

MiCollab Advanced Messaging Avaya Communication Manager SIP Station with Session Manager Integration Technical Note

For version 9.1 and above

Notice

The information contained in this document is believed to be accurate in all respects but is not warranted by Mitel Networks™ Corporation (MITEL®). Mitel makes no warranty of any kind with regards to this material, including, but not limited to, the implied warranties of merchantability and fitness for a particular purpose. The information is subject to change without notice and should not be construed in any way as a commitment by Mitel or any of its affiliates or subsidiaries. Mitel and its affiliates and subsidiaries assume no responsibility for any errors or omissions in this document. Revisions of this document or new editions of it may be issued to incorporate such changes.

No part of this document can be reproduced or transmitted in any form or by any means - electronic or mechanical - for any purpose without written permission from Mitel Networks Corporation.

Trademarks

The trademarks, service marks, logos and graphics (collectively "Trademarks") appearing on Mitel's Internet sites or in its publications are registered and unregistered trademarks of Mitel Networks Corporation (MNC) or its subsidiaries (collectively "Mitel") or others. Use of the Trademarks is prohibited without the express consent from Mitel. Please contact our legal department at legal@mitel.com for additional information. For a list of the worldwide Mitel Networks Corporation registered trademarks, please refer to the website: <http://www.mitel.com/trademarks>.

© Copyright 2020, Mitel Networks Corporation

All rights reserved

Contents

Preface	5
References	6
Documentation	6
Documentation Updates	6
Help	7
Document Conventions	7
Features Supported by This Integration	8
Critical Application Considerations	11
Installation Requirements	14
Telephone System Requirements	14
MiCollab AM Requirements	14
Programming the Telephone System	15
Preparing the Telephone System for the Integration	15
Assigning Node IP Addresses in the Communication Manger	15
Creating a SIP Signaling Group	16
Defining the IP Interfaces	17
Creating a SIP Trunk Group	18
Programming MiCollab AM Ports	19
Configuring the SIP Entities on Avaya Servers	21
Configuring the Session Manager Firewall	22
Configuring the Routing Policies	24
Adding the MiCollab AM Port User Definitions	25
Creating a Hunt Group and Pilot Number	27
Creating a Coverage Path	29
Creating a Route Pattern	29
Modifying Digit Conversion Tables	30
Defining the Telephone System Location	31
Programming Subscriber Telephones	31
Verifying the Local Survivable Processor settings of Telephone System	37
Configuring MiCollab AM	41
Configuring MiCollab AM for the Integration During Initial Installation	41
Configuring Existing MiCollab AM for the Integration	45

Configuring MiCollab AM for SIP Failover	50
Changing the Network Binding Order on the MiCollab AM Platform	53
Windows Server 2012 R2	53
Windows Server 2016 / 2019	54
Configuring Quality of Service (QoS)	55

Preface

This Integration Technical Note (ITN) is written for MiCollab Advanced Messaging (MiCollab AM) certified technicians who are experienced with 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 Aura Communication Manager Telephone system.

This document describes how to integrate MiCollab AM with an Avaya Aura Communication Manager Telephone system, using the Session Initiation Protocol (SIP) integration. This integration operates exclusively over an IP-based network. It uses no analog or digital voice telephony ports, but instead passes voice communication and signaling information over the network.

The Avaya Communication Manager SIP integration consists of the following five major components:

- The Avaya Aura Communication Manager
- The Avaya Media Gateway
- Avaya Aura Session Manager server
- Avaya Aura System Manager server
- MiCollab AM

MiCollab AM registers its SIP ports as terminals or endpoints with the Avaya Aura Session Manager server. The SIP ports are configured as Off-Premises Stations (OPS) and are assigned into a hunt group of the PBX.

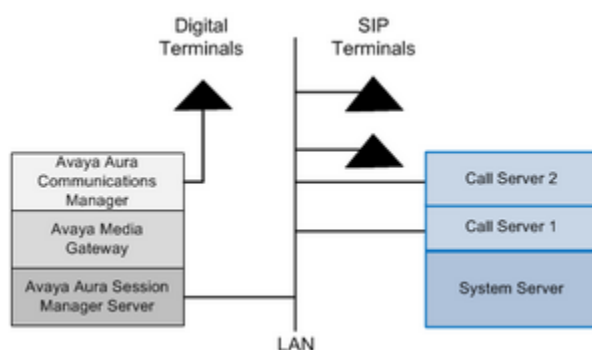


Figure 1. Terminals

Calls intended for MiCollab AM, whether direct or forwarded, are directed to the pilot number of this hunt group. The Call Server uses these same lines to place or transfer calls to the telephone system.

MiCollab AM sets and clears message-waiting indicators (MWIs) by transmitting SIP messages to the Aura Session Manager server. As a result, MWI operations never restrict the number of lines available for calls.

The integration process consists of configuring SIP support on the media server, configuring the telephone system at the gateway, configuring subscriber workstations at the media server, and configuring MiCollab AM. This document also describes the critical application considerations with which you should be familiar before you begin work on the integration.

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 MiCollab AM Documentation Library includes the following documents and resources:

- **Administration Documentation.** Available as a PDF only. Contains the following:
 - **Administration Guides.** Available as a PDF only. Contains administrative guides for administrators about how to manage and configure the messaging system.
 - **Quick Reference Cards (QRC).** Contains shortcuts and quick instructions telling subscribers how to access and use the messaging system.
 - **User Guides.** Available as a PDF only. Contains user guides for subscribers about accessing the messaging system and checking and sending messages.
- **Server Documentation.** Available as a PDF only. Contains the following:
 - **Developer Resources.** Contains programming guides and API references for developers for integrating the server clients and web applications with MiCollab AM.
 - **Installation and Configuration.** Available as a PDF only. Contains installation and configuration guides for server administrators about how to install and configure the messaging system.
 - **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.
 - **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 documents and program files from our partner web site: www.mitel.com

Help

The primary source of information about MiCollab AM is the online help available within any of its administrative utilities. You can access **Help** by clicking the **Help** button in the dialog box or window in which you are working.

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** Titles of other documents are shown in italics.

Example: See the *System Installation and Configuration Guide*.

- **User Interface (UI) Element Names.** Names of UI elements such as dialog boxes, windows, screens, menu items, tabs, buttons, and icons 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 MiCollab AM 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.

For more detailed documents, refer to the following list of references:

Table 1. References

Document Type	Document Title
Administration Documentation	<i>System Administration Guide</i>
Server Documentation	<i>System Installation and Configuration Guide</i>
Online help	MiCollab AM online help system

Features Supported by This Integration

The following tables list the features supported using the Avaya Aura Communication Manager SIP integration.

Table 2. 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	Yes

Table 3. Integration features supported for Avaya Aura Communication Manager SIP

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	Yes	
Overflow from MiCollab AM to attendant	Yes	
Overflow to MiCollab AM from attendant	Yes	
PBX-provided disconnect signaling	Yes	
Revert to operator	Yes	
Silence Timeout	Yes	
SRTP	No	Note 3
TLS	No	Note 3
Transfers, blind	Yes	
Transfers, confirmed	Yes	
Transfers, fully supervised	Yes	
Transfers, monitored	Yes	
Trunk ID for call routing	No	
Multiple Integrations	Yes	Note 4

NOTES

1. Available only when 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 next section, [Critical Application Considerations](#).

3. 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.

To create secure connections, use TLS 1.3 (recommended where available) or 1.2 for the System Server and Call Servers.

4. 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 populate Line extension numbers on the **Lines** tab before starting MiCollab AM or the integration will fail. The extension numbers are registered as SIP stations with the IP PBX during system startup.
- Configure the MiCollab AM **Incoming Hunt Mode** in the **Switch Section Options** dialog box. The hunt mode must match the type of hunting provided by the IP PBX. This helps to alleviate any glare conditions between the IP PBX and the Call Server. The default mode is Terminal.
- You must configure the Hunt Group Access Code in the **Switch Section Options** dialog box. This code cannot conflict with extensions.

For example:

You can use 6000 for the Hunt Group Access Code and start MiCollab AM extensions with 6001.

- 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. 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. For more information, refer to [Changing the Network Binding Order on the MiCollab AM Platform](#).
- 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 SIP Domain Name in the **Integration Options** dialog box must match the domain name configured in the telephone system and on the TFTP server. This value is case sensitive.
- 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.

IMPORTANT Enabling this parameter causes processing overhead and should only be enabled when necessary.

- 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 only to the specific Call Server on which their call is queued.

- If the Avaya H.323 telephones do not provide end-to-end DTMF to MiCollab AM, disable the system-wide parameter IP Shuffling in the System Parameters programming section of the Communication Manager. This is particularly important where multiple Avaya Medpro's are in use. Be sure the parameter, Hairpinning is enabled for all H.323 telephones and the SIP Signaling Group supporting MiCollab AM.
- In an environment with Avaya Communication Manager, Session Manager, Avaya/Nortel CS1000 Call server and Signaling Server, there are limitations in transferring to Avaya/Nortel phones registered to the Signaling Server.
 - If another application requiring a different configuration for G430 or G450 gateways will use Session Manager, a separate gateway will be required.
- Avaya Survivable Remote Server (formerly called Local Survivable Processor [LSP]) support – for Secondary or tertiary failover server scenarios.
 - SIP endpoints supported when defined to have the secondary Survivable Remote Server in Network region. Digital and Analog endpoints only supported if they reside within the G430 or G450 to which the Communication Manager has been set to be defined as the survivable server.

