

**MiCollab Advanced Messaging**  
**Avaya Communication Server 1000**  
**SIP Trunk with NRS**  
**Integration Technical Note**

For version 6.1 and above

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# Contents

<b>Preface</b>	<b>5</b>
References	5
Documentation	5
Documentation Updates	6
Help	6
Document Conventions	6
Features Supported by This Integration	7
<b>Critical Application Considerations</b>	<b>10</b>
<b>Integrating with the SIP Routing Manager</b>	<b>12</b>
<b>Installation Requirements</b>	<b>14</b>
Telephone System Requirements for CS1000 Release 5.5	14
Telephone System Requirements for CS1000 Release 6.0 through 7.6	14
MiCollab AM Requirements	14
<b>Programming the Telephone System</b>	<b>15</b>
Programming the D-Channel	15
Programming the Trunk Route	15
Programming SIP Trunks	16
Programming the Route List Index	17
Programming the Steering Code	18
Programming Customer Data Block (CDB) Network Data	18
Programming Subscriber Telephones	19
<b>Programming the Release 5.50 Signaling Server for Integration with MiCollab AM</b>	<b>20</b>
Configuring the Network Routing Service	20
<b>Programming the Release 5.50 Signaling Server for Integration with MiCollab AM and the SIP Routing Manager</b>	<b>22</b>
Configuring the Network Routing Service	22
<b>Configuring the Element Manager</b>	<b>25</b>
<b>Programming the Release 6.0 through 7.5 Signaling Server for integration with MiCollab AM</b>	<b>26</b>

Configuring the Network Routing Service	26
<b>Programming the Release 6.0 through 7.5 Signaling Server for use with MiCollab AM and the SIP Routing Manager</b>	<b>28</b>
Configuring the Network Routing Service	28
<b>Configuring the Element Manager</b>	<b>31</b>
<b>Configuring MiCollab AM</b>	<b>32</b>
Configuring MiCollab AM for the Integration During Initial Installation	32
Configuring Existing MiCollab AM for the Integration	35
Configuring MiCollab AM for SIP Failover	37
<b>Changing the Network Binding Order on the MiCollab AM Platform</b>	<b>40</b>
Windows Server 2008 R2 with Service Pack 1	40
Windows Server 2012 R2	41
<b>Configuring Quality of Service (QoS)</b>	<b>42</b>

# 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 Avaya Communication Server 1000 telephone system.

This document describes how to integrate MiCollab AM with an Avaya® Communication Server 1000 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.

The Avaya SIP Trunk integration consists of a number of components: the Avaya Call Server (Avaya Communication Server 1000 telephone system), the Avaya Signaling Server, and MiCollab AM. The signaling server keeps track of the IP devices, their locations, and call routing; the call server is responsible for all non-IP processes. MiCollab AM uses SIP trunks to integrate with the switch and thus uses static SIP endpoints. The SIP trunks are configured as virtual trunks on the CS1000 and the corresponding SIP gateway endpoints are configured on the Avaya Signaling Server. The Signaling Server routes all calls originating on the switch to MiCollab AM. Similarly, the Signaling Server routes all of the outgoing calls and MWI requests from MiCollab AM to their proper destinations.

Deploying the SIP Routing Manager Service changes the way in which SIP calls are directed, it does not change the destination of either inbound or outbound calls. The SIP Routing Manager acts as an intermediary; it redirects SIP call traffic to the appropriate destination. For more information on the SIP Routing Manager Service, refer to the section, [Integrating with the SIP Routing Manager](#).

## References

A catalog of technical documentation is included on the MiCollab AM Installation Media. If you are installing any advanced applications, such as Networking and Fax Server applications, you should refer to the appropriate technical documentation for application and installation information.

## Documentation

The technical documentation is produced in the PDF format and requires the PDF reader to view it. The documentation set for this MiCollab AM includes the following documents and resources:

- **Developer Resources.** Contains programming guides and API references for developers for integrating the server clients and web applications with MiCollab AM.
- **Integration Technical Notes (ITN).** Contains a set of guides that describe the integration methods and instructions for a variety of phone systems to work with MiCollab AM. The ITNs are generally used by resellers or administrators who are experienced with MiCollab AM and familiar with the integration procedures and terminology.

- **Quick Reference Card (QRC).** Contains shortcuts and quick instructions telling subscribers how to access and use the messaging system.
- **Server Documentation.** Available as a PDF only. Contains administrative guides for administrators about installing, configuring, and administering the messaging system, and user guides for subscribers about accessing the messaging system and checking and sending messages.
- **Spare Parts Documentation.** Contains a set of guides that describe the instructions for installing and configuring hardware parts to work with MiCollab AM. These documents are written for Mitel certified MiCollab AM technicians who are experienced with MiCollab AM and familiar with the procedures and terminology.
- **Software Release Notice (SRN).** This notice introduces the new features, capabilities, and hardware/software requirements for the corresponding MiCollab AM version.

