Avaya Solution & Interoperability Test Lab

Configuring Connectivity Between the Avaya Meeting Exchange S6200 Conferencing Server, Cisco Unified Communications Manager and the PSTN via the AudioCodes Mediant 2000 - Issue 0.1

Abstract

These Application Notes describe the procedures for configuring connectivity between the Avaya Meeting Exchange S6200 Conferencing Server (Meeting Exchange), Cisco Unified Communications Manager (UCM) and the PSTN via the AudioCodes Mediant 2000. Employing this configuration enables call origination/termination between Avaya Meeting Exchange and endpoints registered to Cisco UCM, as well as endpoints on the PSTN via the AudioCodes Mediant 2000.

1. Introduction

These Application Notes describe the procedures for configuring connectivity between the Avaya Meeting Exchange S6200 Conferencing Server (Meeting Exchange), Cisco Unified Communications Manager (UCM) and the PSTN via the AudioCodes Mediant 2000. Employing this configuration enables call origination/termination between Avaya Meeting Exchange and endpoints registered to Cisco UCM, as well as endpoints on the PSTN via the AudioCodes Mediant 2000.

Figure 1 illustrates the sample configuration utilized for these Application Notes.
Avaya Meeting Exchange is a SIP based voice conferencing solution that provides mid-market enterprise customers with an audio conferencing system that can reside an IP network. For this sample configuration, Avaya Meeting Exchange is provisioned to accept calls from Cisco UCM and the PSTN via the AudioCodes Mediant 2000 through call branding that supported both direct and scan call flows. A direct call flow allows access to conferences provisioned on Avaya Meeting Exchange without entering a passcode. Conversely, to enter a conference via a basic call flow requires a passcode. Avaya Meeting Exchange was also administered for call origination via SIP signaling to endpoints registered to Cisco UCM as well as endpoints on the PSTN via the AudioCodes Mediant 2000.

Cisco UCM provides telephony features for the IP telephones present in this sample configuration. Cisco UCM is provisioned for call origination via SIP signaling to Avaya Meeting Exchange.

The AudioCodes Mediant 2000 is a SIP-based VoIP gateway, offering integrated voice gateway functionality over IP networks. This solution addresses mid-density applications deployed in IP networks by delivering up to 480 simultaneous Voice over IP (VoIP) or Fax over IP (FoIP) calls. The AudioCodes Mediant 2000 routes calls over the IP network using SIP signaling protocol, enabling the deployment of Voice over Packet solutions to PSTN subscribers. The AudioCodes Mediant 2000 was provisioned for call origination via SIP signaling to Avaya Meeting Exchange. The AudioCodes Mediant 2000 was also administered to provide call routing to an emulated PSTN via T1 ISDN-PRI connectivity.

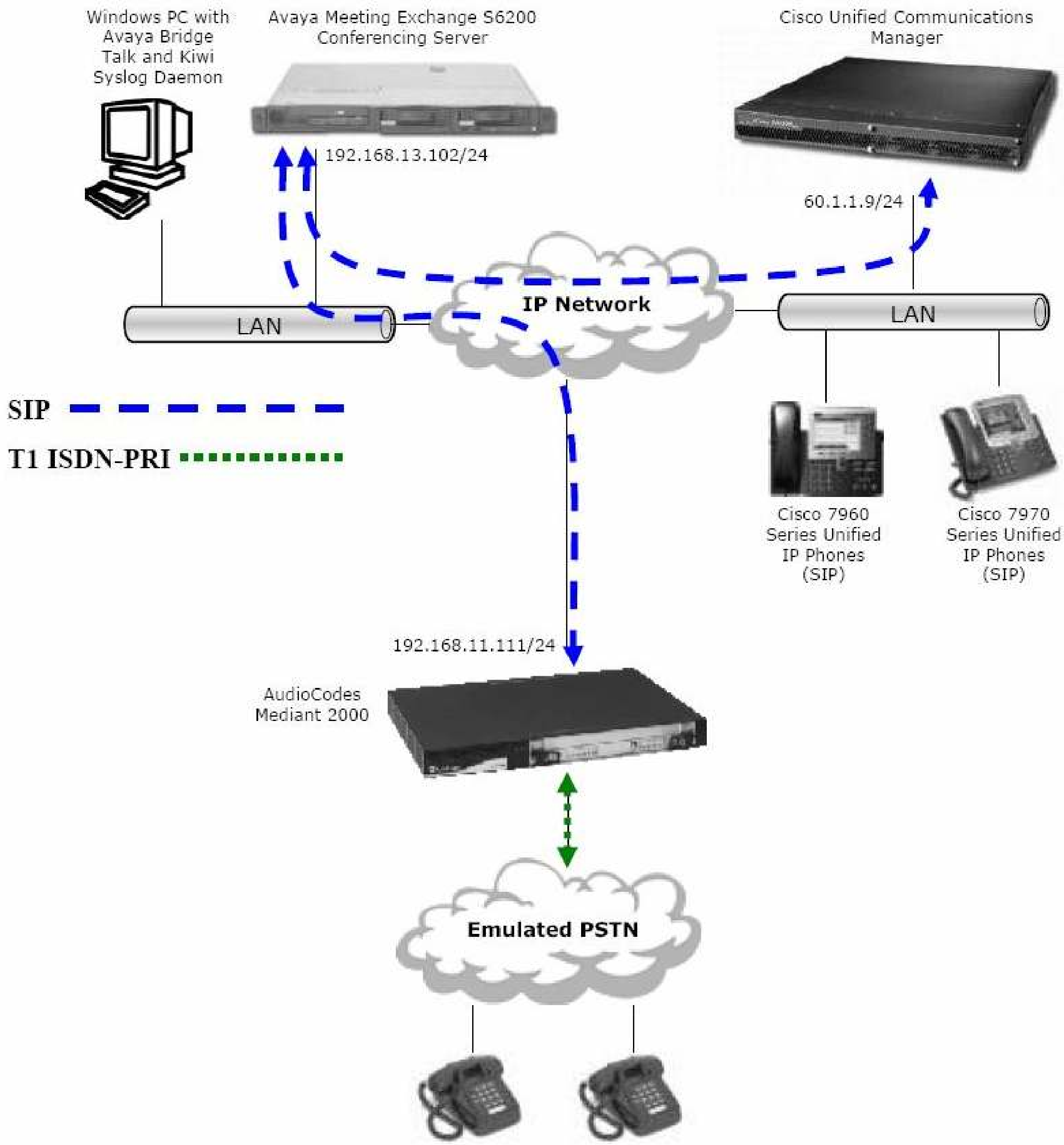


Figure 1: Sample Configuration

2. Equipment and Software Validated

The following equipment and software versions are used for this sample configuration:

Equipment	Software Version
Avaya Meeting Exchange S6200 Conferencing Server	MX 5.0 SP1 (mx5.0.1.0.18)
Avaya Bridge Talk	5.0 Build 11
Cisco Unified Communications Manager	CUCM 6.0 (6.0.1.2000-3)
Cisco 7960 Series IP Phones (SIP)	P0S3-08-6-02
Cisco 7970 Series IP Phones (SIP)	SIP70.8-3-1S
AudioCodes Mediant 2000 • Version ID • DSP Type • DSP Software Version • DSP Software Name • Flash Version • Module Firmware	 5.20A.039.001 2 52016 624AE3 192 0x32
Kiwi Syslog Daemon	V8.1.6

Table 1: Equipment and Software Versions

3. Avaya Meeting Exchange Configuration

This section describes the configuration for enabling Avaya Meeting Exchange to interoperate with Cisco UCM and the AudioCodes Mediant 2000. Call routing, call branding and SIP connectivity are administered on Avaya Meeting Exchange via a Command Line Interface (CLI) accessed via Secure Shell (SSH). Conference related attributes are administered and maintained via the Avaya Bridge Talk application. Refer to [**1**], [**2**] and [**3**] for additional information regarding the administration of Avaya Meeting Exchange.

3.1. Configure Connectivity

This section describes the steps for configuring SIP connectivity between Avaya Meeting Exchange and both Cisco UCM and the AudioCodes Mediant 2000. The provisioning depicted in this section was administered via the CLI.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td rowspan="3">3.1.1</td><td>Administer settings that enable SIP connectivity between Avaya Meeting Exchange and Cisco UCM and the AudioCodes Mediant 2000 by editing the system.cfg file as follows:

From the /usr/ipcb/config directory, edit the system.cfg file with a text editor.
Enter the IP address of Avaya Meeting Exchange (as defined in the /etc/hosts file) for the IPAddress variable.
Enter a SIP Uniform Resource Identifier (URI) for Avaya Meeting Exchange that conforms to SIP standards for the MyListener variable. This entry is used to populate the “From” Header Field in SIP INVITE messages from Avaya Meeting Exchange. The “User” Field, S6200, must conform to SIP standards and is selected to uniquely identify this server. For example, S6200 will be inserted in the “From” Header Field of SIP INVITE messages from Avaya Meeting Exchange and will display on a telephone when a call originates from Avaya Meeting Exchange.
Enter a SIP-URI that conforms to SIP standards and is bounded by angled brackets for the respContact variable. This variable is used to populate the “Contact” Header Field in SIP Response messages from Avaya Meeting Exchange and provides Cisco UCM and the AudioCodes Mediant 2000 a SIP-URI for acknowledging SIP messages from Avaya Meeting Exchange.
Note: if the respContact variable is not configured, Avaya Meeting Exchange will populate the “Contact” Header Field with a default SIP-URI. To enable SIP connectivity over UDP, set the transport-param to udp.
Enter a value in seconds for the minSETimerValue variable. This entry corresponds to the lower bound of the session interval as it pertains to the SIP standards. It is recommended to provision the minSETimerValue variable to a setting that is greater than or equal to the corresponding setting on SIP User Agents (UA) that are intended to interoperate with Avaya Meeting Exchange.
</td></tr>
<tr><td><pre>
# ip address of the server
IPAddress=192.168.13.102

request we will be listening to
MyListener=sip:S6200@192.168.13.102

# if this setting is populated will Overwrite the contact field in responses
# for SIP/UDP: respContact=<sip:S6200@192.168.13.102:5060;transport=udp>
respContact=<sip:S6200@192.168.13.102:5060;transport=tcp>

# Min SE value in seconds for lower bound of Session Interval for SIP Invite
minSETimerValue=1800
</pre></td></tr>
<tr><td></td></tr>
</table>

3.2. Configure Call Routing

The provisioning depicted in this section was administered via the CLI and describes the steps to enable call routing for Avaya Meeting Exchange, where call routing is defined as follows:

- For call termination on Avaya Meeting Exchange, URI to telephone number translations are utilized. These translations associate calls to Avaya Meeting Exchange with corresponding call branding, based on incoming SIP-URIs.
- For call origination from Avaya Meeting Exchange, telephone number to URI translations are utilized. These translations associate a telephone number pattern with a corresponding SIP-URI of a SIP UA, thus allowing call origination from Avaya Meeting Exchange to the SIP UA.

Step	Description
3.2.1	Administer settings to associate incoming calls to Avaya Meeting Exchange with corresponding call branding by adding URI to telephone number translations to the **UriToTelnum.tab** file. These translations extract values for both the Direct Inward Dial (DID, also known as DDI in Europe) and the Automatic Number Identification (ANI).

- From the **/usr/ipcb/config** directory, edit the **UriToTelnum.tab** file with a text editor.
- Add rules, separated by either tabs or single spaces, as a line in the file to match the pattern of the “To” and “From” Header Fields in SIP INVITE messages from either Cisco UCM or the AudioCodes Mediant 2000. If the match is successful, the DID is extracted from the “To” Header Field and the ANI is extracted from the “From” Header Field. Metacharacters such as “*****” or “**?**” may be utilized.
    - The rules under the **TelnumPattern** and **TelnumConversion** columns work in conjunction as follows. Assume Cisco UCM sends a SIP INVITE message with the following “To” and “From” Header Fields. The rule **"*<sip:*@*"** matches the following:
        - To: <sip:*555*@192.168.13.102>, where **$2** utilizes *555*, the variable matched by the second asterisk as the DID value for the call.
        - From: <sip:*917325551236*@60.1.1.9>, where **$2** utilizes *917325551236*, the variable matched by the second asterisk as the ANI for the call.
- [***Not Required***] *Add rules to support operator dial-in. Refer to [2] for information regarding this feature. For this sample configuration, **"*<sip:501@*"** is utilized.*
- Enable an undefined caller to receive a prompt for operator assistance by adding an entry for a wildcard as the last line in this file. This entry accounts for the condition of an unmatched “To” Header Field.

***Note**: Entries in this file are read sequentially, therefore, the entry for the wildcard 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.*

```
# request URI to telnum conversion table
#
# This table converts the Request URI in the SIP INVITE request to the
# appropriate value specified when a pattern is matched. For example, if the
# request Uri was "<sip:3333@10.220.10.4>" and one of the patterns was
# "<sip:*@*" a match would take place. If the conversion for that match was
# $1 then 3333 would be passed as the ddi for the call. If the conversion for
# that match were "0000" then 0000 would be passed as ther ddi for the call.
#
#THE COMMENT COLLUM OR ANY OF THE COLLUMS SHOULD HAVE NO SPACES

TelnumPattern       TelnumConversion       comment
"*<sip:501@*"       "OP501x1"              Op1_From_CiscoUCM
"*<sip:*@*"         $2                     CiscoUCM
*                   $0                     wildcard
```

Step	Description

3.2.2	Administer settings to enable call origination from Avaya Meeting Exchange to both Cisco UCM and the AudioCodes Mediant 2000 by adding telephone number to URI translations to the **telnumToUri.tab** file as follows:
	• From the **/usr/ipcb/config** directory, edit the **telnumToUri.tab** file with a text editor.
	• Add rules, separated by either tabs or single spaces, as a line in the file to route calls from Avaya Meeting Exchange to both Cisco UCM and the AudioCodes Mediant 2000. Metacharacters such as "*****" (refers to a character string) or "**?**" (refers to a single character) may be utilized.
	  o The rules entered under the **TelnumPattern** column match digit patterns with a leading '3' or '9' and correspond to station extensions on Cisco UCM and the PSTN respectively.
	  o The SIP-URI entered under the **TelnumConversion** column route the call to either Cisco UCM or the AudioCodes Mediant 2000 accordingly. To enable SIP connectivity utilizing TCP, the rules must syntactically conform to SIP standards regarding URI and contain **5060** and **transport=tcp**. Avaya Meeting Exchange will replace **$0** with the "dialstring" in outgoing SIP INVITE messages. For example, if *917325551234* is dialed, Avaya Meeting Exchange will format a SIP INVITE message with the following SIP-URI in the Request-Line and "To" Header Field:
	    ▪ sip:*917325551234*@60.1.1.9:5060;transport=tcp
	
	***Note**: Alternatively, call routing to Cisco UCM could have been enabled with the following entry:*
	***9??????????? sip:$0@60.1.1.9:5060;transport=tcp**, where "**?**" is a wildcard and matches any digit. To enable SIP connectivity over UDP, set the transport-param in the SIP-URI to udp. Similarly for the AudioCodes Mediant 2000.*

	# telnum to uri conversion table # # This file is for dialing out from the Bridge to an external party. The # digits that are dialed are converted into the Request URI in the SIP INVITE. # For example, if the digits dialed were 936543 and one of the patterns was # "93????" a match would take place. If the conversion for that match was # $1 then the Request URI for the SIP INVITE would be sip:936543@10.221.11.250 #THE COMMENT COLLUM OR ANY OF THE COLLUMS SHOULD HAVE NO SPACES **TelnumPattern TelnumConversion** comment # 3* sip:$0@192.168.11.10:5060;transport=udp M2K_PSTN_UDP # 9* sip:$0@60.1.1.9:5060;transport=udp CiscoUCM_UDP **3* sip:$0@192.168.11.10:5060;transport=tcp M2K_PSTN_TCP** **9* sip:$0@60.1.1.9:5060;transport=tcp** CiscoUCM_TCP

Step	Description
3.2.3	Restart conferencing related processes on Avaya Meeting Exchange for updates to take effect. At the command prompt, enter "**service mx-bridge restart**". `[S6200]> service mx-bridge restart`

3.3. Configure Call Branding

The following steps provide examples of how to provision direct and scan call branding by utilizing the Call Branding Utility (CBUTIL) on Avaya Meeting Exchange. 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 Dialed Number Identification Service (DNIS) entry. Avaya Meeting Exchange parses these entries in numerically ascending order, with the wildcard character "?" last in the list. For example, 129? follows 1299. The last entry in the table consists entirely of wildcard characters.