NOTE H.323 endpoints are not supported in a failover scenario.

- For additional clarification on survivable server setup for the Avaya Communication Manager, see the following Avaya Documents:
 - *Deploying Avaya Aura Communication Manager* – Release 7.1.1 Issue 2
 - *Converting Avaya Servers and Gateways* (Document ID: 03-602884)
 - *Avaya Aura® Communication Manager Survivability Options* – Release 8.0 Issue 2
 - Know information on installing and configuring survivable core servers and migrating a main server to a survivable core server.
 - *Administering Network Connectivity on Avaya Aura Communication Manager* Release 8.0 Issue 2
- Avaya Survivable Core Server (formerly called Enterprise Survivable Server [ESS]) support – for Secondary or tertiary failover server scenarios.
 - Each Survivable Core Server is administered on the main server, which can be either:
 - a S8500/S8700/S8800 Media Server
 - a server deployed on a VMWare environment (as of Avaya Communication Manager 6.0 and above)
 - a server deployed on the Avaya Appliance Virtualization Platform (AVP) (as of Avaya Communication Manager 7.1 and above)
 - The Avaya Survivable Core Server option is available from Avaya Communication Manager 3.0 and above and requires a software license for the main server and each Survivable Core Server to activate the Survivable Core Server feature. Configuration of Avaya equipment should be performed by Avaya and/or Avaya business partners.
- Direct IP-to-IP communication settings updates for Avaya Aura:

- If the direct IP-to-IP setting is set as 'n', it will route calls directly through the Session Manager and bypass any MedPro configured on the TDM bus. This setting is also required to be used for TLS calls which will route through the Session Manager as well.
- If the direct IP-to-IP setting is set as 'y' it will route the call through the TDM Bus (MedPro) resources.
- When you are using the MedPro on the TDM bus as your IP resource, and you are calling between two SIP endpoints (when a SIP endpoint calls another SIP endpoint), the media stream will initially pass through a TDM resource.

However, once the call has been established and the TDM resource is no longer required, the call is "shuffled" away from the TDM bus and IP flows directly between the two SIP endpoints. This will open up the TDM resource, releasing time-slots on the voice bus, and allow IP media to flow more efficiently

A few rules apply:

- Both SIP endpoints must be administered to allow shuffling. For Avaya phones, enable Intra-region IP-IP Direct Audio, Inter-region IP-IP Direct Audio, and IP Audio Hairpinning for the IP Network Region, and Direct-IP in System Features and the Signaling Group.
- The endpoints must be in the same LAN region or in interconnected LAN regions. The inter-region connection management rules must be met. There is at least one codec in common between the codec lists of the endpoints involved and the Internetwork region connection management codec list.
- The endpoints don't have to do anything special to initiate shuffling. It's all handled by the gateway. The endpoints will know when shuffling is occurring when they receive re-INVITE messages with new media descriptions.
- For additional clarification on network regions defined in the Avaya Communication Manager, see the following two Avaya Documents:
 - *Administering Network Connectivity on Avaya Aura Communication Manager* Release 8.0 Issue 2
 - *Avaya Communication Manager Network Region Configuration Guide* (Document ID: 103244)
- MiCollab AM 9.1 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 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

- Avaya Aura Communication Manager 8.1 or prior
- Avaya Aura Session Manager 8.1 and prior
- Avaya Aura G430 or G450 Media Gateways with possible additional DSP resources required; or G650 with C-LAN and IP Media Processor with current firmware update, to handle voice processing tasks
- One Off-Premises Station (OPS) license per MiCollab AM port
- One Administered SIP license per MiCollab AM port
- Avaya Aura G430/G450 Media Gateways with S8300E for Survivable Remote Server (formerly LSP) failover server scenario. Survivable Remote Server (SIP endpoints supported in this configuration only)
- Avaya Communication Manager 3.0 and above with S8500/S8700/S8800 Media Server for Survivable Core Server (formerly ESS) failover server scenario
- Avaya Communication Manager 6.0 and above with VMWare environments for Survivable Core Server (formerly ESS) failover server scenario
- Avaya Communication Manager 7.1 and above with AVP environments for Survivable Core Server (formerly ESS) failover server scenario

MiCollab AM Requirements

- MiCollab AM version 9.1
- MiCollab AM software key or feature file with the Avaya Communication Manager SIP integration enabled and one Virtual SIP and RTP license enabled for each port involved in the integration
- One 100 Mbps or faster network interface card

Programming the Telephone System

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. Settings that are critical to the integration appear in boldface.

The installing technician should be familiar with programming the telephone system. For detailed information on programming and installing the telephone system, refer to the Avaya documentation.

Preparing the Telephone System for the Integration

Before beginning the integration, make sure that the following configuration tasks are completed on the telephone system.

- Verify the PBX has enough Administered SIP Trunk and OPS Extension licenses available for use with MiCollab AM
- Assigning IP node names and addresses to the components of the Communication Manager, and the Session Manager server platforms
- Defining IP interfaces
- Administering IP network regions

For more information on completing these tasks, refer to the documentation accompanying your telephone system.

Assigning Node IP Addresses in the Communication Manger

Assign the IP addresses on the Communication Manager. These IP address assignments are for communication between the Communication Manager, the Session Manager, and the gateway. Use the command, **change node-names ip** to assign the IP addresses required for the installation.

The following is an example of Assigning Node IP Addresses:

change node-names ip		Page 1 of 2	
		IP NODE NAMES	
Name	IP Address		
AVAYACMSRV	172.16.20.122		
CM-SimplexESS2	172.16.20.150		
CStevenson	10.2.6.13		
CX85LYNC01	172.16.5.10		
CX87AVA	172.16.4.50		
CXCALL05	172.16.4.221		
CXECALL01CS	172.16.26.58		
CXHACALL01	172.16.4.118		
CXHACALL02	172.16.4.82		
CXHASYSTEM	172.16.4.189		
CXsipMeridian	172.16.4.65		
CurtsSRM	172.16.7.161		
Gateway001	172.16.20.1		
IPOffice	172.16.21.101		
Integautotest01	172.16.10.8		
Integautotest02	172.16.10.6		
(16 of 30 administered node-names were displayed)			

Creating a SIP Signaling Group

Using the Communication Manager Element Cut-Through from Avaya System Manager or a SAT terminal, define a Signaling Group associating the Communication Manager and Session Manager servers, as shown in the following example.

To create SIP Signaling Group:

- 1 Specify a node name for the Session Manager and a listening port accessible to both the Session Manager and Communication Manager servers.
- 2 Specify the name of the domain on which the MiCollab AM platform is located; this will be site specific.
- 3 Enable direct IP-to-IP Audio Connections and IP Audio Hairpinning.
- 4 Specify 'rtp-payload' as the method that the telephone system will use in transmitting DTMF tone sequences over the IP network.

change signaling-group 1 Page 1 of 1

SIGNALING GROUP

Group Number: 1 Group Type: sip

IMS Enabled? ☐ Transport Method:

Q-SIP? ☐

IP Video? ☐ Enforce SIPS URI for SRTP? ☐ y

Peer Detection Enabled? ☒ Peer Server: SM

Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? ☐ y

Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? ☐ n

Alert Incoming SIP Crisis Calls? ☐ n

Near-end Node Name: Far-end Node Name:

Near-end Listen Port: Far-end Listen Port:

Far-end Network Region:

Far-end Domain:

Incoming Dialog Loopbacks: Bypass If IP Threshold Exceeded? ☐ n

DTMF over IP: RFC 3389 Comfort Noise? ☐ n

Session Establishment Timer(min): Direct IP-IP Audio Connections? ☒ y

Enable Layer 3 Test? ☒ y IP Audio Hairpinning? ☒ y

H.323 Station Outgoing Direct Media? ☐ n Initial IP-IP Direct Media? ☐ n

Alternate Route Timer(sec):

Defining the IP Interfaces

Define the IP Authoritative Domain and IP interfaces.

IMPORTANT Be sure the Authoritative Domain name is the same throughout the Session Manager server, Communication Manager, and MiCollab AM programming; this is site specific.

Routing / Elements / Routing / Domains

Domain Management

New Edit Delete Duplicate More Actions

1 Item

Name	Type
<input checked="" type="checkbox"/> blvu.avstlabs.local	sip

Select : All, None

Figure 2. Avaya Aura System Manager

display ip-network-region 1
Page 1 of 20

IP NETWORK REGION

Region: 1 NR Group: 1
 Location: 1 Authoritative Domain: blvu.avstlabs.local
 Name: NR 1 Stub Network Region: n

MEDIA PARAMETERS

Intra-region IP-IP Direct Audio:	yes
Inter-region IP-IP Direct Audio:	yes
IP Audio Hairpinning?	y

Codec Set: 1
 UDP Port Min: 2048
 UDP Port Max: 3329

DIFFSERV/TOS PARAMETERS

Call Control PHB Value: 46
 Audio PHB Value: 46
 Video PHB Value: 26

802.1P/Q PARAMETERS

Call Control 802.1p Priority: 6
 Audio 802.1p Priority: 6
 Video 802.1p Priority: 5

AUDIO RESOURCE RESERVATION PARAMETERS

RSVP Enabled? n

H.323 IP ENDPOINTS

H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
 Keep-Alive Interval (sec): 5
 Keep-Alive Count: 5

Creating a SIP Trunk Group

Create a Trunk Group and populate it with the ports that support the MiCollab AM integration, as shown in the following two examples.