## Documentation Updates

Documentation updates may be available from the following sources:

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

## Help

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

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

## Document Conventions

The following conventions are used in this document:

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

Example: **Enter**

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

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

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

Example: Refer to *System Installation Guide*.

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

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

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

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

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

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

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

## Features Supported by This Integration

The following tables list the features supported using the Communication Server 1000 SIP Trunk integration.

Table 1. Call forward to personal greeting support for these common call types

Divert to MiCollab AM on	Supported
No Answer	Yes
Busy	Yes
Forward All	Yes
Do Not Disturb	No

Table 2. Integration features supported for Avaya Communication Server 1000 SIP Trunk with NRS

Feature	Supported	Notes
Automatic subscriber logon	Yes	
ANI/CLI	Yes	
Announce Busy greeting on forwarded calls	Yes	

Call screening	Yes	Note 1
Caller queuing	Yes	Note 2
DNIS	Yes	
End-to-end DTMF, attendant console	Yes	
End-to-end DTMF, proprietary telephones	Yes	
Fax Tone Detection	Yes	
Internal calling party ID for reply	Yes	
Live record, integrated	No	
Live reply to sender	Yes	
Message notification callouts	Yes	
MWI, set/clear	Yes	
MWI, inband/outband	Outband	
Networking, analog	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	No	Note 4
TLS	No	Note 4
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	
Multiple Integrations	Yes	Note 3



## 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. Refer to the [Critical Application Considerations](#) section of this document for limitations on these features
4. 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.

# 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 SIP Domain Name in the Integration Options dialog box must match the domain name configured on the Signaling Server. This is a case sensitive value.
- 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. 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. For more information, refer to [Changing the Network Binding Order](#) on the MiCollab AM Platform later in this document.
- MiCollab AM supports G.729a with support for annex b on the incoming audio stream only. MiCollab AM does not transmit annex b packets.
- When codec negotiation takes place between MiCollab AM and the PBX, MiCollab AM always offers the G.729a audio format as an option. You may configure G.729a as the preferred codec in MiCollab AM; however, the decision whether to use G.729a is always made by the PBX.
- The 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 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 Avaya Communication Manager systems using SIP + 5 Avaya IP Office systems using SIP + 3 Siemens HiPath 4000 systems using Station Set Emulation)
  - SIP timers for Aastra EETS integrations are incompatible with other SIP integrations. Thus, it is not possible to have an EETS integration with any other SIP integration on the Call Server

# Integrating with the SIP Routing Manager

The SIP Routing Manager is a Windows® Service that further enhances the integration capabilities of MiCollab AM to negotiate with the various SIP Servers with which MiCollab AM integrates by providing:

- A seamless SIP integration to multiple Call Servers
- Load Balancing
- Redundancy
- Scalability
- Assistance in locating SIP Servers through DNS (Domain Name Server) and NAPTR (Name Authority Pointer) records
- The use of multiple SIP domains
- Keep Alive messages
- Multiple SIP routes
- Multiple hunt/trunk group numbers
- Multiple SIP integrations

The SIP Routing Manager acts as a redirect server for the Avaya CS1000. It receives the initial SIP messages sent from the PBX and MiCollab AM, redirects the PBX and MiCollab AM messages to the appropriate route or endpoint, and then steps out of the exchange, allowing the PBX and MiCollab AM to negotiate the call connection. When the SIP Routing Manager is running, the SIP Routing Manager redirects all SIP calls within the network.

The SIP Routing Manager Service is an Mitel product. The SIP Routing Manager Service runs on a Windows server platform. It can run on the MiCollab AM System Server (Call Services and Lines are not required to run SIP Routing Manager on the System Server) or a stand-alone Windows server platform with the following caveats:

- The server must be a member of the same network on which the SIP Signaling Server, IP PBX, and MiCollab AM communicate.
- A separate network interface card (NIC) must be installed in the server platform if the SIP integration is on a separate network subnet or VLAN than the primary network connection.

The Avaya CS1000 SIP integration is a SIP trunk integration. MiCollab AM does not register the Avaya SIP ports; the Avaya SIP Server routes calls to a static trunk route directed to MiCollab AM. This type of configuration provides the least amount of integration flexibility because the Avaya NRS cannot route calls to multiple Call Servers and alternate PBX routes do not forward calls in a timely fashion. The CS1000 simply routes the calls based on a single registration or static route and MiCollab AM performs the hunting within a single Call Server.

