

**MiCollab Advanced Messaging
Genband EXPERiUS Application Server
SIP Peering
Integration Technical Note**

For version 9.1 and above

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Contents

Preface	5
References	5
Documentation	5
Documentation Updates	6
Help	6
Document Conventions	6
Features Supported by This Integration	7
Critical Application Considerations	10
Installation Requirements	12
Telephone System Requirements	12
MiCollab AM Requirements	12
System Overview	13
Network Diagram	13
Integration Overview	13
Audiocodes Gateway	14
Routing Calls	14
EXPERiUS AS	16
Overview	16
System Management Console	16
Network Data and Mtc	16
Addresses	16
External Nodes	17
Informational Element: Mitel	18
Informational Element: AudioCodesGW	19
SIP Profile	19
SIP Profile: Mitel	20
SIP Profile: Audiocodes	23
Network Elements	25
Provisioning Manager	26
Root Domain	26

Class of Service (COS)	27
Resources	28
Service Nodes	29
Mitel MiCollab AM Logical Entity	32
Telephony Routes	34
Route List	36
Users	38
EXPERiUS AS Feature	41
Configuring MiCollab AM	45
Configuring MiCollab AM for the Integration During Initial Installation	45
Configuring Existing MiCollab AM for the Integration	48
Configuring MiCollab AM for SIP Failover	51
Changing the Network Binding Order on the MiCollab AM Platform	54
Windows Server 2012 R2	54
Windows Server 2016 / 2019	55
Configuring Quality of Service (QoS)	56

Preface

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

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

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>SIP Routing Manger Installation 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 EXPERiUS 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	No

Table 3. Integration features supported for Genband EXPERiUS Application Server SIP Peering

Feature	Supported	Notes
Automatic subscriber logon	Yes	
ANI/CLI	Yes	
Announce Busy greeting on forwarded calls	Yes	
Call screening	Yes	Note 1
Caller queuing	Yes	Note 1, 2
DNIS	Yes	
End-to-end DTMF, attendant console	Yes	
End-to-end DTMF, proprietary telephones	Yes	
Fax Tone Detection	Yes	
Internal calling party ID for reply	Yes	
Live record, integrated	No	
Live reply to sender	Yes	
Message notification callouts	Yes	
MWI, set/clear	Yes	
MWI, inband/outband	Outband	
Networking, analog	No	
Overflow from MiCollab AM to attendant	Yes	
Overflow to MiCollab AM from attendant	Yes	
PBX-provided disconnect signaling	Yes	
Revert to operator	Yes	
SRTP	Yes	Note 5
TLS	Yes	Note 5
Transfers, blind	Yes	Note 3
Transfers, confirmed	Yes	Note 3

Transfers, fully supervised	Yes	Note 3
Transfers, monitored	Yes	Note 3
Trunk ID for call routing	No	
Spans Multiple Call Servers	Yes	Note 4
Multiple Integrations	Yes	Note 6

NOTES

1. Only available using supervised transfers
2. Caller Queuing is specific to each local Call Server. Call Servers within the system are unaware of queued calls to the same subscriber on other Call Servers. For more information, refer to the [Critical Application Considerations](#) section of this document for limitations on these features.
3. To use supervised transfers, Patch 24 and any dependencies must be installed on the SST.
4. Multiple call servers can be used via the SIP Routing Manager.
5. MiCollab AM supports negotiation for SRTP media streams using the Secure RTP profile defined in RFC 3711 with the offer/answer model defined in RFC 3264. To enable SRTP, RTP, or both, see integration configuration options documentation for the switch. The default setting is RTP. Please note that MiCollab AM doesn't support RFC 5939 which is an extension of RFC 3264. Also, please note that SRTP has not been qualified for this integration, and no switch programming is available for setting up SRTP on the switch side. However, SRTP may be enabled as described above, and technical support will be available on a best effort basis.
6. See the [Critical Application Considerations](#) section.

Critical Application Considerations

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

- You must configure the **Incoming Hunt Mode** in the **Switch Section Options** dialog box. This integration supports terminal, circular, reverse terminal and reverse circular hunt modes only. The default mode is Terminal.
- You must configure the **Trunk Group Access Code** in the **Switch Section Options** dialog box. This code cannot conflict with extensions. This must match with the DN prefix programmed for the Gateway Endpoint associated with this MiCollab AM Call Server.
- 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 Call Queuing feature does not transcend the Call Server. Calls may be queued on multiple Call Servers for the same subscriber but Call Servers do not have knowledge of calls in the queue on other Call Servers within the system. Callers may be prompted with specific information about their place in the queue; however, the information pertains to the specific Call Server on which their call is queued.
- The MiCollab AM **Integration Options** parameter, **Validate Remote Hosts for Media** validates each incoming audio packet and accepts it only if it is sent from a valid endpoint. The parameter is disabled by default. Enabling this parameter causes MiCollab AM to reject RTP packets from invalid endpoints, rejects MWI packets that timeout after a specified number of times, and overcomes port lockups when callers hang up while MiCollab AM is performing a blind transfer.

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

- 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 Genband SIP Trunk 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
- It is recommended to use Windows Server 2016 or later for Integrations that use Session Initiation Protocol (SIP) Transport Layer Security (TLS) when FIPS is enabled on MiCollab AM. Older versions of Windows use algorithms that are not FIPS compliant to export the certificate information used for TLS. Because of this, MiCollab AM will not be able to access certificate-related data.

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

- Genband EXPERiUS Application Server Version 10.4.

MiCollab AM Requirements

- MiCollab AM version 9.1
- Mitel software key diskette or feature file with the Genband SIP Trunk integration enabled and one Virtual SIP and RTP license enabled for each port involved in the integration.
- One or two 10 MB, 100 MB, or 1000 MB (gigabit) network interface cards with cables.

System Overview

Network Diagram

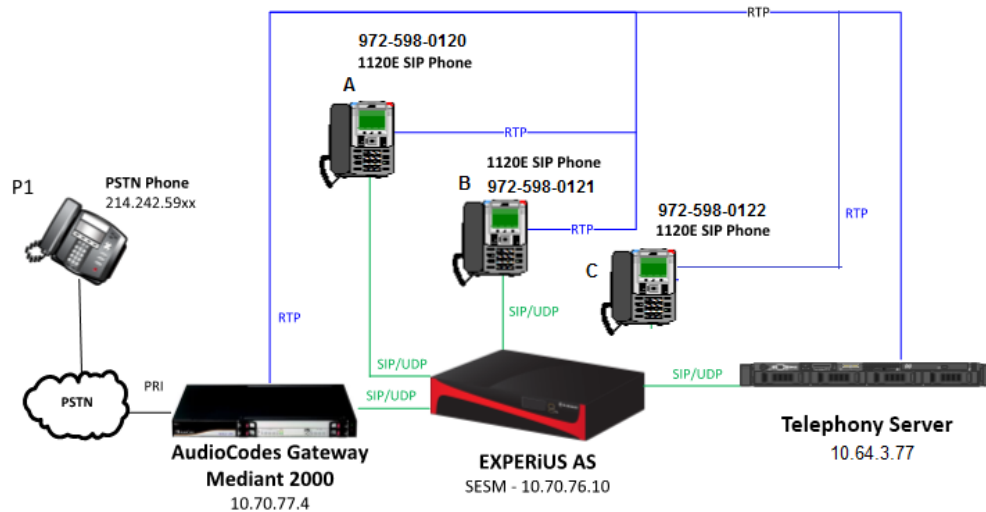


Figure 1. Network Diagram

Integration Overview

Mitel MiCollab AM 9.1 and EXPERiUS AS 10.3 are assumed to be configured within their product configuration settings and this configuration guide is used to highlight the configurations that are required to be added/changed/verified for the integration.

The EXPERiUS AS and Mitel MiCollab AM are deployed in a Peering mode where each supports local subscriber devices/mailboxes with a shared dial plan. The EXPERiUS AS provides PSTN access through a PRI Gateway.

System component software/firmware versions are captured in the following table.

Table 4. System Components and Software/Firmware Versions

Component	Version
Mitel MiCollab AM	6.0 Build 474 Update 1
EXPERiUS AS	10.3 (MCP_17.0.18.1)
Avaya 1120e (User A)	SIP1120e.04.04.20.00
Avaya 1120e (User B)	SIP1120e.04.04.18.00

Audiocodes Gateway

In this section, only the pertinent configurable items that support the integration with EXPERiUS AS are discussed. It is assumed that many items are already configured or to be set to the default values.

