

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

For version 6.1 and above

Notice

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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	29
Service Nodes	29
Mitel MiCollab AM Logical Entity	33
Telephony Routes	35
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	50
Changing the Network Binding Order on the MiCollab AM Platform	53
Windows Server 2008 R2 with Service Pack 1	53
Windows Server 2012 R2	54
Configuring Quality of Service (QoS)	55

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 documentation set for this MiCollab AM includes the following documents and resources:

- **Developer Resources.** Contains programming guides and API references for developers for integrating the server clients and web applications with MiCollab AM.
- **Integration Technical Notes (ITN).** Contains a set of guides that describe the integration methods and instructions for a variety of phone systems to work with MiCollab AM. The ITNs are generally used by resellers or administrators who are experienced with MiCollab AM and familiar with the integration procedures and terminology.
- **Quick Reference Card (QRC).** Contains shortcuts and quick instructions telling subscribers how to access and use the messaging system.
- **Server Documentation.** Available as a PDF only. Contains administrative guides for administrators about installing, configuring, and administering the messaging system, and user guides for subscribers about accessing the messaging system and checking and sending messages.
- **Spare Parts Documentation.** Contains a set of guides that describe the instructions for installing and configuring hardware parts to work with MiCollab AM. These documents are written for Mitel certified MiCollab AM technicians who are experienced with MiCollab AM and familiar with the procedures and terminology.
- **Software Release Notice (SRN).** This notice introduces the new features, capabilities, and hardware/software requirements for the corresponding MiCollab AM version.

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

Table 1. References

Document Type	Document Title
Server Documentation	SIP Routing Manger Installation Guide
Server Documentation	System Administration Guide
Server Documentation	System Installation Guide

Documentation Updates

Documentation updates may be available from the following sources:

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

Help

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

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

Document Conventions

The following conventions are used in this document:

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

Example: **Enter**

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

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

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

Example: Refer to *System Installation Guide*.

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

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

- **User Input.** Information required to be typed is shown in italics.
| **Example:** Type the password *voicemail*.
- **Warning, Caution, Important, and Notes.** Text for the contents that require attention are shown as follows:

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

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

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

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

Features Supported by This Integration

The following tables list the features supported using the 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.
- The primary (public) network interface card (NIC) must be the first network connection in the network binding order. 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. MiCollab AM binds and communicates to other servers and subscribers on this network connection. For more information, refer to the section, [Changing the Network Binding Order on the MiCollab AM Platform](#) section.
- MiCollab AM supports G.729a with support for annex b on the incoming audio stream only. MiCollab AM does not transmit annex b packets.
- When codec negotiation takes place between MiCollab AM and the PBX, MiCollab AM always offers the G.729a audio format as an option. You may configure G.729a as the preferred codec in MiCollab AM; however, the decision whether to use G.729a is always made by the PBX.
- The Call Queuing feature does not transcend the Call Server. Calls may be queued on multiple Call Servers for the same subscriber but Call Servers do not have knowledge of calls in the queue on other Call Servers within the system. Callers may be prompted with specific information about their place in the queue; however, the information pertains to the specific Call Server on which their call is queued.
- The MiCollab AM Integration Options parameter, Validate Remote Hosts for Media validates each incoming audio packet and accepts it only if it is sent from a valid endpoint. The parameter is disabled by default. Enabling this parameter causes MiCollab AM to reject RTP packets from invalid endpoints, rejects MWI packets that timeout after a specified number of times, and overcomes port lockups when callers hang up while MiCollab AM is performing a blind transfer. See the Mitel Technical Bulletin #42285 for more information on this parameter.

