Avaya Solution & Interoperability Test Lab

Configuring SIP Connectivity between the Avaya Meeting Exchange S6200 Conferencing Server and Cisco Unified CallManager - Issue 1.0

Abstract

These Application Notes present the procedures for configuring SIP connectivity between the Avaya Meeting Exchange S6200 Conferencing Server and Cisco Unified CallManager. SIP connectivity is enabled via directly connected SIP trunking between Avaya Meeting Exchange and Cisco Unified CallManager. This configuration provides endpoints registered with Cisco Unified CallManager a rich set of conferencing options available on Avaya Meeting Exchange.

1. Introduction

These Application Notes present the procedures for configuring SIP connectivity between the Avaya Meeting Exchange S6200 Conferencing Server and Cisco Unified CallManager. SIP connectivity is enabled via directly connected SIP trunking between Avaya Meeting Exchange and Cisco Unified CallManager. This configuration provides endpoints registered with Cisco Unified CallManager a rich set of conferencing options available on Avaya Meeting Exchange.

The following conferencing features have been verified:

- Dial-In Conferencing:
  - DNIS direct call function, where conference participants enter a conference as moderator without entering a participant-access-code (passcode).
  - Scan call function, where conference participants enter a conference with a valid passcode.
- Dial-Out Conferencing from Avaya Meeting Exchange:
  - Blast dial
    - Auto, where a conference participant enters a conference via a DNIS direct call function and autonomously invokes a Blast dial to a pre-provisioned dial list of one or more participants.
    - Manual, where a conference participant is already in a conference as moderator and invokes a Blast dial to a pre-provisioned dial list of one or more participants.
  - Originator Dial-Out, where a conference participant is already in a conference as moderator and invokes a Dial-Out to a single participant
  - Operator Fast Dial, where an operator can Dial-Out to a pre-provisioned dial list of one or more participants e participants.
- Operator Dial-Out to establish an Audio Path.
- Operator Dial-In to establish an Audio Path.
- Dial-Out to an FDAPI channel for audio recording.
- Line Transfer initiated from Avaya Bridge Talk.
- Conference Transfer initiated from Avaya Bridge Talk.

The following CODECS were verified:

- G711MU.
- G.711A.

The following SIP feature testing was verified:

- Call Hold/Resume, invoked from and endpoint registered with Cisco Unified CallManager participating in an active conference call.
- Call Transfer, imitated from an endpoint registered with Cisco Unified CallManager participating in an active conference call, transferred to an endpoint registered with Cisco Unified CallManager.

These Application Notes provide the administrative steps for configuring the Avaya Meeting Exchange S6200 Conferencing Server to interoperate with Cisco Unified CallManager via direct SIP trunking (see **Figure 1**).
This configuration enables endpoints registered with Cisco Unified CallManager (for these Application Notes, Cisco 7960 phones) access to Avaya Meeting Exchange. The Avaya Bridge Talk application is utilized for provisioning, scheduling and managing conferences.

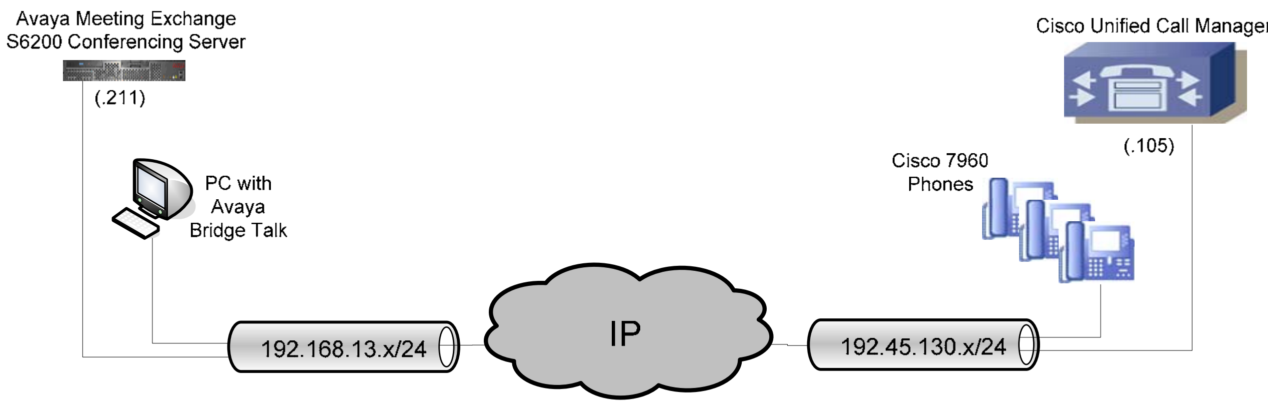


Figure 1: Network Configuration

2. Equipment and Software Validated

The following equipment and software versions were used for the configuration:

Equipment	Software
Avaya Meeting Exchange S6200 Conferencing Server • Software version • IPCB build version	 40103_00_01 mx7_1.3.00-86
Avaya Bridge Talk	4.1.01b
Cisco Unified CallManager • System Version • Administration Version	5.0 (5.0.4.2000-1) (1.1.0.0-1)
Cisco 7960 Phones	8.0 (SCCP)

Table 1: Hardware and Software Versions

3. Avaya Meeting Exchange Configuration

This section describes the steps for configuring Avaya Meeting Exchange to interoperate with Cisco Unified CallManager via direct SIP trunking (see **Figure 1**).

Step	Description
3.1	Log in to the Avaya Meeting Exchange Server console to access the Command Line Interface (CLI) with the appropriate credentials.

3.2	Configure settings that enable SIP connectivity between Avaya Meeting Exchange and other SIP User Agents by editing the **system.cfg** file as follows:
	• cd to **/usr/ipcb/config**
	• Edit the **system.cfg** file with a text editor, e.g., vi.
	• Add a line to identify the IP address of Avaya Meeting Exchange (as defined in the **/etc/hosts** file), e.g.,
	  o **IPAddress=192.168.13.211**
	• Add a line to populate the From Header Field in SIP INVITE messages from Avaya Meeting Exchange, e.g.,
	  o **MyListener=sip:S6200@192.168.13.211**
	    Note*: The string "****S6200****" is arbitrarily chosen.*
	• Add a line to provide User Agents a Contact address to use for Acknowledging SIP messages from Avaya Meeting Exchange, e.g.,
	  o **respContact=<sip:S6200@192.168.13.211:5060;transport=tcp>**
	    Note*: The string "****S6200****" is arbitrarily chosen.*
	• Add the following lines to set the Min-SE timer to **86400** seconds in SIP INVITE messages from Avaya Meeting Exchange, e.g.,
	  o **sessionRefreshTimerValue=86400**
	  o **minSETimerValue=86400**
	Note*: The values for the* ***sessionRefreshTimerValue*** *and the* ***minSETimerValue*** *are defined in seconds and must be provisioned to a value greater than or equal to 1800 seconds (1800 seconds is the value for the Min-SE used by Cisco Unified CallManager). This setting is necessary to enable Dial-Out from Avaya Meeting Exchange to Cisco Unified CallManager.*

Step	Description

3.3	To associate incoming calls to Avaya Meeting Exchange with different call flows, edit the **UriToTelnum.tab** file to extract Direct Inward Dial (DID, also referred to as DDI in some regions) values as follows:
	• cd to **/usr/ipcb/config**
	• Edit the **UriToTelnum.tab** file with a text editor, e.g., vi.
	• Add a line to match the pattern of the To Header Field in SIP INVITE messages from Cisco Unified CallManager to Avaya Meeting Exchange. If a match occurs, the DID is extracted from the To header field, e.g.,
	  o **"*<sip:*@*" $2**
	  where **"*<sip:*@*"** matches:
	  To: **<sip:556@192.168.13.211>** and **$2** utilizes the variable contained in the second ***** as the DID value for the call.
	• Enable an undefined caller to receive a prompt for operator assistance by administering for the condition of an unmatched SIP INVITE message by adding a wildcard entry as the last line in this file, e.g.,
	  o *** $0**
	  ***Note**: Entries in this file are read sequentially, therefore, the line *** $0** must be the last line in the file. Otherwise, all calls to Avaya Meeting Exchange would match the wildcard and thus receive a prompt for operator assistance.*

Step	Description
3.4	To enable Dial-Out from Avaya Meeting Exchange to Cisco Unified CallManager via SIP trunking, edit the **telnumToUri.tab** file as follows: • cd to **/usr/ipcb/config** • Edit the **telnumToUri.tab** file with a text editor, e.g., vi. • Add a line to route outbound calls from Avaya Meeting Exchange to Cisco Unified CallManager, e.g., o **5???? sip:$0@192.45.130.105:5060;transport=tcp** Where the route pattern **5????** matches any five digit number with a leading "**5**" and routes the call to Cisco Unified CallManager (**192.45.130.105**) via SIP/TCP. To enable SIP connectivity utilizing TCP, the entry contains: **5060** and **transport=tcp**. Avaya Meeting Exchange will substitute "**$0**" with the dialed number in outgoing SIP INVITE messages, e.g., if **56011** is dialed, Avaya Meeting Exchange will send a SIP INVITE message with: **sip:56011@192.45.130.105:5060;transport=tcp** in the SIP URI and To header field. ***Note**: Alternatively, routing to Cisco Unified CallManager could have been enabled as a default gateway with a wildcard entry, e.g., *** sip:$0@192.45.130.105:5060;transport=tcp** where ***** routes any dialed digits to Cisco Unified CallManager (**192.45.130.105**) via SIP/TCP.*

Step	Description
3.5	To configure Avaya Meeting Exchange as software media server (softms, which utilizes software based DSP resources), edit the **processTable.cfg** file as follows: • cd to **/usr/ipcb/config** • Edit the **processTable.cfg** file with a text editor, e.g., vi. ***Note**: The **processTable.cfg** for these Application Notes contains IP Addresses of **0.0.0.0**, which are equivalent to the IP address (**192.168.13.211**) of Avaya Meeting Exchange.*

	`# processes file, enumerates the number of processes in the network.`
	`# will have the name of the process   Key ID and the IP address`
	`proccessName   ipcKeyNumber   ProcessExe              ipAddress  route  ProcessArgs`
	`initipcb       110            noexecute                 0.0.0.0`
	`bridget700     100            noexecute                 0.0.0.0`
	`dspEvents/msDispatcher,netEvents/sipAgent`
	`commsProcess   111            /usr/dcb/bin/serverComms   0.0.0.0`
	`sipAgent       101            /usr/dcb/bin/sipagent      0.0.0.0`
	`dspEvents/msDispatcher,appEvents/bridget700`
	`msDispatcher   102            /usr/dcb/bin/msdispatcher  0.0.0.0`
	`netEvents/sipAgent,appEvents/bridget700,dspEvents/mediaServer`
	`mediaServer    103            /usr/dcb/bin/softms        0.0.0.0`
	`appEvents/msDispatcher,netEvents/msDispatcher   1`
	`snmpAgent      120            noexecute                 0.0.0.0`

3.6	**Reboot** Avaya Meeting Exchange for changes to take effect. ***Note**: Rebooting Avaya Meeting Exchange is service impacting.*
	`[S6200]> init 6`

CBUTIL Utility

The following steps provide examples of how to provision DIRECT and SCAN call functions by utilizing the cbutil utility on Avaya Meeting Exchange. DID values (obtained from procedures in the previous section) are associated with call functions to access conferences provisioned on Avaya Meeting Exchange.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td rowspan="3">3.7</td><td>To map DID values obtained in Step 3.3 to DNIS entries, run the cbutil utility as follows:

If not already logged on, log in to the Avaya Meeting Exchange Server console to access the CLI with the appropriate credentials.
At the command prompt enter tcsh to set the UNIX shell on Avaya Meeting Exchange.
At the command prompt run the cbutil utility to verify DNIS entries provisioned on Avaya Meeting exchange.