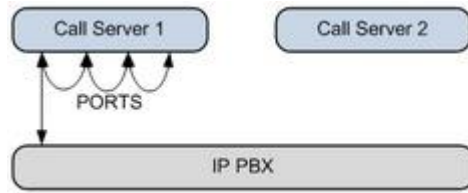


Figure 1. SIP Trunk Integration: MiCollab AM hunting

To integrate the CS1000 to multiple Call Servers you must install and configure the SIP Routing Manager to redirect the SIP calls between the PBX and MiCollab AM.

For more information on Installing and Configuring the SIP Routing Manager, refer to the *SIP Routing Manager Advanced Feature Guide*.

# 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 for CS1000 Release 5.5

- Avaya CS1000 version 4021, release 5.50 Level 4, with SIP Gateway and Converged Desktop (SIP) package 406 and a SIP Access Port License for each MiCollab AM port
- Signaling Server software version SSE-5.50.12
- MC-8, NTVQ01AB VMG Voice Media Gateway Card 8-port assembly; or MC-32, NTVQ01BB VMG Voice Media Gateway Card 32-port Assembly; or MGC, NTDW60BAE5 Media Gateway Controller
- Advanced Networking or higher package (required for Release 3 through Release 5.5)

## Telephone System Requirements for CS1000 Release 6.0 through 7.6

- Avaya CS1000 version 4021, release 6.0 through 7.6, and a SIP Access Port License for each MiCollab AM port
- MC-8, NTVQ01AB VMG Voice Media Gateway Card 8-port assembly; or MC-32, NTVQ01BB VMG Voice Media Gateway Card 32-port Assembly; or MGC, NTDW60BAE5 Media Gateway Controller
- Premium Network Service Package

## MiCollab AM Requirements

- MiCollab AM version 6.1 Mitel software key diskette or feature file with the Avaya Communication Server 1000 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

# Programming the Telephone System

Follow the recommendations and programming examples in this section to program the telephone system for integration with MiCollab AM. Programming examples show commands and parameters that are necessary for integration; they do not represent PBX programming in its entirety.

The installing technician should be familiar with programming the telephone system. For detailed information on programming and installing the telephone system, refer to the *Integrated Services Network, Features Description & Operation Manual*. The Northern Telecom Practice (NTP) library also provides additional information.

## Programming the D-Channel

Configure the D-channel as D-channel over IP Card Type and set the remote capabilities to Network Name Display method 2. In addition, set the MWI using a SIP NOTIFY message. Table 3 provides an example of D-channel programming in LD 17.

Table 3. Example programming for the D-Channel (LD 17)

REQ	CHG
TYPE	ADAN
ADAN	NEW DCH 4
CTYP	DCIP
DES	SIP_PRI
IFC	SL1
RLS	4
RCAP	ND2 MWI

## Programming the Trunk Route

Configure the SIP route, which contains the virtual trunks that act as MiCollab AM lines. Define the protocol ID as SIP and associate the route with the D-channel configured in the previous step. The Node ID should match the node of the Signaling Server. Set the route as incoming and outgoing and assign an access code. To modify the parameters, use the CHG request. Table 4 provides an example of the configuration of route 4 in LD 16.

Table 4. Example programming for the trunk route (LD 16)

REQ	NEW	Comments
TYPE	RDB	
CUST	0	
ROUT	<b>4</b>	
DES	SIP_PRI	
TKTP	TIE	
VTRK	YES	
ZONE	000	
PCID	<b>SIP</b>	
NODE	1110	
ISDN	YES	
MODE	ISLD	
DCH	4	
IFC	SL1	
PNI	00001	Private network identifier (you may change this according to your site's requirements)
NCNA	YES	
NCRD	YES	
TRO	YES	
ICOG	IAO	
TRMB	NO	
ACOD	7003	
CNTL	YES	
NEDC	ETH	
FEDC	ETH	

## Programming SIP Trunks

Configure one IP tie virtual trunk for each MiCollab AM port. Assign the trunks as route members of the route configured in the previous step. Set the Start Arrangement fields to **Immediate** and Supervision Required to **Yes**. Table 5 provides an example of trunk programming in LD 14.



Table 5. Example programming for SIP trunks (LD 14)

REQ	NEW 32 (No. of MiCollab AM lines)
TYPE	IPTI
TN	108 0 1 0
DES	SIP_PRI
CUST	0
RTMB	4 1
CHID	1
STRI	<b>IMM</b>
STRO	<b>IMM</b>
SUPN	<b>YES</b>
CLS	UNR DTN

## Programming the Route List Index

Configure the route list index so it is accessible by the virtual trunk route. Configure any additional site-specific networking features required for the integration. Table 6 provides an example of route list programming in LD 86.

