

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

For version 9.1 and above

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Preface

This Integration Technical Note (ITN) is written for dealers who are experienced with MiCollab Advanced Messaging (MiCollab AM) and are familiar with its procedures and terminology. This document also assumes that you are familiar with the features and programming of the Avaya Communication Manager 8500/8700 Series telephone system.

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 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 8500/8700 Series 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 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.

- 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.
- 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 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.
- On Avaya software versions prior to version 4.0 the integration provides MWI support on SIP-based telephones only. MWI operations on circuit-switched digital telephones require the addition of a secondary, D/82-based integration for MWI purposes. For information on setting up this second integration, refer to [Appendix A—Adding MWI Support for Non-IP Telephones](#).
- 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, 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.

Overview

This document describes how to integrate MiCollab AM with an Avaya 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 four major components (your environment may include a different mix of hardware):

- The Avaya Communication Manager server
- The Avaya Media Gateway server
- The Avaya SIP Enablement Services (SES) proxy server
- MiCollab AM

The Media Gateway coordinates all external connections between the Media Server and the outside world, including station ports and trunk ports as well as SIP communication over the LAN. MiCollab AM uses a station-side integration to connect to the gateway, with all MiCollab AM ports configured as Off-Premises Stations (OPS) and registered individually with the gateway.

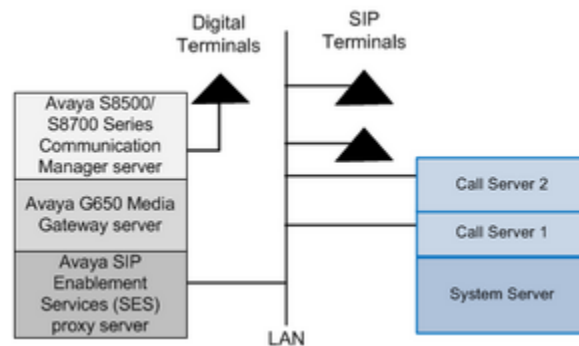


Figure 1. Terminals

The telephone system organizes all lines between itself and MiCollab AM into a hunt group. It then directs all calls intended for MiCollab AM, whether direct or forwarded, to the pilot number of this hunt group. The MiCollab AM 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 SES 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.

Use this document in conjunction with the *System Installation and Configuration Guide*, the *System Administration Guide*, and with the MiCollab AM online help system.

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 Communication Manager 8.1 or prior
- Avaya SIP Enablement Services 5.2 947 SP2a or prior
- TN2302/TN2602 IP Media Processor with current firmware update, to handle voice processing tasks
- TN799D C-LAN to process signaling information, if using a G650 legacy hardware integration
- Media Gateway G430 or G450, if not using a G650 legacy hardware integration
- SES Home Server and SES Edge Server, or
 - Combined SES Home/Edge Server
- One Off-Premises Station (OPS) license per MiCollab AM port
- One Administered SIP Trunk license per MiCollab AM port
- One Home license per MiCollab AM system in SES
- One Edge license per MiCollab AM system in SES
- One Home license per SIP user in SES for MWI

MiCollab AM Requirements

- MiCollab AM version 9.1
- MiCollab AM software key diskette or feature file with the Avaya S8500/S8700 Series SIP integration enabled and one Virtual SIP and RTP license enabled for each port involved in the integration
- One 100 Mbps or 1000 Mbps (1 Gbps) 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.

- Verifying the customer option for SIP trunking
- Assigning IP node names and addresses to the components of the media server and the SES server platform
- Defining IP interfaces
- Mapping the Media Server address
- Administering IP network regions

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

Creating a SIP Signaling Group

Using an Avaya Site Administration (ASA) terminal, define a signaling group associating the MiCollab AM and SES platforms, as shown in the following example.

To create SIP Signaling Group:

- Specify a node name for each of these platforms and a listening port accessible to both.
- Specify the name of the domain where the MiCollab AM platform is located.
- Enable direct IP-to-IP audio connections and IP audio hairpinning.
- Specify a method for the telephone system to use in transmitting DTMF tone sequences over the IP network.

