

**MiCollab Advanced Messaging  
Avaya Communication Server 1000  
SIP Trunk with Session Manager  
Integration Technical Note**

For version 6.1 and above

## Notice

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# Preface

This Integration Technical Note (ITN) is written for technicians who are experienced with MiCollab Advanced Messaging (MiCollab AM) and are familiar with its procedures and terminology. This document also assumes that you are familiar with the features and programming of the Avaya Communication Server 1000E telephone system.

This document describes how to integrate MiCollab AM with a Avaya Communication Server 1000E telephone system, using the Session Initiation Protocol (SIP) integration. This integration operates exclusively over a TCP/IP-based network; it uses no analog or digital voice telephony ports, but passes voice communication and signaling information over the network.

The Avaya SIP Trunk integration consists of three major components: the Avaya Call Server (Avaya Communication Server 1000E telephone system), the Avaya Aura® Session Manager, and MiCollab AM. MiCollab AM uses SIP trunks to integrate with the switch and thus uses static SIP endpoints.

The SIP trunks are configured as virtual trunks on the Communication Server 1000E and the corresponding SIP gateway endpoints are configured on the Session Manager. The Session Manager routes all calls originating on the switch to MiCollab AM.

Similarly, the Session Manager routes all of the outgoing calls and MWI requests from MiCollab AM to their proper destinations.

## References

A catalog of technical documentation is included on the MiCollab AM Installation Media. If you are installing any advanced applications, such as Networking and Fax Server applications, you should refer to the appropriate technical documentation for application and installation information.

## Documentation

The technical documentation is produced in the PDF format and requires the PDF reader to view it. The documentation set for this MiCollab AM includes the following documents and resources:

- **Developer Resources.** Contains programming guides and API references for developers for integrating the server clients and web applications with MiCollab AM.
- **Integration Technical Notes (ITN).** Contains a set of guides that describe the integration methods and instructions for a variety of phone systems to work with MiCollab AM. The ITNs are generally used by resellers or administrators who are experienced with MiCollab AM and familiar with the integration procedures and terminology.
- **Quick Reference Card (QRC).** Contains shortcuts and quick instructions telling subscribers how to access and use the messaging system.
- **Server Documentation.** Available as a PDF only. Contains administrative guides for administrators about installing, configuring, and administering the messaging system, and user guides for subscribers about accessing the messaging system and checking and sending messages.

- **Spare Parts Documentation.** Contains a set of guides that describe the instructions for installing and configuring hardware parts to work with MiCollab AM. These documents are written for Mitel certified MiCollab AM technicians who are experienced with MiCollab AM and familiar with the procedures and terminology.
- **Software Release Notice (SRN).** This notice introduces the new features, capabilities, and hardware/software requirements for the corresponding MiCollab AM version.

## Documentation Updates

Documentation updates may be available from the following sources:

- Mitel certified technicians can view or download the latest/updated documents and program files from our partner web site: [connect.mitel.com/connect](https://connect.mitel.com/connect)

## Help

The primary source of information about MiCollab AM is the online help available within any of its administrative utilities. You can access **Help** as follows:

- Click the **Help** button in the dialog box or window in which you are working
- Press the **F1** key at any time.

## Document Conventions

The following conventions are used in this document:

- **Key Names.** Names of keys on the keyboard are shown in a box.

Example: **Enter**

When two keys must be pressed simultaneously, they are joined by a + sign.

Example: **Alt** + **Tab**

- **Reference to Document.** *Italics* fonts can also signify the titles of other documents.

Example: Refer to *System Installation Guide*.

- **UI Element Names.** Names of UI elements such as dialog windows, screens, menu items, tabs, buttons, icons, etc. are shown in bold.

Example: On the **Startup** screen, click the **Start** icon.

- **User Input.** Information required to be typed is shown in italics.

Example: Type the password *voicemail*.

- **Warning, Caution, Important, and Notes.** Text for the contents that require attention are shown as follows:

**WARNING** A warning paragraph advises you of circumstances that can result in the loss of data, harm to the system server platform, or personal harm.

**CAUTION** Failure to follow these recommendations can result in unauthorized access to the system and consequent loss of data.

**IMPORTANT** An important paragraph gives decision-making information or informs you of the order in which tasks need to be completed.

**NOTE** A note gives additional information, provides an explanation, or indicates an exception to the information in the preceding text.

## Features Supported by This Integration

The following tables list the features supported using the Communication Server 1000E SIP Trunk integration.

Table 1. Call forward to personal greeting support for these common call types

Divert to MiCollab AM on	Supported
No Answer	Yes
Busy	Yes
Forward All	Yes
Do Not Disturb	No

Table 2. Integration features supported for Avaya Communication Server 1000 SIP Trunk with Session Manager

Feature	Supported	Notes
Automatic subscriber logon	Yes	
ANI/CLI	Yes	
Announce Busy greeting on forwarded calls	Yes	
Call screening	Yes	Note 1
Caller queuing	Yes	Note 1, 2

DNIS	Yes	
End-to-end DTMF, attendant console	Yes	
End-to-end DTMF, proprietary telephones	Yes	
Fax Tone Detection	Yes	
Internal calling party ID for reply	Yes	
Live record, integrated	No	
Live reply to sender	Yes	
Message notification callouts	Yes	
MWI, set/clear	Yes	
MWI, inband/outband	Outband	
Networking, analog	No	
Overflow from MiCollab AM to attendant	Yes	
Overflow to MiCollab AM from attendant	Yes	
PBX-provided disconnect signaling	Yes	
Revert to operator	Yes	
S RTP	Yes	Note 5
TLS	Yes	Note 5
Transfers, blind	Yes	Note 3
Transfers, confirmed	Yes	Note 3
Transfers, fully supervised	Yes	Note 3
Transfers, monitored	Yes	Note 3
Trunk ID for call routing	No	
Spans Multiple Call Servers	Yes	Note 4
Multiple Integrations	Yes	Note 6



## NOTES

1. Only available using supervised transfers
2. Caller Queuing is specific to each local Call Server. Call Servers within the system are unaware of queued calls to the same subscriber on other Call Servers. For more information, refer to the [Critical Application Considerations](#) section of this document for limitations on these features.
3. Refer to the [Critical Application Considerations](#) section of this document for limitations on these features.
4. Multiple call servers can be used via the SIP Routing Manager.
5. MiCollab AM supports negotiation for SRTP media streams using the Secure RTP profile defined in RFC 3711 with the offer/answer model defined in RFC 3264. To enable SRTP, RTP, or both, see integration configuration options documentation for the switch. The default setting is RTP. Please note that MiCollab AM doesn't support RFC 5939 which is an extension of RFC 3264. Also, please note that SRTP has not been qualified for this integration, and no switch programming is available for setting up SRTP on the switch side. However SRTP may be enabled as described above, and technical support will be available on a best effort basis.
6. Refer to the [Critical Application Considerations](#) section.

# Critical Application Considerations

Known limitations or conditions within the telephone system and MiCollab AM that affect the integration performance are listed here. General recommendations are provided when ways to avoid these limitations exist.

- You must configure the Incoming Hunt Mode in the Switch Section Options dialog box. This integration supports terminal, circular, reverse terminal and reverse circular hunt modes only. The default mode is Terminal.
- You must configure the Trunk Group Access Code in the Switch Section Options dialog box. This code cannot conflict with extensions. This must match with the DN prefix programmed for the Gateway Endpoint associated with this MiCollab AM Call Server.
- The SIP Domain Name in the Integration Options dialog box must match the domain name configured on the Session Manager. This is a case sensitive value.
- The primary (public) network interface card (NIC) must be the first network connection in the network binding order. MiCollab AM binds and communicates to other servers and subscribers on this network connection. On a MiCollab AM server with two or more NICs, the NIC that supports this integration must not occupy first place in the operating system's binding order. For more information, refer to [Changing the Network Binding Order on the MiCollab AM Platform](#) later in this document.
- MiCollab AM supports G.729a with support for annex b on the incoming audio stream only. MiCollab AM does not transmit annex b packets.
- When codec negotiation takes place between MiCollab AM and the PBX, MiCollab AM always offers the G.729a audio format as an option. You may configure G.729a as the preferred codec in MiCollab AM; however, the decision whether to use G.729a is always made by the PBX.
- The Call Queuing feature does not transcend the Call Server. Calls may be queued on multiple Call Servers for the same subscriber but Call Servers do not have knowledge of calls in the queue on other Call Servers within the system. Callers may be prompted with specific information about their place in the queue; however, the information pertains to the specific Call Server on which their call is queued.
- The MiCollab AM Integration Options parameter, Validate Remote Hosts for Media validates each incoming audio packet and accepts it only if it is sent from a valid endpoint. The parameter is disabled by default. Enabling this parameter causes MiCollab AM to reject RTP packets from invalid endpoints, rejects MWI packets that timeout after a specified number of times, and overcomes port lockups when callers hang up while MiCollab AM is performing a blind transfer. See the Mitel Technical Bulletin #42285 for more information on this parameter.

**NOTE** Enabling this parameter causes processing overhead, enable it only when it is necessary.

- If a different configuration of additional hardware is present, which can include G430 and G450 Media gateways. The Session Manager will manage the DSP resources of the media gateways for SRTP/TLS media encryption.

- MiCollab AM 6.1 supports up to 10 integration types (i.e. licensed integrations) in total per system. However, the following limitations apply to each Call Server:
  - Limited to 3 integration types per Call Server
  - The 3 integration types can be any mix of TDM and SIP (e.g. 1 TDM and 2 SIP)
  - Limited to 1 Mitel MiTAI or 1 Cisco UCM SCCP IP integration. Can be mixed with TDM, but not with SIP
  - Connect up to 10 telephone systems total per Call Server (e.g. 2 Avaya Communication Manager systems using SIP + 5 Avaya IP Office systems using SIP + 3 Siemens HiPath 4000 systems using Station Set Emulation)
  - SIP timers for Aastra EETS integrations are incompatible with other SIP integrations. Thus, it is not possible to have an EETS integration with any other SIP integration on the Call Server

# Installation Requirements

Review the following information before performing any of the procedures in this document. To install this integration successfully, you must meet the installation requirements for both the telephone system and MiCollab AM.

## Telephone System Requirements for Communication Server 1000E Release 7.6

- Avaya Communication Server 1000E version 4021, release 7.6 and a SIP Access Port License for each MiCollab AM port
- Avaya Aura Session Manager 7.0 or prior
- MC-8, NTVQ01AB VMG Voice Media Gateway Card 8-port assembly; or MC-32, NTVQ01BB VMG Voice Media Gateway Card 32-port Assembly; or MGC, NTDW60BAE5 Media Gateway Controller
- Premium Network Service Package

## MiCollab AM Requirements

- MiCollab AM version 6.1
- Mitel software key diskette or feature file with the Avaya Communication Server 1000E SIP Trunk integration enabled and one RADVISION® SIP and RTP license enabled for each port involved in the integration.
- One or two 10 MB, 100 MB, or 1000 MB (gigabit) network interface cards with cables

# Programming the Telephone System

This section provides commands programming examples to program the telephone system for integration with MiCollab AM. The programming in this section can also be done in the System Manager UCM page. To program through UCM page, see [Appendix A](#).

Follow the recommendations and programming examples in this section to program the telephone system for integration with MiCollab AM. Programming examples show commands and parameters that are necessary for integration; they do not represent PBX programming in its entirety.

The installing technician should be familiar with programming the telephone system. For detailed information on programming and installing the telephone system, refer to the *Integrated Services Network, Features Description & Operation Manual*. The Northern Telecom Practice (NTP) library also provides additional information.

## Programming the D-Channel

Configure the D-channel as D-channel over IP Card Type and set the remote capabilities to Network Name Display method 2. In addition, set the MWI using a SIP NOTIFY message. Table 3 provides an example of D-channel programming in LD 17.

Table 3. Example programming for the D Channel (LD 17)

REQ	CHG
TYPE	ADAN
ADAN	NEW DCH 4
CTYP	DCIP
DES	SIP_PRI
IFC	SL1
RLS	4
RCAP	ND2 MWI TAT

## Programming the Trunk Route

Configure the SIP route, which contains the virtual trunks that act as MiCollab AM lines. Define the protocol ID as SIP and associate the route with the D-channel configured in the previous step. The Node ID should match the node of the Signaling Server. Set the route as incoming and outgoing and assign an access code. To modify the parameters, use the CHG request. Table 4 provides an example of the configuration of route 4 in LD 16.