To create a SIP Trunk Group:

- 1 Specify **Group Type** as *sip* and **Service Type** as *tie*.
- 2 Associate the new Trunk Group with the Signaling Group you created previously.

TRUNK GROUP
Page 1 of 21

Group Number: 1 Group Type: sip CDR Reports: y
 Group Name: sip trunk - tls COR: 1 TN: 1 TAC: 201
 Direction: two-way Outgoing Display? n
 Dial Access? n Night Service:
 Queue Length: 0
 Service Type: tie Auth Code? n
 Member Assignment Method: auto
 Signaling Group: 1
 Number of Members: 255

- 3 The number of trunk members is defined on the first page and will be auto-populated as shown in the following figure.

Page 5 of 21

TRUNK GROUP

Administered Members (min/max): 1/ 255

Total Administered Members: 255

GROUP MEMBER ASSIGNMENTS

Port	Name
1: T00001	sip trunk
2: T00002	sip trunk
3: T00003	sip trunk
4: T00004	sip trunk
5: T00005	sip trunk
6: T00006	sip trunk
7: T00007	sip trunk
8: T00008	sip trunk
9: T00009	sip trunk
10: T00010	sip trunk
11: T00011	sip trunk
12: T00012	sip trunk
13: T00013	sip trunk
14: T00014	sip trunk
15: T00015	sip trunk

Programming MiCollab AM Ports

To assign the station ports to the trunk group add IP stations to the system. Assign an Extension Number and a Name to each port and set the Station Type to **4602+**.

Page 1 of 4

STATION

add station 5025
 Extension: 5025
 Type: 4602+
 Port: IP
 Name: 5025

Lock Messages? ☐ n
 Security Code:
 Coverage Path 1:
 Coverage Path 2:
 Hunt-to Station:

BCC: 0
 TN: 1
 COR: 1
 COS: 1
 Tests? ☐ y

STATION OPTIONS

Loss Group: 19
 Speakerphone: 1-way
 Display Language: english
 Survivable GK Node Name:
 Survivable COR: internal
 Survivable Trunk Dest? ☐ y

Time of Day Lock Table:
 Personalized Ringing Pattern: 1
 Message Lamp Ext: 5025
 Mute Button Enabled? ☐ y
 Media Complex Ext:
 IP SoftPhone? ☐ n
 IP Video? ☐ n

Short/Prefixed Registration Allowed: default

Declare the ports you have defined, and any SIP-based extensions that MiCollab AM subscribers use, as parts of an external integration. To do this, enter the command **change off-pbx-telephone station-mapping <number>**, where <number> is the station number for any port involved in the integration.

The command displays a table that you can use to configure all of the station ports you need to change. Associate all of the ports with the trunk group you defined earlier, as the following two examples demonstrate.

(Page 1)

change off-pbx-telephone station-mapping 5025
Page 1 of 3

STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

Station Extension	Application	Dial Prefix	CC	Phone Number	Trunk Selection	Config Set	Dual Mode
5025	OPS			5025	TG1	1	

(Page 2)

change off-pbx-telephone station-mapping 5025
Page 2 of 3

STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

Station Extension	Appl Name	Call Limit	Mapping Mode	Calls Allowed	Bridged Calls	Location
5025	OPS	2	both	all	none	

Configuring the SIP Entities on Avaya Servers

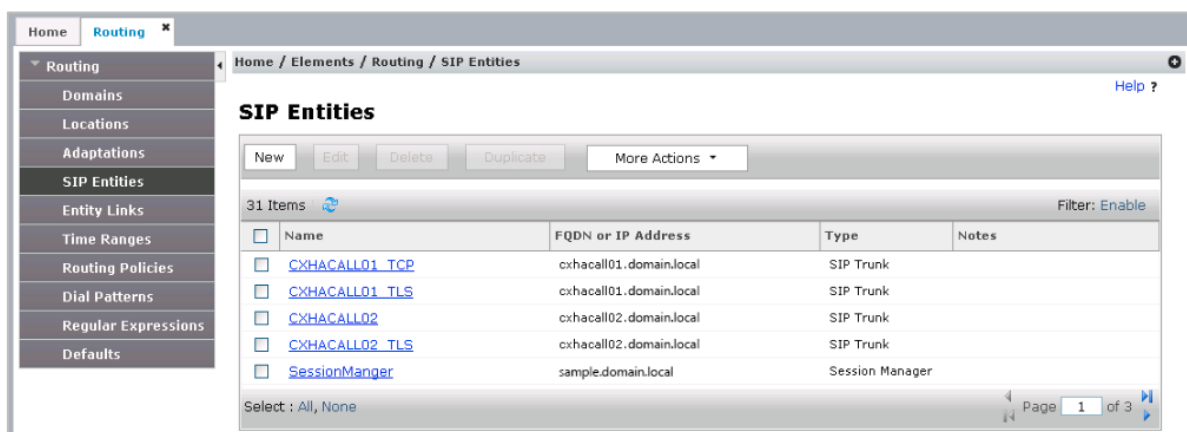
Verify the connection between the Communications Manager and Session Manager servers on the Session Manager server. Make sure the Communication Manager server Interface reflects the same IP addresses as the Media Server Interface.

Configure a SIP Entity for the Communication Manager server and a SIP Entity for the Session Manager server.

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

To configure the SIP Entities:

- 1 Log in to the System Manager, and go to **Elements > Routing > SIP Entities**. The **SIP Entities** page displays.



- 2 From the SIP Entities table, select **Session Manager**, or select the **Session Manager** checkbox and then click **Edit**. The **SIP Entity Details** page displays.
- 3 On the **SIP Entity Details** page, go to the **Port** section, and then assign the **Listen Port** number, select **TCP** or **UDP** as the **Protocol**, and select the **Default Domain**.



IMPORTANT The port number and the protocol you enter here must match the SIP Server Port and the Transport for Outgoing SIP Message of the Required Parameters fields on the MiCollab AM **Integration Options** dialog box.

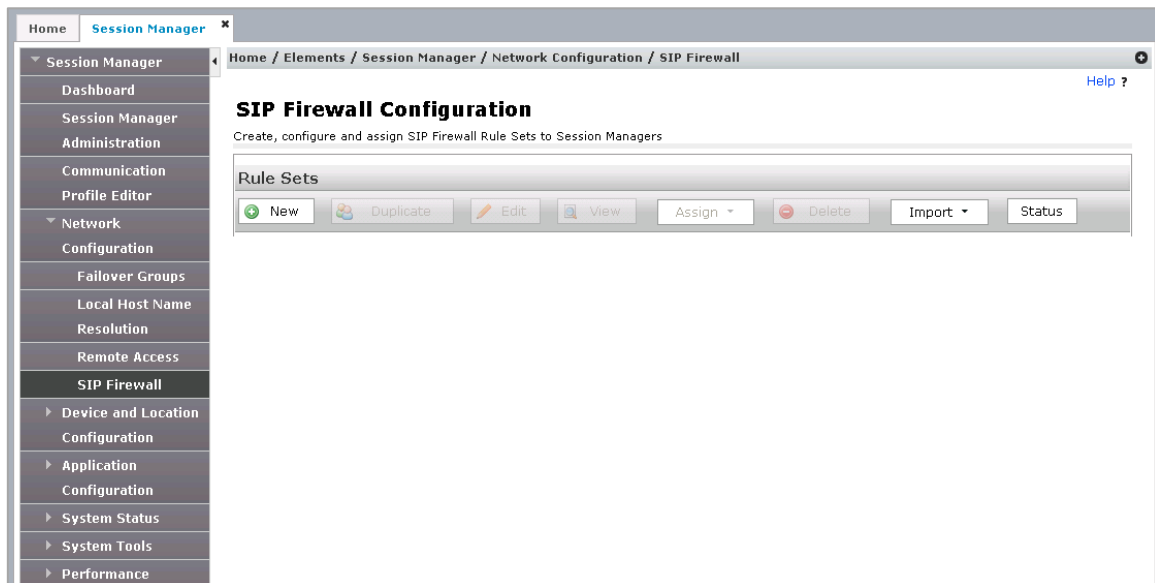
The default protocol is **TCP** and the default port number is **5060**. For more information, refer to the [Configuring MiCollab AM](#) section.

Configuring the Session Manager Firewall

Verify that the MiCollab AM server's IP address is not blocked by the Session Manager firewall. If there is more than one Call Server participating in the integration, ensure that the Session Manager server does not block the IP Address of any Call Server.

To configure the Firewall:

- 1 Log in to the System Manager, and go to **Elements > Session Manager > Network Configuration > SIP Firewall**. The **SIP Firewall Configuration** page displays.



- 2 On the **SIP Firewall Configuration** page, in the **Rule Sets** section, click **New**. The **Rule Set** page displays.