Table 6. Example programming for the route list index (LD 86)

REQ	NEW
CUST	0
FEAT	RLB
RLI	4
ENTR	0
ROUT	4

## Programming the Steering Code

Configure a pilot directory number (PDN) to allow access to the route as if it were an extension, which passes all dialed digits. This number is a steering code that accesses the new route 4. The steering code and route are associated using a route list index, which contains an available route to process the calls. Table 7 provides an example of steering code programming in LD 87.

Table 7. Example programming for a steering code (LD 87)

REQ	NEW
CUST	0
FEAT	CDP
TYPE	DSC
DSC	40
RLI	4

## Programming Customer Data Block (CDB) Network Data

Configure any site specific network settings and options. Table 8 provides an example of network programming in LD 15.

Table 8. Example programming for the CDB network data (LD 15)

REQ	CHG	Comments
TYPE	NET_DATA	
CUST	0	
ISDN	YES	
PNI	1	Private network identifier (you may change this according to your site's requirements)

## Programming Subscriber Telephones

Program subscriber telephones to forward to MiCollab AM on busy and no answer conditions. In the FDN and Hunt field, specify the PDN previously configured. Class of service (CLS) must contain FNA and MWA; enable CFXA for All Call Forwarding. If desired, specify a key as the MWK when programming key assignments. Table 9 provides an example of a 2004P2 set with Key 16 assigned as the MWK function in LD 11.

**NOTE** If you use blind transfers, you must enable HTA. If you use monitored or supervised T-type transfers, do not enable HTA.

Table 9. Example programming for subscriber telephones (LD 11)

REQ	CHG
TYPE	2004P2
TN	112 0 1 0
FDN	4000
HUNT	4000
CLS	FNA HTA MWA CFXA
KEY	16 MWK 4000

# Programming the Release 5.50 Signaling Server for Integration with MiCollab AM

Follow the recommendations and programming examples in this section to program the CS1000 Release 5.50 Signaling Server through the Network Routing Service and the Element Manager for integration with MiCollab AM.

**NOTE** Follow the procedures in this section to program the Signaling Server if you are not implementing the SIP Routing Manager with this integration.

## Configuring the Network Routing Service

In the Network Routing Service (NRS), configure the service level domain, the L1 and L0 domains, the SIP gateway endpoints for MiCollab AM and Call Server and any appropriate routes.

### To configure the NRS:

- 1 Open and sign in to the NRS. Click **NRS Server Settings** on the left menu.
- 2 Type the Host Name and Primary IP (TLAN) in the appropriate boxes, and then click **Save**.
- 3 Click the **Configuration** tab, and then click **Set Standby DB view** at the top of the page.

**IMPORTANT** You must be in standby view to make changes.

- 4 If you are not already in the Service Domain view, click **Service Domains** on the left menu.
- 5 Click **Add**.
- 6 Type in the appropriate service domain name in the Domain name box, and then click **Save**.
- 7 On the left menu, click **L1 Domains (UDP)**.
- 8 Choose the correct Service Domain from the drop-down list, and then click **Add**.
- 9 Type in the appropriate L1 domain name in the Domain name box, and then click **Save**.
- 10 On the left menu, click **L0 Domains (CDP)**.
- 11 Choose the correct domains from the drop-down lists, and then click **Add**.
- 12 Type in the appropriate L0 domain name in the Domain name box, and then click **Save**.
- 13 On the left menu, click **Gateway Endpoints**.
- 14 Type the Call Server endpoint name and the static endpoint address in their appropriate boxes.
- 15 From the SIP Support list, select **Static SIP endpoint**.

- 16 From the SIP transport list, select **UDP**, and then click **Save**.
- 17 On the left menu, click **Routing Entries**.
- 18 Click **Lookup**, and then click **Search** in the new window.
- 19 Select the Call Server gateway endpoint, and then click **Add**.
- 20 From the DN Type list, select **Level 0 regional**.
- 21 In their appropriate boxes, type the MiCollab AM DN Prefix, enter 1 for Route Cost, and then click **Save**.

**IMPORTANT** The DN prefix for the Call Server must be unique from any other DNs in the system.

- 22 To export these changes to the switch click the **Tools** tab.
- 23 On the left menu, click **Database Actions**.
- 24 From the Database Actions list select **Cut over & commit**, and then click **Submit**.
- 25 Continue with programming the Element Manager in the section, *Configuring the Element Manager*.

# Programming the Release 5.50 Signaling Server for Integration with MiCollab AM and the SIP Routing Manager

Follow the recommendations and programming examples in this section to program the CS1000 Release 5.50 Signaling Server through the Network Routing Service and the Element Manager for integration with MiCollab AM using the SIP Routing Manager to redirect SIP traffic.

