

MiCollab Advanced Messaging Genband CS-2100 SIP Trunk Integration Technical Note

For version 9.0 and above

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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	<i>SIP Routing Manger Installation Guide</i>
Server Documentation	<i>System Administration Guide</i>
Server Documentation	<i>System Installation and Configuration Guide</i>

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: See the *System Installation and Configuration 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 or spoken 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 Communication Server 2100 SIP Trunk 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 CS-2100 SIP Trunk

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, 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 [Critical Application Considerations](#).

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.

- If a different configuration of additional hardware is present, which can include G430 and G450 Media gateways.
- MiCollab AM 9.0 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.

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's Nortel CS2100 release 17 or later.

MiCollab AM Requirements

- MiCollab AM version 9.0
- 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

Configuring Genband CS2100

Modify the CS2100 Core Datafilling

Modify the following data tables with proper datafilling. During this procedure, the tuple entries must be **Added** or **Verified**.

Table: CLLI

CLLI	ADNUM	TRKGRSIZ	ADMININF
SIPTAVST	82	0	SIPT_AVST_VM_SYS

Table: TRKGRP

GRPKEY

GRPINFO

SIPTAVST

IBNT2 0 ELO NCRT MTL1 0 MIDL 250 2142399998 ANSDISC 0 Y N N N N
N N 0 250 N 0 0 0 0 N N N N N N N N NATL \$

Table: TRKOPTS

OPTKEY

OPTINFO

SIPTAVST

DPT

DPT SIPT NET_IP Y INTRA_DOMAIN 0

Table: DPTRKMEM

DPTRKKEY

RANGEINF

MAXCALLS

SIPTAVST

SIPT

24

TABLE: SIPLINK

LINKNAME CONNTYPE

AGNTDATA

SIPTAVSTLNK

CS2CS

ISUPTRK

SIPTAVST

TABLE: CLLIMITCE

SIPTAVST AVST

5

10

15

NSS

0

0

N

N (0)

```

Table: TRKSGRP
SGRPKEY CARDCODE SGRPVAR SGRPVAR
-----
SIPTAVST 0 DS1SIG C7UP 2W N N UNEQ NONE Q764 THRH 0 DMSNODE $ NIL CIC

```

```

TABLE: TRKNAME
ADNUM          CLLI
-----
82             SIPTAVST

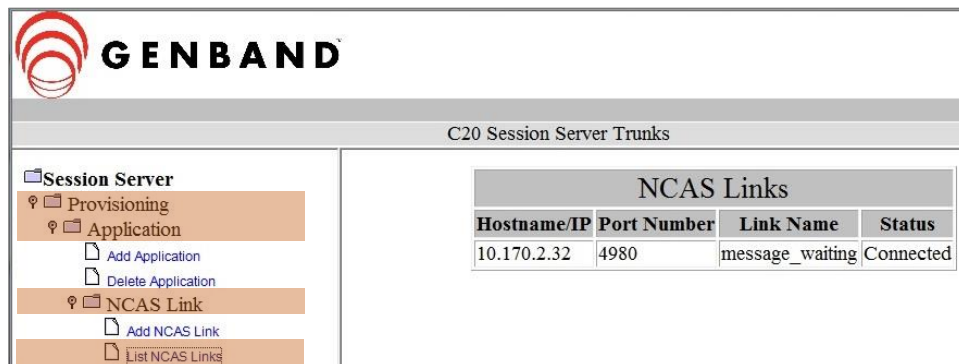
```

Configure the CS2100 Session Server Trunks

Verify NCAS Link

To verify NCAS Link through Session Server Trunk:

- 1 Log in to the Session Server Trunk, and go to **Provisioning > Application > NCAS Link > List NCAS Links**.
- 2 Verify that the NCAS link to the CS2100 core is defined.



Add Profiles

The SIP Gateway Profiles must be mapped correctly for the integration. The server profiles can be added or modified through the following menu: **Provisioning > Application > SIP Gateway > Profiles**.

NOTE If you are required to delete a SIP link, the profiles don't have to be configured. Deleting the SIP Trunk will automatically delete its associated profiles.

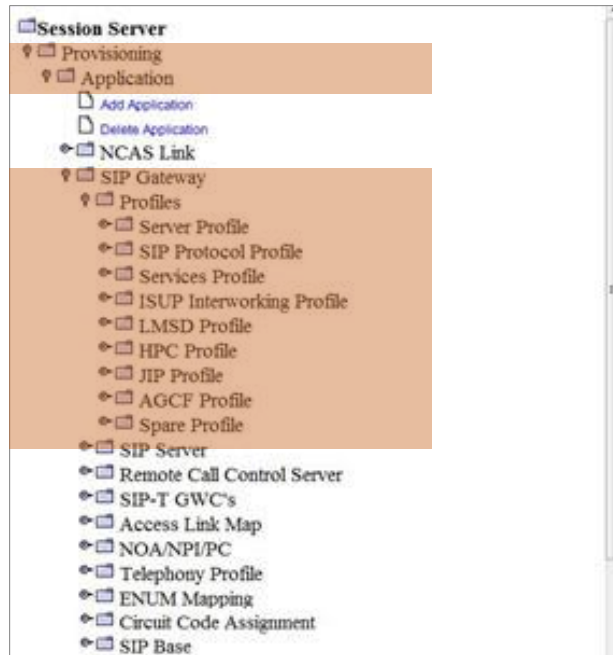


Figure 1. SIP Gateway Profiles Menu

The following profiles require configuration for proper mapping. The rest of the profiles (not mentioned in the list below) use the default settings.