To configure Audiocodes Gateway:

- 1 Login to Audiocodes with appropriate User ID and password.
- 2 On the left pane, click **Configuration**.
- 3 Navigate to Protocol **Definition** > **SIP General Parameters**.

SIP General Parameters	
NAT IP Address	0.0.0.0
PRACK Mode	Supported
Channel Select Mode	Cyclic Ascending
Enable Early Media	Enable
183 Message Behavior	Progress
Session-Expires Time	3600
Minimum Session-Expires	1800
Session Expires Method	Re-INVITE
Asserted Identity Mode	Disabled
Fax Signaling Method	T.38 Relay
Detect Fax on Answer Tone	Initiate T.38 on Preamble
SIP Transport Type	UDP
SIP UDP Local Port	5060
SIP TCP Local Port	5060
SIP TLS Local Port	5061
Enable SIPs	Disable
Enable TCP Connection Reuse	Enable
TCP Timeout	5
SIP Destination Port	5060
Use user=phone in SIP URL	Yes

- 4 Make sure **PRACK Mode** is set to default value **Supported**.

Routing Calls

Use the following procedure to administer call routing for calls originating from PSTN to EXPERiUS AS using UDP.

To route calls:

- 1 Navigate to **Configuration** > **Protocol Configuration** > **Routing Tables** > **Tel to IP Routing**.

Tel to IP Routing

Basic Parameter List

Routing Index: 11-20

Tel To IP Routing Mode: Route calls before manipulation

	Src. Trunk Group ID	Dest. Phone Prefix	Source Phone Prefix	Dest. IP Address	Port	Transport Type	De IPG 1
11	*	9725980120	*	lab.tekvizion.com	5060	UDP	
12	*	9725980121	*	lab.tekvizion.com	5060	UDP	
13	*	9725980122	*	lab.tekvizion.com	5060	UDP	
14	*	9725980123	*	lab.tekvizion.com	5060	UDP	
15	*	9725980129	*	lab.tekvizion.com	5060	UDP	
16	*	9725980124	*	lab.tekvizion.com	5060	UDP	
17	*	9722657203	*	lab.tekvizion.com	5060	UDP	
18	*	9722657204	*	lab.tekvizion.com	5060	UDP	
19	*	9722657205	*	lab.tekvizion.com	5060	UDP	
20	*	*	*	lab.tekvizion.com	5060	UDP	

- 2 In the **Dest. Phone Prefix** field, enter the destination phone numbers to route the incoming calls from the PSTN that matches this pattern to EXPERiUS AS.
- 3 In the **Source Phone Prefix** field, enter a * to allow routing for any source telephone number to the EXPERiUS AS from PSTN.
- 4 **Dest. IP Address** is set to the EXPERiUS AS domain name **lab.tekvizion.com**.
- 5 **Transport Type** is set to **UDP**.

EXPERiUS AS

Overview

EXPERiUS AS is configured through its graphical System Management Console (i.e. System Manager) as well as through its web-based Provisioning Management Interface. Provisioning requirements in both interfaces are discussed in the following sections.

EXPERiUS AS is a very flexible and powerful applications server and its configuration options can be extensive. This document discusses only the pertinently configurable items that support the integration with MiCollab AM at the SIP level. Many items are to be set to the default values.

System Management Console

The System Manager provides a provisioning portal into various data which is largely domain independent. That is, this data is shared within EXPERiUS AS across multiple root domains.

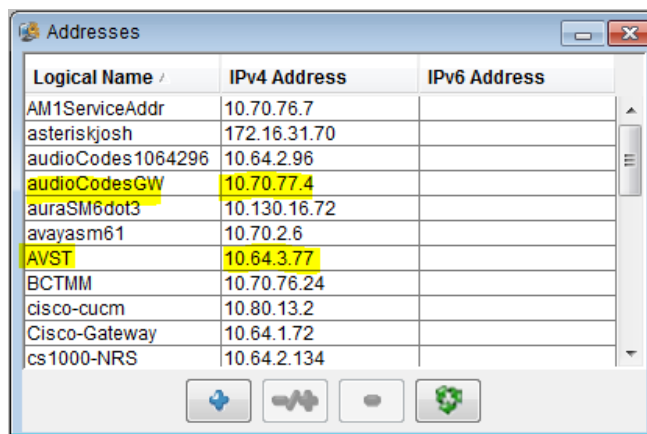
Network Data and Mtc

Within System Manager, all external server nodes in the eco-system must be identified. For this integration, the external nodes are:

- Mitel MiCollab AM
- Audiocodes Gateway (providing PSTN access)

Within System Manager, there are three levels of external node configuration – an IP address, an External Node logical address, and a set of attributes in the Informational Element definition.

Addresses



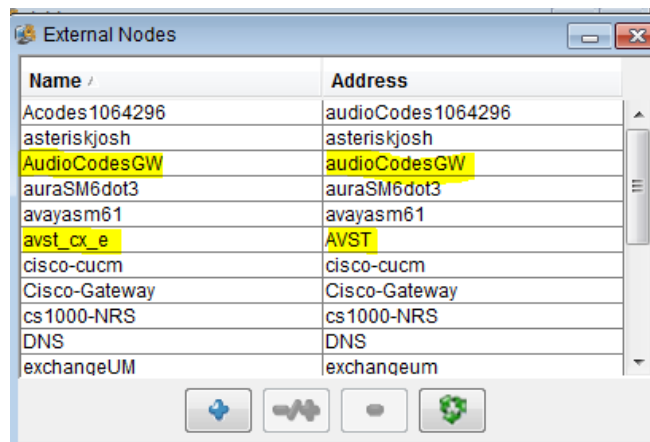
Logical Name	IPv4 Address	IPv6 Address
AM1ServiceAddr	10.70.76.7	
asteriskjosh	172.16.31.70	
audioCodes1064296	10.64.2.96	
audioCodesGW	10.70.77.4	
auraSM6dot3	10.130.16.72	
avayasm61	10.70.2.6	
AVST	10.64.3.77	
BCTMM	10.70.76.24	
cisco-cucm	10.80.13.2	
Cisco-Gateway	10.64.1.72	
cs1000-NRS	10.64.2.134	

Figure 2. System Management Console - Addresses Screen

To configure addresses:

- 1 Click on the + button to add the **Mitel** and **audioCodesGW** addresses.
- 2 Provide a meaningful **Logical Name** to each field.
- 3 Additional configuration is required to specify the IP addresses of the components internal to the AS such as Session Manager, Media Application Server (MAS), EMS, etc. It is assumed that this provisioning information is already in place.

External Nodes



Name	Address
Acodes1064296	audioCodes1064296
asteriskjosh	asteriskjosh
AudioCodesGW	audioCodesGW
auraSM6dot3	auraSM6dot3
avayasm61	avayasm61
avst_cx_e	AVST
cisco-cucm	cisco-cucm
Cisco-Gateway	Cisco-Gateway
cs1000-NRS	cs1000-NRS
DNS	DNS
exchangeUM	exchangeum

Figure 3. System Management Console - External Nodes Screen

To configure external nodes:

- 1 Add an external node to represent each external server identified in the **Address** field.
- 2 Configure the values as highlighted in Figure 3. System Management Console - External Nodes Screen above.

NOTE For simplicity, the values in the **Name** field were made identical to the **Logical Address name**.

Informational Element: Mitel

The screenshot shows the 'Edit Informational Element' window. The 'ShortName' is 'avst'. The 'LongName' is 'avst_cx_e'. 'Trusted' is checked. 'ExemptDoSProtection' is unchecked. 'Identification Port' is '0'. 'Type' is 'Gateway'. 'CTI Support' is a link to 'Configure CTI Support'. Under 'Transport Information', 'Node' is 'avst_cx_e'. 'Enable SIP UDP Port' is checked with a value of '5060'. 'Enable SIP TCP Port' is unchecked with a value of '5060'. 'Enable SIP TLS Port' is unchecked with a value of '5061'. 'Apply' and 'Cancel' buttons are at the bottom.

