



## **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

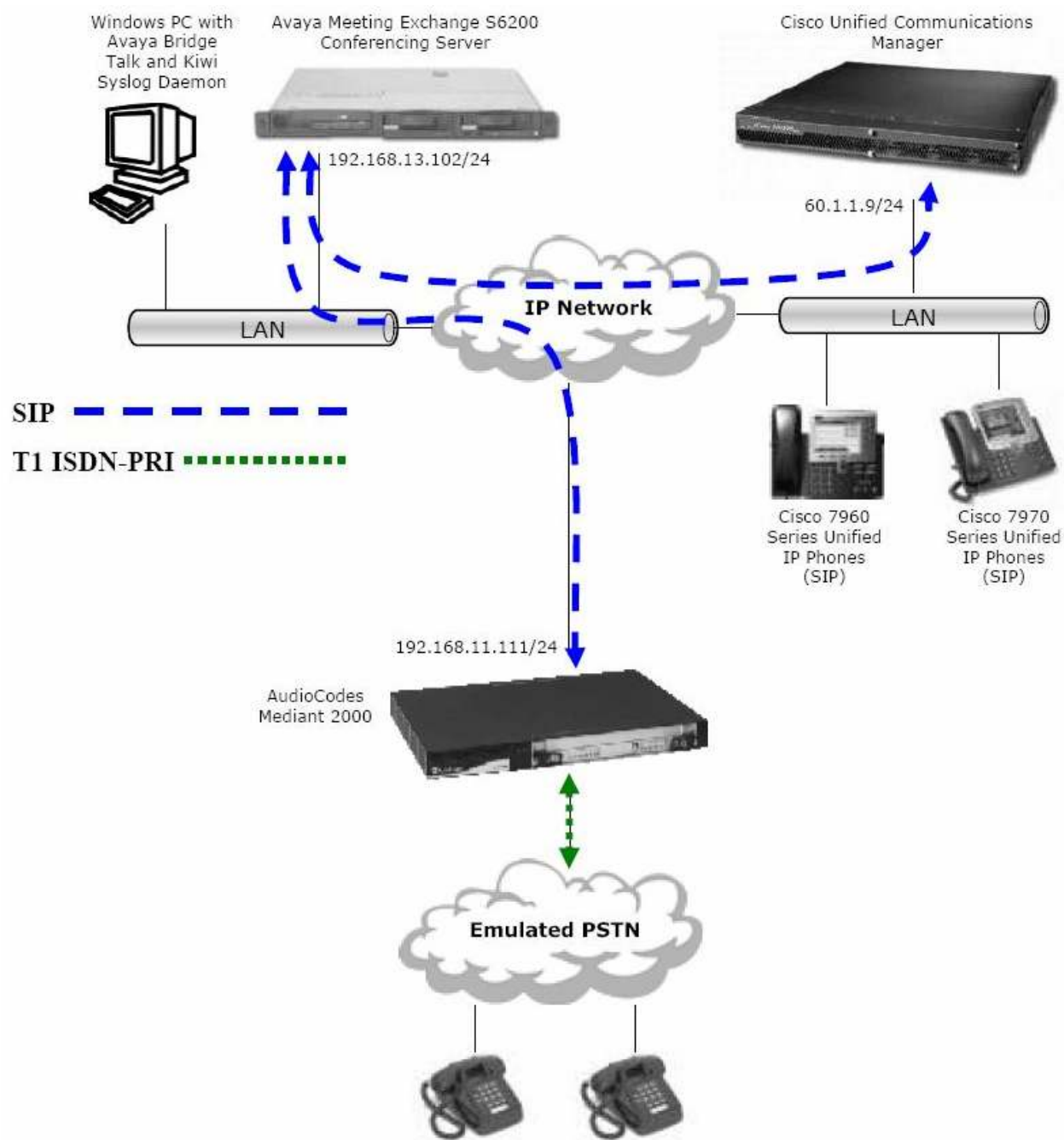
These Application Notes describe the procedures for configuring connectivity between the Avaya Meeting Exchange S6200 Conferencing Server (Meeting Exchange), Cisco Unified Communications Manager (UCM) and the PSTN via the AudioCodes Mediant 2000. Employing this configuration enables call origination/termination 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 on 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 <ul style="list-style-type: none"><li>• Version ID</li><li>• DSP Type</li><li>• DSP Software Version</li><li>• DSP Software Name</li><li>• Flash Version</li><li>• Module Firmware</li></ul>	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.