Step	Description

3.3.1	Administer call branding for a direct call flow as follows:
	• From the **/usr/dcb/bin** directory, add an entry to the call branding table to map the DID value obtained from procedures in **Step 3.2.1** to a conference by entering "**cbutil add 555 0 301 1 n direct**" at the command prompt. The syntax for this command is case insensitive and is defined as follows:
	cbutil add <dnis> <rg> <msg> <ps> <ucps> <func> [-l <ln> -c <cn>], where,
	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 -l <"ln"> Optional line name to associate with caller
	o -c <"cn"> Optional company name to associate with caller
	`S6200-> cbutil add 555 0 301 1 n direct`
	`cbutil`
	`Copyright 2004 Avaya, Inc. All rights reserved.`

3.3.2	Repeat **Step 3.3.1** to add an entry to the call branding table for a scan call flow.
	`S6200-> cbutil add 500 0 1 1 n scan`
	`cbutil`
	`Copyright 2004 Avaya, Inc. All rights reserved.`

Step	Description
3.3.3	At the command prompt, enter "**cbutil list**" to verify the entries provisioned in **Step 3.3.1** and **Step 3.3.2**. ***Note****: The last entry in the call branding table, with a* ***DNIS*** *value* ***???****, was added previously and is a wild card entry. This entry captures any wrong number (e.g., unmatched* ***DID*** *values) and places the call into the enter queue for operator assistance.*

	<pre>S6200-> **cbutil list**
	cbutil
	Copyright 2004 Avaya, Inc. All rights reserved.
	
	DNIS Grp Msg PS CP Function Line Name Company Name
	---------------- --- --- --- -- -------- -------------------- -----------------
	500 0 1 1 N SCAN
	555 0 301 1 N DIRECT
	??? 0 208 1 N ENTER</pre>

3.4. Administer Conferences

The following steps utilize Avaya Bridge Talk to provision conferences on Avaya Meeting Exchange. Avaya Bridge Talk is an application that runs on a standard Windows based PC and is utilized for provisioning and managing conferencing applications on Avaya Meeting Exchange. Refer to [**3**] for information regarding PC requirements. 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.

Figure 2 illustrates the main window of the Avaya Bridge Talk application. The following is a brief description of the task areas that were utilized for these Application Notes.

1. The Menu Bar, which includes menus for both Avaya Meeting Exchange specific and Windows-based commands.
2. The Main Tool Bar, which includes commands for entering command-line text.
3. The Conference Room, which displays information about features and attributes for individual conferences; and lists participants, moderators and their status.
4. The Conference Navigator, which displays a portion of the conferences currently running on the bridge as well as individual conference attributes or features.

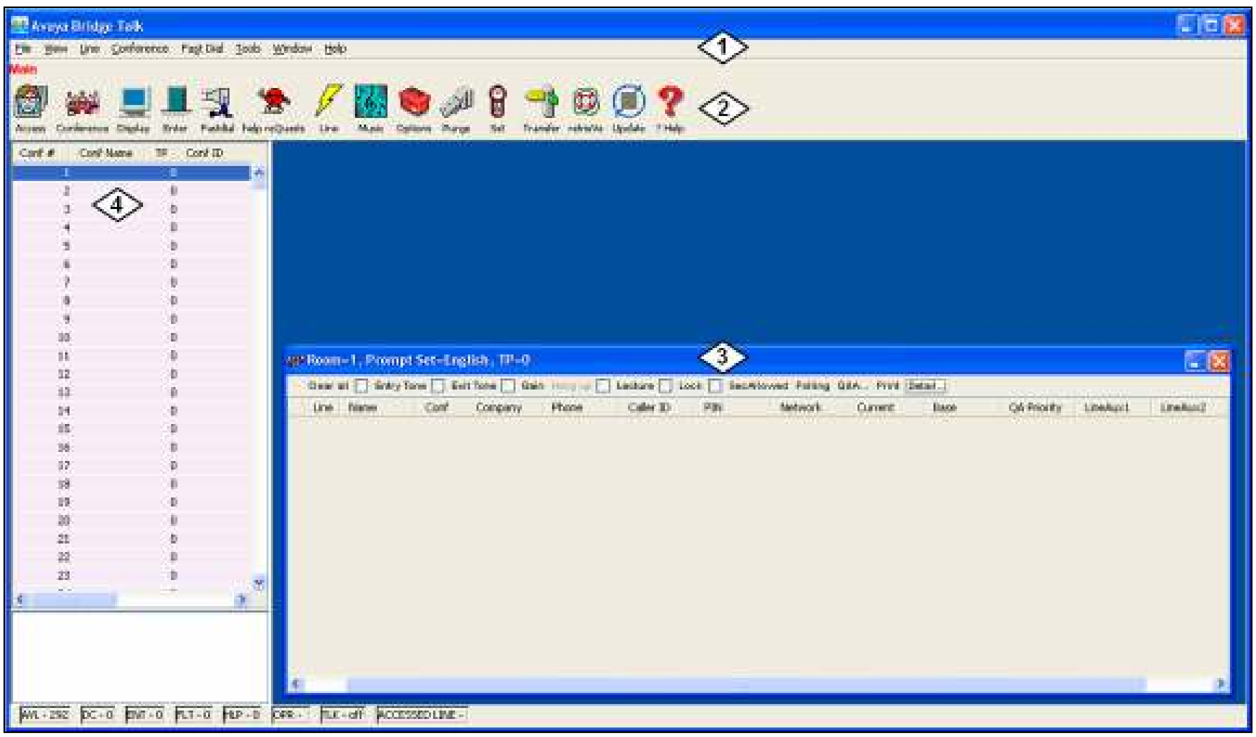


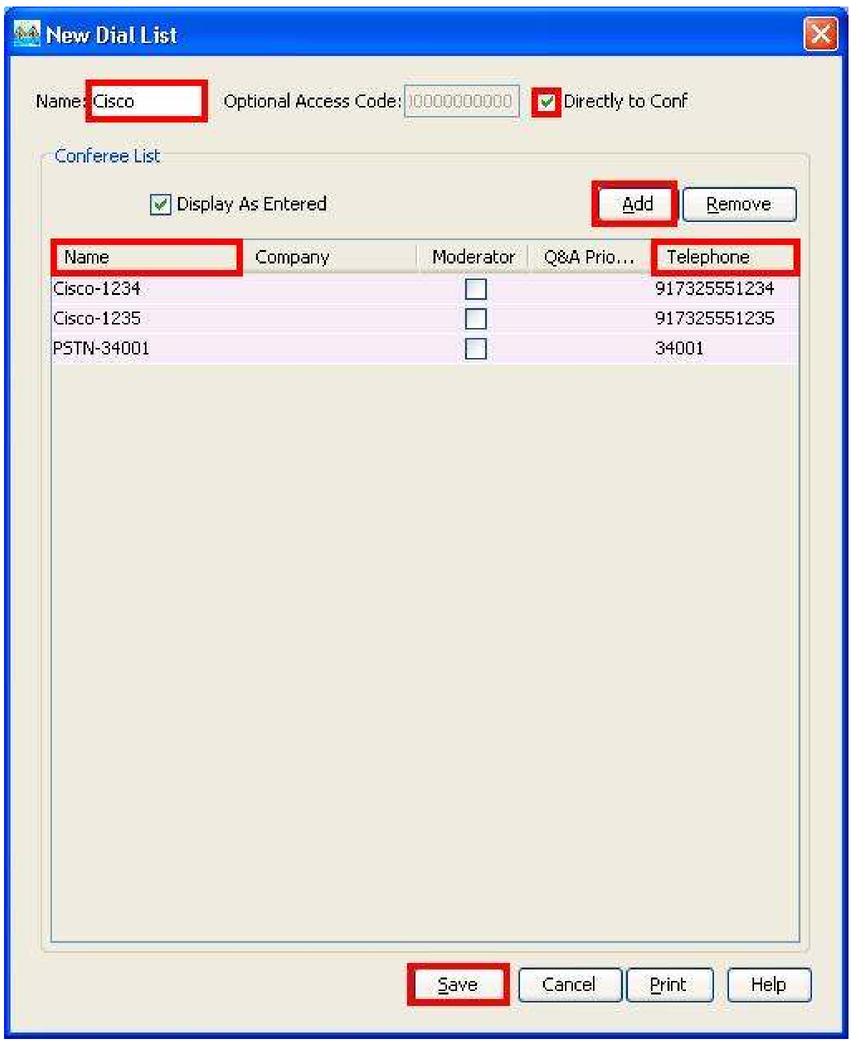
Figure 2: Avaya Bridge Talk Main Window

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.4.1</td><td>Create a dial list of participants on Avaya Meeting Exchange. From the Avaya Bridge Talk Menu Bar, select Fast Dial → New. From the New Dial List window that is displayed, add participants to the dial list as follows:

Enter a descriptive name for this dial list in the Name field.
Add entries to the dial list by clicking Add for each participant.

Enter a descriptive name for each participant in the Name field.
Enter a number in the Telephone field that corresponds to the participants' telephone number.

Enable conference participants on the dial list to enter the conference without a passcode by checking the Directly to Conf box.
Refer to [3] for definitions regarding the remaining fields on this screen.
Click Save.



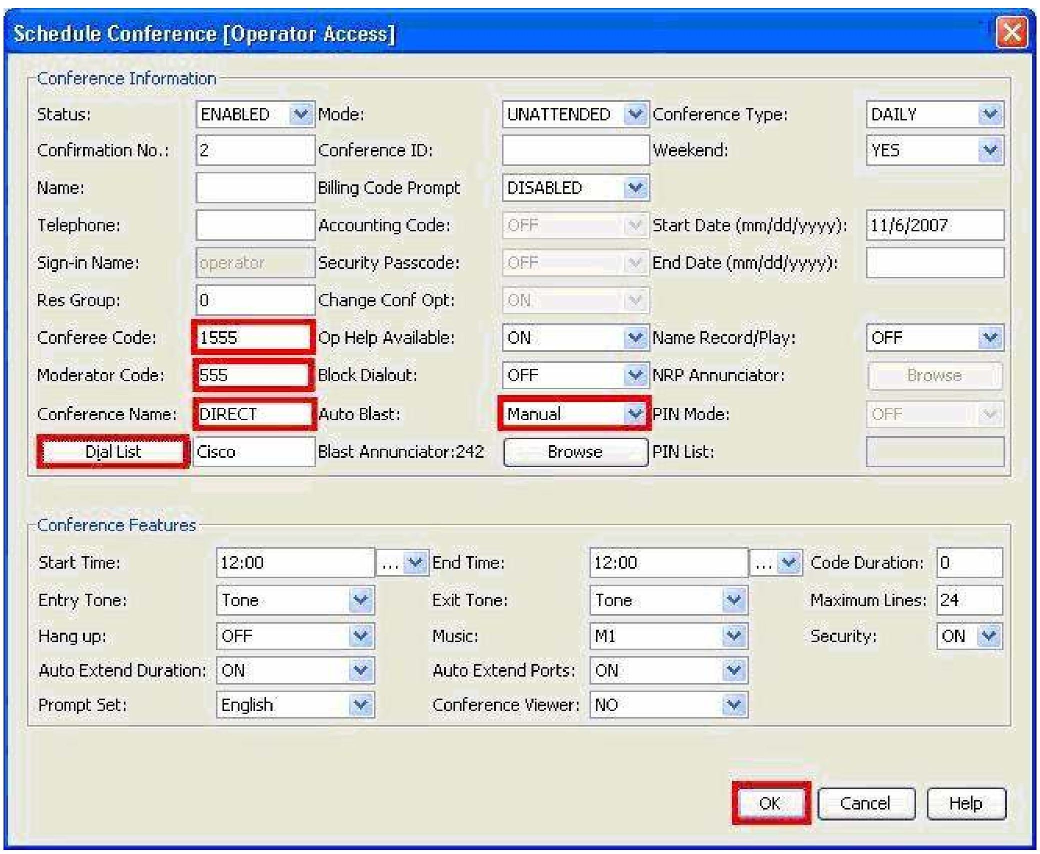
</td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.4.2</td><td>
Schedule conferences that utilize the call branding for a direct call flow provisioned in Step 3.3.1 as follows. From the Menu Bar, click View → Conference Scheduler. From the Conference Scheduler window that is displayed, click File → Schedule Conference. From the Schedule Conference window that is displayed, administer settings as follows:

Enter a unique passcode in the Conferee Code field to allow access to this conference.
Enter a unique passcode in the Moderator Code field to allow access to this conference with moderator/host privileges. Note, to enable access to this conference without entering a passcode, define a Moderator Code that aligns with the provisioning for a direct call flow (see Step 3.3.1).
Enter a descriptive name for this conference in the Conference Name field.
Administer settings to enable a blast dial by setting the Auto Blast field to Manual and selecting the dial list provisioned in Step 3.4.1 in the Dial List field.

Select a dial list by clicking Dial List.
[Not Shown] Select a dial list from the Create, Select or Edit Dial List window that is displayed.

Refer to [3] for definitions regarding the remaining fields on this screen.
Click OK.



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

4. Cisco Unified Communications Manager Configuration

This section describes the configuration for enabling Cisco UCM to interoperate with Avaya Meeting Exchange. Cisco UCM is administered and maintained using a standard web browser over a secure connection by entering **https://<Cisco UCM IP Address or Fully Qualified Domain Name (FQDN)>** into the web browser's Uniform Resource Locator (URL) bar. Refer to [**4**] for additional information regarding the administration of Cisco UCM.