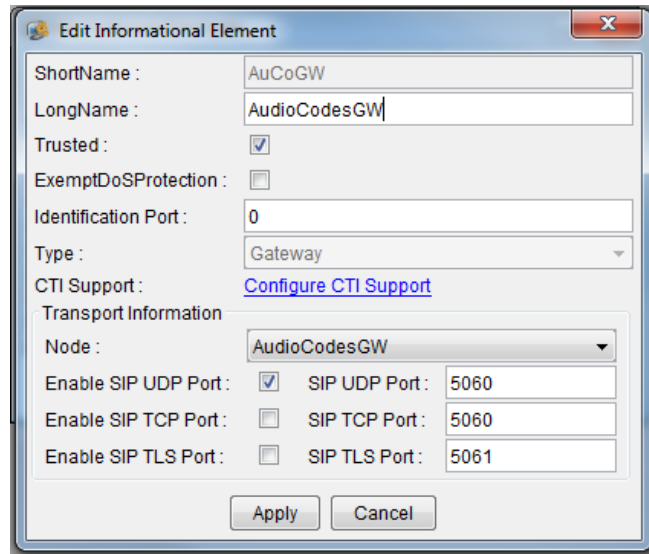
Figure 4. System Management Console - Edit Information Element Screen

To configure information element for Mitel:

- 1 Create an Informational Element for *Mitel_cx_e*.
- 2 Configure the values as follows:
 - **LongName:** Mitel_cx_e
 - **Trusted:** Checked. This will relax the authentication functionality for the calls to Mitel MiCollab AM.
 - **Node:** Select the server name created in the [External Nodes](#) section.
 - **Enable SIP UDP Port:** Checked with the listen port value specified, since the network topology calls for UDP transport between the peer servers.

NOTE Once this configuration is complete, this external server will be referenced by its **LongName** during the configuration procedures using the EXPERiUS AS Provisioning Manager (discussed in the [Provisioning Manager](#) section).

Informational Element: AudioCodesGW



ShortName : AuCoGW

LongName : AudioCodesGW

Trusted : ☒

ExemptDoSPProtection : ☐

Identification Port : 0

Type : Gateway

CTI Support : [Configure CTI Support](#)

Transport Information

Node : AudioCodesGW

Enable SIP UDP Port : ☒ SIP UDP Port : 5060

Enable SIP TCP Port : ☐ SIP TCP Port : 5060

Enable SIP TLS Port : ☐ SIP TLS Port : 5061

Apply Cancel

Figure 5. System Management Console - Edit Information Element Screen for AudioCodesGW

To configure information element for AudioCodesGW:

- 1 Create an Informational Element to represent PSTN access through the Audiocodes.
- 2 Configure the options follows:
 - **LongName:** AudioCodesGW
 - **Trusted:** Checked
 - **Node:** AudioCodesGW
 - **Enable SIP UDP Port:** Checked with the listen port value

SIP Profile

A SIP Profile is a set of various SIP attributes that allow for a fine tuning of the SIP-level communication with external servers. It is associated with an external node in the Service Node configuration from the Provisioning Manager interface as discussed in the Provisioning Manager.

The profiles for both external servers are created as indicated as highlighted in the figure below.

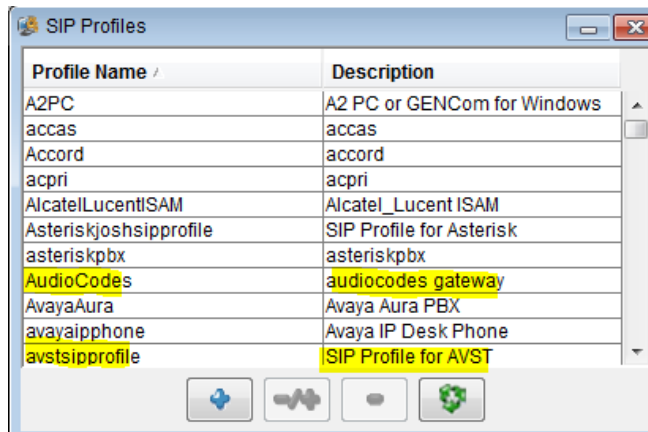


Figure 6. System Management Console - SIP Profiles Screen

SIP Profile: Mitel

Profile Name : mycompanysipprofil

Description : SIP Profile for MyCompany

Signaling

Request Selection : [Select Requests](#)

Redirect Response Allowed : ☒

Header Selection : [Select Headers](#)

Filter Incoming Allow Header Content : [Select Allow Methods](#)

Service Configuration : [Configure Service XML Data](#)

Tags Allowed : ☒

Allow User Info Parameter : ☒

Request x-nt-profile Header : ☐

Add calling party display : ☒

Max Headers : 200

Max Header Length : 1024

Max Block Size : 4096

Hookflash URI username : flash

Digit Timeout URI username : digit_timeout

Emergency Mid Call Reject : ☐

User User Mode1 : ☐

Require Priority RingBack : ☐

Play Announcements : ☐

Unique Call IDs : ☐

Ephemeral Source port : ☐

Continues on the next page

Allow P-Asserted Identity Header : ☒
 Use Calling Party as From : ☒
 Use Options : ☒
 Consult XFer SVC needed : ☐
 Force Homed User : ☐
 Require Conference Parameter Swap : ☐
 Require Refer To Privacy Swap : ☐
 Delay XFer202 : ☐
 Alert Information Set Selection : [Select Alert Information Set](#)
 Subscribe Param Selection : [Select Subscribe Params](#)
 Require Alert Info Header : ☐
 Refer Response : ☐
 Suppress Long call : ☐
 Static Client Type : ☐
 Refer To Substitution : ☐
 IN Session Authentication : ☐
 MCD Update Call Model : ☐
 Use From Header For Subr Lookup : ☐
 Add Diversion Header : ☒
 Use Request URI As TO : ☐
 Remove Unknown Paid : ☐
 Handle Refer On As : ☐
 Use IP as FROM Domain : ☐
 Remove Replaces Support : ☐
 Remove NT-Endpoint from Request URI : ☐
 Remove NT-Endpoint from Contact : ☐
 Alteon 302 Redirection : ☐
 Allow DualCli when Privacy header is Set : ☐
 Require PRACK : ☐
 Use UA-Profile Event Package for MWI : ☐ Special Condition Tone ▾
 Override Host in From URI after Translation : ☐
 Set username for CLI unavailable : ☐ unavailable ▾
 Set username for CLI private : ☐ Private number ▾
 AS Provides Subsequent Ringback : ☐
 Treats Sendonly as Hold : ☐
 Remove Phone Context : ☐
 PIDF-LO : ☐
 Use DN For Paid : ☐
 Use PCharge Info : ☐
 No Ring Alert Info : http://127.0.0.1/No-Ringing ▾
 Multi-Mode Handset (MMH) : ☐
 NTMMH : ☐
 Apply Privacy On Trusted Node : ☐

Continued on the next page

Do Not Send Route Header :	<input type="checkbox"/>
Disable Slow Start :	Select Services
Foreign Server Use As Interapp :	<input type="checkbox"/>
Remove NT parameters from Refer-To :	<input type="checkbox"/>
From Change Header Allowed :	<input type="checkbox"/>
Send "183 Session Progress" Notify For Transfer In Progress :	<input type="checkbox"/>
Use Default IM Encoding :	<input type="checkbox"/>
Add CDPad Parameter :	<input type="checkbox"/>
RFC4235 Compliant Dialog NOTIFY :	<input type="checkbox"/>
Retain Contacts On Active Call :	<input type="checkbox"/>
VM Server Indication in MWI :	<input type="checkbox"/>
Use 401 for Authentication :	<input type="checkbox"/>
Post Progress Signaling Alteration :	<input type="checkbox"/>
Use 401 for Only REGISTER Authentication :	<input type="checkbox"/>
BLF - Same Dialog ID for Forked Calls :	<input type="checkbox"/>
Supported Intercom Header :	<input type="checkbox"/> <input type="text"/>
Use DN for Request URI :	<input type="checkbox"/>
Send "180 Ringing" Notify For Transfer After "202 Accepted" :	<input type="checkbox"/>
Dialog Notify Update For Advatel :	<input type="checkbox"/>
Correct Refer to For Advatel :	<input type="checkbox"/>

Media

Audio Codec Selection :	Select Audio Codecs
Video Codec Selection :	Select Video Codecs
Insert PTimes :	<input type="button" value="None"/>
Info Digit Negotiation :	<input checked="" type="checkbox"/>
Codec Change :	<input checked="" type="checkbox"/>
Pivot Allowed :	<input type="checkbox"/>
All Content :	<input type="checkbox"/>
InfoDigit :	<input type="checkbox"/>
Insert38Desc :	<input type="checkbox"/>
Hold Needed :	<input type="checkbox"/>
Use Network PTime :	<input type="checkbox"/>
Remove SDP From PRACK :	<input type="checkbox"/>
Allow Avaya Enterprise Content :	<input type="checkbox"/>
Remove SRTP :	<input type="checkbox"/>
Multiple Early Media Dialog :	<input checked="" type="checkbox"/>
Remove Redundant SDP from 200 OK :	<input type="checkbox"/>
Supports Early Media Detection :	<input type="checkbox"/>
RFC 3264 Compliant Hold-Retrieve :	<input type="checkbox"/>
Remove Application Media Attribute If Collab Session :	<input type="checkbox"/>
Drop Calls With No Audio Codecs Following Filtering :	<input type="checkbox"/>

