



Configuring SIP Connectivity between the Avaya Meeting Exchange Enterprise S6200 Conferencing Server R5.2 and Cisco Unified Communications Manager R7.0 – Issue 1.0

Abstract

These Application Notes present the procedures for configuring SIP connectivity between the Avaya Meeting Exchange Enterprise S6200 Conferencing Server and Cisco Unified Communications Manager. SIP connectivity is enabled via directly connected SIP trunking between Avaya Meeting Exchange Enterprise and Cisco Unified Communications Manager.

Testing was conducted via the Internal Interoperability Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes present a sample configuration for a network that uses Avaya Meeting Exchange Enterprise S6200 Conferencing Server (MX S6200) and Cisco Unified Communications Manager using SIP trunks. The sample configuration shown in **Figure 1** was used to compliance test Cisco Unified Communications Manager and Cisco 2811 MGCP Gateway interoperability with Avaya Meeting Exchange Enterprise S6200.

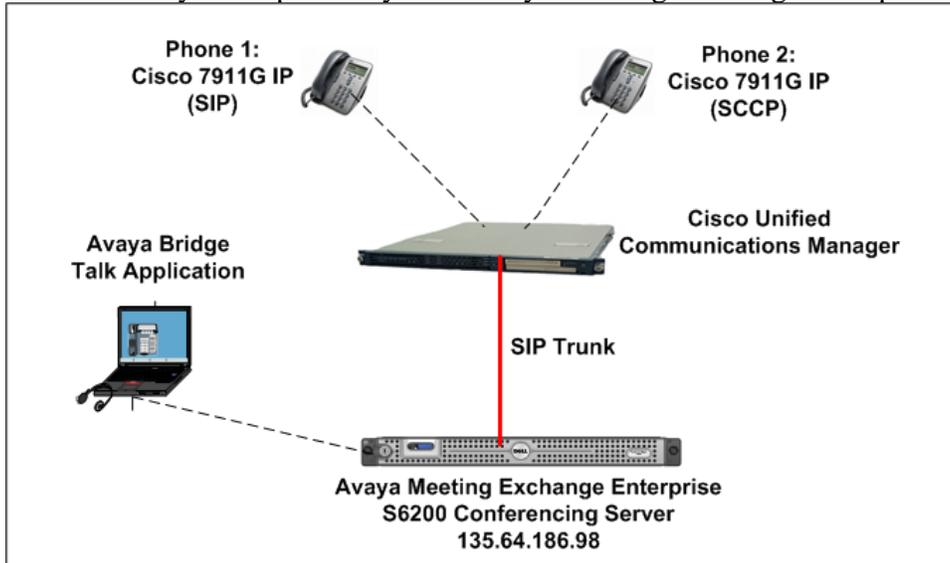


Figure 1 - Avaya Meeting Exchange Enterprise Interop Network Topology

The configuration in **Figure 2** was used to compliance test Cisco Unified Communications Manager interoperability with the Distributed MX S6200 system. The Cisco Unified Communications Manager supports the Cisco 7911G IP Telephone (SIP) and the Cisco 7911G IP Telephone (SCCP).

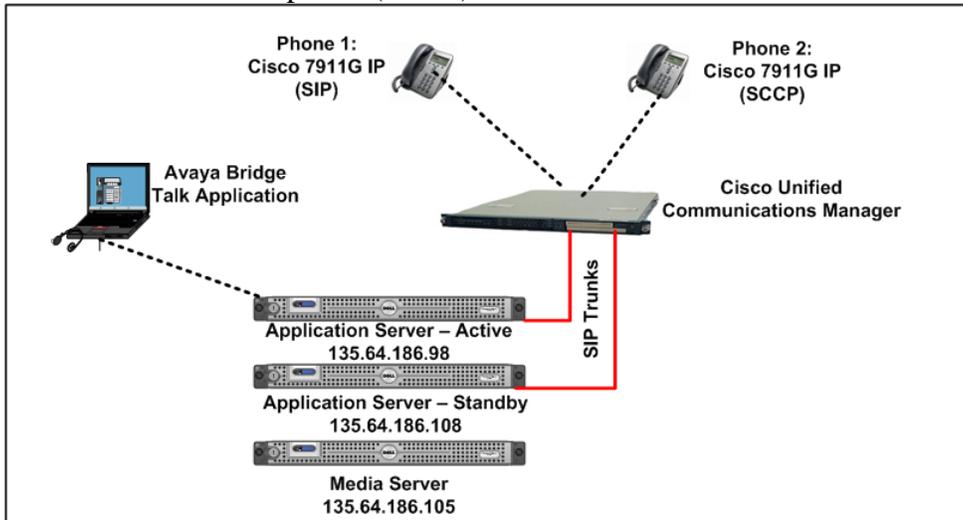


Figure 2 – Distributed Avaya Meeting Exchange Interop Network Topology

2. Equipment and Software Validated

The following equipment and software versions were used for the sample configuration provided in these Application Notes.

Equipment	Software
Avaya S6200 server	Avaya Meeting Exchange Enterprise Edition R5.2 (Build 5.2.0.0.22 + Patch 5.2.0.1.4)
Windows Computer	Avaya Bridge Talk (BT) 5.2.0.0.7
Cisco Unified Communications Manager	7.0.2.100000-18
Cisco 7911G SIP Telephone	SIP 11.8-4-3S
Cisco 7911G SCCP Telephone	SCCP 11.8-3-4SR1S

Table 1: Equipment and Software Versions

3. Configure Avaya Meeting Exchange Enterprise S6200 Conferencing Server

This section describes the steps for configuring the Avaya Meeting Exchange Enterprise S6200 to interoperate with Cisco Unified Communications Manager via SIP trunking. It is assumed that the Meeting Exchange is installed and licensed as described in the product documentation (see reference [1]). The following steps describe the administrative procedures for configuring the Meeting Exchange:

- Configure SIP Connectivity
- Configure Dialout
- Map DNIS Entries
- Configure Audio Preferences
- Configure Application Server
- Configure Bridge Talk

The following instructions require logging in to the Meeting Exchange console using an ssh connection to access the Command Line Interface (CLI) with the appropriate credentials.

3.1. Configuring SIP Connectivity

Log in to the Meeting Exchange server console using ssh (PuTTY) to access the Command Line Interface (CLI) with the appropriate credentials. Configure settings that enable SIP connectivity between the Meeting Exchange server and other devices by editing the **system.cfg** file as follows:

- Edit **/usr/ipcb/config/system.cfg**
 - Add Meeting Exchange S6200 server IP address (**Figure 1**)
 - **IPAddress=(135.64.186.98)**
 - Depending on the SIP signalling protocol, TCP or UDP, add one of the following lines to populate the From Header Field in SIP INVITE messages:
 - **MyListener=<sip:6000@135.64.186.98:5060;transport=tcp>**
 - **MyListener=<sip:6000@135.64.186.98:5060;transport=udp>**
- Note:** The user field 6000, defined for this SIP URI must conform to RFC 3261. For consistency, it is selected to match the user field provisioned for the **respContact** entry (see below).
- Depending on the SIP signalling protocol, TCP or UDP , add one of the following lines to provide SIP Device Contact address to use for acknowledging SIP messages from the Meeting Exchange server:
 - **respContact=<sip:6000@135.64.186.98:5060;transport=tcp>**
 - **respContact=<sip:6000@135.64.186.98:5060;transport=udp>**
 - Add the following lines to set the Min-SE timer to **900** seconds in SIP INVITE messages from the Meeting Exchange server:
 - **sessionRefreshTimerValue= 900**
 - **minSETimerValue= 900**

3.2. Configure Dialout

To enable Dial-Out from the Meeting Exchange to the Cisco Unified Communications Manager, edit the **telnumToUri.tab** file as follows:

- Edit **/usr/ipcb/config/telnumToUri.tab** file with a text editor
- Add the following line to the file to route outbound calls from the Meeting Exchange to the Cisco Unified Communications Manager
6000 sip:\$1@10.10.9.80:5060;transport=tcp

3.3. Map DNIS Entries

To map DNIS entries, run the **cbutil** utility on Meeting Exchange. Log in to the Meeting Exchange with an ssh connection using PuTTY with the appropriate credentials. Enable Dial-In access (via passcode) to conferences provisioned on the Meeting Exchange as follows:

- Add a DNIS entry for a **scan call function** corresponding to DID **1111** by entering the following command at the command prompt:
cbutil add <dnis> <rg> <msg> <ps> <ucps> <func> [-o <of> -l <ln> -c <cn> -crs <n> -cre <n> -cc <code>]

where the variables for add command is defined as follows:

- o **<dnis>** DNIS
- o **<rg>** Reservation Group
- o **<msg>** Annunciator message number
- o **<ps>** Prompt Set number (0-20)
- o **<ucps>** Use Conference Prompt Set (y/n)
- o **<func>** One of: DIRECT/SCAN/ENTER/HANGUP/AUTOVL/FLEX
- o **-o <of>** Optional On-failure function – one of: ENTER/HANGUP
- o **-l <"ln">** Optional line name to associate with caller
- o **-c <"cn">** Optional company name to associate with caller
- o **-crs <n>** Optional conference room start number
- o **-cre <n>** Optional conference room end number