4.1. Configure Connectivity

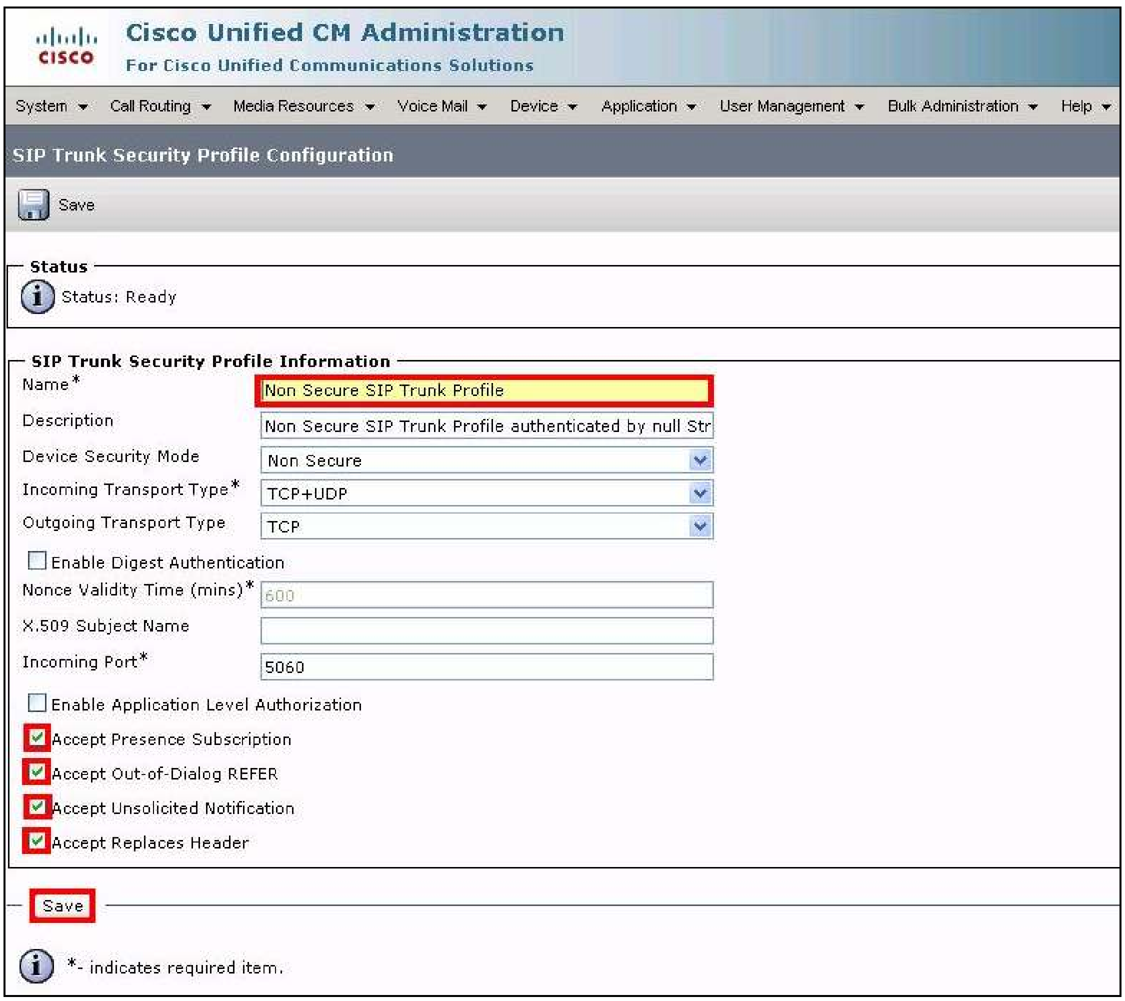
This section describes the steps for configuring SIP connectivity between Cisco UCM and Avaya Meeting Exchange.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.1.1</td><td>

To enable SIP connectivity with Avaya Meeting Exchange utilizing TCP, configure a **SIP Trunk Security Profile** as follows:

- From the Cisco UCM main menu, select **System** ➔ **Security Profile** ➔ **SIP Trunk Security Profile**.
- [***Not Shown***] *Click **Add New** to create a new **SIP Trunk Security Profile**.*
- Provision settings as displayed and click **Save**.

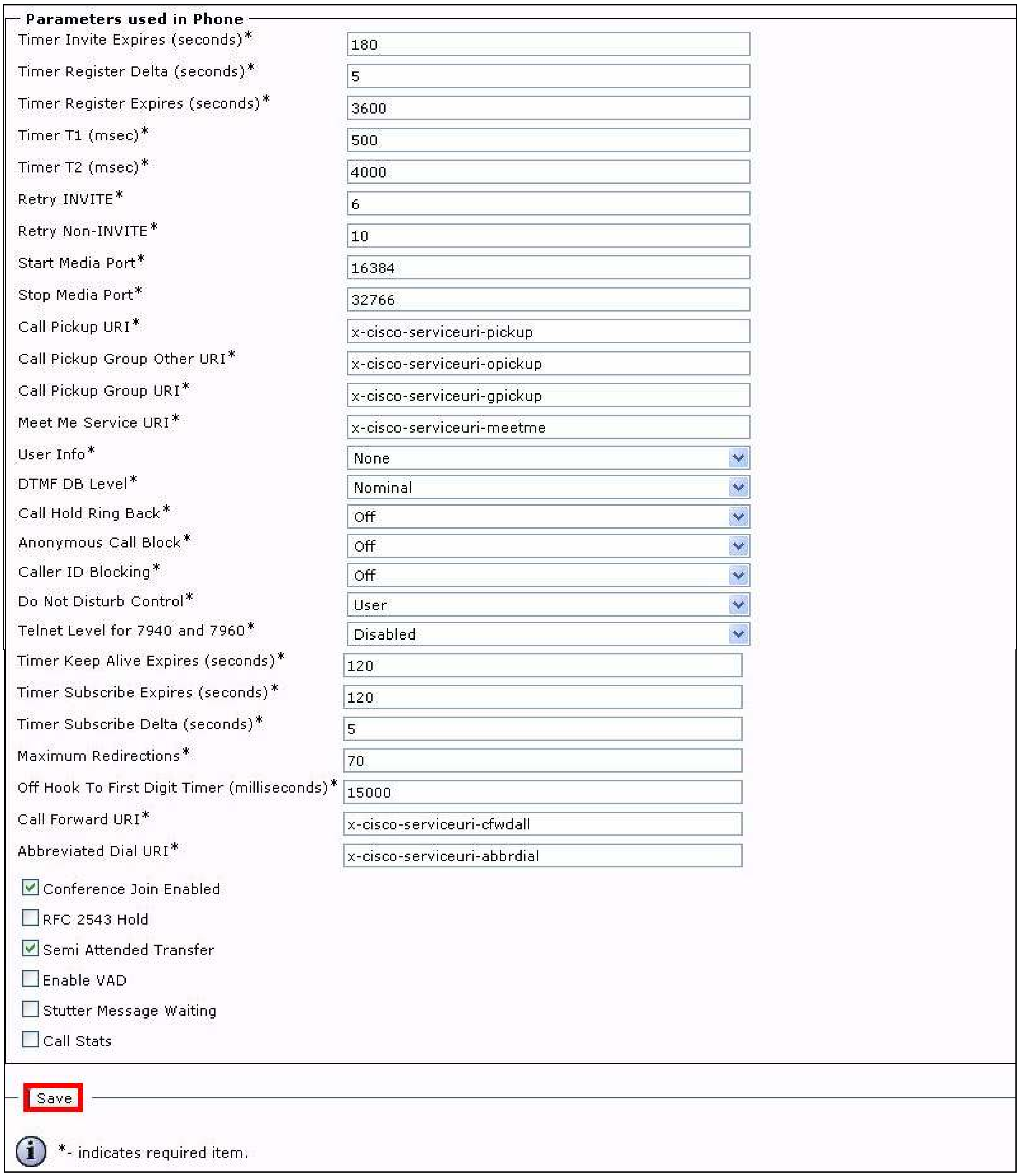
Note: *To enable SIP connectivity to Avaya Meeting Exchange utilizing UDP, set the **Outgoing Transport Type** field to UDP.*



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

Step	Description
4.1.2a	To enable SIP connectivity with Avaya Meeting Exchange, configure a **SIP Profile** as follows: • From the Cisco UCM main menu, select **Device → Device Settings → SIP Profile**. • [***Not Shown***] *Click* ***Add New*** *to create a new* ***SIP Profile***. • Provision settings under **SIP Profile Information** as displayed and scroll down. 

Step	Description
4.1.2b	Use default settings under **Parameters used in Phone** as displayed and click **Save**.



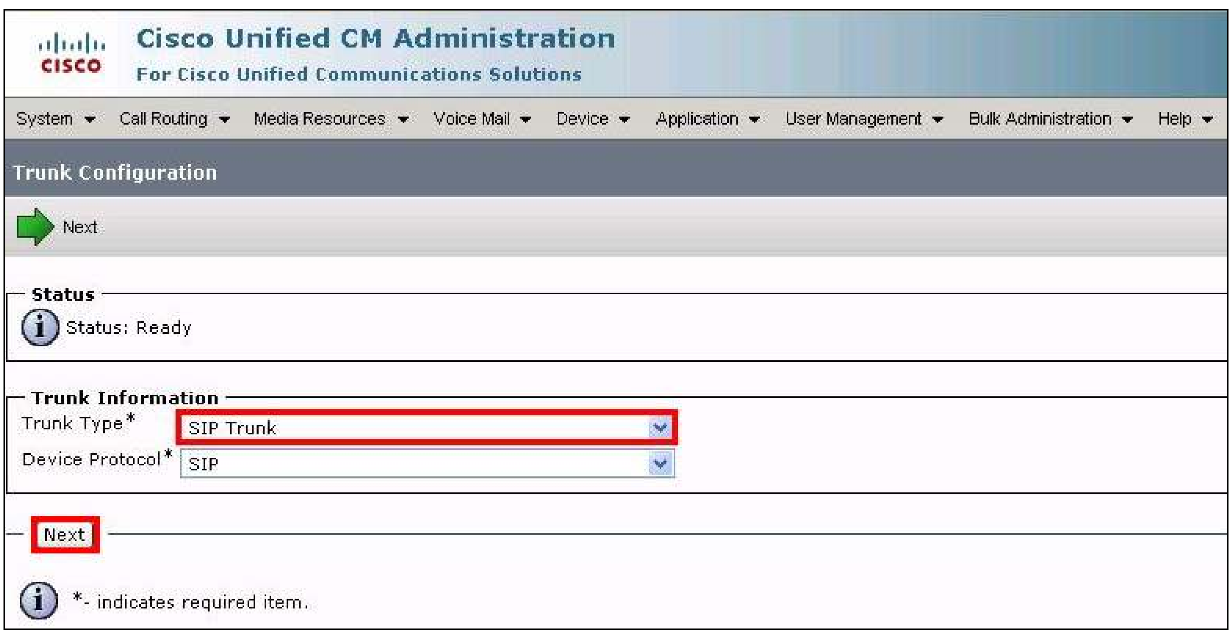
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.1.3</td><td>

To enable SIP connectivity with Avaya Meeting Exchange, configure a **SIP Trunk** as follows:

- From the Cisco UCM main menu, select **Device ➔ Trunk**.
- Click **Add New** to create a new **SIP Trunk**.