<input type="button" value="Apply"/>	<input type="button" value="Cancel"/>	<input type="button" value="Copy"/>
--------------------------------------	---------------------------------------	-------------------------------------

SIP Profile: Audiocodes

Profile Name :	AudioCodes
Description :	audiocodes gateway
Signaling	
Request Selection :	Select Requests
Redirect Response Allowed :	<input checked="" type="checkbox"/>
Header Selection :	Select Headers
Filter Incoming Allow Header Content :	Select Allow Methods
Service Configuration :	Configure Service XML Data
Tags Allowed :	<input checked="" type="checkbox"/>
Allow User Info Parameter :	<input checked="" type="checkbox"/>
Request x-nt-profile Header :	<input type="checkbox"/>
Add calling party display :	<input type="checkbox"/>
Max Headers :	200
Max Header Length :	1024
Max Block Size :	4096
Hookflash URI username :	flash
Digit Timeout URI username :	digit_timeout
Emergency Mid Call Reject :	<input checked="" type="checkbox"/>
User User Mode1 :	<input type="checkbox"/>
Require Priority RingBack :	<input checked="" type="checkbox"/>
Play Announcements :	<input checked="" type="checkbox"/>
Unique Call IDs :	<input type="checkbox"/>
Ephemeral Source port :	<input type="checkbox"/>
Allow P-Asserted Identity Header :	<input checked="" type="checkbox"/>
Use Calling Party as From :	<input type="checkbox"/>
Use Options :	<input type="checkbox"/>
Consult XFer SVC needed :	<input type="checkbox"/>
Force Homed User :	<input type="checkbox"/>
Require Conference Parameter Swap :	<input type="checkbox"/>
Require Refer To Privacy Swap :	<input type="checkbox"/>
Delay XFer202 :	<input type="checkbox"/>
Alert Information Set Selection :	Select Alert Information Set
Subscribe Param Selection :	Select Subscribe Params
Require Alert Info Header :	<input type="checkbox"/>
Refer Response :	<input type="checkbox"/>
Suppress Long call :	<input type="checkbox"/>
Static Client Type :	<input type="checkbox"/>
Refer To Substitution :	<input checked="" type="checkbox"/>
IN Session Authentication :	<input type="checkbox"/>

Continued on the next page

MCD Update Call Model :	<input type="checkbox"/>
Use From Header For Subr Lookup :	<input type="checkbox"/>
Add Diversion Header :	<input type="checkbox"/>
Use Request URI As TO :	<input checked="" type="checkbox"/>
Remove Unknown Paid :	<input checked="" type="checkbox"/>
Handle Refer On As :	<input type="checkbox"/>
Use IP as FROM Domain :	<input type="checkbox"/>
Remove Replaces Support :	<input type="checkbox"/>
Remove NT-Endpoint from Request URI :	<input checked="" type="checkbox"/>
Remove NT-Endpoint from Contact :	<input type="checkbox"/>
Alteon 302 Redirection :	<input type="checkbox"/>
Allow DualCli when Privacy header is Set :	<input type="checkbox"/>
Require PRACK :	<input type="checkbox"/>
Use UA-Profile Event Package for MWI :	<input type="checkbox"/> Special Condition Tone ▾
Override Host in From URI after Translation :	<input type="checkbox"/>
Set username for CLI unavailable :	<input type="checkbox"/> unavailable ▾
Set username for CLI private :	<input type="checkbox"/> Private number ▾
AS Provides Subsequent Ringback :	<input type="checkbox"/>
Treats Sendonly as Hold :	<input checked="" type="checkbox"/>
Remove Phone Context :	<input type="checkbox"/>
PIDF-LO :	<input type="checkbox"/>
Use DN For Paid :	<input checked="" type="checkbox"/>
Use PCharge Info :	<input type="checkbox"/>
No Ring Alert Info :	http://127.0.0.1/No-Ringing ▾
Multi-Mode Handset (MMH) :	<input type="checkbox"/>
NTMMH :	<input type="checkbox"/>
Apply Privacy On Trusted Node :	<input type="checkbox"/>
Do Not Send Route Header :	<input type="checkbox"/>
Disable Slow Start :	Select Services
Foreign Server Use As Interapp :	<input type="checkbox"/>
Remove NT parameters from Refer-To :	<input type="checkbox"/>
From Change Header Allowed :	<input checked="" type="checkbox"/>
Send "183 Session Progress" Notify For Transfer In Progress :	<input type="checkbox"/>
Use Default IM Encoding :	<input type="checkbox"/>
Add CDPad Parameter :	<input checked="" type="checkbox"/>
RFC4235 Compliant Dialog NOTIFY :	<input type="checkbox"/>
Retain Contacts On Active Call :	<input type="checkbox"/>
VM Server Indication in MWI :	<input type="checkbox"/>

Continued on the next page

Use 401 for Authentication :	<input type="checkbox"/>
Post Progress Signaling Alteration :	<input type="checkbox"/>
Use 401 for Only REGISTER Authentication :	<input type="checkbox"/>
BLF - Same Dialog ID for Forked Calls :	<input type="checkbox"/>
Supported Intercom Header :	<input type="checkbox"/> <input type="text"/>
Use DN for Request URI :	<input type="checkbox"/>
Send "180 Ringing" Notify For Transfer After "202 Accepted" :	<input type="checkbox"/>
Dialog Notify Update For Advatel :	<input type="checkbox"/>
Correct Refer to For Advatel :	<input type="checkbox"/>

Media

Audio Codec Selection :	Select Audio Codecs
Video Codec Selection :	Select Video Codecs
Audio PTime Selection :	Select Audio PTimes
Insert PTimes :	<input type="button" value="None"/>
Info Digit Negotiation :	<input type="checkbox"/>
Codec Change :	<input checked="" type="checkbox"/>
Pivot Allowed :	<input checked="" type="checkbox"/>
All Content :	<input type="checkbox"/>
InfoDigit :	<input type="checkbox"/>
Insert38Desc :	<input type="checkbox"/>
Hold Needed :	<input type="checkbox"/>
Use Network PTime :	<input type="checkbox"/>
Remove SDP From PRACK :	<input type="checkbox"/>
Allow Avaya Enterprise Content :	<input type="checkbox"/>
Remove SRTP :	<input type="checkbox"/>
Multiple Early Media Dialog :	<input checked="" type="checkbox"/>
Remove Redundant SDP from 200 OK :	<input type="checkbox"/>
Supports Early Media Detection :	<input checked="" type="checkbox"/>
RFC 3264 Compliant Hold-Retrieve :	<input type="checkbox"/>
Remove Application Media Attribute If Collab Session :	<input type="checkbox"/>
Drop Calls With No Audio Codecs Following Filtering :	<input type="checkbox"/>

