	Transfer Immediate - Send the current call immediately to the target device with no announcement.
	Transfer to Voicemail - Send the current call immediately to the target devices mailbox*.
	Conference - Initiate a conference with the target device. Note: The current call will be put on hold, to complete the conference the 'Conf' button will need to be pressed again after the target device has answered.
	Pickup - Attempts to pick up (reverse transfer) a call that is on hold or ringing at the target device. Detail of the call will be provided if available.
	Queue - Book a queue with the target extension. When the target extension becomes free, the Queue screen will be show prompting a call. Note: If a queue is already booked with the target extension, an option to cancel the queue will be provided.
	Leave a Station Message - Leave a station message with the target extension. Note: If a station message is already in place with the target extension, the option to cancel the message will be provided.
	Leave a Voicemail - Leave a voicemail message with the target extension*.

 * When transferring calls to off node mailboxes, the off node Voicemail application must have been added to the transferring phone's node. There is not import/export option for this, is must be added manually.

CLI Change

The 'CLI Change' softkey can be used to change the caller ID presented from a phone when making external calls. The sections below outline the user interaction when pressing the key when it is not lit (matching caller ID is not currently applied). If pressed when the softkey is lit, the caller ID configured will be removed from the phone and the default caller ID for the trunk will be used.

 This softkey type requires an enhanced 6900 license for the phone.

 If using the [CPN Substitution](#) feature, changing the CLI via a softkey will only affect calls made via the handset.

No Caller ID

If no caller ID is configured against the softkey's parameter, the following dialogue will display prompting for one to be entered:

One Caller ID

If a single caller ID is configured against the softkey's parameter, no dialogue will be displayed and the caller ID will immediately be applied to the phone.

Multiple Caller IDs

If multiple caller IDs are configured against the softkey's parameter (e.g. 01234567890,12345), the following dialogue will be displayed prompting for one to be selected:

To select and apply a code, use the '>' chevron or select a code then press the 'OK' button.

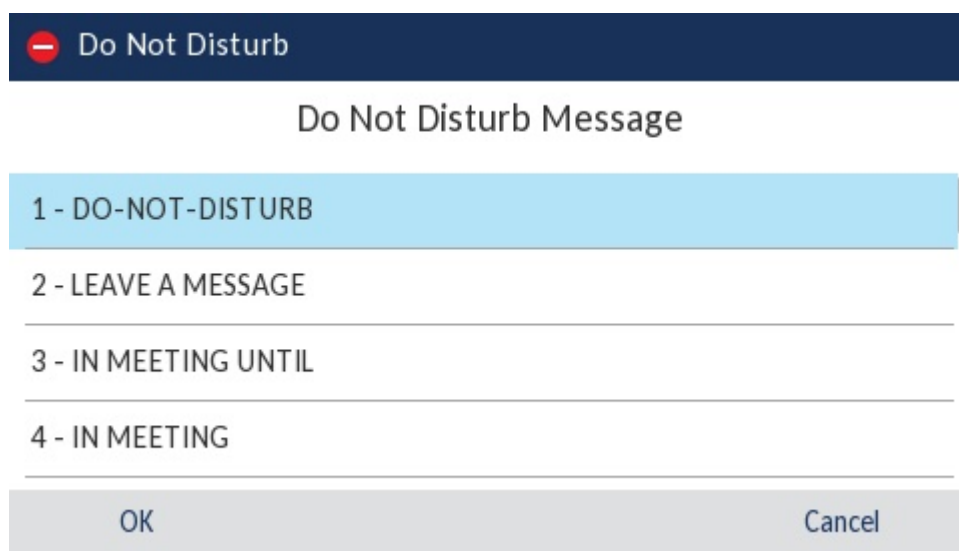
Do Not Disturb

The 'Do Not Disturb' softkey can be used to toggle the phone in and out of do-not-disturb. The sections below outline the user interaction when the softkey is pressed when not in the do-not-disturb state. If in the do-not-disturb state, pressing the softkey will toggle do-not-disturb off.

 When using Presence Profiles, the Do Not Disturb key is not required. DND should be enabled using the appropriate profile.

User Choice

If the DND message parameter against the softkey has been set to 'User Choice', the following dialogue will appear when the softkey is pressed:



Do Not Disturb

Do Not Disturb Message

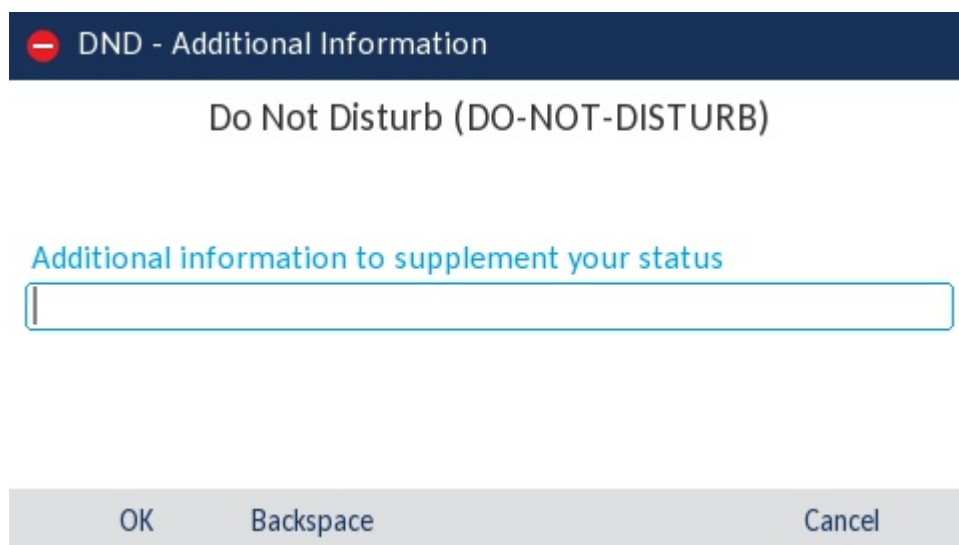
- 1 - DO-NOT-DISTURB
- 2 - LEAVE A MESSAGE
- 3 - IN MEETING UNTIL
- 4 - IN MEETING

OK Cancel

To enable do-not-disturb, select the required message from the list provided by pressing the '>' chevron or selecting a message and pressing 'OK'. The message selected will be visible to other user's when calling and on reporting software. Once a message has been selected, the 'Additional Information' dialogue will be presented as outlined below.

DND Text Empty

If the 'DND Text' parameter has been configured against the softkey, the do-not-disturb state will immediately be applied to the phone once a DND Message has been selected. If the 'DND Text' parameter has not been configured, the following dialogue will appear prompting for up to 12 characters of additional information (e.g. In a Meeting - Until 4pm).



DND - Additional Information

Do Not Disturb (DO-NOT-DISTURB)

Additional information to supplement your status

OK Backspace Cancel

DND Message & DND Text Provided

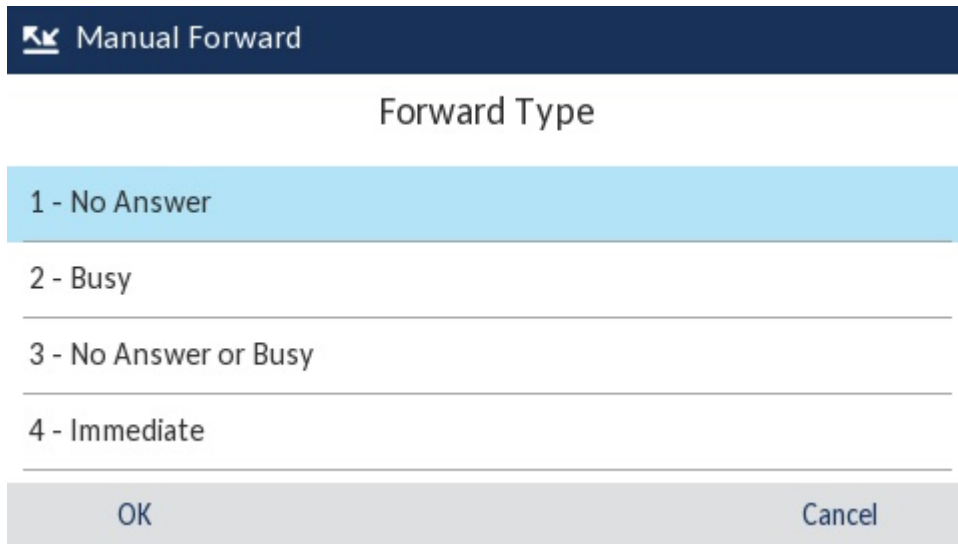
If both the 'DND Message' and 'DND Text' parameters have been configured against the softkey, pressing the softkey will immediately apply the do-not-disturb state with no further user interaction required.

Forward

The forward softkey can be used to apply or remove a manual forward to send calls to an alternate location (internal or external). If the softkey is pressed while a manual forward is in place on the phone, the manual forward will be removed. The sections below outline the user interaction if the key is pressed when no manual forward is in place on the phone.

User Choice

If 'User Choice' is selected as the forward type parameter, the following dialogue will be displayed prompting for the forward type to be selected:



Manual Forward

Forward Type

1 - No Answer

2 - Busy

3 - No Answer or Busy

4 - Immediate

OK Cancel

The forward type can be selected by pressing the '>' chevron or by selecting a type then pressing the 'OK' button. Once the forward type has been selected, the destination dialogue will then be displayed.

Forward Destination

If the 'Destination' parameter has not been configured against the softkey, the following dialogue will appear prompting for the destination to be entered:

Manual Forward - Destination

Enter the destination to forward calls to

OK

Backspace

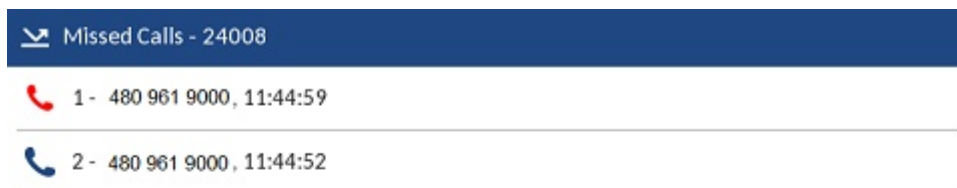
Cancel

An internal extension number or external phone number can be entered (there is no need to enter the Outgoing prefix, just enter the external number).

Once the destination has been correctly entered, pressing the 'OK' button will apply the manual forward to the phone.

Group Missed Calls

The 'Group Missed Calls' softkey is used to display the last 10 missed calls for a specific hunt group.



Cancel

Clear Redials

Pressing the number keypad digit corresponding to a missed call or selecting the missed call using the cursor or touch screen will initiate a redial of the missed call.

Calls which someone has attempted to redial (using the Group Missed Calls softkey only) will be shown with a blue phone icon. The 'Clear Redials' option will remove the red icon against all missed call entries.

 Missed calls will only show if they are abandoned at the hunt group. If they recall to another destination they will not be classed as missed calls on the hunt group.

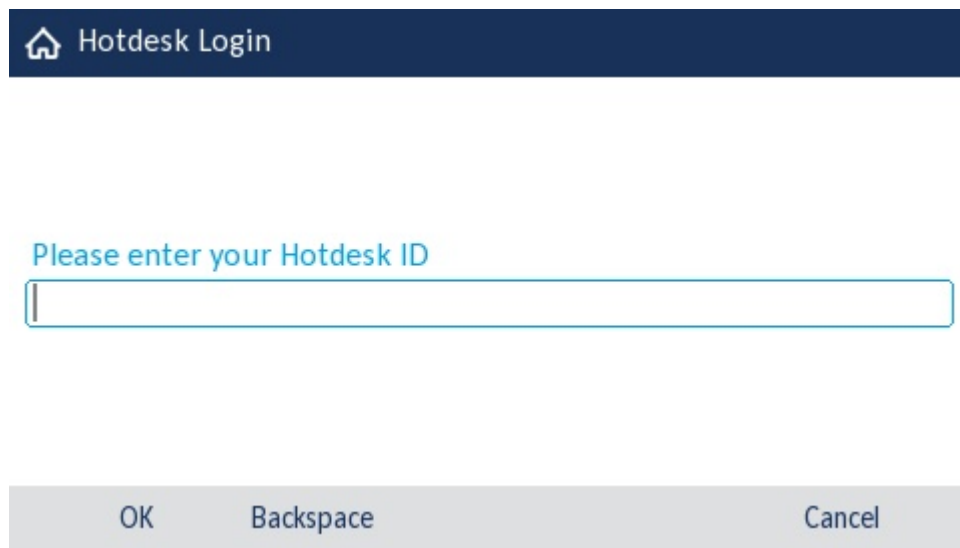
 Missed call details are not stored persistently and will be lost if the CTI Host Service is restarted.

