

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

For version 6.1 and above

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# Contents

<b>Preface</b>	<b>5</b>
References	6
Documentation	6
Documentation Updates	6
Help	6
Document Conventions	7
Features Supported by This Integration	8
<b>Critical Application Considerations</b>	<b>10</b>
<b>Installation Requirements</b>	<b>13</b>
Telephone System Requirements	13
MiCollab AM Requirements	13
<b>Programming the Telephone System</b>	<b>14</b>
Preparing the Telephone System for the Integration	14
Assigning Node IP Addresses in the Communication Manger	14
Creating a SIP Signaling Group	15
Defining the IP Interfaces	15
Creating a SIP Trunk Group	16
Programming MiCollab AM Ports	17
Configuring the SIP Entities on Avaya Servers	18
Configuring the Session Manager Firewall	19
Configuring the Routing Policies	21
Adding the MiCollab AM Port User Definitions	22
Creating a Hunt Group and Pilot Number	24
Creating a Coverage Path	25
Creating a Route Pattern	26
Modifying Digit Conversion Tables	27
Defining the Telephone System Location	28
Programming Subscriber Telephones	28
Verifying the Local Survivable Processor settings of Telephone System	34
<b>Configuring MiCollab AM</b>	<b>38</b>
Configuring MiCollab AM for the Integration During Initial Installation	38
Configuring Existing MiCollab AM for the Integration	42

Configuring MiCollab AM for SIP Failover	47
<b>Changing the Network Binding Order on the MiCollab AM Platform</b>	<b>49</b>
Windows Server 2008 R2 with Service Pack 1	49
Windows Server 2012 R2	50
<b>Configuring Quality of Service (QoS)</b>	<b>51</b>

# 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.

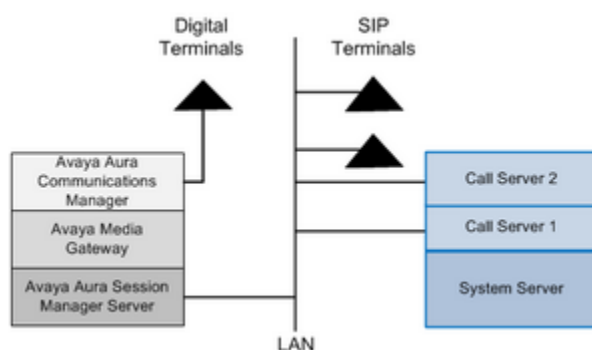


Figure 1. Terminals

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 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](http://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 Avaya Aura Communication Manager SIP 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	Yes

Table 2. 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.
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](#) 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 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 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. See Technical Bulletin #42285 for more information on this parameter.

**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 Local Survivable Processor support – for Secondary or tertiary failover server scenarios.
  - SIP endpoints supported when defined to have the secondary Server (LSP) in Network region. Digital and Analog endpoints only supported if the reside within the G430 or G450 to which the Communication Manager has as 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 refer to the following Avaya Documents:
  - Deploying Avaya Aura Communication Manager – Release 7.0.1 Issue 2.1
  - Converting Avaya Servers and Gateways (Document ID: 03-602884)
  - Avaya Aura® Communication Manager Survivability Options (Document ID: 03-603633)
    - 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* (Document ID: 555-233-504)
- 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 Free 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 refer to the following two Avaya Documents:
  - *Administering Network Connectivity on Avaya Aura Communication Manager* (Document ID: 555-233-504)
  - *Avaya Communication Manager Network Region Configuration Guide* (Document ID: 103244)
- 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

- Avaya Aura Communication Manager 7.0 and prior
- Avaya Aura Session Manager 7.0 and prior
- 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 or G450 with S8300E for failover server scenario. Survivable LSP (SIP endpoints supported in this configuration only)

## MiCollab AM Requirements

- MiCollab AM version 6.1
- MiCollab AM software key or feature file with the Avaya Communication Manager SIP integration enabled and one RADVISION 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:

The screenshot shows a command-line interface for the 'change node-names ip' command. At the top, there is a command prompt 'change node-names ip' followed by buttons for 'send (return)', 'help (f5)', 'cancel (esc)', 'enter (f3)', 'schedule (f9)', 'next (f7)', and 'previous (f8)'. Below the prompt, there are two tabs labeled '1' and '2'. The main area displays a table titled 'IP NODE NAMES' with two columns: 'Name' and 'IP Address'. The table contains four rows of data:

Name	IP Address
CXHACALL01	172.16.4.118
CXHACALL02	172.16.4.82
CXHASYSTEM	172.16.4.189
CXsipMeridian	172.16.4.65

## Creating a SIP Signaling Group

Using an Avaya Site Administration (ASA) 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.
- 3 Enable direct IP-to-IP Audio Connections and IP Audio Hairpinning.
- 4 Specify the rtp-payload method that the telephone system will use in transmitting DTMF tone sequences over the IP network.

The screenshot shows the ASA terminal interface for configuring a SIP Signaling Group. The title bar includes navigation buttons like 'change signaling-group 1', 'send (return)', 'help (f5)', 'cancel (esc)', 'enter (f3)', 'schedule (f9)', 'next (f7)', and 'previous (f8)'. The main window is titled '1 SIGNALING GROUP'. The configuration details are as follows:

- Group Number: 1
- Group Type: sip
- IMS Enabled? ☐ n
- Q-SIP? ☐ n
- IP Video? ☐ n
- Transport Method:
- Enforce SIPS URI for SRTP? ☒ y
- Peer Detection Enabled? ☒ y
- 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:
- DTMF over IP:
- Session Establishment Timer(min):
- Enable Layer 3 Test? ☒ y
- Bypass If IP Threshold Exceeded? ☐ n
- RFC 3389 Comfort Noise? ☐ n
- Direct IP-IP Audio Connections? ☐ n
- IP Audio Hairpinning? ☐ 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.