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

- MiCollab AM 6.1 supports up to 10 integration types (i.e. licensed integrations) in total per system. However, the following limitations apply to each Call Server:
 - Limited to 3 integration types per Call Server
 - The 3 integration types can be any mix of TDM and SIP (e.g. 1 TDM and 2 SIP)

- Limited to 1 Mitel MiTAI or 1 Cisco UCM SCCP IP integration. Can be mixed with TDM, but not with SIP
- Connect up to 10 telephone systems total per Call Server (e.g. 2 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

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

System Overview

Network Diagram

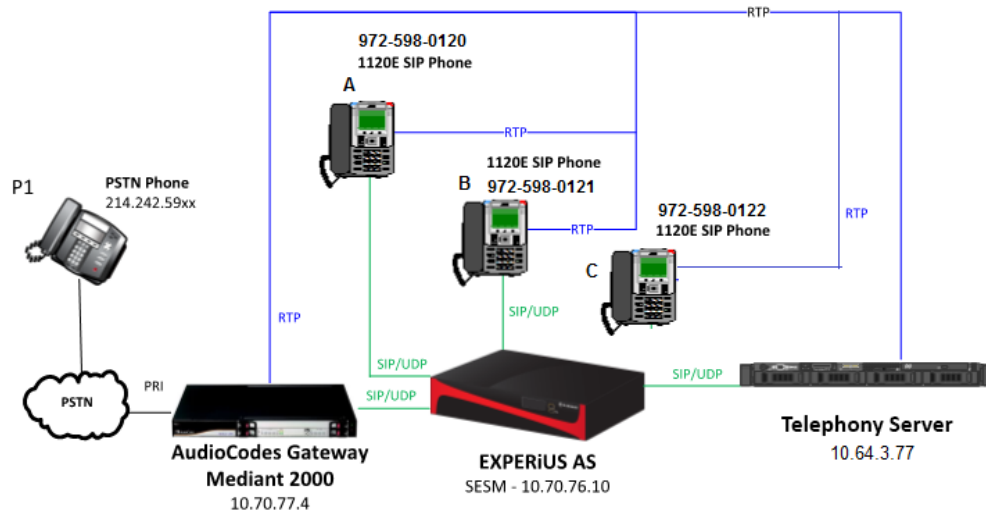


Figure 1. Network Diagram

Integration Overview

Mitel MiCollab AM 6.1 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

Submit

Figure 2. SIP General Parameters

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

Routing Calls

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**.

	Src. Trunk Group ID	Dest. Phone Prefix	Source Phone Prefix	Dest. IP Address	Port	Transport Type	Dest. IP Port
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	

Figure 3. Tel to IP Routing

- 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.