Network Elements

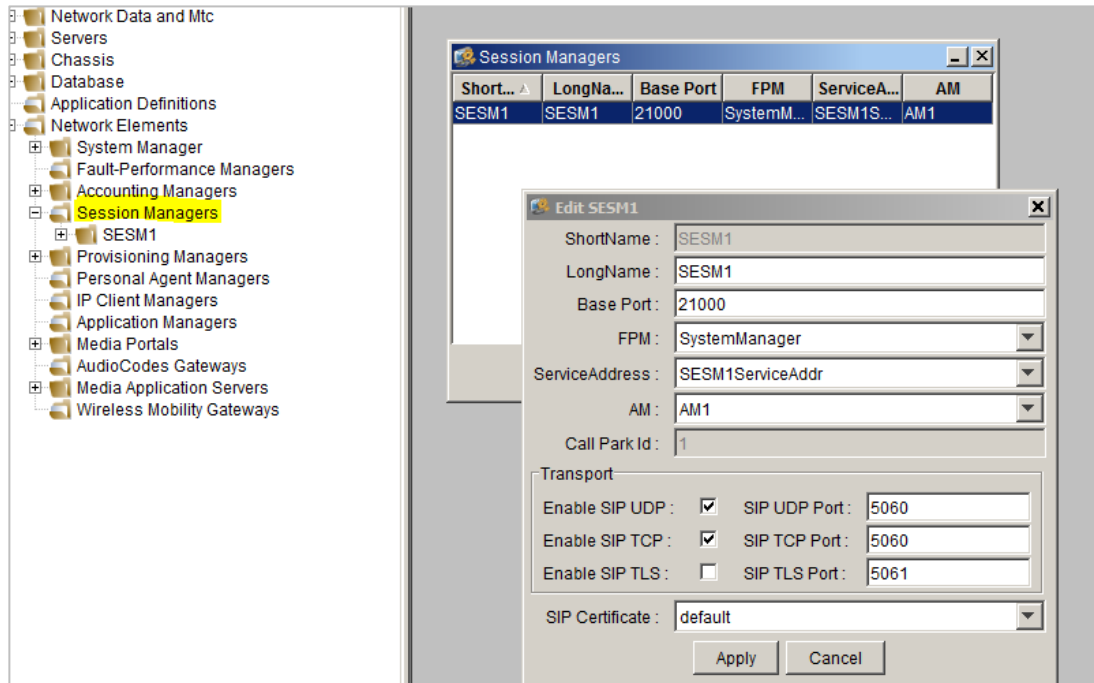
Session Managers

The Session Manager is the AS component which interacts with all external nodes using SIP.

To access the list of Session Managers:

- 1 Go to **System Management Console > Network Elements**.

- 2 Double-click on **Session Managers**.
- 3 In the **Session Managers** window, select a session manager you want to view or edit.
- 4 Verify the configuration as seen in the following figure.



For example:

The selected session manager is configured as follows:

The session manager is configured to utilize both **UDP** and **TCP**, and if necessary, to interact with its external servers.

The **ServiceAddress** references the logical address representing the IP address that floats across the two session manager physical nodes. This logic address is described in the Addresses section.

Provisioning Manager

The EXPERiUS AS Provisioning Manager provides a configuration portal into various data which is largely root domain dependent. That is, this data is configured under a specific root domain and applies only to those communications interactions originating or terminating to users provisioned under the root domain.

The EXPERiUS AS Provisioning Manager is a web-based interface accessible at:

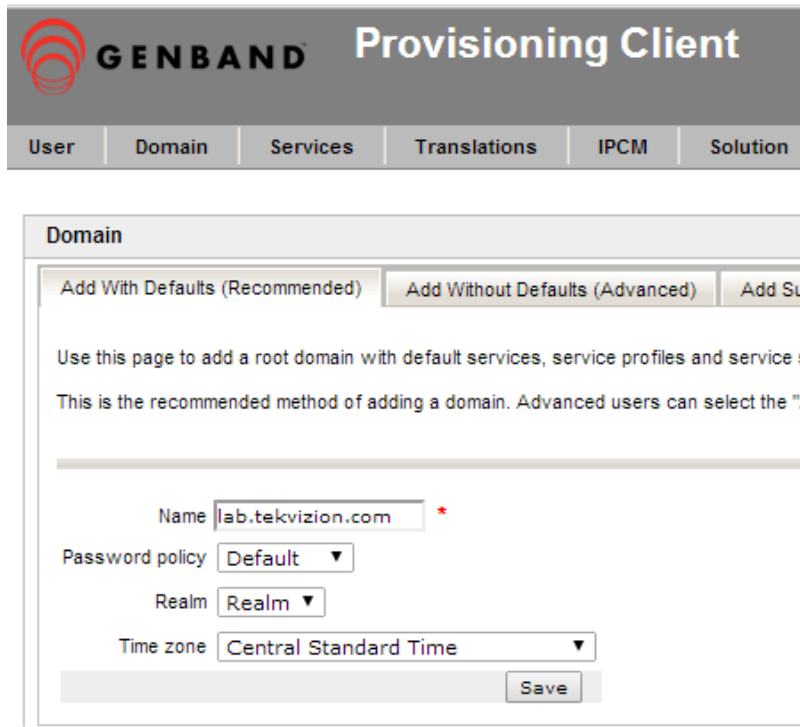
http://<prov blade IP address>:8443/prov

Root Domain

All of the EXPERiUS AS configuration in Provisioning Manager must be grouped under a domain; hence, determine the domain to which the EXPERiUS AS belongs and create it as a root domain.

To add root domain:

- 1 In Provisioning Manager, navigate to **Domain > Add With Defaults (Recommended)**.



The screenshot shows the GENBAND Provisioning Client interface. At the top, there is a navigation bar with tabs: User, Domain, Services, Translations, IPCM, and Solution. Below this, the 'Domain' configuration window is open. It has three tabs: 'Add With Defaults (Recommended)', 'Add Without Defaults (Advanced)', and 'Add Su'. The 'Add With Defaults (Recommended)' tab is selected. The text below the tabs reads: 'Use this page to add a root domain with default services, service profiles and service s' and 'This is the recommended method of adding a domain. Advanced users can select the "'. Below this text are four input fields: 'Name' with the value 'lab.tekvizion.com' and a red asterisk, 'Password policy' with a dropdown menu set to 'Default', 'Realm' with a dropdown menu set to 'Realm', and 'Time zone' with a dropdown menu set to 'Central Standard Time'. A 'Save' button is located at the bottom right of the form.

Figure 7. Provisioning Manager – Domain

Class of Service (COS)

A default Class of Service must be defined for reference in various provisioning activities.

To add/modify COS:

- 1 In Provisioning Manager, navigate to **Translations > Class of Service > Domain**.
The **Class of Service** window appears.

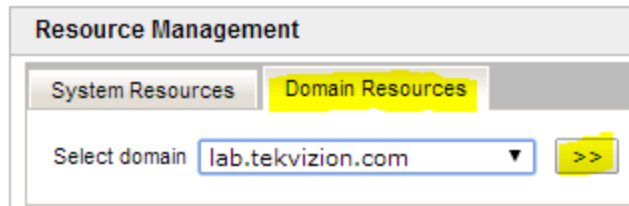
- 2 In the **Select domain** drop-down list, select the proper domain and click the >> icon.
- 3 In the **Class of service name** field, create a COS called *DefaultCOS*.
- 4 Click on the **Assign/Unassign** link in the COS summary table. The **Assign** tab appears.

- 5 In the **Available Domains** list, select a domain name, and then click the **Copy** button.
The selected domain is moved in the **Selected Domains** list.
- 6 Click **Save**.

Resources

To allocate resources to your root domain:

- 1 In Provisioning Manager, navigate to **Services > Domain Resources**.



- 2 Minimally, the following resources must be allocated:

- Call Forward Subscribers

- Subscribers

- Unified Messaging Subscribers

Service Nodes

Service Nodes are needed within the Provisioning Manager to represent the servers that were created and given addressing information in the System Manager. Service Nodes are required for:

- Mitel MiCollab AM

- Audiocodes (for PSTN access)

These Service Nodes will bridge the configuration gap from the Provisioning Manager across to the System Manager as the Service Node definition in Provisioning will reference a System Manager Informational Element.

Mitel MiCollab AM:

To configure the Mitel MiCollab AM Service Nodes:

- 1 In Provisioning Manager, navigate to **Translations > Service Node**.

- 2 In the **Node address** field, select **Address Name**.
- 3 In the **Address Name** drop-down list, select **Mitel_cx_e**, the Informational Element defined by the System Manager.
- 4 In the **Node type** drop-down list, select **Mitelsipprofile**.
This gives the operator control over SIP interworking behavior as defined in a SIP Profile.
- 5 Check the **Is trusted** box.
- 6 Click **Save**.

To add the environment's root domain to the domains supported by the new service node:

- 1 In the **Service Nodes** table, find the service node you are configuring and click the **Domains** hyperlink for that node.

Name	Type	Address	Domains	Delete
asteriskjosh	Asteriskjoshsipprofile	asteriskjosh	Domains	Delete
audiocodesgw	AudioCodes	AudioCodesGW	Domains	Delete
audiocodesgw2	AudioCodes	AudioCodes 1064296	Domains	Delete
aurasm63	AvayaAura	auraSM6dot3	Domains	Delete
avayasm61	ciscocmpbx	avayaSM6dot1	Domains	Delete
avst_node	avstsipprofile	avst_cx_e	Domains	Delete

The **Assign Domain** screen appears.

- 2 In the **Available domains** drop-down list, select the target root domain you want to add to the node, and then click **Add**.

The selected domain appears in the **Assigned domains** field.

- 3 Click **Save**.