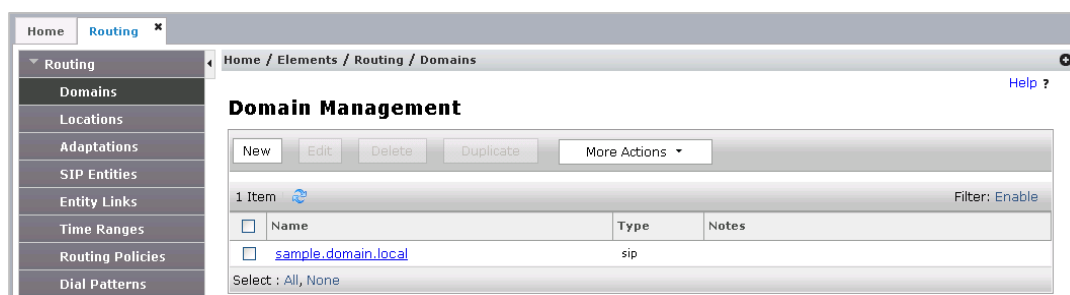


Figure 2. Avaya Aura System Manager

display ip-network-region 1 send (return) help (f5) cancel (esc) enter (f3) schedule (f9) next (f7) previous (f8)

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20
---	---	---	---	---	---	---	---	---	----	----	----	----	----	----	----	----	----	----	----

IP NETWORK REGION

```

Region: 1
Location: 1      Authoritative Domain: blvu.avstlabs.local
Name: NR 1      Stub Network Region: n
MEDIA PARAMETERS
  Codec Set: 1      Intra-region IP-IP Direct Audio: yes
                   Inter-region IP-IP Direct Audio: yes
                   IP Audio Hairpinning? y
  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
H.323 IP ENDPOINTS
  H.323 Link Bounce Recovery? y
  Idle Traffic Interval (sec): 20
  Keep-Alive Interval (sec): 5
  Keep-Alive Count: 5
AUDIO RESOURCE RESERVATION PARAMETERS
  RSVP Enabled? n
  
```

## 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*.

change trunk-group 1 send (return) help (f5) cancel (esc) enter (f3) schedule (f9) next (f7) previous (f8)

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21
---	---	---	---	---	---	---	---	---	----	----	----	----	----	----	----	----	----	----	----	----

TRUNK GROUP

```

Group Number: 1      Group Type: sip      CDR Reports: y
Group Name: sip trunk - t1s      COR: 1      TN: 1      TAC: 201
Direction: two-way      Outgoing Display? n      Night Service:
Dial Access? n
Queue Length: 0
Service Type: tie      Auth Code? n
Member Assignment Method: auto
Signaling Group: 1
Number of Members: 255
  
```

- 2 Associate the new Trunk Group with the Signaling Group you created previously.

change trunk-group 1 send (return) help (f5) cancel (esc) enter (f3) schedule (f9) next (f7) previous (f8)