Note: A command line utility, cbutil enables administrators to assign a specific annunciator message, line name, company name, system function, reservation group and prompt sets to a maximum of 30,000 DNIS or DID patterns. Each reservation group may use one passcode to enter a conference. In this way, administrators can create different reservation groups on Avaya Meeting Exchange, rather than relying on a single, bridge-wide passcode. Avaya Meeting Exchange stores this assignment information in the Call Branding table of the database. Avaya Meeting Exchange sorts the information in the Call Branding table in ascending order of the DNIS or DID number with the wildcard character “?” last in a series. For example, 129? follows 1299. The last entry in the table consists entirely of wildcard characters. The number of characters in this entry corresponds to the number of DNIS/DID digits specified in the Digit Parameters configuration.</td></tr>
<tr><td><pre>
S6200>cbutil
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.

Usage: cbutil <command> [command-specific args...]
 where <command> may be one of:
  add   Add an entry to the Call Branding table
  remove   Remove an entry from the Call Branding table
  update   Update an entry in the Call Branding table
  lookup   Display an entry in the Call Branding table
  count   Display the number of entries in the Call Branding table
  list   List entries in the Call Branding table
  dnissize   Set system configured max dnis length (1-16)
   Note: This command should only be used when the bridge is not running.
Use "cbutil<command> -help" to get help on a specific command
</pre></td></tr>
<tr><td></td></tr>
</table>

Step	Description

3.8 Enable Dial-In access (via passcode) to conferences provisioned on Avaya Meeting Exchange as follows:

- Add a DNIS entry for a **scan call function** corresponding to DID **501** by entering the following command at the command prompt:
  cbutil add <**dnis**> <**rg**> <**msg**> <**ps**> <**ucps**> <**func**> [**-l** <**ln**> **-c** <**cn**>], where the variables for add command is defined as follows:
  - <**dnis**> DNIS
  - <**rg**> Reservation Group
  - <**msg**> Annunciator message number
  - <**ps**> Prompt Set number (0-20)
  - <**ucps**> Use Conference Prompt Set (y/n)
  - <**func**> One of: DIRECT/SCAN/ENTER/HANGUP/AUTOVL/FLEX
  - **-l** <**"ln"**> Optional line name to associate with caller
  - **-c** <**"cn"**> Optional company name to associate with caller

```
S6200>cbutil add 501 0 1 1 n scan
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.
```

3.9 Enable Dial-In access (as moderator without entering a passcode) to conferences provisioned on Avaya Meeting Exchange by adding a DNIS entry for a **direct call function** corresponding to DID **556**.

```
S6200>cbutil add 556 0 301 1 n direct
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.
```

3.10 At the command prompt enter **cbutil list** to verify the DNIS entries provisioned in **Steps 3.8** and **3.9** were provisioned and entered correctly.

***Note**: The last entry in the call brand table is the wild card entry "**???**". This entry captures any wrong number (e.g., unmatched **DID** values) and places the call into enter queue for operator assistance.*

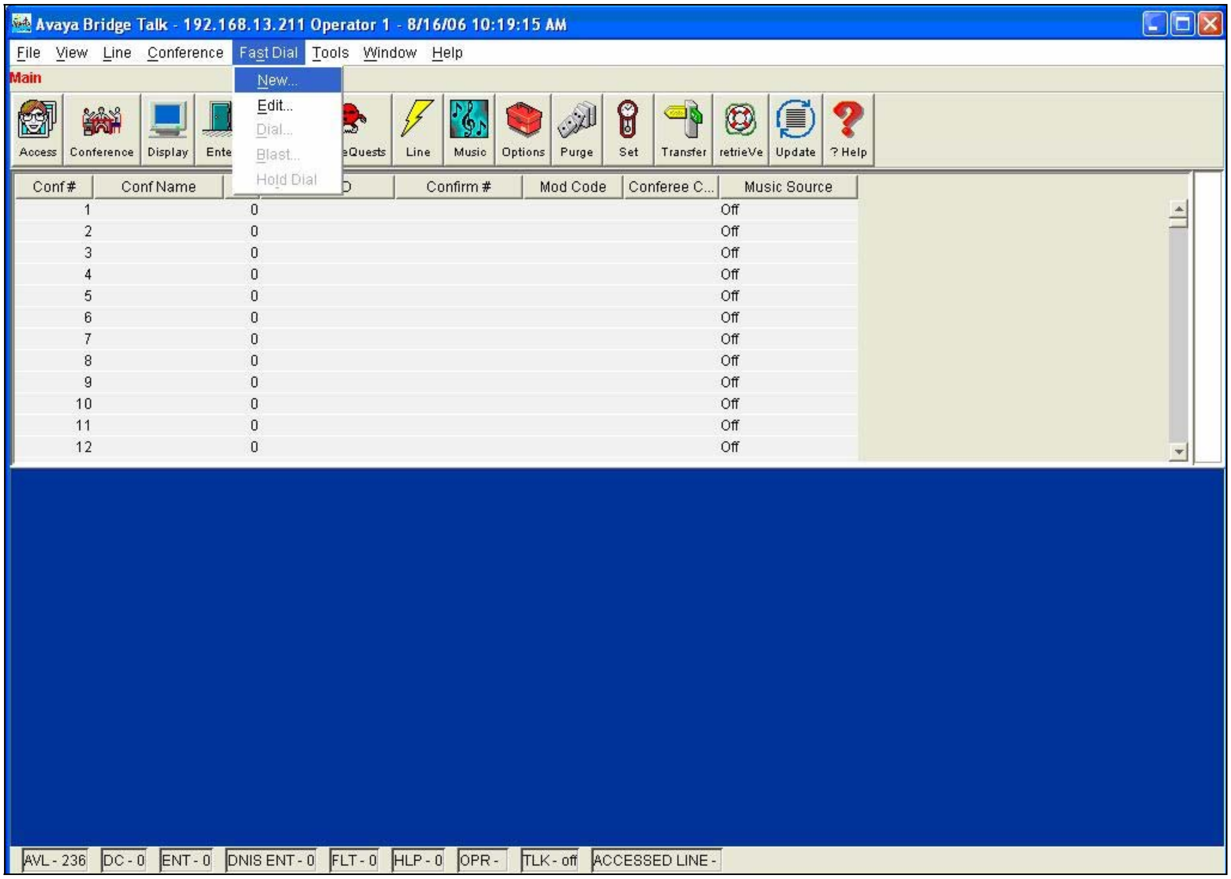
```
S6200>cbutil list
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.

DNIS             Grp Msg PS  CP Function Line Name            Company Name
---------------- --- --- --- -- -------- -------------------- -----------------
501              0   1   1   N  SCAN
556              0   301 1   N  DIRECT
???              0   208 1   N  ENTER
```