- Server Profile
- SIP Protocol Profile
- Services Profile
- ISUP Interworking Profile

NOTE If you are upgrading NGSS to version SE17 from any previous version, the default profile names are created automatically. Select each profile and ensure that the configuration is correct as shown in this section.

Server Profile

Modify Server Profile

WARNING: Making provisioning changes at high traffic level may impact call processing!

1

Server Profile Name: PIP2IP

2

SIP PROTOCOL PROFILE: PIP2IP
ISUP INTERWORKING PROFILE: LAB_ISUP
SERVICES PROFILE: PIP2IP
LMSD PROFILE: DEFAULT
HPC PROFILE: DEFAULT

Image continues on next page

Image continued from next page

2

JIP PROFILE:	DEFAULT
AGCF PROFILE:	DEFAULT
SPARE PROFILE:	DEFAULT
ISUP to SIP Cause Map:	DEFAULT
SIP to ISUP Cause Map:	DEFAULT
SIP Redirection Map:	DEFAULT
ISUP Variant to SIP Ver Map:	DEFAULT
NOA/NPI to Phone Context Map:	DEFAULT

Modify Abort Operation

Figure 2. Server Profile Configuration

- 1 Create a **Server Profile Name** that is simple and noticeable, which will make it easier for mapping.
- 2 Note the profile names to line up.

SIP Protocol Profile

Modify SIP Protocol Profile

WARNING: Making provisioning changes at high traffic level may impact call processing!

1 SIP Protocol Profile Name: **PIP2IP**

Methods Supported:

☒ SUBSCRIBE ☒ NOTIFY
☒ UPDATE ☒ INFO

SIP Headers Supported: ☒ Content-Disposition ☒ Remote-Party-ID ☒ P-Asserted-ID
☐ Privacy ☒ Reason ☒ Replaces
☒ Referred-By ☒ Diversion ☐ History Info
☐ Generic Digits ☐ Charge Number ☐ Resource-Priority

URI Parameters Supported: ☐ CIC ☐ RN ☐ NPDI ☒ Phone-Context
☐ OLI ☐ CIP ☐ JIP ☐ Carrier Selection
☐ NOC ☐ CPC ☐ RN Context

SIP Header Format: ☐ Compact ☒ Long

Overlap Method: ☒ Info ☐ ReInvite

Long Call Audit Mechanism: Info

Session Timer Value: 20

Out of Band DTMF Payload: application/dtmf-relay

Unknown Header: NULL

Anonymous Header: NULL

ContentDispositionHandling: OPTIONAL

Modify Abort Operation

Figure 3. SIP Protocol Profile Configuration

- 1 Create a **SIP Protocol Profile Name** that is simple and noticeable, which will make it easier for mapping.

Services Profile

Modify Services Profile

WARNING: Making provisioning changes at high traffic level may impact call processing!

1

Services Profile Name: PIP2IP

☐ Alert-Info

Non-Standard MIME Type:

☐ UUI

☐ Location XML

Use OPTIONS for Heartbeat:

☒ Yes

☐ No

Use SIPS URI for TLS:

☒ Yes

☐ No

Post-Answer Media Exchange:

☒ ReInvite

☐ Update

Hold via Port 0:

☐ Yes

☒ No

Map DNIS/DNID to To Header:

NONE

Map To Header to DNIS/DNID:

NONE

OLI Interworking:

☒ Yes

☐ No

OLI Values:

0

Telephony Profile Support:

☒ Yes

☐ No

Accepts Early SDP:

☐ Yes

☒ No

Accept Invite Without SDP:

☐ Yes

☒ No

Enforce CODEC-Compatibility:

☐ Yes

☒ No

Accepts Encapsulated ISUP:

☐ Yes

☒ No

Conn Mode Allowed:

☐ Yes

☒ No

OCN and To Header Interworking:

☐ Yes

☒ No

Default 183:

☐ Yes

☒ No

EnhancedRetryAfterHandling:

☐ Yes

☒ No

Add Record Route values:

☐ Yes

☒ No

T.38AnnexD Supported:

☐ Yes

☒ No

Figure 4. Services Profile Configuration

- 1 Create a **Services Profile Name** that is simple and noticeable, which will make it easier for mapping.

ISUP Interworking Profile

NOTE This Profile needs to be built only once and will be common for all future builds.

Modify ISUP Interworking Profile

WARNING: Making provisioning changes at high traffic level may impact call processing!