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	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
  
```

*Image continues on next page*



Image continued from previous page

7: T00007	sip trunk
8: T00008	sip trunk
9: T00009	sip trunk
10: T00010	sip trunk
11: T00147	sip trunk
12: T00148	sip trunk
13: T00149	sip trunk
14: T00150	sip trunk
15: T00151	sip trunk

## Programming MiCollab AM Ports

Assign the station ports to the trunk group you have created (and have therefore included in the integration). Assign an Extension Number and a Name to each port and set the Station Type to **4602+**.

add station 5001   send (return)   help (f5)   cancel (esc)   enter (f3)   schedule (f9)   next (f7)   previous (f8)

1 | 2 | 3 | 4

**STATION**

Extension: 5001   Lock Messages? ☐   BCC: 0  
 Type: 4602+   Security Code:   TN: 1  
 Port: IP   Coverage Path 1:   COR: 1  
 Name: 5001   Coverage Path 2:   COS: 1  
 Hunt-to Station:   Tests? ☒

**STATION OPTIONS**

Loss Group: 19   Time of Day Lock Table: ☐  
 Personalized Ringing Pattern: 1  
 Message Lamp Ext: 6001  
 Speakerphone: 1-way   Mute Button Enabled? ☒  
 Display Language: english  
 Survivable GK Mode Name:   Media Complex Ext:   IP SoftPhone? ☐  
 Survivable COR: internal  
 Survivable Trunk Dest? ☒  
 IP Video? ☐  
 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 these ports with the trunk group you defined earlier, as the following two examples demonstrate.

(Page 1)

display off-pbx-telephone station   send (return)   help (f5)   cancel (esc)   enter (f3)   schedule (f9)   next (f7)   previous (f8)

1 | 2 | 3

**STATIONS WITH OFF-PBX TELEPHONE INTEGRATION**

Station Extension	Application	Dial Prefix	CC	Phone Number	Trunk Selection	Config Set	Dual Mode
5001	OPS	-		5001	1	1	
5002	OPS	-		5002	1	1	
5003	OPS	-		5003	1	1	
5004	OPS	-		5004	1	1	
5005	OPS	-		5005	1	1	
5006	OPS	-		5006	1	1	
5007	OPS	-		5007	1	1	

(Page 2)

display off-pbx-telephone station	send (return)	help (f5)	cancel (esc)	enter (f3)	schedule (f9)	next (f7)	previous (f8)
1	2	3					
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION							
Station Extension	Appl Name	Call Limit	Mapping Mode	Calls Allowed	Bridged Calls	Location	
5001	OPS	2	both	all	none		
5002	OPS	2	both	all	none		
5003	OPS	2	both	all	both		
5004	OPS	2	both	all	both		
5005	OPS	2	both	all	both		
5006	OPS	2	both	all	both		
5007	OPS	2	both	all	both		

## Configuring the SIP Entities on Avaya Servers

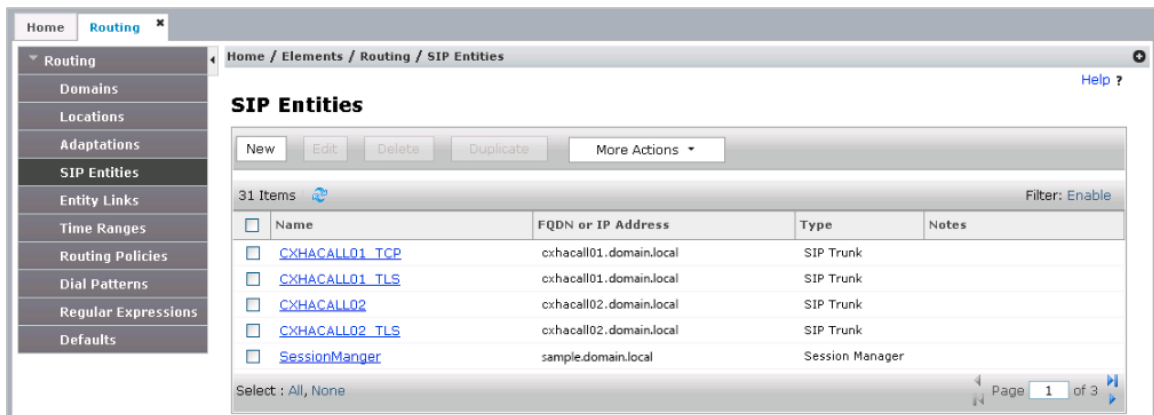
Verify the connection between the Communications Manager and Session Manager servers on the Session Manager server. Make sure the Communication Manager server Interface reflects the same IP addresses as the Media Server Interface.

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**.

**Listen Ports**

TCP Failover port:

TLS Failover port:

Add Remove

3 Items Filter: Enable

<input type="checkbox"/>	Listen Ports	Protocol	Default Domain	Notes
<input type="checkbox"/>	<input type="text" value="5060"/>	TCP	<input type="text"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="5060"/>	TCP	<input type="text"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="5061"/>	TCP	<input type="text"/>	<input type="text"/>

Select : All, None

**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.

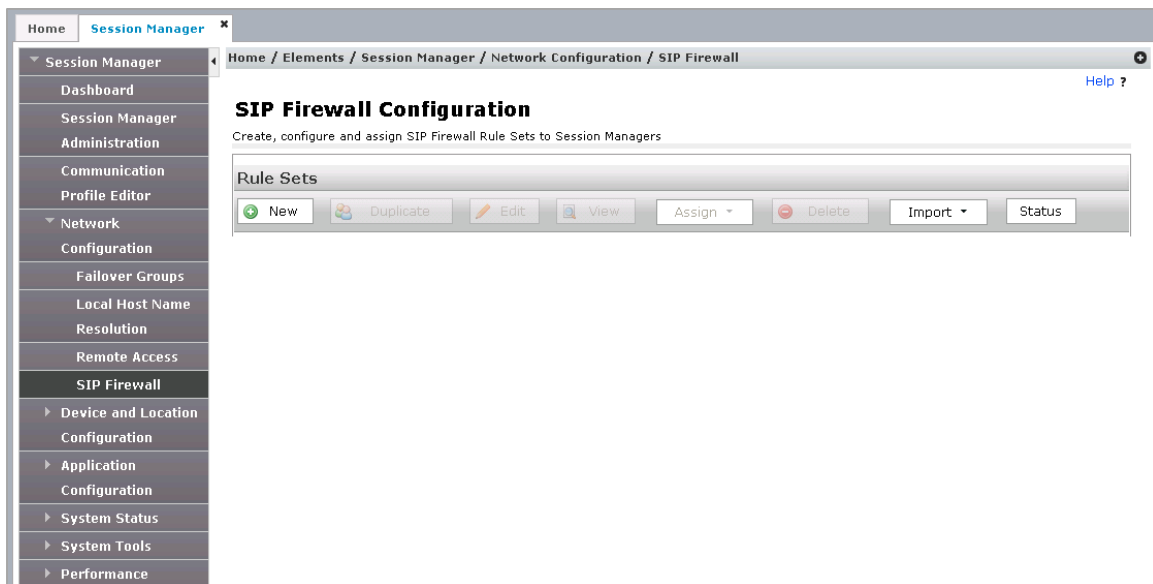
The default protocol is **TCP** and the default port number is **5060**. For more information, refer to the [Configuring MiCollab AM](#) section.

## 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.

Rule Set

CommitCancel

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

\*Name

Description

RulesBlacklistWhitelist

Enabled☐

NewEditViewDeleteUpDown

<input type="checkbox"/>	Enabled	Name	Action Type	Log Type	Log Message
--------------------------	---------	------	-------------	----------	-------------

- 3 In the **Name** field, select or type *Rule Set for SessionManager*.

- 4 In the **Rules** tab, click **New**. The **Rule** page displays.

Rule

CancelDone

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

General

Enabled:☒

\*Name:

Rule Default

\*Action Type:

Permit

Log Type:

None

Log Message:

IP Layer Match Options

Protocol:

Any

Remote IP Address:

Any

Remote Port:

Any

Local Port:

Any

SIP Layer Match Options

NewDelete

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

IP/SIP Layer Track

Track:

None

Threshold

Count (packets):

20

Period (secs):

20

Timeout (secs):

900

Connection

Connection Type:

Any

\*Required

CancelDone

- 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**.

- 7 Define an IP address to ensure that an 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.