Table 4. Example programming for the trunk route (LD 16)

REQ	NEW	Comments
TYPE	RDB	
CUST	0	
ROUT	<b>4</b>	
DES	SIP_PRI	
TKTP	TIE	
VTRK	YES	
ZONE	000	
PCID	<b>SIP</b>	
NODE	1110	
ISDN	YES	
MODE	ISLD	
DCH	4	
IFC	SL1	
PNI	00001	Private network identifier (you may change this according to your site's requirements)
NCNA	YES	
NCRD	YES	
TRO	YES	
ICOG	IAO	

TRMB	NO
ACOD	7003
CNTL	YES
NEDC	ETH
FEDC	ETH

## Programming SIP Trunks

Configure one IP Tie virtual trunk for each MiCollab AM port. Assign the trunks as route members of the route configured in the previous step. Set the Start Arrangement fields to **Immediate** and Supervision Required to **Yes**. Table 5 provides an example of trunk programming in LD 14.

Table 5. Example programming for SIP trunks (LD 14)

REQ	NEW 32 (No. of MiCollab AM lines)
TYPE	IPTI
TN	108 0 1 0
DES	SIP_PRI
CUST	0
RTMB	4 1
CHID	1
STRI	<b>IMM</b>
STRO	<b>IMM</b>
SUPN	<b>YES</b>
CLS	UNR DTN

## Programming the Route List Index

Configure the route list index so it is accessible by the virtual trunk route. Configure any additional site-specific networking features required for the integration. Table 6 provides an example of route list programming in LD 86.

Table 6. Example programming for the route list index (LD 86)

REQ	NEW
CUST	0
FEAT	RLB
RLI	4
ENTR	0
ROUT	4

## Programming the Steering Code

Configure a pilot directory number (PDN) to allow access to the route as if it were an extension, which passes all dialed digits. This number is a steering code that accesses the new route 4. The steering code and route are associated using a route list index which contains an available route to process the calls. Table 7 provides an example of steering code programming in LD 87.

Table 7. Example programming for a steering code (LD 87)

REQ	NEW
CUST	0
FEAT	CDP
TYPE	DSC
DSC	40
RLI	4



## Programming Customer Data Block (CDB) Network Data

Configure any site-specific network settings and options. Table 8 provides an example of network programming in LD 15.

Table 8. Example programming for the CDB network data (LD 15)

REQ	CHG	Comments
TYPE	NET_DATA	
CUST	0	
ISDN	YES	
PNI	1	Private network identifier (you may change this according to your site's requirements)

## Programming Subscriber Telephones

Program subscriber telephones to forward to MiCollab AM on busy and no answer conditions. In the FDN and Hunt field, specify the PDN previously configured. Class of service (CLS) must contain FNA and MWA; enable CFXA for All Call Forwarding. If desired, specify a key as the MWK when programming key assignments. Table 9 provides an example of a 2004P2 set with Key 16 assigned as the MWK function in LD 11.

**NOTE** If you use blind transfers, you must enable HTA. If you use monitored or supervised T-type transfers, do not enable HTA.

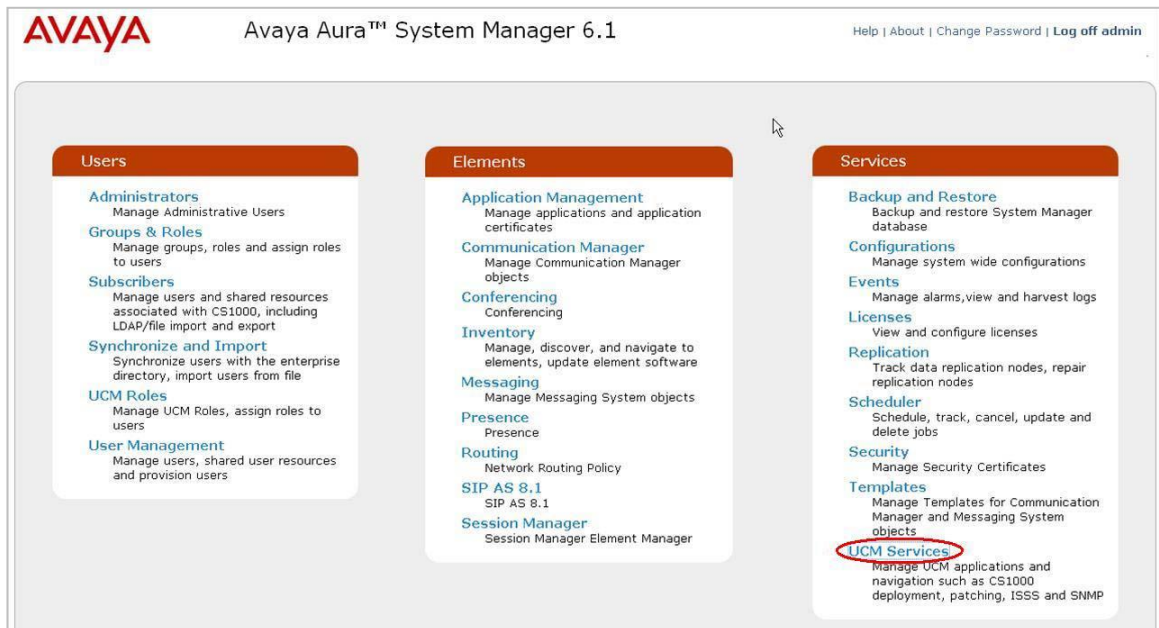
Table 9. Example programming for subscriber telephones (LD 11)

REQ	CHG
TYPE	2004P2
TN	112 0 1 0
FDN	4000
HUNT	4000
CLS	FNA HTA MWA CFXA
KEY	16 MWK 4000

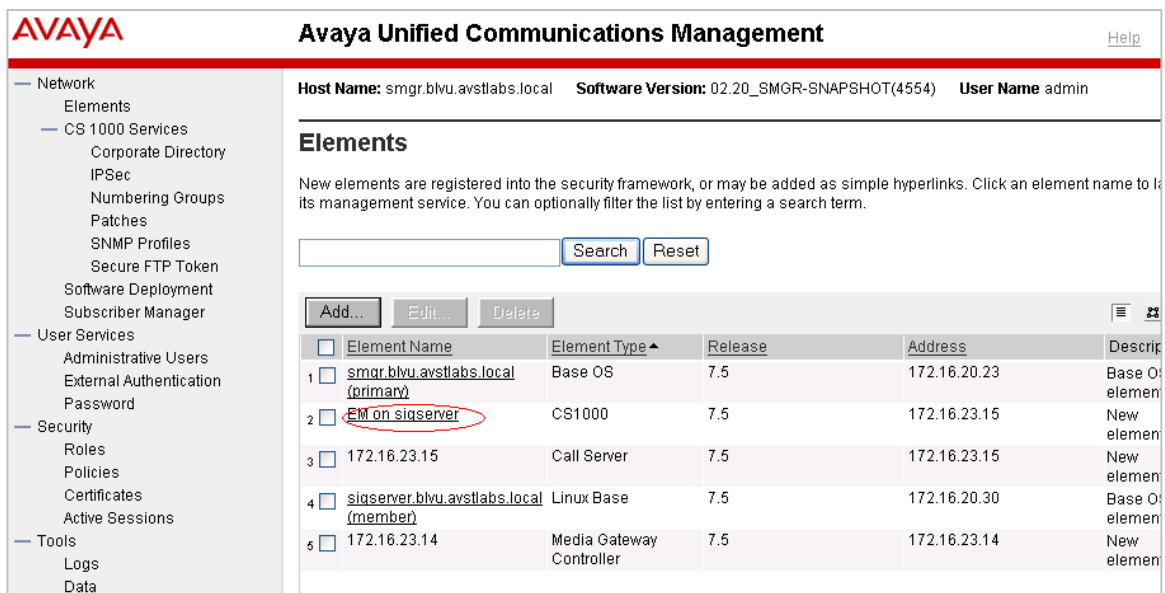
# Configuring the Element Manager

To configure the Element Manager:

- 1 Launch Communication Server 100E Unified Communications Manager.



- 2 Click on the Element Name corresponding to the Element Type of CS1000



- 3 Click **System > IP Network > Nodes: Servers, Media Cards**. The **IP Telephony Nodes** page displays.

- 4 Select the link for the Node ID to open the **Node Details** page.
- 5 Enter the Embedded LAN (ELAN) IP address and the Telephony LAN (TLAN) IP address as required.

**AVAYA CS1000 Element Manager**

Managing: 172.16.23.15 Username: admin  
System > IP Network > IP Telephony Nodes > Node Details

**Node Details (ID: 1110 - SIP Line, LTPS, Gateway ( SIPGw ))**

Node ID: 1110 \* (0-9999)

Call server IP address: 172.16.23.15 \* TLAN address type: ☒ IPv4 only ☐ IPv4 and IPv6

**Embedded LAN (ELAN)** Gateway IP address: 172.16.23.1 \* Subnet mask: 255.255.255.0 \*

**Telephony LAN (TLAN)** Node IPv4 address: 172.16.20.40 \* Subnet mask: 255.255.255.0 \*

Node IPv6 address:

\* Required Value. Save Cancel

**Associated Signaling Servers & Cards**

Select to add Add Remove Make Leader Print Refresh

Hostname	Type	Deployed Applications	ELAN IP	TLAN IPv4	Role
sigserver	Signaling_Server	LTPS, Gateway, PD, Presence Publisher, IP Media Services	172.16.23.16	172.16.20.30	Leader

Show: ☐ IPv6 address

Note: Only server(s) that are not part of any other IP telephony node and deployed application(s) that match the service(s) selected for this node are available in the servers list.

- 6 Click **Gateway**.

**AVAYA CS1000 Element Manager**

Managing: 172.16.23.15 Username: admin  
System > IP Network > IP Telephony Nodes > Node Details

**Node Details (ID: 1110 - SIP Line, LTPS, Gateway ( SIPGw ))**

Subnet mask: 255.255.255.0 \* Subnet mask: 255.255.255.0 \*

Node IPv6 address:

**IP Telephony Node Properties**

- Voice Gateway (VGW) and Codecs
- Quality of Service (QoS)
- LAN
- SNTP
- Numbering Zones
- MCDN Alternative Routing Treatment (MALT) Causes

**Applications (click to edit configuration)**

- SIP Line
- Terminal Proxy Server (TPS)
- Gateway (SIPGw)**
- Personal Directories (PD)
- Presence Publisher
- IP Media Services

\* Required Value. Save Cancel

**Associated Signaling Servers & Cards**

Select to add Add Remove Make Leader Print Refresh

Hostname	Type	Deployed Applications	ELAN IP	TLAN IPv4	Role
sigserver	Signaling_Server	LTPS, Gateway, PD, Presence Publisher, IP Media Services	172.16.23.16	172.16.20.30	Leader

Show: ☐ IPv6 address

Note: Only server(s) that are not part of any other IP telephony node and deployed application(s) that match the service(s) selected for this node are available in the servers list.

- 7 Under the General tab, select **Vtrk Gateway Application as SIP Gateway**, and then add the SIP domain name.