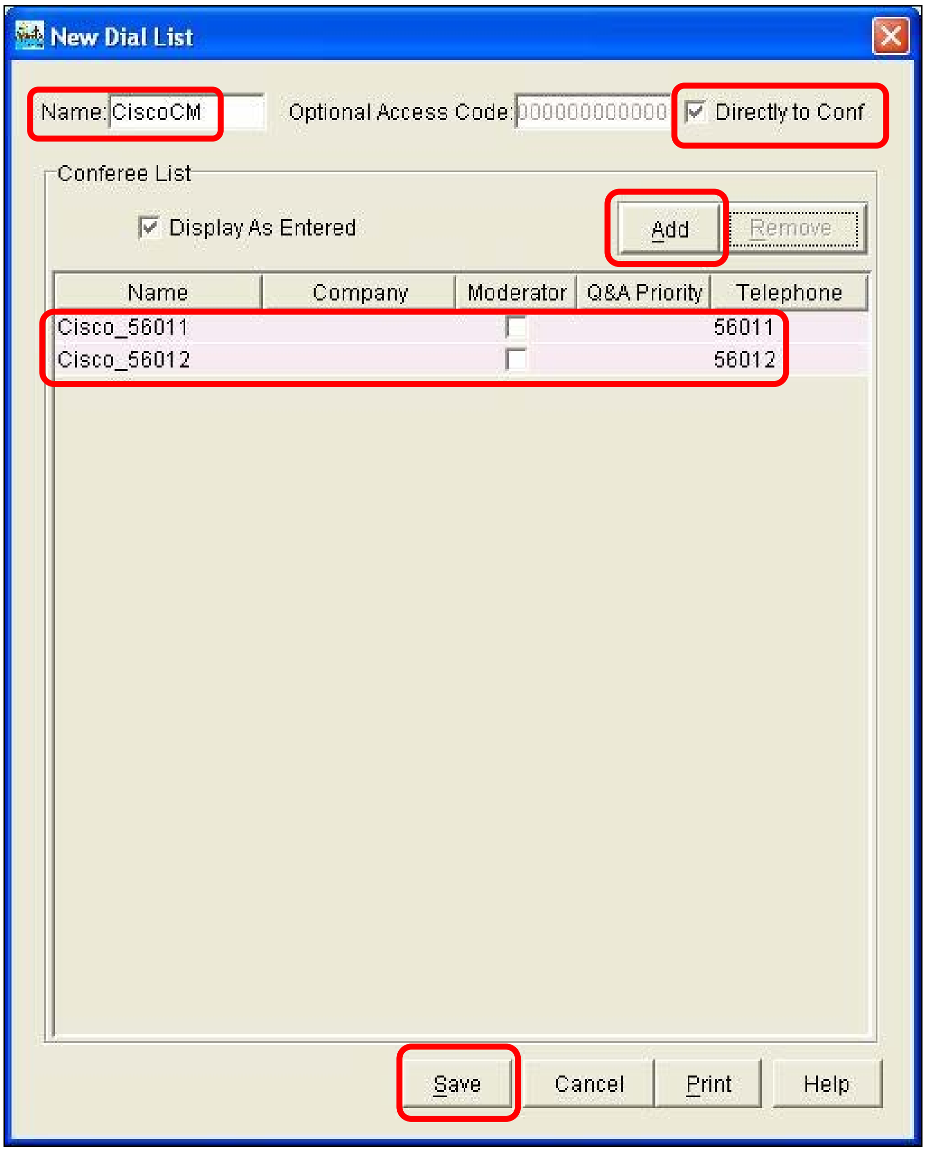
Bridge Talk

The following steps provide an example of how to provision a conference on Avaya Meeting Exchange from the Avaya Bridge Talk application. This sample conference is utilized in conjunction with the DIRECT and SCAN call functions (provisioned in the previous steps) to enable both Dial-In and Dial-Out access to audio conferencing for endpoints registered to Cisco Unified CallManager.

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

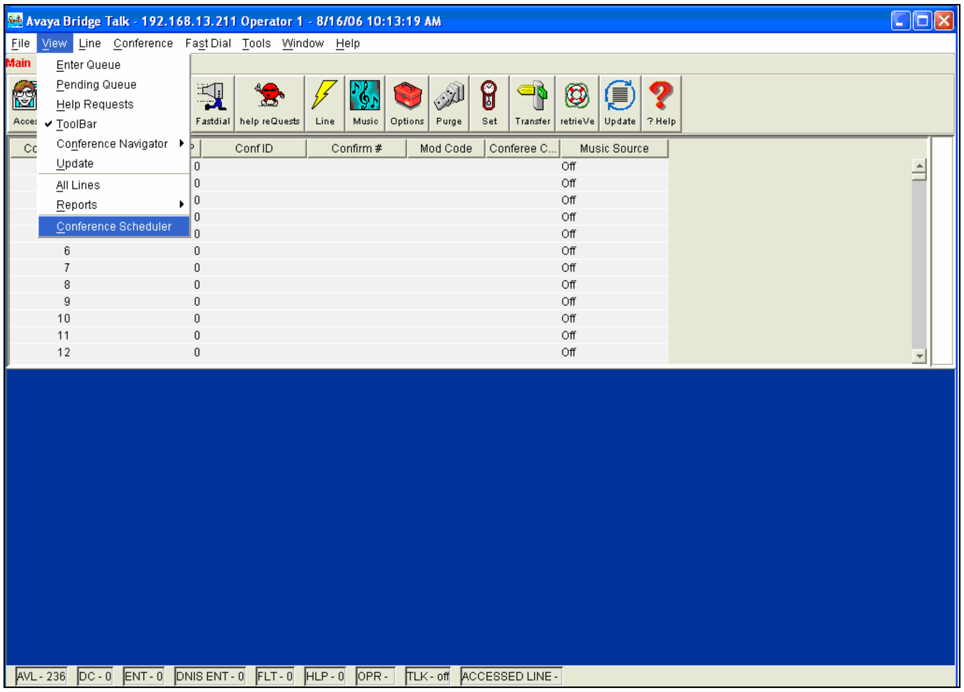
Step	Description
3.11	Open the Avaya Bridge Talk application and log in to Avaya Meeting Exchange with the appropriate credentials. 

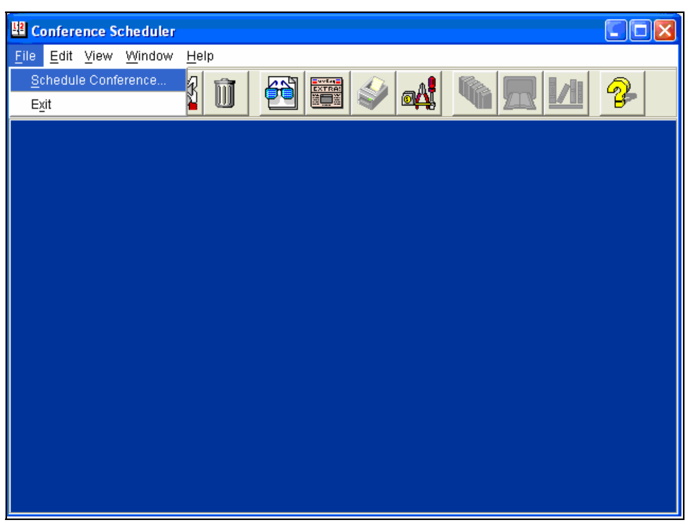
Step	Description
3.12	Provision a dial list that is utilized for Dial-Out (e.g., Blast dial and Fast Dial) from Avaya Meeting Exchange. From the Avaya Bridge Talk menu bar, click **Fast Dial → New**. 

Step	Description
3.13	From the **New Dial List** window that is displayed: • Enter a descriptive name for the **Name** field. • Enable conference participants on the dial list to enter the conference without a passcode by checking the **Directly to Conf** box as shown. • Add entries to the dial list by clicking the **Add** button for each entry. o Give moderator privileges to a conference participant by checking the **Moderator** box. • See **Reference 3** in **Section 7** for provisioning the remaining entries in this screen. • When finished, click the **Save** button on the bottom of the screen. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.14</td><td>Provision a conference with Auto Blast enabled.

From the Avaya Bridge Talk menu bar, click View → Conference Scheduler.


</td></tr>
<tr><td>3.15</td><td>From the Conference Scheduler window that is displayed, click File → Schedule Conference.

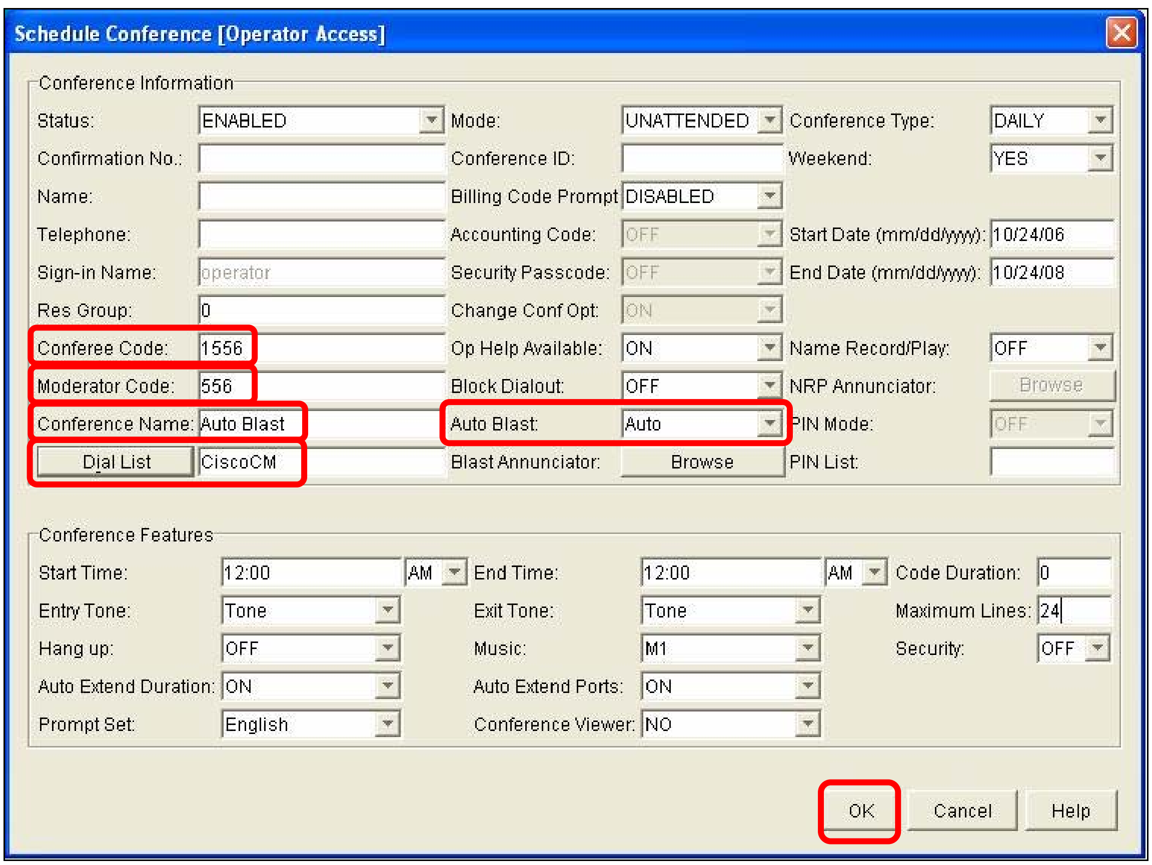

</td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.16</td><td>

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. Enable moderator access without a passcode for this conference call by configuring the following:
    - The **Moderator Code** “**556**” must have an associated **direct call function** provisioned for “**556**” (see **Step 3.9**).

  Note*: This conference remains open for participants to enter as either moderator or participant by entering the appropriate code when prompted.*
- Enter a descriptive name for the **Conference Name** field.
- Administer settings to enable an Auto Blast dial by setting **Auto Blast** to **Auto** and selecting the dial list provisioned in **Step 3.13**.
    - [***Not Shown***] *Select a dial list by clicking the **Dial List** button ➔ select a dial list from the **Create, Select or Edit Dial List** window that is displayed ➔ click the **Select** button.*
- See **Reference 3** in **Section 7** for provisioning the remaining entries in this screen.
- When finished, click the **OK** button on the bottom of the screen.