The following is an example of creating a SIP Signaling Group

```
display signaling-group 1
      SIGNALING GROUP

Group Number: 1      Group Type: sip      Transport Method: tls

Near-end Node Name: clan1      Far-end Node Name: avayasip
Near-end Listen Port: 5061      Far-end Listen Port: 5061
Far-end Network Region: 1
Far-end Domain: blvu.AVSTlabs.local

Bypass If IP Threshold Exceeded? n

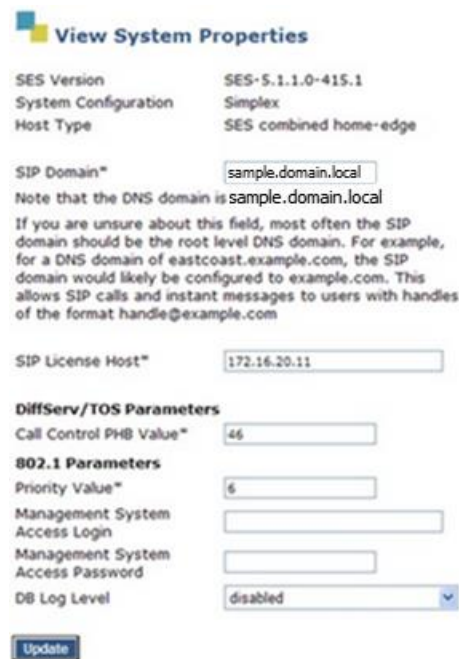
DTMF over IP: rtp-payload      Direct IP-IP Audio Connections? y
                                IP Audio Hairpinning? y

Session Establishment Timer(min): 120
```

Configuring the SIP Domain

Verify the SIP domain to the SES server is assigned correctly. If this is a new installation, you must assign a SIP domain name.

IMPORTANT The authoritative domain name must be the same throughout the SES, Communication Manager, and MiCollab AM programming.



The screenshot shows a 'View System Properties' window with the following fields and values:

Field	Value
SES Version	SES-5.1.1.0-415.1
System Configuration	Simplex
Host Type	SES combined home-edge
SIP Domain*	sample.domain.local
Note that the DNS domain is sample.domain.local	
If you are unsure about this field, most often the SIP domain should be the root level DNS domain. For example, for a DNS domain of eastcoast.example.com, the SIP domain would likely be configured to example.com. This allows SIP calls and instant messages to users with handles of the format handle@example.com	
SIP License Host*	172.16.20.11
DiffServ/TOS Parameters	
Call Control PHB Value*	46
802.1 Parameters	
Priority Value*	6
Management System Access Login	
Management System Access Password	
DB Log Level	disabled

An 'Update' button is located at the bottom left of the window.

Figure 2: SIP Domain


```

display ip-network-region 1
                                IP NETWORK REGION

  Region: 1
Location: 1      Authoritative Domain: sample.domain.local (example only)
  Name: region 1
MEDIA PARAMETERS                                Intra-region IP-IP Direct Audio: yes
  Codec Set: 1                                Intra-region IP-IP Direct Audio: yes
UDP Port Min: 2048                                IP Audio Hairpinning? y
UDP Port Max: 65531
DIFFSERV/TOS PARAMETERS                        RTCP Reporting Enabled? y
Call Control PHB Value: 46
                                RTCP MONITOR SERVER PARAMETERS
  Audio PHB Value: 46                        Use Default Server Parameters? y
  Video PHB Value: 26
802.1P/Q PARAMETERS
Call Control 802.1p Priority: 7
  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): 6
  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:

- Specify **SIP** as the group type and **tie** as the service type.
- Associate the new trunk group with the signaling group you created in the previous section.