In this sample configuration, the DNIS entry for a **scan call function** was added corresponding to DNIS 11111 by entering the following command at the command prompt:

```
[MXSIL]# cbutil add 11111 0 247 1 N SCAN
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.
```

At the command prompt, enter **cbutil list** to verify the DNIS entries provisioned.

```
[MXSIL]# cbutil list
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.

DNIS   Grp Msg PS   CP Function On Failure Line Name Company Name Room Start
Room End
-----
-----
11111           0   247 1   N  SCAN      DEFAULT
```

3.4. Configure Audio Preferences file

The **audioPreference.cfg** file located at **/usr/ipcb/config/** specifies the order in which codecs are offered in the Session Description Protocol.

```
# audioPreferences.cfg
# This table is an ordered list of MIME subtypes specifying the codecs
supported
# by this media server. The list is specified in the order in which an SDP
offer
# will list the various MIME subtypes on the m=audio line.
# For static payload type numbers (i.e. numbers between 0 - 96) please use the
# iana registered numbering scheme.
# See: http://www.iana.org/assignments/rtp-parameters
mimeSubtype      payloadType
PCMU              0
PCMA              8
G722              9
G729              18
iLBC30            97
iLBC20            98
wbPCMU            102
wbPCMA            103
telephone-event  120
iSAC              104
G726_16           105
G726_24           106
G726_32           107
G726_40           108
```

3.5. Configure Application Server

To configure the Meeting Exchange server, edit the **processTable.cfg** file as follows:

- Edit the **/usr/ipcb/config/processTable.cfg** file with a text editor.
- Configure the file using the IP address of Application Server 1 and Media Server.

This applies to the configuration in **Figure 2**.

processName	ipcKeyNumber	autoStart	ProcessExe	ipAddress	route	ProcessArgs
initipcb	100	0	noexecute	0.0.0.0		
bridget700	102	0	noexecute	0.0.0.0		
commsProcess	101	1	/usr/dcb/bin/serverComms	0.0.0.0		
sipAgent	131	1	/usr/dcb/bin/sipagent	<135.64.186.98>		
msDispatcher	132	1	/usr/dcb/bin/msdispatcher	<135.64.186.98>		
mediaServer	120	1	/usr/dcb/bin/msInterface	<135.64.186.98>		
mediaServer	121	1	/usr/dcb/bin/msInterface	<135.64.186.98>		1
mediaServer	122	1	/usr/dcb/bin/msInterface	<135.64.186.98>		2
mediaServerExt	140	1	/usr/dcb/bin/softms	<135.64.186.105>		3
mediaServerExt	141	1	/usr/dcb/bin/softms	<135.64.186.105>		1
mediaServerExt	142	1	/usr/dcb/bin/softms	<135.64.186.105>		2
			appEvents/msDispatcher, netEvents/msDispatcher			3

3.6. Bridge Talk

The following steps utilize the Avaya Bridge Talk application to provision a sample conference on the Meeting Exchange. This sample conference enables both Dial-In and Dial-Out access to audio conferencing for endpoints on the Public Switched Telephone Network.

Note: If any of the features displayed in the Avaya Bridge Talk screen captures are not present, contact an authorized Avaya Sales representative to make the appropriate changes.

3.6.1. Initializing Bridge Talk

Invoke the Avaya Bridge Talk application as follows:

- Double-click on the desktop icon from a Personal Computer loaded with the Avaya Bridge Talk application and with network connectivity to the Meeting Exchange (Not shown).
- Enter the appropriate credentials in the **Sign-In** and **Password** fields.
- Enter the IP address of the Meeting Exchange server (**135.64.186.98** for this sample configuration) in the **Bridge** field as shown below.

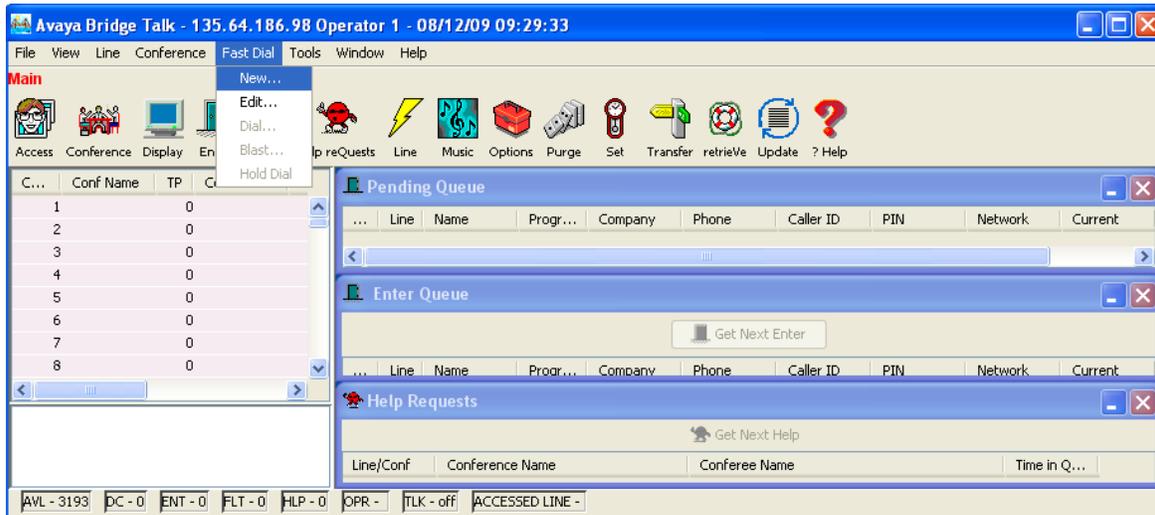


The image shows a screenshot of the 'Avaya Bridge Talk login' dialog box. The dialog has a blue title bar with the text 'Avaya Bridge Talk login' and a red close button. The main area is light gray and contains four input fields: 'Sign-In' with the text 'user', 'Password' with four black dots, 'Bridge' with the IP address '135.64.186.98' and a dropdown arrow, and 'Operator' with the text 'Next available' and a dropdown arrow. At the bottom, there are two buttons: 'OK' and 'Exit'.

3.6.2. Creating a Dial Out list

Provision a dial list that is utilized for Dial-Out (e.g., Blast dial and Fast dial) from the Meeting Exchange.

- From the Avaya Bridge Talk Menu Bar, click **Fast Dial** → **New**.



3.6.3. Creating a Dial List

From the **Dial List Editor** window that is displayed below:

- Enter a descriptive label in the **Name** field.
- Enable conference participants on the dial list to enter the conference without a passcode by selecting the **Directly to Conf** box as displayed.
- Add entries to the dial list by clicking on the **Add** button and enter **Name**, **Company** and **Telephone** number for dial out for each participant. [Optional] Moderator privileges may be granted to a conference participant by checking the **Moderator** box.

When finished, click on the **Save** button on the bottom of the screen.

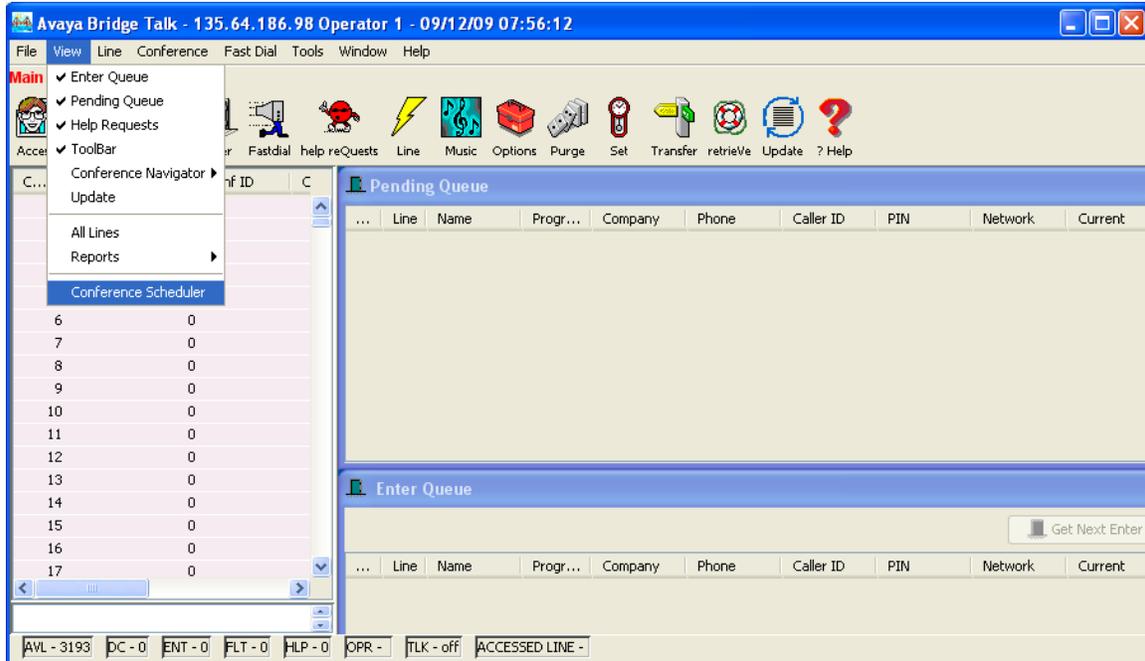
The screenshot shows the 'Dial List Editor' window. At the top, there are fields for 'Name' (containing 'blast'), 'Optional Access Code' (containing '1000000000'), and a checked 'Directly to Conf' checkbox. Below this is a section titled 'Conferee List' with a checked 'Display As Entered' checkbox and 'Add' and 'Remove' buttons. A table with the following data is displayed:

Name	Company	Moderator	Q&A Priority	Telephone
Phone1	Avaya	<input type="checkbox"/>		6002
Phone2	Avaya	<input type="checkbox"/>		6010

At the bottom of the window are 'Save', 'Cancel', 'Print', and 'Help' buttons.