Hot Desk

The 'Hot Desk' softkey is used to log in or out a SIP Hot Desk from the phone. The sections below outline the user interaction when pressing the hot desk softkey.

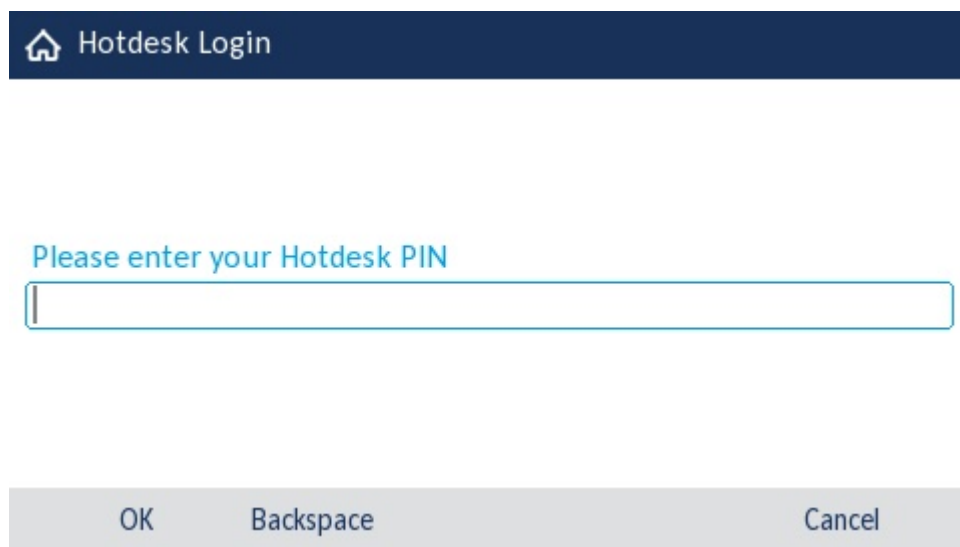
Logging In

Pressing the hot desk softkey when logged out will start the log in process. The dialogue below will be displayed prompting for a Hot Desk ID to be entered:



The dialog box has a dark blue header bar with a white house icon and the text 'Hotdesk Login'. Below the header is a light blue prompt 'Please enter your Hotdesk ID'. Underneath the prompt is a white text input field with a blue border. At the bottom of the dialog is a light gray bar containing three buttons: 'OK', 'Backspace', and 'Cancel'.

Once a Hot Desk ID has been entered, pressing the 'OK' button will display the Hot Desk PIN request dialogue:



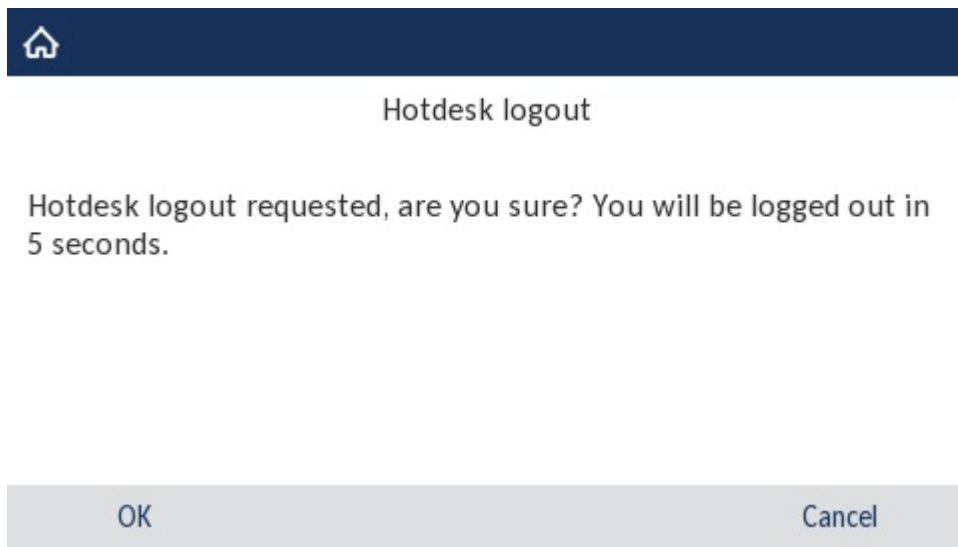
The dialog box has a dark blue header bar with a white house icon and the text 'Hotdesk Login'. Below the header is a light blue prompt 'Please enter your Hotdesk PIN'. Underneath the prompt is a white text input field with a blue border. At the bottom of the dialog is a light gray bar containing three buttons: 'OK', 'Backspace', and 'Cancel'.

Once the Hot Desk PIN has been entered, press the 'OK' button to complete the log in.

Logging Out

Pressing the hot desk softkey when logged in will start the logout process. The dialogue below will display prompting the user to confirm that a log out is required, this is to handle scenarios where the softkey is accidentally pressed. Pressing the 'Cancel' button within 5 seconds will stop the logout process from

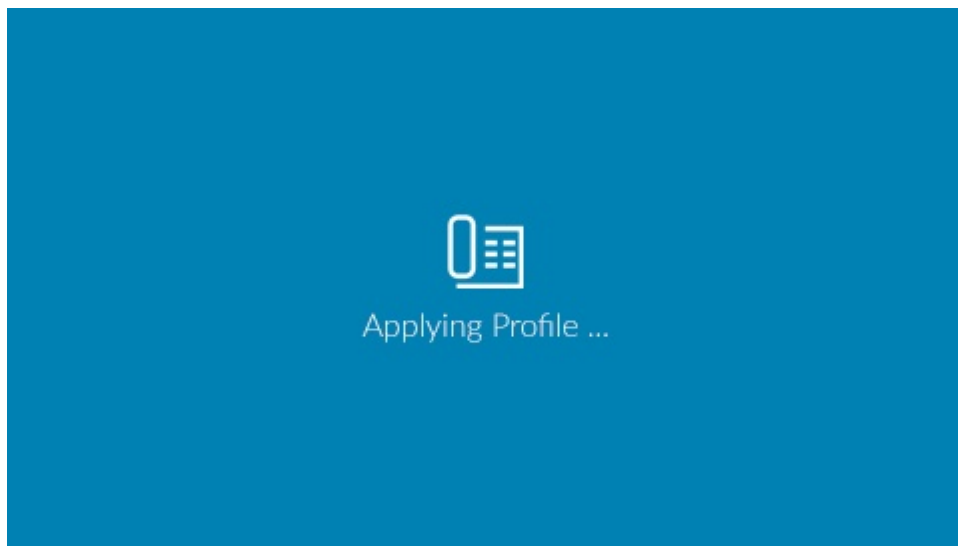
continuing.



If the 'OK' button is pressed or the 5 second timer expires, the logout process will be initiated.

Applying Profile

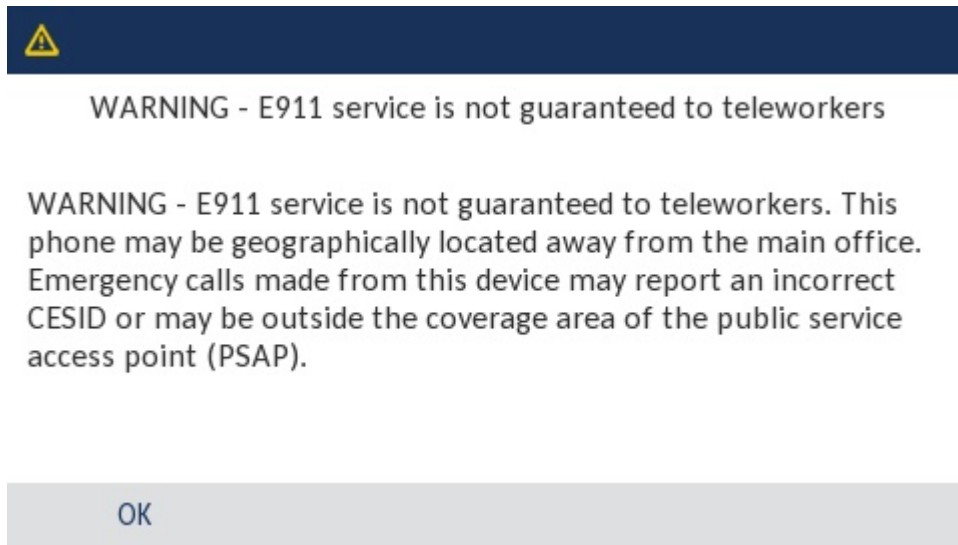
Whenever a hot desk login or logout operation is in progress, the 'Applying Profile' screen will display on the phone. The screen will display while the softkeys for the hot desk device are being added or removed. For remote phones, the softkey configuration can take longer and the applying profile screen will display for a longer period of time.



E911 Notification

The E911 screen is displayed every time a remote user hot desks into a phone.

 The E911 screen also appears the first time a phone connects to the MiVoice Office Application and on an ongoing monthly basis for remote phones.



Before the phone can be used, the content of the warning must be read and accepted by the user by pressing the 'OK' button.

Paging (PBX)

The Paging (PBX) softkey provides the user access to the page zones configured on the telephone system for paging 5300 and/or digital phones (not 6900 phones). The sections below outline the user interaction when the paging (PBX) softkey is pressed.

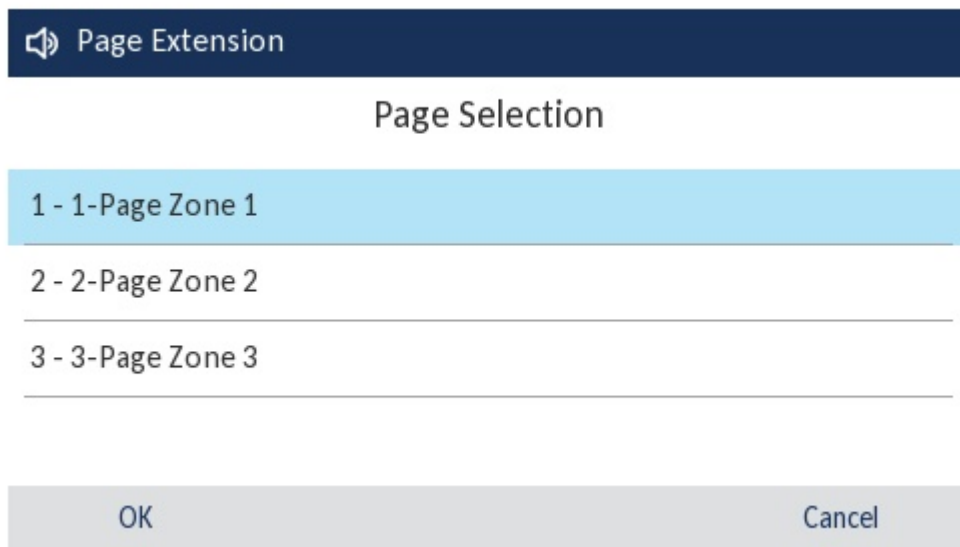
 To page other 6900 phones, use the Page (Phone) softkey.

Specific Page Zone

If a specific page zone ID has been selected in the parameters for the softkey, the page zone will be dialled when the softkey is pressed with no further interaction from the user required.

User Choice

If no specific page zone ID has been selected in the parameters for the softkey, the following dialogue will be displayed showing the page zones available to the user:



A page zone can be select by pressing the '>' chevron or by selecting a page zone and then pressing the 'OK' button.

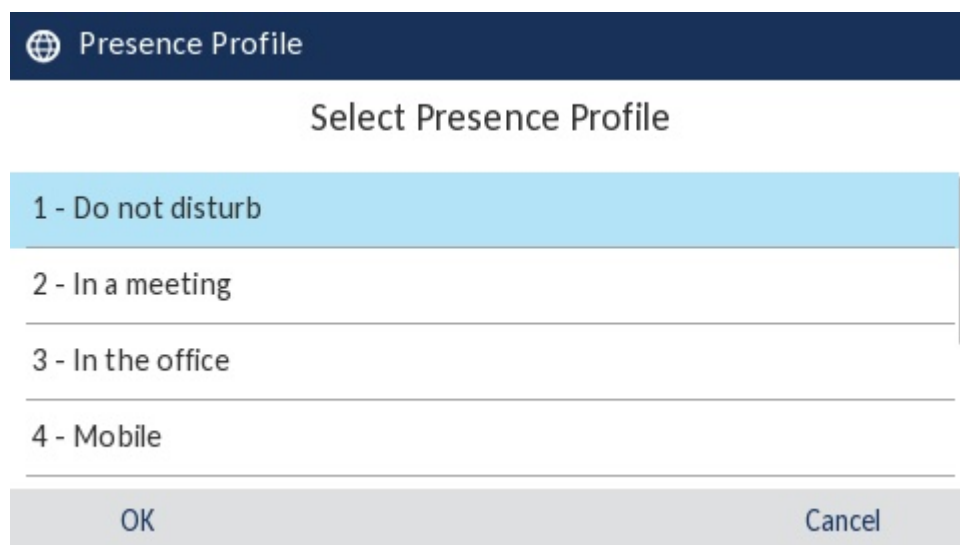
Presence Profile

The presence profile softkey can be used to a change a user's currently selected [Presence Profile](#). The name of the currently selected profile will be displayed in the label of the softkey.