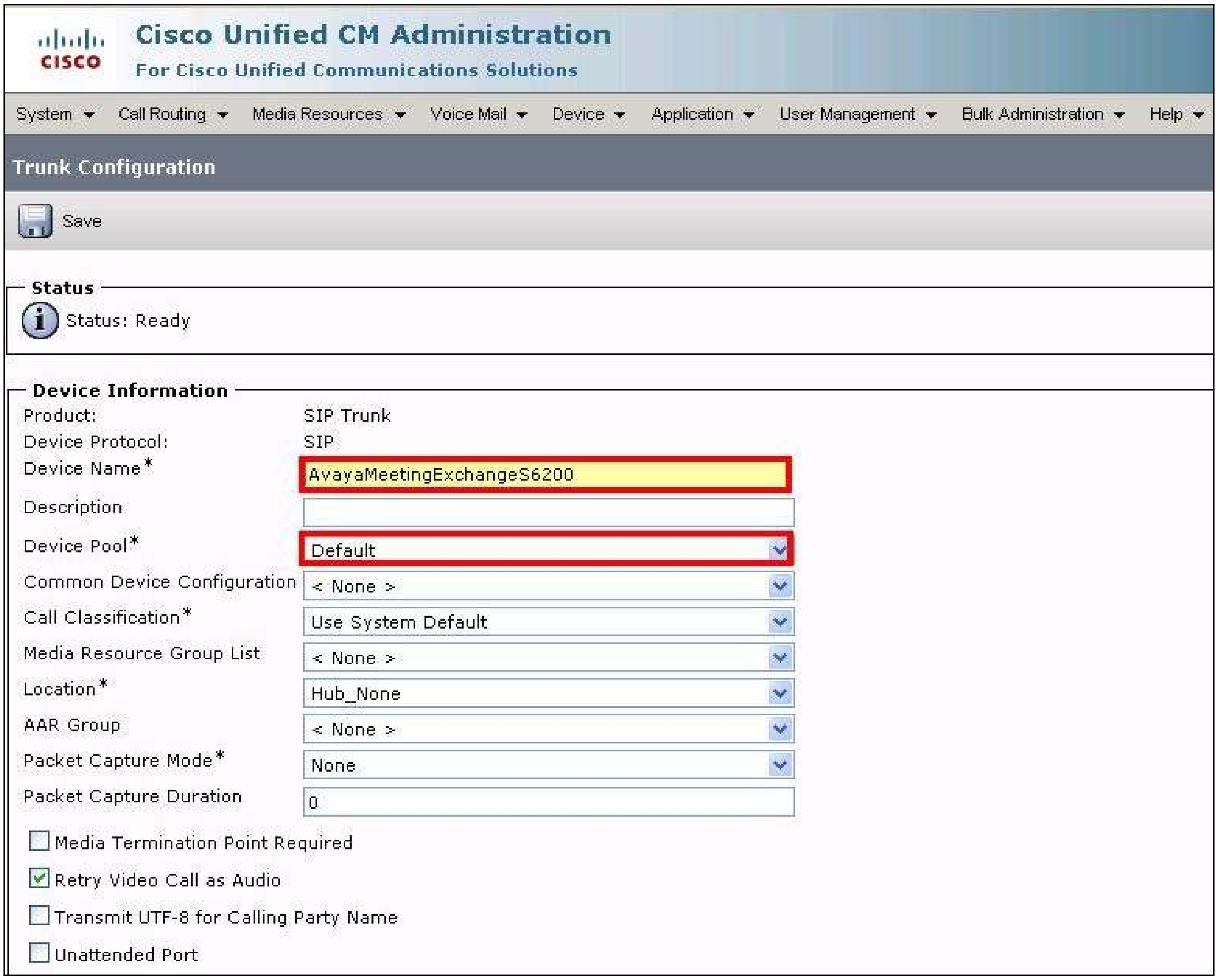
</td></tr>
<tr><td>4.1.4</td><td>

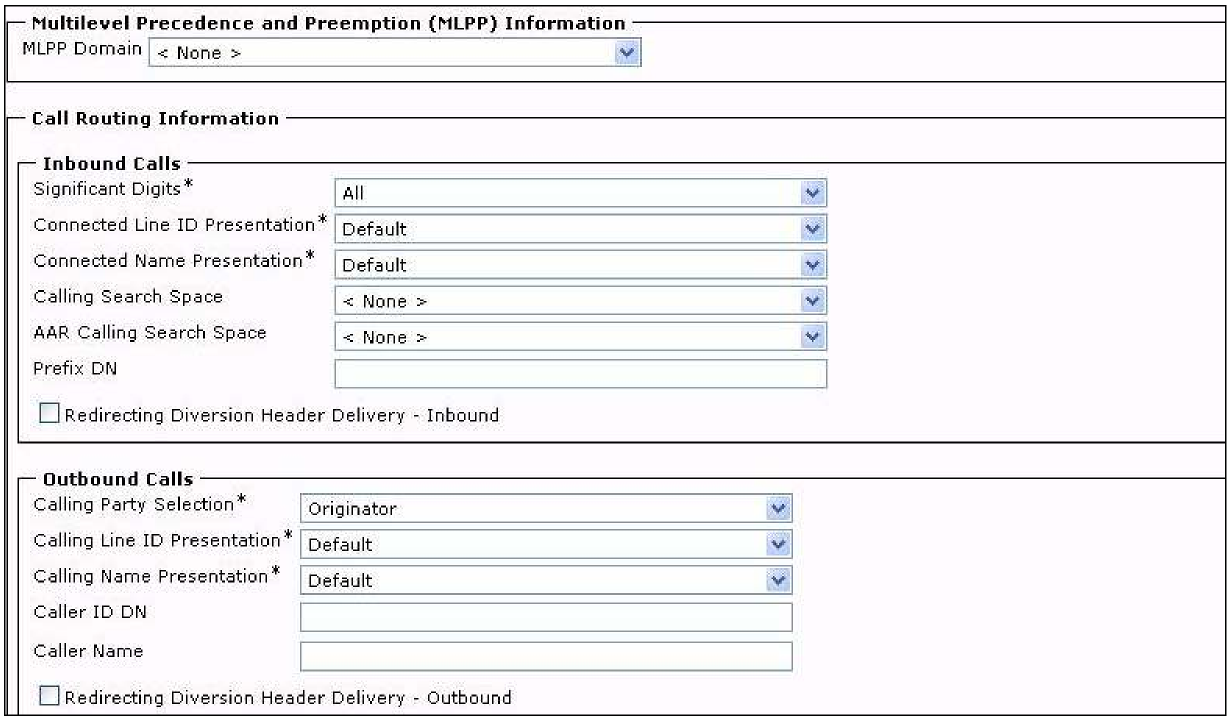
Select **SIP Trunk** from the drop-down list for the **Trunk Type** field. Accept the default setting for the **Device Protocol** field and click **Next**.



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

Step	Description
4.1.5a	Provision settings under **Device Information** as displayed and scroll down. The **Location** field specifies the total bandwidth that is available for calls between this location and the central location, or hub. Using the default setting **Hub_None** specifies unlimited available bandwidth.

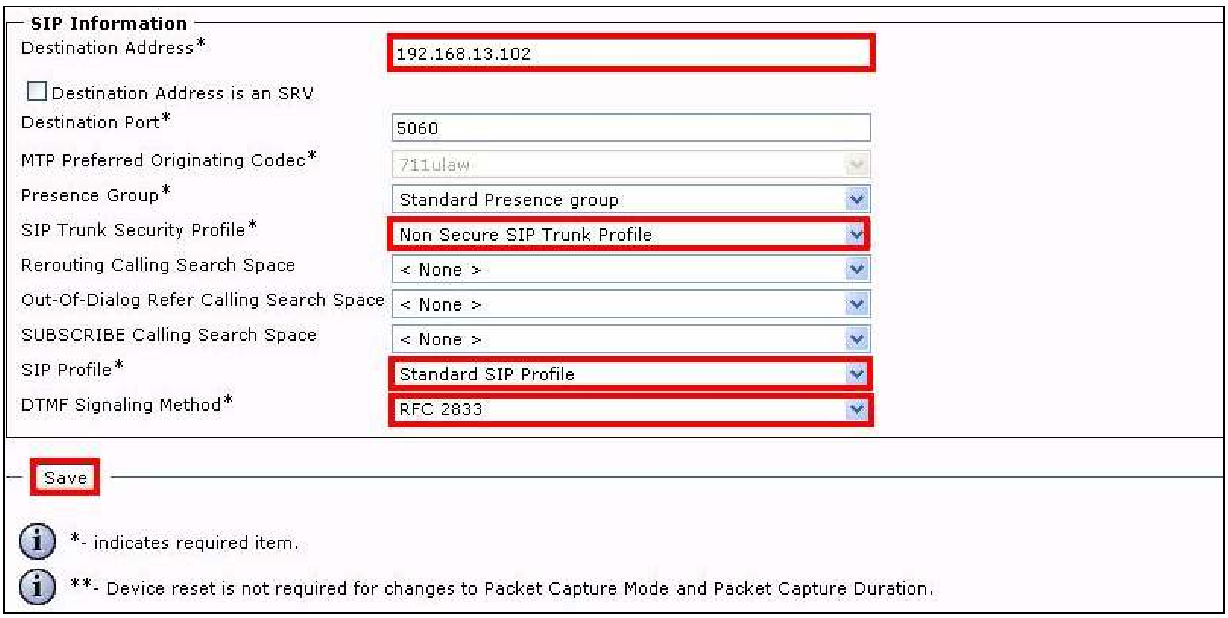


Step	Description
4.1.5b	Use default settings as displayed and scroll down. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.1.5c</td><td>

Provision settings under **SIP Information** as displayed.

- Enter the IP address of Avaya Meeting Exchange in the **Destination Address** field.
- Select the SIP Trunk Security Profile provisioned in **Step 4.1.1** from the drop-down list for the **SIP Trunk Security Profile** field.
- Select the SIP Profile provisioned in **Step 4.1.2** from the drop-down list for the **SIP Profile** field.
- Select **RFC 2833** from the drop-down list for the **DTMF Signaling Method** field.
- Click **Save**.



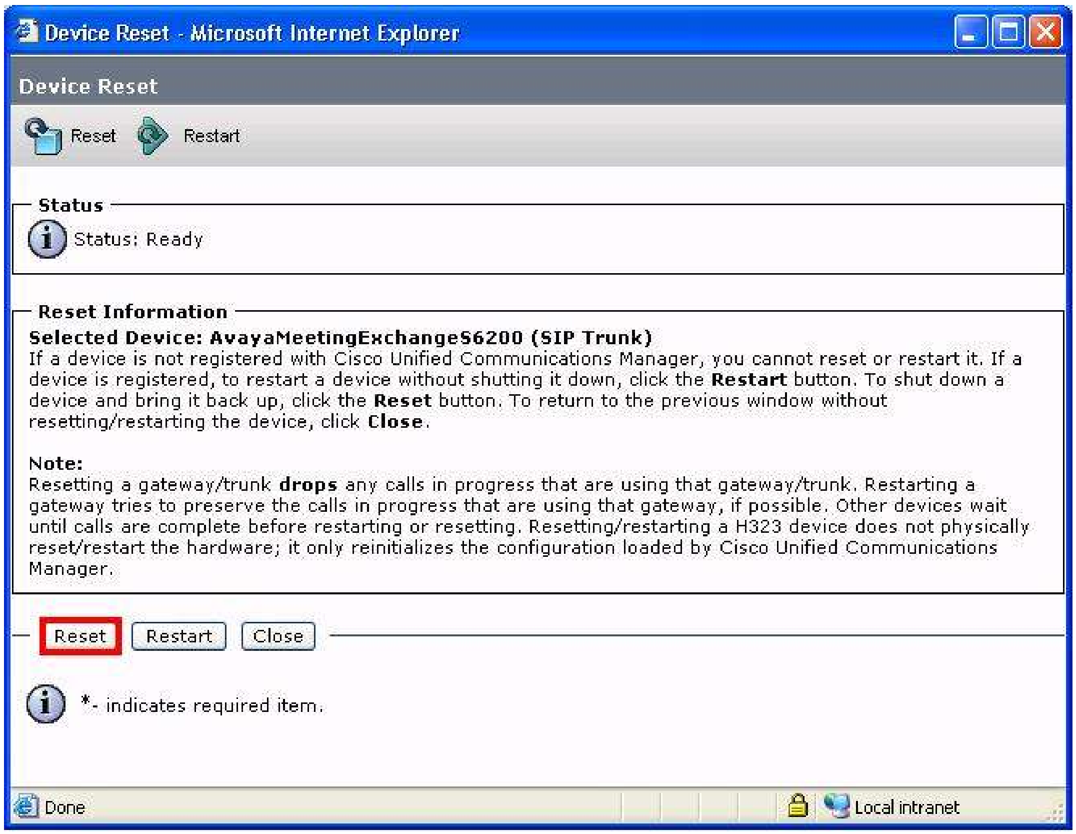
</td></tr>
<tr><td>4.1.6</td><td>

From the pop-up window, click **OK** and reset the trunk by clicking **Reset**, 

[***Not Shown****, located at the bottom of the* ***SIP Trunk*** *page*].

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

Step	Description
4.1.7	From the pop-up window, click **Reset**.



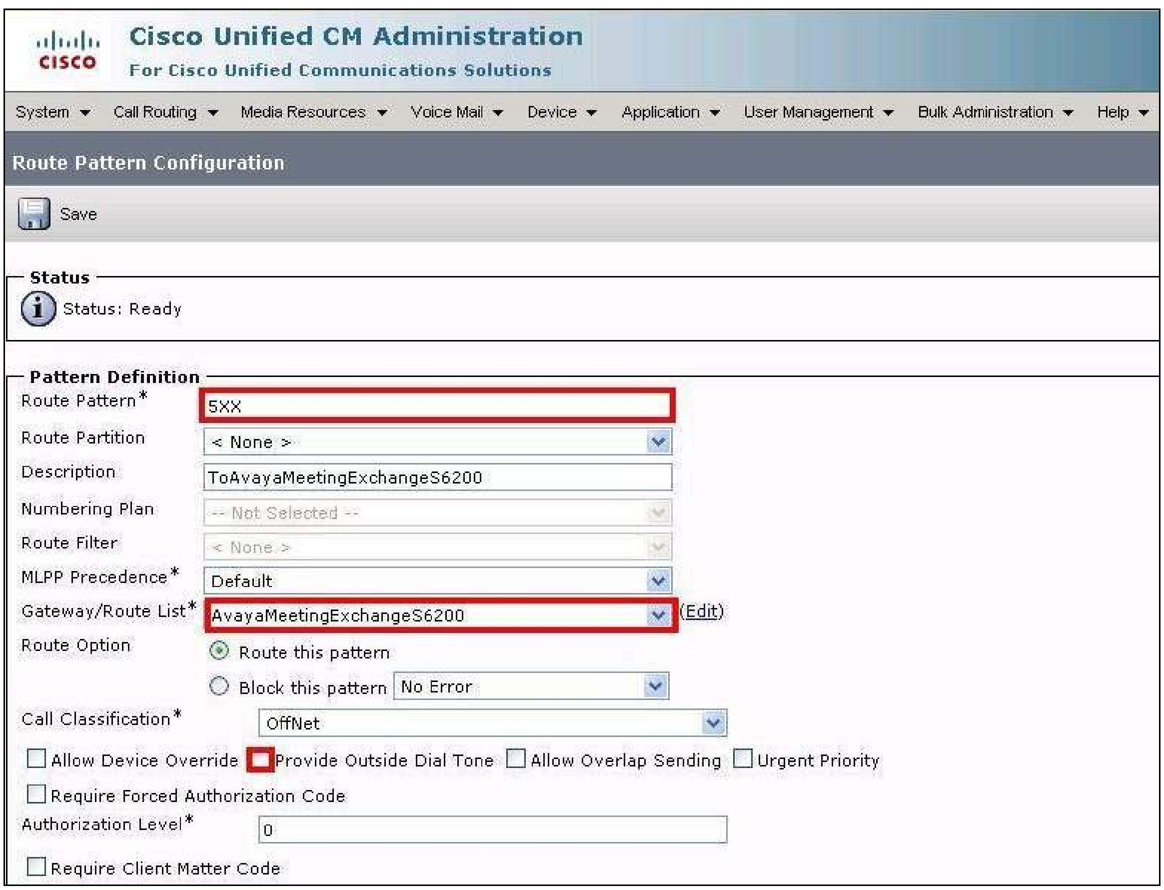
4.2. Configure Call Routing

This section describes the steps for configuring call routing from Cisco UCM to Avaya Meeting Exchange.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.2.1a</td><td>To enable routing from Cisco UCM to Avaya Meeting Exchange utilizing the SIP trunk provisioned in Section 4.1, configure a Route Pattern as follows:

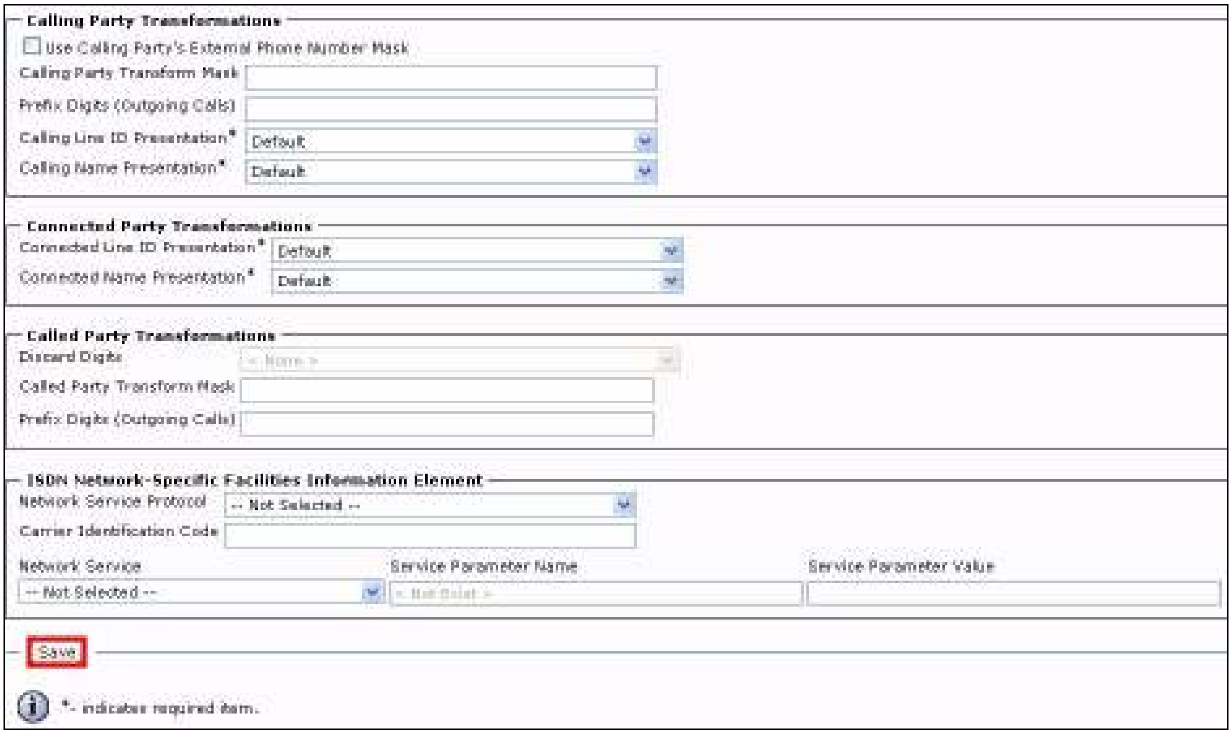
From the Cisco UCM main menu, select Call Routing → Route/Hunt → Route Pattern.
[Not Shown] Click Add New to create a new Route Pattern.
Provision settings under Pattern Definition as displayed and scroll down.