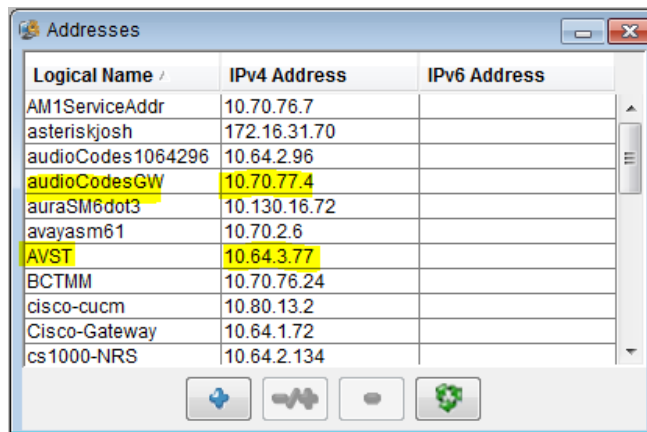
Network Data and Mtc

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



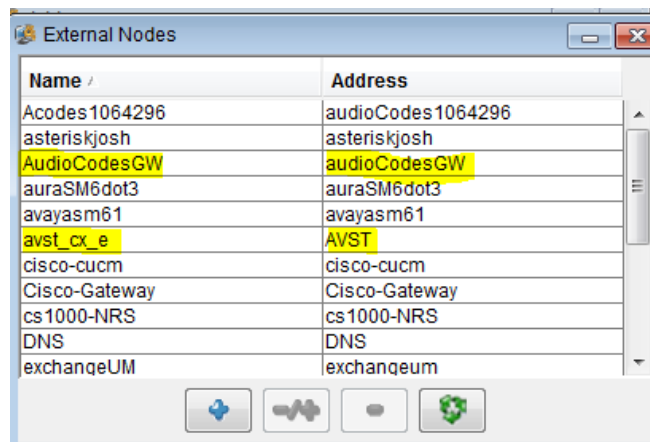
Logical Name	IPv4 Address	IPv6 Address
AM1ServiceAddr	10.70.76.7	
asteriskjosh	172.16.31.70	
audioCodes1064296	10.64.2.96	
audioCodesGW	10.70.77.4	
auraSM6dot3	10.130.16.72	
avayasm61	10.70.2.6	
AVST	10.64.3.77	
BCTMM	10.70.76.24	
cisco-cucm	10.80.13.2	
Cisco-Gateway	10.64.1.72	
cs1000-NRS	10.64.2.134	

Figure 4. 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 5. 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 5. 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 'LongName' field is 'avst_cx_e'. The 'Trusted' checkbox is checked. The 'Identification Port' is '0'. The 'Type' is 'Gateway'. The 'CTI Support' link is 'Configure CTI Support'. Under 'Transport Information', the 'Node' is 'avst_cx_e'. The 'Enable SIP UDP Port' checkbox is checked, and the 'SIP UDP Port' is '5060'. The 'Enable SIP TCP Port' and 'Enable SIP TLS Port' checkboxes are unchecked, with their respective ports set to '5060' and '5061'.

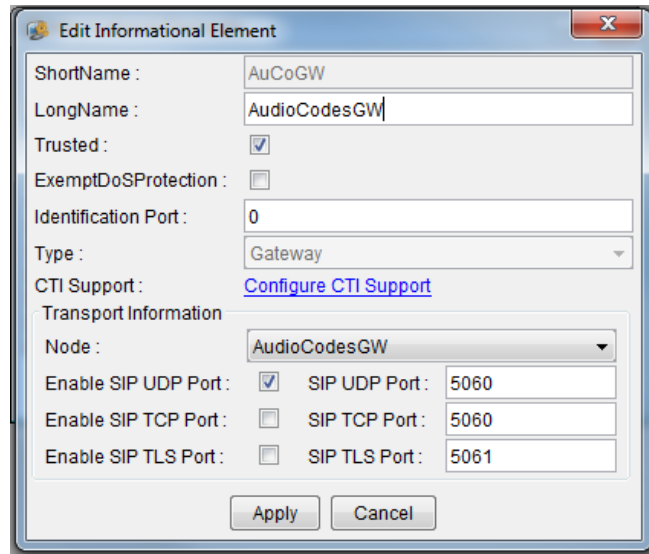
Figure 6. System Management Console - Edit Information Element Screen

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 7. 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.

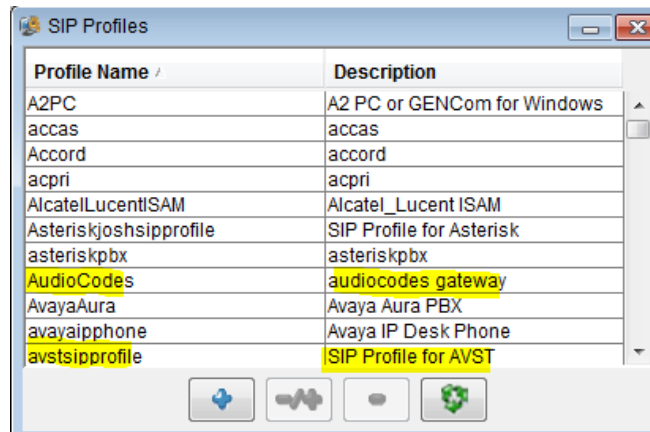


Figure 8. System Management Console - SIP Profiles Screen

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 :	<input checked="" type="checkbox"/>
Use Calling Party as From :	<input checked="" type="checkbox"/>
Use Options :	<input checked="" 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 type="checkbox"/>
IN Session Authentication :	<input type="checkbox"/>
MCD Update Call Model :	<input type="checkbox"/>
Use From Header For Subr Lookup :	<input type="checkbox"/>
Add Diversion Header :	<input checked="" type="checkbox"/>
Use Request URI As TO :	<input type="checkbox"/>
Remove Unknown Paid :	<input 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 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 type="checkbox"/>
Remove Phone Context :	<input type="checkbox"/>
PIDF-LO :	<input type="checkbox"/>
Use DN For Paid :	<input 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"/>

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**.