 For the Presence Profile softkey to operate, the phone must be assigned to a [user's](#) Primary, Secondary or DEE device.

Changing Profile

Pressing the presence profile softkey will cause the following dialogue to be displayed listing all the user's presence profiles.



To change the current profile, press the '>' chevron or select a profile and then press the 'OK' button.

Depending on the configuration of the profile selected, the dialogues associated with the Do Not Disturb and/or Forward softkeys may be displayed prompting the user for additional information.

Station Monitor

The station monitor feature can be used to listen in on a call in progress at another extension on the phone system. Once the monitor is in progress, the monitoring party can barge in or steal the call if required.

 For station monitor to work, permission must have been granted on the telephone system. If a 'Call Failed' message is received, check that the correct permissions have been applied.

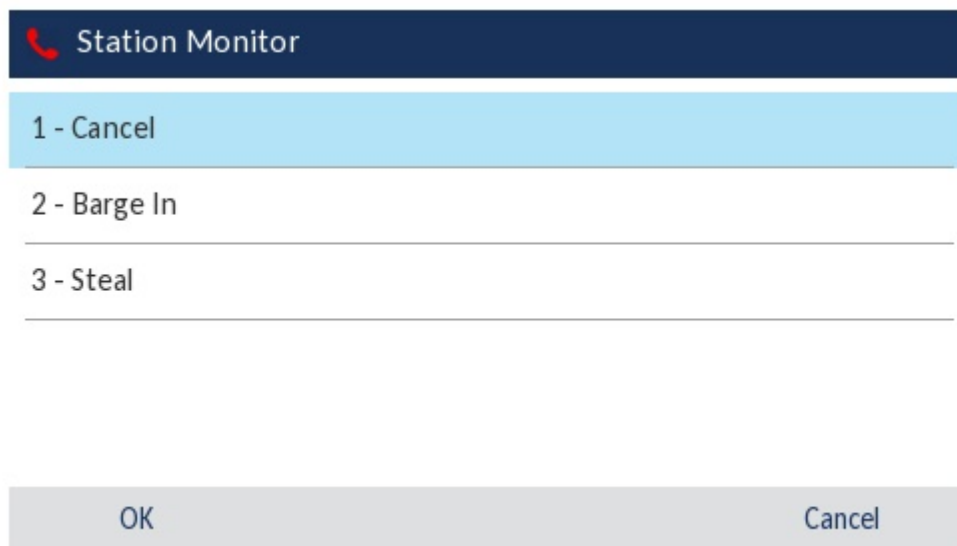
Monitoring

Pressing the monitor key will cause the monitor feature code to be dialed on the telephone. To complete the monitoring process, enter the number of the extension you wish to monitor or press one of the BLF keys configured on the phone.

The monitor can be cancelled by terminating the call.

Barge In / Steal

If a station monitor is in progress, pressing the station monitor key will load the options menu:



The image shows a softkey menu titled 'Station Monitor' with a red telephone handset icon. The menu contains three options: '1 - Cancel', '2 - Barge In', and '3 - Steal'. Each option is preceded by a right-pointing chevron '>'. At the bottom of the menu are two buttons: 'OK' and 'Cancel'.

To toggle the barge in status or steal the call, press the '>' chevron next to the entry or select an entry and press the 'OK' button.

System Speed Dial

The system speed dial softkey can be used to make a call to one of the speed dials configured on the telephone system. The sections below outline the user interaction when pressing the system speed dial softkey.

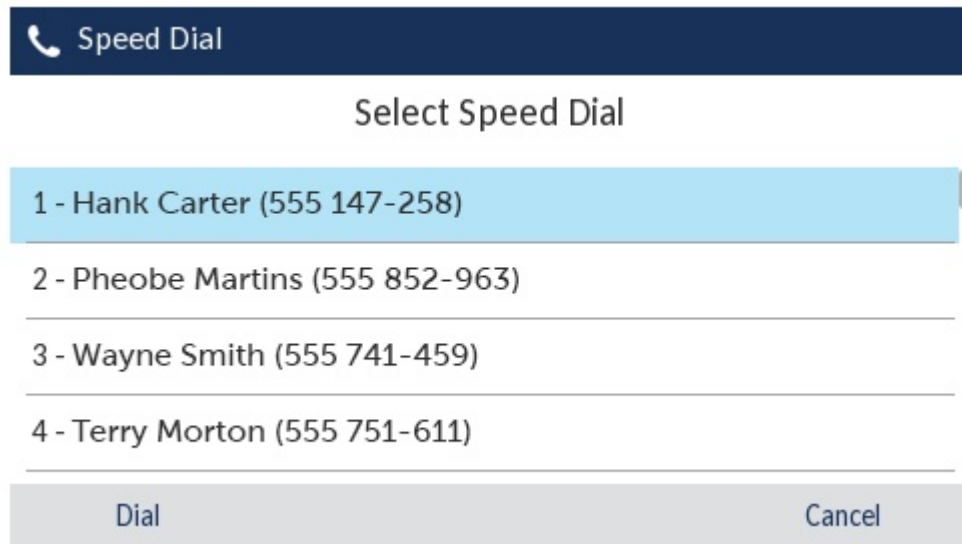
 Currently there is no way to dial system speed dials set to private from a 6900 phone.

Specific Speed Dial

If a specific speed dial has been selected in the parameters of the softkey, the number will be dialled as soon as the button is pressed.

User Choice

If no specific speed dial has been configured in the parameters of the softkey, a list of speed dials available will be presented on the screen:



To dial a speed dial entry, press the '>' chevron next to the entry or select an entry and press the 'dial' button.

 The system speed dial can be searched more easily by using the directory built into the phone. Pressing the  button on the phone will access the directory search screen.

Transfer Immediate

The 'Transfer Immediate' softkey can be used to transfer a connected call directly to another device with no enquiry call. The sections below outline the user interaction when the softkey is pressed.

Target Device

If the target device parameter is configured against the softkey, the call will immediately be transferred to the device with no further user interaction required.

No Target Device

If no target device is configured against the softkey, the following dialogue will appear prompting for the device to be entered:

→ Immediate Transfer Destination

Enter the destination to transfer to

OK Backspace Cancel

Once the required device number has been entered, pressing the 'OK' button will transfer the call.

 If using an Expansion Module (PKM), a BLF key can be pressed instead of entering a target device number when the dialogue is displayed.

Transfer to Voicemail

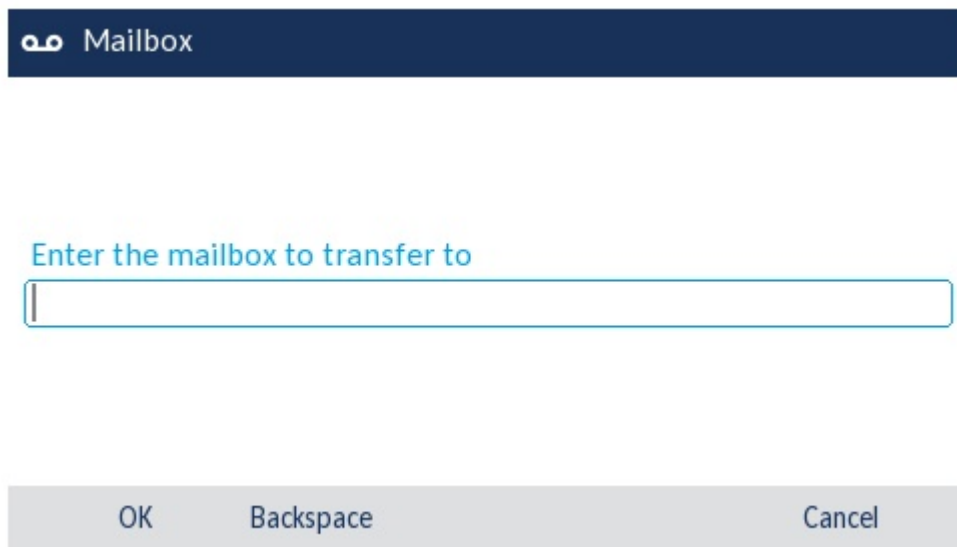
The 'Transfer to Voicemail' softkey can be used to transfer a connected call directly into someone else's mailbox. The sections below outline the user interaction when the softkey is pressed.

Mailbox

If the mailbox parameter is configured against the softkey, the call will immediately be transferred to the mailbox with no further user interaction required.

No Mailbox

If no mailbox is configured against the softkey, the following dialogue will appear prompting for the mailbox to be entered:



Mailbox

Enter the mailbox to transfer to

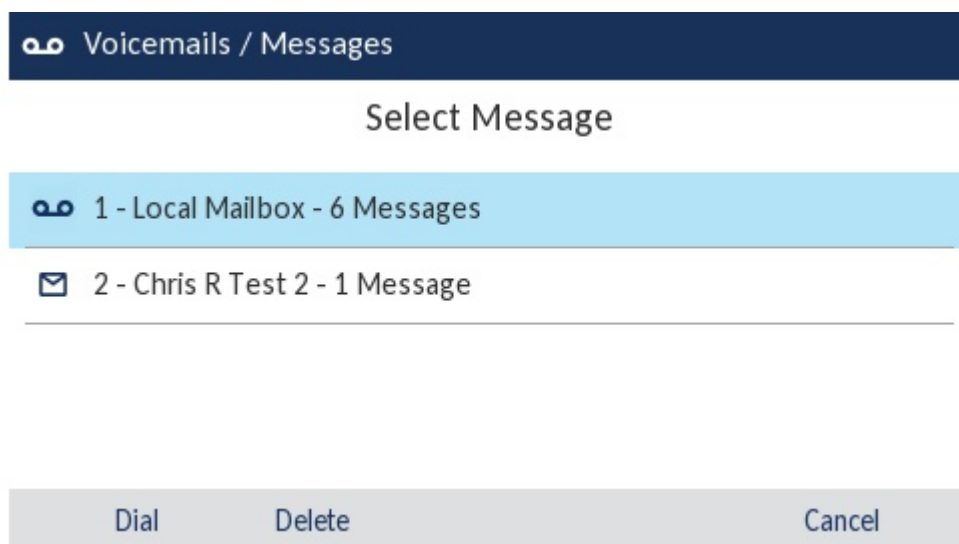
OK Backspace Cancel

Once the required mailbox has been entered, pressing the 'OK' button will transfer the call.

-  When transferring calls to off node mailboxes, the off node Voicemail application must have been added to the transferring phone's node. There is not import/export option for this, it must be added manually.
-  If using an Expansion Module (PKM), a BLF key can be pressed instead of entering a mailbox number when the dialogue is displayed.

Voicemail

The voicemail screen appears when the physical 'Voicemail' button () is pressed on the phone. Depending on the current voicemail and message status of the phone, the button will perform different actions.



Voicemails / Messages

Select Message

1 - Local Mailbox - 6 Messages

2 - Chris R Test 2 - 1 Message

Dial Delete Cancel

Status	Action
No voicemails/messages and the phone is idle	The phone will dial into the associated mailbox for admin access.
Voicemails (local mailbox only) and the phone is idle	The phone will dial into the associated mailbox to retrieve voicemails.
Voicemails (no local mailboxes) and the phone is idle	The 'Voicemail / Messages' screen will display showing details of any voicemails or messages left.
Station Messages	
Outbound Ringing Call	The phone will leave a station message with the target extension and hang-up
Connected Call	The phone will display the Transfer to Voicemail screen.

The sections below outline how voicemails and station messages appear on the voicemail screen.

Voicemails

Any voicemail messages will be listed in the selection screen with the voicemail icon () next to them and a description of the mailbox the message is in. To listen to the message, select the mailbox and press the 'Dial' key.

Station Messages

Station messages are direct messages that have been left from another user on the system. Any station

messages will be listed in the selection screen with the message icon () . To return a message (call the sending extension back), select the message on the screen and press the 'Dial' key. The message will automatically be removed from the phone when the returned call is answered.

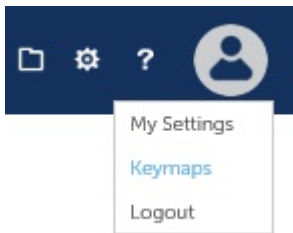
To delete the message without returning the call, press the 'Delete' key.