Enter a pattern in the Route Pattern field that corresponds to the call branding for direct and scan call flows provisioned on Avaya Meeting exchange in Section 3.3. Note that “X” is a wildcard and represents any digit 0 through 9.
Select the SIP trunk group provisioned in Section 4.1 from the drop-down list for the Gateway/Route List field.
Verify that the Provide Outside Dial Tone field is not selected.

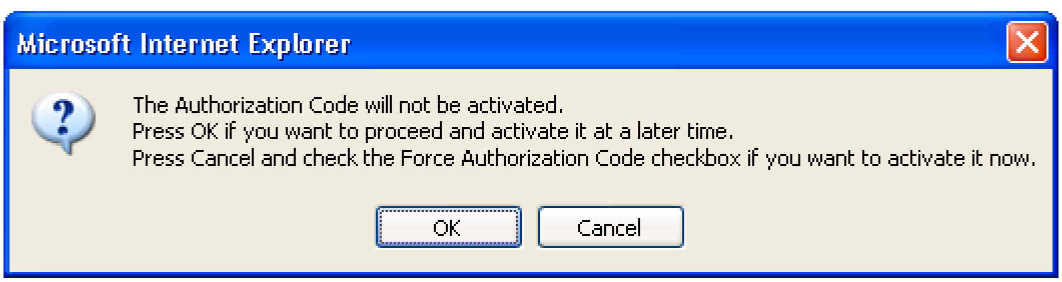


</td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.2.1b</td><td>Use default settings as displayed and click Save.



</td></tr>
<tr><td>4.2.2</td><td>The Require Forced Authorization Code option was not enabled in Step 4.2.1, click OK.



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

5. Configure the AudioCodes Mediant 2000

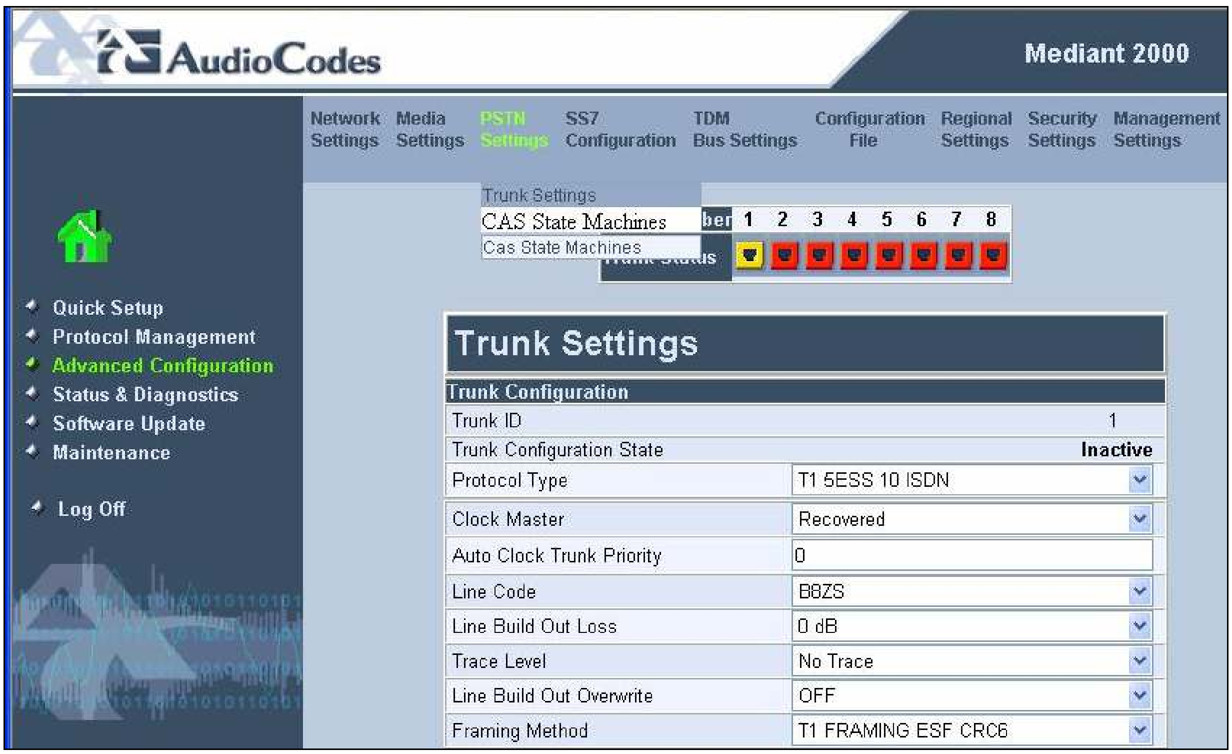
This section describes the steps for configuring connectivity between Avaya Meeting Exchange and the PSTN via the AudioCodes Mediant 2000. The AudioCodes Mediant 2000 is administered using either the AudioCodes Element Management System (EMS) or an embedded web server. For this sample configuration, the provisioning is administered via the embedded web server by entering **http://<AudioCodes Mediant 2000 IP Address or Fully Qualified Domain Name (FQDN)>** into a web browser's Uniform Resource Locator (URL) bar. Refer to [**5**] and [**6**] for additional information regarding the administration of the AudioCodes Mediant 2000.

5.1. Configure T1 Connectivity

This section describes the steps for configuring T1 connectivity between the AudioCodes Mediant 2000 and the PSTN.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>5.1.1a</td><td>To enable connectivity to the PSTN, administer Trunk Settings as follows:

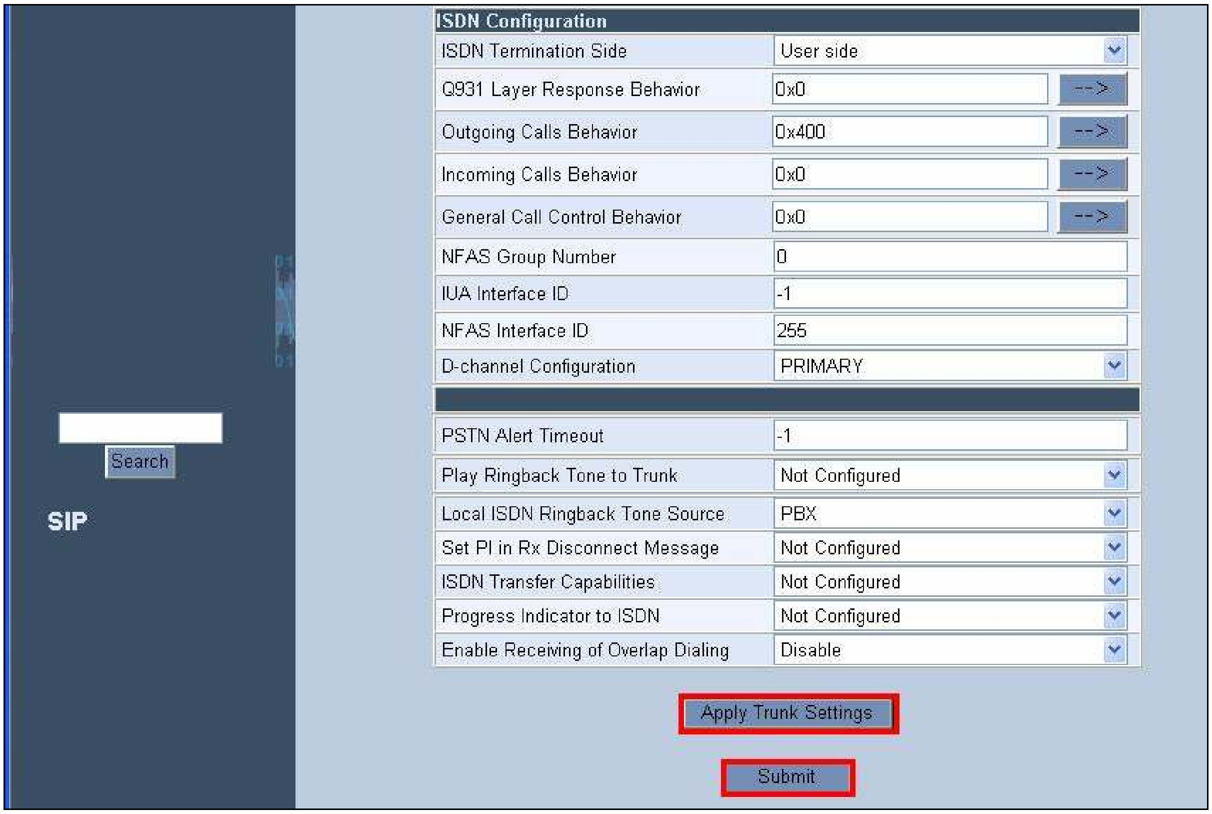
From the embedded web server, select Advanced Configuration ➔ PSTN Settings ➔ Trunk Settings.
Select the appropriate trunk to provision. For this sample configuration, T1 connectivity to the PSTN uses trunk 1.
[Not Shown] Click the Stop Trunk button to modify the selected trunk’s parameters.
The status of the Trunk Configuration State parameter changes to Inactive.
Configure Trunk Settings for this interface to enable T1 ISDN-PRI connectivity to the PSTN according to requirements defined by the PSTN service provider.
Scroll down.



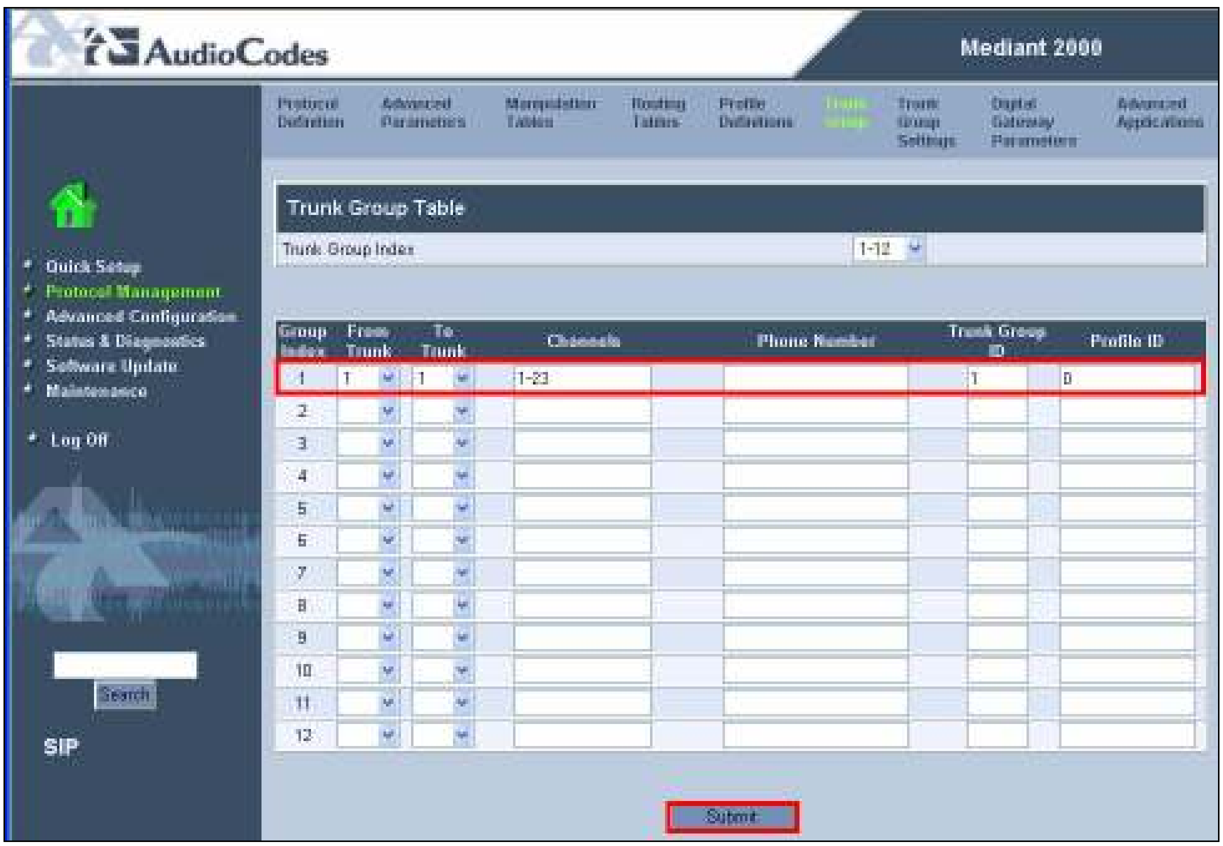
</td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>5.1.1b</td><td>Provision settings as required.

Click Submit.
Click Apply Trunk Settings.