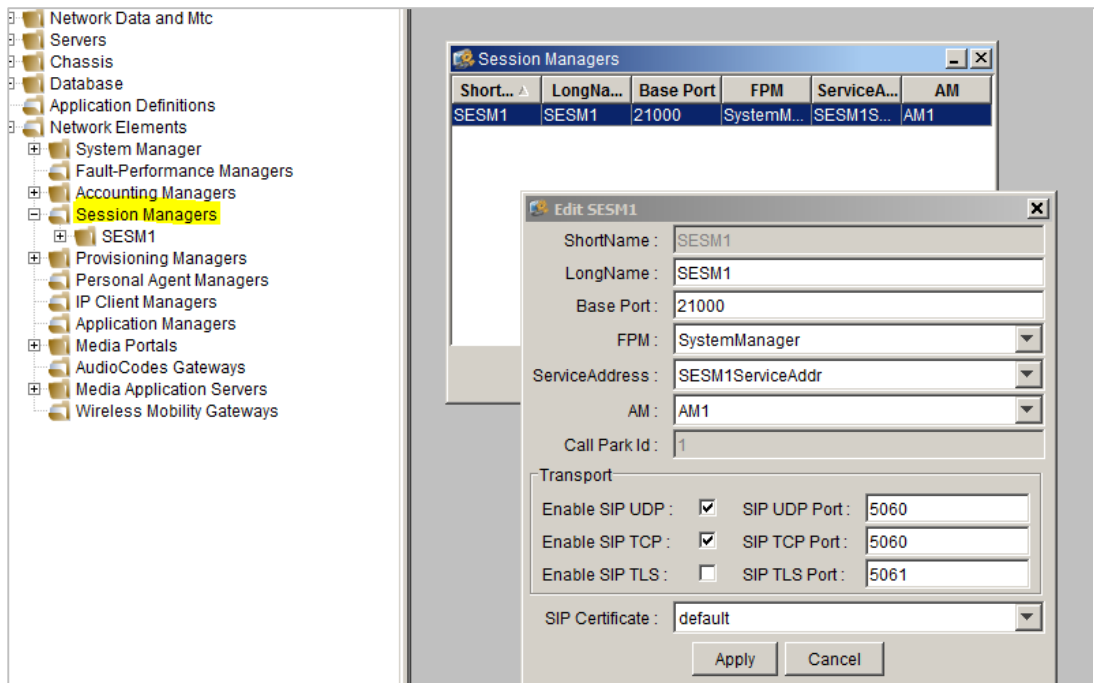


Figure 9. System Management Console - Session Manager

- 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 [Figure 9. System Management Console - Session Manager](#).

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:

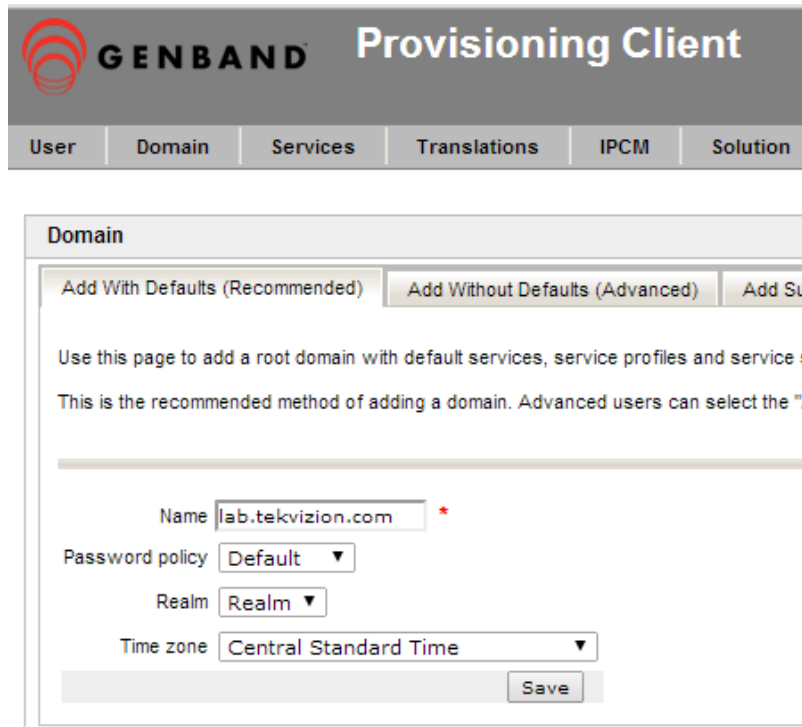
http://<prov blade IP address>:8443/prov

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' section is active. It contains three tabs: 'Add With Defaults (Recommended)', 'Add Without Defaults (Advanced)', and 'Add Su'. The 'Add With Defaults (Recommended)' tab is selected. Below the tabs, there is a text block: '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, there is a form with the following fields: 'Name' (text input with 'lab.tekvizion.com' and a red asterisk), 'Password policy' (dropdown menu with 'Default' selected), 'Realm' (dropdown menu with 'Realm' selected), and 'Time zone' (dropdown menu with 'Central Standard Time' selected). At the bottom right of the form is a 'Save' button.

Figure 10. 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 displays.

Figure 11. Provisioning Manager - Class of Service Screen

- 2 In the **Select domain** dropdown 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 displays.

Figure 12. Provisioning Manager - Assign Screen

- 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**.

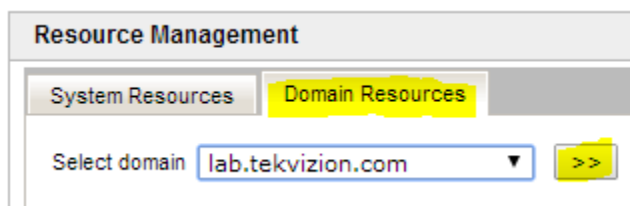


Figure 13. Provisioning Manager – Resources

- 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.

Figure 14. Provisioning Manager - Mitel_CX_E Service Node Screen

- 2 In the Node address field, select Address Name.
- 3 In the **Address Name** dropdown list, select Mitel_cx_e, the Informational Element defined by the System Manager.
- 4 In the **Node type** dropdown 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 Domains hyperlink for that node.

Name	Type	Address	Domains	Delete
asteriskjosh	Asteriskjoshsipprofile	asteriskjosh	Domains	Delete
audiocodesgw	AudioCodes	AudioCodesGW	Domains	Delete
audiocodesgw2	AudioCodes	AudioCodes1064296	Domains	Delete
aurasm63	AvayaAura	auraSM6dot3	Domains	Delete
avayasm61	ciscocmpbx	avayaSM6dot1	Domains	Delete
avst_node	avstsipprofile	avst_cx_e	Domains	Delete

Figure 15. Provisioning Manager - Service Nodes Table for Mitel MiCollab AM

The **Assign Domain** screen displays.

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

Figure 16. Provisioning Manager - Assign Domain Screen for Mitel MiCollab AM

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**.

Figure 17. Provisioning Manager - AudioCodesGW Service Node Screen

- 2 In the **Node** address field, select **Address Name**.
- 3 In the **Address Name** dropdown list, select **AudioCodesGW**, the Informational Element defined by the System Manager.

- 4 In the **Node type** dropdown 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	rtc	lynMS61	Domains	Delete
lynms63	lynMS	lynMS63	Domains	Delete

Figure 18. Provisioning Manager - Service Nodes Table for AudioCodesGW

The **Assign Domain** screen displays.