Audiocodes Gateway

To configure the Audiocodes Gateway Service Node

- 1 In Provisioning Manager, navigate to **Translations > Service Node**.

- 2 In the **Node** address field, select **Address Name**.
- 3 In the **Address Name** drop-down list, select **AudioCodesGW**, the Informational Element defined by the System Manager.
- 4 In the **Node type** drop-down list, select **AudioCodes**.
This gives the operator control over SIP interworking behavior as defined in a SIP Profile.
- 5 Check the **Is trusted** box.

6 Click **Save**.

To add the environment's root domain to the domains supported by the new service node

- 1 In the **Service Nodes** table, find the service node you are configuring and click **Domains** hyperlink for that node.

Name	Type	Address	Domains	Delete
audiocodesgw	AudioCodes	AudioCodesGW	Domains	Delete
audiocodesgw2	AudioCodes	AudioCodes1064296	Domains	Delete
aurasm63	AvayaAura	auraSM6dot3	Domains	Delete
avayasm61	ciscocmpbx	avayaSM6dot1	Domains	Delete
cisco_gateway	CiscoSIPGateway	Cisco-Gateway	Domains	Delete
cs1k	cs1kpbx	cs1k-IE	Domains	Delete
cucmnode	ciscocmpbx	cisco-cucm	Domains	Delete
exchangeum	A2PC	exchangeUM	Domains	Delete
gbimnode	IMM	GenbandIMM	Domains	Delete
ipcalliance	ipcalliance	ipc-alliance	Domains	Delete
lynms61	rtcc	lynMS61	Domains	Delete
lynms63	lynMS	lynMS63	Domains	Delete

The **Assign Domain** screen appears.

- 2 In the **Available domains** drop-down list, select the target root domain you want to add to the node, and then click **Add**.

Service Node

Assign Domain

Assign domains to node - audiocodesgw

Available domains: lab.tekvizion.cor [Add]

Assigned domains: system, cibc.com, abcenterprise.cor, lab.tekvizion.com, experius-aura.coi, sippbx-qflex.com, lynclabsp.local [Remove]

[Save]

The selected domain appears in the **Assigned domains** field.

- 3 Click **Save**.

Mitel MiCollab AM Logical Entity

A Logical Entity is utilized in the EXperiUS AS/Mitel MiCollab AM integration to facilitate the sharing of a single root domain across the Mitel MiCollab AM and EXperiUS AS.

The Logical Entity allows for the custom specification of a domain name to be placed in an outgoing INVITE request-URI host field. This logical entity technique is needed to force the AS to specify a remote domain name which is identical to its own root domain.

To create the Mitel MiCollab AM Logical Entity:

- 1 In Provisioning Manager, navigate to **Translations > Service Node**.

Service Node

Modify Logical Entity

Modify - avst_le

Entity name

Available routable services

Selected routable services

Branding Announcement
Colorful RingBack Tones
Number Translation
Unified Conferencing
Chat
Meet Me Conferencing
Mobility Gateway
Unified Communications

>> Copy all
> Copy
< Remove
<< Remove All

Routable services

Selection algorithm

Add route(s) to logical entity

Name	Node	Parms	Weight (0-10)
<input type="text"/>	<input type="text" value="asteriskjosh"/>	<input type="text" value="Trunk Group"/> Add Remove all	<input type="text" value="0"/>

Save

avst_node avst_node;facilityDomain=lab.tekvizion.com 10.0 (100.0%) Remove

First
Up
Down
Last

Remove all

Save

- 2 In the **Entity name** field, type the name.
- 3 In **Routable services**, click **Copy all** to move all services from the **Selected routable services** list to the **Available routable services** list.
- 4 In **Add a route to the logical entity**, configure the following options:
 - a In the **Name** field, type the name.
 - b In the **Node** drop-down list, select a service node which provides connectivity to Mitel MiCollab AM.
 - c In the **Parms** drop-down list, select a facility domain, and then click the **Add** button.
 - d In the **Facility domain** field, type the desired domain name to be used for the INVITE requests sent to Mitel MiCollab AM.
 - e In the **Weight (0-10)** field, type 10 to represent all calls.

f Click the **Save** button to add the route entry to the logical entity.

5 Click the **Save** button at the end of the logical entity page.

To assign the location entity to the root domain:

1 In the **Logical Entities** table, find the new entity you just added, and then click the **Assign/Remove** hyperlink.

Name	Domains
asteriskjosh_le	Assign/Remove
aurasm63_le	Assign/Remove
avst_le	Assign/Remove

The **Assign Domain** screen appears.

2 In the **Available domains** drop-down list, select the domain you want to assign, and then click **Add**.

Service Node

Assign Domain

Assign/Remove domains to logical entity - avst_le

Available domains: a2.tekvizionlabs ▼ Add

Assigned domains: lab.tekvizion.com ▲ Remove

Save

3 Click **Save**.

Telephony Routes

Telephony Routes provide the ways the AS routes SIP messages. In this setup, Gateway Telephony Routes are utilized in the AS to route calls to Mitel MiCollab AM.

To add a Telephony Route:

1 In Provisioning Manager, navigate to **Translations > Telephony Routes**.

2 From the **Root domain** drop-down list, select a domain and click **Add Gateway Telephony Routes**.

3 Click the >> button.

8012Xto72XX

Create a Gateway Telephony Route to represent routing of 4-digit VM number to Mitel MiCollab AM. This route handles calls originated from the AS users.

Telephony Route

Gateway Telephony route

Name *

Description *

From digit *

To digit *

Min number of digit *

Max number of digit *

Remove *

Prefix

Recursive ☐

Gateway route type ▼

Gateway route ▼

Number qualifier ▼

Override name ☐

Override name value

Override charge value

Override charge ID ☐

Interop profile ▼

Off Net ☐

Figure 8. Provisioning Manager - Gateway Telephony Route for Mitel MiCollab AM

This route handles a single 4-digit VM number.

This route terminates to the AS logical entity as referenced in the **Gateway Route** field.

PSTN_to_Mitel

Create a Gateway Telephony Route to represent routing of 10-digit target numbers to Mitel from PSTN. This will facilitate routing of PSTN-originated calls to Mitel via AS.

Telephony Route

Gateway Telephony route

Name PSTN_to_AVST *

Description PSTN_to_AVST *

From digit 9725980120 *

To digit 9725980120 *

Min number of digit 10 *

Max number of digit 10 *

Remove 10 *

Prefix 5500

Recursive ☐

Gateway route type PUBLIC ▼

Gateway route avst_le ▼

Number qualifier --None Selected-- ▼

Override name ☐

Override name value

Override charge value

Override charge ID ☐

Interop profile None ▼

Off Net ☐

Figure 9. Provisioning Manager - Telephony Route Screen for PSTN to Mitel

This route handles a 10-digit number from PSTN to route to Mitel MiCollab AM.

Set **Prefix** to 5500. Set **Remove** to 10. This facilitates the number dialed from PSTN to hear announcement from Mitel MiCollab AM.

This route terminates to the AS logical entity as referenced in the **Gateway route** field.

Route List

EXPERiUS AS requires a Route List as an organizational envelope of one or more telephony routes targeting a single server. A dedicated Route List is created to represent the following server targets:

Mitel MiCollab AM

Audiocodes

To create a Route List:

- 1 In the Provisioning Manager, navigate to **Translations > Telephony Routes**.
- 2 Select the **Add New Route List** tab, and then click the >> button next to your domain name.

- 3 Define the Mitel MiCollab AM Route List as seen below.

Mitel_cx

Telephony Route

Modify Route list

Name *

Description

Incoming other domain tree call routing

Incoming same domain tree call routing

Class of service

Available Telephony route(s)

214toPSTN

Selected Telephony route(s)

AStoAVST-CX
PSTN_to_AVST

» Copy all
» Copy
« Remove
« Remove All

Save

Figure 10. Provisioning Manager - Mitel_cx Route

From the **Available Telephony route(s)** list, make sure to **Copy** the applicable telephony routes over to the **Select Telephony route(s)** list.

Save the entry.

AudiocodesGW

Modify Route list

Name: AudiocodesGW *

Description:

Incoming other domain tree call routing: ALLOW ▼

Incoming same domain tree call routing: ALLOW ▼

Class of service: Default ▼

Available Telephony route(s)

- IMM
- 972265toAvayaaura
- 972toPSTN
- 719377922xtoCS1000
- 8012Xto72XX
- 7922xtoCS1000
- 80122

Buttons: Copy all, Copy, Remove, Remove All

Selected Telephony route(s)

- 214311toPSTN
- 214toPSTN
- 866-888
- 518toPSTN
- 800toPSTN

Save

Figure 11. Provisioning Manager - AudiocodesGW Route

From the **Available Telephony route(s)** list, make sure to **Copy** the applicable telephony routes over the **Selected Telephony route(s)** list.