1

ISUP Interworking Profile Name: LAB_ISUP

Use Network Number:

☐ Yes

☒ No

Suspend Support For SIP:

No Support

SIP Cause Precedence Priority:

Encapsulated Msg

First

Retry after reason

Second

Cause Code Map

Third

Image continues on next page

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BCI Data:

2 Interworking Indication: Interworking Encountered

ISUP / BICC Indicator: ISUP Not Used All the Way

ISDN Access Indicator: Terminating Access is not ISDN

FCI Data:

3 Domestic/International Call Indication: Treat as Domestic Call

Interworking Indicator: Interworking Encountered

ISDN User Part Indicator: ISUP Used All the Way

ISUP Preference Indicator: ISUP/BICC Not Required

ISDN Access Indicator: Originating Access is not ISDN

Hop-Counter Factor: 7

National Circuit Code: 8

Trunk Group ID Prefix: 0

Modify Abort Operation

Figure 5. ISUP Interworking Profile Configuration

- 1 Create a common name. For example, **LAB_ISUP**.
- 2 The defaults will not allow you to change this value.
- 3 The defaults will not allow you to change this value.

Add SIP Server

To add SIP Server:

- 1 From the Session Server Trunk, go to **SIP Gateway > Remote SIP Server > Add Server**.

GENBAND

C20 Session Server

Session Server

- Provisioning
 - Application
 - Add Application
 - Delete Application
 - NCAS Link
 - SIP Gateway
 - Remote SIP Server
 - Add Server
 - List Servers
 - Remote Call Control Server
 - SIP-T GWC's
 - Access Link Map
 - NOA/NPI/PC
 - Telephony Profile
 - ENUM Mapping
 - Circuit Code Assignment
 - SIP Base
 - ISUP and SIP Mappings
 - Switch

List Remote SIP Servers

Number of Remote SIP Servers = 15

Server Name	Details	Delete
MitelSVR	Details	Delete
CISCOCCM	Details	Delete
ESNA	Details	Delete
EXCHANGE	Details	Delete
GPS2IP	Details	Delete
MASMEETME	Details	Delete
OCSVOICE	Details	Delete
PIP2A2E	Details	Delete
PIP2IP	Details	Delete
RMS1AS	Details	Delete
S2100IP	Details	Delete

2 On the **SIP Server** screen:

Modify SIP Server

WARNING: Making provisioning changes at high traffic level may impact call processing!

Server Name: **PIP2IP**

Server Type: **Session Server**

Domain Name: **NULL**

Enable DNS Lookup ☐ Yes ☒ No

Protocol:

Port:

IP Address: **10.224.6.56** Port: **5060** Protocol: **UDP** Remote Config: **send+receive** IPsec e

Add Opt IP Address

Server Identifier: **NONE**

Server Profile: **PIP2IP**

Modify **Abort Operation**

- a Define the **Server Name** for the Mitel server.
- b In **Server Type**, select **Session Server**.
- c In **IP Address**, set the IP Address of the Mitel server.
- d In **Server Profile**, select the SIP server defined for the Mitel server.

3 Configure the remaining options as follows:

Advanced SIP Options

Methods Supported: ☐ INVITE ☒ CANCEL ☒ BYE ☒ OPTIONS
☒ SUBSCRIBE ☒ NOTIFY ☒ REFER ☒ PRACK
☐ UPDATE ☐ INFO

SIP Headers Supported: ☒ Content-Disposition ☒ Remote-Party-ID ☒ P-Asserted-ID
☐ Privacy ☒ Alert-Info ☒ Reason ☐ Replaces
☒ Referred-By ☒ Diversion ☐ History Info

☐ Generic Digits ☐ Charge Number ☐ Resource-Priority

URI Parameters Supported: ☒ CIC ☒ RN ☒ NPDI ☒ Phone-Context
☐ OLI ☐ CIP ☐ JIP ☐ Carrier Selection
☐ NOC ☐ CPC

Non-Standard MIME Type: ☐ UII ☐ Location XML

SIP Header Format: ☐ Compact ☒ Long

Use OPTIONS for Heartbeat: ☐ Yes ☒ No

Use SIPS URI for TLS: ☒ Yes ☐ No

Post-Answer Media Exchange: ☒ ReInvite ☐ Update

Overlap Method: ☒ Info ☐ ReInvite

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Image continued from previous page

LMSD Supported:	<input type="radio"/> Yes <input checked="" type="radio"/> No	Default
Alert-Info Handling Policy:		
Busy Tone	<input checked="" type="radio"/> Map to RIs Cause Value	17 - User Busy
	<input type="radio"/> Tandem	
Caller Waiting Tone	<input checked="" type="radio"/> Map to request Ring Back Tone	
	<input type="radio"/> Tandem	
Congestion Tone	<input checked="" type="radio"/> Map to Cause Value	42 - Network Congestion
	<input type="radio"/> Tandem	
Comfort Tone	<input checked="" type="radio"/> Map to request Tone	Silent Tone
	<input type="radio"/> Tandem	
Hold via Port 0:	<input type="radio"/> Yes <input checked="" type="radio"/> No	
Map DNIS/DNID to To Header:	NONE	
Map To Header to DNIS/DNID:	NONE	
OLI Interworking:	<input checked="" type="radio"/> Yes <input type="radio"/> No	