## Configuring the Network Routing Service

In the Network Routing Service (NRS), configure the service level domain, the L1 and L0 domains, the SIP gateway endpoints for MiCollab AM and Call Server and any appropriate routes.

### To configure the NRS:

- 1 Open and sign in to the NRS. Click **NRS Server Settings** on the left menu.
- 2 Type the Host Name and Primary IP (TLAN) in the appropriate boxes, and then click **Save**.
- 3 Click the **Configuration** tab, and then click **Set Standby DB View** at the top of the page.

**IMPORTANT** You must be in standby view to make changes.

- 4 If you are not already in the Service Domain view, click **Service Domains** on the left menu.
- 5 Click **Add**.
- 6 Type in the appropriate service domain name in the Domain name box, and then click **Save**.
- 7 On the left menu, click **L1 Domains (UDP)**.
- 8 Choose the correct Service Domain from the drop-down list, and then click **Add**.
- 9 Type in the appropriate L1 domain name in the Domain name box, and then click **Save**.
- 10 On the left menu, click **L0 Domains (CDP)**.
- 11 Choose the correct domains from the drop-down lists, and then click **Add**.
- 12 Type in the appropriate L0 domain name in the Domain name box, and then click **Save**.

### To add a Gateway Endpoint for the SIP Routing Manager:

- 1 On the left menu, click **Gateway Endpoints**.

- 2 Type the SIP Routing Manager endpoint name and the static endpoint IP address in their appropriate boxes.

**For example:**

SIPRoutingMgr, 192.168.1.100

- 3 From the SIP Support list, select **Static SIP endpoint**.
- 4 From the SIP transport list, select **UDP**, and then click **Save**.

### To add a Gateway Endpoint for the MiCollab AM Call Servers:

- 1 On the left menu, click **Gateway Endpoints**.
- 2 Type the Call Server endpoint name and the static endpoint IP address in their appropriate boxes.  
**For example:**  
CallSrvr1, 192.168.1.101
- 3 From the SIP Support list, select **Static SIP endpoint**.
- 4 From the SIP transport list, select **UDP**, and then click **Save**.
- 5 Repeat steps 1 through 4 for each Call Server in the system.

**IMPORTANT** Each Call Server must have a unique endpoint name and IP Address.

### To add a DN prefix for the SIP Routing Manager:

- 1 On the left menu, click **Routing Entries**.
- 2 Click **Lookup**, and then click **Search** in the new window.
- 3 Select the gateway endpoint that points to the SIP Routing Manager, and then click **Add**.
- 4 From the DN Type list, select **Level 0 regional**.
- 5 In their appropriate boxes, type your **DN Prefix**, enter 1 for Route Cost, and then click **Save**.

**NOTE** The DN prefix for the SIP Routing Manager routes all calls destined to the Call Servers to the SIP Routing Manager first. The SIP Routing Manager redirects the calls to the Call Servers.

### To add a DN prefix for the MiCollab AM Call Servers:

- 1 On the left menu, click **Routing Entries**.
- 2 Click **Lookup**, and then click **Search** in the new window.
- 3 Select the gateway endpoint that points to the appropriate Call Server, and then click **Add**.
- 4 From the DN Type list, select **Level 0 regional**.
- 5 In their appropriate boxes, type the **DN Prefix**, enter 1 for Route Cost, and then click **Save**.

**IMPORTANT** The DN prefix for each MiCollab AM Call Server must be unique from any other DNs in the system, including other Call Servers and the SIP Routing Manager.

- 6 Repeat steps 1 through 5 for each Call Server in the system.

### To complete the NRS configuration:

- 1 To export these changes to the switch click the **Tools** tab.
- 2 On the left menu, click **Database Actions**.
- 3 From the Database Actions list select **Cut over & commit**, and then click **Submit**.
- 4 Continue with programming the Element Manager in the section, *Configuring the Element Manager*.



# Configuring the Element Manager

In the Element Manager, configure the node.

## To configure the Element Manager:

- 1 Open, and then sign in to the Element Manager.
- 2 Click IP Network, select Nodes: Servers, and then select Media Cards.
- 3 Click Edit.
- 4 In the list, click SIP GW Settings.
- 5 Under SIP GW Settings, type the TLAN IP address in the Primary Proxy or Re-direct (TLAN) IP address box.
- 6 On the same page, click Signaling Servers, and then click Signaling Server Properties.
- 7 Change Enable IP Peer Gateway (Virtual Trunk TPS) to SIP Only and then enter the following required information:
  - Embedded LAN (ELAN) IP address
  - Embedded LAN (ELAN) MAC address
  - Telephony LAN (TLAN) IP address
  - Hostname
- 8 On the same page, click SIP URI Map, and then enter the following required information:
  - Public E.164/National domain name
  - Public E.164/Subscriber domain name
  - Private/UDP domain name
  - Private/CDP domain name
- 9 Click Save and Transfer, and then click OK to confirm your changes.