```

display trunk-group 1                                Page 1 of 21
                                TRUNK GROUP

Group Number: 1      Group Type: sip      CDR Reports: y
Group Name: sip trunks  COR: 1      TN: 1      TAC: 201
Direction: two-way      Outgoing Display? n
Dial Access? n      Night Service:
Queue Length: 0
Service Type: tie      Authorization Code? n

                                Signaling Group: 1
                                Number of Members: 50

```

```

display trunk-group 1                               Page 5 of 21
      TRUNK GROUP
      Administered Members (min/max): 1/50

GROUP MEMBER ASSIGNMENTS      Total Administered Members: 50

      Port      Name
1: T00001      sip trunks
2: T00002      sip trunks
3: T00003      sip trunks
4: T00004      sip trunks
5: T00005      sip trunks
6: T00006      sip trunks
7: T00007      sip trunks
8: T00008      sip trunks
9: T00009      sip trunks
10: T00010     sip trunks
11: T00064     sip trunks
12: T00065     sip trunks
13: T00066     sip trunks
14: T00067     sip trunks
15: T00068     sip trunks

```

Programming MiCollab AM Ports

Assign the station ports to the trunk group you have created for the integration. Assign an extension number and a name to each port, and then set the station type to **4602+**.

```

add station 5001                                     Page 1 of 3
      STATION

Extension: 5001      Lock Messages? n      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:

STATION OPTIONS
      Loss Group: 19      Personalized Ringing Pattern: 1
                          Message Lamp Ext: 5001
      Speakerphone: 1-way      Mute Button Enabled? y
      Display Language: english
      Survivable GK Node Name:
      Survivable COR: internal      Media Complex Ext:
      Survivable Trunk Dest? y      IP SoftPhone? N

```

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.

Example One, Page 1 of 2

```
display off-pbx-telephone station-mapping      Page 1 of 2
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION
```

Station Extension	Application Prefix	Dial Prefix	Phone Number Selection	Trunk Set	Configuration
4000	OPS	- 4000	1	1	
4001	OPS	- 4001	1	1	
4002	OPS	- 4002	1	1	
4010	OPS	- 4010	1	1	
4011	OPS	- 4011	1	1	
4012	OPS	- 4012	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	
5008	OPS	- 5008	1	1	
5009	OPS	- 5009	1	1	
5010	OPS	- 5010	1	1	

Example Two, Page 2 of 2