OLI Values:	0
Telephony Profile Support:	<input checked="" type="radio"/> Yes <input type="radio"/> No
Accepts Early SDP:	<input checked="" type="radio"/> Yes <input type="radio"/> No
Accept Invite Without SDP:	<input checked="" type="radio"/> Yes <input type="radio"/> No
Enforce CODEC-Compatibility:	<input type="radio"/> Yes <input checked="" type="radio"/> No
Accepts Encapsulated ISUP:	<input checked="" type="radio"/> Yes <input type="radio"/> No
Conn Mode Allowed:	<input type="radio"/> Yes <input checked="" type="radio"/> No
OCN and To Header Interworking:	<input type="radio"/> Yes <input checked="" type="radio"/> No
Default 183:	<input type="radio"/> Yes <input checked="" type="radio"/> No
EnhancedRetryAfterHandling:	<input type="radio"/> Yes <input checked="" type="radio"/> No
Add Record Route values:	<input type="radio"/> Yes <input checked="" type="radio"/> No
T.38AnnexD Supported:	<input type="radio"/> Yes <input checked="" type="radio"/> No
Re-Invite for Voice Band Data:	<input type="radio"/> Yes <input checked="" type="radio"/> No
Enhanced Media Cut Through:	<input checked="" type="radio"/> Yes <input type="radio"/> No
BufferACMonEMCT:	<input type="radio"/> Yes <input checked="" type="radio"/> No
SupportForGlobalRN:	<input type="radio"/> Yes <input checked="" type="radio"/> No
MBG Supported:	<input type="radio"/> Yes <input checked="" type="radio"/> No
PRACK with SDP Supported:	<input type="radio"/> Yes <input checked="" type="radio"/> No
Auto-Subscribe:	<input type="radio"/> Yes <input checked="" type="radio"/> No
Retain Contact Info:	<input checked="" type="radio"/> Yes <input type="radio"/> No
Validate Request URI:	<input type="radio"/> Yes <input checked="" type="radio"/> No
2 CLI Supported:	<input type="radio"/> Yes <input checked="" type="radio"/> No
Use Network Number:	<input type="radio"/> Yes <input checked="" type="radio"/> No
Reject REFER with 403:	<input type="radio"/> Yes <input checked="" type="radio"/> No
E.164 Format Allowed:	<input type="radio"/> Yes <input checked="" type="radio"/> No
Use DefaultLD for LD calls:	<input type="radio"/> Yes <input checked="" type="radio"/> No
Exhaust Overload Processing:	<input type="radio"/> Yes <input checked="" type="radio"/> No

Wait Timer(Sec) :	10
Stop Timer(Sec) :	10
Call Gap Increment %:	10
Max Call Gap Increment %:	99
Country Code:	0
Long Call Audit Mechanism:	Info
Session Timer Value:	20
Out of Band DTMF Payload:	application/telephone-event
Network Number:	NULL
Unknown Header:	NULL
Anonymous Header:	NULL
Server Identifier:	NONE
Overload Response Code:	503
ContentDispositionHandling:	OPTIONAL
Suspend Support For SIP:	No Support

Image continued on next page

Image continued from previous page

Segmentation Message Support:	<input type="radio"/> Yes <input checked="" type="radio"/> No
T34 Timer (secs) :	4
MSRID:	NULL
Connected Party Information:	<input type="radio"/> Yes <input checked="" type="radio"/> No

HPC Options

HPC Privilege :	NetworkCore
RPH Requirement :	requires-optional
HPC Calls Allowed :	<input checked="" type="radio"/> Yes <input type="radio"/> No

HPC Handling :	reject
Max Simultaneous HPC Session :	500
DSCPx Marking Allowed:	<input type="radio"/> Yes <input checked="" type="radio"/> No
DSCPx Value :	40
Use RSS Based HPC Strings :	<input type="radio"/> Yes <input checked="" type="radio"/> No

Add HPC String

SIP PSTN Interworking Options

ISUP to SIP Cause Map:	DEFAULT
SIP to ISUP Cause Map:	DEFAULT
SIP Redirection Map:	AAM302
ISUP Variant to SIP Ver Map:	DEFAULT
NOA/NPI to Phone Context Map:	DEFAULT

SIP Cause Precedence Priority:	
Encapsulated Msg	First
Retry after reason	Second
Cause Code Map	Third

BCI Data:	
Interworking Indication	Interworking Encountered
ISUP / BICC Indicator	ISUP Not Used All the Way
ISDN Access Indicator:	Terminating Access is not ISDN

Configure CS2100 to Allow the Route Advance Feature

In order to allow the **Route Advance** to the function, modifications need to be made to the Software Optionality Control (SOC) and Session Server Trunks. Follow the procedures in the section to configure required SOC and Session Server Trunk.

Configure the Required Software Optionality Control (SOC)

In order to allow **Re-routing** based on the **Release Cause Value**, Software Optionality Control (SOC) must be configured properly. Follow the procedure in this section to configure the required SOC.

To configure the required SOC to allow the Route Advance feature:

- 1 Enable the **SOC** shown in the following example.

```
GROUP:ISP

OPTION      NAME                RTU STATE  USAGE  LIMIT  UNITS  LAST_CHG
-----
ISP70008    Flexible CAUSEMAP      Y    ON    -      -      -      08/09/30
```

- 2 After the **SOC** is enabled, the option **RTEADV** (Route Advance) is added to the table, **FLXCMAP**.

```
TABLE: FLXCMAP
-----
CSEMPKEY TREAT RTEADV
```

- 3 Change the **RTEADV** value to **Y** only in the tuple shown in the following example. This will allow **Route Advance**.

```
TABLE: FLXCMAP
-----
CSEMPKEY TREAT RTEADV
CSE_102      CCITT_STANDARD SSTD Y
```

Configure the Session Server Trunk

To configure the Session Server Trunk to allow the Route Advance feature:

- 1 Log in to the Session Server Trunk, and go to **Provisioning > Application > SIP Gateway > ISUP and SIP Mappings > SIP to ISUP Map > Add mapping**.