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

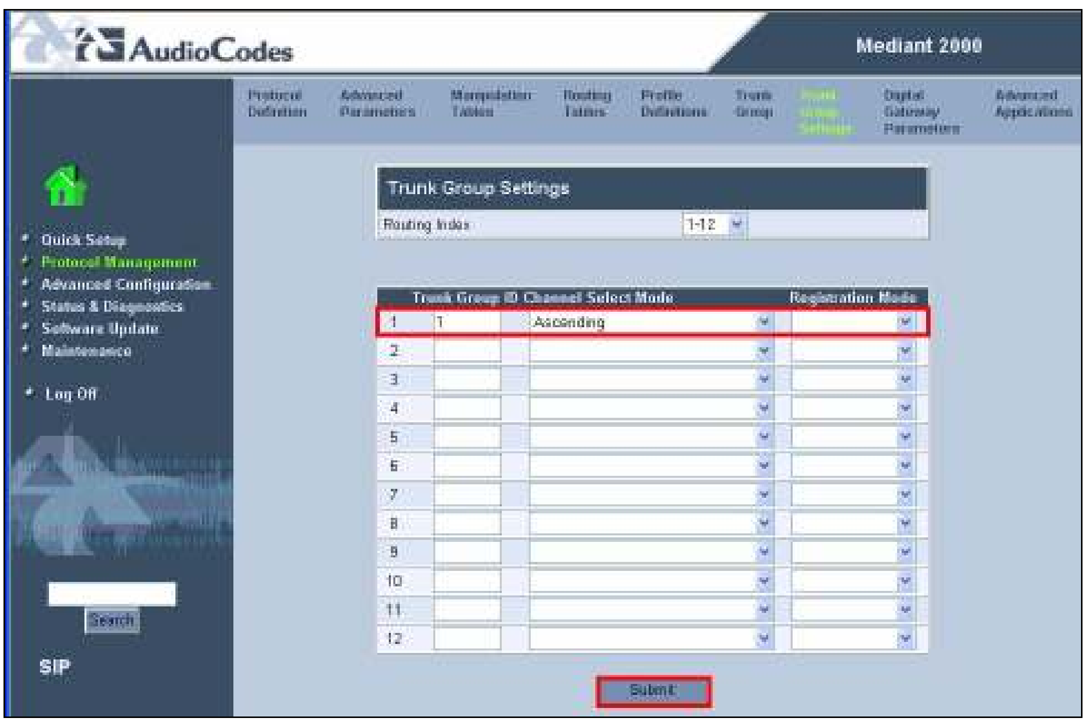
Step	Description
5.1.2	To assign common rules for routing IP-to-PSTN calls, administer a **Trunk Group** as follows: • From the embedded web server, select, **Protocol Management → Trunk Group**. • Select the appropriate trunk from the **Group Index** column. For this sample configuration, T1 connectivity to the PSTN uses trunk 1. • Set the **From Trunk** and **To Trunk** fields to the starting and ending number of the trunk to the PSTN. • Set the **Channels** field to correspond to number of B-channels available on this trunk. For this sample configuration, T1 ISDN-PRI trunking is used, thus 23 B-channels. • Set the **Trunk Group ID** field to an available number. • Click **Submit**. ***Note***: *Trunk groups are logical entities that are used for routing IP to telephone calls with common rules in which calls are assigned to B-channels within each trunk group*. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>5.1.3</td><td>
To determine the method in which new calls are assigned to B-channels within a trunk group, administer Trunk Group Settings as follows:

From the embedded web server, select Protocol Management ➔ Trunk Group Settings.
Add an entry that corresponds to trunk group provisioned in Step 5.1.2 as displayed.

Set the Trunk Group ID field to the Trunk Group ID assigned to the trunk group provisioned in Step 5.1.2.
Set the Channel Select Mode field to determine the method in which call origination from Avaya Meeting Exchange are assigned to B-channels within a trunk group. For this sample configuration, this trunk group is administered to select B-channels in Ascending mode, while call origination from the PSTN selects B-channels in a descending fashion.
Note: To reduce the probability of glare, which occurs when both sides of a trunk select the same B-channel for call origination, the network should be administered so both sides of the trunk select B-channels from opposite ends of the trunk. This is called linear hunting, ascending or descending. For example, on a 23B+D T1 ISDN-PRI trunk, the user side should be administered to select B-channels starting at channel 1 (ascending) if the network side is administered to select B-channels starting at channel 23 (descending).

Click Submit.



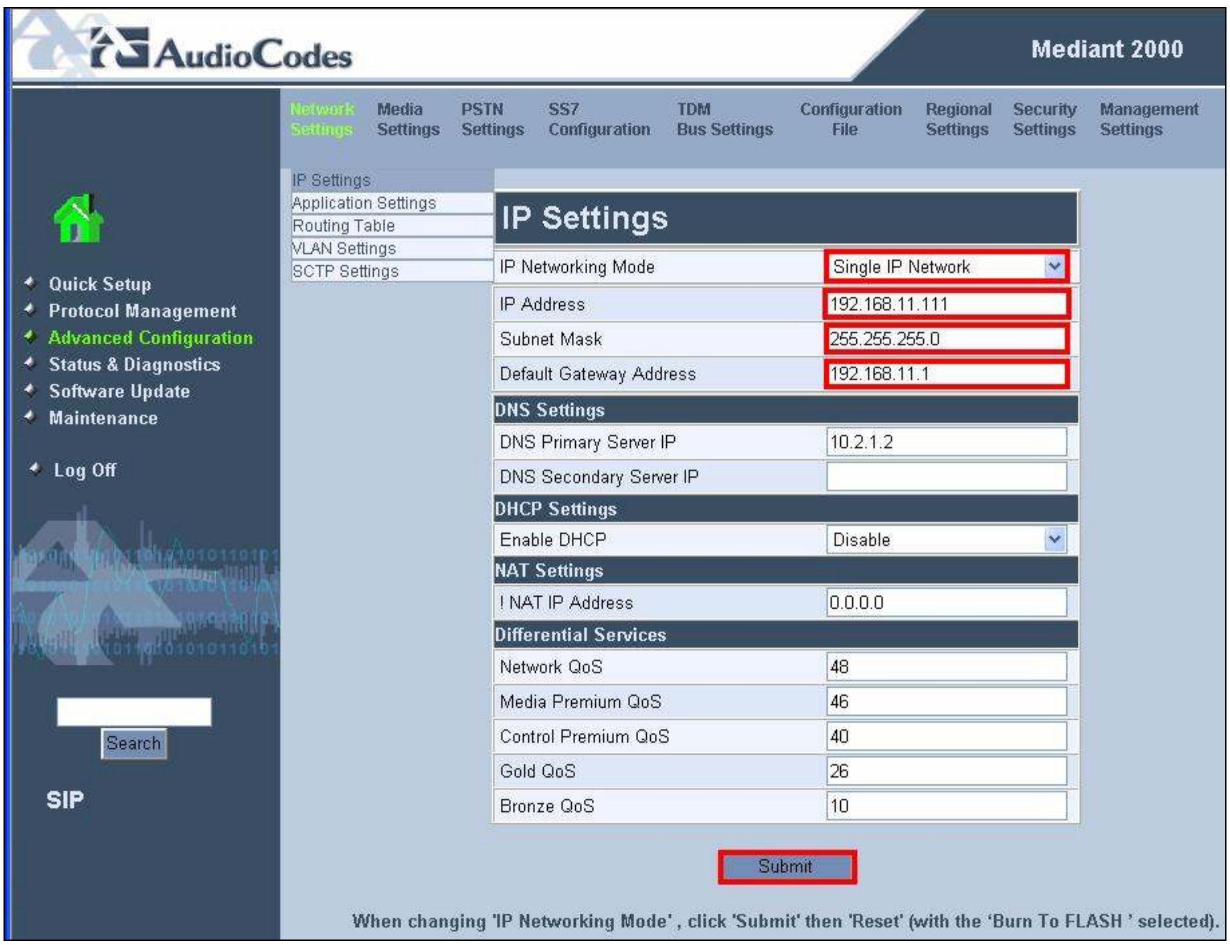
</td></tr>
</table>

5.2. Configure SIP Connectivity

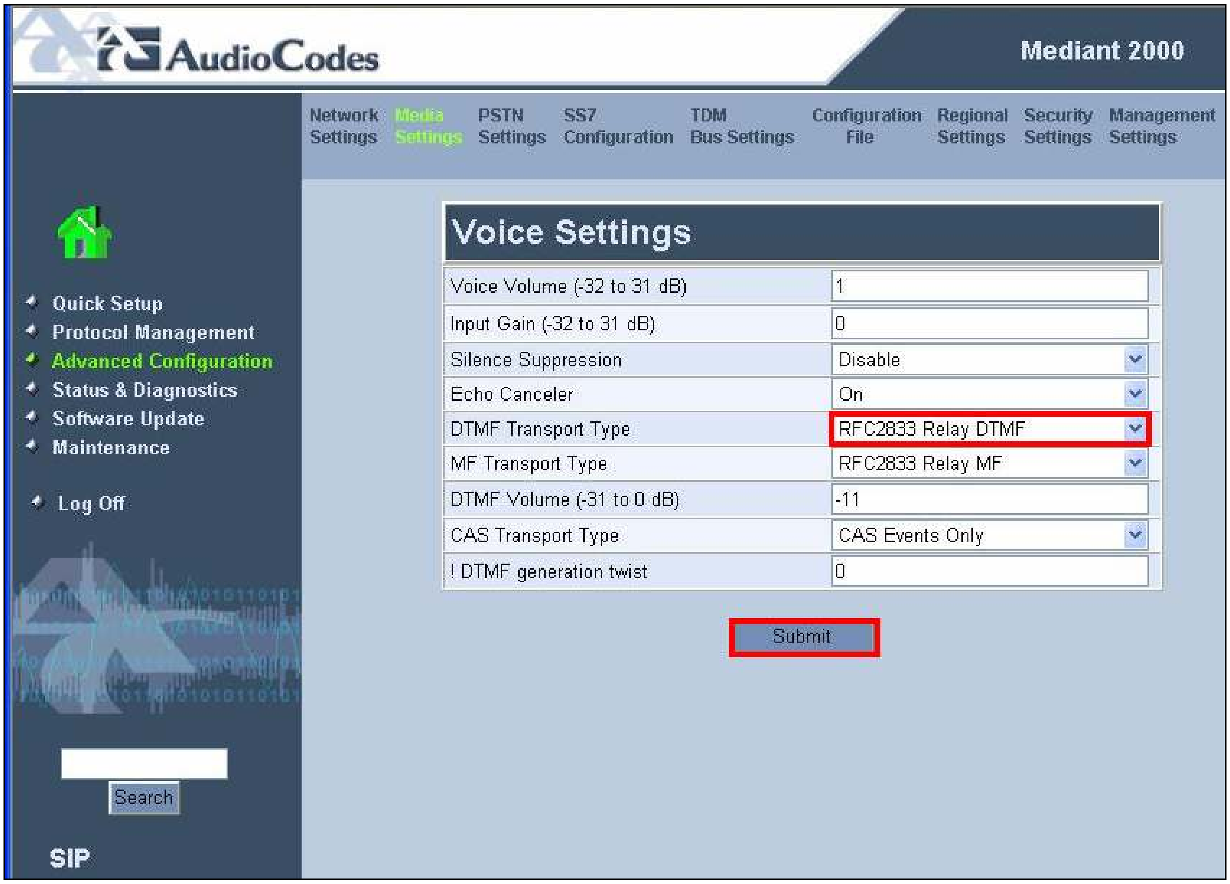
This section describes the steps for configuring SIP connectivity between the Avaya Meeting Exchange the AudioCodes Mediant 2000.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>5.2.1</td><td>
To enable IP connectivity to the AudioCodes Mediant 2000, administer IP Settings as follows:

From the embedded web server, select Advanced Configuration ➔ Network Settings ➔ IP Settings.
Select the appropriate IP network configuration from the drop-down list for the IP Networking Mode field. For this sample configuration, Media, Control and OAM are on the same network.
Set the IP Address, Subnet Mask and Default Gateway Address fields accordingly.
Use default settings for remaining fields.
Click Submit.



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

Step	Description
5.2.2	Administer **Media Settings** as displayed. • Click on **Advanced Configuration → Media Settings → Voice Settings**. • Select **RFC2833 Relay DTMF** from the drop-down list for the **DTMF Transport Type** field. • Click **Submit**. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>5.2.3a</td><td>

To enable SIP connectivity with the Avaya Meeting Exchange, administer **General Parameters** as follows:

- From the embedded web server, select **Protocol Management ➔ Protocol Definition ➔ General Parameters**.
- Set the **SIP Transport Type**, **SIP TCP Local Port** and **SIP Destination Port** fields to enable SIP connectivity with the Avaya Meeting Exchange utilizing TCP.

  ***Note**: To enable SIP connectivity with Avaya Meeting Exchange utilizing UDP, set the **SIP Transport Type** and **SIP UDP Local Port** fields for UDP.*
- Use default settings for remaining fields.
- Scroll down.

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

Step	Description
5.2.3b	Use default settings and click **Submit**. 

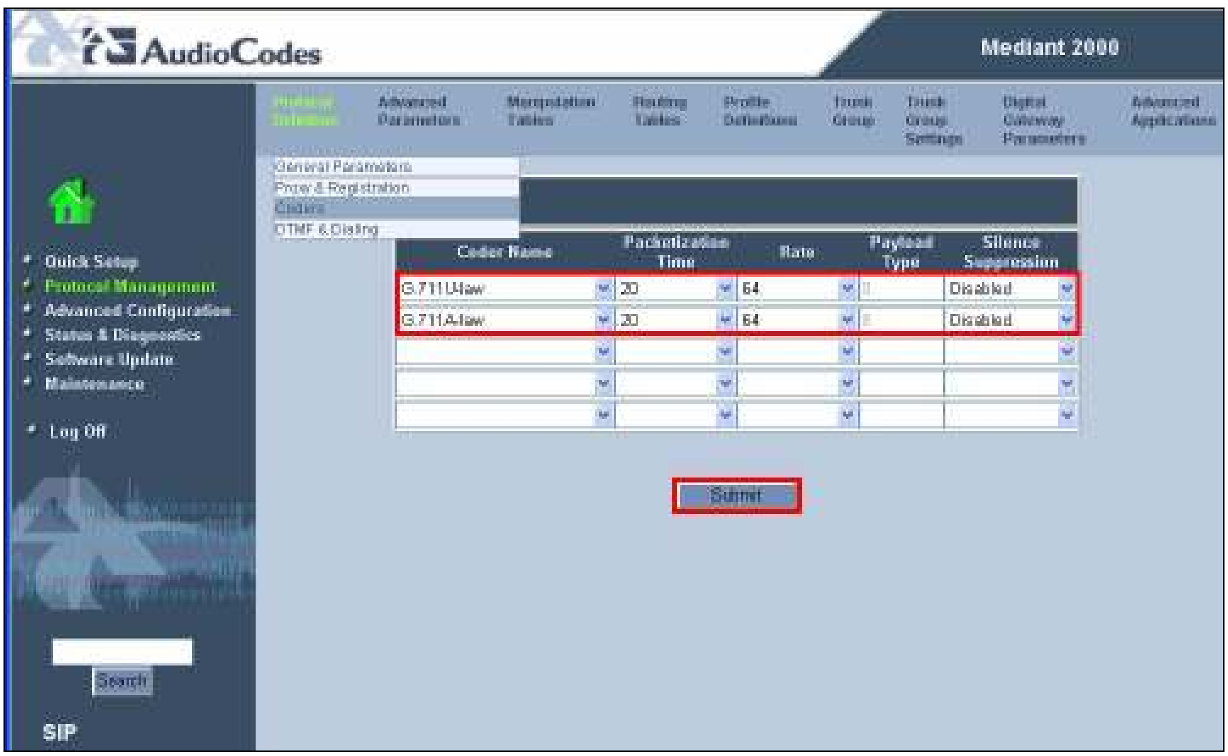
Step	Description
5.2.4	Administer settings for **Proxy & Registration** as follows: • From the embedded web server, select **Protocol Management ➔ Protocol Definition ➔ Proxy & Registration**. • Select **Don't Use Proxy** from the drop-down list for the **Enable Proxy** field. *Note: SIP connectivity between the AudioCodes Mediant 2000 and Avaya Meeting Exchange is direct.* • Use default settings for remaining fields. • Click **Submit**. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>5.2.5</td><td>

Administer codec preferences and attributes for the SIP trunk between the AudioCodes Mediant 2000 and Avaya Meeting Exchange as follows:

- From the embedded web server, select **Protocol Management ➔ Protocol Definition ➔ Coders**.
- Add entries for codecs that are supported on the Convedia CMS-6000 Media Server (see **Step 3.16**) as displayed.
  - Select a codec from the drop-down list for the **Coder Name** field that is compatible with the codecs supported on Avaya Meeting Exchange.
  - Use default settings for remaining fields.
- Click **Submit**.