The screenshot shows the 'Rule Set' configuration page. At the top, there are 'Commit' and 'Cancel' buttons. Below them is the instruction 'Edit or view SIP Firewall Rule Set whitelist, blacklist, and rules.' The form includes fields for '*Name' and 'Description'. There are three tabs: 'Rules' (selected), 'Blacklist', and 'Whitelist'. Under the 'Rules' tab, there is an 'Enabled' checkbox. Below the tabs are buttons for 'New', 'Edit', 'View', 'Delete', 'Up', and 'Down'. At the bottom, there is a table with the following headers: 'Enabled', 'Name', 'Action Type', 'Log Type', and 'Log Message'. The table is currently empty.

- 3 In the **Name** field, select or type *Rule Set for SessionManager*.
- 4 In the **Rules** tab, click **New**. The **Rule** page displays.

Rule Cancel Done

General | IP Layer Match Options | SIP Layer Match Options | IP/SIP Layer Track | Threshold | Connection |
Expand All | Collapse All

General

Enabled: ☒

*Name:

*Action Type:

Log Type:

Log Message:

IP Layer Match Options

Protocol:

Remote IP Address:

Remote Port:

Local Port:

SIP Layer Match Options

New Delete

<input type="checkbox"/>	Key Type	Value Type	Value
<input type="checkbox"/>			

IP/SIP Layer Track

Track:

Threshold

Count (packets):

Period (secs):

Timeout (secs):

Connection

Connection Type:

*Required Cancel Done

- 5 On the **Rule** page, configure the firewall appropriately to allow access to each Call Server in the integration. When finished, click **Done**.
- 6 Click the **Whitelist** tab and click **New**.

Rule Set Commit Cancel

Edit or view SIP Firewall Rule Set whitelist, blacklist, and rules.

*Name:

Description:

*SM Type:

Rules **Blacklist** **Whitelist**

Enabled: ☒

New Delete

<input type="checkbox"/>	Key	Value	Mask
<input type="checkbox"/>	Remote IP Address	192.11.13.2	255.255.255.255
<input type="checkbox"/>	Remote IP Address	172.16.4.127	255.255.255.0
<input type="checkbox"/>	Remote IP Address	172.16.4.109	255.255.255.0

Select: All, None

- 7 Add a Remote IP address for the MiCollab AM to ensure that the IP address is not being blocked by a firewall. Click **Commit**.

Configuring the Routing Policies

Configure the routing policies and the dialing pattern for the MiCollab AM hunt group and the non-SIP subscriber directory number range.

To configure the routing policies:

- 1 Log in to the System Manager, and go to **Elements > Routing > Routing Policies**. The **Routing Policies** page displays.
- 2 On the **Routing Policies** page, click **New** to create a new Routing Policy. The **Routing Policy Details** page displays.

Home / Elements / Routing / Routing Policies

Routing Policy Details

Commit Cancel

General

* Name: Example

Disabled: ☐

* Retries: 0

Notes:

- 3 In the **General** section, enter the name for the policy and number of retries.

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
CX_Buffalo_TLS	172.16.4.127	SIP Trunk	SIP trunk to CX Buffalo

Time of Day

Add Remove View Gaps/Overlaps

1 Item

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time
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

Select : All, None

Add Remove

7 Items Filter: Enable

Pattern	Min	Max	Emergency Call	SIP Domain	Originating Location	Notes
40	4	4	<input type="checkbox"/>	-ALL-	Bothell_LAB	
41	4	4	<input type="checkbox"/>	-ALL-	-ALL-	CS1000E SIP Trunk
42	4	4	<input type="checkbox"/>	-ALL-	Bothell_LAB	
43	4	4	<input type="checkbox"/>	-ALL-	Bothell_LAB	CX_to_ML150-01
56	4	4	<input type="checkbox"/>	blvu.avstlabs.local	-ALL-	SIP TLS trunk to Buffalo CX
5700	4	4	<input type="checkbox"/>	blvu.avstlabs.local	-ALL-	

- 4 In the **SIP Entity as Destination** and **Dial Pattern** sections, configure the options for the MiCollab AM hunt group number and the non-SIP subscriber directory number range according to the requirements of the site. This enables the non-SIP calls to route to the Communication Manager.

IMPORTANT When you add user definitions for the MiCollab AM ports, you must assign the same password to all users and all ports. If you do not, the integration cannot function correctly.

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

Adding the MiCollab AM Port User Definitions

Add a user definition for each MiCollab AM port in the integration.

To add a user definition:

- 1 Log in to the System Manager, and go to **Users > User Management > Manage Users**. The **User Management** page displays.
- 2 On the **User Management** page, click **New**. The **New User Profile** page displays in the **Identity** tab.

The screenshot shows the 'New User Profile' page in the System Manager. The left sidebar contains a 'User Management' menu with options: Manage Users, Public Contacts, Shared Addresses, System Presence, ACLs, Communication, Profile Password, and Policy. The main content area has a breadcrumb trail: Home / Users / User Management / Manage Users. The title is 'New User Profile'. Below the title are four tabs: Identity (selected), Communication Profile, Membership, and Contacts. Under the 'Identity' tab, there is a 'User Provisioning Rule' dropdown menu. Below that, the 'Identity' section contains several input fields: Last Name (with a red asterisk), Last Name (Latin Translation), First Name (with a red asterisk), First Name (Latin Translation), Middle Name, Description (with a red asterisk), and Login Name (with a red asterisk). The 'Last Name' and 'First Name' fields are highlighted with red borders.

- 3 In the **Identity** tab, fill in the appropriate fields.
- 4 Click the **Communication Profile** tab.

New User Profile

Identity * **Communication Profile** Membership Contacts

Communication Profile

Communication Profile Password:

Confirm Password: [Generate](#)

Name

☒ Primary

Select : None

* Name:

Default : ☒

Communication Address

Type	Handle	Domain
No Records found		

☐ Session Manager Profile ▶

☐ CM Endpoint Profile ▶

☐ CS 1000 Endpoint Profile ▶

☐ CallPilot Messaging Profile ▶

- 5 In the **Communication Profile** section, enter the same numeric password for MiCollab AM.

NOTE The password is required later to configure the MiCollab AM integration.

- 6 In the **Communication Address** section, click **New**. The options for adding a new communication address display.

Communication Address

Type	Handle	Domain
No Records found		

Type:

* Fully Qualified Address: @

- 7 From the **Type** drop-down menu, select **Avaya SIP**.
- 8 In the **Fully Qualified Address** fields, enter the extension number assigned to the port and the domain. And then click **Add**.

NOTE Assign each port the same extension number as you assigned it in the Element Manager configuration screens.

- 9 In the **Session Manager Profile** section, select the arrow ▶ to open the section.

☒ **Session Manager Profile** ▼

SIP Registration

Primary Session Manager

Secondary Session Manager

Survivability Server

Max. Simultaneous Devices ▼

Block New Registration When Maximum Registrations Active? ☐

Application Sequences

Origination Sequence ▼

Termination Sequence ▼

Call Routing Settings

Home Location ▼

Conference Factory Set ▼

Call History Settings

Enable Centralized Call History? ☐

10 In the **Session Manager Profile** section:

- a** In the **SIP Registration** section, select the **Primary Session Manager**.
- b** In the **Call Routing Settings** section, select the **Home Location**.

11 No administration is required for the **CM Endpoint Profile** section.

12 Click **Commit**.

Creating a Hunt Group and Pilot Number

Define a Hunt Group for MiCollab AM and assign a Pilot Number to it that is not associated with any port or extension. Set the ISDN/SIP Caller Display to **mbr-name** to allow the stations in the group to display the name of the group member receiving the call.

Add all MiCollab AM port extensions to the new hunt group. The following examples show a typical hunt group configuration for this integration.

(Page 1)

add hunt-group 40Page 1 of 60

HUNT GROUP

Group Number:	40	ACD?	<input type="text" value="n"/>
Group Name:	<input type="text" value="SIP VM"/>	Queue?	<input type="text" value="n"/>
Group Extension:	<input type="text" value="5030"/>	Vector?	<input type="text" value="n"/>
Group Type:	<input type="text" value="ucd-mia"/>	Coverage Path:	<input type="text"/>
TN:	<input type="text" value="1"/>	Night Service Destination:	<input type="text"/>
COR:	<input type="text" value="1"/>	MM Early Answer?	<input type="text" value="n"/>
Security Code:	<input type="text"/>	Local Agent Preference?	<input type="text" value="n"/>
ISDN/SIP Caller Display:	<input type="text" value="mbr-name"/>		

(Page 2)

add hunt-group 40Page 2 of 60

HUNT GROUP

LWC Reception:	<input type="text" value="none"/>	AUDIX Name:	<input type="text"/>
Message Center:	<input type="text" value="none"/>		

(Page 3)

add hunt-group 40Page 3 of 60

HUNT GROUP

Group Number:	40	Group Extension:	5030	Group Type:	ucd-mia
Member Range Allowed:	1 - 1500		Administered Members (min/max):	0 / 0	
			Total Administered Members:	0	

GROUP MEMBER ASSIGNMENTS

Ext	Name(19 characters)	Ext	Name(19 characters)
1: <input type="text" value="5001"/>		14: <input type="text"/>	
2: <input type="text" value="5002"/>		15: <input type="text"/>	
3: <input type="text" value="5003"/>		16: <input type="text"/>	
4: <input type="text"/>		17: <input type="text"/>	
5: <input type="text"/>		18: <input type="text"/>	
6: <input type="text"/>		19: <input type="text"/>	
7: <input type="text"/>		20: <input type="text"/>	
8: <input type="text"/>		21: <input type="text"/>	
9: <input type="text"/>		22: <input type="text"/>	
10: <input type="text"/>		23: <input type="text"/>	
11: <input type="text"/>		24: <input type="text"/>	
12: <input type="text"/>		25: <input type="text"/>	
13: <input type="text"/>		26: <input type="text"/>	

At End of Member List

Creating a Coverage Path

Define a Coverage Path to use on all MiCollab AM subscriber extensions, as shown in the following example.