# Programming the Release 6.0 through 7.5 Signaling Server for integration with MiCollab AM

Follow the recommendations and programming examples in this section to program the CS1000 Release 6.0 Signaling Server through the Network Routing Service and the Element Manager in UCM for integration with MiCollab AM.

## Configuring the Network Routing Service

In the Network Routing Service (NRS), configure the service level domain, the L1 and L0 domains, the SIP gateway endpoints for MiCollab AM, the Avaya Call Server and any appropriate routes.

### To configure the NRS:

- 1 Open the NRS Manager Navigator. Click **NRS Server** on the left menu.
- 2 In the Server Configuration window, Type the Host Name and the Primary IP (TLAN) in the appropriate boxes, and then enable the **UDP Transport**.
- 3 On the left menu of the NRS, select **Numbering Plans > Domains**, and then click **Standby database** at the top of the page.

**IMPORTANT** You must be in standby view to make changes.

- 4 Click **Add**.
- 5 Type in the appropriate service domain name in the Domain name box, and then click **Save**.
- 6 Click **L1 Domains (UDP)**.
- 7 Choose the correct Service Domain from the drop-down list, and then click **Add**.
- 8 Type in the appropriate L1 domain name in the Domain name box, and then click **Save**.
- 9 Click **L0 Domains (CDP)**.
- 10 Choose the correct domains from the drop-down lists, and then click **Add**.
- 11 Type in the appropriate L0 domain name in the Domain name box, and then click **Save**.
- 12 On the left menu, click **Numbering Plans > Endpoints**.
- 13 Type the Call Server endpoint name and static endpoint address in their appropriate boxes.
- 14 From the SIP Support list, select **Static SIP endpoint**.

- 15 Select the SIP UDP transport enable check box, and then click **Save**.
- 16 On the left menu, click **Routes**.
- 17 Select the Call Server gateway endpoint, and then click **Add**.
- 18 From the DN Type list, select **Level 0 regional**.
- 19 In their appropriate boxes, type your DN Prefix, enter 1 for Route Cost, and then click **Save**.

**IMPORTANT** The DN prefix for the Call Server must be unique from any other DNs in the system.

- 20 To export these changes to the switch select **System > Database**.
- 21 Click the Cut over button, and then perform a database **Commit Submit**.
- 22 Continue with programming the Element Manager in the section, *Configuring the Element Manager*.

# Programming the Release 6.0 through 7.5 Signaling Server for use with MiCollab AM and the SIP Routing Manager

Follow the recommendations and programming examples in this section to program the CS1000 Release 6.0 Signaling Server through the Network Routing Service and the Element Manager in UCM for integration with MiCollab AM.

## Configuring the Network Routing Service

In the Network Routing Service (NRS), configure the service level domain, the L1 and L0 domains, the SIP gateway endpoints for MiCollab AM, the Avaya Call Server and any appropriate routes.

### To configure the NRS:

- 1 Open the NRS Manager Navigator. Click NRS Server on the left menu.
- 2 In the Server Configuration window, Type the Host Name and the Primary IP (TLAN) in the appropriate boxes, and then enable the UDP Transport.
- 3 On the left menu of the NRSM, select Numbering Plans > Domains, and then click Standby database at the top of the page.

**IMPORTANT** You must be in standby view to make changes.

- 4 Click **Add**.
- 5 Type in the appropriate service domain name in the Domain name box, and then click **Save**.
- 6 Click **L1 Domains (UDP)**.
- 7 Choose the correct Service Domain from the drop-down list, and then click **Add**.
- 8 Type in the appropriate L1 domain name in the Domain name box, and then click **Save**.
- 9 Click **L0 Domains (CDP)**.
- 10 Choose the correct domains from the drop-down lists, and then click **Add**.
- 11 Type in the appropriate L0 domain name in the Domain name box, and then click **Save**.

### To add a Gateway Endpoint for the SIP Routing Manager:

Programming the Release 6.0 through 7.5 Signaling Server for use with MiCollab AM and the SIP Routing Manager

- 1 On the left menu, click Numbering Plans > Endpoints.
- 2 Type the SIP Routing Manager endpoint name and the static endpoint IP address in their appropriate boxes.  
**For example:**  
SIPRoutingMgr, 192.168.1.100
- 3 From the SIP Support list, select Static SIP endpoint.
- 4 Select the SIP UDP transport enable check box, and then click Save.