```
display off-pbx-telephone station-mapping      Page 2 of 2
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION
```

Station Extension	Call Limit	Mapping Mode	Calls Allowed	Bridged Calls
4000	3	both	all	both
4001	3	both	all	both
4002	3	both	all	both
4010	3	both	all	both
4011	2	both	all	both
4012	4	both	all	both
5001	3	both	all	both
5002	2	both	all	both
5003	2	both	all	both
5004	2	both	all	both
5005	2	both	all	both
5006	2	both	all	both
5007	2	both	all	both
5008	2	both	all	both
5009	2	both	all	both
5010	2	both	all	both

Defining MiCollab AM Ports on the Media Gateway

Use the **Administration Web Interface** of the Avaya Integrated Management Console to modify the existing media server configuration, if needed. Later in this section, create an address map that associates the MiCollab AM ports with this media server.

Make sure the Communication Manager Server Interface reflects the same IP addresses as the Media Server Interface and SIP Trunk IP Address.

NOTE Write down the name and host IP address of the media server interface. You need to refer to them later.

Edit Communication Manager Server Interface

Communication Manager Server Interface Name* 172.16.20.13

Host 172.16.20.11

SIP Trunk

SIP Trunk Link Type ☐ TCP ☒ TLS

SIP Trunk IP Address* 172.16.20.13

Communication Manager Server

Communication Manager Server Admin Address* (see Help) 172.16.20.10

Communication Manager Server Admin Port* 5022

Communication Manager Server Admin Login* siplogin

Communication Manager Server Admin Password* *****

Communication Manager Server Admin Password Confirm* *****

SMS Connection Type ☒ SSH ☐ Telnet ☐ Not Available

Note: If the Communication Manager Server changed, changing connection type to SSH and changing connection type to Telnet will

Fields marked * are required.

Update

Figure 3: Communication Manager Server Interface

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 will not function correctly.

Assigning User Definitions to MiCollab AM Ports

- Add one user definition for each MiCollab AM port. Verify that the Host field is set to the host IP address assigned to the media server interface. Set the Primary Handle, User ID, First Name, and Last Name fields to the extension number assigned to the port, and then select the Add Media Server Extension check box.
- Assign each port the same extension number here as you assigned it in the ASA configuration screens. Verify that the Media Server field is set to the name of the media server interface and change it if necessary.

IMPORTANT After you have completed all changes find, and then click the Update link on the navigation menu at the left side of the Integrated Management page. You must click this link to save your changes.



The 'Add User' form contains the following fields and controls:

- Primary Handle*: Text input with '5001' entered.
- User ID: Text input with '5001' entered.
- Password*: Password input with 6 dots.
- Confirm Password*: Password input with 6 dots.
- Host*: Dropdown menu with '172.16.20.11' selected.
- First Name*: Text input with '5001' entered.
- Last Name*: Text input with '5001' entered.
- Address 1: Text input.
- Address 2: Text input.
- Office: Text input.
- City: Text input.
- State: Text input.
- Country: Text input.
- Zip: Text input.
- Add Media Server: Checkmark icon.
- Extension: Text input.
- Fields marked * are required.
- Add: Button.

Figure 4: Add User

Mapping the Media Server Address to the MiCollab AM ports

To add a Media Server Address Map:

From the **List Communication Manager** page, select **Map**, and then click the **Edit** command.



The 'List Communication Manager Server Address Map' form displays a table with the following structure:

Commands	Name	Commands	Contact
Edit Delete	Voicemail_in	Edit Delete	sip:\${user}@172.16.20.13:5060
Add Another Map		Add Another Contact	

Enter a descriptive name and pattern for the SIP domain address to denote the routing. The full regular expression pattern in this example is:

^sip:[4][0][0-1][0-9]@blvu.AVSTlabs.local

Edit Communication Manager Map Entry

Name*

Pattern*

Replace URI ☒

Fields marked * are required.

Match the extensions to the pattern. The pattern uses regular expression syntax.

Creating a Hunt Group and Pilot Number

Define a hunt group for MiCollab AM and assign it a pilot number that is not associated with any port or extension. Set the caller display mode for the stations in the group so that they display the name of the group member receiving the call. Add all MiCollab AM extensions to the new hunt group. The following examples show typical hunt group configuration for this integration.

```
display hunt-group 2                               Page 1 of 60
      HUNT GROUP

Group Number: 2                                     ACD? n
Group Name: SIP VMail                               Queue? n
Group Extension: 5000                               Vector? n
Group Type: ucd-mia                                Coverage Path:
TN: 1                                                Night Service Destination:
COR: 1                                              MM Early Answer? n
Security Code:                                     Local Agent Preference? n
ISDN/SIP Caller Display: mbr-name
```

```
display hunt-group 2                               Page 2 of 60
      HUNT GROUP

LWC Reception: none                                AUDIX Name:

Message Center: none
```

```
display hunt-group 2                               Page 3 of 60
      HUNT GROUP

Group Number: 2  Group Extension: 5000  Group Type: ucd-mia
Member Range Allowed: 1 - 1500 Administered Members (min/max): 1/24
      Total Administered Members: 24
GROUP MEMBER ASSIGNMENTS
      Ext  Name (24 characters)  Ext  Name (24 characters)
1: 5001  5001                  14: 5014  5014
2: 5002  5002                  15: 5015  5015
3: 5003  5003                  16: 5016  5016
4: 5004  5004                  17: 5017  5017
5: 5005  5005                  18: 5018  5018
```

6: 5006	5006	19: 5019	5019
7: 5007	5007	20: 5020	5020
8: 5008	5008	21: 5021	5021
9: 5009	5009	22: 5022	5022
10: 5010	5010	23: 5023	5023
11: 5011	5011	24: 5024	5024
12: 5012	5012	25:	
13: 5013	5013	26:	

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, define the MiCollab AM hunt group 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, ringing without answer (RNA), or set to do-not-disturb mode (DND).