**AVAYA** **CS1000 Element Manager**

Managing: 172.16.23.15 Username: admin  
System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration

**Node ID: 1110 - Virtual Trunk Gateway Configuration Details**

General | SIP Gateway Settings | SIP Gateway Services

Vtrk gateway application: ☒ Enable gateway service on this node

**General**

Vtrk gateway application: SIP Gateway (SIPGw) \*  
SIP domain name: blvu.avstlabs.local \*  
Local SIP port: 5060 \* (1 - 65535)  
Gateway endpoint name: node1110 \*  
Gateway password: \*  
Application node ID: 1110 \* (0-9999)  
Enable failsafe NRS: ☐  
SIP ANAT: ☒ IPv4 ☐ IPv6

**Virtual Trunk Network Health Monitor**

☐ Monitor IP addresses (listed below)  
Information will be captured for below.  
Monitor IP:   
Monitor addresses:

8 On the same page, click the **SIP GW Settings** tab.

9 Under SIP GW Settings, type the Session Manager IP address in the Primary TLAN IP Address box.

**AVAYA** **CS1000 Element Manager**

Managing: 172.16.23.15 Username: admin  
System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration

**Node ID: 1110 - Virtual Trunk Gateway Configuration Details**

General | SIP Gateway Settings | SIP Gateway Services

**Proxy Server Route 1:**

Primary TLAN IP address: 172.16.20.11  
The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"  
Port: 5060 (1 - 65535)  
Transport protocol: TCP  
Options: ☐ Support registration  
☐ Primary CDS proxy  
Secondary TLAN IP address: 0.0.0.0  
The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"  
Port: 5060 (1 - 65535)  
Transport protocol: TCP  
Options: ☐ Support registration

\* Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved.

10 On the same page, enter the appropriate information in the SIP URI Map section.

Node ID: 1110 - Virtual Trunk Gateway Configuration Details

General

SIP Gateway Settings

SIP Gateway Services

SIP URI Map:

Public E.164 domain names

National:

Subscriber:

Special number:

PublicSpecial

Unknown:

PublicUnknown

Private domain names

UDP:

udp

CDP:

cdp.udp

Special number:

PrivateSpecial

Vacant number:

PrivateUnknown

Unknown:

UnknownUnknown

- 11 Click **Save and Transfer** to confirm your changes.

# Programming the Session Manager for Release 7.5 integration with MiCollab AM

This section provides the procedures for configuring Session Manager. It is assumed that the basic Session Manager installation and configuration has been completed. For further information on Session Manager, see the Mitel Avaya SIP Trunk ITN and Avaya Session Manager documents.

## Log in to Avaya Aura System Manager

Access the Avaya Aura System Manager used to manage Session Manager. Select Routing.

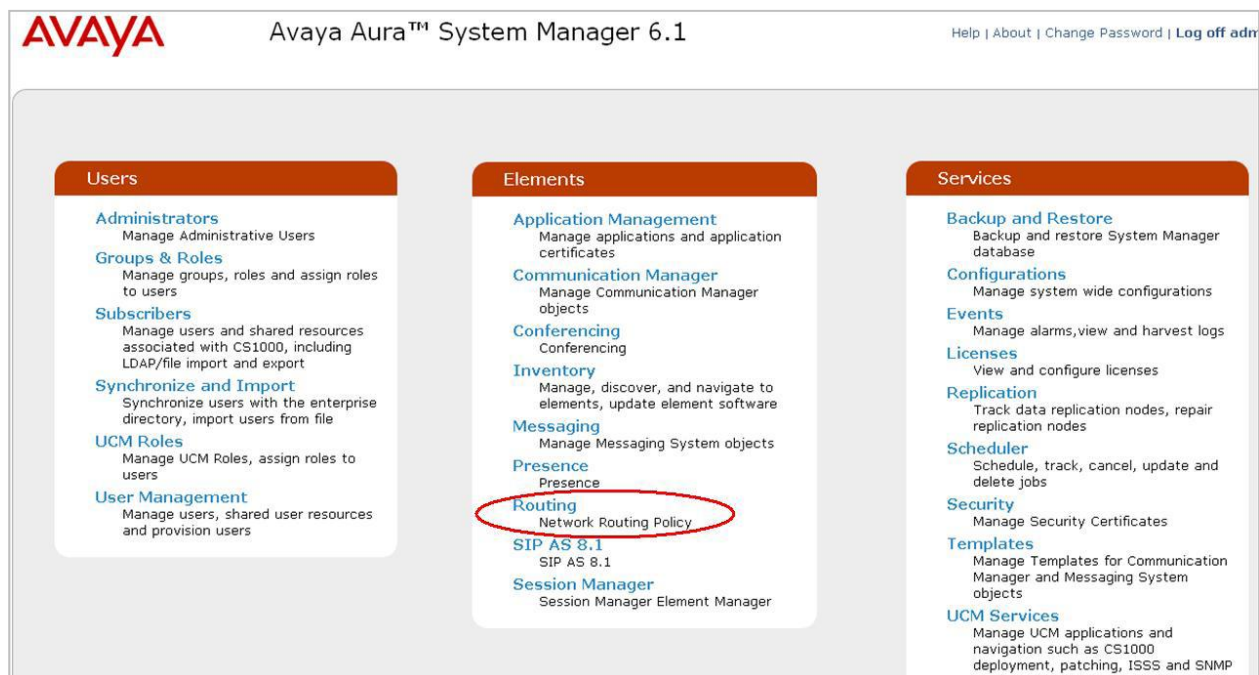


Figure 1. System Manager

## Obtaining SIP Domain

Read the SIP domain by selecting Domains on the left panel menu. This entry should be consistent with that configured both in the Communication Server 1000E and MiCollab AM.

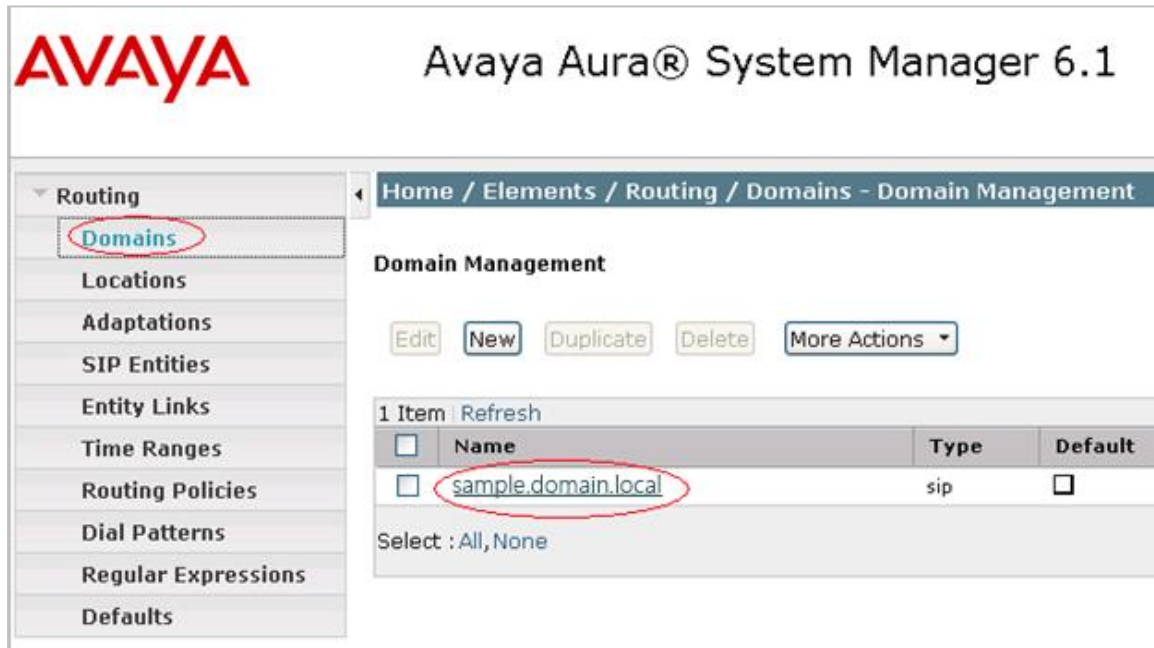


Figure 2. Avaya Aura® System Manager 6.1

## Configuring the SIP Firewall

Configure the SIP firewall and ensure the MiCollab AM server's IP address is not blocked by the Session Manager firewall.

To configure the Firewall:

- 1 On the System Manager Web Interface, select **Elements** > **Session Manager** > **Network Configuration**, and then select **SIP Firewall**.
- 2 Configure the firewall appropriately to allow access to the MiCollab AM server's IP address.

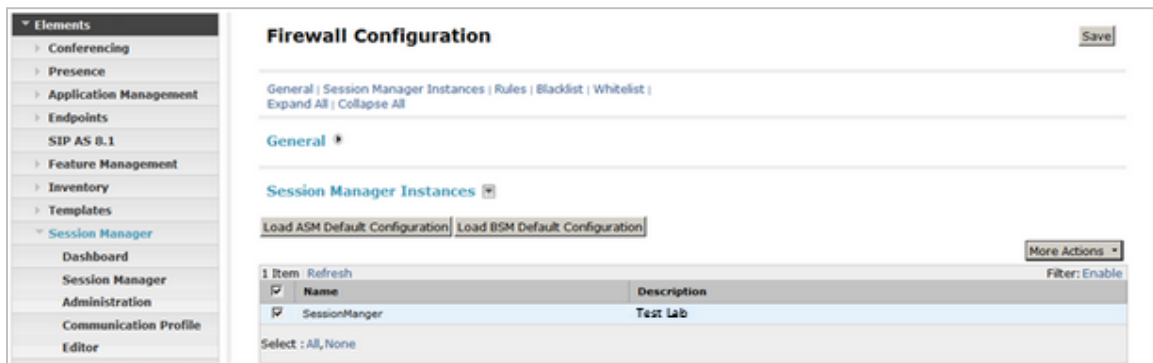
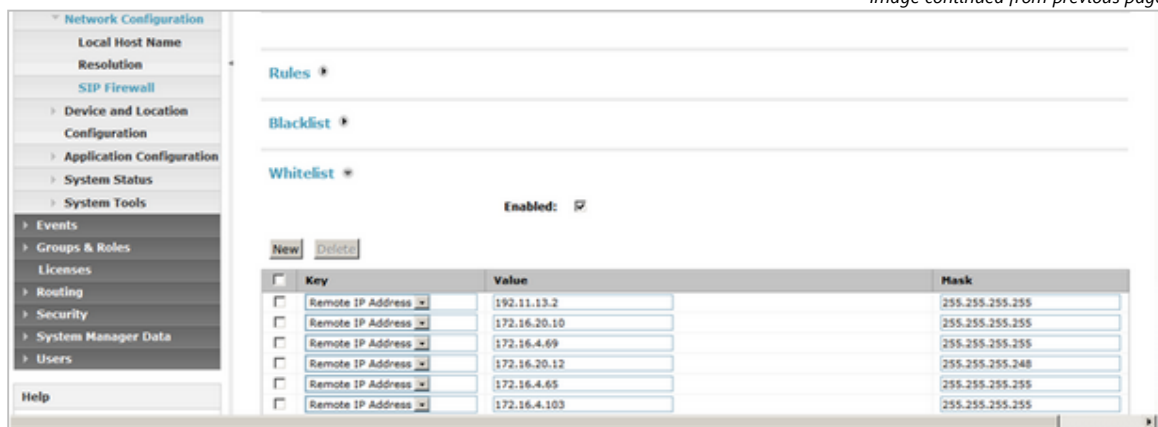
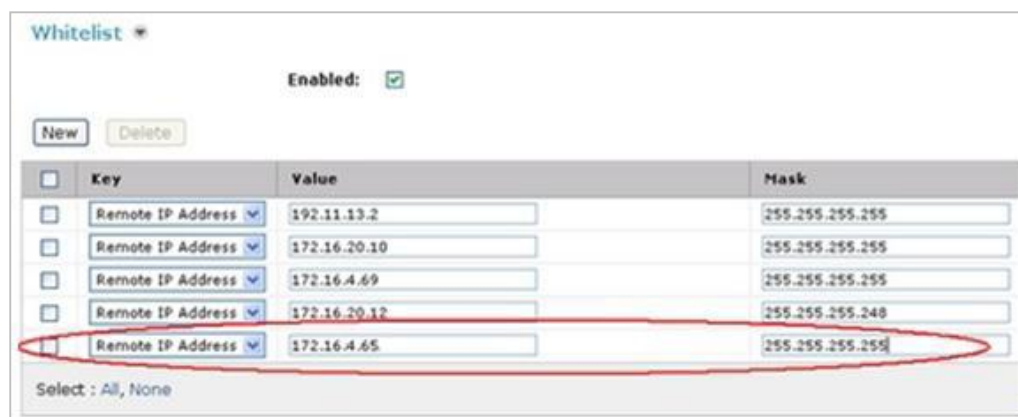


Image continues on next page

Image continued from previous page



- 3 Add an IP address to the Whitelist to ensure that an IP address is not being blocked by a firewall.