The screenshot shows the Genband C20 Session Server Trunks web interface. On the left is a navigation tree with the following structure:

- Session Server
 - Provisioning
 - Application
 - Add Application
 - Delete Application
 - NCAS Link
 - SIP Gateway
 - Profiles
 - SIP Server
 - Remote Call Control Server
 - SIP-T GWC's
 - Access Link Map
 - NOA/NPI/PC
 - Telephony Profile
 - ENUM Mapping
 - Circuit Code Assignment
 - SIP Base
 - ISUP and SIP Mappings
 - ISUP to SIP Map
 - SIP to ISUP Map
 - Add Mapping
 - Delete Mapping
 - List Mappings
 - SIP Redirection
 - ISUP Variant Map

The main content area is titled "Add a SIP to ISUP Mapping". It contains the following text and form elements:

This command will add a new SIP Response Code to ISUP Cause Mapping based on the "base mapping" but can then be modified as needed by selecting it from the List Mappings menu.

New Mapping Name:

Base Mapping Name:

WARNING: Making provisioning changes at high traffic level may impact call processing!

NOTE: SIP to ISUP Mapping names will be converted to all upper case.

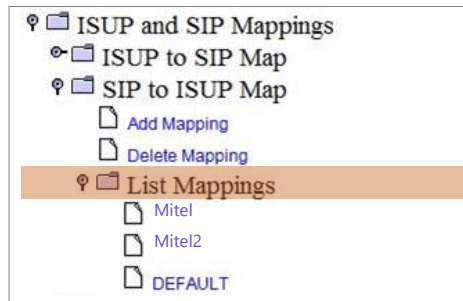
- 2 In the **Add a SIP to ISUP Mapping** screen, create two new mappings.
- The first mapping is used by the first and second Mitel Servers for **Route Advance**.
 - The second mapping is used by the 3rd Mitel server to treat calls properly in the case of **All Server Down** or **All Trunks Busy**.
- 3 To create a mapping, follow these steps:

NOTE In this guide, *Mitel* (first mapping) and *Mitel2* (second mapping) will be used as examples.

New Mapping Name:

Base Mapping Name: **DEFAULT** ▼

- a In the **New Mapping Name** field, type *Mitel*.
 - b In the **Base Mapping Name** dropdown list, select **Default**. This sets the default mapping.
 - c Click **Add**.
 - d Using the same procedure, create *Mitel2*.
- 4 On the **Session Server** tree menu, select **List Mappings** to view the mappings created.



- 5 Click the first mapping, **Mitel**, to modify the **Cause Code** map.
- 6 On the **Cause Code** map, modify only **486** and **503**.

485	1	Modify
486	102	Modify
487	16	Modify
488	31	Modify
500	41	Modify
501	79	Modify
502	38	Modify
503	102	Modify

- 7 Click **Modify** next to the **486** cause code.

Modify a SIP to ISUP Cause Mapping Tuple

Mapping Name: **Mitel**

SIP Response Code: **486**

ISUP Release Cause: **102**

WARNING: Making provisioning changes at high traffic level may impact call processing!

- 8 In the **ISUP Release Cause** box, change the code to *102*, and then click the **Modify** button.
- 9 Repeat **Steps 7 & 8** to modify the **503** cause code.

10 Now modify the second mapping. From the **List Mappings** tree, select the second mapping, **Mitel2**.

11 On the **Cause Code** map, modify only **486**.

484	28	Modify
485	1	Modify
486	34	Modify
487	16	Modify
488	31	Modify

12 Click **Modify** next to the **486** cause code.

13 In the **ISUP Release Cause** box, change the code to 34, and then click the **Modify** button.

14 Next, change the SIP to ISUP Cause Map for each **Mitel** SIP Server.

15 From the **Session Server** tree menu, go to **Provisioning > Application > SIP Gateway > Server Profile > List Server Profiles**.

Session Server

- Provisioning
 - Application
 - Add Application
 - Delete Application
 - NCAS Link
 - SIP Gateway
 - Profiles
 - Server Profile
 - Add Server Profile
 - List Server Profiles
 - SIP Protocol Profile
 - Services Profile
 - ISUP Interworking Profile
 - LMSD Profile
 - HPC Profile
 - JIP Profile
 - AGCF Profile
 - Spare Profile
 - SIP Server
 - Remote Call Control Server
 - SIP-T GWC's
 - Access Link Map
 - NOA/NPI/PC
 - Telephony Profile
 - ENUM Mapping

Modify Server Profile

WARNING: Making provisioning changes at high traffic level may impact call processing

Server Profile Name: MitelSVR

SIP PROTOCOL PROFILE: MitelSVR

ISUP INTERWORKING PROFILE: LAB_ISUP

SERVICES PROFILE: MitelSVR

LMSD PROFILE: DEFAULT

HPC PROFILE: DEFAULT

JIP PROFILE: DEFAULT

AGCF PROFILE: DEFAULT

SPARE PROFILE: DEFAULT

ISUP to SIP Cause Map: DEFAULT

SIP to ISUP Cause Map: Mitel

SIP Redirection Map: DEFAULT

ISUP Variant to SIP Ver Map: DEFAULT

NOA/NPI to Phone Context Map: DEFAULT

[Modify](#) [Abort Operation](#)

16 From the list of servers, select the first **Mitel Server**.

17 In the **Modify Server Profile** screen, change the **SIP to ISUP Cause MAP** value to the first mapping you created (**Mitel** in this example).