Step	Description
3.1.1	<p>Administer settings that enable SIP connectivity between Avaya Meeting Exchange and Cisco UCM and the AudioCodes Mediant 2000 by editing the <b>system.cfg</b> file as follows:</p> <ul style="list-style-type: none"> <li>From the <b>/usr/ipcb/config</b> directory, edit the <b>system.cfg</b> file with a text editor.</li> <li>Enter the IP address of Avaya Meeting Exchange (as defined in the <b>/etc/hosts</b> file) for the <b>IPAddress</b> variable.</li> <li>Enter a SIP Uniform Resource Identifier (URI) for Avaya Meeting Exchange that conforms to SIP standards for the <b>MyListener</b> variable. This entry is used to populate the “From” Header Field in SIP INVITE messages from Avaya Meeting Exchange. The “User” Field, <b>S6200</b>, must conform to SIP standards and is selected to uniquely identify this server. For example, <b>S6200</b> 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.</li> <li>Enter a SIP-URI that conforms to SIP standards and is bounded by angled brackets for the <b>respContact</b> 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.  <i>Note: if the <b>respContact</b> 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 <b>transport-param</b> to <b>udp</b>.</i></li> <li>Enter a value in seconds for the <b>minSETimerValue</b> 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 <b>minSETimerValue</b> 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.</li> </ul> <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=&lt;sip:S6200@192.168.13.102:5060;transport=udp&gt; respContact=&lt;sip:S6200@192.168.13.102:5060;transport=tcp&gt;  # Min SE value in seconds for lower bound of Session Interval for SIP Invite minSETimerValue=1800</pre>

## 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	<p>Administer settings to associate incoming calls to Avaya Meeting Exchange with corresponding call branding by adding URI to telephone number translations to the <b>UriToTelnum.tab</b> file. These translations extract values for both the Direct Inward Dial (DID, also known as DDI in Europe) and the Automatic Number Identification (ANI).</p> <ul style="list-style-type: none"> <li>From the <b>/usr/ipcb/config</b> directory, edit the <b>UriToTelnum.tab</b> file with a text editor.</li> <li>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. <ul style="list-style-type: none"> <li>The rules under the <b>TelnumPattern</b> and <b>TelnumConversion</b> columns work in conjunction as follows. Assume Cisco UCM sends a SIP INVITE message with the following “To” and “From” Header Fields. The rule “*&lt;sip:*@*” matches the following: <ul style="list-style-type: none"> <li>To: &lt;sip:555@192.168.13.102&gt;, where \$2 utilizes 555, the variable matched by the second asterisk as the DID value for the call.</li> <li>From: &lt;sip:917325551236@60.1.1.9&gt;, where \$2 utilizes 917325551236, the variable matched by the second asterisk as the ANI for the call.</li> </ul> </li> </ul> </li> <li>[<b>Not Required</b>] Add rules to support operator dial-in. Refer to [2] for information regarding this feature. For this sample configuration, “*&lt;sip:501@*” is utilized.</li> <li>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.</li> </ul> <p><i><b>Note:</b> 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.</i></p> <pre># 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 "&lt;sip:3333@10.220.10.4&gt;" and one of the patterns was # "&lt;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 "*&lt;sip:501@*"      "OP501x1"             Opl_From_CiscoUCM "*&lt;sip:*@*"        \$2                    CiscoUCM *                  \$0                    wildcard</pre>