## Configuring the Location

Session Manager uses locations to determine which dial patterns to look when routing a call. Each SIP entity has a particular IP address. Depending on the physical and geographic location of each SIP entity, some of the SIP entities can be grouped into a single location. You may use an already existing suitable location for MiCollab AM or create a new one.

To create a Location:



- 1 On the System Manager Web Interface, select **Routing**, and then **Locations**.
- 2 Click **New** to create a new location. This location is used to create a SIP Entity for the MiCollab AM Call Server.

**Location Details** Commit Cancel

**General**

\* **Name:**

**Notes:**

**Managed Bandwidth:**  Kbit/sec

\* **Average Bandwidth per Call:**  Kbit/sec

*Image continues on next page*

*Image continued from previous page*

**Location Pattern**

Add Remove

1 Item Refresh Filter: Enable

IP Address Pattern	Notes
* <input type="text" value="172.16.0.0"/>	<input type="text" value="255.255.0.0"/>

Select : All, None

\* **Input Required** Commit Cancel

- 3 In the **Name** field, enter a Name for the location.
- 4 In the **Average Bandwidth per Call** field, enter a value based on the site policies (the default is blank).  
For example:  
G.711 Mu law uses roughly 80kbps of bandwidth per call. Use this value for the Average Bandwidth per Call.
- 5 In the **Location Pattern** field, enter a pattern that allows the MiCollab AM Call Server IP address. In this example, IP addresses in the range of 172.16.0.0 through 172.16.255.255 are allowed.
- 6 Click **Commit** to save the changes.

## Creating an Adaptation

Adaptations modify the SIP messages that leave the Session Manager. The associated adaptation is used when routing calls to the SIP Entity.

### To create an Adaptation:

- 1 On the System Manager Web Interface, select **Routing** and then **Adaptations**.
- 2 Click **New** to create a new adaptation.

**Adaptation Details** Commit Cancel

**General**

\* Adaptation name: Voicemail

Module name: DigitConversionAdapter

Module parameter:

Egress URI Parameters:

Notes:

**Digit Conversion for Incoming Calls to SM**

Add Remove

0 Items Refresh Filter: Enable

Matching Pattern	Min	Max	Delete Digits	Insert Digits	Address to modify	Notes

**Digit Conversion for Outgoing Calls from SM**

Add Remove

1 Item Refresh Filter: Enable

Matching Pattern	Min	Max	Delete Digits	Insert Digits	Address to modify	Notes
*+	*1	*36	*1		both	

Select : All, None

\* Input Required Commit Cancel

- 3 In the **Adaptation** name field, enter an Adaptation name.
- 4 In the **Module** name field, select **DigitConversionAdapter** from the list.
- 5 The remaining fields can retain their default values.
- 6 Click **Commit** to save the changes.

## Configuring a SIP Entity

SIP Entities are all network entities that are part of the SIP domain. A SIP Entity must be added for Session Manager and for each SIP-based telephony system supported by a SIP Trunk. Verify the IP connections between the Communication Server 1000E and Session Manager server. The following screen shows the SIP Entity for the Communication Server 1000E.

**AVAYA** Avaya Aura® System Manager 6.1 Help

Home / Elements / Routing / SIP Entities - SIP Entity Details

**SIP Entity Details**

**General**

\* Name:

\* FQDN or IP Address:

Type:

Notes:

Adaptation:

Location:

Time Zone:

Override Port & Transport with DNS SRV: ☐

\* SIP Timer B/F (in seconds):

Credential name:

Call Detail Recording:

**SIP Link Monitoring**

SIP Link Monitoring:

**Entity Links**

Figure 3. SIP Entity Details

Create a SIP entity for MiCollab AM. The Session Manager uses a SIP Entity to route calls to the Call Server.

**NOTE** For more information on SIP Entities, refer to the Avaya document 03-603324. See the topic, *Administering Avaya Aura Session Manager*.

### To create a SIP Entity:

- 1 On the System Manager Web Interface, select **Routing** > **SIP Entities**.
- 2 Click **New** to create a new SIP Entity for MiCollab AM Call Server.

- 3 In the **Name** field, enter a name for the SIP Entity.
- 4 In the **FQDN or IP address** field, enter the FQDN or IP Address of the MiCollab AM Call Server.
- 5 In the **Type** field, select **SIP Trunk** from the list.
- 6 (Optional) In the **Notes** field, enter a descriptive note that helps identify the SIP Entity.
- 7 In the **Adaptation** field, select the Adaptation you created in the procedure, Creating an Adaptation from the list. In this example, select **Voicemail**.
- 8 In the **Location** field, select the location from the list.
- 9 In the **Time Zone** field, select the appropriate time zone from the list.
- 10 In the **SIP Timer B/F (in seconds)** field, enter **4**.
- 11 In the **Call Detail Recording** field, select an available option based on site policies.
- 12 In the **SIP Link Monitoring** field, select **Use Session Manager Configuration** from the list.

**NOTE** This selection uses the settings from **Elements > Session Manager > Session Manager Administration > Monitoring**. If monitoring is disabled, then the administrator should select Link Monitoring Enabled from the list in step 12. The default values may be changed based on the site policy.

- 13 Click **Commit** to save the changes.

## Configuring Entity Links

Configure an entity link for the Session Manager that allows it to send and receive messages with MiCollab AM.

### To configure the Entity Link:

- 1 On the System Manager Web Interface, select **Routing**, and then **Entity Links**.
- 2 Click **New** to create a new Entity Link between the Session Manager and MiCollab AM Call Server.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
SM - Stingray	SessionManager	TCP	5060	Stingray	5060	<input checked="" type="checkbox"/>	Stingray

- 3 In the **Name** field, enter a Name for the Entity Link.
- 4 In the **SIP Entity 1** field, select **Session Manager** from the list.
- 5 In the **Protocol** field, select **TCP** from the list.
- 6 In the **Port** field, enter **5060**.
- 7 In the **SIP Entity 2** field, select the SIP Entity you created for MiCollab AM in the previous procedure, Configuring a SIP Entity from the list, in this example, Stingray.
- 8 In the **Port** field, enter **5060**.

**NOTE** This port number must match the listening port configured on MiCollab AM in the Integrations Options dialog box.

- 9 Selected the **Trusted** checkbox to make the Entity Link trusted.
- 10 (Optional) In the **Notes** field, enter a description for identification.
- 11 Click **Commit** to save changes.

## Configuring the Routing Policies

Configure the routing policies associated with the SIP Entity created for MiCollab AM.

### To configure the routing policies:

- 1 On the System Manager Web Interface, select **Routing**, and then click **Routing Policies**.

- 2 Click **New** to create a new Routing Policy.

The screenshot shows the 'Routing Policy Details' configuration page. On the left is a navigation menu with categories: Elements, Events, Groups & Roles, Licenses, Routing (selected), Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, Defaults, Security, System Manager Data, and Users. Below the menu is a 'Help' section with links to various help topics. The main content area is titled 'Routing Policy Details' and contains several sections: 'General' with fields for Name (To Stingray), Disabled (checkbox), and Notes (Stingray); 'SIP Entity as Destination' with a 'Select' button and a table listing SIP entities; 'Time of Day' with 'Add', 'Remove', and 'View Gaps/Overlaps' buttons, a table of time ranges, and a 'Select' dropdown; 'Dial Patterns' with 'Add' and 'Remove' buttons and a table of dial patterns; and 'Regular Expressions' with 'Add' and 'Remove' buttons and a table of regular expressions. At the bottom are 'Commit' and 'Cancel' buttons.

**Routing Policy Details** Commit Cancel

**General**

\* Name:

Disabled: ☐

Notes:

**SIP Entity as Destination**

Select

Name	FQDN or IP Address	Type	Notes
Stingray	Stingray.sample.domain.local	SIP Trunk	SIP Trunk to Stingray

**Time of Day**

Add Remove View Gaps/Overlaps

1 Item Refresh Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select: All, None

**Dial Patterns**

Add Remove

0 Items Refresh Filter: Enable

Pattern	Min	Max	Emergency Call	SIP Domain	Originating Location	Notes
---------	-----	-----	----------------	------------	----------------------	-------

**Regular Expressions**

Add Remove

0 Items Refresh Filter: Enable

Pattern	Rank Order	Deny	Notes
---------	------------	------	-------

\* Input Required Commit Cancel

- 3 In the **Name** field, enter a Name for the policy.
- 4 (Optional) In the **Notes** field, enter a descriptive note for identification.
- 5 Click **Select** under SIP Entity as Destination.
- 6 Select the SIP Entity you created in the procedure, Configuring a SIP Entity for MiCollab AM and then click **Select**.
- 7 Adjust the **Time of Day** per site policies or keep the default values.
- 8 Click **Commit** to save the changes.

## Configuring Dial Patterns

Configure a dial pattern to route calls to MiCollab AM based on the dialed digits.

### To configure a Dial Pattern:

- 1 On the System Manager Web interface, select **Routing** and then **Dial Patterns**.
- 2 Click **New** to create a new Dial Pattern.

**Dial Pattern Details** [Commit] [Cancel]

**General**

\* Pattern: 6000

\* Min: 4

\* Max: 4

Emergency Call: ☐

SIP Domain: -ALL-

Notes: SIP Trunk to Voicemail

**Originating Locations and Routing Policies**

[Add] [Remove]

1 Item Refresh

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Test Lab		To Stingray	0	<input type="checkbox"/>	Stingray	Stingray

Select : All, None

**Denied Originating Locations**

[Add] [Remove]

0 Items Refresh

<input type="checkbox"/>	Originating Location	Notes
--------------------------	----------------------	-------

\* Input Required [Commit] [Cancel]

- 3 In the **Pattern** field, enter a pattern that allows calls directed to the MiCollab AM hunt group number to be routed to MiCollab AM.

**For example:**

A pattern that matches the Hunt group number.

**NOTE** The pattern can be 1 to 36 digits long. A valid pattern format is '[+\*#0-9x][0-9x]{0,35}'.

- 4 In the **Min** field, enter the minimum number of matching digits.
- 5 In the **Max** field, enter the maximum number of matching digits.
- 6 In the **SIP Domain** field, select the SIP Domain to which this dial pattern should be restricted. You may allow it for all SIP Domains.
- 7 (Optional) In the **Notes** field, enter a descriptive note for identification.
- 8 Click **Add**, to add an Originating Location and a Routing Policy.
- 9 In the **Originating Location Name** field, select a location from the list of available Originating Locations.
- 10 In the **Routing Policy Name** field, select the Routing policy that you created for MiCollab AM in the procedure, configuring the Routing Policies from the list of available Routing Policies, and then click **Select**.
- 11 Click **Commit** to save the changes.

# Configuring MiCollab AM

Once the telephone system is programmed, you must configure MiCollab AM for the integration. There are two ways you can configure MiCollab AM: (1) Configuring MiCollab AM for the telephone system integration when you are installing MiCollab AM for the first time, or (2) Configuring the existing MiCollab AM with the new telephone system integration.

Click the appropriate steps that your system requires from below and follow the steps:

- [Configuring MiCollab AM for the Integration During Initial Installation](#): Integrate the telephone system while you install MiCollab AM for the first time.
- [Configuring Existing MiCollab AM for the Integration](#): Integrate a new telephone system on your existing MiCollab AM system.

**NOTE** For general information on integrations, refer to the **Integrating MiCollab AM with the Telephone System** chapter in *System Installation Guide*, and the topic, **Integrate the Telephony Server with the Telephone System**, in the online help.

## Configuring MiCollab AM for the Integration During Initial Installation

To configure MiCollab AM for the integration during the initial installation:

- 1 In the **Database Initialization Parameters** dialog box, configure the following options:
  - a In the **Mailbox Length** box, enter the mailbox length in digits.
  - b In the **First Extension** box, enter first extension number for the first line. You can also leave the **First Extension** box empty.
  - c From the **Manufacturer** dropdown list, select **Avaya**.
  - d From the **Model** dropdown list, select **Communication Server 1000**.
  - e From the **Integration Type** dropdown list, select **SIP Trunk**.
- 2 Click **Next**. The **Board Options** dialog box displays.
- 3 In the **Board Options** dialog box, configure the following options:
  - a From the **Manufacturer** dropdown list, select **RadVision**.
  - b From the **Model** dropdown list, select **SIP STACK**.
  - c In the **Name** field, the name for this board is automatically generated. Enter a new name if necessary.
  - d From the **Protocol** dropdown list, select **SIP IP RTP**.