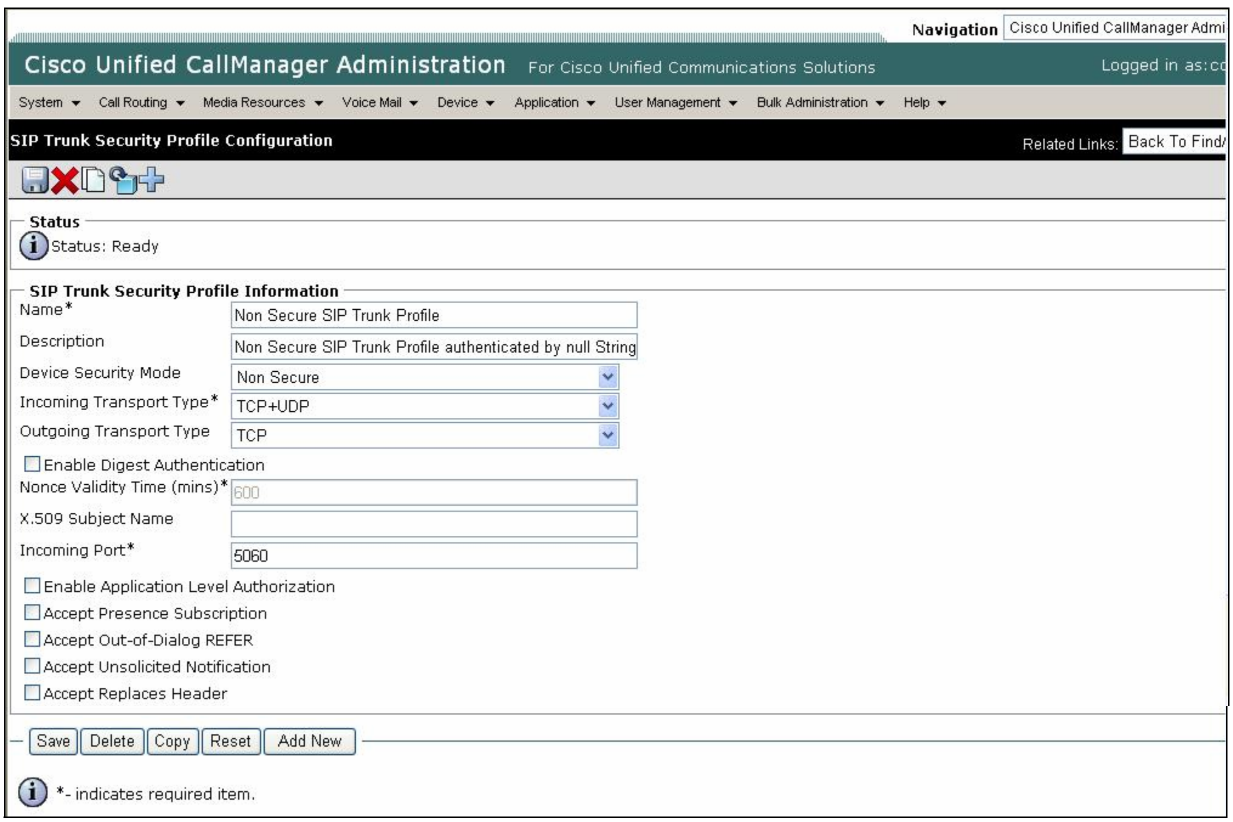


</td></tr>
</table>

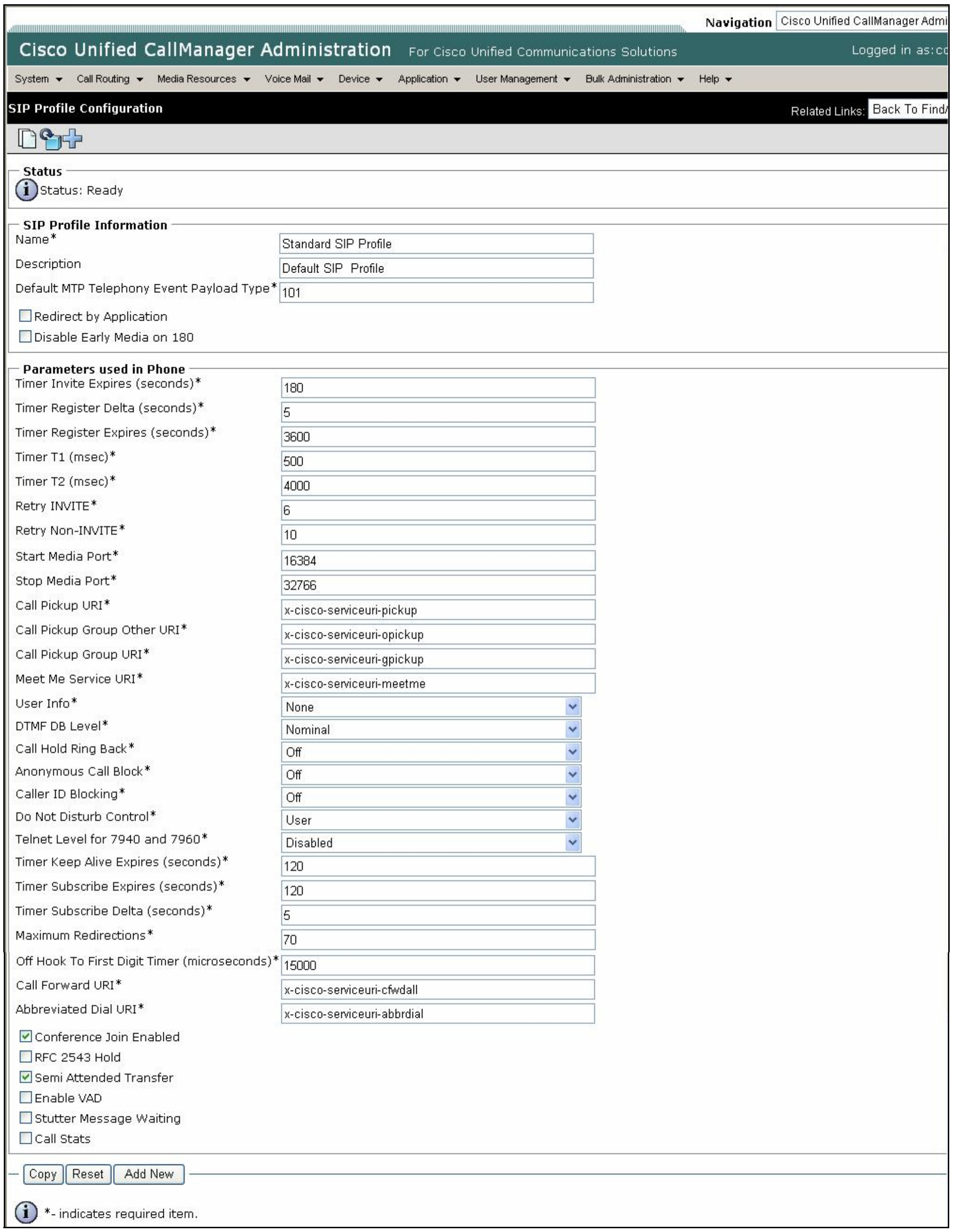
4. Cisco Unified Call Manager Configuration

This section describes the steps for configuring Cisco Unified CallManager to interoperate with Avaya Meeting Exchange via direct SIP trunking (see **Figure 1**).

Step	Description
4.1	Administer settings for Cisco Unified CallManager as follows: • Open a web browser and enter the following URL: **https://<IP Address of Cisco Unified CallManager>** • Log in to Cisco Unified CallManager with the appropriate credentials.
4.2	The following screen shows the **SIP Trunk Security Profile** settings that are utilized for these Application Notes. • View settings by clicking **System → Security Profile → SIP Trunk Security Profile**. • Click the **Find** button [***Not Shown***] to initiate a search for **SIP Trunk Security Profiles**. • When the search is complete, select the appropriate **SIP Trunk Security Profile** from the search results.