18 Change the **SIP to ISUP Cause MAP** value to **Mitel** for the second **Mitel Server** as well.

19 From the list of servers, select the third **Mitel2 Server**.

20 Change the **SIP to ISUP Cause MAP** value to **Mitel2**.

Modify Server Profile

WARNING: Making provisioning changes at high traffic level may impact call processing!

Server Profile Name:	Mitel2SVR
SIP PROTOCOL PROFILE:	Mitel2SVR
ISUP INTERWORKING PROFILE:	LAB_ISUP
SERVICES PROFILE:	Mitel2SVR
LMSD PROFILE:	DEFAULT
HPC PROFILE:	DEFAULT
JIP PROFILE:	DEFAULT
AGCF PROFILE:	DEFAULT
SPARE PROFILE:	DEFAULT
ISUP to SIP Cause Map:	DEFAULT
SIP to ISUP Cause Map:	Mitel2
SIP Redirection Map:	DEFAULT
ISUP Variant to SIP Ver Map:	DEFAULT
NOA/NPI to Phone Context Map:	DEFAULT

Modify Abort Operation

21 This completes the SIP Gateway configuration to use the **Route Advance** functionality.

Modify the CS2100 Core Translations

After configuring the SIP to ISUP mapping, mo CS2100

To modify the CS2100 Core Translations:

- 1 In table **IBNRTE**, create a route for the third Mitel system.

RTE	RTELIST
	OPTIONS
-----	-----
892	(N N N N N TOM2AVST 0) \$
	\$

- 2 Next, create a route for the Second Mitel System.

RTE	RTELIST
	OPTIONS
-----	-----
891 (N N N N N TOM1AVST 0) (CND NRR T IBNRTE 892) \$	
	\$

- 3 In this route, the **CND NRR** option is used to **Advance** to the third Mitel system if a **486** or **503** cause code is received.

- 4 Next create a primary route for the Mitel Systems.

RTE	RTELIST
	OPTIONS
-----	-----
890 (N N N N N SIPTAVST 0) (CND NRR T IBNRTE 891)	
(N N N N N TOM1AVST 0) (CND NRR T IBNRTE 892)	
(N N N N N TOM2AVST 0) \$	

- 5 This route is used for primary routing. Each route is followed by an **NRR** route to advance the routing if a **486** or **503** cause code is received.
- 6 In table **DNROUTE**, the **Primary** dn is pointed to the **Primary** route.

AREACODE	OFCCODE	STNCODE	DNRESULT		
214	239	9990	T	IBNRTE	890

- 7 This completes the modification that are needed.

Configure CS2100 to Enable MWI

SIP NMS Support Based on RFC 3842

Voice Mail (VM) allows a user to have incoming calls forwarded to a message desk, to be notified that recorded messages are waiting (via a Message Waiting Indicator – MWI), and to retrieve and play back those messages later. Network Message Service (NMS) supports VM across a network, which enables message desk functionality to be provided centrally for a number of different nodes.

In releases prior to ISN09, the C20 supported only a circuit-based network implementation for NMS. This feature extends NMS support to packet networks using the Session Initiation Protocol (SIP) event package for MWI defined in RFC 3842.

Features 00007544 and 00013457 provide C20 support for NMS from a VMS using SIP RFC3842 MWI notifications to the C20. This supports the following types of end user:

- Traditional Phones (e.g., POTS, IBN) on IAD Gateways (H248, NCS, MGCP) on the C20
- CentrexIP Client Manager (CICM) IP sets on the C20
- SIP lines on the C20
- Remote Lines on a TDM switch connected to C20 by ETSI SS7
- Remote Lines on a remote SIP Server connected to C20 by SIP-T (ETSI ISUP)
- Remote Lines on a TDM PBX connected to C20 by ETSI PRI

Feature 00012847 extends this support to **Remote Lines** on TDM PBXs connected to C20 by DPNSS.

Feature 00007544 also enables a VMS using SMDI MWI to serve SIP clients by C20 converting the SMDI MWI notifications to SIP RFC3842 MWI for:

- C20 hosted SIP Lines
- A remote SIP Server connected to C20 using SIP-T (ETSI ISUP) trunks.

NOTES

- Networking of MWI via C20 to remote clients on SIP PBXs, H323 PBXs and remote SIP Servers connected by pure SIP or SIP-T encapsulating anything other than ETSI ISUP are not yet supported in the International load.

- Currently, MWI and NMS support directory numbers of up to 10 digits, with numbers longer than this having their leading digits dropped.

PRINCIPAL BENEFITS

Provides support for Voice Mail services over packet networks (public networks and VPNs) without the need for a separate CCS7 network, reducing operating costs and providing the potential for increased revenue for network operators and improved customer satisfaction. It is also a stepping stone towards IP-based Unified Messaging (UM) combining multiple types of mail messages (e.g. voicemail, email, paging).