3.5.3 6900 User Keymaps

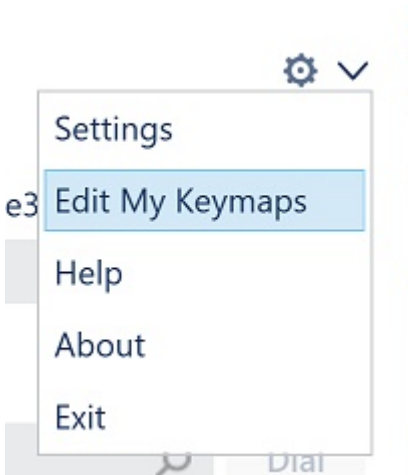
This section of the website provides a way for users to configure any available softkeys on any 6900 handsets they are using.

There are two ways for users to access their own keymap configuration, directly through the MCS website or by using the hyper link from within their Phone Manager Desktop client:

Access via MCS:



Access via Phone Manager:



 If a Phone Manager Desktop user is remotely connected to the MCS, they will not be able to access the User Keymaps page if the website port has not been forwarded through the firewall. User Keymap editing should be performed when on a local connection or through a VPN.

In addition, user keymaps can also be accessed from the [Phones](#) page to allow engineers/administrators to configure user keymaps on a user's behalf.

Editing User Keymaps

A list of 6900 extensions assigned to the user will be displayed in the grid. If there are no extensions displayed there are no 6900 handsets associated with the user's account, please refer to the [Users & Business Units](#) section for more information.

To edit a keymap, press the edit icon () next to an extension. The keymap for that extension will be displayed. Any softkeys that have been configured centrally (either System, Model or Handset level) will be displayed on the keymap for the user to see but will be read-only. Any softkey that has not been configured centrally can be manually configured by the user.

For information on the features available, please refer to the [6900 Softkey Features](#) section.

3.6 6900 Handset Configuration Profiles

Many of the features normally controlled by the telephone system (such as House Phone, Headset Mode, Time zone etc) are instead configured directly on 6900 series phones. Configuration profiles provide a way to configure the handsets centrally.

Configuration profiles can be assigned to handsets in three different ways:

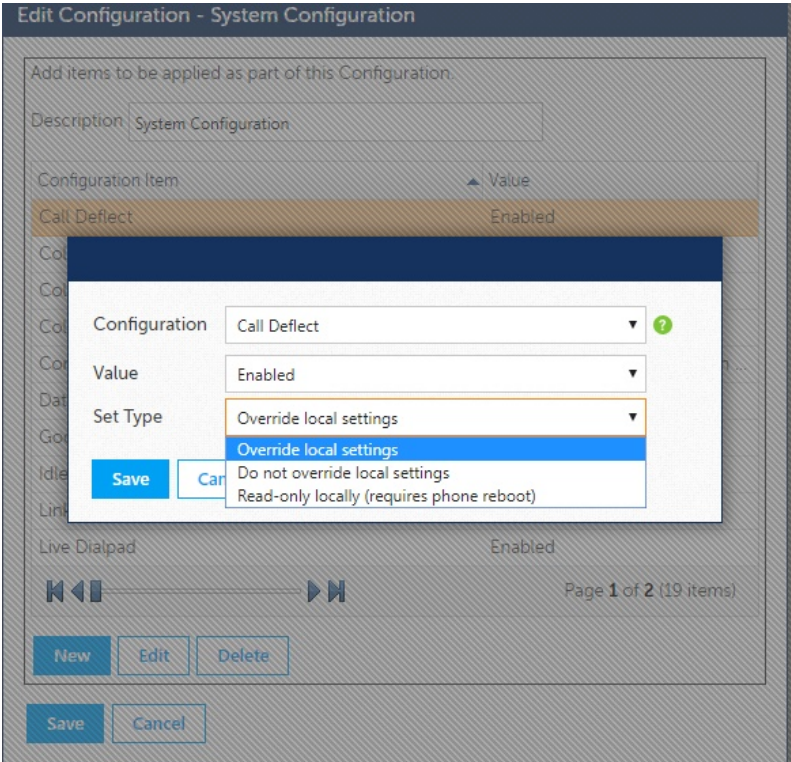
- Default Configuration, assigned to all handsets by default
- Model Configuration, assigned to all handsets of a specific model
- Phone Configuration, assigned directly to a phone or Hot Desk Device

When a handset is provided a configuration, the Default (system-wide) configuration and any model or phone configurations are merged together. Configuration options set at model level will override any set at Default level and configuration options set at phone level will override those at model level.

Please refer to the [Configuration Options](#) section for information about each options available.

Editing a Configuration

Any of the available configuration items can be added to a configuration profile by pressing the 'New' button. Select the configuration from the drop down list then select the value. Depending on the configuration item selected, the value available will be different. For information on the valid values, please refer to the tooltip.



The 'Set Type' option controls how the configuration will be applied to the phone. The available types are:

- Override local settings -> The local setting on the phone will be ignored
- Do not override local settings -> Will be used only if the local phone setting has not be changed from the default
- Read-only locally (requires phone reboot) -> Will override the local setting on the phone and will become greyed out (read-only)

Restarting

In addition to the read-only option, some configuration items will only take effect after a restart. For information about which configuration items this applies to, please refer to the [Configuration Options](#) section.

Default Configuration

To help with the initial rollout of handsets, the default configuration is pre-populated with commonly configured items. The table below shows the options that are enabled by default and what they are configured as.

Option	Default	Description
Alert Auto Call Distribution (ACD)	Bellcore-dr5	Sets the ringtone for ACD calls to be Bellcore-dr4.
Auto External	Bellcore-dr2	Sets the ringtone for External calls to be Bellcore-dr2.
Auto Group (UCD)	Bellcore-dr4	Sets the ringtone for UCD calls to be Bellcore-dr4.
Auto Resync Mode	3 (Firmware and Config)	Checks for updates of configuration files and firmware automatically at the specified time.
Auto Resync Time	04:00	Sets the time to 04:00 that it will do the Auto Resync of firmware and config. Note this causes the phone to reboot at this time.
Call Deflect	Enabled	Enables the display of the 'deflect' softkey when an inbound call is alerting the handset
Call Hold Reminder	Enabled	Enables the reminder ring splash as soon as you place a call on hold. When enabled, the phone initiates a reminder ring splash periodically for the single call on hold. When disabled, no reminder ring splash is audible.
Call Waiting Tone Period	5	Causes call waiting tone to play every 5 seconds if you have an incoming call while on an active call
Collapsed Context User Softkey Screen	Enabled	Causes Softkeys to collapse onto the first page if there is space when on a call
Collapsed More Softkey Screen	Enabled	Causes the More button to be removed if there is the exact number of keys to fit on the screen.
Collapsed Softkey Screen	Disabled	Causes Top Softkeys to retain their position and not move up to empty spaces
Collapsed Softkey Screen Offset Bottom	23	Causes Bottom Softkeys to compress so that they are visible where possible.
Conference/Transfer In Live Dial Mode	Live Dial Mode (With Dial Plan Matching)	When a user initiates a conference call or transfer, they hear a dial tone before dialling begins. The phone automatically dials out if the number matches the local dial plan or if it reaches the set digit timeout.
Date Format	Depends on region	• UK / Canada - DD MMM YYYY • US - MM DD YYYY
Goodbye Key Cancels Incoming Call	Disabled	When disabled, the 'Goodbye' key will instead clear the call in progress.
HTTPS Auth Whitelist Period	10	Trust the HTTPS server for 10 minutes after each request.
Idle Screen Mode	Secondary	Secondary mode displays the extension name and number on the main idle area of the screen.
Live Dialpad	Enabled	Enables real-time validation of the SIP dial plan so that calls are made without having to press the 'dial' key. This mimics normal MiNET/Digital handset behaviour.
Mask SIP Password	Enabled	Enables the enhanced security feature whereby a user's SIP account password is hidden/masked in the server.cfg and local.cfg files (downloaded from the IP phone's Web UI troubleshooting page for debug purposes).
Max Boot Count	0	Disables the phone going into web recovery mode automatically.
Priority Alerting Enabled	Enabled	Enables priority ringtones for ACD, External & Group calls.
Picture Refresh Timeout	4	Indicates the timeout in hours for a downloaded picture.
Screensaver Background Image	Mitel Screensaver	Sets the Mitel Screensaver images to appear on the phone after 240 seconds idle and rotate image every 5 minutes
Screensaver Refresh Timer (minutes)	5	
Screensaver Time (seconds)	240	
SIP Allow Auto Answer	Disabled	This stops the phone answering all internal calls automatically by default.
SIP Intercom Allow Barge In	Disabled	This stops the phone putting calls on hold when a page occurs.
SIP Intercom Mute Mic	False - Microphone is not muted	This means the microphone is not muted when an intercom call is made.
SIP Intercom Type	Phone	This enables the 6900 intercom feature (direct page)
SIP Keepalive Timer	60	Keeps NAT table entries refreshed between phone and server
SIP Refer-to With Replaces	Enabled	Ensures the correct OAI is generated when doing supervised transfers.

SIP Registration Period	1200	This sets the phone registration period to 20 minutes (causing the phone to re-register with the telephone system every 10 minutes).
Switch Focus to Ringing Line	Disable	This is disabled to stop an incoming call interfering with a transfer.
Time Format	Depends on region	• UK / Canada - HH:mm • US - hh:mm
Time Server 1	1.mitel.pool.ntp.org	Provides a default set of public NTP servers to keep the time on the phone accurate. These may need adjusting to suit your network setup <i>Note: If the 'Use MCS as local NTP Server' setting has been enabled, Time Server 3 will be populated with the IP address of the MCS server and will automatically be added to the configuration of local phones.</i>
Time Server 2	2.mitel.pool.ntp.org	
Time Server 3	3.mitel.pool.ntp.org	
Time Zone Name	Depends on region	• Canada - CA-Eastern • UK - GB-London • US - US-Eastern

Restricted Configurations

Some of the configuration options that are available on the handsets are not accessible through a configuration profile. These are designated as 'Restricted' options because they are provided to the phone automatically by the MCS server and are key to the normal operation of the phone in conjunction with the MCS server.

The following options are restricted and cannot be changed by profile or manually on the phone:

Area	Option	Description
SIP Configuration (Device)	SIP Auth Name	These settings are automatically populated with the relevant information based on the handset's fixed extension or logged in hot desk device. 'Display Name' is set to the extension number. 'Screen Name' is set to the mapped user's name unless there is no user, in which case it is set to the extension description. 'SIP Vmail' is set to the extension's message retrieval extension
	SIP Display Name	
	SIP Password	
	SIP Screen Name	
	SIP Screen Name 2	
	SIP User name	
	SIP Vmail	
SIP Configuration (Server)	SIP Proxy IP	These settings are automatically populated with the SIP configuration details for the node of the handset's fixed extension or logged in hot desk device. This information can be configured against the relevant node in the Phone Systems section.
	SIP Proxy Port	
	SIP Registrar IP	
	SIP Registrar Port	
	SIP Registration Timeout Retry Timer	30
	SIP XML Notify Event	Enabled
	SIP Use Basic Codecs	1 (limits to using just the basic codecs G.711 u-Law, G.711 a-Law,G.729)
Configuration Server	Firmware Server	Set to the 6900 Handset Server Address
	Image Server URI	Set to the 6900 Handset Server Address
	Download Protocol	https
	Https Path	config
	Https Port	8202
	Https Server	Set to the 6900 Handset Server Address
	Https Validate Certificates	Enabled
	Upload System Info Server	Set to the 6900 Handset Server Address
	Web Interface Blacklist Duration	0
	XML Application Post List	Set to the 6900 Handset Server Address
Phone UI	Idle Screen Avatar	Set to enabled to show the user's profile image on the handset
	Softkey Selection List	Limits the softkeys the user can configure through the UI: <i>speeddial,lcr,speeddialxfer,speeddialconf,directory,callers,redial,conf,xfer,icom,phonelock,paging,discreetringing,callhistory,empty</i>
	DND UI	Enables/disable access to DND through the UI: <i>disabled</i>
Diagnostics	Log Module XML	If Diagnostics are enabled against a handset in the 6900 Phones section, this will be enabled.
	Log Server IP	If Diagnostics are enabled against a handset in the 6900 Phones section, this will be set to the 6900 Handset Server Address, either local or remote. Note: For logging to operate, the 6900 Handset Server address must be an IP Address, not a DNS name.
	Log Server Port	514
Action URIs	Action URI Connected	Actions URIs configure the phone to update the server with information about call events. This is required for Phone Manager to be able to track phone status and provide call control
	Action URI Incoming	
	Action URI Disconnected	

	Action URI Registered	
	Action URI Registration Event	
	Action URI Offhook	
	Action URI Onhook	
	Action URI Outgoing	
	Action URI Startup	
	Action URI XML SIP Notify	
Directories	Directory 1	This directory is automatically populated with the telephone system intercom & system speed dial directories.
	Directory 1 Name	
	Directory 1 Enabled	
	Directory 2	This directory is automatically populated with the user's Phone Manager Personal Directory contacts.
	Directory 2 Name	
	Directory 2 Enabled	
	LDAP Enabled	This directory is automatically populated with the required configuration when server based Directory Access has been enabled.
	LDAP Name	
	LDAP Base DN	
	LDAP Server	
	LDAP Downloaded	
Dial Plan	SIP Dial Plan	This option is automatically populated with information from the Dial Plan configured on the MCS unless it has been overridden in the 6900 General Settings .
Paging	Paging Group Listening	This option is automatically populated with multicast addresses based on which pages zones (6900 Handset Page Zones) that handset has been placed in. For more information, please refer to the 6900 Paging section.
Phone Lock	XML Lock Override	This option is automatically set to 1 to ensure that the configuration server can post softkey & configuration updates to the handsets even when they are locked.