```
display coverage path 2
                        COVERAGE PATH

Coverage Path Number: 2
Hunt after Coverage? n
Next Path Number:                Linkage

COVERAGE CRITERIA

Station/Group Status   Inside Call   Outside Call
                        Active?         n
                        Busy?           y
                        Don't Answer?   y
                        All?            n
DND/SAC/Goto Cover?    y
Holiday Coverage?      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 a SIP Trunk Group.

IMPORTANT You must deactivate Secure SIP in this route pattern.

```
display route-pattern 1                               Page 1 of 3

Pattern Number: 1                                     Pattern Name: SIP
SCCAN? n      Secure SIP? 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 No. Numbering LAR
  0 1 2 3 4 W   Request      Dgts Format
      Subaddress
1: y y y y y n n      rest      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 Conversation 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.

```
display aar analysis 1                               Page 1 of 2
      AAR DIGIT ANALYSIS TABLE
      Percent Full: 1

Dialed      Total  Route  Call  Node ANI
String      Min Max  Pattern  Type  Num  Reqd
2           7   7   999    aar      n
3           7   7   999    aar      n
4           7   7   999    aar      n
4300        4   4   10    lev0      n
4400        4   4   9     aar      n
5           4   4   1     aar      n
6           7   7   999    aar      n
7           7   7   999    aar      n
8           7   7   999    aar      n
9           7   7   999    aar      n
          n
          n
```


n
n
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.

```
display aar digit-conversion 0          Page 1 of 2
      AAR DIGIT CONVERSION TABLE
      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
x11       3         3   0   ars  y      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 under Creating a Route Pattern.

```
display locations
      LOCATIONS

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

Loc. Name  Timezone Rule NPA          Proxy Sel.
No.        Offset
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. This section discusses each of these tasks.

IMPORTANT On Avaya software versions prior to version 4.0 the integration provides MWI support on SIP-based telephones only. MWI operations on circuit-switched digital telephones require the addition of a secondary, D/82-based integration for MWI purposes.

The Avaya TFTP server, which is installed with the telephone system, initializes IP-based telephones when they are connected to the network. The necessary telephone setting information is located in a text file to which the TFTP server has access. You must specify the name and location of this settings file in the Outbound file path box in the Server Options dialog box on the TFTP server. Make the following changes to the settings file for 4600-series telephones:

To update the Station Settings file:

- Set the internal dial plan length to the number of digits used in extension numbers.
- Allow call forwarding under busy and RNA conditions.
- Specify valid dialing strings in the DIALPLAN variable.
- Set the MWI server address variable to the IP address of the SIP trunk used in the integration.
- Specify the name of the domain containing the MiCollab AM server.

The following example shows a settings file configured for this integration.

```
## Telephone dial plan length (the length of an internal ext number).
SET PHNDPLENGTH 4

##### SIP-Specific Settings #####

SET CALLFWDSTAT 6
SET DIALPLAN      [4]xxx|[7]xxx|[5]xxx|*xx|#xx|9lxxxxxxxxxxx|
9[2-9]xxxxxx
SET MWISVR        "172.16.20.13"
SET SIPDOMAIN      "blvu.AVSTlabs.local"
```

To create a station definition for subscriber telephones:

At the ASA terminal, create a station definition for each subscriber extension as shown in the following examples. Set the MWI extension number equal to the station's own extension number, set the MWI type to **sip-adjunct**, and direct coverage to the path you created earlier.