MWI & NMW E.164 SUPPORT 206848/00023764

KEY CAPABILITIES

Currently, Message Waiting Indication and Network Message Waiting functionality support 10 digit numbers, with numbers longer than this having their leading digits dropped. Use of greater than 10 greater than 10 digit numbers in a variable length dial plan combined with the broader geographic deployment enabled by packet network technology can result in number ambiguity, with messages being directed to the wrong voice mailbox.

The objective of this feature is to provide C20 Core and SST support for E.164 numbers of up to 15 digits for SIP voice mail interfaces and with Message Waiting and Network Message Waiting services.

Note that this enhancement does not apply to SMDI-based voicemail, because the SMDI interface does not currently support more than 10 digits.

PRINCIPAL BENEFITS

Provides MWI and NMW compatibility with E.164 numbering and ensures that voicemail is not delivered to the wrong user's mailbox because of ambiguous numbering arising from long number truncation.

KEY CAPABILITIES

This feature increases the maximum number of preset conferences that can be supported on a C20, which is determined by the size of table PRECONF, from 64 to 1024. This table expansion is controlled by new SOC MDC00080.

PRINCIPAL BENEFITS

Ensures that network operators can provide adequate preset conference capacity to fulfill customer requirements, notably in the public services sector (fire, rescue, etc.).

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:

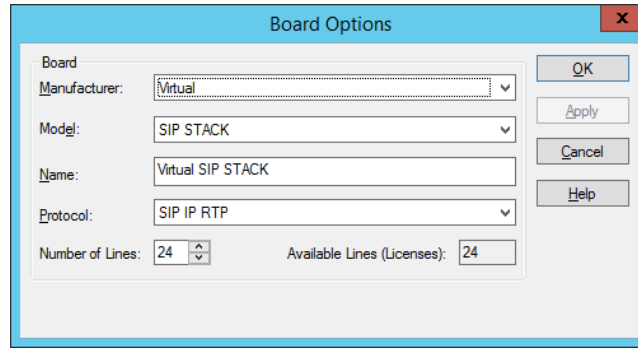
- [Configuring MiCollab AM for the Integration During Initial Installation](#): Integrate the telephone system while you install MiCollab AM for the first time.
- [Configuring Existing MiCollab AM for the Integration](#): Integrate a new telephone system on your existing MiCollab AM system.

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** dropdown list, select **Genband**.
 - d From the **Model** dropdown list, select **CS2100**.
 - e From the **Integration Type** dropdown list, select **SIP Trunk**.
- 2 Click **Next**. The **Board Options** dialog box appears.



- 3 In the **Board Options** dialog box, configure the following options:
 - a From the **Manufacturer** dropdown list, select **Virtual**.
 - 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 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.
 - 7 In the **Integration Options** dialog box, verify that the communication settings are correct.
 - 8 Click **OK**. The **Switch Section Options** dialog box appears.

9 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 4 Required Parameters View – Integration Options

Field	Required Value
SIP Server Address	Enter the IP address of the switch.
SIP Server Port	Enter the port number on which the switch listens for SIP messages. The default port number is 5060 .
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> <p>The extension that will be used as the hunt number on the PBX followed by the @ symbol and the IP of the call server, such as 5000@172.16.4.202. The hunt number must be unique across all IP integrations.</p> <p>The Fully Qualified Domain Name (FQDN) of the switch, such as pbx1.sipdomain.com.</p>

	<p>NOTE This setting should match a string in the SIP header that is unique to this particular integration.</p>
SIP Domain Name	<p>Enter the SIP domain name. This value must be the same value as in the procedure, Obtain SIP Domain.</p> <p>IMPORTANT Be sure the Authoritative Domain name is the same throughout the server, Communication Server 1000E, and MiCollab AM programming.</p>
Transport for outgoing SIP messages	<p>Enter TCP or UDP.</p> <p>NOTE This value must match the protocol selected on the Entity Link created for MiCollab AM Call Server.</p>
Local IP Address to bind on	<p>Enter the IP address of the network interface card (NIC) on the Call Server platform that supports the SIP integration. If there is only one NIC on the MiCollab AM server platform, this field typically contains the IP address of that NIC already.</p>
SIP Location Connection Port	<p>Enter the port number on which MiCollab AM listens for incoming SIP messages.</p> <p>The default value is 5060.</p>
Media packet size (milliseconds)	<p>MiCollab AM sends/receives packets containing the number of milliseconds worth of audio data set here.</p> <p>The default value is 20.</p>

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

- Select the **Validate Remote Hosts for Media** checkbox.
- 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.

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

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

- 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.

NOTE This integration supports **Terminal**, **Circular**, **Reverse Terminal**, and **Reverse Circular** hunt modes only.

- c** In the **Hunt Group Access Code** field, enter the number that matches with the DN prefix programmed for the **Gateway Endpoint** associated with this MiCollab AM Call Server.
 - d** Click **OK**.
- 12** Continue through and complete the configuration. At the end of the configuration, a confirmation dialog box appears. Click **OK**.
- 13** If **MiCollab AM Configuration** does not open automatically after the configuration completes, open **MiCollab AM Configuration**, and select the **Lines** tab.
- 14** 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.
- 15** 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** Depending on the type of Aculab card you have installed, configure the board options. Refer to the appropriate *Spare Parts document* for more information on the Aculab card you are installing.
 - b** Click **OK**.
- 4** Select the **Switch** tab and click the **Add** button. The **Switch Integration Data Setup** dialog box appears.
 - a** From the **Manufacturer** dropdown list, select **Genband**.
 - b** From the **Model** dropdown list, select **CS2100**.
 - c** From the **Integration Type** dropdown 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 *System Installation and Configuration Guide*.