In this Coverage Path (9), define the MiCollab AM hunt group (2) as the only Coverage Point. Configure the Coverage Path so that the telephone system forwards calls to this Coverage Point when a subscriber extension is busy, ring-no-answer (RNA), or set to do-not-disturb mode (DND).

add coverage path 9Page 1 of 1

COVERAGE PATH

Coverage Path Number: 9

Cvg Enabled for VDN Route-To Party? Hunt after Coverage?

Next Path Number: Linkage

COVERAGE CRITERIA

Station/Group Status	Inside Call	Outside Call	
Active?	<input type="text" value="n"/>	<input type="text" value="n"/>	
Busy?	<input type="text" value="y"/>	<input type="text" value="y"/>	
Don't Answer?	<input type="text" value="y"/>	<input type="text" value="y"/>	Number of Rings: <input type="text" value="5"/>
All?	<input type="text" value="n"/>	<input type="text" value="n"/>	
DND/SAC/Goto Cover?	<input type="text" value="y"/>	<input type="text" value="y"/>	
Holiday Coverage?	<input type="text" value="n"/>	<input type="text" value="n"/>	

COVERAGE POINTS

Terminate to Coverage Pts. with Bridged Appearances?

Point1: Rng:

Point2:

Point3:

Point4:

Point5:

Point6:

Creating a Route Pattern

Define a call routing pattern as shown in the following example. Associate this pattern with the trunk group you defined earlier under Creating the SIP Trunk Group.

IMPORTANT You must deactivate Secure SIP in this route pattern.

Page 1 of 3

Pattern Number: 1 Pattern Name: SIP

SCCAN? ☐ n Secure SIP? ☐ n Used for SIP stations? ☐ n

Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted	DCS/	IXC
No	Mrk	Lmt	List	Del	Digits			QSIG	
					Dgts			Intw	
1:	1	0						n	user
2:								n	user
3:								n	user
4:								n	user
5:								n	user
6:								n	user

BCC	VALUE	TSC	CA-TSC	ITC	BCIE	Service/Feature	PARM	Sub	Numbering	LAR
0	1	2	M	4	W	Request			Dgts	Format
1:	y	y	y	y	y	n	n	rest		lev0-pvt none
2:	y	y	y	y	y	n	n	rest		none
3:	y	y	y	y	y	n	n	rest		none
4:	y	y	y	y	y	n	n	rest		none
5:	y	y	y	y	y	n	n	rest		none
6:	y	y	y	y	y	n	n	rest		none

Modifying Digit Conversion Tables

Update the Automatic Alternate Routing (AAR) digit analysis table so that the hunt pilot number is a valid dialed string that maps to the route pattern you have defined for the MiCollab AM hunt group, as shown in the following example.

Page 1 of 2

change aar analysis 5

AAR DIGIT ANALYSIS TABLE

Location: all Percent Full: 1

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd
5	7	7	999	aar		n
5030	4	4	1	aar		n
51	4	4	88	aar		n
5300	4	4	53	aar		n
5400	4	4	54	aar		n
5500	4	4	27	aar		n
5600	4	4	1	aar		n
5700	4	4	57	aar		n
5800	4	4	58	aar		n
5900	4	4	59	aar		n
6	7	7	999	aar		n

Update the AAR digit conversion table so that all ranges of extension numbers used for MiCollab AM integration ports and subscriber extensions are defined as valid extension patterns, as shown in the following example.

change aar digit-conversion 0 Page 1 of 2

AAR DIGIT CONVERSION TABLE

Location: all Percent Full: 0

Matching Pattern	Min	Max	Del	Replacement String	Net	Conv	ANI	Req
0	1	28	0		ars	y		n
1	4	28	0		ars	y		n
4	4	4	0		ext	y		n
5	4	4	0		ext	y		n
6	4	4	0		ext	n		n
8	4	4	0		ext	n		n
x11	3	3	0		ars	y		n
								n
								n
								n
								n
								n
								n
								n
								n
								n
								n
								n
								n
								n
								n
								n

Defining the Telephone System Location

Update the location definition as shown in the following example, so that the definition specifies the route pattern you defined earlier in the [Creating a Route Pattern](#) section.

change locations Page 1 of 1

LOCATIONS

ARS Prefix 1 Required For 10-Digit NANP Calls? ☐ y

Loc No	Name	Timezone	DST	City/ Area	Proxy Sel
		Offset			Rte Pat
1	Main	+ 00	: 00 0	425	1

Programming Subscriber Telephones

Subscriber telephone programming involves setting up initialization parameters for SIP-based telephones and configuring the corresponding extensions in the telephone system.

NOTE There are several ways to setup initialization parameters for 9600 SIP phones. For more information, refer to the *Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones Administrator's Guide* from Avaya, which contains detailed information about initializing switch parameters for 9600 SIP phones.

To create a station definition for subscriber telephones:

- 1 At the Avaya Communication Manager Element Cut-Through or SAT terminal, create a station definition for each subscriber extension as shown in the examples in Step 2.
- 2 Make the MWI LAMP Ext number the same as the station's extension number, and set the **Coverage Path** to the one you created earlier.
- 3 Set the **MWI Served User Type** as, **sip-adjunct**, and then associate the station with the Trunk Group you defined previously.

(Page 1)

change station 4010

Page 1 of 6

STATION

Extension: 4010

Type: 9620SIP

Port: S00005

Name: 4010SIP, stn4010

Lock Messages?

Security Code:

Coverage Path 1:

Coverage Path 2:

Hunt-to Station:

BCC: 0

TN:

COR:

COS:

STATION OPTIONS

Loss Group:

Display Language:

Survivable COR:

Survivable Trunk Dest?

Time of Day Lock Table:

Message Lamp Ext:

IP SoftPhone?

IP Video?

(Page 2)

change station 4010		Page 2 of 6	
FEATURE OPTIONS		STATION	
LWC Reception:	<input type="text" value="spe"/>	Coverage Msg Retrieval?	<input type="text" value="y"/>
LWC Activation?	<input type="text" value="y"/>	Auto Answer:	<input type="text" value="none"/>
CDR Privacy?	<input type="text" value="n"/>	Data Restriction?	<input type="text" value="n"/>
Per Button Ring Control?	<input type="text" value="n"/>	Idle Appearance Preference?	<input type="text" value="n"/>
Bridged Call Alerting?	<input type="text" value="n"/>	Bridged Idle Line Preference?	<input type="text" value="n"/>
Active Station Ringing:	<input type="text" value="single"/>	Restrict Last Appearance?	<input type="text" value="y"/>
H.320 Conversion?	<input type="text" value="n"/>	Per Station CPN - Send Calling Number?	<input type="text" value=""/>
MWI Served User Type:	<input type="text" value="sip-adjunct"/>	EC500 State:	<input type="text" value="enabled"/>
Emergency Location Ext:	<input type="text" value="4010"/>	Coverage After Forwarding?	<input type="text" value="s"/>
Always Use?	<input type="text" value="n"/>	Direct IP-IP Audio Connections?	<input type="text" value="y"/>
		IP Audio Hairpinning?	<input type="text" value="n"/>

- 4 Associate the station with the SIP trunk. This is required for MWI purposes.

(Page 6)

change station 4010		Page 6 of 6	
SIP FEATURE OPTIONS		STATION	
Type of 3PCC Enabled:	<input type="text" value="None"/>		
SIP Trunk:	<input type="text" value="tg1"/>		

- 5 Add the station to the off-pbx-telephone station-mapping. AAR is used for routing of 4011 and 4012 in the AAR form.

STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

Station Extension	Application	Dial Prefix	CC	Phone Number	Trunk Selection	Config Set	Dual Mode
4010	OPS	-		4010	aar	1	
4011	OPS	-		4011	aar	1	
4012	OPS	-		4012	aar	1	
4013	OPS	-		4013	aar	1	
4014	OPS	-		4014	aar	1	
4015	OPS	-		4015	aar	1	
4016	OPS	-		4016	aar	1	
4017	OPS	-		4017	aar	1	
4019	OPS	-		4019	aar	1	
4020	OPS	-		4020	aar	1	
4021	OPS	-		4021	aar	1	
5001	OPS	-		5001	tg99	1	
5002	OPS	-		5002	tg99	1	
5003	OPS	-		5003	tg99	1	

- 6 Add the extension number into the public-unknown-numbering form.