- 2 In the **Available domains** dropdown 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: lab.tekvizion.cor [Remove]

system
cibc.com
abcenterprise.cor
experius-aura.coi
sippbx-qflex.com
lynclabsp.local

[Save]

Figure 19. Provisioning Manager - Assign Domain Screen for AudioCodesGW

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

Routable services

Selected routable services

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

Selection algorithm

Add route(s) to logical entity

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

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

Figure 20. Provisioning Manager - Logical Entity Screen for Mitel MiCollab AM

- 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** dropdown list, select a service node which provides connectivity to Mitel MiCollab AM.
 - c In the Parms dropdown 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

Figure 21. Provisioning Manager - Logical Entities Table

The Assign Domain screen appears.

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

Figure 22. Provisioning Manager - Assign Node Screen

- 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** dropdown 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.

The screenshot shows the 'Telephony Route' configuration form in Provisioning Manager. The form is titled 'Telephony Route' and has a sub-tab 'Gateway Telephony route'. The fields are as follows:

- Name: AStoAVST-CX *
- Description: AStoAVST-CX *
- From digit: 5500 *
- To digit: 5500 *
- Min number of digit: 4 *
- Max number of digit: 10 *
- Remove: 0 *
- Prefix: (empty)
- Recursive: ☐
- Gateway route type: PUBLIC ▼
- Gateway route: avst_le ▼
- Number qualifier: --None Selected-- ▼
- Override name: ☐
- Override name value: (empty)
- Override charge value: (empty)
- Override charge ID: ☐
- Interop profile: None ▼
- Off Net: ☐

Figure 23. 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.

The screenshot shows the 'Telephony Route' configuration interface. The 'Gateway Telephony route' tab is selected. The configuration fields are as follows:

Field	Value	Required
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	<input type="checkbox"/>	
Gateway route type	PUBLIC	
Gateway route	avst_le	
Number qualifier	--None Selected--	
Override name	<input type="checkbox"/>	
Override name value		
Override charge value		
Override charge ID	<input type="checkbox"/>	
Interop profile	None	
Off Net	<input type="checkbox"/>	

Figure 24. 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: Avst_cx *

Description: Avst_cx

Incoming other domain tree call routing: ALLOW

Incoming same domain tree call routing: ALLOW

Class of service: Default

Available Telephony route(s)

- 214toPSTN

Selected Telephony route(s)

- AStoAVST-CX
- PSTN_to_AVST

Buttons: Copy all, Copy, Remove, Remove All

Save

Figure 25. 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

The screenshot shows the 'Modify Route list' window in the Provisioning Manager. The 'Name' field is set to 'AudiocodesGW'. The 'Description' field is empty. The 'Incoming other domain tree call routing' and 'Incoming same domain tree call routing' are both set to 'ALLOW'. The 'Class of service' is set to 'Default'. The 'Available Telephony route(s)' list contains: IMM, 972265toAvayaaura, 972toPSTN, 719377922xtoCS1000, 8012Xto72XX, 7922xtoCS1000, and 80122. The 'Selected Telephony route(s)' list contains: 214311toPSTN, 214toPSTN, 866-888, 518toPSTN, and 800toPSTN. Buttons for 'Copy all', 'Copy', 'Remove', and 'Remove All' are located between the two lists. A 'Save' button is at the bottom left.

Figure 26. 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 ▼ >>

Figure 27. Provisioning Manager - Add Screen for User

- 2 From the **Type** dropdown 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 ▼

Figure 28. Provisioning Manager - Add User With Defaults Screen

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

User Details for: as_80122@lab.tekvizion.com

Base data Password Services Extended data Actions Links

Aliases

Directory numbers 9725980122 80122

Save

Figure 29. Provisioning Manager - Extended Data Tab

- 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

Figure 30. Provisioning Manager –

- 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

Figure 31. Provisioning Manager - User Data Tab

- 10 In the **User's system profile** dropdown 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.