- 7 Click **OK**. The **Integration Options** dialog box appears.
- 8 In the **Integration Options** dialog box, verify that the communication settings are correct.
- 9 Click **OK**. The **Switch Section Options** dialog box appears.

- 10 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 switch.
SIP Server Port	Enter the port number on which the switch listens for SIP messages. The default port number is 5060 .
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.

	<ul style="list-style-type: none"> • Multiple SIP integrations on the call server: Use a string that is unique to each SIP integration. <p>For example:</p> <p>The extension that will be used as the hunt number on the PBX followed by the @ symbol and the IP of the call server, such as 5000@172.16.4.202. The hunt number must be unique across all IP integrations.</p> <p>The Fully Qualified Domain Name (FQDN) of the switch, such as pbx1.sipdomain.com.</p> <p>NOTE This setting should match a string in the SIP header that is unique to this particular integration.</p>
SIP Domain Name	<p>Enter the SIP domain name. This value must be the same value as in the procedure, Obtain SIP Domain.</p> <p>IMPORTANT Be sure the Authoritative Domain name is the same throughout the server, Communication Server 1000E, and MiCollab AM programming.</p>
Transport for outgoing SIP messages	<p>Enter TCP or UDP.</p> <p>NOTE This value must match the protocol selected on the Entity Link created for MiCollab AM Call Server.</p>
Local IP Address to bind on	<p>Enter the IP address of the network interface card (NIC) on the Call Server platform that supports the SIP integration. If there is only one NIC on the MiCollab AM server platform, this field typically contains the IP address of that NIC already.</p>
SIP Location Connection Port	<p>Enter the port number on which MiCollab AM listens for incoming SIP messages.</p> <p>The default value is 5060.</p>
Media packet size (milliseconds)	<p>MiCollab AM sends/receives packets containing the number of milliseconds worth of audio data set here.</p> <p>The default value is 20.</p>

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

- Select the **Validate Remote Hosts for Media** checkbox.
- 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.

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

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

- 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.

NOTE This integration supports **Terminal**, **Circular**, **Reverse Terminal**, and **Reverse Circular** hunt modes only.

- c** In the **Hunt Group Access Code** field, enter the number that matches with the DN prefix programmed for the **Gateway Endpoint** associated with this MiCollab AM Call Server.

- d** Click **OK**.

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

14 Select the **Lines** tab.

15 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.

16 Click **OK** to save all changes.

Configure 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 6. Secondary SIP Server Address and the Secondary SIP Server Port example

Field	Value
-------	-------

Secondary SIP Server Address Enter the TCP/IP address or an FQDN of the secondary node.

For example:

The IP address 123.45.6.789 as displayed on the Review/Modify SIP Gateway screen.

NOTE This integration requires the machine name to be a fully qualified domain name. Therefore, use the Machine Name field as displayed on the Review/Modify SIP Gateway screen during the integration process.

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

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

7 From the **View** dropdown 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 7. Integration Specific Parameters

Field	Value
Enable SIP server failover	Select this check box to allow for failover and to enable the failover server setting changes.
Delay (in ms) between Failover attempts	The delay in milliseconds before MiCollab AM attempts to register its port with the SIP server. The default is 1000 ms.
Incoming off hook delay	800
Outgoing off hook delay	0
On hook delay	300
Type of Call Progress to use for External Calls	How this should be set depends on the gateway used for the integration. <ul style="list-style-type: none">• If the gateway supports call progress through to the endpoint, set to Digital.

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

- 9 Click **Apply** to save the changes.
- 10 To add another failover server repeat **Steps 4-9**.
- 11 Click **OK** to close the **Integration Options** dialog box.

To remove a SIP Failover Server:

- 1 From **MiCollab AM Configuration**, click the **Integrations** tab.
- 2 From the **Integrations** list, select your integration, and then click **Edit**.
- 3 In the **Integration Options** dialog box, go to the **Local Integration Settings** section.
- 4 From the **View** 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**.

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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 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

To change the binding order of multiple NICs:

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

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

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

Configuring Quality of Service (QoS)

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

Table 8. QoS Configuration

Field	Setting
Application Name	At_TelephonyServer.exe
Protocol	Match the setting used for the integration UDP or TCP
Source Port	<p>MiCollab AM requires a range of ports for audio support. The MiCollab AM audio ports start at the Local Media Base UDP Port configured in the Server tab. Each MiCollab AM line reserves 10 ports. Hence, the port range starts from the number configured there, and goes to the last port of the last line. The formula for calculating the highest port number in the range is as follows:</p> $\text{BasePortNumber} + (\text{NumberOfCXPorts} * 10) - 1.$ <p>Hence, if the base port is 10000, and MiCollab AM has 8 lines, then the port range to use would be:</p> <p>10000:10079</p>
DSCP Value	46