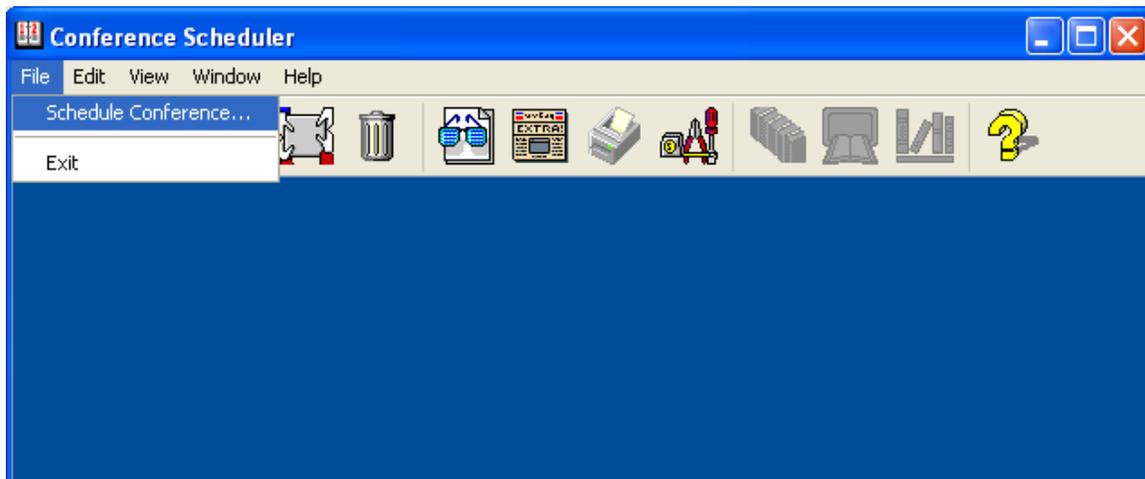
3.6.4. Conference Scheduler

From the **Avaya Bridge Talk** menu bar, click **View** → **Conference Scheduler** to provision a conference.



3.6.5. Scheduling a Conference

From the **Conference Scheduler** window, click **File** → **Schedule Conference**.



3.6.6. Provision a Conference

From the **Schedule Conference** window that is displayed, provision a conference as follows:

- Enter a unique **Conferee Code** to allow participants access to this conference.
- Enter a unique **Moderator Code** to allow participants access to this conference with moderator privileges.
- Enter a descriptive label in the **Conference Name** field.
- Administer settings to enable an **Auto Blast** dial by setting Auto/Manual as desired.

Select a dial list by clicking on the **Dial List** button, select a dial list from the **Create, Select or Edit Dial List** window that is displayed (not shown), and click on the **Select** button (to verify Dial out and Blast Dial out).

- When finished, click on the **OK** button on the bottom of the screen.

Schedule Conference [Administrator Access]

Conference Information

Status: Mode: Conference Type:
Confirmation No.: Conference ID: Weekend:
Name: Billing Code Prompt:
Telephone: Accounting Code: Start Date (dd/mm/yyyy):
Sign-in Name: Security Passcode: End Date (dd/mm/yyyy):
Res Group: Change Conf Opt:
Conferee Code: Op Help Available: Name Record/Play:
Moderator Code: Block Dialout: NRP Annunciator:
Conference Name: Auto Blast: PIN Mode:
 Blast Annunciator: PIN List:

Conference Features

Start Time: End Time: Code Duration:
Entry Tone: Exit Tone: Maximum Lines:
Hang up: Music: Security:
Auto Extend Duration: Auto Extend Ports:
Prompt Set: Conference Viewer:

4.0. Configure Cisco Unified Communications Manager

This section provides the procedures for configuring Cisco Unified Communications Manager. These Application Notes assume that the basic configuration needed to support Cisco IP telephones has been completed. For further information on Cisco Unified Communications Manager, please consult **References** [3] and [4]. The procedures include configuration of the following items:

- Log in to Cisco Unified Communications Manager
- Administer SIP Trunk Security Profile
- Administer SIP Trunk
- Administer Route Pattern
- Administer Route Group
- Administer Phone

4.1. Log in to Cisco Unified Communications Manager

Open the Cisco Unified Communications Manager Administration web interface by using the URL “<http://<ip-address>>” in an Internet browser window, where “<ip-address>” is the IP address of the Cisco Unified Communications Manager. Click on **Cisco Unified Communications Manager Administration** at the bottom of the screen.

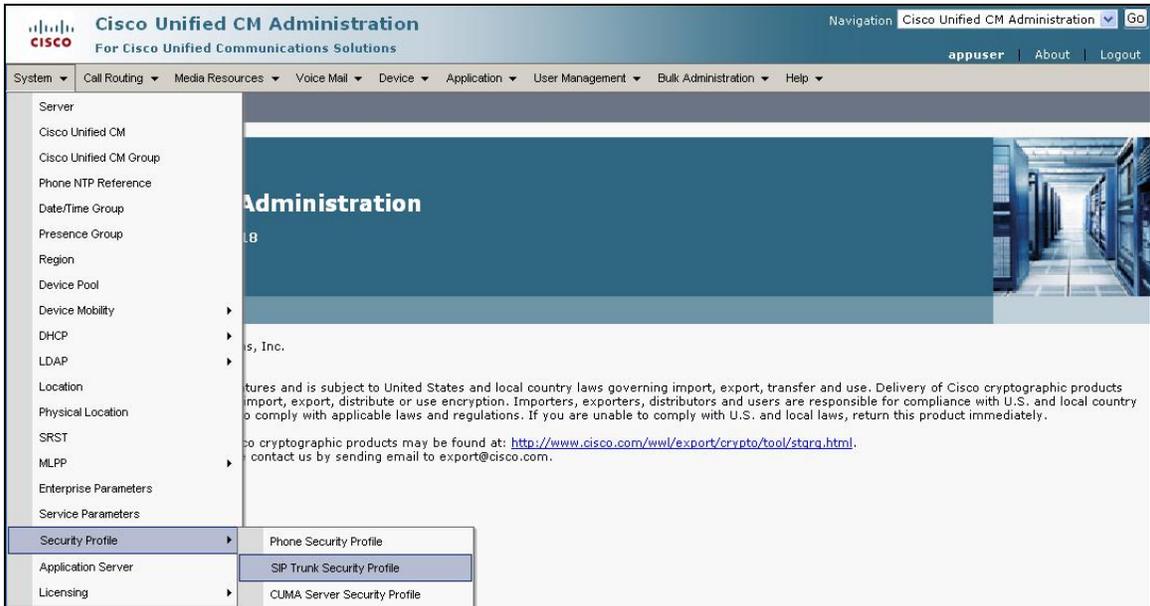


The **Cisco Unified CM Administration** screen is displayed. Select **Cisco Unified CM Administration** from the **Navigation** drop-down list, and log in with appropriate credentials.



4.2. Administer SIP Trunk Security Profile

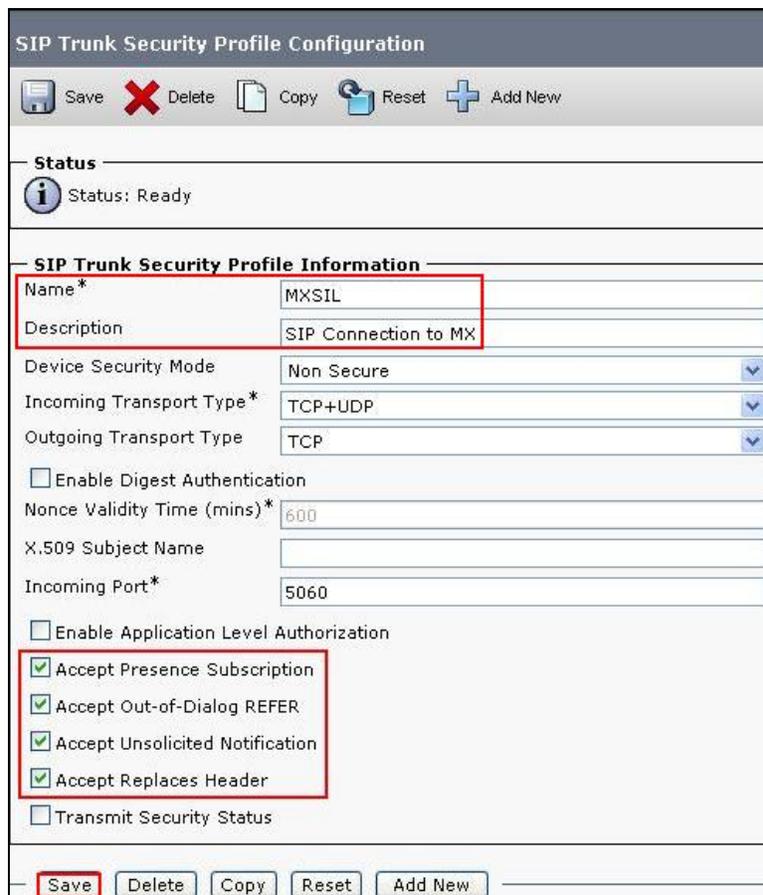
Scroll to the top of the screen, and select **System** → **Security Profile** → **SIP Trunk Security Profile** as shown below.