*Image continues on next page*

Image continued from previous page

The screenshot shows two configuration sections: 'Dial Patterns' and 'Regular Expressions'. Each section has an 'Add' and 'Remove' button at the top. Below these are tables with columns for configuration. The 'Dial Patterns' table has columns: Pattern, Min, Max, Emergency Call, SIP Domain, Originating Location, and Notes. The 'Regular Expressions' table has columns: Pattern, Rank Order, Deny, and Notes. Both tables currently show '0 Items'. At the bottom right of the entire form are 'Commit' and 'Cancel' buttons.

- 3 In the **General** section, enter the name for the policy and number of retries.
- 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 like 'Manage Users', 'Public Contacts', 'Shared Addresses', 'System Presence', 'ACLs', 'Communication', 'Profile Password', and 'Policy'. The main content area has tabs for 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Identity' tab is active, showing a 'User Provisioning Rule' dropdown and a form with fields for 'Last Name', 'First Name', 'Middle Name', 'Description', and 'Login Name'. Each field has a red asterisk indicating it is required. At the top right of the form are 'Commit & Continue', 'Commit', and 'Cancel' buttons.

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

Home / Users / User Management / Manage Users

**New User Profile**

Identity \* **Communication Profile** Membership Contacts

Communication Profile

Communication Profile Password:

Confirm Password:

New Delete Done Cancel

Name

Primary

Select : None

\* Name: Primary

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

\* Required

Commit & Continue Commit Cancel

- 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

New Edit Delete

Type	Handle	Domain
No Records found		

Type: Avaya SIP

\* Fully Qualified Address:  @

Add Cancel

- 7 From the **Type** dropdown 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 ASA configuration screens.

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

The screenshot shows the 'Session Manager Profile' configuration window. It has a title bar with a close button and a help icon. The window is divided into several sections:

- SIP Registration**: Contains fields for 'Primary Session Manager', 'Secondary Session Manager', and 'Survivability Server', each with a search icon. Below these is a 'Max. Simultaneous Devices' dropdown set to '1' and a checkbox for 'Block New Registration When Maximum Registrations Active?' which is unchecked.
- Application Sequences**: Contains two dropdown menus for 'Origination Sequence' and 'Termination Sequence', both set to '(None)'.
- Call Routing Settings**: Contains a dropdown for 'Home Location' set to 'Select' and a dropdown for 'Conference Factory Set' set to '(None)'.
- Call History Settings**: Contains a checkbox for 'Enable Centralized Call History?' which is unchecked.

- 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 2																									send (return)	help (f5)	cancel (esc)	enter (f3)	schedule (f9)	next (f7)	previous (f8)
1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25							
HUNT GROUP																															
Group Number: 2															ACD? <input type="text" value="n"/>																
Group Name: SIP UM															Queue? <input type="text" value="n"/>																
Group Extension: 5000															Vector? <input type="text" value="n"/>																
Group Type: ucd-mia															Coverage Path: <input type="text" value=""/>																
TN: 1															Night Service Destination: <input type="text" value=""/>																
COR: 1															MM Early Answer? <input type="text" value="n"/>																
Security Code: <input type="text" value=""/>															Local Agent Preference? <input type="text" value="n"/>																
ISDN/SIP Caller Display: mbr-name																															

(Page 2)

add hunt-group 2																									send (return)	help (f5)	cancel (esc)	enter (f3)	schedule (f9)	next (f7)	previous (f8)
1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25							
HUNT GROUP																															
LWC Reception: none															AUDIX Name: <input type="text" value=""/>																
Message Center: none																															

(Page 3)

add hunt-group 2																									send (return)	help (f5)	cancel (esc)	enter (f3)	schedule (f9)	next (f7)	previous (f8)
1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25							
HUNT GROUP																															