3.6.1 6900 Handset Configuration Options

The following table outlines the different 6900 handset configuration options that are made available through configuration profiles. Not all the configuration options available on the handsets are directly supported by the MCS server. For information on how to apply a configuration option which is not directly supported, please refer to the [Custom Configuration Options](#) section.



Some configuration options require a reboot of the phone to take effect. Please refer to the table below for information about specific options.



For detailed information about each configuration option, please refer to the Mitel 6900 SIP Administration Guide

Option	Description	Requires Phone Reboot
Admin Password	Allows you to set a new administrator password for the IP phone.	No
Alert Auto Call Distribution (ACD)	If 'Priority Alerting' is enabled, the configured Bellcore tone will be used for alerting ACD calls.  Alert Auto Call Distribution (ACD) will not work if the SIP device was created on the PBX as a generic SIP device - only if it was created as a 69xx SIP Device	No
Alert External	If 'Priority Alerting' is enabled, the configured Bellcore tone will be used for alerting external calls.  Alert External will not work if the SIP device was created on the PBX as a generic SIP device - only if it was created as a 69xx SIP Device	No
Alert Group (UCD)	If 'Priority Alerting' is enabled, the configured Bellcore tone will be used for alerting UCD calls.  Alert Group (UCD) will not work if the SIP device was created on the PBX as a generic SIP device - only if it was created as a 69xx SIP Device	No
Audio Diagnostic	Enables/disabled audio diagnostic features of the phone.	No
Audio Mode	Allows you to configure how the "handsfree" button on the IP phone operates.	No
Auto Resync Days	Specifies the amount of days that the phone waits between checksync operations. Note: A value of 0 causes the phone to checksync every time the clock reads the proper time. A value of 1 forces the phone to wait 24 hours prior to doing the first checksync.	No
Auto Resync Max Delay	Specifies the maximum time, in minutes, the phone waits past the scheduled time before starting a checksync.	No
Auto Resync Mode	Enables and disables the phone to be updated automatically once a day at a specific time in a 24-hour period. This parameter works with TFTP, FTP, and HTTP servers.	No
Auto Resync Time	Sets the time of day in a 24-hour period for the IP phone to be automatically updated. This parameter works with TFTP, FTP, and HTTP servers.	No
Background Image	Specifies the location of the background image for the idle screen on phones.	No
Backlight Mode	Allows you to turn the backlight on the LCD, Off (always off) or Auto. The "Auto" setting sets the phone to turn off the backlight after a period of inactivity. You can set the amount of time before the backlight goes off using the "Backlight on Time" option ("bl on time" parameter).	Yes
Backlight On time	Allows you to set the amount of time, in seconds, that the backlight stays ON before turning OFF because of inactivity.	Yes
Brightness Level	Specifies the brightness level when the phone is active.	Yes

Call Deflect	Enables or disables the ability to deflect a call to another number during a ringing state.	Yes
Call Hold Reminder	Enables or disables the reminder ring splash timer to start as soon as you put a call on hold (even when no other calls are active on the phone). When enabled, the phone initiates a reminder ring splash periodically for the single call on hold. When disabled, no reminder ring splash is audible. Note: When enabled, by default the reminder will notify after 10 seconds then every 60 seconds thereafter.	Yes
Call Hold Reminder During Active Calls	Enables or disables the ability for the phone to initiate a continuous reminder tone on the active call when another call is on hold. For example, when the call on Line 1 is on hold, and the User answers a call on Line 2 and stays on that line, a reminder tone is played in the active audio path on Line 2 to remind the User that there is still a call on hold on Line 1. When this feature is disabled, a ring splash is heard when the active call hangs up and there is still a call on hold.	Yes
Call Hold Reminder Frequency	Specifies the time interval, in seconds, between each ring splash sound on the active line. For example, if a call comes into Line 1, and then a call comes into Line 2 and you answer Line 2, Line 1 is automatically placed on hold. While on the active Line 2, after 7 seconds, a ring splash audio sounds on the line reminding you that the call on Line 1 is still on hold (determined by the “call hold reminder timer” parameter), and then the ring splash is heard again after 60 seconds (determined by this parameter).	Yes
Call Hold Reminder Timer	Specifies the time delay, in seconds, that a ring splash is heard on an active call when another call was placed on hold. For example, if a call comes into Line 1, and then a call comes into Line 2 and you answer Line 2, Line 1 is automatically placed on hold. While on the active Line 2, after 7 seconds, a ring splash audio sounds on the line reminding you that the call on Line 1 is still on hold. This timer begins to increment after Line 2 is answered.	Yes
Call Transfer Disabled	Enables or disables the Xfer key on the IP phone. If this parameter is set to false, the key is active and can be pressed by the user. If this parameter is set to true, pressing the Xfer key is ignored.	No
Call Waiting	Allows you to enable or disable Call Waiting on the IP phone.	Yes
Call Waiting Tone	Enable or disables the playing of a call waiting tone when a caller is on an active call and a new call comes into the phone.	Yes
Call Waiting Tone Period	Specifies the time period, in seconds, that the call waiting tone is audible on an active call when another call comes in. When enabled, the call waiting tone plays at regular intervals for the amount of time set for this parameter.	Yes
Collapsed Context User Softkey Screen	When enabled, user configured softkeys on the applicable phones will collapse and fill in any unused softkeys starting on the first page of softkeys during the following states: • outgoing • ringing • connected • hold	Yes
Collapsed More	Controls how softkeys are displayed on the 6920, 6930, 6940 and 6970 IP	Yes

Softkey Screen	phones' screens when the number of softkeys configured matches the exact number of softkey buttons on the phone.	
Collapsed Softkey Screen	Enables or disables the 6920/6930/6940/6970 from collapsing the softkeys to remove blank keys	Yes
Collapsed Softkey Screen Offset Top	Defines the offset for locking or collapsing the top softkeys. If the "collapsed softkey screen" parameter is enabled, the phone will lock the respective top softkeys (from top softkey 1 to the value defined) and collapse the rest of the top softkeys. If the "collapsed softkey screen" parameter is disabled, the phone will collapse the respective top softkeys (from top softkey 1 to the value defined) and lock the rest of the top softkeys.	Yes
Collapsed Softkey Screen Offset Bottom	Defines the offset for locking or collapsing the bottom softkeys. If the "collapsed softkey screen" parameter is enabled, the phone will lock the respective bottom softkeys (from bottom softkey 1 to the value defined) and collapse the rest of the bottom softkeys. If the "collapsed softkey screen" parameter is disabled, the phone will collapse the respective bottom softkeys (from bottom softkey 1 to the value defined) and lock the rest of the bottom softkeys.	Yes
Conference Disabled	Enables or disables the Conf key on the IP phone. If this parameter is set to 0 - False, the key is active and can be pressed by the user. If this parameter is set to 1 - True, pressing the Conf key is ignored.	No
Conference/Transfer In Live Dial Mode	Enables/disables support for live dial mode when initiating a conference call or transfer.	Yes
Date Format	This parameter allows the user to change the date to various formats.	No
Directory Digits Match	Specifies how many digits of an incoming call's phone number the phone will take into consideration when performing a directory lookup.	Yes
Display DTMF Digits	Enables and disables the display of DTMF digits when dialling on the IP phone.	Yes
DST Config	Enables/disables the use of daylight savings time.	No
Ethernet Port Mirroring	Enables/disables the mirroring of the Ethernet port to the PC port.	No
Goodbye Key Cancels Incoming Call	Enable or disables the behaviour of the Goodbye Key on the IP phone.	Yes
HTTPS Auth Whitelist Period	Speed up HTTPS communication by trusting the HTTPS server for the specified number of minutes.	Yes
Idle Screen Font Color	Specifies the font colour of the date, time, and status message text on the home/idle screen of the 6920, 6930, 6940 and 6970 IP phones.	Yes
Idle Screen Mode	Used to switch between the two Home/Idle screen modes. The primary screen mode provides users with a larger date and time and displays the Screen Name ("sip screen name") parameter beside the line number in the top status bar. The secondary screen mode displays both the Screen Name and Screen Name 2 ("sip screen name 2") parameters above the smaller, repositioned date and time.	Yes
Inactivity Brightness Level	Specifies the brightness level when the phone is inactive.	No
Input Language	Allows you to specify the language to use for inputs on the IP Phone. Entering a	Yes

	language value for this parameter allows users to enter text and characters in the IP Phone UI and in XML applications via the keypad on the phone (or for the 6940 & 6970 on-screen keyboard), in the language(s) specified.	
Language	The language you want to display for the IP Phone UI.	No
Line Show Caller Id	Enable or disable the caller ID display on the line and mobile softkeys.	Yes
Live Dialpad	Turns the “Live Dialpad” feature ON or OFF. With live dial pad ON, the IP phone automatically dials out and turns ON Handsfree mode as soon as a dial pad key or softkey is pressed. With live dial pad OFF, if you dial a number while the phone is on-hook, lifting the receiver or pressing the Speaker/Headset key initiates a call to that number.	No
Link Layer Discovery Protocol	Enables or disables Link Layer Discovery Protocol for Media Endpoint Devices (LLD-MED) on the IP Phone.	Yes
Log Issue	Enables or disables the Log Issue button on the phone. This is automatically enabled when diagnostics are enabled on the phone through the Phones page.	No
Map Conf Key To	Sets the Conf key as a Speed Dial key if a value is entered for this parameter. If you leave this parameter blank, the Conf key returns to its original functionality.	No
Map Redial key To	Sets the Redial key as a Speedial key if a value is entered for this parameter. If you leave this parameter blank, the Redial key returns to its original functionality.	No
Mask SIP Password	Enables the enhanced security feature whereby a user’s SIP account password is hidden/masked in the server.cfg and local.cfg files (downloaded from the IP phone’s Web UI troubleshooting page for debug purposes).	Yes
Max Boot Count	Specifies the number of faulty boots that occur before the phone is forced into Web recovery mode.	Yes
Minimum Ringer Volume	Specifies the minimum ringer volume level.	Yes
Missed Calls Indicator Disabled	Enables or disables the Missed Calls Indicator. If the “missed calls indicator disabled” parameter is set to false, the indicator increments as unanswered calls come into the IP phone. If the “missed calls indicator disabled” parameter is set to true, the indicator is disabled and will NOT increment as unanswered calls come into the IP phone.	Yes
Mobile Contacts Enabled	Enable or disable the mobile contacts import.	Yes
Mobile Contacts Name	Override the default folder name “Mobile Contacts” to provide a customized name to the mobile contacts folder.	Yes
Mute DTMF Playback	Disabled (default), the user hears locally generated DTMF. Enabled, locally generated DTMF is muted.	No
Options Password Enabled	Enables or disables password protection of the Options key on the IP phone. If enabled, upon pressing the Options key, a user has to enter a password at the IP phone UI.	No
PC Port PassThru Enabled	Enables or disables the PC port.	Yes
Picture Refresh Timeout	Indicates the timeout in hours for a downloaded picture.	Yes