- e In the **Number of Lines** field, enter the number of lines this board uses. The total number of lines is limited by the capacity of the board and the number of **Available Line Licenses**.
- 4 Click **OK**. The **Switch Options** dialog box displays.
- 5 If necessary, make any changes to the default settings your site requires in the **Switch Options** dialog box.

**NOTE** The settings related to the telephone system in the **Switch Options** dialog box are filled in automatically when you select the correct telephone system during setup.

If you need to customize settings on the **Switch Options** dialog box to meet requirements specific to your site, refer to the documentation accompanying the telephone system, the online help, and the guide, *System Installation Guide*.

- 6 Click **OK**. The **Integration Options** dialog box displays.
- 7 In the **Integration Options** dialog box, configure the options as follows:
  - a In the **Local Integration Settings** section, select the **Required Parameters** view.

Table 10. Required Parameters for Integrations Options

Field	Required Value
SIP Server Address	Enter the IP address of the Session Manager server.
SIP Server Port	Enter the port number on which the Session Manager listens for SIP messages. This port <b>must</b> match the Session Manager port configured in the procedure, <a href="#">Configuring Entity Links</a> . The default port number is <b>5060</b> .
SIP parser qualifier string	<ul style="list-style-type: none"> <li>• <b>Single SIP integration on the call server:</b> Enter the local IP address to which the integration is bound. This field is used by MiCollab AM to match SIP packets to the appropriate SIP integration.</li> <li>• <b>Multiple SIP integrations on the call server:</b> Use a string that is unique to each SIP integration.</li> </ul> <p><b>For example:</b></p> <ul style="list-style-type: none"> <li>• The extension that will be used as the hunt number on the PBX followed by the @ symbol and the IP of the call server, such as 5000@172.16.4.202. <i>The hunt number must be unique across all IP integrations.</i></li> <li>• The Fully Qualified Domain Name (FQDN) of the switch, such as pbx1.sipdomain.com.</li> </ul>
	<p><b>NOTE</b> This setting must match a string in the SIP header that is unique to this particular integration.</p>
SIP Domain Name	Enter the SIP domain name. This value must be the same value as in the procedure, <a href="#">Obtaining SIP Domain</a> .

	<b>IMPORTANT</b> Be sure the Authoritative Domain name is the same throughout the Session Manager server, Communication Server 1000E, and MiCollab AM programming.
Transport for outgoing SIP messages	Enter <b>TCP</b> or <b>UDP</b> .  <b>NOTE</b> This value must match the protocol selected on the Entity Link created for MiCollab AM Call Server.
Local IP Address to bind on	Enter the IP address of the network interface card (NIC) on the Call Server platform that supports the SIP integration. If there is only one NIC on the MiCollab AM server platform, this field typically contains the IP address of that NIC already.
SIP Location Connection Port	Enter the port number on which MiCollab AM listens for incoming SIP messages.  The default value is <b>5060</b> .
Media packet size (milliseconds)	MiCollab AM sends/receives packets containing the number of milliseconds worth of audio data set here.  The default value is <b>20</b> .

**b** In the **Local Integration Settings** section, select the **Integration Specific Parameters** view and configure the following parameters.

- Select the Validate Remote Hosts for Media check box.
- Set the **Type of Call Progress to use for External Calls** value. How this should be set depends on the gateway used for the integration as follows:
  - **Digital:** Select Digital if the gateway supports call progress through to the endpoint.
  - **Media:** Select Media if the gateway reports early that the call is connected, such as before the phone rings or while the phone is ringing.

**8** Click **OK**. The **Switch Section Options** dialog box displays.

**9** In the **Switch Section Options** dialog box, configure the following options:

**a** In the **Local Integration Settings** section, select the **Required Parameters** view.

**b** In the **Incoming Hunt Mode** field, enter the mode for this integration.

**NOTE** This integration supports **Terminal**, **Circular**, **Reverse Terminal**, and **Reverse Circular** hunt modes only.

**c** In the **Hunt Group Access Code** field, enter the pilot number that matches with the DN prefix programmed for the Gateway Endpoint associated with this MiCollab AM Call Server.

**d** Click **OK**.

**10** Continue through and complete the configuration. At the end of the configuration, a confirmation dialog box displays. Click **OK**.

- 11 If **MiCollab AM Configuration** does not open automatically after the configuration completes, open **MiCollab AM Configuration**, and select the **Lines** tab.
- 12 In the table from the **Lines** tab, configure callouts for the application. For information on configuring callout settings, see the topic *Configuring Callout Settings*, in the online help system.
- 13 Click **OK** to save all changes.

## Configuring Existing MiCollab AM for the Integration

To configure exiting MiCollab AM for the telephone integration:

- 1 Open **MiCollab AM Configuration**, and go to the **Main** tab.
- 2 In the **Main** tab, click **Shutdown** to stop the system. Wait until the **Current Status** shows **Stopped**.

**NOTE** If you have not configured the virtual board with your MiCollab AM system yet, complete **Step 3**. If your MiCollab AM already has the virtual board configured, skip to **Step 4**.

- 3 **[Optional]** Select the **Board** tab, and then click the **Add** button. The **Board** dialog box displays.
  - a From the **Manufacturer** dropdown list, select **RadVision**.
  - b From the **Model** dropdown list, select **SIP STACK**.
  - c In the **Name** field, the name for this board is automatically generated. Enter a new name if necessary.
  - d From the **Protocol** dropdown list, select **SIP IP RTP**.
  - e In the **Number of Lines** field, enter the number of lines this board uses. The total number of lines is limited by the capacity of the board and the number of **Available Line Licenses**.
  - f Click **OK**.
- 4 Select the **Switch** tab and click the **Add** button. The **Switch Integration Data Setup** dialog box displays.
  - a From the **Manufacturer** dropdown list, select **Avaya**.
  - b From the **Model** dropdown list, select **Communication Server 1000**.
  - c From the **Integration Type** dropdown list, select **SIP Trunk**.
- 5 Click **OK**. The **Switch Options** dialog box displays.
- 6 If necessary, make any changes to the default settings your site requires in the **Switch Options** dialog box.

**NOTE** The settings related to the telephone system in the **Switch Options** dialog box are filled in automatically when you select the correct telephone system during setup.

If you need to customize settings on the **Switch Options** dialog box to meet requirements specific to your site, refer to the documentation accompanying the telephone system, the online help, and the guide, *System Installation Guide*.

- 7 Click **OK**. The **Integration Options** dialog box displays.
- 8 In the **Integration Options** dialog box, configure the options as follows:
  - a In the **Local Integration Settings** section, select the **Required Parameters** view.

Table 11. Required Parameters for Integrations Options

Field	Required Value
SIP Server Address	Enter the IP address of the Session Manager server.
SIP Server Port	Enter the port number on which the Session Manager listens for SIP messages. This port <b>must</b> match the Session Manager port configured in the procedure, <a href="#">Configuring Entity Links</a> . The default port number is <b>5060</b> .
SIP parser qualifier string	<ul style="list-style-type: none"> <li>• <b>Single SIP integration on the call server:</b> Enter the local IP address to which the integration is bound. This field is used by MiCollab AM to match SIP packets to the appropriate SIP integration.</li> <li>• <b>Multiple SIP integrations on the call server:</b> Use a string that is unique to each SIP integration.  <b>For example:</b> <ul style="list-style-type: none"> <li>• The extension that will be used as the hunt number on the PBX followed by the @ symbol and the IP of the call server, such as 5000@172.16.4.202. <i>The hunt number must be unique across all IP integrations.</i></li> <li>• The Fully Qualified Domain Name (FQDN) of the switch, such as pbx1.sipdomain.com.</li> </ul> </li> </ul> <p><b>NOTE</b> This setting must match a string in the SIP header that is unique to this particular integration.</p>
SIP Domain Name	Enter the SIP domain name. This value must be the same value as in the procedure, <a href="#">Obtaining SIP Domain</a> .  <b>IMPORTANT</b> Be sure the Authoritative Domain name is the same throughout the Session Manager server, Communication Server 1000E, and MiCollab AM programming.
Transport for outgoing SIP messages	Enter <b>TCP</b> or <b>UDP</b> .  <b>NOTE</b> This value must match the protocol selected on the Entity Link created for MiCollab AM Call Server.
Local IP Address to bind on	Enter the IP address of the network interface card (NIC) on the Call Server platform that supports the SIP integration. If there is

	only one NIC on the MiCollab AM server platform, this field typically contains the IP address of that NIC already.
SIP Location Connection Port	Enter the port number on which MiCollab AM listens for incoming SIP messages. The default value is <b>5060</b> .
Media packet size (milliseconds)	MiCollab AM sends/receives packets containing the number of milliseconds worth of audio data set here. The default value is <b>20</b> .

**b** In the **Local Integration Settings** section, select the **Integration Specific Parameters** view and configure the following parameters.

- Select the Validate Remote Hosts for Media check box.
- Set the **Type of Call Progress to use for External Calls** value. How this should be set depends on the gateway used for the integration as follows:
  - **Digital:** Select Digital if the gateway supports call progress through to the endpoint.
  - **Media:** Select Media if the gateway reports early that the call is connected, such as before the phone rings or while the phone is ringing.

**9** Click **OK**. The **Switch Section Options** dialog box displays.

**10** In the **Switch Section Options** dialog box, configure the following options:

- a** In the **Local Integration Settings** section, select the **Required Parameters** view.
- b** In the **Incoming Hunt Mode** field, enter the mode for this integration.

**NOTE** This integration supports **Terminal**, **Circular**, **Reverse Terminal**, and **Reverse Circular** hunt modes only.

- c** In the **Hunt Group Access Code** field, enter the pilot number that matches with the DN prefix programmed for the Gateway Endpoint associated with this MiCollab AM Call Server.
- d** Click **OK**.

**11** In **MiCollab AM Configuration**, verify that the telephone system is properly added and configured in the **Switches**, **Switch Sections**, and **Integrations** tabs.

**12** Select the **Lines** tab.

**13** In the table from the **Lines** tab, configure callouts for the application. For information on configuring callout settings, see the topic *Configuring Callout Settings*, in the online help system.

**14** Click **OK** to save all changes.

## Configuring MiCollab AM for SIP Failover

MiCollab AM can be configured for automatic failover to the secondary SIP server in the event of the primary/host SIP server failure. Use the instructions provided in this section to add or remove secondary SIP server(s) for failover.

## To add a SIP failover server:

- 1 From **MiCollab AM Configuration**, click the **Integrations** tab.
- 2 From the **Integrations** list, select your integration, and then click **Edit**.
- 3 In the **Integration Options** dialog box, go to the **Local Integration Settings** section.
- 4 From the **View** dropdown list, select **Failover Server Settings**.
- 5 Click the **Add Failover Server** button. Two new rows are added to configure the secondary SIP server.
- 6 In the **Secondary SIP Server Address** and **Secondary SIP Server Port** rows, enter the appropriate value as follows:

Table 12. Secondary SIP Server Address and the Secondary SIP Server Port example

Field	Value
Secondary SIP Server Address	<p>Enter the TCP/IP address or an FQDN of the secondary node.</p> <p><b>For example:</b> The IP address 123.45.6.789 as displayed on the Review/Modify SIP Gateway screen.</p> <p><b>NOTE</b> This integration requires the machine name to be a fully qualified domain name. Therefore, use the Machine Name field as displayed on the Review/Modify SIP Gateway screen during the integration process.</p> <p><b>IMPORTANT</b> This value must match the configuration on the Gateway of the secondary node.</p>
Secondary SIP Server Port	Enter the port number of the secondary node. The default value is <b>5060</b> .

- 7 From the **View** dropdown list, select **Integration Specific Parameters**. The **Integration Specific Parameters** view displays.
- 8 In the **Integration Specific Parameters** list, enter the information as shown in the following table:

**NOTE** The parameters in the following table is listed in alphabetical order. The actual Integration Specific Parameters on your system may not be listed in the same order presented in the table below.