Step	Description
3.2.2	<p>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 <b>telnumToUri.tab</b> file as follows:</p> <ul style="list-style-type: none"> <li>From the <b>/usr/ipcb/config</b> directory, edit the <b>telnumToUri.tab</b> file with a text editor.</li> <li>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. <ul style="list-style-type: none"> <li>The rules entered under the <b>TelnumPattern</b> column match digit patterns with a leading ‘3’ or ‘9’ and correspond to station extensions on Cisco UCM and the PSTN respectively.</li> <li>The SIP-URI entered under the <b>TelnumConversion</b> 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 <b>5060</b> and <b>transport=tcp</b>. Avaya Meeting Exchange will replace <b>\$0</b> with the “dialstring” in outgoing SIP INVITE messages. For example, if <b>917325551234</b> is dialed, Avaya Meeting Exchange will format a SIP INVITE message with the following SIP-URI in the Request-Line and “To” Header Field: <ul style="list-style-type: none"> <li><b>sip:917325551234@60.1.1.9:5060;transport=tcp</b></li> </ul> </li> </ul> </li> </ul> <p><i><b>Note:</b> Alternatively, call routing to Cisco UCM could have been enabled with the following entry:</i>  <b>9?????????? sip:\$0@60.1.1.9:5060;transport=tcp</b>, where “?” is a wildcard and matches any digit. To enable SIP connectivity over UDP, set the transport-param in the SIP-URI to <b>udp</b>. Similarly for the AudioCodes Mediant 2000.</p> <pre># 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</pre>



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

### 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	<p>Administer call branding for a direct call flow as follows:</p> <ul style="list-style-type: none"> <li>From the <b>/usr/dcb/bin</b> directory, add an entry to the call branding table to map the DID value obtained from procedures in <b>Step 3.2.1</b> to a conference by entering “<b>cbutil add 555 0 301 1 n direct</b>” at the command prompt. The syntax for this command is case insensitive and is defined as follows:  <b>cbutil add &lt;dnis&gt; &lt;rg&gt; &lt;msg&gt; &lt;ps&gt; &lt;ucps&gt; &lt;func&gt; [-l &lt;ln&gt; -c &lt;cn&gt;]</b>, where, <ul style="list-style-type: none"> <li>&lt;dnis&gt; DNIS</li> <li>&lt;rg&gt; Reservation group</li> <li>&lt;msg&gt; Annunciator message number</li> <li>&lt;ps&gt; Prompt set number (0-20)</li> <li>&lt;ucps&gt; Use conference prompt set (y/n)</li> <li>&lt;func&gt; One of: DIRECT/SCAN/ENTER/HANGUP/AUTOVL/FLEX</li> <li>-l &lt;“ln”&gt; Optional line name to associate with caller</li> <li>-c &lt;“cn”&gt; Optional company name to associate with caller</li> </ul> </li> </ul>
	<pre>S6200-&gt; cbutil add 555 0 301 1 n direct cbutil Copyright 2004 Avaya, Inc. All rights reserved.</pre>
3.3.2	Repeat <b>Step 3.3.1</b> to add an entry to the call branding table for a scan call flow.
	<pre>S6200-&gt; cbutil add 500 0 1 1 n scan cbutil Copyright 2004 Avaya, Inc. All rights reserved.</pre>