Priority Alerting Enabled	Enables and disables distinctive ringing on the IP phone for incoming external, ACD & UCD calls. Use the Alert Auto Call Distribution, Alert External & Alert Group settings to choose ringtones.  Priority Alerting Enabled will not work if the SIP device was created on the PBX as a generic SIP device - only if it was created as a 69xx SIP Device	No
Priority Non-IP	Specifies the priority value for non-IP packets.	Yes
Proximity Lock	Enable or disable phone automatic lock on proximity detection.	Yes
Proximity Lock Delay	When proximity lock is enabled, delay after proximity event detection before locking the phone.	Yes
Proximity Unlock	Enable or disable phone automatic unlock on proximity detection.	Yes
Proximity Unlock Delay	When proximity unlock is enabled, delay after proximity event detection before unlocking the phone.	Yes
Redial Disabled	Enables or disables the Redial key on the IP phone. If this parameter is set to 0 - False, the key is active and can be pressed by the user. If this parameter is set to 1 - True, pressing the Redial key is ignored, and the dialled number is not saved to the "Outgoing Redial List".	No
Ring Audibly Enable	Enables/disables the feature whereby the ring tone of an incoming call is played through the IP phone's speaker if a user is on an active call.	Yes
Ringback Timeout	Specifies the timeout period (in seconds) before the phone terminates the call. For outgoing calls, the originating phone will send a SIP CANCEL to stop the ringing at the destination phone after the timeout expires. For incoming calls, the terminating phone will send a SIP 486 Busy Here to stop the ringback at the originating phone after the timeout expires.	No
Ringtone	Globally sets the type of ring tone on the IP phone. Ring tone can be set to one of 15 pre-defined rings (excluding silence) or one of 8 custom ringtones.  To use a custom ring tone, it must be uploaded via the phone's web UI or using a TFTP server.	No
RTP Symmetric Port	By default the phones support symmetrical RTP port handling (i.e. the phone will only play an RTP stream if it comes from a "source" port that is the same as the "listening" port negotiated by the call manager. By disabling this parameter, administrators can configure the phones to support asymmetrical RTP port handling. If disabled, the phone will accept an RTP stream coming from a different "source" port and send RTP traffic to the caller at the "listening" port.	Yes
Screensaver Time (seconds)	Specifies the amount of time (in seconds) the phone must be idle before the 6920/6930/6940/6970 IP phone's screen saver initiates.	Yes

Screensaver Background Image	Specifies the location and file name of the background image for the screen saver on 6920, 6930, 6940 and 6970 phones.	Yes
Screensaver Refresh Timer (minutes)	Specifies the interval (in minutes) when the screen saver background image is refreshed.	Yes
Simplified Options Menu	Allows you to enable a simplified options menu or enable the full menu on the IP Phone UI.	Yes
SIP Allow Auto Answer	Enables or disables the IP phone to allow automatic answering for an Intercom call. If auto-answer is enabled on the IP phone, the phone plays a tone to alert the user before answering the intercom call. If auto-answer is disabled, the phone treats the incoming intercom call as a normal call.	No
SIP Auto-Dial Number	Globally or on a per-line basis, specifies the SIP phone number that the IP phone autodial when the handset is lifted from the phone cradle. An empty (blank) value disables autodial on the phone.	No
SIP Auto-Dial Timeout	Globally or on a per-line basis, specifies the time, in seconds, that the phone waits to dial a pre-configured number after the handset is lifted from the IP phone cradle.	No
SIP Dial Plan Terminator	Specifies whether or not pressing the hash/pound (i.e. "#") key, while performing an outgoing call on an open line, should be sent as %23 to the proxy in the dial string or if the key should be used as a dial plan terminator (i.e. dials out the call immediately).	No
SIP Digit Timeout	Represents the time, in seconds, between consecutive key presses on the IP phone. The default for this parameter is 4 seconds. If you press a key on the phone and wait 4 seconds before pressing the next key, the key times out and cancels the digit selection. You must press consecutive keys before the timeout occurs.	No
SIP Early Media Mute Mic	Enables or disables the microphone while in early media.	No
SIP Intercom Allow Barge In	Enable or disables how the phone handles incoming intercom calls while the phone is on an active call as well as how the phone handles multicast paging calls while the phone is in a dialling state.	No
SIP Intercom Line	Specifies the line for which the IP phone uses the configuration from, when making the Intercom call. The IP phone uses the first available line for physically making the call but uses the configuration from the line you set for this parameter.	No
SIP Intercom Mute Mic	Enables or disables the microphone on the IP phone for Intercom calls made by the originating caller.	No
SIP Intercom Type	Determines whether the IP phone or the server is responsible for notifying the recipient that an Intercom call is being placed.	No
SIP Intercom Warning Tone	Enables or disables a warning tone to play when the phone receives an incoming intercom call on an active line.	No
SIP Keepalive Timer	This value is how many seconds to wait before sending a SIP UDP keep alive packet to the configured SIP servers. A zero value disables this feature. Note: This is only for UDP transport protocol.	No
SIP Outbound	Enables and disables whether or not the phone uses Draft 15 (draft-ietf-sip-	No

Support	outbound-15) support for SIP outbound packets. A SIP User Agent (UA) behind a firewall, reuses an existing connection (usually the REGISTER outbound connection) for the inbound request if the proxy supports it. The UA uses keep-alive packets to monitor the connection status. Notes: • If the Global SIP parameter “Persistent TLS” is set on the phone, then only one TLS persistent connection can be established since the phone uses the local port 5061 for connection. If the Global SIP parameter “TLS” is set on the phone, more than one connection can be setup since the phone uses a random local port for connection. • This parameter must be enabled for this feature to start keep-alive for a particular transport.	
SIP Refer-to With Replaces	Flag for controlling the mode of a semi-attended transfer.	No
SIP Registration Period	The requested registration period, in seconds, from the registrar.	No
SIP RPort	Allows you to enable (1) or disable (0) the use of Rport on the IP phone. “Rport” in RFC 3581, allows a client to request that the server send the response back to the source IP address and the port from which the request came. Note: Configuring the Rport parameter is recommended for clients behind a firewall.	No
SIP RTP Port	Indicates the port through which the RTP packets are sent. This value must specify the beginning of the RTP port range on the gateway or router. The RTP port is used for sending DTMF tones and for the audio stream. Your network administrator may close some ports for security reasons. You may want to use this parameter to send RTP data using a different port.	No
SIP Update Caller Id	Enables or disables the updating of the Caller ID information during a call.	No
Suppress DTMF Playback	Enables and disables suppression of DTMF playback when a number is dialed from the softkeys or programmable keys.	No
Suppress Incoming DTMF Playback	Suppress playback of both SIP INFO and RFC2833 DTMF tones.	Yes
Switch Focus to Ringing Line	Specifies whether or not the UI focus is switched to a ringing line while the phone is in the connected state.	No
Tagging Enabled	Enables or disables VLAN on the IP phones.	Yes
Time Format	This parameter changes the time to 12 hour or 24 hour format. Use “0” for the 12 hour format and “1” for the 24 hour format.	No
Time Server 1	This parameter allows you to set the IP address of Time Server 1 in dotted decimal format.	No
Time Server 2	This parameter allows you to set the IP address of Time Server 2 in dotted decimal format.	No
Time Server 3	This parameter allows you to set the IP address of Time Server 3 in dotted decimal format.	No
Time Server Disabled	Enables or disables the time server. This parameter affects the time server1, time server2, and time server3 parameters. Setting this parameter to 0 allows the use	No

	of the configured Time Server(s). Setting this parameter to 1 prevents the use of the configured Time Server(s).	
Time Zone Minutes	The number of minutes the timezone is offset from UTC (Coordinated Universal Time). This can be positive (West of the Prime Meridian) or negative (East of the Prim Meridian). Eastern Standard Time (EST) has a value of 300 which is the default.	No
Time Zone Name	This parameter allows you to set the time zone code or customize the time zone for their area as required.	No
Tone Set	Globally sets a tone set for a specific country.	No
User Password	Allows you to set a new user password for the IP phone.	No
VLAN Id	Allows you to configure a VLAN ID that associates with the physical Ethernet Port 0 (LAN port).	Yes
VLAN Id Port 1	Allows you to configure a VLAN ID that associates with the physical Ethernet Port 1 (PC port).	Yes
Web Interface Enabled	Specifies whether or not to disable the web user interface.	Yes

Custom Configuration Options

To support configuration options that might be added to the phones in the future (and ones that have not been added to the configuration profile template), it is possible to add custom entries to a configuration profile.

To do this, select the 'Custom...' option from the list when adding an option to a configuration profile.

When this is selected, a 'Key' as well as a 'Value' is requested. Enter in the required values and then save the option down.

3.7 6900 Directories

Each 6900 phone provides the user with a number of directories they can use to access internal and external contacts. The phone will automatically match calls against numbers it has in its contact directory.

Directory	Description	Source
Local	This is a directory which is local to the phone. Any contacts added here are stored only on the phone and are not accessible anywhere else or backed up.	Locally Update
Phone System	This directory contains contact information for other users on the telephone system and additional items, including: • Extensions • Hunt Groups • System Speed Dials*	CSV Download
Personal	These are contacts from the user's Phone Manager Personal directory. Any contact added by the user through Phone Manager will be accessible here.  This does not include Outlook contacts that are accessible through the Phone Manager Desktop interface.	CSV Download
Server	Contacts added to the one or more of the server based directories, including directories linked to CRM databases or imported.	LDAP Query
Mobile Contacts	This directory contains the contacts that have been synced from a mobile device which has been paired locally with the phone. This feature is available on 6930 & 6940 (but not 6970) models.	Bluetooth Sync

 * Private system speed dials are not available through this directory.

Directory Updates

The phone system, personal & server directories are provided to the phone when it first boots. Directory updates are requested by the phone every 24 hours. Updates to the directory can take up to 24 hours to appear on all phones.

Directory Limitations

Each contact in each directory is download locally to the phone. It is recommended that no more than 5,000 contact records are provided to the phone to ensure that there are no performance issues when access the directories or contact matching on calls.

If there is a requirement to access directories which contain more than 5,000 records, an asynchronous LDAP lookup can be used for searching (but no contact matching on calls).

 For more information on asynchronous lookups, please refer to the [General Settings](#) section of the server documentation.

3.8 6900 Handset Images

Mitel 6900 series handsets have the ability to display background images and screen savers. Both of these types of image can be customised as required.

Handset images can be created on the server and then assigned to phones as a background image or screen saver through a [Configuration Profile](#).

Adding / Editing Handset Images

To create or edit a handset image on the server, an image file for each Model type must be provided. The image sizes for the models currently supported are shown in the table below:

Model No	Screen/Image Size in Pixels (Width x Height)
6920	320 x 240
6930	480 x 272
6940, 6970	800 x 480

Any images file uploaded must also have the following properties:

- JPEG format
- 24 or 32bit depth
- Less than 1MB in size

To add a handset image, press the 'New' button and enter a unique description into the window that appears. Upload images for all handset models and then press 'Save'.

Background Images

To use a handset image as a background image, the configuration option 'Background Image' must be used. Edit or create a new configuration profile, add a new configuration option for 'Background Image' and then select the handset image from the list provided. If applied to the *[Default]* configuration profile, the background image will display on all phones. Background images can then be overridden in specific configuration files if required for different departments etc.

If a dark image is provided, it may not be possible to read the extension number, time and other idle screen properties. If this is the case, the 'Idle Screen Font Color' configuration option can be used.

 Background images only appear on the handset while it is idle.

 Background images can be useful for showing hot desk status. By configuring a different images for fixed phones and hot desk phones, user will be able to see from a distance which phones are free to logon to.

For information on using handset images in screen savers, please refer to the [6900 Screen Saver](#) section.

3.9 6900 Handset Screen Saver

Each 6900 handset by default runs a local screen saver with pre-loaded images. using [Handset Images](#) uploaded to the MCS, it is possible to customise the screen saver. Screen savers configured on the MCS can be applied to handsets using [Configuration Profiles](#).

Creating / Editing Screen Savers

A screen saver is a group of one or more handset images. Before screen savers can be created, the images required must already have been uploaded through the [Handset Images](#) configuration section of the MCS website.

To create a new screen saver, press the 'New' button then give enter a unique description in the form that appears.

Press the plus icon to the right of the Images property and select an image from the context menu. Repeat this process adding as many images as needed. If required, the same image can be added more than once. when all the images required have been added, press the 'Save' button.

To apply the screen saver to a handset, create a new [Configuration Profile](#) or edit an existing one. Add a new configuration option and select 'Screen Saver Background Image'. Select a screen saver from the list provided then save the option and profile.

Screen savers applied in the *[Default]* configuration profile will apply to all handsets.

The following configuration options can be used to control how the screen saver appears on the handsets:

- ScreenSaver Time, time in seconds before the screen saver appears on the phone
- ScreenSaver Refresh Timer, time in minutes for the phone to switch between images in the screen saver

If required, different screen savers can be applied to different handsets / models by using multiple configuration profiles.

3.10 6900 Handset Hot Desking

The MCS server provides a SIP Hot Desking feature specifically for 6900 series handsets. This differs from the existing MiVoice Office 250 Hot Desking and Agent Hot Desking.

SIP Hot Desking allows users to logon to any 6900 handset that has hot desking enabled (even ones on other nodes*). Once logged in, the handset becomes their own, all keymaps, configuration options and SIP configuration gets transferred to the target handset.