Table 13. Integration Specific Parameters

Field	Value
Enable SIP server failover	Select this check box to allow for failover and to enable the failover server setting changes.

Delay (in ms) between Failover attempts	The delay in milliseconds before MiCollab AM attempts to register its port with the SIP server. The default is <b>1000 ms</b> .
Incoming off hook delay	800
Outgoing off hook delay	0
On hook delay	300
Type of Call Progress to use for External Calls	<p>How this should be set depends on the gateway used for the integration.</p> <ul style="list-style-type: none"> <li>• If the gateway supports call progress through to the endpoint, set to <b>Digital</b>.</li> <li>• If the gateway reports early that the call is connected, such as before the phone rings or while the phone is ringing, set to <b>Media</b>.</li> </ul>

- 9 Click **Apply** to save the changes.
- 10 To add another failover server repeat **Steps 4-9**.
- 11 Click **OK** to close the **Integration Options** dialog box.

### To remove a SIP Failover Server:

- 1 From **MiCollab AM Configuration**, click the **Integrations** tab.
- 2 From the **Integrations** list, select your integration, and then click **Edit**.
- 3 In the **Integration Options** dialog box, go to the **Local Integration Settings** section.
- 4 From the **View** dropdown list, select **Failover Server Settings**.
- 5 In the **Failover Server Settings** view, click the **Remove Failover Server** button.
- 6 At the confirmation prompt, click **Yes** to confirm the deletion.

**NOTE** If multiple servers are listed, the last server address and port pair on the list is deleted first.

- 7 Click **Apply** to save the changes, and then click **OK** to close the **Integration Options** dialog box.

# Changing the Network Binding Order on the MiCollab AM Platform

MiCollab AM uses the primary (public) network interface card (NIC) in the platform. It must be the first network connection in the network binding order. If your MiCollab AM server platform is a component of two or more local or wide area networks (LANs or WANs), you must make sure that this integration does not interfere with the normal network operation of the server.

**NOTE** The operating system gives precedence to the first network connection in the list followed by the remaining connections based on their position in the list.

The instructions in this section ensure that the binding order is correct when you set up the integration. If you replace a NIC on the MiCollab AM server platform later, the platform's operating system registers the new adapter at the bottom of its binding order. Restoring the original binding order should correct any problems caused by the change.

**IMPORTANT** The following procedure shifts the binding order of the network interface cards. To determine which NIC is associated with a specific network connection, right-click the connection in the Network Connections window, and then select Properties.

## Windows Server 2008 R2 with Service Pack 1

To change the binding order of multiple NICs:

- 1 From the taskbar, click **Start** > **Control Panel**.
- 2 In the **Control Panel**, click **Network and Sharing Center**.
- 3 On the left pane, select **Change Adapter Settings**.
- 4 Press **Alt** to display the menu bar.
- 5 On the menu bar, select **Advanced**, and then click **Advanced Settings**.
- 6 On the **Adapters and Bindings** tab of **Advanced Settings**, click the network connection that serves MiCollab AM.
- 7 Click the up arrow button to the right of the **Connections** list as many times as needed to move the connection to the top of the list.
- 8 Click **OK**, and then close the **Network Connections** window and the **Control Panel**.



## Windows Server 2012 R2

To change the binding order of multiple NICs:

- 1 From the taskbar, click **Start > Control Panel**.
- 2 In the **Control Panel**, click **Network and Sharing Center**.
- 3 On the left pane, select **Change Adapter Settings**.
- 4 Press **Alt** to display the menu bar.
- 5 On the menu bar, select **Advanced**, and then click **Advanced Settings**.
- 6 On the **Adapters and Bindings** tab of **Advanced Settings**, click the network connection that serves MiCollab AM.
- 7 Click the up arrow button to the right of the **Connections** list as many times as needed to move the connection to the top of the list.
- 8 Click **OK**, and then close the **Network Connections** window and the **Control Panel**.

# Configuring Quality of Service (QoS)

As of version 6.0, MiCollab AM has no internal support for QoS. QoS must now be implemented externally via group policies as Policy-Based QoS. Refer to your operating system's documentation for details.

Table 14. QoS Configuration

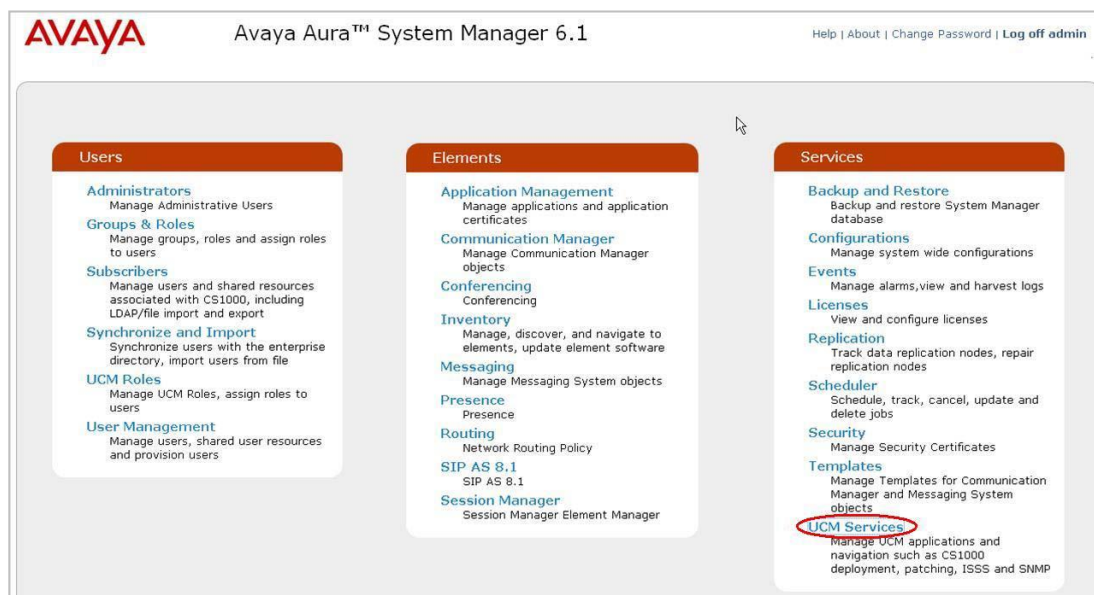
Field	Setting
Application Name	At_TelephonyServer.exe
Protocol	Match the setting used for the integration UDP or TCP
Source Port	<p>MiCollab AM requires a range of ports for audio support. The MiCollab AM audio ports start at the Local Media Base UDP Port configured in the Server tab. Each MiCollab AM line reserves 10 ports. Hence, the port range starts from the number configured there, and goes to the last port of the last line. The formula for calculating the highest port number in the range is as follows:</p> $\text{BasePortNumber} + (\text{NumberOfCXPorts} * 10) - 1.$ <p>Hence, if the base port is 10000, and MiCollab AM has 8 lines, then the port range to use would be:</p> <p>10000:10079</p>
DSCP Value	46

# Appendix A

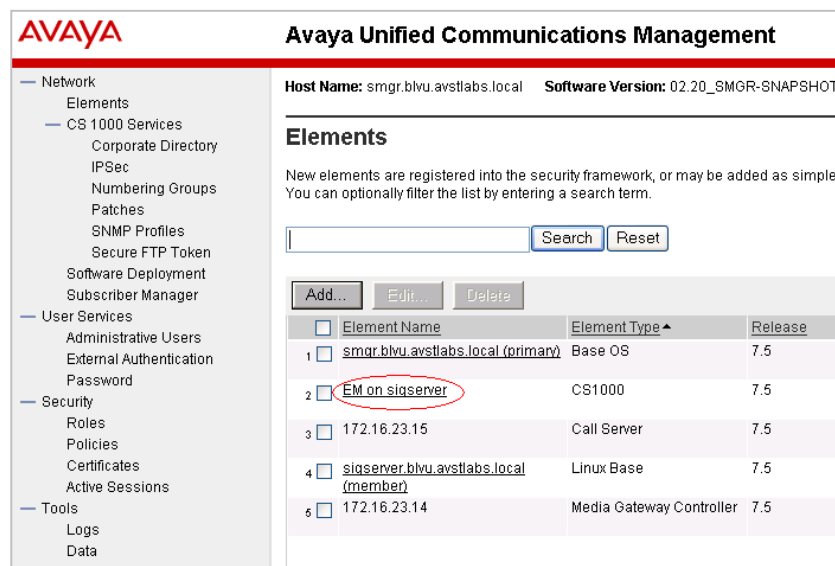
## Programming ISDN through Avaya System Manager UCM Services

To Launch Communication Server 1000E Unified Communications Manager:

- 1 Open **Unified Communications Manager (UCM)** on the **System Manager** page.

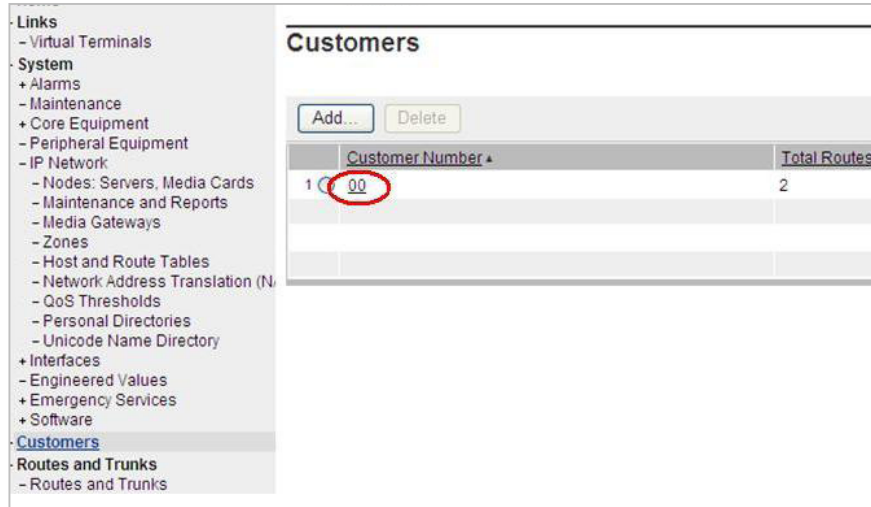


- 2 Click the **Element Name** that corresponds to the **Communication Server 1000** element type.

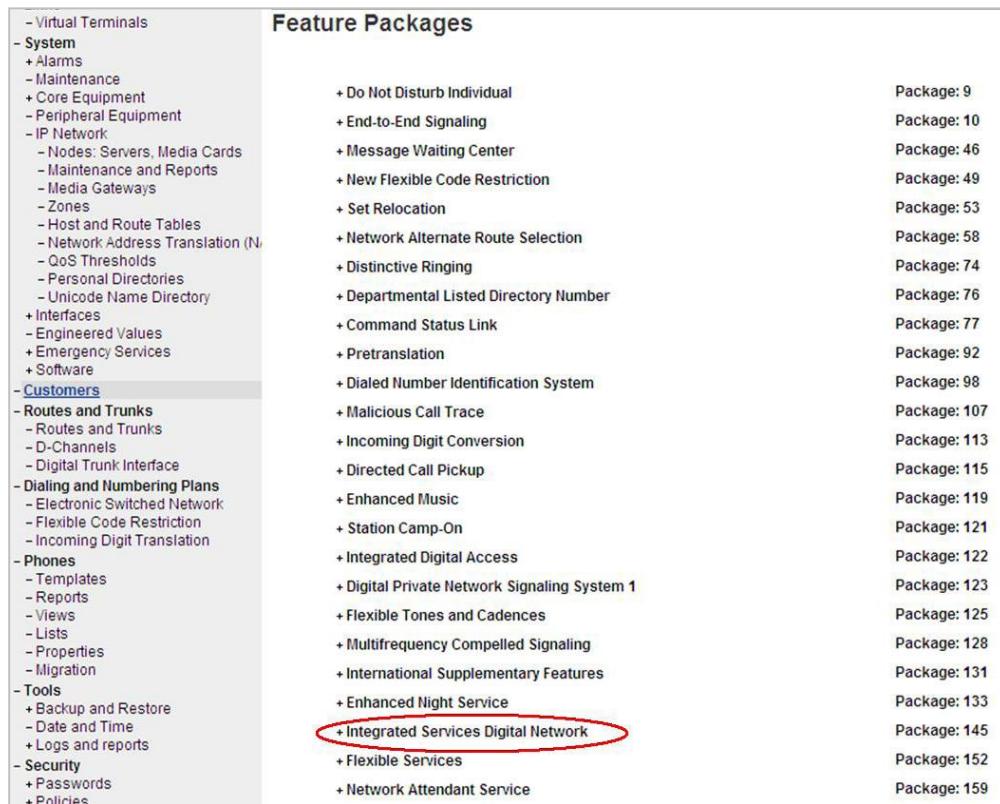


### 3 Administer ISDN.

In the left pane, select Customers. Click the link associated with the appropriate customer. The system can support multiple customers with different network settings and options. In the sample configuration, only one customer was configured on the system.