***Note**: The first coder is the highest priority coder and is used by the AudioCodes Mediant 2000 whenever possible. If the far end SIP User Agent cannot use the coder assigned as the first coder, the gateway attempts to use the next coder and so forth.*



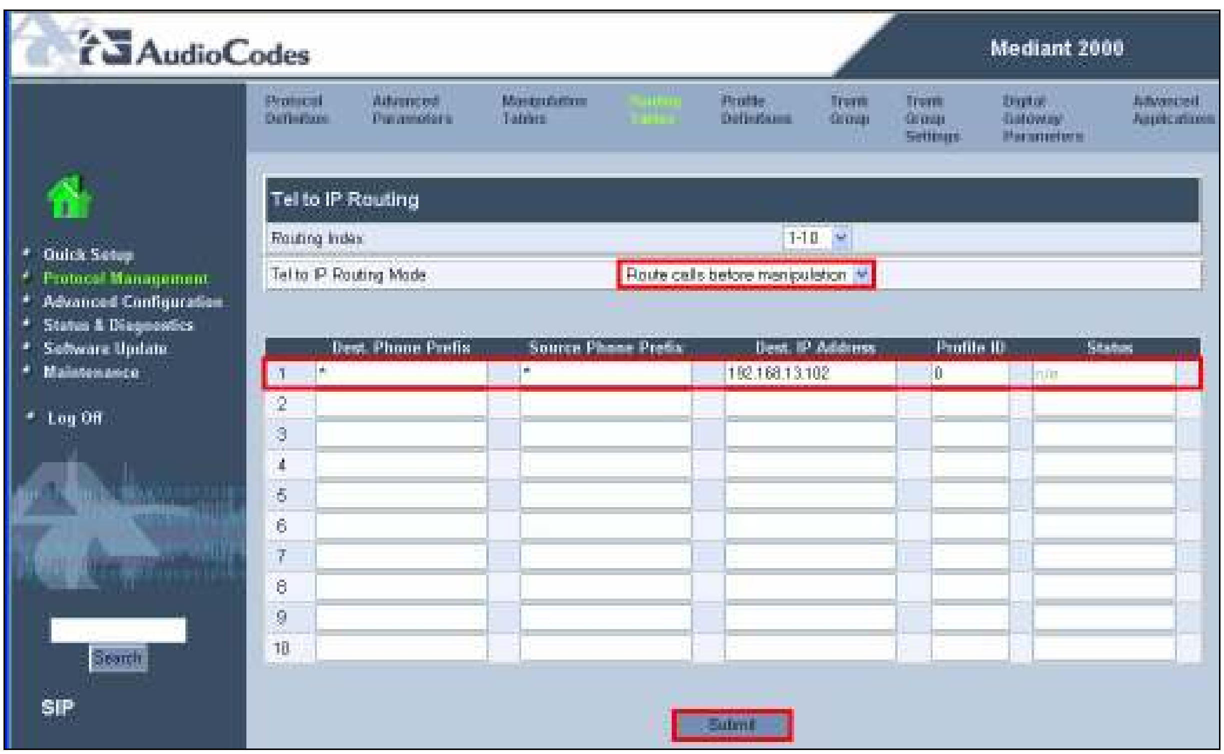
</td></tr>
</table>

5.3. Configure Call Routing

This section describes the steps for configuring call routing between Avaya Meeting Exchange and the PSTN via the AudioCodes Mediant 2000.

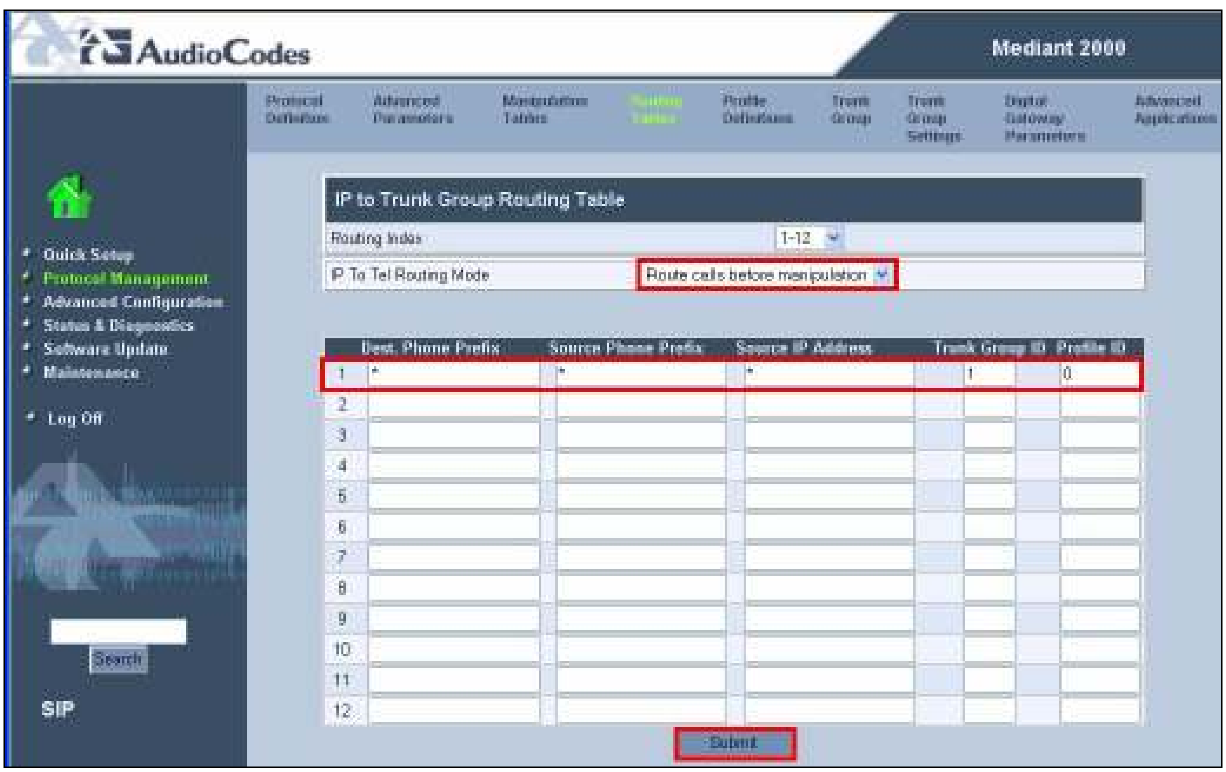
Step	Description

5.3.1	Administer call routing rules that are applied to calls originating from the PSTN to Avaya Meeting Exchange by adding **Tel to IP Group Routing** rules as follows:
	• From the embedded web server, select **Protocol Management ➔ Routing Tables ➔ Tel to IP Group Routing**.
	• Select **Route calls before manipulation** from the drop-down list for the **Tel to IP Routing Mode** field. Add an entry to enable call origination from the PSTN to Avaya Meeting Exchange as displayed.
	    o Enter a rule in the **Dest. Phone Prefix** field that matches the pattern of the called party number assigned to Avaya Meeting Exchange. For this sample configuration, the rule "*" is utilized, where "*" is a wildcard and matches any digit string, thus routing all calls to Avaya Meeting Exchange.
	    o Enter a rule in the **Source Phone Prefix** field to match the calling party number for calls from the PSTN.
	    o Enter the IP address of Avaya Meeting Exchange in the **Dest. IP Address** field.
	• Click **Submit**.



Step	Description

5.3.2	Administer call routing rules that are applied to calls originating from Avaya Meeting Exchange to the PSTN by adding **IP to Hunt Group Routing** rules as follows:
	• From the embedded web server, select **Protocol Management ➔ Routing Tables ➔ IP to Hunt Group Routing**.
	• Select **Route calls before manipulation** from the drop-down list for the **IP To Tel Routing Mode** field. Add an entry to enable call origination from Avaya Meeting Exchange to the PSTN as displayed. Add an entry to enable call origination from Avaya Meeting Exchange to the PSTN as displayed.
	  o Enter a rule in the **Dest. Phone Prefix** field that matches the pattern of the called party number assigned to the PSTN. For this sample configuration, the rule "*" is utilized, where "*" is a wildcard and matches any digit string, thus routing all calls to the PSTN.
	  o Enter a rule in the **Source Phone Prefix** field to match the calling party number for calls from the Avaya Meeting Exchange.
	  o Enter a rule in the **Source IP Address** field to match the IP address of Avaya Meeting Exchange.
	  o Enter the Trunk Group ID for the T1 ISDN-PRI trunk group provisioned in **Section 5.1** in the **Trunk Group ID** field.
	• Click **Submit**.



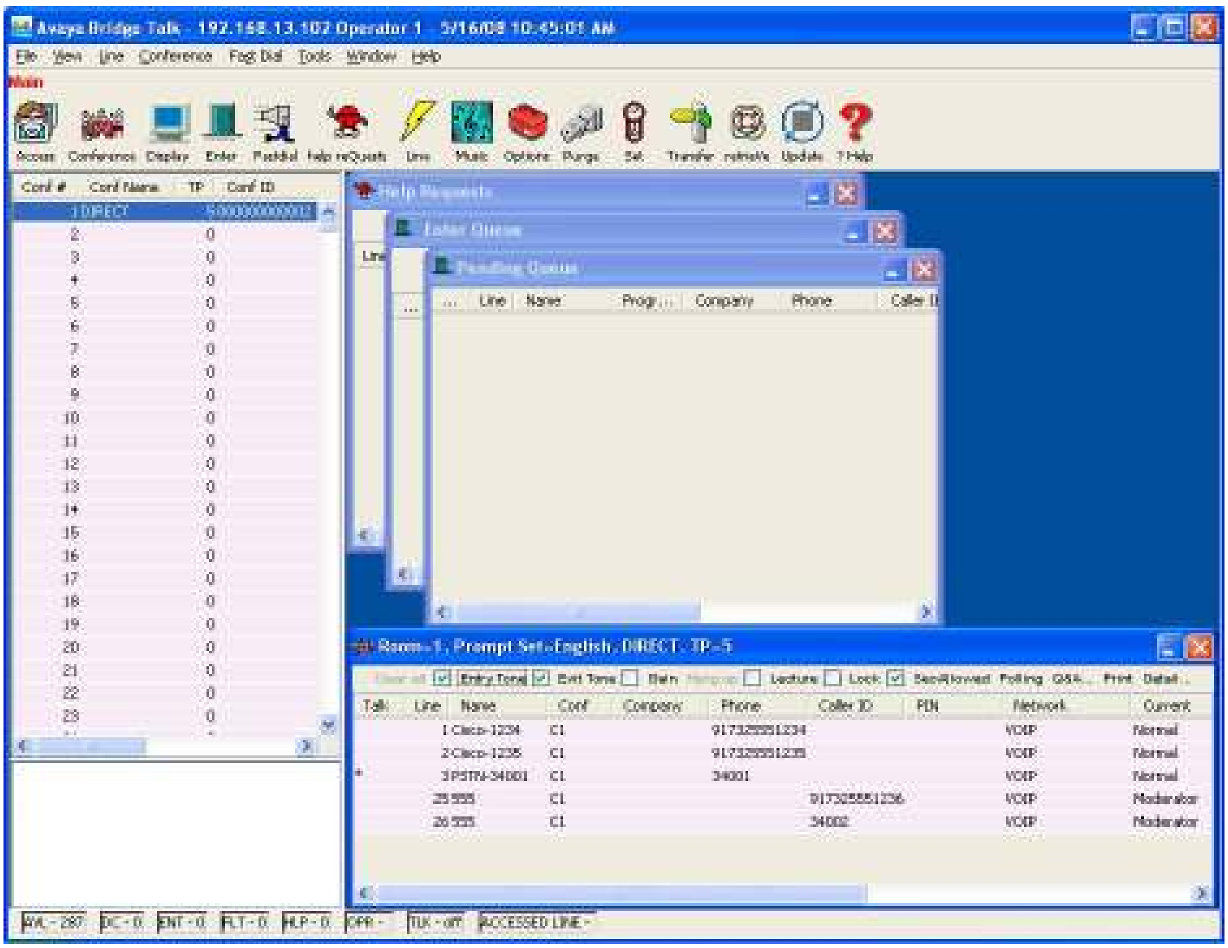
6. Verification Steps

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

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>6.1</td><td>Verify conferencing related processes are running on Avaya Meeting Exchange. From the Avaya Meeting Exchange CLI, enter “service mx-bridge status” at the command prompt and verify that a Process ID (PID) is present for all processes.
<pre>S6200-> service mx-bridge status
 2373 pts/1   00:00:00 initdcb
 2420 pts/1   00:00:00 log
 2423 pts/1   00:00:00 bridgeTranslato
 2424 pts/1   00:00:00 netservices
 2431 pts/1   00:00:00 timer
 2432 pts/1   00:00:00 traffic
 2433 pts/1   00:00:00 chdbased
 2434 pts/1   00:00:00 startd
 2435 pts/1   00:00:00 cdr
 2436 pts/1   00:00:00 modapid
 2437 pts/1   00:00:00 schapid
 2438 pts/1   00:00:00 callhand
 2439 pts/1   00:00:00 initipcb
 2443 pts/1   00:00:00 sipagent
 2451 pts/1   00:00:00 msdispatcher
 2454 pts/1   00:00:00 softms
 2457 pts/1   00:00:00 serverComms
 2311 pts/1   00:00:00 sqlexecd with 5 children</pre></td></tr>
<tr><td>6.2</td><td>Validate signaling and media connectivity for call origination from Cisco UCM to Avaya Meeting Exchange. This is accomplished by verifying that the trunk group provisioned in Section 4.1 is utilized when a call from a telephone registered to Cisco UCM dials in to a conference provisioned on Avaya Meeting Exchange. From a telephone registered to Cisco UCM, dial 555 to enter the conference provisioned in Section 3.4 as moderator via the call branding for a direct call flow provisioned in Step 3.3.1.</td></tr>
<tr><td>6.3</td><td>Validate signaling and media connectivity for call origination from the PSTN to Avaya Meeting Exchange via the AudioCodes Mediant 2000. This is accomplished by verifying that the trunk groups provisioned in Section 5.1 and Section 5.2 are utilized when a call from a telephone associated with the PSTN dials in to a conference provisioned on Avaya Meeting Exchange. From a telephone associated with the PSTN, dial 555 to enter the conference provisioned in Section 3.4 as moderator via the call branding for a direct call flow provisioned in Step 3.3.1.</td></tr>
</table>

Step	Description
6.4	Validate signaling and media connectivity for call origination from Avaya Meeting Exchange to both Cisco UCM and the PSTN via the AudioCodes Mediant 2000. This is accomplished by verifying that the trunk groups provisioned in **Section 4.1**, **Section 5.1** and **Section 5.2** are utilized when a call from a participant in a conference on Avaya Meeting Exchange is placed to telephones registered to Cisco UCM and associated with the PSTN respectively. From a telephone already in conference (see **Step 6.2**, or **Step 6.3**), enter the appropriate touchtone command to initiate the blast dial feature as provisioned in **Section 3.4**. Note that the goal of this step is to validate call origination from Avaya Meeting Exchange to Cisco UCM, thus any form of call origination from Avaya Meeting Exchange may be utilized, e.g., originator dial-out.