For information on configuring a SIP Hot Desking implementation, please refer to the [6900 Engineering Guidelines](#).



* SIP Hot Desking is supported across nodes if all 6900 handsets are controlled by the same MiVoice Office Application Suite.

Managing SIP Hot Desk Phones

The hot desking configuration section displays all SIP hot desk phones that have been configured on the system.

Each hot desk phone has the following properties:

- Device Number, the SIP device on the telephone system
- PIN, the number used by the user when logging on and off
- Keymap Profile, a keymap profile to apply when logged on (Optional)
- Configuration Profile, a configuration profile to apply when logged on (Optional)

To add a new hot desk phone, press the 'Add' button. In the form that appears, enter the number of the SIP device on the telephone system to configure as a hot desk phone (the device picker will filter out any SIP device which are already have a fixed mapping to a handset).

Once a hot desk phone has been created, it can be used to logon to any handset that has been enabled for accepting hot desk logons. Please see the [6900 Handset Phone](#) section for more information.

The effective keymap option can be used to see the combination of hot desk keymap and user keymap. The actual keymap that gets sent to the phone will differ depending on the phone the hot desk device is logged into.



To login to teleworker phones, SIP Hot Desk phones must also be configured on the MiVoice Border Gateway. Please refer to the [Teleworker](#) section of the 6900 Engineering Guidelines for more information.



Each SIP extension which is to be used as a SIP Hot Desk phone must be configured on the PBX and MCS as any other Softphone/6900 would be. Please refer to the [Softphone/6900 Support](#) section for more information.

3.11 6900 Handset Phone Page Zones

6900 series handsets have a built in paging facility that uses IP multicast to broadcast RTP from one handset to multiple handsets at the same time.

The 6900 Paging section provides an interface for creating one or more 'Page Zones' that can be used to page between groups of handsets.

Due to the nature of IP multicast, 6900 handset paging can only be used where the handset are on the same network and will receive the multicast packets.

 6900 Handset Page Zones are separate to the page zones configured on the telephone system. 6900 handsets can access telephone system page zones to page MiNET and Digital handsets, but this cannot be done in reverse.

Page Zones

Each page zone created requires three properties:

- A list of handsets that will listen to this address and port combination
- An IP multicast address (calculated from the Base Paging Multicast address)
- A port number (uses the Base Paging Port number)

In normal use the IP address and port are configured automatically by the system and will not need to be altered. If they do need to be altered they can be amended under the Advanced Options.

How the multicast address is calculated for each page zone

The 'Base Paging Multicast Address' is used to calculate the IP address for each page zone created. Every time a page zone is added, it is assigned a unique ID which start from 1 and are incremented for each new page zone. This ID is added to the base address to guarantee a unique address for each zone.

This means if you have 10 paging zones they will use (by default) IP addresses 224.0.0.151-224.0.0.160. If a page zone is deleted the IP address will not be re-used, if you delete and add a new one it will use 224.0.0.161.

 The default base paging multicast address is configured to 224.0.0.150.

The base address may need to be changed if there is other equipment on site using multicast addresses in the range from the base address up to the number of paging zones you created. The valid range of IP multicast addresses is: 224.0.0.0 to 239.255.255.255.

 The default base paging port is 33100. This is used for all paging zones. The port number has a valid range of 1 to 65535.

Configuring Page Zones

To add a new page zone, press the 'Add' button. On the form that displays enter a unique description. (To override the automatically generated multicast address at this time, tick the Advanced Options and enter an IP multicast address E.g. 244.0.0.3).

Once a page zone has been created, the 'Apply' icon can be used to select the phones that will be configured to listen on this address/port combination.

 Pages cannot be sent to hot desk phones, only fixed phones. Pages will still be sent to a base phone if a hot desk phone is logged in.

At this point, the page zone has been created and the phones have been configured to listen. A page button will need to be added to the keymap of any handset required to make pages. Please refer to the [Keymap](#)

[Profile](#) section for more information.

 Page Zones can not be sent to the handsets in real-time. Handsets will not start to receive pages until they are rebooted. If a Handset is removed from a page zone it will continue to receive pages on that zone until it is rebooted.

 A Handset can be in a maximum of 5 Page Zones. If it is added to more than 5 Page Zones, additional zones will be ignored.

3.12 6900 External Paging

SIP paging units made by Algo can be used in conjunction with the MiVoice Office 250 and 6900 series phones to provide external paging features.

 The Algo 8180 G2 has been tested with both scenarios listed below. Other Algo devices should work with where applicable, please refer to Algo documentation for more information.

The table below outlines the two different modes in which Algo SIP Paging Units can be used.

Option	Description
Master/Sender Mode	Connect the SIP paging unit as a SIP device on the MiVoice Office 250. • Allows MiNET/Digital/69xx phones to page externally • Allows MiNET/Digital phones to page 69xx phones
Slave/Receiver Mode	Configure the SIP paging unit to receive MultiCast pages directly from 6900 series phones • Allows 69xx phones to page externally without the need for an additional SIP licence on the phone system.

Configuring a Slave/Receiver to Receive Direct 6900 Pages

The following instructions explain how to configure an Algo SIP Unit to receive direct pages from a 6900 series phone. The instructions assume that the unit has already been configured on the customer's network and that you have access to the unit's web configuration interface.

 Please refer to Algo documentation for information on configuring IP settings and accessing the Web configuration interface.

Enable Slave/Receiver Mode

Navigate to the 'Basic Settings\Multicast' tab and enable the following options:

- Multicast Mode -> Slave/Receiver
- Multicast Type -> Regular (RTP)
- Basic Slave Zones -> Select the zones the unit should listen to pages in

The screenshot shows the 'Multicast Settings' page in the ALGO 8180G2 SIP Audio Alerter Control Panel. The page has a navigation bar with 'Status', 'Basic Settings', 'Additional Features', 'Advanced Settings', 'System', and 'Logout'. The 'Multicast' tab is selected. The settings are organized into sections: 'Multicast Mode' with options for 'Multicast Mode' (None, Master/Sender, Slave/Receiver) and 'Multicast Type' (Regular (RTP), Polycom Group Page, Polycom Push-to-Talk); 'Number of Zones' with options for 'Basic Zones Only' and 'Basic and Expanded Zones'; and 'Slave/Receiver Zone Settings' with checkboxes for 'Priority Call', 'All Call', 'Music', and 'Zone 1' through 'Zone 6'. A 'Save' button is at the bottom right.

Advanced Multicast Settings

The page zones now need to be configured to match those on the MiVoice Office Application Suite. Browse to the 'Configuration\Features\6900 Handsets\Paging' section of the MiVoice Office Application Suite website and enable 'Show/hide Advanced Options'.

Make a note of the base paging address and port. The first page zone will use this port with each additional zone using the next sequential port number.

The screenshot shows the 'Paging' configuration page in the MiVoice Office Application Suite. The left sidebar contains a tree view with 'Sites' expanded, showing 'Default site' and 'Features'. Under 'Features', 'Paging' is selected. The main content area has a title 'Paging' and a description: 'Configure multicast page zones that can be used by handsets. Once a page zone has been created, you can add a Page Zone'. Below this is a table with columns 'Page Zone Name', 'Phones on Page Zone', 'Edit', 'Delete', and 'Assign'. The table contains one row: 'Page Zone 1' with '1' phones. Below the table are 'Add', 'Edit', and 'Delete' buttons. A 'Show/Hide Advanced Options' checkbox is checked. Below this are two text input fields: 'Base Paging Multicast Address' with the value '224.0.2.59' and 'Base Paging Port' with the value '50002'. A 'Save' button is at the bottom.

If you have overridden the address/port combination against any individual page zone this will need to be taken into account.

On the Algo unit's web configuration page, navigate to 'Advanced Settings\Advanced Multicast'. Update the base address and port to match those of the MiVoice Office Application Suite.

ALGO 8180G2 SIP Audio Alerter Control Panel Firmware: 1.6.4

Navigation: Status | Basic Settings | Additional Features | **Advanced Settings** | System | Logout

Sub-navigation: Network | Admin | Time | Provisioning | Tones | File Manager | Advanced Audio | Advanced SIP | **Advanced Multicast**

Advanced Multicast Settings

Current multicast mode: Slave
Multicast mode can be set in "Basic Settings > Multicast"

Slave Settings

Audio Sync (milliseconds, 0 ~ 1000)

When using multicast with other third-party devices that have a delay in their audio path, the audio on the 8180G2 may be heard slightly earlier than on these other devices. Use this feature to add a small delay to the audio output on the 8180G2 in order to synchronize with these other devices. Applies to Multicast Slave mode only.

Basic Zone Definition

If using an Algo device as a Multicast master, it is recommended to set the slave tones to "None" to avoid conflicts, as the Algo devices already multicast a tone by default.

Zone	IP Address and Port	Page Tone	Page Volume
Priority Call (DTMF:9)	224.0.2.60:50000	<None>	<Use Default Page Volume>
All Call (DTMF:0/8)	224.0.2.60:50001	<None>	<Use Default Page Volume>
Zone 1 (DTMF:1)	224.0.2.60:50002	<None>	<Use Default Page Volume>
Zone 2 (DTMF:2)	224.0.2.60:50003	<None>	<Use Default Page Volume>
Zone 3 (DTMF:3)	224.0.2.60:50004	<None>	<Use Default Page Volume>
Zone 4 (DTMF:4)	224.0.2.60:50005	<None>	<Use Default Page Volume>
Zone 5 (DTMF:5)	224.0.2.60:50006	<None>	<Use Default Page Volume>
Zone 6 (DTMF:6)	224.0.2.60:50007	<None>	<Use Default Page Volume>
Music (DTMF:7)	224.0.2.60:50008	<None>	<Use Default Page Volume>

Once configured, 6900 series phones will be able to page directly to the Algo SIP paging unit over multicast, with no SIP registration with the MiVoice Office 250 necessary.

Configuring a Master/Sender to Receive Pages Via SIP

The Algo unit can be configured with a 'Base/Page' extension which is a SIP extension on the telephone system. The paging device will register as this extension and will automatically answer any SIP call presented, sending the audio out via the attached speaker.

ALGO
8180G2 SIP Audio Alerter Control Panel
Firmware: 1.6.4

Status
Basic Settings
Additional Features
Advanced Settings
System
Logout

SIP
Features
Multicast

SIP Settings

SIP

ⓘ This section allows the SIP server information & account credentials to be entered. This information should be obtained from your telephone system administrator or hosted account provider. After saving these settings, see the [Status](#) tab to confirm successful registration.

SIP Domain (Proxy Server)

ⓘ Default port is 5060. To specify a different port, enter PROXY:PORT, e.g. my_proxy.com:5070, or 192.168.1.10:5080.

Ring/Alert Mode

☐ Monitor "Ring" event on registered SIP extension

☐ Use "Subscribe/Notify" dialog event (RFC 4235) to monitor event on different extension

☐ Use "Subscribe/Notify" presence event (RFC 3856/3863 PIDF) to monitor event on different extension

☒ None

Base/Page Extension

Authentication ID

Authentication Password

ⓘ The device will auto-answer any inbound call received on this extension and provide a voice paging path (and multicast if configured).

Save

To use the paging unit in this manner, create a SIP extension on the telephone system and then add the extension, authentication ID and password on the 'Basic Settings\SIP' tab.

To send the page to other Algo device or 6900 phones, enable 'Master/Sender' on the 'Basic Settings\Multicast' tab then select the zone which the device should distribute pages to.

ALGO
8180G2 SIP Audio Alerter Control Panel
Firmware: 1.6.4

Status
Basic Settings
Additional Features
Advanced Settings
System
Logout

SIP
Features
Multicast

Multicast Settings

Multicast Mode

Multicast Mode
☐ None
 ☒ Master/Sender
 ☐ Slave/Receiver

ⓘ Multicast Zone Definitions can be found in "Advanced Settings > [Advanced Multicast](#)".

Multicast Type
☒ Regular (RTP)
 ☐ Polycom Group Page
 ☐ Polycom Push-to-Talk
 ☐ Regular RTP + Polycom Group Page
 ☐ Regular RTP + Polycom Push-to-Talk

ⓘ Regular mode uses RTP audio packets compatible with all Algo SIP endpoints, and most multicast-enabled phones.
 ⓘ Both "RTP + Polycom" multicast types will enable local speaker playback for all groups and zones.

Number of Zones
☒ Basic Zones Only
 ☐ Basic and Expanded Zones

Master/Sender Zone Settings

Zone Selection Mode
☐ DTMF Selectable Zone
 ☒ Single Zone

ⓘ For additional capabilities allowing unique SIP extensions per zone, see "Additional Features > [More Page Extensions](#)".