The **SIP Trunk Security Profile** screen is displayed. Click **Add New** to add a new SIP Trunk Security Profile.

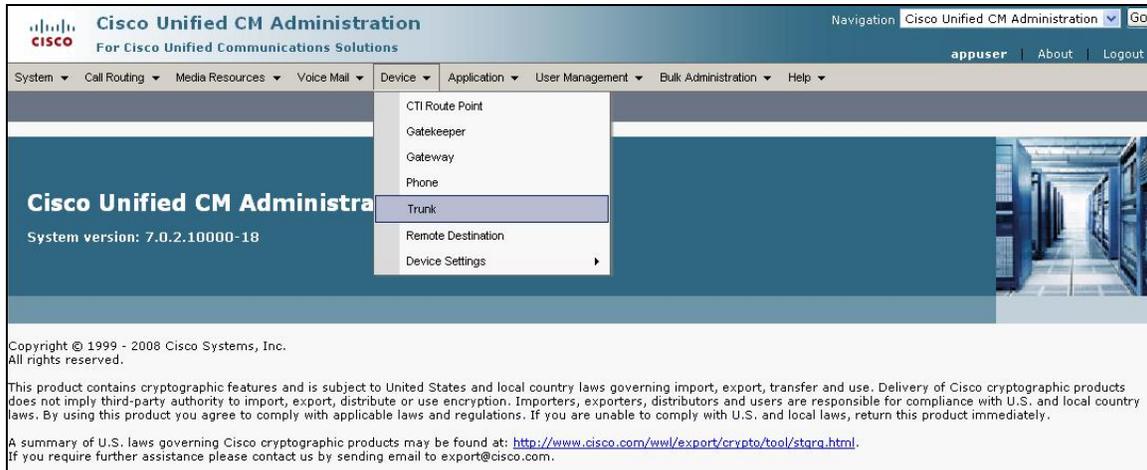


The **SIP Trunk Security Profile Information** configuration screen is displayed which was used in the sample network. Configure the highlighted areas as shown, and retain the default values for the remaining fields. Click **Save** to commit the changes.



4.3. Administer SIP Trunk

Scroll to the top of the screen, and select **Device** → **Trunk** as shown below.



The **Find and List Trunks** screen is displayed. Click **Add New** to add a new SIP Trunk.



Select **SIP Trunk** as the **Trunk Type** and the **Device Protocol** field will automatically be changed to **SIP**. Click **Next** to continue.

The screenshot shows the Cisco Unified CM Administration interface. At the top, there is a navigation bar with the Cisco logo and the text "Cisco Unified CM Administration For Cisco Unified Communications Solutions". Below this is a menu bar with options like System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled "Trunk Configuration" and includes a "Next" button with a green arrow. Below this, there is a "Status" section showing "Status: Ready". The "Trunk Information" section contains two dropdown menus: "Trunk Type*" set to "SIP Trunk" and "Device Protocol*" set to "SIP". At the bottom, there is another "Next" button and a note: "* - indicates required item."

The **SIP Trunk Configuration** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields. Click **Save** to commit the changes.

- **Device Name** An informative name
- **Description** Any note for this trunk

The screenshot shows the "SIP Trunk Configuration" screen. At the top, there is a "Save" button highlighted with a red box, along with "Delete", "Reset", and "Add New" buttons. Below this is the "Device Information" section. The "Device Name*" field and "Description" field are both highlighted with red boxes and contain the value "MXSIL". Other fields include "Product" (SIP Trunk), "Device Protocol" (SIP), "Device Pool*" (Default), "Common Device Configuration" (< None >), "Call Classification*" (Use System Default), "Media Resource Group List" (< None >), "Location*" (Hub_None), "AAR Group" (< None >), "Packet Capture Mode*" (None), and "Packet Capture Duration" (0).

Navigate to the SIP Information section and enter the following configuration:

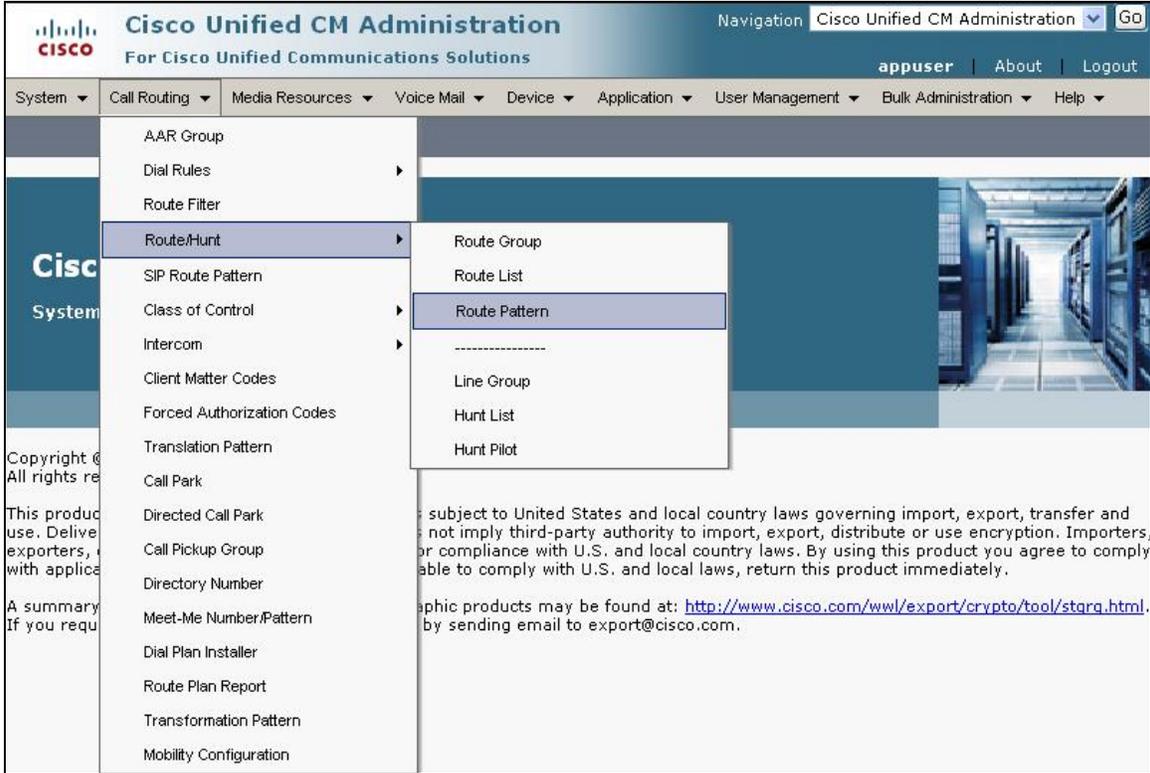
- **Destination Address** IP address of the Meeting Exchange or if distributed, then the Application Server
- **Destination Port** Destination port number use for SIP Communications
- **SIP Trunk Security Profile** Profile configured at **Section 4.2**
- **DTMF Signaling Method** Select **RFC 2833**

Click **Save** to commit the changes.