Group Number: 2										Group Extension: 5000										Group Type: ucd-mia											
Member Range Allowed: 1 - 1500										Administered Members (min/max): 2 /4																					
Total Administered Members: 3																															
GROUP MEMBER ASSIGNMENTS																															
Ext												Name(19 characters)																			
1: <input type="text" value=""/>												14: <input type="text" value=""/>																			
2: 5002												15: <input type="text" value=""/>																			
3: 5003												16: <input type="text" value=""/>																			
4: 5004												17: <input type="text" value=""/>																			
5: <input type="text" value=""/>												18: <input type="text" value=""/>																			
6: <input type="text" value=""/>												19: <input type="text" value=""/>																			
7: <input type="text" value=""/>												20: <input type="text" value=""/>																			
8: <input type="text" value=""/>												21: <input type="text" value=""/>																			
9: <input type="text" value=""/>												22: <input type="text" value=""/>																			
10: <input type="text" value=""/>												23: <input type="text" value=""/>																			
11: <input type="text" value=""/>												24: <input type="text" value=""/>																			
12: <input type="text" value=""/>												25: <input type="text" value=""/>																			
13: <input type="text" value=""/>												26: <input type="text" value=""/>																			
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 (2), 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 2
send (return)
help (f5)
cancel (esc)
enter (f3)
schedule (f9)
next (f7)
previous (f8)

1

COVERAGE PATH

Coverage Path Number: 2  
Cvg Enabled for VDN Route-To Party? ☐ n  
Next Path Number:   
Hunt after Coverage? ☐ n  
Linkage

COVERAGE CRITERIA

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

COVERAGE POINTS

Terminate to Coverage Pts. with Bridged Appearances? ☐ n  
Point1:  h2 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.

add route-pattern 1    send (return)    help (f5)    cancel (esc)    enter (f3)    schedule (f9)    next (f7)    previous (f8)

1    2    3

Pattern Number: 1    Pattern Name: SIP

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

Grp No	FRL	NPA	Pfx	Hop	Toll	No. Del	Inserted Dgts	DCS/ QSIG	IXC 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	n	n	rest		lev0-pvt	none
2:	y	y	y	y	n	n	rest			none
3:	y	y	y	y	n	n	rest			none
4:	y	y	y	y	n	n	rest			none
5:	y	y	y	y	n	n	rest			none
6:	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.

change aar analysis 1    send (return)    help (f5)    cancel (esc)    enter (f3)    schedule (f9)    next (f7)    previous (f8)

1    2

**AAR DIGIT ANALYSIS TABLE**  
Location: all    Percent Full: 1

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Req'd
2	7	7	999	aar		n
3	7	7	999	aar		n
4	7	7	999	aar		n
4200	4	4	10	lev0		n
4300	4	4	10	lev0		n
4400	4	4	9	aar		n
4500	4	4	45	aar		n
4600	4	4	46	aar		n
4800	4	4	48	aar		n
490	4	4	99	aar		n
5	7	7	999	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

[illegible]

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.

## Programming Subscriber Telephones

**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 ASA 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 1** 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)

display station 4010	send (return)	help (f5)	cancel (esc)	enter (f3)	schedule (f9)	next (f7)	previous (f8)
1	2	3	4	5	6		
<b>STATION</b>							
Extension: 4010		Lock Messages? n		BCC: 0			
Type: 9620SIP		Security Code: *		TN: 1			
Port: S00101		Coverage Path 1: 58		COR: 1			
Name: 4010SIP, stn4010		Coverage Path 2:		COS: 1			
		Hunt-to Station:					
<b>STATION OPTIONS</b>							
Loss Group: 19				Time of Day Lock Table:			
				Message Lamp Ext: 4010			
Display Language: english							
Survivable COR: internal				IP SoftPhone? n			
Survivable Trunk Dest? y				IP Video? n			

(Page 2)

display station 4010	send (return)	help (f5)	cancel (esc)	enter (f3)	schedule (f9)	next (f7)	previous (f8)
1	2	3	4	5	6		
<b>STATION</b>							
<b>FEATURE OPTIONS</b>							
LWC Reception: spe		Coverage Msg Retrieval? y					
LWC Activation? y		Auto Answer: none					
CDR Privacy? n		Data Restriction? n					
Per Button Ring Control? n		Idle Appearance Preference? n					
Bridged Call Alerting? n		Bridged Idle Line Preference? n					
Active Station Ringing: single		Restrict Last Appearance? n					
H.320 Conversion? n		Per Station CPN - Send Calling Number?					
		EC500 State: enabled					
MWI Served User Type:		Coverage After Forwarding? s					