Save the entry.

Users

The EXPERiUS AS users are provisioned to represent A (SIP), B(SIP), and C (SIP) in the test environment.

Managing Resources

- 1 In Provisioning Manager, navigate to **Services > Resources**.
- 2 Select **Domain Resources** for the target domain.
- 3 In the **Subscribers** row and the **Assign** column from the table, type the number of subscriber(s) you want to activate for allocation.
- 4 Click the **Save** button.

Adding Users

Add a user for both phones that are registering directly with EXPERiUS AS.

NOTE The User records are protocol-independent.

- 1 In Provisioning Manager, navigate to **User > Add**.

User

Search Advanced Search Add Add User With Defaults Domain Defaults

Select domain lab.tekvizion.com ▼

Type Standalone ▼ >>

- 2 From the **Type** drop-down list, select **Standalone**, and then click the >> button.
- 3 In the **Add User with Defaults** tab, provide the user details as illustrated in the figure below.

User

Search Advanced Search Add Add User With Defaults Domain Defaults

Select domain lab.tekvizion.com ▼

Type Standalone ▼ >>

User name as_80122 *

Password ***** *

Confirm password ***** *

Service set --None Selected-- ▼

Status reason ACTIVE ▼

Location Other ▼

First name as_80122 * ?

Last name Phone * ?

E-mail

Business phone

Home phone

Cell phone

Pager

Fax

Time zone Eastern Standard Time ▼

Locale English ▼

Home language --None Selected-- ▼ ?

Home country --None Selected-- ▼ ?

Class of Service

Class of service Default ▼

Redirection class of service Default ▼

- 4 Select the **Extended data** tab to assign a telephone number to the user.

User

User Details for: as_80122@lab.tekvizion.com

Base data Password Services Extended data Actions Links

Aliases

Directory numbers 9725980122 80122

Save

- 5 In the **Directory numbers** field, enter a 10-digit phone number and a 5-digit extension.
- 6 Click **Save**.
- 7 Select the **Links** tab.

User Details for: as_80122@lab.tekvizion.com

Base data Password Services Extended data Actions Links

System Service Links
Application Manager
Class of Service
Core Services Session Manager
IP Restriction
Miscellaneous Groups
SIP Line
Service Capabilities Interaction Manager
User Routing

- 8 Click **Core Services Session Manager**.
- 9 In the **Core Services Session Manager Service** screen, select the **User Data** tab.

Core Services Session Manager Service

System Profile User Data

Select user (user@domain) as_80120@lab.tekvizion.cc >>

User's system profile

sesm1 Save Delete

- 10 In the **User's system profile** drop-down list, select the AS's active Session Manager, *sesm1*.
- 11 Click **Save**.
- 12 Return to the **Links** tab, and then click **User Routing**.
- 13 In the **User Routing Service** screen, select the **Charge ID** tab.

User Routing Service

Static Routes | **Charge ID** | User Identification

Select user (user@domain) >>

Private charge ID

Public charge ID

- 14 Enter the values in the **Private charge ID** and **Public charge ID** fields.

The 10-digit **Public charge ID** is provisioned so that it calls to the PSTN/Mitel MiCollab AM include a 10 digit caller ID to satisfy the SIP Trunk carrier requirements.

- 15 Select the **User Identification** tab and ensure the **Public DN** value is set to 9725980122 and the **Private DN** value is set to 80122.

User Routing Service

Static Routes | Charge ID | **User Identification**

Select user (user@domain) >>

Public DN ?

Public Number Qualifier ?

Private DN ?

Private Number Qualifier ?

- 16 Add a similar User record as needed for additional accounts using a unique username and directory number.

EXPERiUS AS Feature

Setting up Call Forwarding:

- 1 In Provisioning Manager, navigate to **Services > Call Routing > Call Forward Variants**.
- 2 In the **Profile name** field from **System Profile** tab, create a profile with the type of CF desired.

Call Forward Variants Service

System Profile

Modify - cf-greg

Profile name

Call forward variants busy ☒

Call forward variants no answer ☒

Call forward immediate ☒

Call forward unreachable ☐

Call forward not logged in ☐

Profile name	Copy	Delete	Domain
_system_default_	Copy		Assign/Remove
afu	Copy	Delete	Assign/Remove
cf	Copy	Delete	Assign/Remove
cf-greg	Copy	Delete	Assign/Remove

- 3 In the **Profile name** column, find the profile you created, and then click the **Assign/Remove** hyperlink.
- 4 In the **Profile – <Profile Name>** page, copy the system profile you created to your root.

Call Forward Variants Service

System Profile **User Data**

Profile - cf-greg

Available Domains

- abcenterprise.com
- avaya-aura-qflex-tv.com
- clus23pubsub.lab.tekvizi
- ed-test.com
- experius-qflex.com
- imttekvision.com
- sippbx-qflex.com
- sinnbx.com

Selected Domains

- a2.tekvizionlabs.com
- cibc.com
- experius-aura.com
- experius-cucm.com
- lab.tekvizion.com
- lynclabsp.local

- 5 Go to your **User** account.

User

User Details for: as_80122@lab.tekvizion.com

Base data Password **Services** Extended data Actions Links

► User service(s) assigned successfully. You need to assign specific service profile(s)

Assign all services ☐

Show Profile

<input type="checkbox"/>	Assistant Support
<input checked="" type="checkbox"/>	Call Forward Variants
<input type="checkbox"/>	Call Grabber
<input type="checkbox"/>	Call Park
<input checked="" type="checkbox"/>	Call Pickup
<input type="checkbox"/>	Call Return
<input type="checkbox"/>	Call Type Based Screening

Save

- 6 In the **Service name** column, make sure the **Call Forward Variants** box is checked either directly or as part of a Service Set.
- 7 Click the **Call Forward Variants** hyperlink.
- 8 From the **User's system profile** drop-down list in the **User Data** tab, select the CF profile created in Step 2.

Call Forward Variants Service


System Profile User Data

Select user (user@domain) as_80122@lab.tekvizion.cc >>

User's system profile

cf-greg Save Delete

- 9 Go to the user's **Personal Agent**.

 **GENBAND** Personal Agent Home

Welcome as_80122@lab.tekvizion.com

Routes

- Call Routes
- Call Forwarding Immediate
- Call Forward Variants**

Call Forward Variants

Forward your calls to a location or number based upon your availability.

The call forward variants option is: ☒ On ☐ Off

Under the following conditions forward my calls:

Number of Rings: ▼

If busy, route to
 ▼

If no answer, route to
 ▼

If unreachable, route to
 ▼

If not logged in, route to
 ▼

10 Select **Call Forwarding Variants**.

11 In the **Call Forward Variants** screen, assign a CF target as shown in the previous figure.

12 Click **OK**.

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:

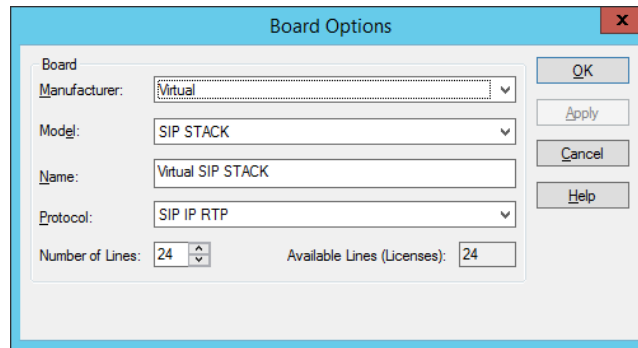
- [Configuring MiCollab AM for the Integration During Initial Installation](#): Integrate the telephone system while you install MiCollab AM for the first time.
- [Configuring Existing MiCollab AM for the Integration](#): Integrate a new telephone system on your existing MiCollab AM system.

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 MiCollab AM 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 **Genband**.
 - d From the **Model** drop-down list, select **EXPERiUS Application Server**.
 - e From the **Integration Type** drop-down list, select **SIP Trunk**.
- 2 Click **Next**. The **Board Options** dialog box appears.