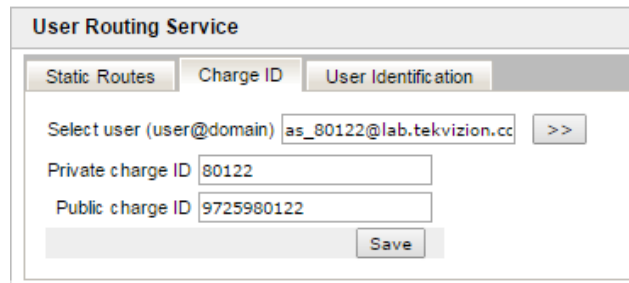


Figure 32. Provisioning Manager - Charge ID Tab

- 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.

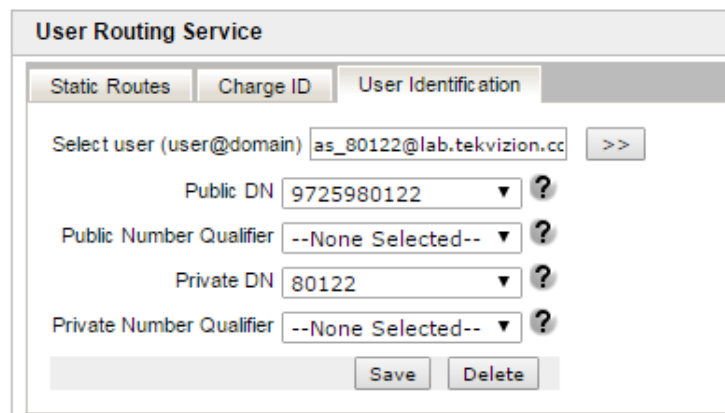


Figure 33. Provisioning Manager - User Identification Tab

- 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

cf-greg

Call forward variants busy

☒

Call forward variants no answer

☒

Call forward immediate

☒

Call forward unreachable

☐

Call forward not logged in

☐

Save

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

Figure 34. Provisioning Manager - System Profile Screen

- In the **Profile name** column, find the profile you created, and then click the **Assign/Remove** hyperlink.
- 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

Copy all

Copy

Remove

Remove All

Save

Figure 35. Provisioning Manager - Profile Domains Screen

- 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

Figure 36. Provisioning Manager - User Services Screen

- 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** dropdown 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