AUDIX Name:		Direct IP-IP Audio Connections? y					
Emergency Location Ext: 4010		Always Use? n IP Audio Hairpinning? n					

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

(Page 6)

display station 4010   send (return)   help (f5)   cancel (esc)   enter (f3)   schedule (f9)   next (f7)   previous (f8)

1 | 2 | 3 | 4 | 5 | 6

**STATION**

**SIP FEATURE OPTIONS**

Type of 3PCC Enabled: None  
SIP Trunk: 1

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

display off-pbx-telephone station   send (return)   help (f5)   cancel (esc)   enter (f3)   schedule (f9)   next (f7)   previous (f8)

1 | 2 | 3

**STATIONS WITH OFF-PBX TELEPHONE INTEGRATION**

Station Extension	Application	Dial Prefix	CC	Phone Number	Trunk Selection	Config Set	Dual Mode
4001	OPS	-		4001	1	1	
4010	OPS	-		4010	1	1	
4011	OPS	-		4011	aar	1	
4012	OPS	-		4012	aar	1	
4013	OPS	-		4013	1	1	
4014	OPS	-		4014	1	1	
4015	OPS	-		4015	1	1	
4017	OPS	-		4017	1	1	
4019	OPS	-		4019	1	1	
5001	OPS	-		5001	1	1	
5002	OPS	-		5002	1	1	
5003	OPS	-		5003	1	1	
5004	OPS	-		5004	1	1	
5005	OPS	-		5005	1	1	
5006	OPS	-		5006	1	1	
5007	OPS	-		5007	1	1	

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

display public-unknown-number   send (return)   help (f5)   cancel (esc)   enter (f3)   schedule (f9)   next (f7)   previous (f8)

1 | 2

**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

Total Administered: 7  
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 User Management x

Home / Users / User Management / Manage Users

**New User Profile** [Help ?](#)

**Identity** \* **Communication Profile** **Membership** **Contacts**

**User Provisioning Rule**

**Identity** \*

\* Last Name:   
 Last Name (Latin Translation):   
 \* First Name:   
 First Name (Latin Translation):   
 Middle Name:   
 Description:   
 \* Login Name:   
 Authentication Type: Basic   
 Password:   
 Confirm Password:   
 Localized Display Name:   
 Endpoint Display Name:   
 Title:   
 Language Preference:   
 Time Zone:   
 Employee ID:   
 Department:   
 Company:

**Address**   
**Localized Names**

\*Required

- 8 In the **Identity** tab, add in the name and password as required.
- 9 Click the **Communication Profile** tab.

Home / Users / User Management / Manage Users

### New User Profile

Commit & Continue Commit Cancel

Identity \* Communication Profile Membership Contacts

**Communication Profile**

Communication Profile Password:

Confirm Password:

New Delete Done Cancel

Name

Primary

Select : None

\* Name: Primary

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 ▶

\* Required

Commit & Continue Commit Cancel

- 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 display.

**Communication Address**

New Edit Delete

Type	Handle	Domain
No Records found		

Type: Avaya SIP

\* Fully Qualified Address:  @

Add Cancel

- 12 From the **Type** dropdown 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

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? ☐

- 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

Use Existing Endpoints ☐

\* Extension  [Endpoint Editor](#)

\* Template

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**, **Template**, **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 ASA 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	AVAYACHSRU 172.16.20.122 No 06 Entry	LSP	y	n	13:48 9/14/2016	2

- 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 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] )

IP ADDRESS MAPPING				
IP Address	Subnet Bits	Network Region	U LAN	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:				
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)

The screenshot shows a CLI window with a command prompt at the top: `display ip-network:region 2`. Below the prompt is a series of numbered tabs from 1 to 20. The main display area shows the configuration for IP Network Region 2. The configuration is organized into several sections: Region, Location, MEDIA PARAMETERS, DIFFSERV/TOS PARAMETERS, 802.1P/Q PARAMETERS, H.323 IP ENDPOINTS, and AUDIO RESOURCE RESERVATION PARAMETERS. The settings include Region: 2, Location: 1, Name: LSP, Authoritative Domain: blvu.avstlabs.local, Stub Network Region: n, Intra-region IP-IP Direct Audio: yes, Inter-region IP-IP Direct Audio: yes, IP Audio Hairpinning? y, UDP Port Min: 2048, UDP Port Max: 3029, Call Control PHB Value: 46, Audio PHB Value: 46, Video PHB Value: 26, Call Control 802.1p Priority: 6, Audio 802.1p Priority: 6, Video 802.1p Priority: 5, H.323 Link Bounce Recovery? y, Idle Traffic Interval (sec): 20, Keep-Alive Interval (sec): 5, Keep-Alive Count: 5, and RSVP Enabled? n.