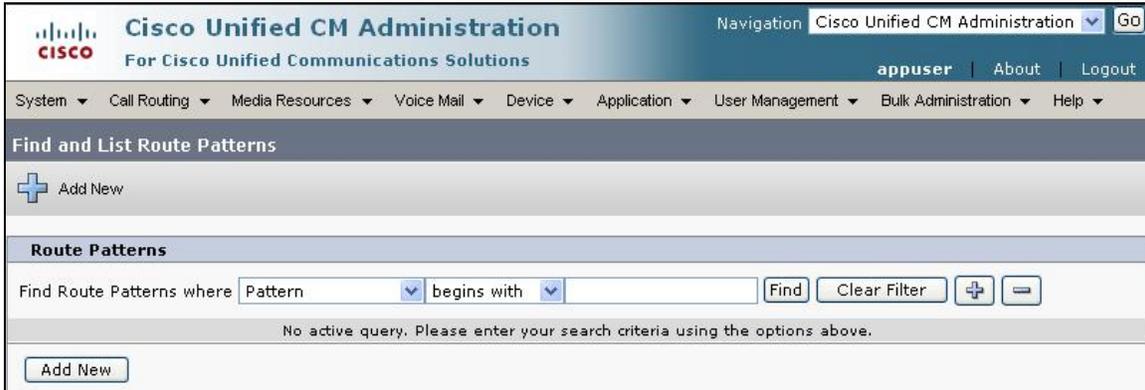
SIP Information	
Destination Address*	135.64.186.98
<input type="checkbox"/> Destination Address is an SRV	
Destination Port*	5060
MTP Preferred Originating Codec*	711ulaw
Presence Group*	Standard Presence group
SIP Trunk Security Profile*	MXSIL
Rerouting Calling Search Space	< None >
Out-Of-Dialog Refer Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile
DTMF Signaling Method*	RFC 2833

4.4. Administer Route Pattern

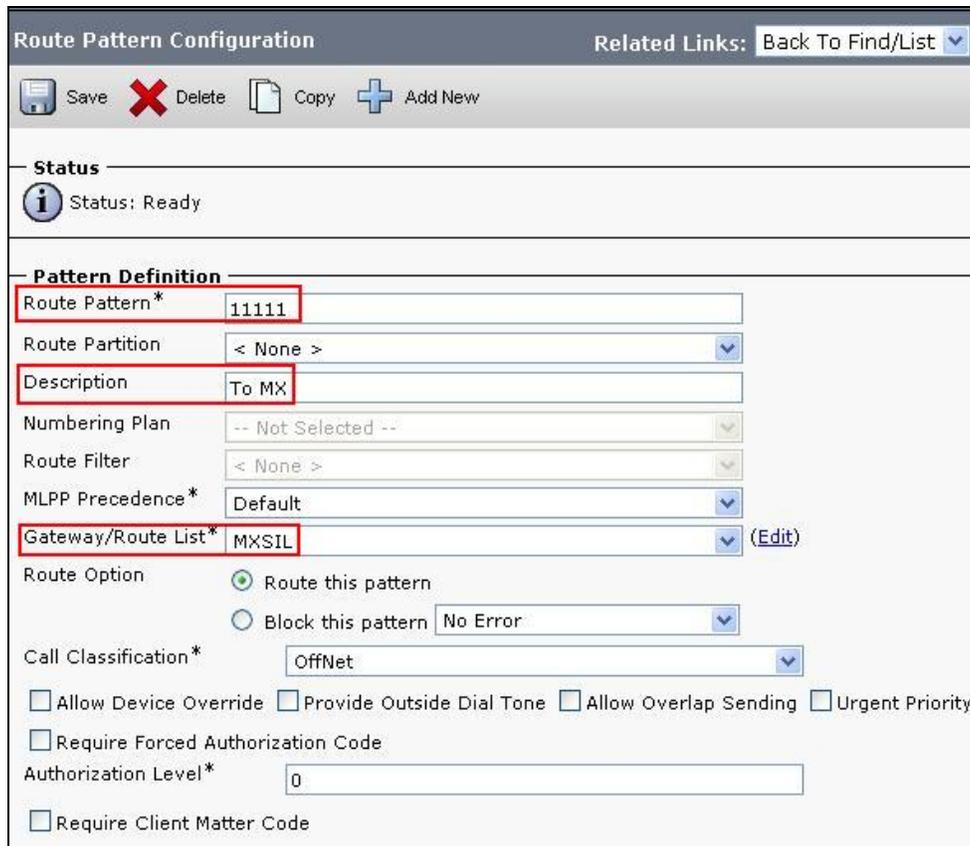
Scroll to the top of the screen, and select **Call Routing** → **Route/Hunt** → **Route Pattern** as shown below.



The **Find and List Route Patterns** screen is displayed. Click **Add New** to add a new Route Pattern.



The following screen shows the route pattern used in the sample network. The route pattern **11111** will cause calls to be routed through the MXSIL SIP Trunk defined in **Section 4.3**. Click **Save** to commit the changes (not shown).



Click OK on the two subsequent pop up dialog boxes.



4.5. Administer Route Groups

Route Groups must be administered to use multiple Application servers using the same Route pattern. In the example below two SIP trunks are created for each of the Application servers, MXSIL_Active and MXSIL_Standby, as per **Section 4.3**.

Find and List Trunks

Status

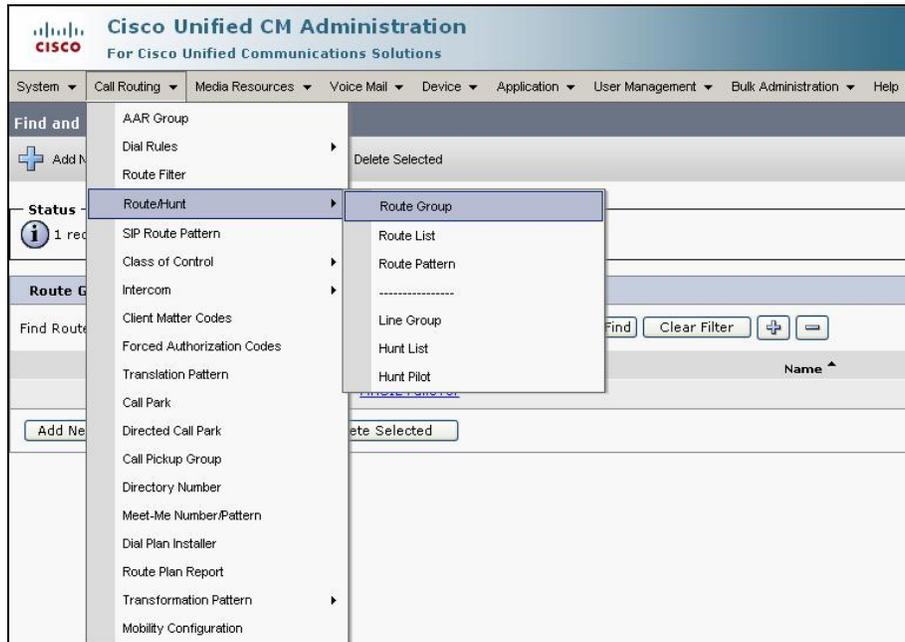
11 records found

Trunks (1 - 11 of 11) Rows per Page 50

Find Trunks where Device Name begins with

<input type="checkbox"/>	Name	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Security Profile
<input type="checkbox"/>	ASM-Silstack	To SM100		Default	37XXX				SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	ASM-Silstack	To SM100		Default	50000				SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	ASM-Silstack	To SM100		Default	320XX				SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	ASM-Silstack	To SM100		Default	200XX				SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	ASM-Silstack	To SM100		Default	300XX				SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	ASM-Silstack	To SM100		Default	39999				SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	ASM-Silstack	To SM100		Default	34XXX				SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	ASM-Silstack	To SM100		Default	80950				SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	CUBE	SIP Trunk to CUBE		Default	5XXX				SIP Trunk	CUBE SIP Trunk
<input type="checkbox"/>	MXSIL_Active	MXSIL_Active		Default					SIP Trunk	MXSIL
<input type="checkbox"/>	MXSIL_Standby	MXSIL_Standby		Default					SIP Trunk	MXSIL

Next is to administer Route Group. Scroll to the top of the screen, and select **Call Routing** → **Route/Hunt** → **Route Group** as shown below.



The **Find and List Route Patterns** screen is displayed. Click **Add New** to add a new Route Group.



The following screen shows the route group used in the sample network. The **Route Group Name** is any informative name. In the **Find Devices to Add to Route Group** the Trunk names created will be in the **Available Devices** table. Select both devices and select **Add to Route Group**. These devices will be shown in the **Current Route Group Members** table in **Selected Devices**. Click **Save** to commit the changes. Once saved ensure the **Route Group Members** table displays the group members which have just been added.