### 4 In the **Customer Details** screen, select **Feature Packages**. The **Feature Packages** contains a listing of feature packages (not all features shown below). Select **Integrated Services Digital Network** to edit its parameters.



The screen is updated with parameters populated below **Integrated Services Digital Network**.

- 5 Check the **Integrated Services Digital Network** checkbox. Leave the default values in place for all remaining fields. Scroll down to the bottom of the screen, and click **Save** (not shown).

**AVAYA CS1000 Element Manager**

**Integrated Services Digital Network** Package: 145

+ Dial Access Prefix on CLID table entry option

Integrated Services Digital Network: ☒

- Virtual private network identifier:  (1 - 16383)

- Private network identifier:  (1 - 16383)

- Node DN:

Multi-location business group:  (0 - 65535)

Business sub group consult-only:  (0 - 65535)

## To Administer D-Channel:

- 1 Select **Routes and Trunks** > **D-Channels** from the left pane to display the **D-Channels** screen.
- 2 In the **Choose a D-Channel Number** field, select an available D-channel from the drop-down.
- 3 Click to **Add**. The **D-Channels Property Configuration** screen is displayed next.
- 4 Enter the following values for the specified fields. The screen below displays the parameters subsequent to adding the D-channel.
  - a D channel Card Type: D-Channel is over IP (DCIP).
  - b Designator: A descriptive name.

**AVAYA CS1000 Element Manager**

Managing: 172.16.23.15 Username: admin  
Routes and Trunks > D-Channels > D-Channels 4 Property Configuration

**D-Channels 4 Property Configuration**

**- Basic Configuration**

Input Description	Input Value
Action Device And Number (ADAN):	DCH
D channel Card Type :	DCIP
Designator:	SIP_PRI
Recovery to Primary:	<input type="checkbox"/>
PRI loop number for Backup D-channel:	<input type="text"/>
User :	Integrated Services Signaling Link Dedicated (ISLD)
Interface type for D-channel:	Meridian Meridian1 (SL1)
D-Channel PRI loop number:	<input type="text"/>
Primary Rate Interface:	<input type="text"/> more PRI
Secondary PRI2 loops:	<input type="text"/>
Meridian 1 node type:	Slave to the controller (USR)
Release ID of the switch at the far end:	25
Central Office switch type:	100% compatible with Bellcore standard (STD)
Integrated Services Signaling Link Maximum:	4000 Range: 1 - 4000
Signalling server resource capacity:	3700 Range: 0 - 3700
Primary D-channel for a backup DCH:	<input type="text"/> Range: 0 - 254
- PINX customer number:	<input type="text"/>

**- Basic options (BSCOPT)**

- Click to expand the **Basic options (BSCOPT)**, and then click the **EDIT** button next to Remote Capabilities.

The screenshot shows a configuration interface with a left-hand navigation pane and a main configuration area on the right. In the left pane, the 'Basic options (BSCOPT)' section is highlighted with a red oval. In the main area, the 'Remote Capabilities' section is visible, and the 'Edit' button next to it is also highlighted with a red oval. Other visible fields include 'Secondary PRI2 loops', 'Meridian 1 node type', 'Release ID of the switch at the far end', 'Central Office switch type', 'Integrated Services Signaling Link Maximum', 'Signalling server resource capacity', 'Primary D-channel for a backup DCH', 'PINX customer number', 'Progress signal', 'Calling Line Identification', 'Output request Buffers', 'D-channel transmission Rate', 'Channel Negotiation option', and 'Remote Capabilities'.

- Check the checkbox next to Network name display method 2 (ND2) and MWI. Click Return – **Remote Capabilities**. Then Click on Submit (not shown).

The screenshot shows the 'Remote Capabilities' section of the configuration interface. The left-hand navigation pane is visible, and the main area contains a list of checkboxes for various options. The following options are checked: 'Message waiting interworking with DMS-100 (MWI)', 'Network name display method 2 (ND2)', and 'Trunk anti-tromboning operation (TAT)'. Other options include 'MCDN QSIG conversion (MQC)', 'Remote D-channel is on a MSDL card (MSL)', 'Network access data (NAC)', 'Network call trace supported (NCT)', 'Network name display method 1 (ND1)', 'Network name display method 3 (ND3)', 'Name display - integer ID coding (NDI)', 'Name display - object ID coding (NDO)', 'Path replacement uses integer values (PRI)', 'Path replacement uses object identifier (PRO)', 'Release Link Trunks over IP (RLTI)', 'Remote virtual queuing (RVQ)', 'User to user service 1 (UUS1)', 'NI-2 name display option (NDS)', 'Message waiting indication using integer values (QMWI)', 'Message waiting indication using object identifier (QMWI)', and 'User to user signalling (UUI)'.

## To Add a Route:

- Select **Routes and Trunks** > **Routes and Trunks** from the left pane. Next to the applicable Customer row, click **Add route**.
- Enter the following values for the specified fields, and retain the default values for the remaining fields.
  - Route number (ROUT): Select an available route number.

- b** Designator field for trunk (DES): Descriptive text.
- c** Trunk type (TKTP): TIE trunk data block (TIE).
- d** Incoming and outgoing trunk (ICOG): Incoming and Outgoing (IAO).
- e** Access Code for the trunk route (ACOD): An available access code.
- f** This route is for a virtual trunk route (VTRK): Check the checkbox.
- g** Management (ZONE): zone number.
- h** Node ID of signaling server of this route (NODE): Node ID.
- i** Protocol ID for the route (PCID): SIP (SIP).
- j** Mode of operation (MODE): Route uses ISDN Signaling Link (ISLD).
- k** D channel number (DCH): D-Channel number from previous section.
- l** Private network identifier (PNI): Private network identifier according to your site's requirements.
- m** Network calling name allowed (NCNA): Check the checkbox.
- n** Network call redirection (NCRD): Check the checkbox.

Below is an example of a SIP route.

The screenshot displays the AVAYA CS1000 Element Manager interface. On the left is a navigation tree with categories like UCM Network Services, System, Interfaces, Customers, Routes and Trunks, and Phones. The 'Routes and Trunks' section is expanded, showing 'Routes and Trunks' as the selected item. The main area is titled '- Basic Configuration' and contains various fields for configuring a route. The fields are as follows:

- Route data block (RDB) (TYPE): RDB
- Customer number (CUST): 00
- Route number (ROUT): 4
- Designator field for trunk (DES): SIP\_PRI
- Trunk type (TKTP): TIE
- Incoming and outgoing trunk (ICOG): Incoming and Outgoing (IAO) (dropdown)
- Access code for the trunk route (ACOD): 7003
- Trunk type M911P (M911P): ☐
- The route is for a virtual trunk route (VTRK): ☒
- Zone for codec selection and bandwidth management (ZONE): 00000 (0 - 8000)
- Node ID of signaling server of this route (NODE): 1110 (0 - 9999)
- Protocol ID for the route (PCID): SIP (SIP) (dropdown)
- Print correlation ID in CDR for the route (CRID): ☐
- Integrated services digital network option (ISDN): ☒
- Mode of operation (MODE): Route uses ISDN Signaling Link (ISLD) (dropdown)
- D channel number (DCH): 4 (0 - 254)
- Interface type for route (IFC): Meridian M1 (SL1) (dropdown)
- Private network identifier (PNI): 00001 (0 - 32700)
- Network calling name allowed (NCNA): ☒
- Network call redirection (NCRD): ☒
- Trunk route optimization (TRO): ☐
- Recognition of DTI2 ABCD FALT signal for ISL (FALT): ☐

*Image continues on next page*

Image continued from previous page

- Channel type (CHTY) : B-channel (BCH)

- Call type for outgoing direct dialed TIE route (CTYP) : Unknown Call type (UKWN)

- Insert ESN access code (INAC) : ☐

- Integrated service access route (ISAR) : ☐

- Display of access prefix on CLID (DAPC) : ☐

- Mobile extension route (MBXR) : ☐

- Mobile extension outgoing type (MBXOT) : National number (NPA)

- Mobile extension timer (MBXT) : 0 (0 - 8000 milliseconds)

Calling number dialing plan (CNDP) : Unknown (UKWN)

+ Basic Route Options

+ Network Options

+ General Options

+ Advanced Configurations

Submit Refresh Delete Cancel

## To Add a Trunk:

- 1 Click the Add trunk button next to the newly added route.

**Links**

- Virtual Terminals

**System**

- + Alarms
- Maintenance
- Core Equipment
  - Loops
  - Superloops
  - MSDLMISP Cards
  - Conference/TDS/Multifrequen
  - Tone Senders and Detectors
- Peripheral Equipment
- IP Network
  - Nodes: Servers, Media Cards
  - Maintenance and Reports
  - Media Gateways
  - Zones
  - Host and Route Tables
  - Network Address Translation
  - QoS Thresholds
  - Personal Directories
  - Unicode Name Directory

**Routes and Trunks**

Customer: 0	Total routes: 8	Total trunks: 159	Add route
+ Route: 0	Type: COT	Description: CX4500	Edit Add trunk
+ Route: 1	Type: COT	Description: EXT4100	Edit Add trunk
+ Route: 2	Type: DID	Description: DID	Edit Add trunk
+ Route: 3	Type: COT	Description: CX4001	Edit Add trunk
+ Route: 4	Type: TIE	Description: SIP_PRI	Edit Add trunk
+ Route: 5	Type: TIE	Description: EURO_PRI	Edit Add trunk
+ Route: 6	Type: TIE	Description: US_PRI	Edit Add trunk
+ Route: 7	Type: COT	Description: XFR	Edit Add trunk

- 2 The trunk configuration screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- Trunk data block:** IP Trunk (IPTI)
- Terminal number:** An available terminal number in your site
- Designator field for trunk:** Descriptive text
- Member number:** Starting member number
- Card Density** Select Octal Density (8D)
- Start arrangement Incoming:** IMM
- Start arrangement Outgoing:** IMM
- Channel ID for this trunk:** An available starting channel ID



**AVAYA** **CS1000 Element Manager** Help |

---

Managing: **172.16.23.15** Username: admin  
Routes and Trunks » [Routes and Trunks](#) » Customer 0, Route 4, Trunk 1 Property Configuration

### Customer 0, Route 4, Trunk 1 Property Configuration

- UCM Network Services
- Home
- Links
- Virtual Terminals
- System
  - + Alarms
  - Maintenance
  - Core Equipment
  - Loops
  - Superloops
  - MSDU/MISP Cards
  - Conference/TDS/Multifrequen
  - Tone Senders and Detectors
  - Peripheral Equipment
  - IP Network
  - Nodes: Servers, Media Cards
  - Maintenance and Reports
  - Media Gateways
  - Zones
  - Host and Route Tables
  - Network Address Translation
  - QoS Thresholds
  - Personal Directories
  - Unicode Name Directory
- + Interfaces
- Engineered Values
- + Emergency Services
- + Geographic Redundancy
- + Software
- Customers
- Routes and Trunks
  - [Routes and Trunks](#)
  - D-Channels
  - Digital Trunk Interface
- Dialing and Numbering Plans

**- Basic Configuration**

Auto increment member number: ☒

Trunk data block:

Terminal number:

Designator field for trunk:

Extended trunk:

Member number:  \*

Level 3 Signaling:

Card density:

Start arrangement Incoming:

Start arrangement Outgoing:

Trunk group access restriction:

Channel ID for this trunk:

Class of Service:

**+ Advanced Trunk Configurations**

3 Click on the **Edit** button next to Class of Service.

**AVAYA** **CS1000 Element Manager**

---

Managing: **172.16.23.15** Username: admin  
Routes and Trunks » [Routes and Trunks](#) » Customer 0, Route 4, Trunk 1 Property Configuration

### Customer 0, Route 4, Trunk 1 Property Configuration

- Software
- Call Server PEPs
- Loadware PEPs
- File Upload
- IP Phone Firmware
- Voice Gateway Media Card
- Media Cards PEPs
- Plug-ins
- Customers
- Routes and Trunks
  - [Routes and Trunks](#)
  - D-Channels
  - Digital Trunk Interface
- Dialing and Numbering Plans
- Electronic Switched Network
- Flexible Code Restriction
- Incoming Digit Translation
- Phones
- Templates
- Reports
- Views
- Lists
- Properties
- Migration
- Tools
- + Backup and Restore
- Call Server Initialization
- Date and Time
- + Logs and reports
- Security
- Passwords
- Customer Passwords
- Policies

**- Basic Configuration**

Auto increment member number: ☒

Trunk data block:

Terminal number:

Designator field for trunk:

Extended trunk:

Member number:  \*

Level 3 Signaling:

Card density:

Start arrangement Incoming:

Start arrangement Outgoing:

Trunk group access restriction:

Channel ID for this trunk:

Class of Service:

**+ Advanced Trunk Configurations**

#### Class of Service Example:

Input Description	Input Value
- ACD Priority :	ACD Priority not required (APN) ▼
- Analog Semi-Permanent Connections :	Analog Semi-Permanent Connections Denied (SPCD) ▼
- ARF Supervised COT:	▼
- Barring:	▼
- Battery Supervised COT :	▼
- Busy Tone Supervised COT:	▼
- Centrex Switchhook Flash:	Centrex Switchhook Flash Denied (THFD) ▼
- Dial Pulse:	Digitone (DTN) ▼
- DTR PAD value:	▼
- Echo Canceling:	Echo Canceling Denied (ECD) ▼
- Hong Kong DTI :	▼
- Loop Break Supervised COT:	▼
- Make-break ratio for dial pulse:	10 pulses per second (P10) ▼
- Manual Incoming:	Manual Incoming Denied (MID) ▼
- Media Security:	Media Security Best Try (MSBT) ▼
- Network Hook Flash Over M911P:	▼
- Polarity:	▼
- Priority:	Low Priority (LPR) ▼
- Restriction level:	Unrestricted (UNR) ▼
- Short or long line:	▼
- Transmission Class of Service:	Non-Transmission Compensated (NTC) ▼
- Warning Tone:	Warning Tone Allowed (WTA) ▼
- ARF Supervised COT:	▼

- 4 In the **Advanced Trunk Configurations**, Select **Answer and disconnect supervision required**, then select **PIP** as the Supervision Type.

**AVAYA**  
**CS1000 Element Manager**

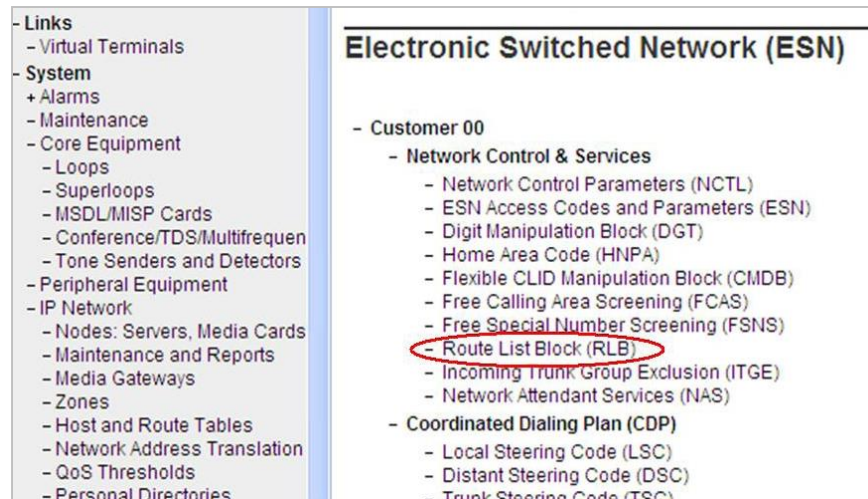
- UCM Network Services  
- Home  
- Links  
- Virtual Terminals  
- System  
+ Alarms  
- Maintenance  
+ Core Equipment  
- Peripheral Equipment  
+ IP Network  
+ Interfaces  
- Engineered Values  
+ Emergency Services  
+ Geographic Redundancy  
+ Software  
- Customers  
- Routes and Trunks  
- Routes and Trunks  
- D-Channels  
- Digital Trunk Interface  
- Dialing and Numbering Plans  
- Electronic Switched Network  
- Flexible Code Restriction  
- Incoming Digit Translation  
- Phones  
- Templates  
- Reports  
- Views  
- Lists  
- Properties  
- Migration  
- Tools  
- Backup and Restore  
- Call Server  
- Personal Directories  
- Call Server Initialization  
- Date and Time  
+ Logs and reports  
- Security

Channel ID for this trunk: 1  
Class of Service: Edit  
- Advanced Trunk Configurations

CTI trunk Monitoring and Control: ☐  
Auto Terminate DN:   
Music conference loop:  ( 0 - 159 )  
Call modification features restriction: ☐  
Digit collection ready: ☐  
Forced Charge Account: ☐  
Multifrequency digit level: 0 ▼  
Multifrequency PAD: ☐  
Manual Directory Number:   
Network Class of Service group: 0 ▼  
Night service group number: 0 ▼  
Night service directory number:   
Pulse code modulation law: ▼  
Pad category table number for digital trunks: 1 ▼  
Private line directory number:   
Signaling category table number: 1 ▼  
Answer and disconnect supervision required: ☒  
- Supervision Type : Polarity Insensitive Pack (PIP) ▼  
Step-by-step CO trunk: ☐  
Termination Impedance: 600 ohms (600) ▼  
Trunk identifier:

## To Administer Route List Block and Distant Steering Code:

- 1 Select **Dialing and Numbering Plans > Electronic Switched Network** from the left pane to display the **Electronic Switched Network (ESN)** screen. Select **Route List Block (RLB)**.



- 2 In the **Please enter a route list index** field; enter an available route list block number (in this case **4**). Click **Add**.

The **Route List Block** screen is displayed with a listing of parameters. For the **Route Number** field, select the route number from previous **Section**. Retain the default values for the remaining fields.

**Data Entry of a Route List Block**

Route List Block Index: 4

---

**General Properties**

Entry Number for the Route List:

**Indexes**

Time of Day Schedule:

Facility Restriction Level:  (0 - 7)

Digit Manipulation Index:

ISL D-Channel Down Digit Manipulation Index:  (0 - 1999)

Free Calling Area Screening Index:

Free Special Number Screening Index:

Business Network Extension Route: ☐

Incoming CLID Table:  (0 - 0)

**Options**

Local Termination entry: ☐

Route Number:

Skip Conventional Signaling: ☐

Use Tone Detector: ☐

Conversion to LDN: ☐

Expensive Route: ☐

Strategy on Congestion:

*Image continues on next page*

Image continued from previous page

- QSIG Alternate Routing Causes: QSIG Alternate Routing Cause 1

Preferred Routing: Preferred Route 1

ISDN Drop Back Busy: Drop Back Disabled (DBD)

ISDN Off-Hook Queuing Option: ☐

Off-Hook Queuing Allowed: ☐

Call Back Queuing Allowed: ☐

**VNS Options**

Entry is a VNS Route: ☐

Submit Refresh Delete

Figure 4: Data Entry of a Route List Block

- 3 Select **Dialing and Numbering Plans > Electronic Switched Network** again from the left pane to display the **Electronic Switched Network (ESN)** screen. Select **Distant Steering Code (DSC)**

- Links
  - Virtual Terminals
- System
  - + Alarms
  - Maintenance
  - Core Equipment
    - Loops
    - Superloops
    - MSDLMISP Cards
    - Conference/TDS/Multifrequen
    - Tone Senders and Detectors
  - Peripheral Equipment
  - IP Network
    - Nodes: Servers, Media Cards
    - Maintenance and Reports
    - Media Gateways
    - Zones
    - Host and Route Tables
    - Network Address Translation
    - QoS Thresholds
    - Personal Directories
    - Unicode Name Directory
  - + Interfaces
    - Engineered Values
    - + Emergency Services
    - + Software
  - Customers
  - Routes and Trunks
    - Routes and Trunks
    - D-Channels
    - Digital Trunk Interface
  - Dialing and Numbering Plans
    - **Electronic Switched Network**
    - Flexible Code Restriction

## Electronic Switched Network (ESN)

- Customer 00
  - Network Control & Services
    - Network Control Parameters (NCTL)
    - ESN Access Codes and Parameters (ESN)
    - Digit Manipulation Block (DGT)
    - Home Area Code (HNPA)
    - Flexible CLID Manipulation Block (CMDB)
    - Free Calling Area Screening (FCAS)
    - Free Special Number Screening (FSNS)
    - Route List Block (RLB)
    - Incoming Trunk Group Exclusion (ITGE)
    - Network Attendant Services (NAS)
  - Coordinated Dialing Plan (CDP)
    - Local Steering Code (LSC)
    - **Distant Steering Code (DSC)**
    - Trunk Steering Code (TSC)
  - Numbering Plan (NET)
    - Access Code 1
      - Home Location Code (HLOC)
      - Location Code (LOC)
      - Numbering Plan Area Code (NPA)
      - Exchange (Central Office) Code (NXX)
      - Special Number (SPN)
      - Network Speed Call Access Code (NSCL)
    - Access Code 2
      - Home Location Code (HLOC)
      - Location Code (LOC)
      - Numbering Plan Area Code (NPA)

The **Distant Steering Code** screen is displayed. This is an example of steering code 7000. For the **Route List to be accessed for trunk steering code** field, select the route list index previously configured from the drop-down list. Retain the default values in all remaining fields and click on **Submit**.

- Software
  - Call Server PEPs
  - Loadware PEPs
  - File Upload
  - IP Phone Firmware
  - Voice Gateway Media Card
  - Media Cards PEPs
  - Plug-Ins
- Customers
- Routes and Trunks
  - Routes and Trunks
  - D-Channels
  - Digital Trunk Interface
- Dialing and Numbering Plans
  - [Electronic Switched Network](#)
  - Flexible Code Restriction
  - Incoming Digit Translation
- Phones
  - Templates
  - Reports
  - Views
  - Lists
  - Properties

Managing: **172.16.23.15** Username: admin  
 Dialing and Numbering Plans » [Electronic Switched Network \(ESN\)](#) » Customer 00 » Coordinated Dialing Plan (C

## Distant Steering Code

Distant Steering Code:

Flexible Length number of digits:  (0 - 10)

Display:

Remote Radio Paging Access: ☐

Route List to be accessed for trunk steering code:

Collect Call Blocking: ☐

Maximum 7 digit NPA code allowed:

Maximum 7 digit NXX code allowed:

# Subscriber Administration in UCM

Subscriber administration includes:

- Configure Phones to cover to the MiCollab AM 'pilot' extension
- From the left-pane of Element Manager select **Phones**.
  - Use the **Search Criteria** to select the station to edit a station.
  - Set the following
    - CFXA – Call Forward External = Allowed**
    - FNA = Allowed**
    - FDN** – In the adjacent field enter the Pilot Number, for MiCollab AM
    - HTA – Hunting** set the following:

If you use blind transfers, you must enable HTA. If you use monitored or supervised T-type transfers, do not enable HTA.
    - HUNT** = Enter the Pilot Number, 7000, for MiCollab AM.
    - MWA – Message Waiting at Message Service = Allowed**
- Go to the **Keys** table, Scroll down to Key No. **16**. Use the adjacent drop-down menu in the Key Type column and select **MWK – Message Waiting**. In the **Message Center DN** field, enter the Pilot Number of 7000 for MiCollab AM.
- Ensure you **SAVE** and backup your configuration.