NUMBERING - PUBLIC/UNKNOWN FORMAT

Ext Len	Ext Code	Trk Grp(s)	CPN Prefix	Total CPN Len
4	4			4
5	4	27		4
4	5	1		4
6	8		88	8
7	8	11	555	10
4	4400	11	425111	10
4	5700	57		4
4	5800	58		4

Total Administered: 8
Maximum Entries: 9999

Note: If an entry applies to a SIP connection to Avaya Aura(R) Session Manager, the resulting number must be a complete E.164 number.

Communication Manager automatically inserts a '+' digit in this case.

- 7 To add new users, log in to the System Manager, and go to **Users > User Management > Manage Users**. And the click **New**. The **New User Profile** page displays in the **Identity** tab.

Home / Users / User Management / Manage Users

New User Profile

Identity * Communication Profile Membership Contacts

User Provisioning Rule: [dropdown]

Identity

* Last Name: [text box]
Last Name (Latin Translation): [text box]

* First Name: [text box]
First Name (Latin Translation): [text box]
Middle Name: [text box]

Description: [text box]

* Login Name: [text box]

8 In the **Identity** tab, add in the name and other required fields.

9 Click the **Communication Profile** tab.

Home / Users / User Management / Manage Users

New User Profile

Identity * Communication Profile Membership Contacts

Communication Profile

Communication Profile Password: [text box]
Confirm Password: [text box] [Generate](#)

[New](#) [Delete](#) [Done](#) [Cancel](#)

Name

☒ Primary

Select : None

* Name: Primary [text box]
Default : ☒

Communication Address

[New](#) [Edit](#) [Delete](#)

Type	Handle	Domain
No Records found		

☐ Session Manager Profile

☐ CM Endpoint Profile

☐ CS 1000 Endpoint Profile

☐ CallPilot Messaging Profile

10 In the **Communication Profile** section, add password as required.

11 In the **Communication Address** section, click **New**. The options for adding a new communication address appear.

Communication Address

New Edit Delete

Type	Handle	Domain
No Records found		

Type: Avaya SIP

* Fully Qualified Address: @

Add Cancel

- 12 From the **Type** drop-down menu, select **Avaya SIP**.
- 13 Fill in appropriate address in the **Fully Qualified Address** fields, and then click **Add**.
- 14 Repeat **Steps 10 to 12** to add **Avaya E.164**.
- 15 In the **Session Manager Profile** section, select the arrow ► to open the section.

☒ Session Manager Profile

SIP Registration

Primary Session Manager SessionManger

Secondary Session Manager

Survivability Server

Max. Simultaneous Devices 10

Block New Registration When Maximum Registrations Active? ☒

Application Sequences

Origination Sequence CM_Feature

Termination Sequence CM_Feature

Call Routing Settings

Home Location Bothell_LAB

Conference Factory Set (None)

Call History Settings

Enable Centralized Call History? ☐

- 16 In the **Session Manager Profile** section, fill in the following options:
 - a In the **Primary Session Manger** field, enter or select **Session Manager**.
 - b In the **Application Sequences** section, for the **Origination Sequence** and **Termination Sequence** options, select **CM Features**.
 - c In the **Call Routing Settings** section, for the **Home Location** option, select the appropriate location.
- 17 Select the arrow ► icon at the end of the **CM Endpoint Profile** option to open the section.

☒ **CM Endpoint Profile** ▼

System

Profile Type

Extension

Set Type

Security Code

Port

Voice Mail Number

Preferred Handle

Calculate Route Pattern ☐

Sip Trunk

Enhanced Callr-Info display for 1-line phones ☐

Delete Endpoint on Unassign of Endpoint from User or on Delete User. ☒

Override Endpoint Name and Localized Name ☒

Allow H.323 and SIP Endpoint Dual Registration ☐

- 18** In the **CM Endpoint Profile** section, fill in the appropriate options for **System**, **Profile Type**, **Extension**, **Set Type**, **Security Code**, **Port**, and **Voice Mail Number**.

Verifying the Local Survivable Processor settings of Telephone System

To identify the available survivable processors associated with the Avaya Communication Manager:

- 1 At the Avaya Communication Manager Element Cut-Through or SAT terminal, type the following command. **List Survivable-processor**. If none are listed you do not have a secondary system that will failover and can ignore this section.

SURVIVABLE PROCESSORS						
Record Number	Name/ IP Address	Type	Reg	Act	Translations Updated	Net Rgn
1	AVAYACMSRV 172.16.20.122 No V6 Entry	LSP	y	n	22:00 8/21/2017	2
2	CM-SimplexESS2 172.16.20.150 No V6 Entry	ESS S	y	n	22:00 8/21/2017	1

- 2 Write down the **Record Name** and **IP Address** of the Survivable Processors listed here. You will need them later in the [Configuring MiCollab AM for SIP Failover](#) section.
- 3 Display associated network region(s).

```
display network-region-group 2
```

```
NETWORK REGION GROUP
```

```
Network Region Group: 2
Name: LSP
```

```
display network-region-group 1
```

```
NETWORK REGION GROUP
```

```
Network Region Group: 1
Name: NR 1
```

- 4 You can also verify the screens associated with the Network Regions used for Fail-over within the Avaya Aura Communication Manager.

Display IP-Network-Map (this will show the range of IP addresses associated with each network region excluding the primary region [1])

display ip-network-map

Page 1 of 63

IP ADDRESS MAPPING

IP Address	Subnet Bits	Network Region	Emergency Location	Ext
FROM: 172.16.20.121	/	2	n	
TO: 172.16.20.130				
FROM:	/		n	
TO:				
FROM:	/		n	
TO:				
FROM:	/		n	
TO:				
FROM:	/		n	
TO:				

Display IP-Network-Region 2 (this will show the settings of the network region associated with each Survivable Processor)

display ip-network-region 2

Page 1 of 20

IP NETWORK REGION

Region:	2	NR Group:	2
Location:	1	Authoritative Domain:	blvu.avstlabs.local
Name:	LSP	Stub Network Region:	n
MEDIA PARAMETERS		Intra-region IP-IP Direct Audio:	yes
Codec Set:	1	Inter-region IP-IP Direct Audio:	yes
UDP Port Min:	2048	IP Audio Hairpinning?	y
UDP Port Max:	3329		
DIFFSERV/TOS PARAMETERS			
Call Control PHB Value:	46		
Audio PHB Value:	46		
Video PHB Value:	26		
802.1P/Q PARAMETERS			
Call Control 802.1p Priority:	6		
Audio 802.1p Priority:	6		
Video 802.1p Priority:	5	AUDIO RESOURCE RESERVATION PARAMETERS	
H.323 IP ENDPOINTS		RSVP Enabled?	n
H.323 Link Bounce Recovery?	y		
Idle Traffic Interval (sec):	20		
Keep-Alive Interval (sec):	5		
Keep-Alive Count:	5		

Keep in mind that there are multiple pages of configuration settings for each Network Region.

- 5 On the Avaya Aura System Manager there also are a few screens that should be verified.

From the **Home** screen, select the **Session Manager** from the **Elements** Menu. A screen similar to the following will be displayed, showing the primary server and any Survivable servers.

NOTE Security module will only show the BSM up when the failover server is running.

- 6 Select the **Session Manager Administration** tab to display the Global Settings. Then select the **Session Manager Instances** tab. Select the Session Manager and click on View to see the current settings.

Session Manager Administration

This page allows you to administer Session Manager instances and configure their global settings.

Session Manager Administration

This page allows you to administer Session Manager instances and configure their global settings.

NOTE For additional information, please refer to the Avaya Aura documentation for System Manager and Session Manager.

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:

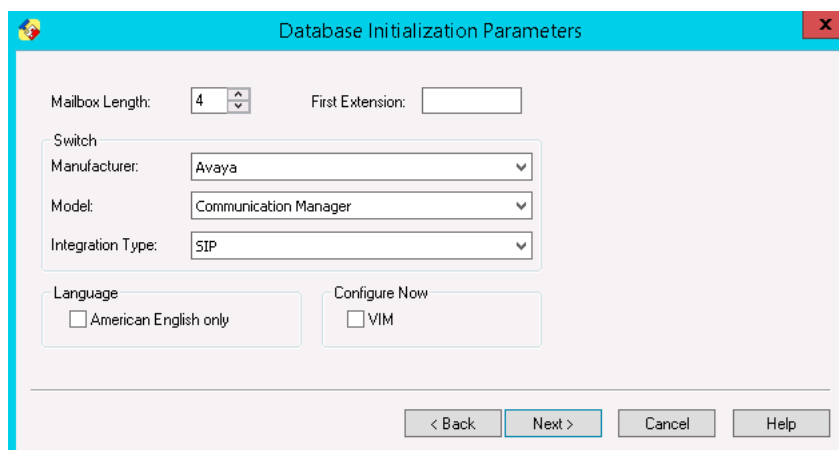
- [Configuring MiCollab AM for the Integration During Initial Installation](#): Integrate the telephone system while you install MiCollab AM for the first time.
- [Configuring Existing MiCollab AM for the Integration](#): Integrate a new telephone system on your exiting MiCollab AM system.