Route Group Configuration
Related Links:

Save
 Delete
 Add New

Route Group Information

Route Group Name*

Distribution Algorithm* Circular

Route Group Member Information

Find Devices to Add to Route Group

Device Name contains Find

Available Devices**

- MXSIL_Active
- MXSIL_Standby

Port(s) None Available

Add to Route Group

Current Route Group Members

Selected Devices***

- MXSIL_Active (All Ports)
- MXSIL_Standby (All Ports)

▼

▲

Reverse Order of Selected Devices

▼ ▲

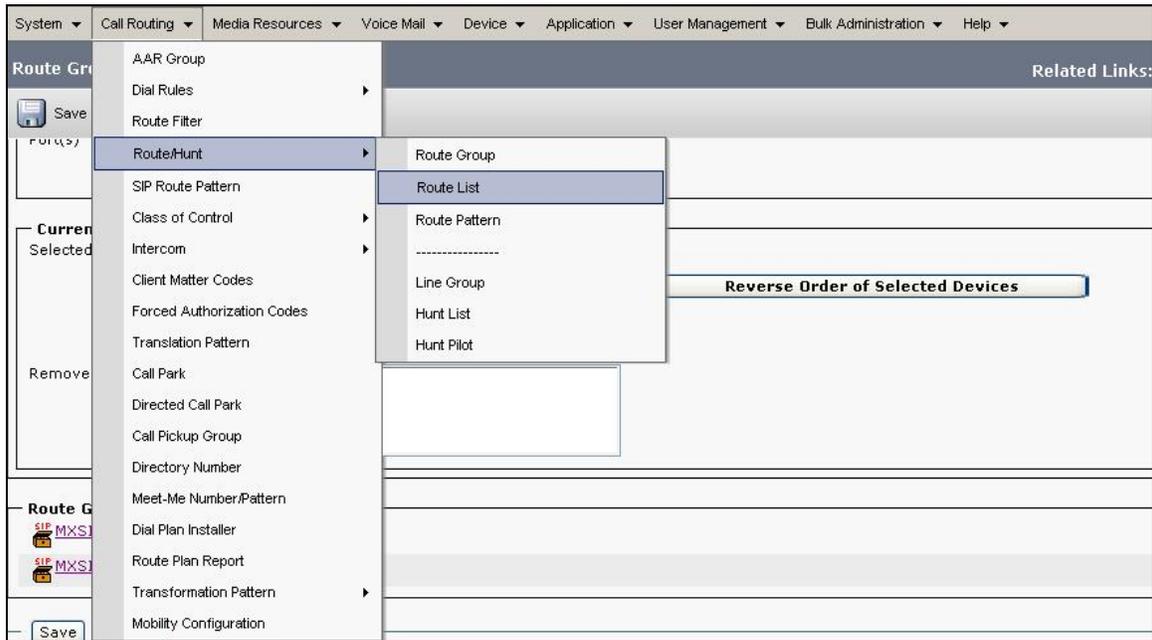
Removed Devices****

Route Group Members

- SIP
MXSIL_Active
- SIP
MXSIL_Standby

Save
Delete
Add New

Next is to administer Route List, scroll to the top of the screen and select **Call Routing** → **Route/Hunt** → **Route List** as shown below.



The **Find and List Route Patterns** screen is displayed. Click **Add New** to add a new Route List.



The **Route Group Name** is any informative name. The **Cisco Unified Communications Manager Group** is set to default. Click **Save** to commit the changes.

Route List Configuration Related Links: [Back To Find/List](#)

Status

Status: Ready

Route List Information

Name*

Description

Cisco Unified Communications Manager Group*

*- indicates required item.

**Ordered by highest priority

***Will be removed from Route List when you click Save

The following screen shows the **Route List Configuration**, select **Add Route Group**.

Route List Configuration Related Links: [Back To Find/List](#)

Status

Add successful

Route List Information

Name*

Description

Cisco Unified Communications Manager Group*

Enable this Route List (change effective on Save; no reset required)

Route List Member Information

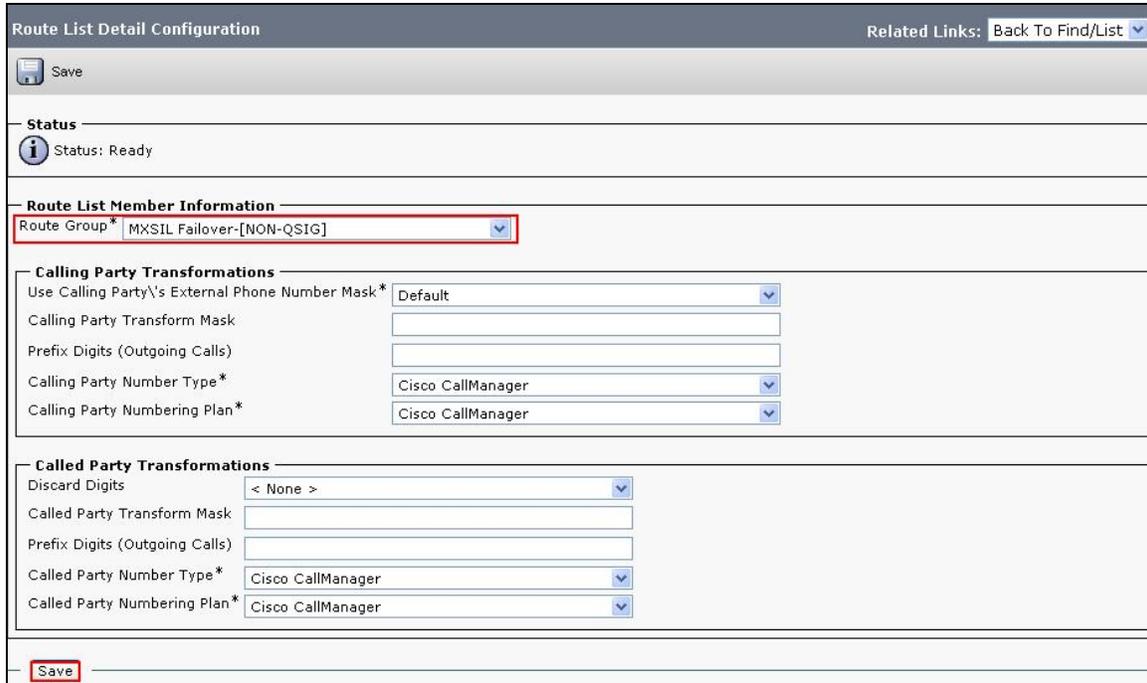
Selected Groups**

v ^

Removed Groups***

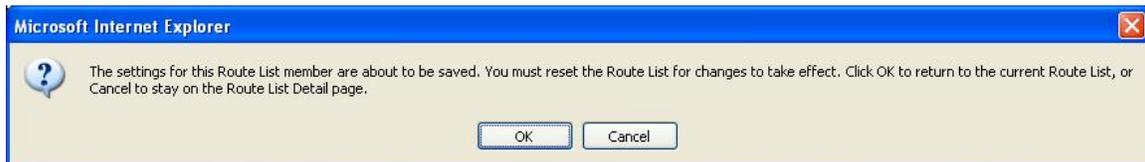
v ^

The following screen shows the **Route List Detail Configuration**. Configure the highlighted area as shown, where the **Route Group MXSIL Failover-[NON-QSIG]** is selected from the drop down menu, and retain the default values for the remaining fields. Click **Save** to commit the changes.



The screenshot shows the 'Route List Detail Configuration' window. At the top right, there is a 'Related Links: Back To Find/List' dropdown. Below the title bar is a 'Save' button. The 'Status' section shows 'Status: Ready'. The 'Route List Member Information' section is highlighted with a red box and contains a dropdown menu for 'Route Group*' with 'MXSIL Failover-[NON-QSIG]' selected. Below this are two sections: 'Calling Party Transformations' and 'Called Party Transformations'. The 'Calling Party Transformations' section includes fields for 'Use Calling Party\'s External Phone Number Mask*' (Default), 'Calling Party Transform Mask', 'Prefix Digits (Outgoing Calls)', 'Calling Party Number Type*' (Cisco CallManager), and 'Calling Party Numbering Plan*' (Cisco CallManager). The 'Called Party Transformations' section includes fields for 'Discard Digits' (< None >), 'Called Party Transform Mask', 'Prefix Digits (Outgoing Calls)', 'Called Party Number Type*' (Cisco CallManager), and 'Called Party Numbering Plan*' (Cisco CallManager). At the bottom left, there is a 'Save' button.

Click **OK** on the subsequent pop up dialog boxes.



The following screen shows the **MXSIL Failover** added as a Route List member. In the **Route List Details** table ensure it displays the group members which have just been added.

Route List Configuration Related Links: [Back To Find/List](#)

Save Delete Copy Reset Add New

Status
Add successful

Route List Information

Name*

Description

Cisco Unified Communications Manager Group*

Enable this Route List (change effective on Save; no reset required)

Route List Member Information

Selected Groups**

Removed Groups***

Route List Details

[MXSIL Failover](#)

Save Delete Copy Reset Add New

Ensure the Route List created status shows **Registered with callMgr** as per the screen below.

Find and List Route Lists

Add New Select All Clear All Delete Selected Reset Selected

Status
1 records found

Route List (1 - 1 of 1) Rows per Page 50

Find Route List where Name begins with

<input type="checkbox"/>	Name ^	Description	Enabled	Status
<input type="checkbox"/>	MX Redundancy	MX Route List Redundancy	true	Registered with callMgr

Add New Select All Clear All Delete Selected Reset Selected

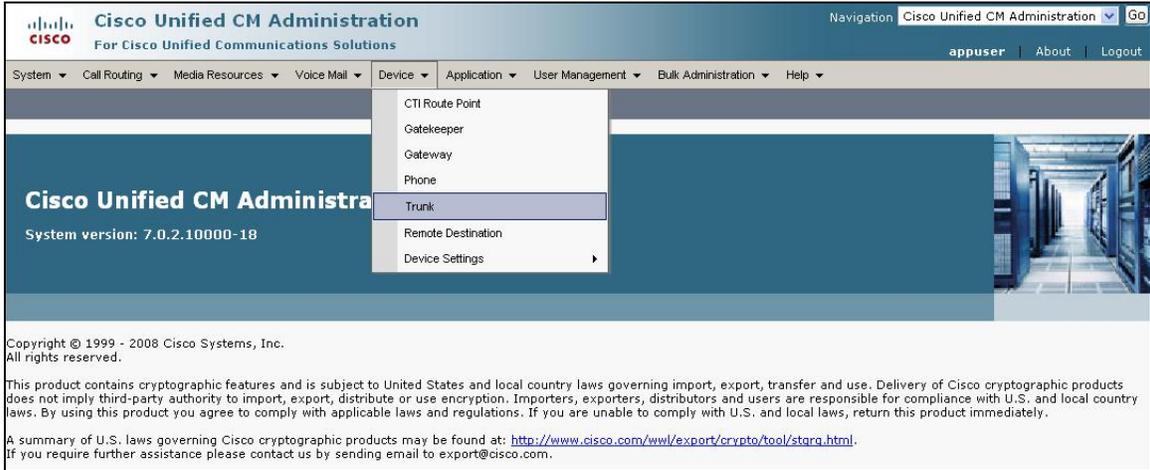
Edit the route pattern as defined in **Section 4.4**. Under **Gateway/Route List** select the route list defined above. Click **Save** to commit the changes (not shown).