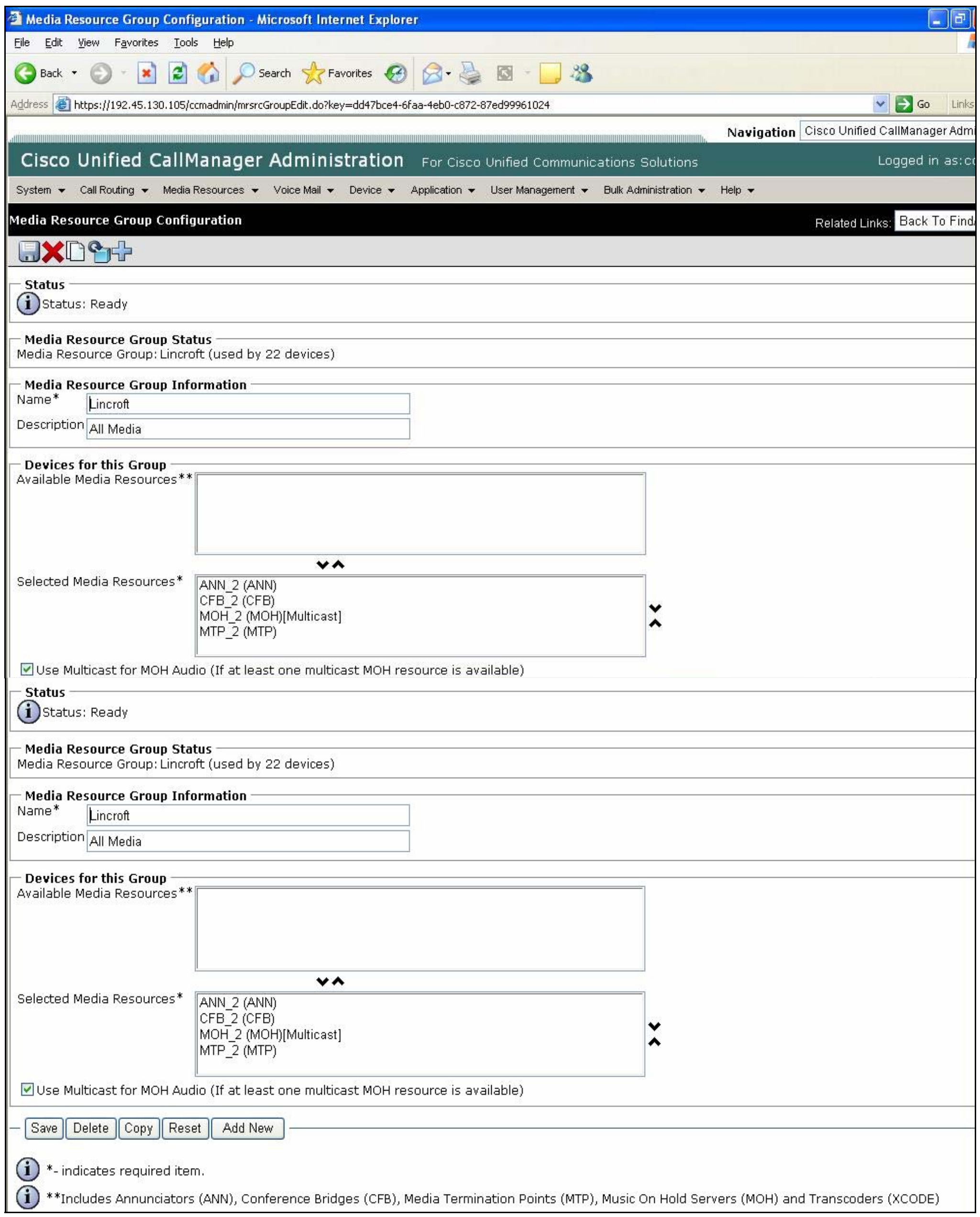


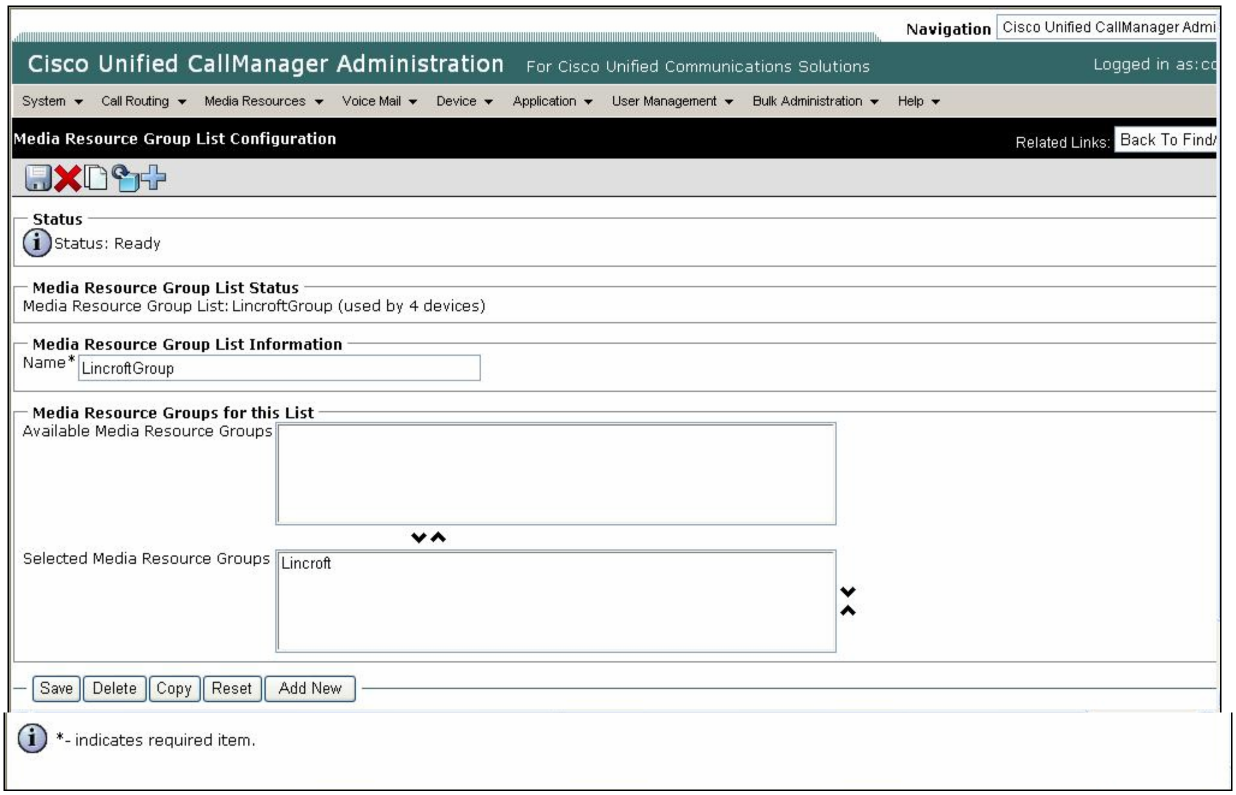
Step	Description
4.3	The following screen shows the **SIP Profile** settings that are utilized for these Application Notes. • View settings by clicking **Device → Device Settings → SIP Profile**. • Click the **Find** button [***Not Shown***] to initiate a search for **SIP Profiles**. • When the search is complete, select the appropriate **SIP Profile** from the search results.



Step	Description
4.4	The following screen shows the **Media Resource Group** settings that are utilized for these Application Notes.

- View settings by clicking **Media Resources → Media Resource Group**.
- Click the **Find** button [***Not Shown***] to initiate a search for **Media Resource Groups**.
- When the search is complete, select the appropriate **Media Resource Group** from the search results.

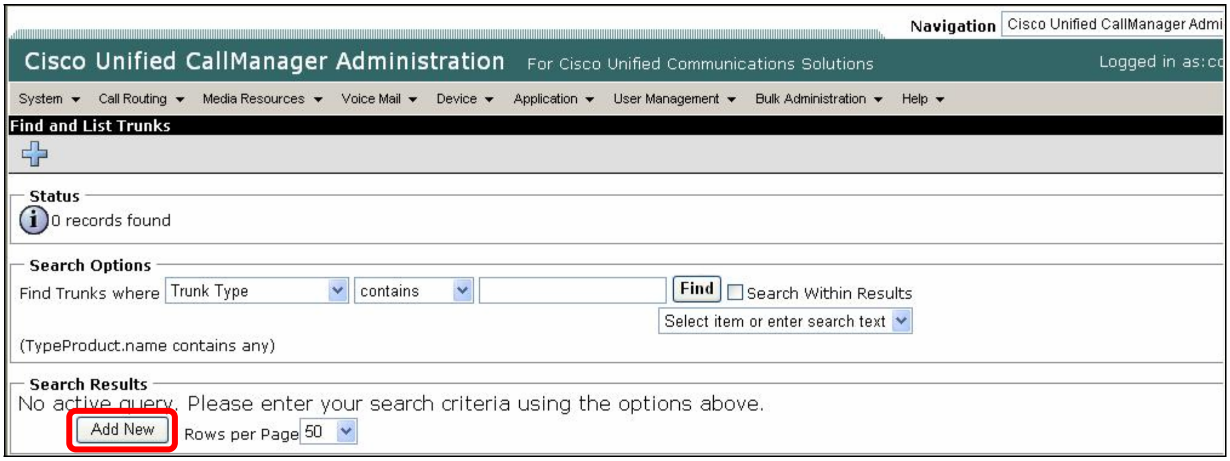


Step	Description
4.5	The following screen shows the **Media Resource Group List** settings that are utilized for these Application Notes. • View settings by clicking **Media Resources → Media Resource Group List**. • Click the **Find** button [***Not Shown***] to initiate a search for **Media Resource Group Lists**. • When the search is complete, select the appropriate **Media Resource Group List** from the search results. ***Note****: The Media Resource Group displayed in **Step 4.4** is utilized below.* 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.6</td><td>

To enable SIP connectivity to Avaya Meeting Exchange, add a **SIP Trunk** as follows:

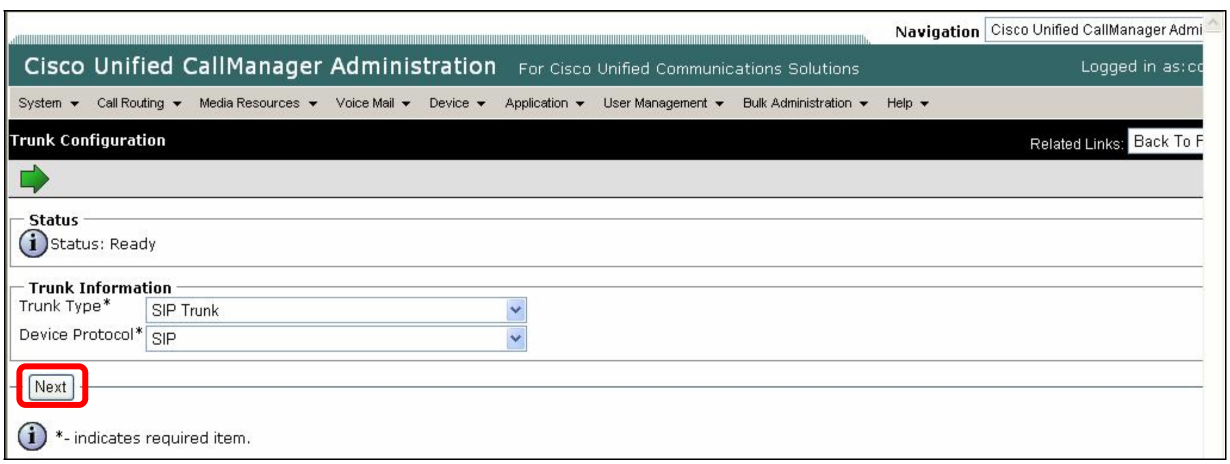
- Click **Device → Trunk**.
- Click the **Add New** button.