```
display station 4010                               Page 1 of 4
          STATION

Extension: 4010      Lock Messages? n              BCC: 0
Type: 4610           Security Code: *              TN: 1
```


Configuring MiCollab AM

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

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

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

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

Configuring MiCollab AM for the Integration During Initial Installation

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

- 1 In the **Database Initialization Parameters** dialog box, configure the following options:
 - a In the **Mailbox Length** box, enter the mailbox length in digits.
 - b In the **First Extension** box, enter first extension number for the first line. You can also leave the **First Extension** box empty.
 - c From the **Manufacturer** 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**.

Click **Next**. The **Board Options** dialog box appears.

- f From the **Manufacturer** drop-down list, select **Virtual**.
- g From the **Model** drop-down list, select **SIP STACK**.
- h In the **Name** field, the name for this board is automatically generated. Enter a new name if necessary.
- i From the **Protocol** drop-down list, select **SIP IP RTP**.
- j 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**.

- 2 Click **OK**. The **Switch Options** dialog box appears.

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*.

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

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	The IP address of the SES platform
SIP Server Port	Enter the port number on which the SES platform listens for SIP messages. This port must match the SES platform's port. The default port number is 5060 .
SIP Domain Name	Enter the domain name containing the MiCollab AM server. This case-sensitive value must be the same as the Far-End Domain Name in the signaling group. NOTE This value is case-sensitive.
Transport for outgoing SIP messages	Select TCP (the default value)
Local IP Address to bind on	Enter the IP address of the network interface card (NIC) on the MiCollab AM server platform that supports the media server. 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 TCP port MiCollab AM listens for incoming SIP messages. The default value is 5060 .
SIP parser qualifier string	In cases of a single SIP integration 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.

In cases where there are multiple SIP integrations on the call server, use a string that is unique to each SIP integration.

For example:

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.*

The Fully Qualified Domain Name (FQDN) of the switch, such as pbx1.sipdomain.com.

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

PBX Registration Password	Enter the password that you assigned to the user definitions for the integrated ports in the section, Defining MiCollab AM Ports on the Media Gateway , earlier in this document
Media packet size (milliseconds)	MiCollab AM sends/receives packets containing the number of milliseconds worth of audio data set here. The default value is 20 .

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

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

c In the **Local Integration Settings** section, select the **Media Settings** view and configure the following option:

- Select the checkbox in the **Validate Remote Hosts for Media**, if you want to use this feature.

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

d Click **OK**. The **Switch Section Options** dialog box appears.

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

e In the **Local Switch Section Settings** section, select the **Required Parameters** view.

f In the **Incoming Hunt Mode** field, select the mode for this integration.

g In the **Hunt Group Access Code** field, type the hunt pilot number you defined earlier in the section, [Creating a Hunt Group and Pilot Number](#). This is the pilot number that users dial to reach MiCollab AM.

h Click **OK**.

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

If **MiCollab AM Configuration** does not open automatically after the configuration completes, open **MiCollab AM Configuration**, and select the **Lines** tab.

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.

IMPORTANT You must enter the PBX extension numbers that the Call Server is configured to answer or the integration will fail. The extension numbers are registered as SIP stations with the IP PBX during system startup.

Click **OK** to save all changes.

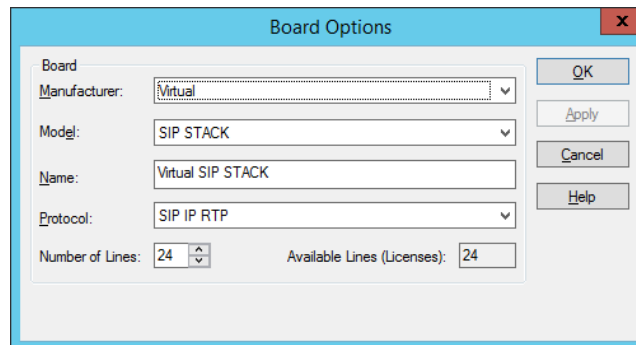
Configuring Existing MiCollab AM for the Integration

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.



- 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**.