The screenshot shows the 'Route Pattern Configuration' dialog box. At the top, there are buttons for 'Save', 'Delete', 'Copy', and 'Add New'. Below that is a 'Status' section showing 'Status: Ready'. The main area is titled 'Pattern Definition' and contains several fields: 'Route Pattern*' (11111), 'Route Partition' (< None >), 'Description' (MXSIL Failover), 'Numbering Plan' (-- Not Selected --), 'Route Filter' (< None >), 'MLPP Precedence*' (Default), 'Resource Priority Namespace Network Domain' (< None >), 'Gateway/Route List*' (MX Redundancy), and 'Route Option' (Route this pattern). The 'Gateway/Route List*' field is highlighted with a red border, and an '(Edit)' link is visible to its right.

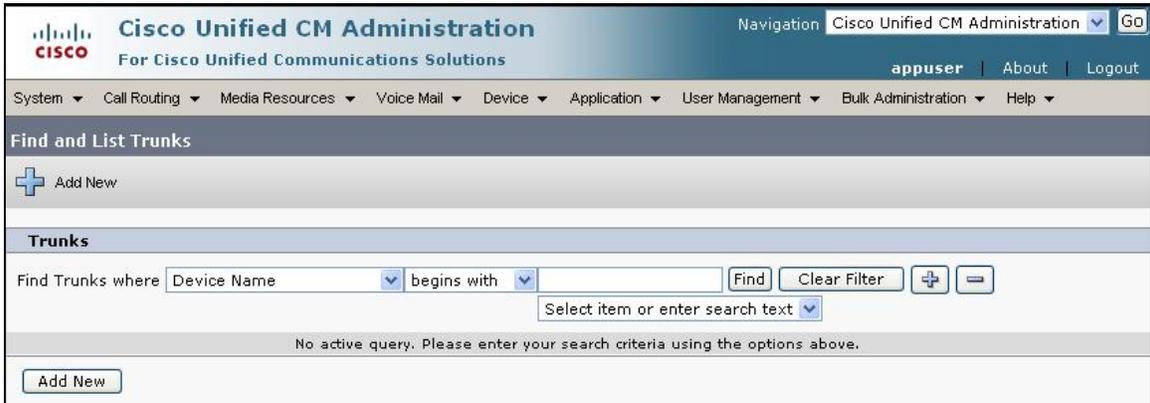
Click OK on the two subsequent pop up dialog boxes.



Scroll to the top of the screen and select **Device** → **Trunk** as shown below.



The **Find and List Trunks** screen is displayed. Click **Add New** to add a new SIP Trunk.



The List **Trunk Configuration** screen is displayed. It shows both the MXSIL_Active and MXSIL_Backup servers have now been configured in the Route Group with the selected Priority.

Find and List Trunks

Status

11 records found

Trunks (1 - 11 of 11) Rows per Page 50

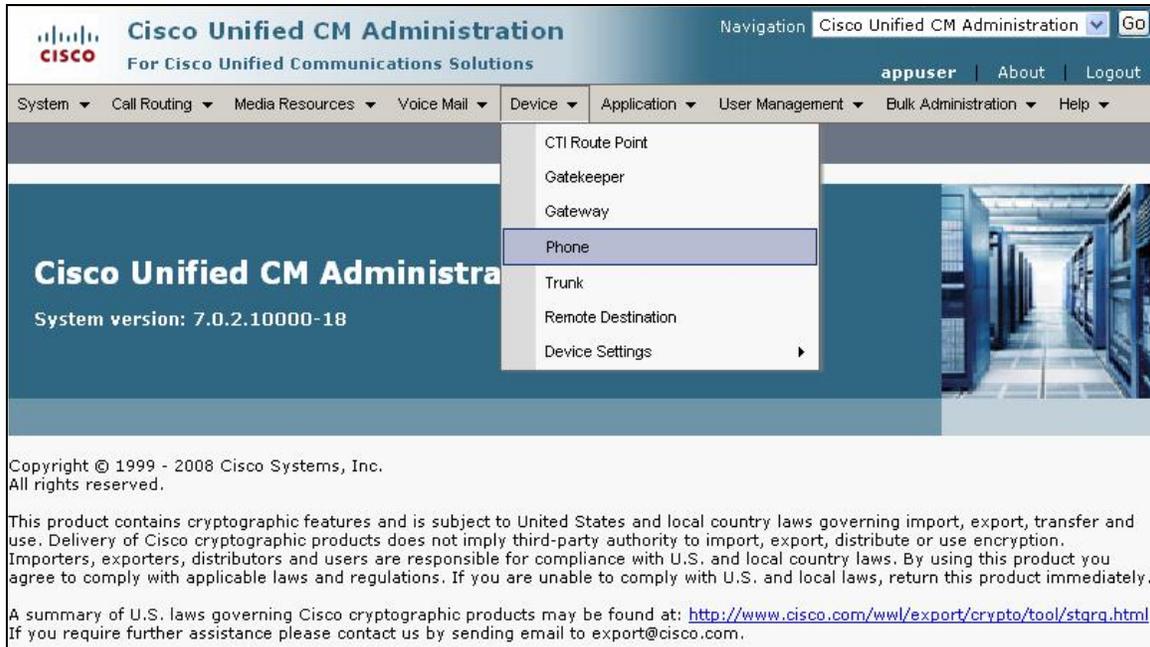
Find Trunks where Device Name begins with Find

Select item or enter search text

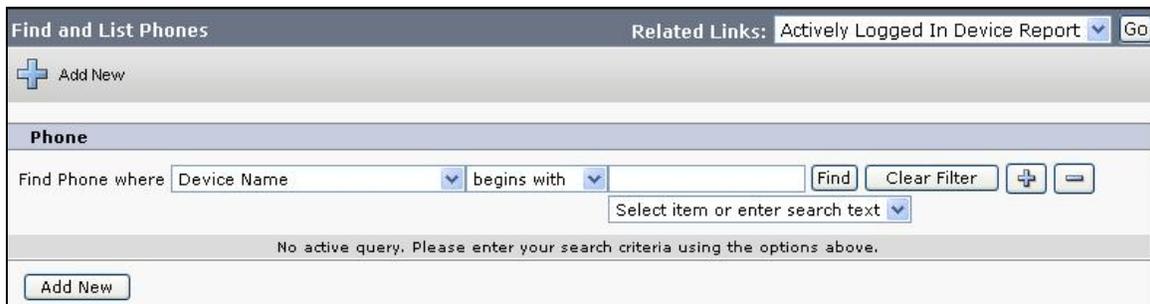
<input type="checkbox"/>	Name ^	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Security Profile
<input type="checkbox"/>	ASM-Silstack	To SM100		Default	37XXX				SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	ASM-Silstack	To SM100		Default	50000				SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	ASM-Silstack	To SM100		Default	320XX				SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	ASM-Silstack	To SM100		Default	200XX				SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	ASM-Silstack	To SM100		Default	300XX				SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	ASM-Silstack	To SM100		Default	39999				SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	ASM-Silstack	To SM100		Default	34XXX				SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	ASM-Silstack	To SM100		Default	80950				SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	CUBE	SIP Trunk to CUBE		Default	5XXX				SIP Trunk	CUBE SIP Trunk
<input type="checkbox"/>	MXSIL_Active	MXSIL_Active		Default				MXSIL Failover 1	SIP Trunk	MXSIL
<input type="checkbox"/>	MXSIL_Backup	MXSIL_Backup		Default				MXSIL Failover 2	SIP Trunk	MXSIL

4.5. Administer Phones

Scroll to the top of the screen and select **Device** → **Phone** as shown below.



The **Find and List Phones** screen is displayed.



The following screen shows the display after a device has been selected. Click on the line for the device as highlighted in the screen below.