Figure 37. Provisioning Manager - User Data Screen

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

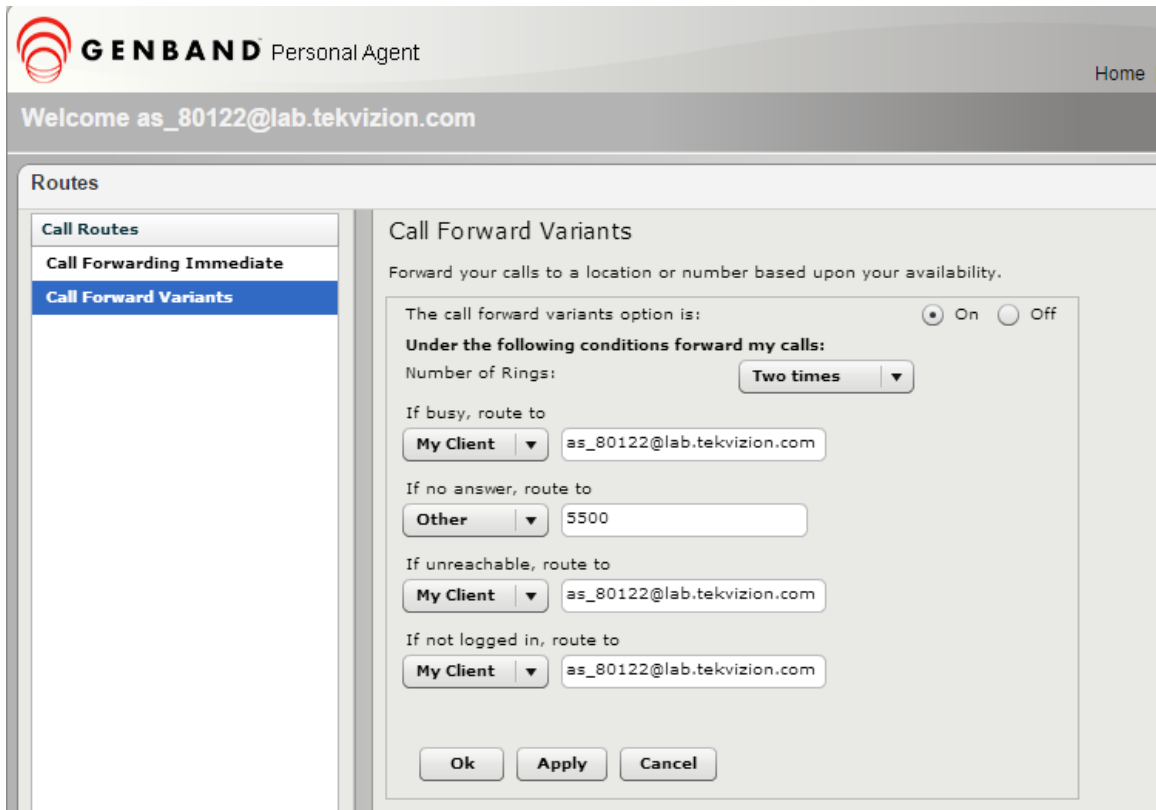


Figure 38. Personal Agent - Call Forward Variants

- 10 Select **Call Forwarding Variants**.
- 11 In the **Call Forward Variants** screen, assign a CF target as shown in Figure 38. Personal Agent - Call Forward Variants.
- 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 *System Installation Guide*, and the topic, **Integrate the Telephony Server with the Telephone System**, in the online help.

Configuring MiCollab AM for the Integration During Initial Installation

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

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

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

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

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

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

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

Table 5. Required Parameter Settings for 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

- **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:

- 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 displays.
- 9** In the **Switch Section Options** dialog box, configure the following options:
- a** In the **Local Integration Settings** section, select the **Required Parameters** view.
 - b** In the **Incoming Hunt Mode** field, enter the mode for this integration.
 - 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 displays. Click **OK**.
- 11** If **MiCollab AM Configuration** does not open automatically after the configuration completes, open **MiCollab AM Configuration**, and select the **Lines** tab.
- 12** In the table from the **Lines** tab, configure callouts for the application. For information on configuring callout settings, see the topic *Configuring Callout Settings*, in the online help system.
- 13** Click **OK** to save all changes.

Configuring Existing MiCollab AM for the Integration

To configure exiting MiCollab AM for the telephone integration:

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

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

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

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

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

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

8 In the **Integration Options** dialog box, configure the options as follows:

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

Table 6. Required Parameter Settings for 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. <p>For example:</p> <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.
- 9** Click **OK**. The **Switch Section Options** dialog box displays.
- 10** In the **Switch Section Options** dialog box, configure the following options:
 - a** In the **Local Integration Settings** section, select the **Required Parameters** view.
 - b** In the **Incoming Hunt Mode** field, enter the mode for this integration.
 - 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** dropdown list, select **Failover Server Settings**.
- 5 Click the **Add Failover Server** button. Two new rows are added to configure the secondary SIP server.
- 6 In the **Secondary SIP Server Address** and **Secondary SIP Server Port** rows, enter the appropriate value as follows:

Table 7. Secondary SIP Server Address and the Secondary SIP Server Port example

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

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

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

Table 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** dropdown list, select **Failover Server Settings**.
- 5 In the **Failover Server Settings** view, click the **Remove Failover Server** button.
- 6 At the confirmation prompt, click **Yes** to confirm the deletion.

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

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

Changing the Network Binding Order on the MiCollab AM Platform

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

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

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

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

Windows Server 2008 R2 with Service Pack 1

To change the binding order of multiple NICs:

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

Windows Server 2012 R2

To change the binding order of multiple NICs:

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

Configuring Quality of Service (QoS)

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

Table 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