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

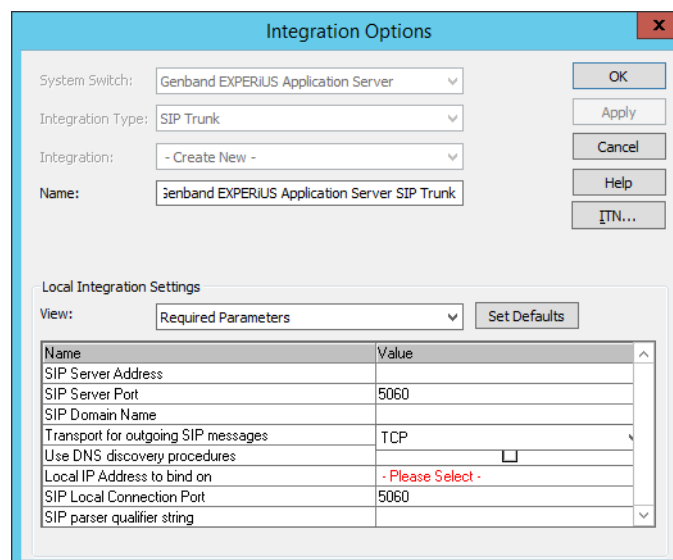
- Manufacturer:** A drop-down menu with "Virtual" selected.
- Model:** A drop-down menu with "SIP STACK" selected.
- Name:** A text field containing "Virtual SIP STACK".
- Protocol:** A drop-down menu with "SIP IP RTP" selected.
- Number of Lines:** A numeric field set to 24.
- Available Lines (Licenses):** A numeric field set to 24.
- Buttons: OK, Apply, Cancel, and Help.

- 3 In the **Board Options** dialog box, configure the following options:
 - a From the **Manufacturer** drop-down list, select **Virtual**.
 - b From the **Model** drop-down list, select **SIP STACK**.
 - c In the **Name** field, the name for this board is automatically generated. Enter a new name if necessary.
 - d From the **Protocol** drop-down list, select **SIP IP RTP**.
 - e In the **Number of Lines** field, enter the number of lines this board uses. The total number of lines is limited by the capacity of the board and the number of **Available Line Licenses**.
- 4 Click **OK**. The **Switch Options** dialog box appears.
- 5 If necessary, make any changes to the default settings your site requires in the **Switch Options** dialog box.

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

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

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



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

- System Switch:** A drop-down menu with "Genband EXPERIUS Application Server" selected.
- Integration Type:** A drop-down menu with "SIP Trunk" selected.
- Integration:** A drop-down menu with "- Create New -" selected.
- Name:** A text field containing "Genband EXPERIUS Application Server SIP Trunk".
- Buttons: OK, Apply, Cancel, Help, and ITN...
- Local Integration Settings** section:
 - View:** A drop-down menu with "Required Parameters" selected.
 - Set Defaults** button.
 - Table:**

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 options as follows:

- a In the **Local Integration Settings** section, select the **Required Parameters** view, and configure the following parameters.

Table 5. Required Parameters View – Integration Options

Field	Required Value
SIP Server Address	Enter the IP address of the PBX.
SIP Domain Name	Enter your LAN or VLAN domain (the domain in which both the MiCollab AM Call Server and EXPERiUS AS reside).
Local IP Address to bind on	Select the IP address of the NIC on the MiCollab AM platform that should support the integration. NOTE If the MiCollab AM platform only has one NIC, leave the Local IP Address to bind on field blank.
SIP parser qualifier string	<ul style="list-style-type: none">• 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.• Multiple SIP integrations on the call server: Use a string that is unique to each SIP integration. For example:<ul style="list-style-type: none">• 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.

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

- In the **SIP Domain Name** field, enter your LAN or VLAN domain (the domain in which both the MiCollab AM Call Server and EXPERiUS AS reside).
- Set the **Type of Call Progress to use for External Calls** value. How this should be set depends on the gateway used for the integration as follows:
 - **Digital:** Select Digital if the gateway supports call progress through to the endpoint.
 - **Media:** Select Media if the gateway reports early that the call is connected, such as before the phone rings or while the phone is ringing.

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

- 9 In the **Switch Section Options** dialog box, configure the following options:
 - a In the **Local Integration Settings** section, select the **Required Parameters** view.
 - b In the **Incoming Hunt Mode** field, enter the mode for this integration.
 - c In the **Hunt Group Access Code** field, enter the pilot number you configured previously. This is the internal number subscribers dial to reach MiCollab AM.
 - d Click **OK**.
- 10 Continue through and complete the configuration. At the end of the configuration, a confirmation dialog box appears. Click **OK**.
- 11 If **MiCollab AM Configuration** does not open automatically after the configuration completes, open **MiCollab AM Configuration**, and select the **Lines** tab.
- 12 In the table from the **Lines** tab, configure callouts for the application. For information on configuring callout settings, see the topic *Configuring Callout Settings*, in the online help system.
- 13 Click **OK** to save all changes.

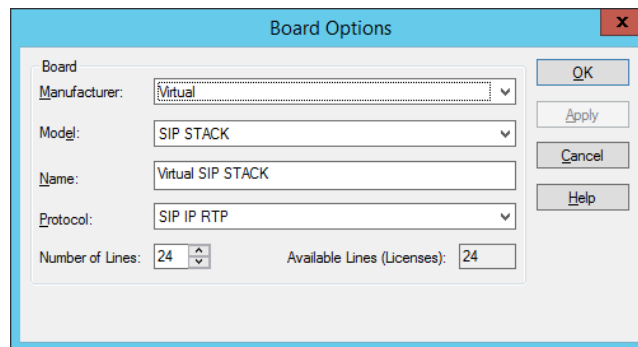
Configuring Existing MiCollab AM for the Integration

To configure exiting MiCollab AM for the telephone integration:

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

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

- 3 **[Optional]** Select the **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**.
- 4 Select the **Switch** tab and click the **Add** button. The **Switch Integration Data Setup** dialog box appears.
- a From the **Manufacturer** drop-down list, select **Genband**.
 - b From the **Model** drop-down list, select **EXPERiUS Application Server**.
 - c From the **Integration Type** drop-down list, select **SIP Trunk**.
- 5 Click **OK**. The **Switch Options** dialog box appears.
- 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 and Configuration Guide*.

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

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

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

Table 6. Required Parameters View – Integration Options

Field	Required Value
SIP Server Address	Enter the IP address of the PBX.
SIP Domain Name	Enter your LAN or VLAN domain (the domain in which both the MiCollab AM Call Server and EXPERiUS AS reside).
Transport for outgoing SIP messages	Enter TCP or UDP .
Local IP Address to bind on	Select the IP address of the NIC on the MiCollab AM platform that should support the integration. NOTE If the MiCollab AM platform only has one NIC, leave the Local IP Address to bind on field blank.
SIP parser qualifier string	<ul style="list-style-type: none"> • 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. • Multiple SIP integrations on the call server: Use a string that is unique to each SIP integration. For example: <ul style="list-style-type: none"> • 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. <p>NOTE This setting must match a string in the SIP header that is unique to this particular integration.</p>

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

- In the **SIP Domain Name** field, enter your LAN or VLAN domain (the domain in which both the MiCollab AM Call Server and EXPERiUS AS reside).
- Set the **Type of Call Progress to use for External Calls** value. How this should be set depends on the gateway used for the integration as follows:
 - **Digital:** Select Digital if the gateway supports call progress through to the endpoint.
 - **Media:** Select Media if the gateway reports early that the call is connected, such as before the phone rings or while the phone is ringing.

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

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

- a** In the **Local Integration Settings** section, select the **Required Parameters** view.
 - b** In the **Incoming Hunt Mode** field, enter the mode for this integration.
 - c** In the **Hunt Group Access Code** field, enter the pilot number you configured previously. This is the internal number subscribers dial to reach MiCollab AM.
 - d** Click **OK**.
- 11** In **MiCollab AM Configuration**, verify that the telephone system is properly added and configured in the **Switches**, **Switch Sections**, and **Integrations** tabs.
 - 12** Select the **Lines** tab.
 - 13** In the table from the **Lines** tab, configure callouts for the application. For information on configuring callout settings, see the topic *Configuring Callout Settings*, in the online help system.
 - 14** Click **OK** to save all changes.

Configuring MiCollab AM for SIP Failover

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

To add a SIP failover server:

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

Table 7. 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**.

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

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

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

Table 8. 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.

9 Click **Apply** to save the changes.

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

11 Click **OK** to close the **Integration Options** dialog box.

To remove a SIP Failover Server:

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

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

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

Changing the Network Binding Order on the MiCollab AM Platform

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

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

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

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

Windows Server 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 9. 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