Phone Configuration
Related Links: [Back To Find/List](#)

Save
Delete
Copy
Reset
Add New

Status

Status: Ready

Association Information

[Modify Button Items](#)

1	Line [1] - 6002 (no partition)
2	None
3	Add a new SD
4	Add a new SD
5	Add a new SD
6	Add a new SD
----- Unassigned Associated Items -----	
7	Line [2] - Add a new DN
8	Add a new SD
9	Privacy
10	None

Phone Type

Product Type: Cisco 7911

Device Protocol: SIP

Device Information

Registration: Registered with Cisco Unified Communications Manager callmgr

IP Address: [135.64.186.239](#)

MAC Address*:

Description:

Device Pool*: [View Details](#)

Common Device Configuration: [View Details](#)

Phone Button Template*:

Softkey Template:

Common Phone Profile*:

Calling Search Space:

The following screen shows the display after the line has been selected. Enter information for **Directory Number**, **Alerting Name** and **ASCII Alerting Name**.

Directory Number Configuration Related Links: [Configure Device \(SEP0023049CDB7B\)](#)

Save ✖ Delete ↺ Reset + Add New

Status

i Status: Ready

Directory Number Information

Directory Number*

Route Partition

Description

Alerting Name

ASCII Alerting Name

Allow Control of Device from CTI

Associated Devices

▼ ^

Dissociate Devices

Navigate to **Line 1 on Device** section and enter information for **Display (Internal Caller ID)** and **ASCII Display (Internal Caller ID)**. This will be displayed on the called party phone on all outgoing calls. Check all boxes in **Forwarded Call Information Display on Device** section. Click **Save** to complete.

Line 1 on Device SEP0023049CDB7B	
Display (Internal Caller ID)	Cisco SIP <small>Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.</small>
ASCII Display (Internal Caller ID)	Cisco SIP
Line Text Label	<input type="text"/>
ASCII Line Text Label	<input type="text"/>
External Phone Number Mask	<input type="text"/>
Visual Message Waiting Indicator Policy*	Use System Policy <input type="button" value="v"/>
Audible Message Waiting Indicator Policy*	Default <input type="button" value="v"/>
Ring Setting (Phone Idle)*	Ring <input type="button" value="v"/>
Ring Setting (Phone Active)	Use System Default <input type="button" value="v"/> <small>Applies to this line when any line on the phone has a call in progress.</small>
Call Pickup Group Audio Alert Setting (Phone Active)	Use System Default <input type="button" value="v"/>
Recording Option*	Call Recording Disabled <input type="button" value="v"/>
Recording Profile	< None > <input type="button" value="v"/>
Monitoring Calling Search Space	< None > <input type="button" value="v"/>
Multiple Call/Call Waiting Settings on Device SEP0023049CDB7B	
<small>Note: The range to select the Max Number of calls is: 1-6</small>	
Maximum Number of Calls*	<input type="text" value="4"/>
Busy Trigger*	<input type="text" value="2"/> <small>(Less than or equal to Max. Calls)</small>
Forwarded Call Information Display on Device SEP0023049CDB7B	
<input checked="" type="checkbox"/> Caller Name	
<input checked="" type="checkbox"/> Caller Number	
<input checked="" type="checkbox"/> Redirected Number	
<input checked="" type="checkbox"/> Dialed Number	
Users Associated with Line	
<input type="button" value="Associate End Users"/>	
<input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Reset"/> <input type="button" value="Add New"/>	

5. Verification Steps

The following steps were used to verify the administrative steps presented in these Application Notes and are applicable for similar configurations in the field. The verification steps in this section validated the following:

- The Avaya Meeting Exchange Enterprise S6200 Conferencing Server configuration

5.1. Avaya Meeting Exchange Enterprise S6200 Conferencing Server Processes

Verify all conferencing related processes are running on the Meeting Exchange as follows:

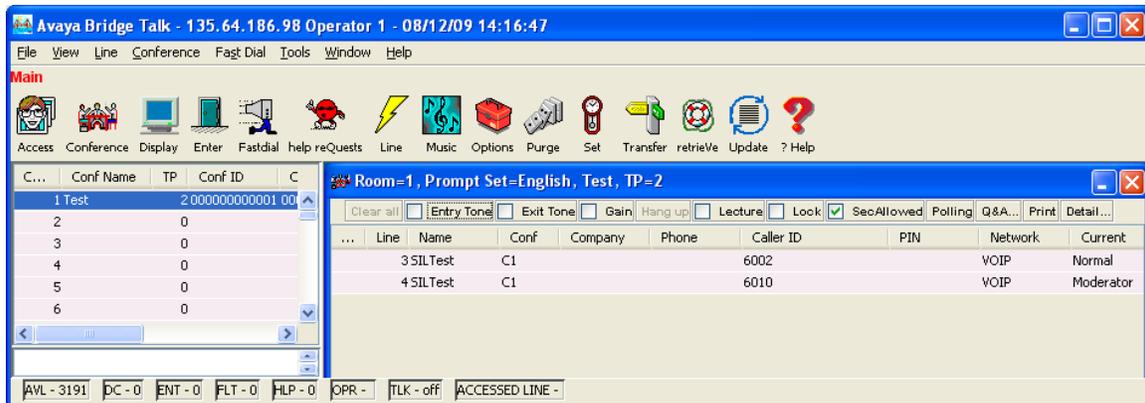
- Log in to the Meeting Exchange server console to access the CLI with the appropriate credentials.
- cd to **/usr/dcb/bin**
- At the command prompt, run the script **service mx-bridge status** and confirm all processes are running by verifying an associated Process ID (PID) for each process.

```
[sroot@MXSIL ~]# service mx-bridge status
5042 ?      00:00:01  initdcb
5604 ?      00:00:00  log
5607 ?      00:00:00  bridgeTranslato
5608 ?      00:00:00  netservices
5626 ?      00:00:00  timer
5627 ?      00:00:00  traffic
5628 ?      00:00:00  chdbased
5629 ?      00:00:00  startd
5630 ?      00:00:00  cdr
5631 ?      00:00:00  modapid
5632 ?      00:00:00  schapid
5633 ?      00:00:01  callhand
5634 ?      00:00:00  initipcb
5644 ?      00:00:00  sipagent
5645 ?      00:00:00  msdispatcher
5646 ?      00:00:00  serverComms
5648 ?      00:00:00  softms
5649 ?      00:00:00  softms
5650 ?      00:00:00  softms
5651 ?      00:00:00  softms
5652 ?      00:00:00  softms
5653 ?      00:00:00  softms
4022 ?      00:00:00  postmaster with 9 children
```

5.1.1. Verify Call Routing

Verify end to end signalling/media connectivity between the Meeting Exchange and the Cisco Unified Communications Manager. This is accomplished by placing calls from the Cisco end points to the Meeting Exchange. This step utilizes the Avaya Bridge Talk application to verify calls to and from the Meeting Exchange are managed correctly, e.g., callers are added/removed from conferences. This step will also verify the conferencing applications provisioned.

- Configure a conference with Auto Blast enabled and provision a dial list. From an endpoint on the Public Switched Telephone Network, dial a number that corresponds to DNIS **11111** to enter a conference as **Moderator** (with passcode) and blast dial is invoked automatically. When answered these callers enter the conference.
- If not already logged on, log in to the Avaya Bridge Talk application with the appropriate credentials
- **Double-Click on the** highlighted **Conf #** to open a **Conference Room** window
- Verify conference participants are added/removed from conferences by observing the Conference Navigator and/or Conference Room windows.



5.2. Verified Scenarios

The following scenarios have been verified for the configuration described in these Application Notes.

- Place a call from the 7911G IP Telephone (SIP) and the Cisco 7911G IP Telephone (SCCP) to a scheduled conference on the Meeting Exchange.
- Ensure the welcome message is played from the Conferencing Bridge and there is audio between callers in the conference.
- Initiate dial out by dialling *1 on the phone's touch pad and entering the phone number. Enter the number and press 1 to make the call. When the callers answer dial *2 to return them to the main conference.

6. Conclusion

As illustrated in these Application Notes, Avaya Meeting Exchange Enterprise S6200 Conferencing Server can interoperate with Cisco Unified Communications Manager using SIP trunks. No verification of TLS was performed between Avaya Meeting Exchange Enterprise S6200 Conferencing Server and Cisco Unified Communications Manager. The following interoperability items were observed during testing:

- No outgoing audio from Cisco SIP phone with codec ILBC30
- G.726 is not supported by Call Manager in 7.0.2.100000-18

7. Additional References

Avaya Meeting Exchange references are available at <http://support.avaya.com>

- [1] *Meeting Exchange S6200 5.2 Administration and Maintenance S6200/S6800*
- [2] *Avaya Meeting Exchange Enterprise Groupware Edition Version 5.2 User's Guide for Bridge Talk*

Cisco references are available at <http://cisco.com>

- [3] *Cisco Unified Communications Manager Administration Guide for Cisco Unified Communications Manager Business Edition*, Release 7.0(1), Part Number: OL-15405-01
- [4] *Cisco Unified Communications Manager Features and Services Guide for Cisco Unified Communications Manager Business Edition*, Release 7.0(1), Part Number: OL-15409-01
- [5] *Cisco Unified Real-Time Monitoring Tool Administration Guide*, Release 7.0(1), Part Number: OL-14994-01

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