### To add a Gateway Endpoint for the MiCollab AM Call Servers:

- 1 On the left menu, click Numbering Plans > Endpoints.
- 2 Type the Call Server endpoint name and the static endpoint IP address in their appropriate boxes.  
**For example:**  
CallSrvr1, 192.168.1.101
- 3 From the SIP Support list, select Static SIP endpoint.
- 4 Select the SIP UDP transport enable check box, and then click Save.
- 5 Repeat steps 1 through 4 for each Call Server in the system.

**IMPORTANT** Each Call Server must have a unique endpoint name and IP Address.

### To add a DN prefix for the SIP Routing Manager:

- 1 On the left menu, click Routes.
- 2 Select the SIP Routing Manager gateway endpoint, and then click Add.
- 3 From the DN Type list, select Level 0 regional.
- 4 In their appropriate boxes, type your DN Prefix, enter 1 for Route Cost, and then click Save.

**NOTE** The DN prefix for the SIP Routing Manager routes all calls destined to the Call Servers to the SIP Routing Manager first. The SIP Routing Manager redirects the calls to the Call Servers.

### To add a DN prefix for the MiCollab AM Call Servers:

- 1 On the left menu, click Routes.
- 2 Select the Call Server gateway endpoint, and then click Add.
- 3 From the DN Type list, select Level 0 regional.
- 4 In their appropriate boxes, type your DN Prefix, enter 1 for Route Cost, and then click Save.

**IMPORTANT** The DN prefix for each MiCollab AM Call Server must be unique from any other DNs in the system, including other Call Servers and the SIP Routing Manager.

- 5 Repeat steps 1 through 5 for each Call Server in the system.

### To complete the NRS configuration:

- 1 To export these changes to the switch select System > Database.
- 2 Click the Cut over button, and then perform a database Commit Submit.
- 3 Continue with programming the Element Manager in the section, *Configuring the Element Manager*.

# Configuring the Element Manager

In the Element Manager, configure the node.

## To configure the Element Manager:

- 1 Open and sign in to the Element Manager.
- 2 Click **System > IP Network > Nodes: Servers, Media Cards**. The IP Telephony Nodes page displays.
- 3 Select the link for the Node ID to open the Node Details page.
- 4 Enter the Embedded LAN (ELAN) IP address and the Telephony LAN (TLAN) IP address as required.
- 5 Click **Gateway**.
- 6 Under the General tab, select **Vtrk Gateway Application as SIP Gateway**, and then add the SIP domain name.
- 7 On the same page, click the **SIP GW Settings** tab.
- 8 Under SIP GW Settings, type the TLAN IP address in the Primary TLAN IP Address box.
- 9 On the same page, enter the following required information in the SIP URI Map section.
  - Public E.164/National domain name
  - Public E.164/Subscriber domain name
  - Private/UDP domain name
  - Private/CDP domain name
- 10 Click **Save and Transfer** to confirm your changes.

# 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 **Avaya**.
  - d From the **Model** dropdown list, select **Communication Manager**.
  - 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 following options:

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

Table 10. Required Parameters for Integrations Options

Field	Value
SIP Server Address	<b>If...</b>
	The SIP Routing Manager is <b>not</b> installed
	The SIP Routing Manager is installed
SIP Server Port	<b>Then...</b>
	Enter the Avaya Signaling Server IP address
	Enter the Sip Routing Manager IP address
SIP Server Port	<b>If...</b>
	The SIP Routing Manager is <b>not</b> installed
	The SIP Routing Manager is installed
SIP Server Port	<b>Then...</b>
	Enter the listen port configured on the Avaya Signaling Server
	Enter the listen port configured
The default port number is <b>5060</b> .	
SIP Domain Name	Enter the service domain name configured on the Signaling Server.
	<b>NOTE</b> This value is case-sensitive.
SIP parser qualifier string	In cases of a single SIP integration on the call server, enter the local IP address to which the integration is bound. This field is used by MiCollab AM to match SIP packets to the appropriate SIP integration.
	In cases where there are multiple SIP integrations on the call server, use a string that is unique to each SIP integration.
	<b>For example:</b>

	<p>The extension that will be used as the hunt number on the PBX followed by the @ symbol and the IP of the call server, such as 5000@172.16.4.202. The hunt number must be unique across all IP integrations.</p> <p>The Fully Qualified Domain Name (FQDN) of the switch, such as pbx1.sipdomain.com.</p>
	<p><b>NOTE</b> This setting must match a string in the SIP header that is unique to this particular integration.</p>
Transport for outgoing SIP messages	Depending on the telephone system's configuration, enter UDP or TCP. The default is UDP
Local IP Address to bind on	Enter the IP address of the NIC on the MiCollab AM platform that connects to the Avaya Signaling Server and the Avaya Call Server
SIP Local Connection Port	Enter the TCP port MiCollab AM listens for incoming SIP messages. The default port is 5060
Media Packet Size (milliseconds)	MiCollab AM sends/receives packets containing the number of milliseconds worth of audio data set here. The default value is 20.