Master Single Zone
 Zone 1 ▼

ⓘ If "DTMF Selectable Zone" is selected above, then this single zone setting will not apply to Paging (since the zone can now be dynamically selected per call using DTMF), but it will still apply to the Ring Extension and Relay triggered events.

Speaker Playback Zones
☒ Priority Call
 ☒ All Call
 ☒ Music
☒ Zone 1
 ☒ Zone 2
 ☒ Zone 3
☒ Zone 4
 ☒ Zone 5
 ☒ Zone 6

ⓘ Allows master device to play audio for selected zones only. This is useful if using DTMF Selectable Zone mode (or [More Page Extensions](#) per zone) and wishing to make the Master unit a member of only certain zones.

Save

Using a paging unit in 'Master/Slave' mode allows MiNET/Digital extension to page 69xx units.

 A Cat-F licence will be required on the MiVoice Office 250 for each SIP extension the Algo Units registers with.

4 6900 Installation Check List

The following table contains all the installation tasks which should have been completed before starting to rollout a 6900 phone.

Category	Item	Description	Completed?
Installed Version Check	MiVoice Office 250	Minimum version required for 6900 support is 6.3 SP2.	
	MiVoice Office Application Suite	Minimum version required for 6900 support is 5.1 SP2.	
	Mitel CT Gateway	Minimum version required for 6900 support is 5.0.64.	
MiVoice Office 250 Programming	Encryption Password	Has the 'App Suite Server Configuration - Encryption Password' been configured against each node on the network?	
	6900 Phones	Have the 6900 devices been created on the system (including any for SIP Hot Desking)? Note: Do not use standard SIP devices or the Handsfree Intercom feature will not work.	
Networking	DHCP Option 66	Has the DHCP option 66 been added to update the 6900 phone's 'Configuration Server' to be the IP Address/DNS address of the MiVoice Office Application Suite server?	
	DHCP Options 43 & 125	Have the MiNET options 43 & 125 been removed from the DHCP server (use 128, 129 & 130 instead for supporting non-6900 MiNET phones)?	
MiVoice Office Application Suite Configuration	Client Location	Have the Local and Remote client location IP address or DNS names been set on the MiVoice Office Application Suite?	
	Node IP Addresses	Have the Node IP address and SIP connection details been configured on the MiVoice Office Application Suite?	
	Encryption Passwords	Have the Encryption Passwords for each node been configured on the MiVoice Office Application Suite?	

5 6900 Handset Diagnostics & Troubleshooting

Diagnostics

There may be circumstances where Mitel support ask for information to diagnose faults. The service that handles 6900 handset requests will log to the normal Mitel Communication Service destinations. Logging for this service is enabled/disabled using the standard 'Logging' settings within the MCS website.

For more information, please refer to the [Logging](#) section of the Technical Manual.

Handset Log Retrieval

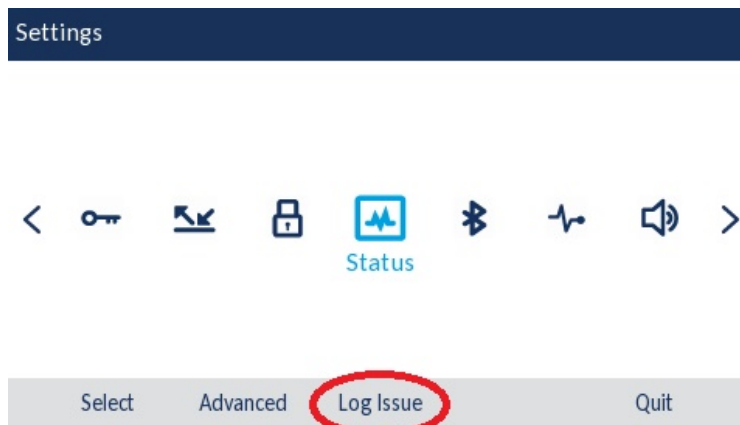
In addition to the logs generated by the server, each handset can be configured to upload diagnostic information to the server.

To enable logging for a specific handset, navigate to the 'Configuration\Features\6900 Handsets\Phones' page of the MCS website. Select the specific handset and press the 'Edit' button. Check the 'Enable Diagnostics' button then press 'Save'.

For information on enabling this feature on a handset centrally, please refer to the [6900 Handset Phones](#) section of the Technical Manual for more information. Ensure that the Syslog IP address has been configured for the 6900 Handsets in the [External Addresses](#) section of the MCS website.

Once logging has been enabled and any problem/fault has been recreated on the phone, the logs can be downloaded from the phone using the following steps:

1. Press the settings  button on the phone and the press the 'Log Issue' button:



 After pressing the 'Log Issue' button, a message will appear on the phone stating 'Please wait...'. Wait until this message disappears before continuing to the next step.

 The 'Log Issue' button will not appear until 'Enable Diagnostics' has been configured against the phone on the MCS website.

2. Navigate to the website built into the phone using a web browser (default login - admin/22222). Load the 'Troubleshooting' page, then locate the 'Get Log Files' option within the 'Support Information' section. Press the 'Save As...' button to download the logs from the phone.

Support Information

Get local.cfg

Save As...

Get server.cfg

Save As...

Get Crash Log

Save As...

Get Log Files

Save As...

Show Task and Stack Status

Show

Troubleshooting

No Keymap After a Reboot

If there is also a 'No Service' message showing on the phone, address this issue first.

The most likely cause for this is incorrect configuration server details stored persistently in the phone. This can be checked by accessing the phone's Web UI (admin/22222) and browsing to the 'Troubleshooting' section. Press the 'Save As...' button next to the 'Get server.cfg' item and open the file in notepad or similar. Check the 'Created ...' date, this should be within 24 hours if the phones are still configured to auto update every night (which they are by default). Also, check the 'https server:' configuration item, it should be the DNS path for the server which is configured in the 'Client Locations' section of the MCS website.

To resolve the issue, either default the phone so that it picks the DHCP option 66 or mDNS settings again to find the MCS again or manually retype the configuration server settings in the 'Configuration Server' area of the website:

Download Protocol: HTTPS

HTTPS Server: [external IP Address or external DNS name of the MCS server]

HTTPS Port: 8202



If the configuration server details already show this but the server.cfg show different details, the phone is currently using updated details passed through XML but has not stored them locally so is losing them on reboot. To update the Configuration Server details, change the 'HTTPS Server' through the Web UI to something different, then change it back again. This will force the phone to store them locally.

No Service showing on the Handset

The most likely cause of this is incorrect match of the device username and password set on the PBX and in the MCS.

This can be identified in the HandsetLifeCycle logs for the handset stored by default in
%ProgramData%\Mitel\Mitel Communication Service\Logs\Handset\logs\Handsets\<MAC of handset>

You will see a line similar to below:

```
14:34:20.7933 : [ 35] RequestController : SIP Registration Regevent registration params
SipUsername=14638,DisplayName=14638,SipAuthName=14638,ProxyUrl=192.168.106.1,ActiveProxy=192.168.106.1,IncomingName=Unknown
Name,LocalIp=172.19.23.7,CallDuration=0,RegistrationCode=401,LineIndex=0,RegistrationState=REFUSED
```

The highlighted items show a 401 error indicating authorization failure.

Ensure the SIP username and password match for the device between the PBX and MCS device configuration. Changes will be pushed to the handset and it should connect.



Remember that when connecting any SIP device to the MiVoice Office 250, the 'SIP UDP Listening Port' must be enabled in the 'Advanced IP Settings' section. Currently a reboot of the phone system is required after enabling this.

Remote Handset Connections

For remote handsets, ensure the following items have been checked:

- MiVoice Office 250
 - SIP extension has configured Use Registered Username and NAT address type set to Native (see 6900 Handset SIP configuration - MiVoice Office 250 section in the Online help or Technical Manual)
 - Check the Set-Side username and password matches the SIP device username and password
- MCS Server
 - Local and remote nodes are configured and the external address of the MBG is in the NAT IP Address section (see 6900 Handset SIP configuration - Node IP Addressing section in the Online help or Technical Manual)
 - Check the Set-Side username and password matches the SIP device username and password in the MCS server (see 6900 Handset SIP configuration - SIP Device Authentication section in the Online help or Technical Manual)
 - Check the ICP-side username and password match the SIP device username and password on the MiVoice Office 250



If any changes have been made to a remote configuration while the handset has 'No Service' then the change will only take effect once you reboot the handset

Web Recovery Mode

When deploying a 6900 handset or after an environmental change you may find your 6900 handset using SIP firmware is in an unrecoverable state. This can occur if the handset fails to receive a DHCP address or it is in a constant loop and the settings of the handset or the web interface cannot be accessed.

Once the phone has rebooted and failed 10 times in a row it will put itself into 'Web Recovery Mode'.



To manually place a handset into 'Web Recovery Mode', hold down '1' and '#' while the handset is booting up.

The handset should have an IP address which should be displayed on the screen. Navigate to the phone's web interface using the IP address provided.

Once on the website, select 'Upgrade Software' from the left-hand menu. You will need to browse to the relevant model number's '.st' file (there is a specific file for each model type) and upgrade the phone to the chosen firmware version.

The '.st' files are available from the MiVoice Application Suite server in the following location:

'C:\ProgramData\Mitel\Mitel Communication Service\Net Store\Firmware\[Version]'

Once complete, the phone will reboot and will be back to default.

 If using a Bluetooth handset or other peripherals, these will now need to be re-paired.

Manual Firmware Updates

This section outlines how to perform a manual firmware update on a 6900 handset.

 This should be used to update a 6900 Handset from MiNET to SIP Firmware at a remote location followed by configuring the Configuration Server described in the next section

Requirements:

- A TFTP server (such as TFTP64)
- The .st firmware file for the model handset you require (These are available from the MCS server %ProgramData%\Mitel\Mitel Communication Service\Net Store\Firmware)
- IP address of the 6900 Handset (use settings and status on handset to find this)
- IP address of the TFTP server

Steps:

1. Place .st firmware file in the TFTP server directory (ensure that no MiNET files are present).
2. Press the gear button to go into setting on the 6900 phone.
3. Select 'Advanced' settings (default password is 73738)
4. Select 'Network' then 'Static' settings.
5. Enter the IP address of the TFTP server then press 'Save' (ensure your IP phone has an IP address assigned)
6. Select 'Restart' from the base settings screen on the phone.
7. The IP phone will attempt to download MiNET firmware on boot up. When not found, it will subsequently download the SIP firmware.

Remote Phone Configuration for SIP Firmware

If you have just updated the firmware on a remote 6900 Handset from MiNET to SIP then complete the following steps to enable the phone to connect to the MCS server

- Add the configuration server details on the phone manually, press the settings button (⚙) on the handset, then press the 'Advanced' key along the bottom of the screen.
- At this point you will be prompted to enter the administrator password. The default SIP password is '22222'.
- Once the password has been accepted, use the navigation keys (D-pad) or touch screen on 6940 & 6970 to navigate to the 'Configuration Server' section.
- Populate the following entries:

*Download Protocol: HTTPS
HTTPS Server: [external IP Address or external DNS name of the MCS server]
HTTPS Port: 8202
Cert Validation: false*

- Press 'Save' and then reboot the handset.

After a reboot, the phone will connect to the MCS server. The MCS server will provide over HTTPS connection information and will then tell the phone to reboot.

 It is possible that there will be more than one reboot at this stage as the firmware update is completed

The handset should now be registered with the MCS server.

The Phone will have Line 1 and Line 2 showing as the Top Sofkeys.

Unlike for local deployment as 'Setup' button will not be available and the SIP extension needs to be configured in the MCS server.

The 'Phones' page within the configuration section of the MCS website can be used to view whether the handset has been identified. The MCS uses the handsets MAC Address to uniquely identify it.

Restoring MiNET Firmware

If required, the firmware on the handset can be restored to a MiNET version.

Requirements:

- A TFTP server (such as TFTP64)
- The MiNET firmware file for the model handset you require
- IP address of the 6900 Handset (use settings and status on handset to find this)
- IP address of the TFTP server

Steps:

1. Place the MiNET firmware file in the TFTP server directory
2. Start the TFTP server
3. Using a web browser, connect to the web page of the phone using the IP address of the phone (Default Username - 'root', Default Password - '73738' on MiNET firmware and Default Username - 'admin', Default Password '22222' on SIP firmware)
4. Select 'Reset' then select 'Restore To Factory Defaults'. At this point the phone will reboot. When it has completed it's reboot, repeat step 3 to log back into the website.
5. Select 'Firmware Update', complete the form:
 - Enter the name of the firmware file in the filename box
 - Enter the IP address of the TFTP server in the Server box
 - Select Download firmware
 - Firmware will download and then the phone reboot (note this can take several minutes)



It is important to factory default the phone before restoring to MiNET firmware to avoid complications.

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