</td></tr>
<tr><td>4.7</td><td>

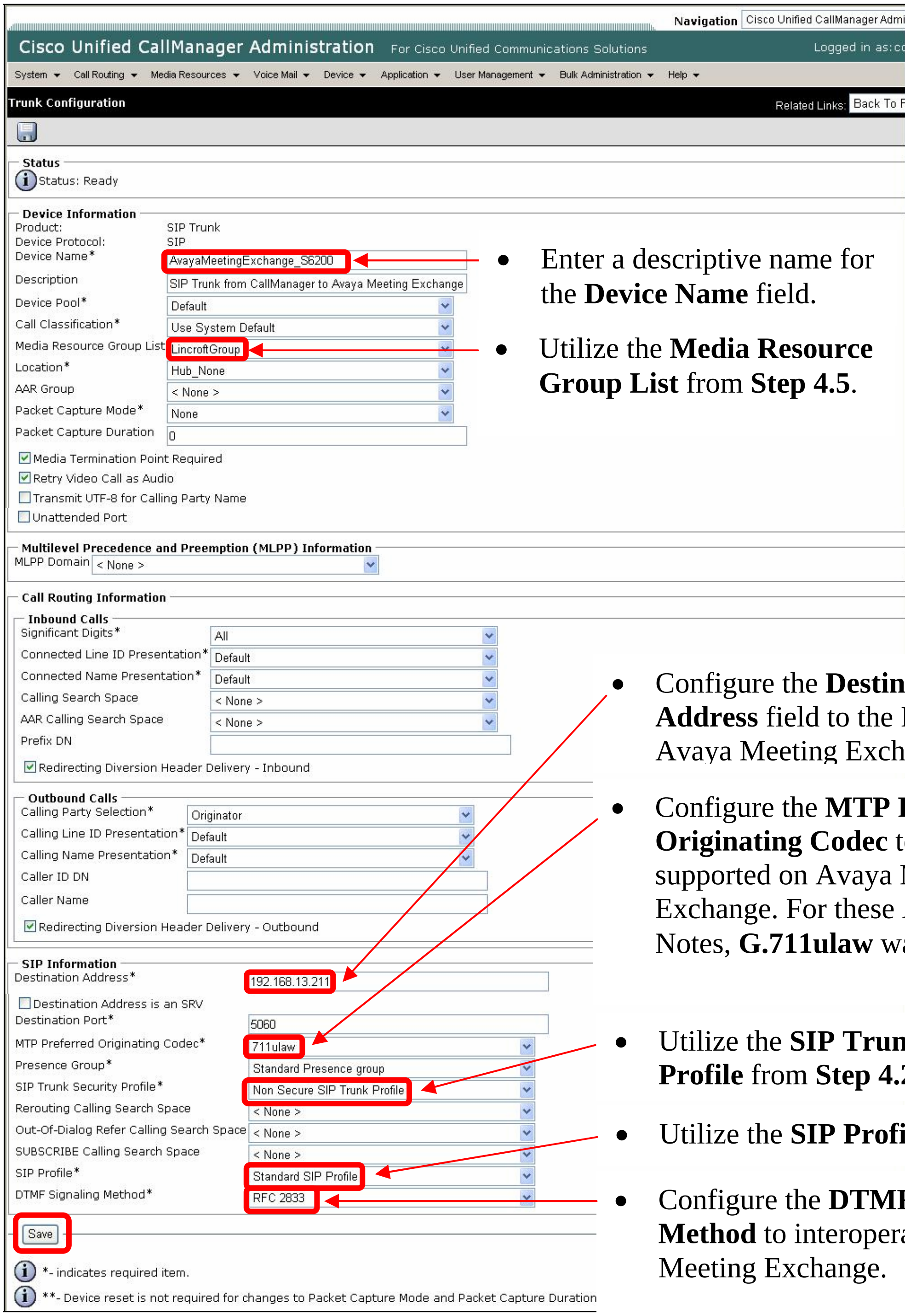
From the **Trunk Configuration** Screen that is displayed:

- Set the **Trunk Type** field to **SIP Trunk**.
- Set the **Device Protocol** field to **SIP**.
- When finished, click the **Next** button.

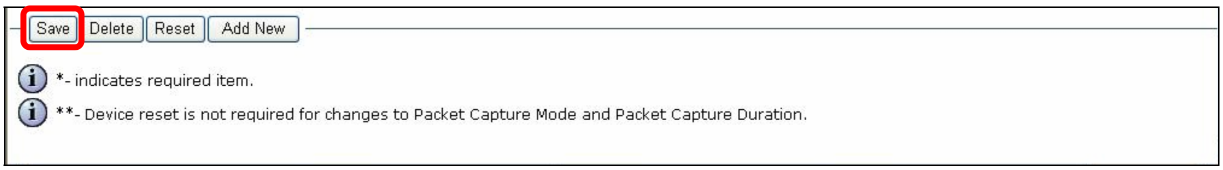


</td></tr>
</table>

Step	Description
4.8	From the **Trunk Configuration** Screen that is displayed, administer settings for the **SIP Trunk** as follows, when finished, click the **Save** button.

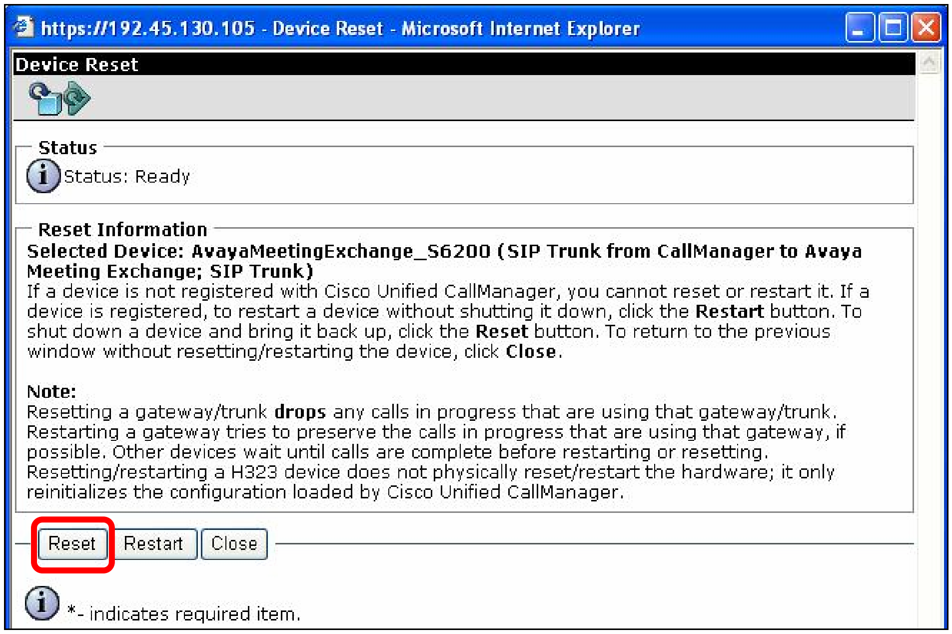


- Enter a descriptive name for the **Device Name** field.
- Utilize the **Media Resource Group List** from **Step 4.5**.
- Configure the **Destination Address** field to the IP address for Avaya Meeting Exchange.
- Configure the **MTP Preferred Originating Codec** to a Codec that is supported on Avaya Meeting Exchange. For these Application Notes, **G.711ulaw** was selected
- Utilize the **SIP Trunk Security Profile** from **Step 4.2**.
- Utilize the **SIP Profile** from **Step 4.3**.
- Configure the **DTMF Signaling Method** to interoperate with Avaya Meeting Exchange.

Step	Description
4.9	Click the **OK** button. 
4.10	The **SIP Trunk** to Avaya Meeting Exchange was added successfully, click the **Reset** button. 

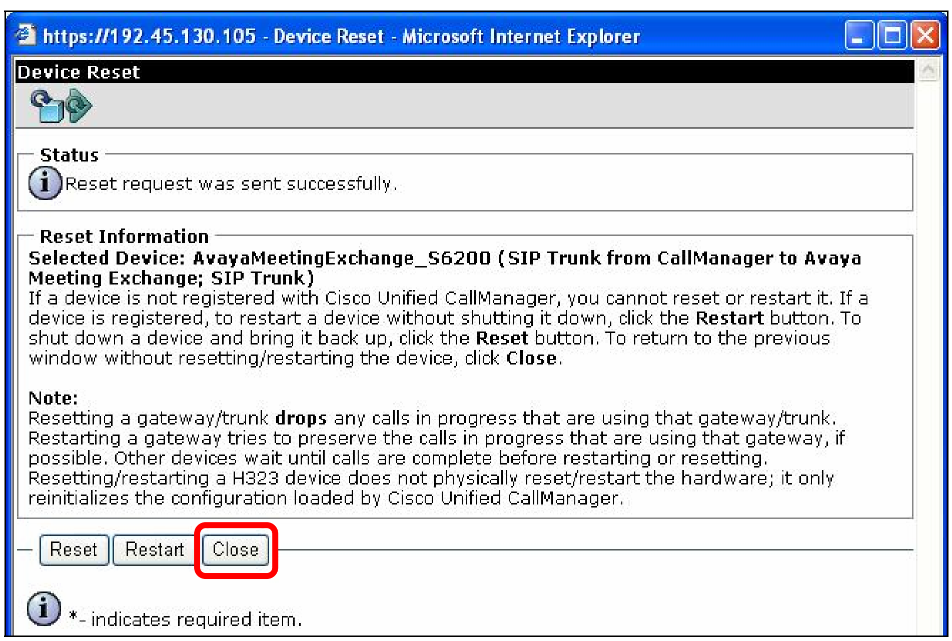
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.11</td><td>

Click the **Reset** button in the pop-up window that is displayed.

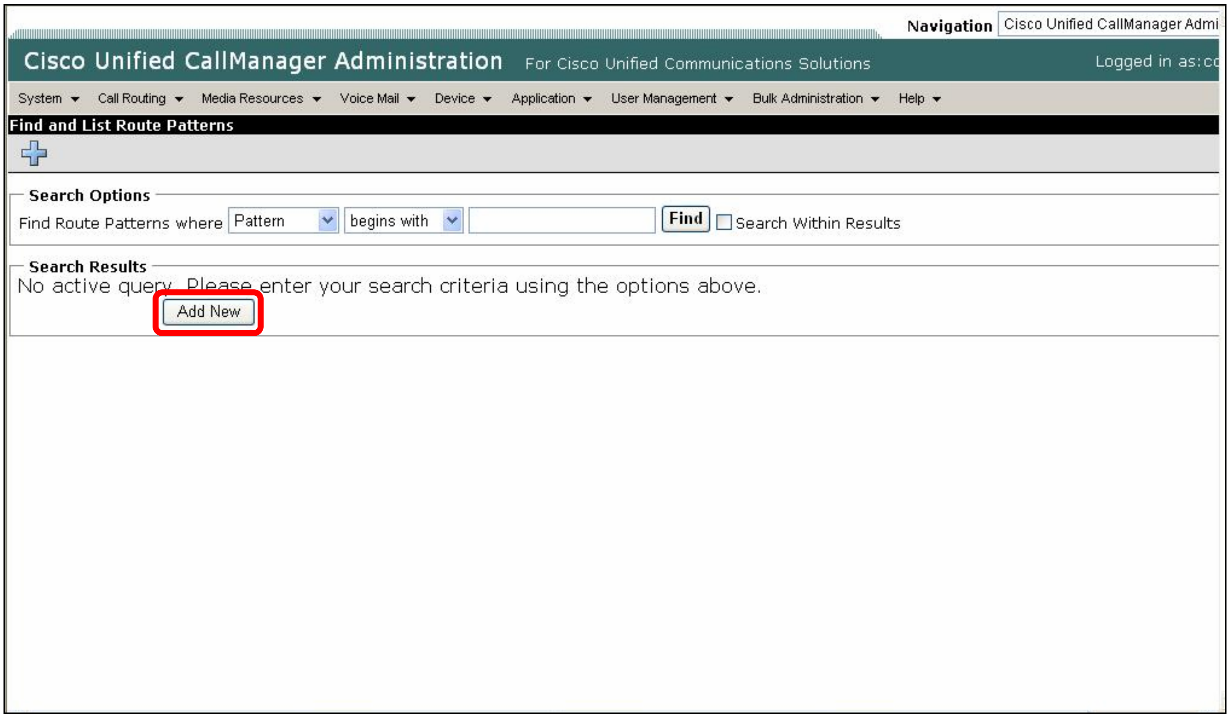


The Reset was successful.