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

Configuring MiCollab AM for the Integration During Initial Installation

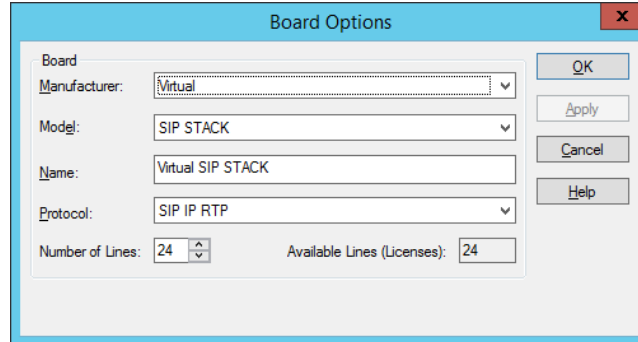
To configure MiCollab AM with the integration for the first time:

- 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** drop-down list, select **Avaya**.
 - d From the **Model** drop-down list, select **Communication Manager**.
 - e From the **Integration Type** drop-down list, select **SIP**.
- 2 Click **Next**. The **Board Options** dialog box displays for the virtual board configuration.

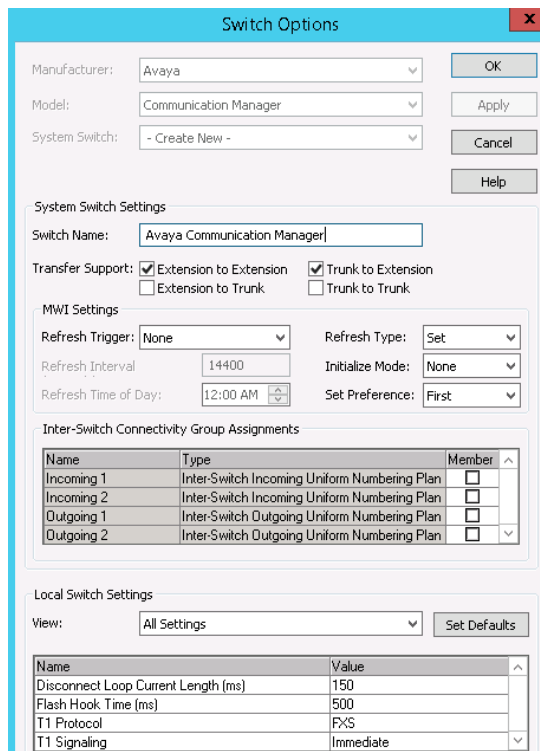


The **Board Options** dialog box is shown with the following configuration:

- Board**
 - Manufacturer: Virtual
 - Model: SIP STACK
 - Name: Virtual SIP STACK
 - Protocol: SIP IP RTP
 - Number of Lines: 24
 - Available Lines (Licenses): 24

Buttons on the right: OK, Apply, Cancel, Help.

- 3 In the **Board Options** dialog box, configure the following options:
- a From the **Manufacturer** drop-down list, select **Virtual**.
 - b From the **Model** drop-down 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** drop-down 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 appears.



The **Switch Options** dialog box is shown with the following configuration:

- System Switch Settings**
 - Manufacturer: Avaya
 - Model: Communication Manager
 - System Switch: - Create New -
 - Switch Name: Avaya Communication Manager
 - Transfer Support:
 - ☒ Extension to Extension
 - ☒ Trunk to Extension
 - ☐ Extension to Trunk
 - ☐ Trunk to Trunk
- MWI Settings**
 - Refresh Trigger: None
 - Refresh Interval: 14400
 - Refresh Time of Day: 12:00 AM
 - Refresh Type: Set
 - Initialize Mode: None
 - Set Preference: First
- Inter-Switch Connectivity Group Assignments**

Name	Type	Member
Incoming 1	Inter-Switch Incoming Uniform Numbering Plan	<input type="checkbox"/>
Incoming 2	Inter-Switch Incoming Uniform Numbering Plan	<input type="checkbox"/>
Outgoing 1	Inter-Switch Outgoing Uniform Numbering Plan	<input type="checkbox"/>
Outgoing 2	Inter-Switch Outgoing Uniform Numbering Plan	<input type="checkbox"/>
- Local Switch Settings**
 - View: All Settings
 - Set Defaults

Name	Value
Disconnect Loop Current Length (ms)	150
Flash Hook Time (ms)	500
T1 Protocol	FXS
T1 Signaling	Immediate

Buttons on the right: OK, Apply, Cancel, Help.

- 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 *System Installation and Configuration Guide*.

- 6 Click **OK**. The **Integration Options** dialog box appears.

Name	Value
SIP Server Address	
SIP Server Port	5060
SIP Domain Name	
Transport for outgoing SIP messages	TCP
Use DNS discovery procedures	<input type="checkbox"/>
Local IP Address to bind on	- Please Select -
SIP Local Connection Port	5060
SIP parser qualifier string	

- 7 In the **Integration Options** dialog box, configure the following options:
 - a In the **Local Integration Settings** section, select the **Required Parameters** view, and configure the following options:

Table 4. Required Parameters View – Integration Options

Field	Required Value
SIP Server Address	For the SIP trunk with SM integrations: Enter the IP address of the Session Manager server. For the Direct SIP trunk integrations. Enter the IP address of the CM procr.
SIP Server Port	Enter the port number on which the Session Manager or CM listens for SIP messages. For the SM, this port must match the Session Manager port configured as an Entity Link for MiCollab AM. The default port number is 5060.
SIP Domain Name	Enter the SIP domain name. This case-sensitive value must be the

	same as the Far-End Domain Name in the signaling group.
	<p>IMPORTANT Be sure the Authoritative Domain name is the same throughout the Session Manager server, Communication Manager, and MiCollab AM programming.</p>
	<p>NOTE This value is case-sensitive.</p>
Transport for outgoing SIP messages	<p>Enter TCP or UDP (TCP is the default value.)</p> <p>NOTE This value must match the protocol selected on the Entity Link created for MiCollab AM Call Server.</p>
Use DNS discovery procedures	Select this box to use DNS discovery.
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 5060.
Sip parser qualifier string	<p><i>In cases of a single SIP integration</i> on the call server, 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.</p> <p>In cases where there are multiple SIP integrations on the call server, use a string that is unique to each SIP integration.</p> <p>For example:</p> <p>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. The hunt number must be unique across all IP integrations.</p> <p>The Fully Qualified Domain Name (FQDN) of the switch, such as pbx1.sipdomain.com.</p> <p>NOTE This setting must match a string in the SIP header that is unique to this particular integration.</p>

- b** In the **Local Integration Settings** section, select the **Integration Specific Parameters** view and configure the following option:
- In the **Type of Call Progress to use for External Calls** field, select the type. How this should be set depends on the gateway used for the integration.

- **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.

IMPORTANT Enabling this parameter causes processing overhead and should only be enabled when necessary. For information on this setting, see the note in the [Critical Application Considerations](#) section.

- c In the **Local Integration Settings** section, select the **Media Settings** view and configure the following option:
 - In the **Validate Remote Hosts for Media** field, select the box if you want to use this feature.
- d Click **OK**. The **Switch Section Options** dialog box appears.
- 8 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**, select the mode for this integration.
 - c In the **Hunt Group Access Code** field, type the hunt pilot number you defined earlier under the [Creating a Hunt Group and Pilot Number](#) section.
 - d Click **OK**.
- 9 Continue through and complete the configuration. At the end of the configuration, a confirmation dialog box appears. Click **OK**.
- 10 If **MiCollab AM Configuration** does not open automatically after the configuration completes, open **MiCollab AM Configuration**, and select the **Lines** tab.
- 11 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.
- 12 Click **OK** to save all changes.

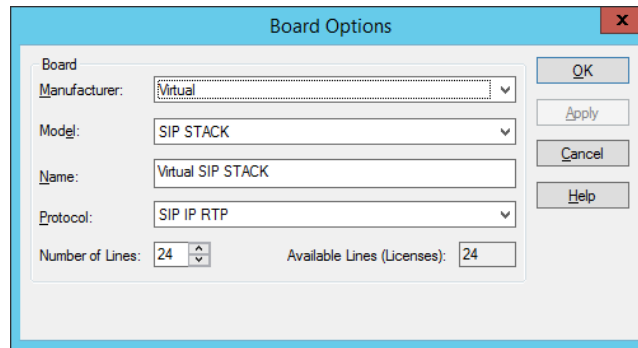
Configuring Existing MiCollab AM for the Integration

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**.

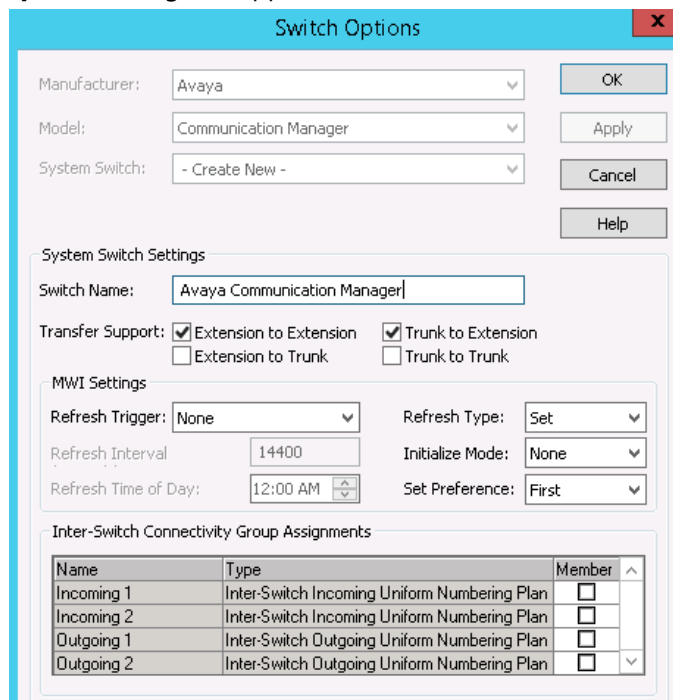
- 3 **[Optional]** Select the **Boards** tab, and then click the **Add** button. The **Board Options** dialog box appears.