Step	Description
6.5	Verify that calls to and from Avaya Meeting Exchange are managed correctly, e.g., participants are added/removed from conferences. This is accomplished by utilizing the Avaya Bridge Talk application. • If not already logged on, log in to the Avaya Bridge Talk application with the appropriate credentials. • From the Conference Navigator, double-click the appropriate entry to open the corresponding Conference Room. • Verify conference participants are added/removed from conferences by observing the Conference Navigator and/or Conference Room window. ***Note**: The screen capture below displays the conference that was initiated in **Step 6.2**, **Step 6.3** and **Step 6.4**.*



<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>6.6</td><td>

Verify **Channel Status** on the AudioCodes Mediant 2000 as follows:

- From the embedded web server, select **Status & Diagnostics → Channel Status**.
  *Note: The **Trunk & Channel Status** displays **Active - OK** for T1 ISDN-PRI trunk group provisioned in **Section 5.1**.*
- This screen capture displays the channel selection pattern for the **Active** channels that are associated with the conference that was initiated in **Step 6.3** and **Step 6.4**.
  *Note: The PSTN selects channels in a descending pattern over the ISDN-PRI trunk between the PSTN and the AudioCodes Mediant 2000. This display shows **Channel 23** on **Trunk 1** is selected by the PSTN for call origination to Avaya Meeting Exchange. Conversely, **Channel 1** on **Trunk 1** is selected by the AudioCodes Mediant 2000 for call origination to the PSTN.*

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

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>6.7</td><td>Below is a SIP call flow of the scenario that was initiated in Step 6.2, Step 6.3 and Step 6.4. This trace is intended to display the provisioning presented in these Application Notes.

Cisco UCM (60.1.1.9) sends a SIP INVITE message to Avaya Meeting Exchange (192.168.13.102). Avaya Meeting Exchange extracts the DID (555) using the provisioning in Step 3.2.1 and places the call in conference using the call branding provisioned in Section 3.3.
The AudioCodes Mediant 2000 (192.168.11.111) sends a SIP INVITE message to Avaya Meeting Exchange (192.168.13.102). Avaya Meeting Exchange extracts the DID (555) using the provisioning in Step 3.2.1 and places the call in conference using the call branding provisioned in Section 3.3.
Avaya Meeting Exchange sends SIP INVITE messages to both Cisco UCM and the AudioCodes Mediant 2000 using the provisioning in Step 3.2.2.

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

7. Conclusion

These Application Notes present a sample configuration comprised of the Avaya Meeting Exchange S6200 Conferencing Server (Meeting Exchange), Cisco Unified Communications Manager (UCM) and the AudioCodes Mediant 2000. Employing this configuration enables call origination/termination between Avaya Meeting Exchange and endpoints registered to Cisco UCM, as well as endpoints on the PSTN via the AudioCodes Mediant 2000.

8. Additional References

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

[1] Meeting Exchange 5.0 S6200/6800 Administration and Maintenance Guide, Issue 2, Doc ID 04-602167, August 2007.

[2] Meeting Exchange 5.0 Service Pack 1 S6200/6800 Configuration Guide, Issue 4, Doc ID 04-602171, December 2007.

[3] Meeting Exchange 5.0 Bridge Talk User's Guide, Doc ID 04-602163, Issue 1, August 2007.

Cisco references are available at http://www.cisco.com.

[4] Cisco Unified Communications Manager Administration Guide Release 6.0(1), Document #: OL-12525-01.

AudioCodes references are available at http://www.audiocodes.com.

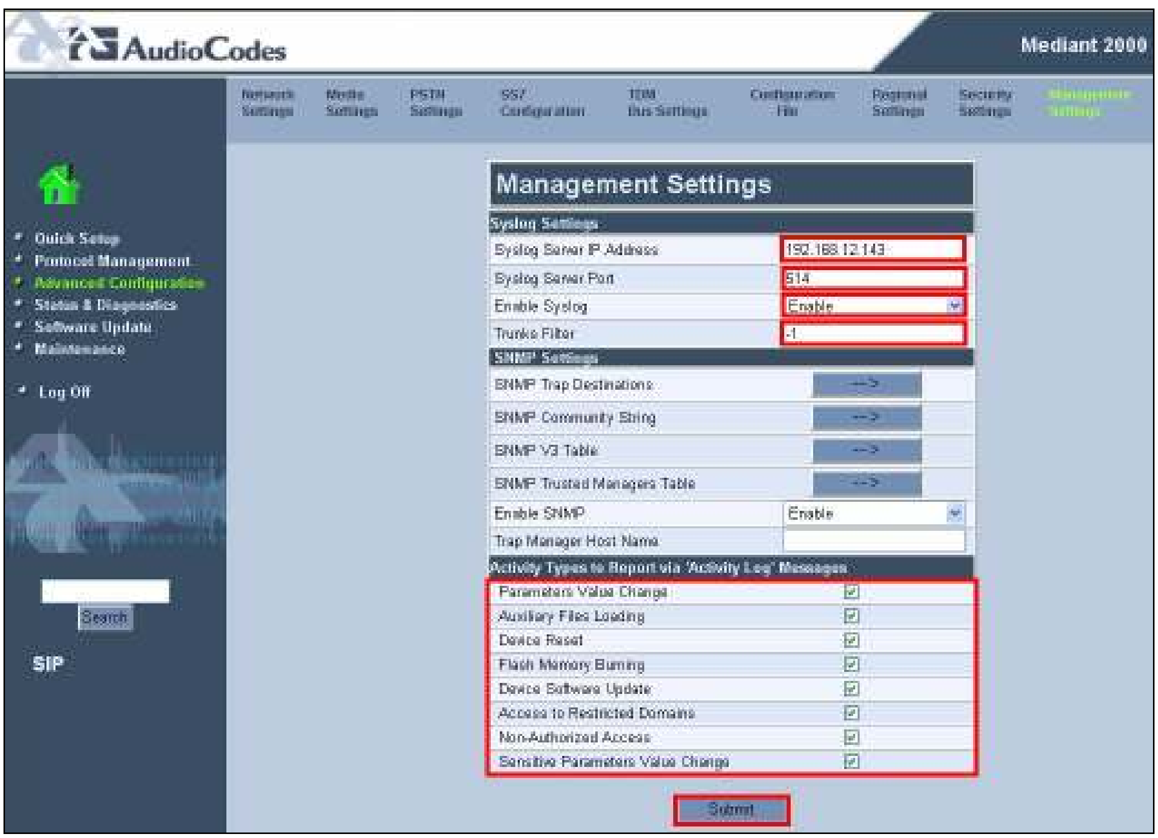
[5] Mediant™ 2000 User’s Manual version 5.2, Document # LTRT-69806, August 2007.

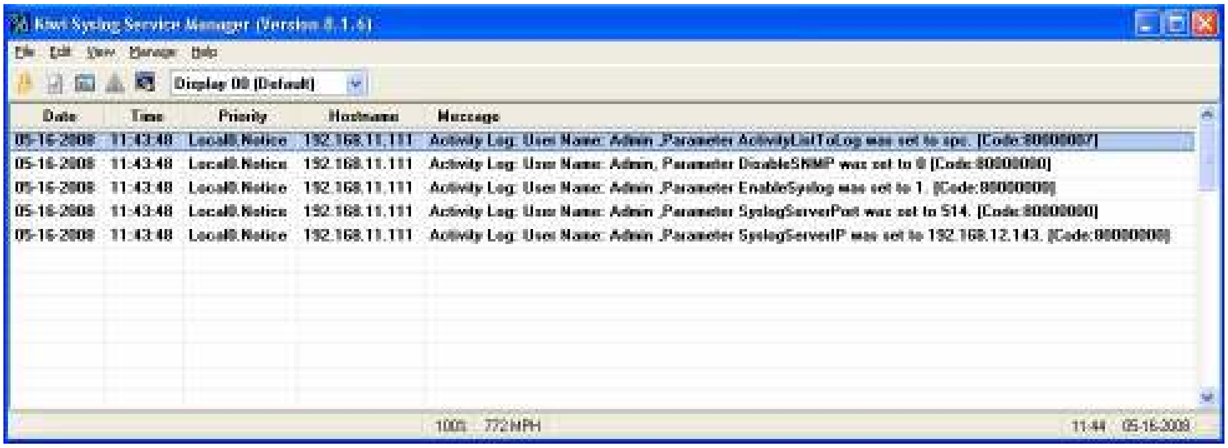
[6] Mediant™ 2000 & TP-1610 & TP-260-UNI SIP User's Manual Ver 5.2, Document #: LTRT-68806, September 2007.

9. Appendix

Appendix A - AudioCodes Mediant 2000 Syslog Configuration

Step	Description
A1	Enable syslog functionality on the AudioCodes Mediant 2000 as follows: • From the embedded web server, select **Advanced Configuration ➔ Management Settings**. • Set the **Syslog Server IP Address** and **Syslog Server Port** fields appropriately. • Select **Enable** from the drop-down list for the **Enable Syslog** field. • Set the **Trunks Filter** field to enable syslog reporting for the appropriate trunk(s). For this sample configuration, **-1** is used, which sets no filter, thus allowing reporting for all trunks to appear in the syslog. • Enable logging by selecting the appropriate fields under the **Activity Types to Report via 'Activity Log' Messages** heading. If enabled, any action associated with the enabled field will be logged (see **Step A2** for a sample syslog regarding 'Activity Types'). • Click **Submit**.



Step	Description
A2	For this sample configuration, it is assumed that a syslog server is configured and has IP network connectivity with the AudioCodes Mediant 2000. Below is a sample syslog obtained from the syslog server that display the events initiated in **Step A1**. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>A3</td><td>
Configure the log level to capture messages for debug traces as follows:

From the embedded web server, select Protocol Management ➔ Advanced Parameters ➔ General Parameters.
Select the appropriate setting from the drop-down list for the Debug Level field. For this sample configuration, the Debug Level field is set to 4 (see Step A4 and Step A5 for sample logging regarding ‘debug traces’).
[Button Not Shown] Click Submit.

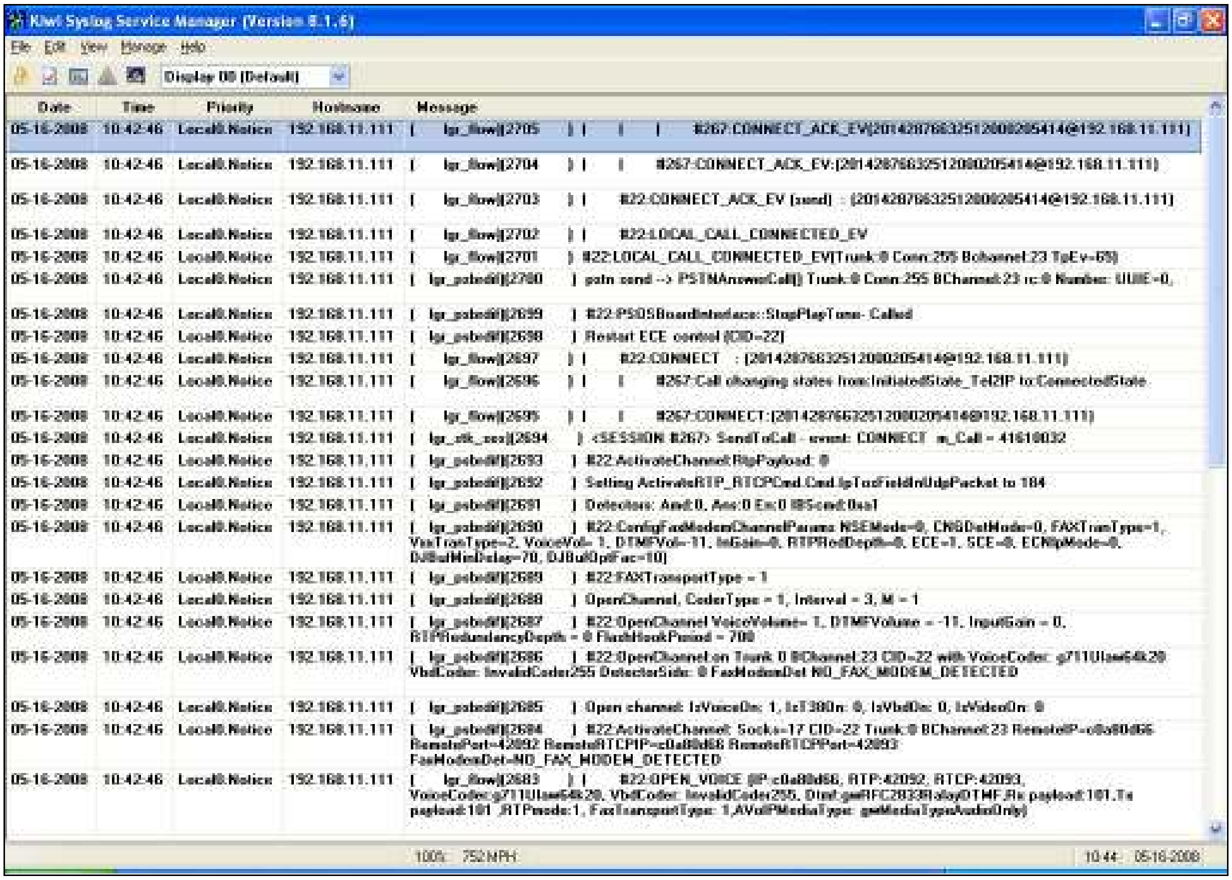
Note: It is recommended to provision the Debug Level field to the highest setting for debug traces. Setting the Debug Level field to 0 turns off logging for debug traces and will not affect the syslog of ‘Activity Type’ messages, as configured in Step A1.

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

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>A4</td><td>With logging set at the appropriate level, messages will get logged to the embedded web server on the AudioCodes Mediant 2000.

Note: This display is from the call originating from the PSTN to Avaya Meeting Exchange via the AudioCodes Mediant 2000 (see Step 6.3).


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

Step	Description
A5	With logging set at the appropriate level and syslog enabled, messages will get logged to the syslog server. ***Note****: This display is from the call originating from the PSTN to Avaya Meeting Exchange via the AudioCodes Mediant 2000 (see* ***Step 6.3****).* 

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