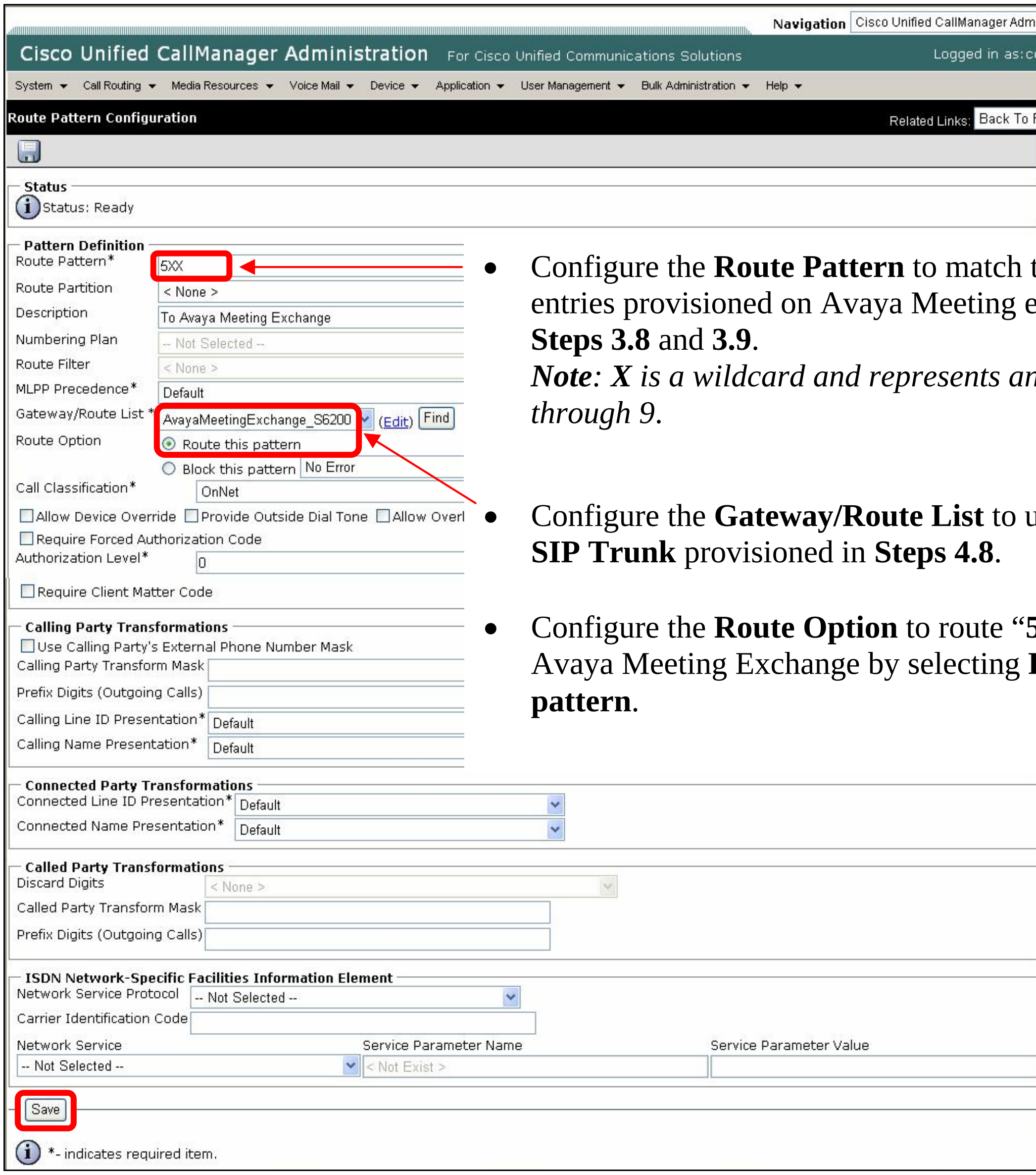
- Click the **Close** button.



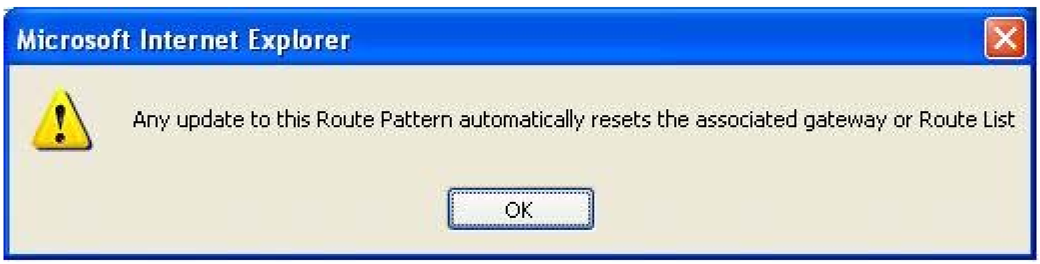
</td></tr>
</table>

Step	Description
4.12	To enable routing from Cisco Unified CallManager to Avaya Meeting Exchange utilizing the SIP Trunk provisioned in **Step 4.8**, add a **Route Pattern** as follows: • Click **Call Routing → Route/Hunt → Route Pattern**. • Click the **Add New** button. 

Step	Description
4.13	From the **Route Pattern Configuration** Screen, administer settings for a **Route Pattern** as follows, when finished, click the **Save** button.



- Configure the **Route Pattern** to match the DNIS entries provisioned on Avaya Meeting exchange in **Steps 3.8** and **3.9**.
  Note*: **X** is a wildcard and represents any digit 0 through 9.*
- Configure the **Gateway/Route List** to utilize the **SIP Trunk** provisioned in **Steps 4.8**.
- Configure the **Route Option** to route "**5XX**" to Avaya Meeting Exchange by selecting **Route this pattern**.

Step	Description
4.14	The **Require Forced Authorization Code** option was not enabled in **Step 4.13**, click the **OK** button to proceed and activate authorization at a later time. 
4.15	Click the **OK** button. 

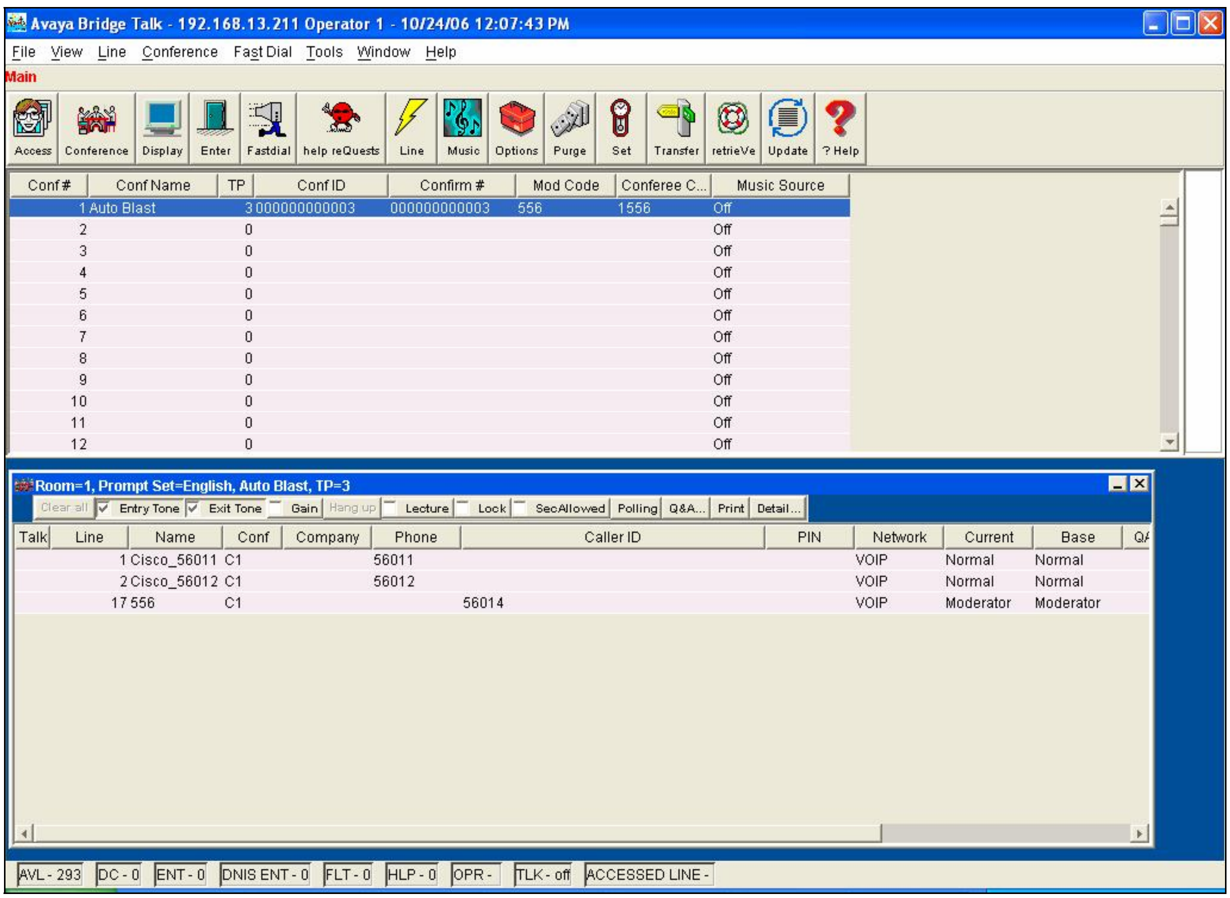
5. Verification Steps

The following steps can be used to verify the configuration described in these Application Notes.