```

display ip-network:region 2
1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 20
IP NETWORK REGION

Region: 2
Location: 1      Authoritative Domain: blvu.avstlabs.local
Name: LSP        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: 3029
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
H.323 IP ENDPOINTS
H.323 Link Bounce Recovery? y
Idle Traffic Interval (sec): 20
Keep-Alive Interval (sec): 5
Keep-Alive Count: 5
AUDIO RESOURCE RESERVATION PARAMETERS
RSVP Enabled? n
  
```

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

- 4 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.

The screenshot shows the Avaya Aura System Manager 7.0 interface. The left sidebar contains a navigation menu with options: Home, Session Manager, Dashboard, Session Manager Administration, Communication Profile Editor, Network Configuration, Device and Location Configuration, Application Configuration, System Status, System Tools, and Performance. The main content area is titled 'Session Manager Dashboard' and includes a 'Session Manager Instances' table. The table has columns for Service State, Shutdown System, As of, Type, Tests Pass, Alarms, Security Module, Service State, Entity Monitoring, Active Call Count, Registrations, Data Replication, User Data Storage Status, License Mode, and Version. The table lists two instances: Session Manager (Core) and AYASMSIGSRV (BSM). The Session Manager instance is in a 'Up' state with 'Accept New Service' and '21/20' entities. The AYASMSIGSRV instance is in a 'Down' state with 'Accept New Service' and '0/0' entities.

Service State	Shutdown System	As of	Type	Tests Pass	Alarms	Security Module	Service State	Entity Monitoring	Active Call Count	Registrations	Data Replication	User Data Storage Status	License Mode	Version
Up		As of 1:20 PM	Core	0/0/0	0/0/0	Up	Accept New Service	21/20	0	5/9	✓	✓	Normal	7.0.0.0.700007
Down			BSM	0/0/0	0/0/0	Down	Accept New Service	---	0	0/0	⚠	---	Normal	7.0.0.0.700007

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

Select the **Session Manager Administration** tab to display the Instances and their individual settings.

### Session Manager Administration

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

#### Global Settings

Save

Allow Unauthenticated Emergency Calls ☐

Allow Unsecured PPM Traffic ☒

Fallback Policy 

Auto

ELIN SIP Entity 

None

Better Matching Dial Pattern or Range in Location ALL Overrides Match in Originator's Location ☐

Ignore SDP for Call Admission Control ☐

Disable Call Admission Control Threshold Alarms ☐

Disable Loop Detection Alarms ☐

\*Loop Detection Alarms Threshold (hours) 

24

Enable TLS Endpoint Certificate Validation ☐

Enable Dial Plan Ranges ☐

Enable Implicit Users Applications for SIP users ☐

Enable End to End Secure Call Indication ☐

#### Session Manager Instances

New

View

Edit

Delete

1 Item 

Filter: Enable

Name	License Mode	Primary Communication Profiles	Secondary Communication Profiles	Maximum Active Communication Profiles	Description
<div>SessionManger</div>	Normal	57	0	57	AVST Lab

Select : None

#### Branch Session Manager Instances

New

View

Edit

Delete

1 Item 

Filter: Enable

Name	License Mode	Main CM for LSP	SIP Communication Profiles	Description
<div>AVAYASMSGSRV</div>	Normal	CM_Lab	0	

Select : None

**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:

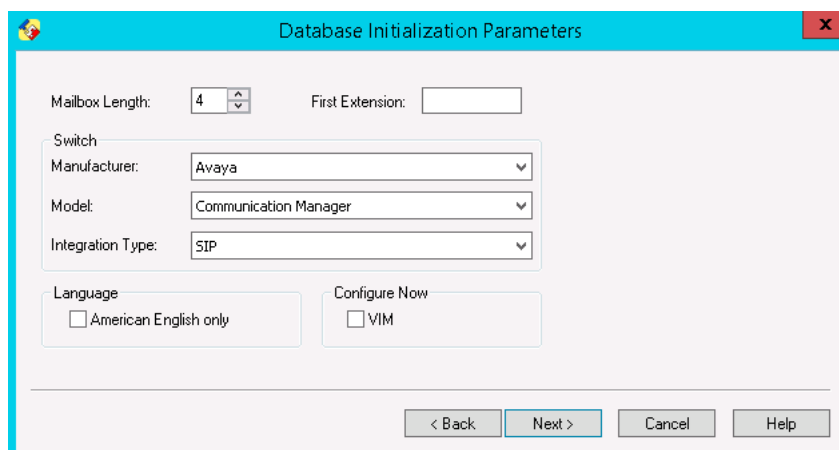
- [Configuring MiCollab AM for the Integration During Initial Installation](#): Integrate the telephone system while you install MiCollab AM for the first time.
- [Configuring Existing MiCollab AM for the Integration](#): Integrate a new telephone system on your exiting MiCollab AM system.

**NOTE** For general information on integrations, refer to the **Integrating MiCollab AM with the Telephone System** chapter in *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

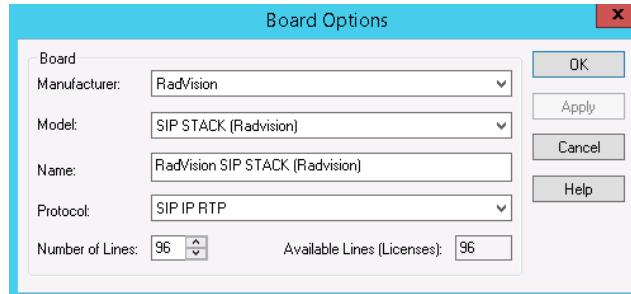
To configure MiCollab AM with the integration for the first time:

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

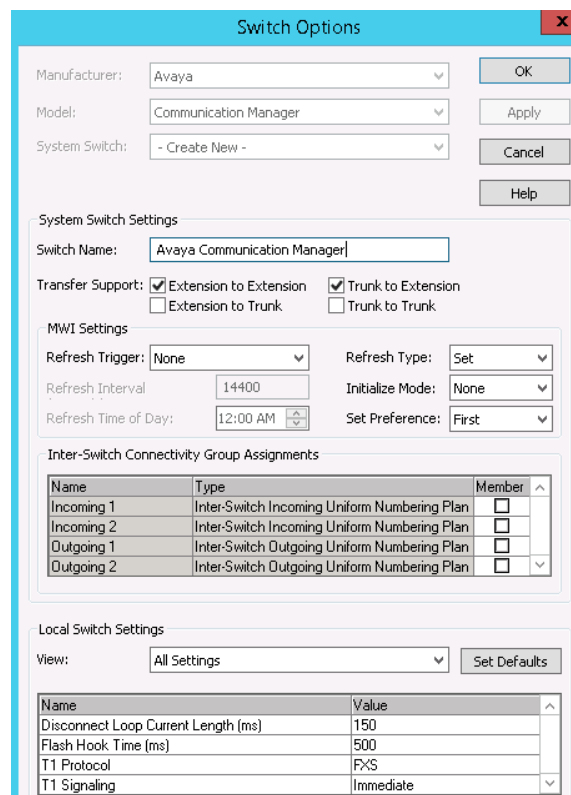


The **Board Options** dialog box is shown with the following configuration:

- Board Manufacturer:** RadVision
- Model:** SIP STACK (Radvision)
- Name:** RadVision SIP STACK (Radvision)
- Protocol:** SIP IP RTP
- Number of Lines:** 96
- Available Lines (Licenses):** 96

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

- 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.



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

- Manufacturer:** Avaya
- Model:** Communication Manager
- System Switch:** - Create New -
- System Switch Settings:**
  - 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 Type:** Set
    - Refresh Interval:** 14400
    - Initialize Mode:** None
    - Refresh Time of Day:** 12:00 AM