Select the **Switches** tab, and click the **Add** button. The **Switch Integration Data Setup** dialog box appears.

g From the **Manufacturer** drop-down list, select **Avaya**.

h From the **Model** drop-down list, select **Communication Manager**.

i From the **Integration Type** drop-down list, select **SIP**.

4 Click **OK**. The **Switch Options** dialog box appears.

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*.

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

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	The IP address of the SES platform
SIP Server Port	Enter the port number on which the SES platform listens for SIP messages. This port must match the SES platform's port. The default port number is 5060 .
SIP Domain Name	Enter the domain name containing the MiCollab AM server. This case-sensitive value must be the same as the Far-End Domain Name in the signaling group. NOTE This value is case-sensitive.
Transport for outgoing SIP messages	Select TCP (the default value)
Local IP Address to bind on	Enter the IP address of the network interface card (NIC) on the MiCollab AM server platform that supports the media server. 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 TCP port MiCollab AM listens for incoming SIP messages. The default value is 5060 .
SIP parser qualifier string	<p>In cases of a single SIP integration 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. <i>The hunt number must be unique across all IP integrations.</i></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>
PBX Registration Password	Enter the password that you assigned to the user definitions for the integrated ports in the section, Defining MiCollab AM Ports on the Media Gateway , earlier in this document
Media packet size (milliseconds)	MiCollab AM sends/receives packets containing the number of milliseconds worth of audio data set here. The default value is 20 .

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

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

c In the **Local Integration Settings** section, select the **Media Settings** view and configure the following option:

- Select the checkbox in the **Validate Remote Hosts for Media**, if you want to use this feature.

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

d Click **OK**. The **Switch Section Options** dialog box appears.

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

- e** In the **Local Switch Section Settings** section, select the **Required Parameters** view.
- f** In the **Incoming Hunt Mode** field, select the mode for this integration.
- g** In the **Hunt Group Access Code** field, type the hunt pilot number you defined earlier in the section, [Creating a Hunt Group and Pilot Number](#). This is the pilot number that users dial to reach MiCollab AM.
- h** Click **OK**.

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

Select the **Lines** tab.

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.

IMPORTANT You must enter the PBX extension numbers that the Call Server is configured to answer or the integration will fail. The extension numbers are registered as SIP stations with the IP PBX during system startup.

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**.

In the **Integration Options** dialog box, go to the **Local Integration Settings** section.

From the **View** drop-down list, select **Failover Server Settings**.

Click the **Add Failover Server** button. Two new rows are added to configure the secondary SIP server.

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	Enter the TCP/IP address or an FQDN of the secondary node. For example: The IP address 123.45.6.789 as displayed on the Review/Modify SIP Gateway screen.

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.

IMPORTANT This value must match the configuration on the Gateway of the secondary node.

Secondary SIP Server Port	Enter the port number of the secondary node. The default value is 5060 .
---------------------------	---------------------------------------------------------------------------------

From the **View** drop-down list, select **Integration Specific Parameters**. The **Integration Specific Parameters** view appears.

In the **Integration Specific Parameters** list, enter the information as shown in the following table:

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

Table 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	<p>How this should be set depends on the gateway used for the integration.</p> <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.

Click **Apply** to save the changes.

To add another failover server repeat **Steps 4-9**.

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**.

In the **Integration Options** dialog box, go to the **Local Integration Settings** section.