Step	Description
5.1	Log in to the Avaya Meeting Exchange Server console to access the CLI with the appropriate credentials. Run the **dcbps** script to verify all conferencing related processes are running on Avaya Meeting Exchange. • cd to **/usr/dcb/bin** • At the command prompt, run the **dcbps** script and confirm all processes are running by verifying an associated Process ID (PID) for each process.

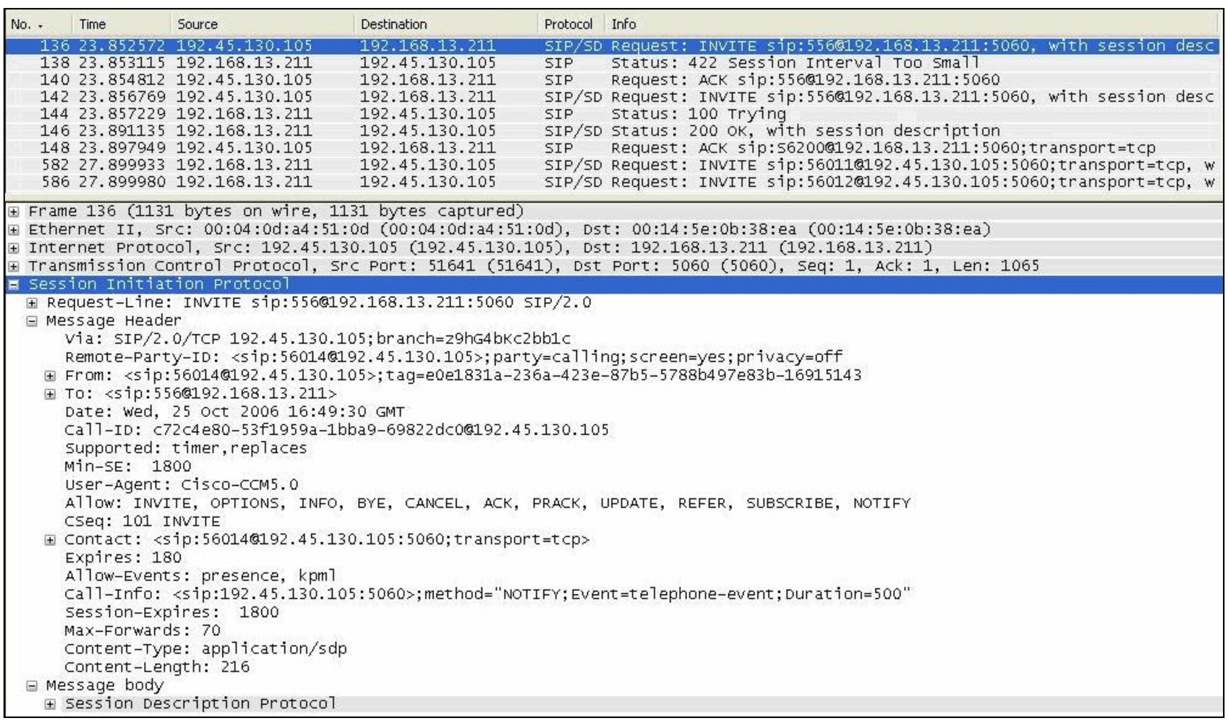
```
S6200>dcbps
  1786   FP 101 ?        0:00 log
  1776   FP 144 ?        0:01 initdcb
  1787   FP 101 ?        0:00 bridgeTr
  1788   FP 105 ?        0:00 netservi
  1791   FP 129 ?        0:00 timer
  1792   FP 101 ?        0:00 traffic
  1793   FP 104 ?        0:00 chdbased
  1794   FP 101 ?        0:00 startd
  1795   FP 109 ?        0:00 cdr
  1796   FP 101 ?        0:00 modapid
  1797   FP 101 ?        0:00 schapid
  1798   FP 104 ?        0:00 callhand
  1799   FP 139 ?        0:00 initipcb
  1800   FP 139 ?        0:00 sipagent
  1801   FP 139 ?        0:00 msdispat
  1802   FP 158 ?        0:00 softms
  1803   FP 139 ?        0:00 serverCo
  1554   TS  80 ?        0:00 sqlexecd with 5 children
```

Step	Description
5.2	The primary focus of this step is to verify SIP trunking between Avaya Meeting Exchange and Cisco Unified CallManager. This is accomplished by placing calls to and from Avaya Meeting Exchange. This step utilizes the Avaya Bridge Talk application to verify calls to and from Avaya Meeting Exchange are managed correctly, e.g., callers are added/removed from conferences. This step will also verify conferencing applications provisioned in **Section 3**. • From an endpoint registered to Cisco Unified CallManager, Dial **556** to enter an conference as **Moderator** (without passcode) while simultaneously invoking the associated Auto Blast dial feature for this conference (see **Step 3.16**). • If not already logged on, log in to the Avaya Bridge Talk application with the appropriate credentials. • **Double-Click** 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.



Step	Description

5.3	Below is a SIP trace of the scenario invoked in **Step 5.2**. This trace is intended to display the provisioning presented in these Application Notes.
	• An endpoint registered to Cisco Unified CallManager (**192.45.130.105**) Dials-In (e.g., sends an **INVITE**, message **No. 136** in the SIP trace) to Avaya Meeting Exchange (**192.168.13.211**).
	    o The Route Pattern “5XX” configured for Cisco Unified CallManager in **Step 4.13** matches both the **Request Line** and **To** fields in the SIP trace.
	      Request-Line: INVITE sip:**556**@192.168.13.211
	      To: <sip:**556**@192.168.13.211>
	• There is a **422 Session Timer Too Small** response from Avaya Meeting Exchange to the initial INVITE from Cisco Unified CallManager. This is due to setting the minSETimerValue to 86400 seconds on Avaya Meeting Exchange in **Step 3.2**. This value is greater that the Min-SE (1800 seconds) used by Cisco Unified CallManager and thus the 422 response from Avaya Meeting Exchange to negotiate a new value for the Min-SE.
	• The **ACK** from Cisco Unified CallManager (message **No. 148** in the SIP trace) uses the **respContact** provisioned on Avaya Meeting Exchange in **Step 3.2**.
	• Avaya Meeting Exchange sends two INVITES (messages **No. 582** and **No. 586** in the SIP trace) to endpoints registered to Cisco Unified CallManager.
	    o The **Request Line** in both **INVITE** messages utilize the parameters in the **telnumToUri.tab** file configured in **Step 3.4**.



```
No.  Time       Source          Destination     Protocol Info
136 23.852572 192.45.130.105   192.168.13.211   SIP/SD Request: INVITE sip:556@192.168.13.211:5060, with session desc
138 23.853115 192.168.13.211   192.45.130.105   SIP    Status: 422 Session Interval Too Small
140 23.854812 192.45.130.105   192.168.13.211   SIP    Request: ACK sip:556@192.168.13.211:5060
142 23.856769 192.45.130.105   192.168.13.211   SIP/SD Request: INVITE sip:556@192.168.13.211:5060, with session desc
144 23.857229 192.168.13.211   192.45.130.105   SIP    Status: 100 Trying
146 23.891135 192.168.13.211   192.45.130.105   SIP/SD Status: 200 OK, with session description
148 23.897949 192.45.130.105   192.168.13.211   SIP    Request: ACK sip:S6200@192.168.13.211:5060;transport=tcp
582 27.899933 192.168.13.211   192.45.130.105   SIP/SD Request: INVITE sip:56011@192.45.130.105:5060;transport=tcp, w
586 27.899980 192.168.13.211   192.45.130.105   SIP/SD Request: INVITE sip:56012@192.45.130.105:5060;transport=tcp, w

Frame 136 (1131 bytes on wire, 1131 bytes captured)
Ethernet II, Src: 00:04:0d:a4:51:0d (00:04:0d:a4:51:0d), Dst: 00:14:5e:0b:38:ea (00:14:5e:0b:38:ea)
Internet Protocol, Src: 192.45.130.105 (192.45.130.105), Dst: 192.168.13.211 (192.168.13.211)
Transmission Control Protocol, Src Port: 51641 (51641), Dst Port: 5060 (5060), Seq: 1, Ack: 1, Len: 1065
Session Initiation Protocol
  Request-Line: INVITE sip:556@192.168.13.211:5060 SIP/2.0
  Message Header
    Via: SIP/2.0/TCP 192.45.130.105;branch=z9hG4bKc2bb1c
    Remote-Party-ID: <sip:56014@192.45.130.105>;party=calling;screen=yes;privacy=off
    From: <sip:56014@192.45.130.105>;tag=e0e1831a-236a-423e-87b5-5788b497e83b-16915143
    To: <sip:556@192.168.13.211>
    Date: Wed, 25 Oct 2006 16:49:30 GMT
    Call-ID: c72c4e80-53f1959a-1bba9-69822dc0@192.45.130.105
    Supported: timer,replaces
    Min-SE:  1800
    User-Agent: Cisco-CCM5.0
    Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
    CSeq: 101 INVITE
    Contact: <sip:56014@192.45.130.105:5060;transport=tcp>
    Expires: 180
    Allow-Events: presence, kpml
    Call-Info: <sip:192.45.130.105:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
    Session-Expires:  1800
    Max-Forwards: 70
    Content-Type: application/sdp
    Content-Length: 216
  Message body
    Session Description Protocol
```

6. Conclusion

These Application Notes provide administrators with the procedures to configure connectivity between the Avaya Meeting Exchange S6200 Conferencing Server and Cisco Unified CallManager utilizing direct SIP trunking.

7. Additional References

Avaya references, available at http://support.avaya.com

1. *Meeting Exchange 4.1 Administration and Maintenance S6200/S6800 Media Server*, Issue 1, Doc ID 04-601168, July 2006.
2. *Meeting Exchange 4.1 Configuring S6200, S6500, and S6800 Conferencing Servers*, Issue 1, Doc ID 04-601338, July 2006.
3. *Avaya Meeting Exchange Groupware Edition Version 4.1 User's Guide for Bridge Talk*, Doc ID 04-600878, Issue 2, July 2006.

Cisco reference, available at http://www.cisco.com/techsupport

4. *Cisco CallManager System Guide Release 5.0(1)*, Document #: OL-8140-01.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya Solution & Interoperability Test Lab at interoplabnotes@list.avaya.com