    - 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** button
  - Table:**

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 guide, *System Installation Guide*.

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

Integration Options

System Switch: Avaya Communication Manager

Integration Type: SIP Trunk

Integration: - Create New -

Name: Avaya Communication Manager SIP Trunk

Local Integration Settings

View: Required Parameters

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 enter the information as shown in the following table:

Table 3. Require Parameter Values – Integration Options

Field	Required Value
SIP Server Address	<b>For the SIP trunk with SM integrations:</b> Enter the IP address of the Session Manager server. <b>For the Direct SIP trunk integrations.</b> 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 <b>must</b> 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><b>IMPORTANT</b> Be sure the Authoritative Domain name is the same throughout the Session Manager server, Communication Manager, and MiCollab AM programming.</p>
	<p><b>NOTE</b> This value is case-sensitive.</p>
Transport for outgoing SIP messages	<p>Enter TCP or UDP (TCP is the default value.)</p> <p><b>NOTE</b> 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><b>For example:</b></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><b>NOTE</b> 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 displays.
- 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 displays. 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.

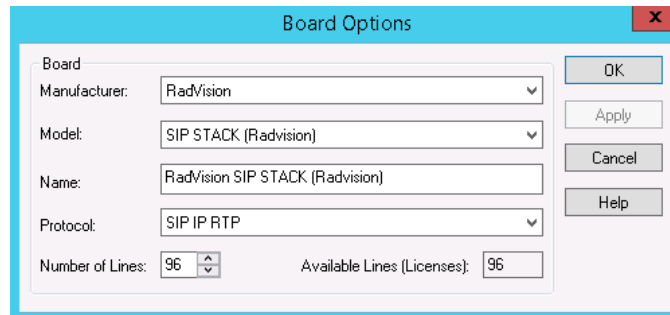
## Configuring Existing MiCollab AM for the Integration

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**.

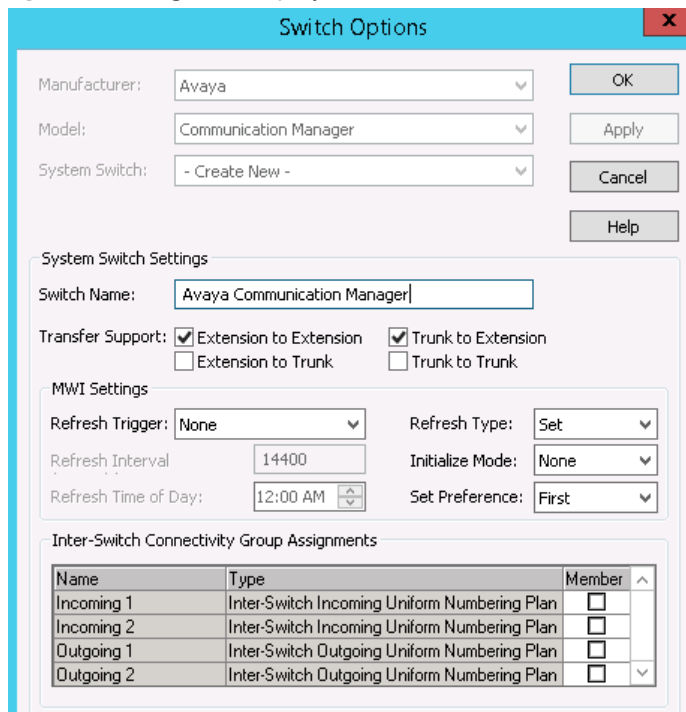
- 3 **[Optional]** Select the **Board** tab, and then click the **Add** button. The **Board** dialog box displays.



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

- Manufacturer:** RadVision (dropdown)
- Model:** SIP STACK (Radvision) (dropdown)
- Name:** RadVision SIP STACK (Radvision) (text field)
- Protocol:** SIP IP RTP (dropdown)
- Number of Lines:** 96 (spin box)
- Available Lines (Licenses):** 96 (spin box)
- Buttons:** OK, Apply, Cancel, Help

- 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 Manager**.
  - c From the **Integration Type** dropdown list, select **SIP**.
- 5 Click **OK**. The **Switch Options** dialog box displays.



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 (spin box)
  - 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 guide, *System Installation Guide*.

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

Integration Options

System Switch: Avaya Communication Manager OK

Integration Type: SIP Trunk Apply

Integration: - Create New - Cancel

Name: Avaya Communication Manager SIP Trunk Help

Read Me...

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 enter the information as shown in the following table:

Table 4. Require Parameter Values – Integration Options

Field	Required Value
SIP Server Address	<b>For the SIP trunk with SM integrations:</b>

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 <b>must</b> 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><b>IMPORTANT</b> Be sure the Authoritative Domain name is the same throughout the Session Manager server, Communication Manager, and MiCollab AM programming.</p> <p><b>NOTE</b> This value is case-sensitive.</p>
Transport for outgoing SIP messages	<p>Enter TCP or UDP (TCP is the default value.)</p> <p><b>NOTE</b> 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><b>For example:</b></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 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**, 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** 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 5. Secondary SIP Server Address and the Secondary SIP Server Port example

Field	Value
Secondary SIP Server Address	<div>Enter the TCP/IP address or an FQDN of the secondary node.</div> <div><b>For example:</b> The IP address 172.16.20.122 as displayed on the Review/Modify SIP Gateway screen.</div> <div><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.</div> <div><b>IMPORTANT</b> This value must match the configuration on the Gateway of the secondary node.</div>
Secondary SIP Server Port	<div>Enter the port number of the secondary node. The default value is <b>5060</b>.</div>

- 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 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 6. 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</b> ms.
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

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 card 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 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 7. 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