From the **View** drop-down list, select **Failover Server Settings**.

In the **Failover Server Settings** view, click the **Remove Failover Server** button.

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.

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

Appendix A—Adding MWI Support for Non-IP Telephones

IMPORTANT This section is required for Avaya software versions prior to version 4.0. If the software version of your Avaya telephone system is 4.0 or later, skip this section.

The SIP integration supports message-waiting indicator (MWI) operations on all IP-based telephones connected to the telephone system. If the system uses older extensions that are not based on IP technology however, the SIP integration cannot set or clear MWIs at those extensions. To provide proper MWI operation on non-IP telephones prior to Avaya version 4.0, you must add a secondary D/82 Station Set integration to MiCollab AM.

Refer to the Avaya 8500 D/42 or D/82 Station Set Emulation Integration Technical Note, P/N 1081-50538-00 Rev 06 for more specific information on programming the telephone system for the D/82 integration.

IMPORTANT You must order and install licenses for the additional MWI ports and the secondary Avaya D/82 Station Set integration before you begin the following procedure. In addition, before beginning, you must order and install a Dialogic D/42 or D/82 linecard in your MiCollab AM server platform. For instructions on installing the linecard, refer to the Dialogic D/42JCT-U or D/82JCT-U Linecard Installation and Replacement spare parts document.

To add MWI support for non-IP telephones to the Avaya Communication Manager SIP integration:

- 1 From the MiCollab AM Configuration utility, shut down the system server.
- 2 Click the **Integrations** tab, select **Avaya Communication Manager SIP** from the list of integrations, and then click **Edit** to modify the integration.
- 3 From the **View** list in the **Integration Options** dialog box, select **Advanced Message Waiting Settings**.
- 4 In the list of settings, clear the **Allow MWI Operations** check box, and then click **OK**.
- 5 Click the **Switches** tab, and then click **Add**.
- 6 From the **Manufacturer** list in the **Switch Options** dialog box, click **Avaya**.
- 7 Select **Definity** from the **Model** list, and then click **OK**.
- 8 Click the **Integrations** tab, and then click **Add**.
- 9 From the **Switch Name** box in the **Integration Options** dialog box, select **Avaya Definity**.
- 10 From the **Type** list, select **Dialogic D/82 2-wire set emulation**.
- 11 Click **OK** to return to MiCollab AM Configuration, and then click the **Lines** tab.

- 12 In the **Integration** column for the first D/82 line port, pull down the list of available integrations and select **Avaya Definity Dialogic D/82 2-wire** set emulation.
- 13 Make sure that the **Callouts** and **Open Lines** check boxes are selected for the line port.
- 14 Repeat steps 12 and 13 for the remaining D/82 line ports available.
- 15 Click **Apply**, and then restart MiCollab AM.

Programming Dialogic Configuration Manager

By default, the Dialogic System Release Configuration Manager program sets the parameter PBXSwitch to Nortel_Norstar. You must change this parameter to the appropriate PBX type you are integrating with MiCollab AM.

IMPORTANT If this is an existing MiCollab AM system with a previous version of Dialogic software installed, you must remove it and any Dialogic point release software before you install MiCollab AM version 9.1 and the Dialogic Software Support Components on the Call Server platform.

If the MiCollab AM version 9.1 InstallShield Wizard detects an existing version of Dialogic software during the setup process, the installation is aborted and a message displays to un-install all Dialogic software first.

For more information on removing previous versions of Dialogic software, refer to the MiCollab AM online help or the *Dialogic and Aculab System Administrator Guide*.

To change the PBXSwitch parameter for each D/42 or D/82 linecard installed:

- 1 Select **Start > Programs > Dialogic System Software**, and then select **Configuration Manager-DCM**.
- 2 Stop the **Dialogic** service if it is running.
- 3 Double-click the installed D/42 or D/82 linecard to open the **Properties** sheet.
- 4 Click the **Miscellaneous** tab, and then select the **PBXSwitch** parameter.
- 5 In the **Values** list, select **Lucent_2_wire** as the type of integration.
- 6 On the **Telephony Bus** tab, verify that the correct PCM encoding scheme is selected. The default value is automatic or U-Law; you must change this value to A-Law outside of the U.S. and Japan.
- 7 Click **OK**.
- 8 Repeat steps 3-7 for each D/42 or D/82 linecard that is installed.
- 9 Restart the **Dialogic** service and close **Dialogic Configuration Manager**.