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

- Select the **Validate Remote Hosts for Media** check box.
- 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 Hung Mode** field, select 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, type the access code. This number must match the DN prefix programmed for the Gateway Endpoint associated with this MiCollab AM Call Server.

**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 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 displays.
  - a From the **Manufacturer** dropdown list, select **Avaya**.
  - b From the **Model** dropdown list, select **Communication Manager**.
  - c From the **Integration Type** dropdown list, select **SIP 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 following options:
  - a In the **Local Integration Settings** section, select the **Required Parameters** view, and configure the following parameters:

Table 11. Required Parameters for Integrations Options

Field	Value
SIP Server Address	<b>If...</b>
	The SIP Routing Manager is <b>not</b> installed
	The SIP Routing Manager is installed
SIP Server Port	<b>Then...</b>
	Enter the Avaya Signaling Server IP address
	Enter the Sip Routing Manager IP address
SIP Server Port	<b>If...</b>
	The SIP Routing Manager is <b>not</b> installed
	The SIP Routing Manager is installed
SIP Server Port	<b>Then...</b>
	Enter the listen port configured on the Avaya Signaling Server
	Enter the listen port configured
The default port number is <b>5060</b> .	
SIP Domain Name	Enter the service domain name configured on the Signaling Server.
	<b>NOTE</b> This value is case-sensitive.
SIP parser qualifier string	<p>In cases of a single SIP integration on the call server, enter the local IP address to which the integration is bound. This field is used by MiCollab AM to match SIP packets to the appropriate SIP integration.</p> <p>In cases where there are multiple SIP integrations on the call server, use a string that is unique to each SIP integration.</p> <p><b>For example:</b></p> <p>The extension that will be used as the hunt number on the PBX followed by the @ symbol and the IP of the call server, such as 5000@172.16.4.202. The hunt number must be unique across all IP integrations.</p> <p>The Fully Qualified Domain Name (FQDN) of the switch, such as pbx1.sipdomain.com.</p> <p><b>NOTE</b> This setting must match a string in the SIP header that is unique to this particular integration.</p>
Transport for outgoing SIP messages	Depending on the telephone system's configuration, enter UDP or TCP. The default is <b>UDP</b> .
Local IP Address to bind on	Enter the IP address of the NIC on the MiCollab AM platform that connects to the Avaya Signaling Server and the Avaya Call Server
SIP Local Connection Port	Enter the TCP port MiCollab AM listens for incoming SIP messages. The default port is <b>5060</b> .

Media Packet Size (milliseconds)	MiCollab AM sends/receives packets containing the number of milliseconds worth of audio data set here. The default value is <b>20</b> .
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- 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** check box.
  - 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 Hung Mode** field, select 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, type the access code. This number must match the DN prefix programmed for the Gateway Endpoint associated with this MiCollab AM Call Server.
  - 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 12. 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><b>For example:</b> The IP address 123.45.6.789 as displayed on the Review/Modify SIP Gateway screen.</p> <p><b>NOTE</b> This integration requires the machine name to be a fully qualified domain name. Therefore, use the Machine Name field as displayed on the Review/Modify SIP Gateway screen during the integration process.</p> <p><b>IMPORTANT</b> This value must match the configuration on the Gateway of the secondary node.</p>
Secondary SIP Server Port	<p>Enter the port number of the secondary node. The default value is <b>5060</b>.</p>

- 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 13. Integration Specific Parameters

Field	Value
Enable SIP server failover	Select this check box to allow for failover and to enable the failover server setting changes.
Delay (in ms) between Failover attempts	The delay in milliseconds before MiCollab AM attempts to register its port with the SIP server. The default is <b>1000</b> ms.
Incoming off hook delay	800
Outgoing off hook delay	0

On hook delay	300
Type of Call Progress to use for External Calls	<p>How this should be set depends on the gateway used for the integration.</p> <ul style="list-style-type: none"> <li>• If the gateway supports call progress through to the endpoint, set to <b>Digital</b>.</li> <li>• If the gateway reports early that the call is connected, such as before the phone rings or while the phone is ringing, set to <b>Media</b>.</li> </ul>

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

### To remove a SIP Failover Server:

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

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

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

# Changing the Network Binding Order on the MiCollab AM Platform

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 14. 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