The **Board Options** dialog box contains the following fields and controls:

- Manufacturer:** Virtual (dropdown)
- Model:** SIP STACK (dropdown)
- Name:** Virtual SIP STACK (text field)
- Protocol:** SIP IP RTP (dropdown)
- Number of Lines:** 24 (spin box)
- Available Lines (Licenses):** 24 (spin box)
- Buttons: OK, Apply, Cancel, Help

- a From the **Manufacturer** drop-down list, select **Virtual**.
 - b From the **Model** drop-down 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** drop-down 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 **Switches** tab and click the **Add** button. The **Switch Integration Data Setup** dialog box appears.
 - a From the **Manufacturer** drop-down list, select **Avaya**.
 - b From the **Model** drop-down list, select **Communication Manager**.
 - c From the **Integration Type** drop-down list, select **SIP**.
- 5 Click **OK**. The **Switch Options** dialog box appears.



The **Switch Options** dialog box contains the following sections and controls:

- Manufacturer:** Avaya (dropdown)
- Model:** Communication Manager (dropdown)
- System Switch:** - Create New - (dropdown)
- Buttons: OK, Apply, Cancel, Help
- System Switch Settings**
 - Switch Name:** Avaya Communication Manager (text field)
 - Transfer Support:**
 - ☒ Extension to Extension
 - ☒ Trunk to Extension
 - ☐ Extension to Trunk
 - ☐ Trunk to Trunk
 - MWI Settings**
 - Refresh Trigger:** None (dropdown)
 - Refresh Interval:** 14400 (text field)
 - Refresh Time of Day:** 12:00 AM (spin box)
 - Refresh Type:** Set (dropdown)
 - Initialize Mode:** None (dropdown)
 - Set Preference:** First (dropdown)
- Inter-Switch Connectivity Group Assignments**

Name	Type	Member
Incoming 1	Inter-Switch Incoming Uniform Numbering Plan	<input type="checkbox"/>
Incoming 2	Inter-Switch Incoming Uniform Numbering Plan	<input type="checkbox"/>
Outgoing 1	Inter-Switch Outgoing Uniform Numbering Plan	<input type="checkbox"/>
Outgoing 2	Inter-Switch Outgoing Uniform Numbering Plan	<input type="checkbox"/>

Image continued on next page

Image continued from previous page

Local Switch Settings

View: All Settings Set Defaults

Name	Value
Disconnect Loop Current Length (ms)	150
Flash Hook Time (ms)	500
T1 Protocol	FXS
T1 Signaling	Immediate

- 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 *System Installation and Configuration Guide*.

- 7 Click **OK**. The **Integration Options** dialog box appears.

Integration Options

System Switch: Avaya Communication Manager OK

Integration Type: SIP Trunk Apply

Integration: - Create New - Cancel

Name: Avaya Communication Manager SIP Trunk Help

ITN...

Local Integration Settings

View: Required Parameters Set Defaults

Name	Value
SIP Server Address	
SIP Server Port	5060
SIP Domain Name	
Transport for outgoing SIP messages	TCP
Use DNS discovery procedures	<input type="checkbox"/>
Local IP Address to bind on	- Please Select -
SIP Local Connection Port	5060
SIP parser qualifier string	

- 8 In the **Integration Options** dialog box, configure the following options:
- a In the **Local Integration Settings** section, select the **Required Parameters** view, and configure the following options:

Table 5. Required Parameters View – Integration Options

Field	Required Value
SIP Server Address	For the SIP trunk with SM integrations:

Enter the IP address of the Session Manager server.

For the Direct SIP trunk integrations:

Enter the IP address of the CM procr.

SIP Server Port	Enter the port number on which the Session Manager or CM listens for SIP messages. For the SM, this port must match the Session Manager port configured as an Entity Link for MiCollab AM. The default port number is 5060.
SIP Domain Name	<p>Enter the SIP domain name. This case-sensitive value must be the same as the Far-End Domain Name in the signaling group.</p> <p>IMPORTANT Be sure the Authoritative Domain name is the same throughout the Session Manager server, Communication Manager, and MiCollab AM programming.</p> <p>NOTE This value is case-sensitive.</p>
Transport for outgoing SIP messages	<p>Enter TCP or UDP (TCP is the default value.)</p> <p>NOTE This value must match the protocol selected on the Entity Link created for MiCollab AM Call Server.</p>
Use DNS discovery procedures	Select this box to use DNS discovery.
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 5060.
Sip parser qualifier string	<p><i>In cases of a single SIP integration</i> on the call server, 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.</p> <p>In cases where there are multiple SIP integrations on the call server, use a string that is unique to each SIP integration.</p> <p>For example:</p> <p>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. The hunt number must be unique across all IP integrations.</p> <p>The Fully Qualified Domain Name (FQDN) of the switch, such as pbx1.sipdomain.com.</p>

NOTE This setting must match a string in the SIP header that is unique to this particular integration.

- b** In the **Local Integration Settings** section, select the **Integration Specific Parameters** view and configure the following option:
- In the **Type of Call Progress to use for External Calls** field, select the type. How this should be set depends on the gateway used for the integration.
 - **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.

IMPORTANT Enabling this parameter causes processing overhead and should only be enabled when necessary. For information on this setting, see the note in the [Critical Application Considerations](#) section.

- c** In the **Local Integration Settings** section, select the **Media Settings** view and configure the following option:
- In the **Validate Remote Hosts for Media** field, select the box if you want to use this feature.
- d** Click **OK**. The **Switch Section Options** dialog box appears.

- 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**, select the mode for this integration.
 - c** In the **Hunt Group Access Code** field, type the hunt pilot number you defined earlier under the [Creating a Hunt Group and Pilot Number](#) section.
 - d** Click **OK**.
- 10** In **MiCollab AM Configuration**, verify that the telephone system is properly added and configured in the **Switches**, **Switch Sections**, and **Integrations** tabs.
- 11** 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 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** drop-down 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 6. 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>For example:</p> <p>The IP address for a Survivable Remote Server (formerly LSP) is 172.16.20.122 as displayed on the Review/Modify SIP Gateway screen.</p> <p>The IP address for a Survivable Core Server (formerly ESS) is 172.16.20.150 as displayed on the Review/Modify SIP Gateway screen.</p> <p>NOTE 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>IMPORTANT This value must match the configuration on the Gateway of the secondary node.</p>
Secondary SIP Server Port	<p>Enter the port number of the secondary node. The default value is 5060.</p>

- 7 From the **View** drop-down 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 are 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 7. 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 1000 ms.
Incoming off hook delay	800
Outgoing off hook delay	0
On hook delay	300
Type of Call Progress to use for External Calls	How this should be set depends on the gateway used for the integration. <ul style="list-style-type: none"> If the gateway supports call progress through to the endpoint, set to Digital.

- If the gateway reports early that the call is connected, such as before the phone rings or while the phone is ringing, set to **Media**.
-

- 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** drop-down 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

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. By default, MiCollab AM uses the primary (public) network interface card (NIC) in the platform, the first NIC in the network binding order. If you want MiCollab AM to use a NIC other than the first one, you must make several required configuration changes. It is much easier to configure the Integration to use another NIC by simply setting the integration parameter **Local IP Address to bind on** to the address of the NIC connected to the PBX.

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 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 Internet > 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 2016 / 2019

To change the binding order of multiple NICs:

- 1 From the taskbar, select **Start > Control Panel**.
- 2 In the **Control Panel**, click **Network and Internet > Network and Sharing Center**.
- 3 On the left pane, select **Change Adapter Settings**.
- 4 Right-click the network connection that serves MiCollab AM and then select **Properties**.
- 5 On the **Networking** tab of the **Local Area Connection Properties** dialog box, select **Internet Protocol Version 4 (TCP/IPv4)**, and then click **Properties**.
- 6 On the **General** tab of the **Internet Protocol Version 4 (TCP/IPv4) Properties** dialog box, click the **Advanced** button.
- 7 On the **IP Settings** tab of the **Advanced TCP/IP Settings** dialog box, clear the **Automatic metric** check box and then type in a low value in the **Interface metric** field. The lower the value, the higher the priority.

NOTE For all Windows systems, the value 1 is reserved for the loopback adapter. It is recommended to use a value of 2 or higher for the network connection that serves MiCollab AM.

- 8 Click **OK** on all of the dialog boxes to save the settings, and then close the **Local Area Connection Properties** dialog box.
- 9 Repeat steps 4 through 8 to assign an Interface metric value to all other network adapters.

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 8. 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