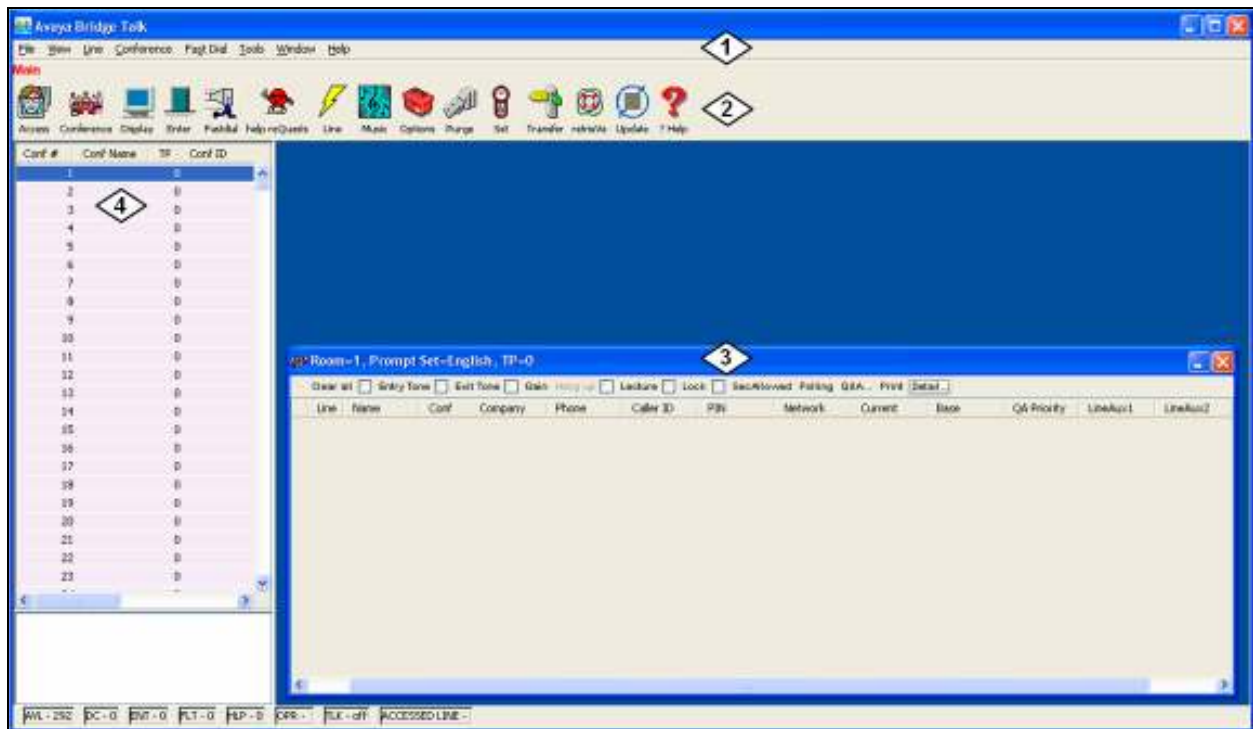
Step	Description																																								
3.3.3	<p>At the command prompt, enter “<b>cbutil list</b>” to verify the entries provisioned in <b>Step 3.3.1</b> and <b>Step 3.3.2</b>.</p> <p><i><b>Note:</b> The last entry in the call branding table, with a <b>DNIS</b> value <b>???</b>, was added previously and is a wild card entry. This entry captures any wrong number (e.g., unmatched <b>DID</b> values) and places the call into the enter queue for operator assistance.</i></p> <pre>S6200-&gt; cbutil list cbutil Copyright 2004 Avaya, Inc. All rights reserved.</pre> <table><thead><tr><th>DNIS</th><th>Grp</th><th>Msg</th><th>PS</th><th>CP</th><th>Function</th><th>Line Name</th><th>Company Name</th></tr><tr><th>-----</th><th>---</th><th>---</th><th>---</th><th>---</th><th>-----</th><th>-----</th><th>-----</th></tr></thead><tbody><tr><td>500</td><td>0</td><td>1</td><td>1</td><td>N</td><td>SCAN</td><td></td><td></td></tr><tr><td>555</td><td>0</td><td>301</td><td>1</td><td>N</td><td>DIRECT</td><td></td><td></td></tr><tr><td>???</td><td>0</td><td>208</td><td>1</td><td>N</td><td>ENTER</td><td></td><td></td></tr></tbody></table>	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		
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555	0	301	1	N	DIRECT																																				
???	0	208	1	N	ENTER																																				

### 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**

Step	Description
3.4.1	<p>Create a dial list of participants on Avaya Meeting Exchange. From the Avaya Bridge Talk Menu Bar, select <b>Fast Dial → New</b>. From the <b>New Dial List</b> window that is displayed, add participants to the dial list as follows:</p> <ul style="list-style-type: none"> <li>• Enter a descriptive name for this dial list in the <b>Name</b> field.</li> <li>• Add entries to the dial list by clicking <b>Add</b> for each participant. <ul style="list-style-type: none"> <li>○ Enter a descriptive name for each participant in the <b>Name</b> field.</li> <li>○ Enter a number in the <b>Telephone</b> field that corresponds to the participants' telephone number.</li> </ul> </li> <li>• Enable conference participants on the dial list to enter the conference without a passcode by checking the <b>Directly to Conf</b> box.</li> <li>• Refer to [3] for definitions regarding the remaining fields on this screen.</li> <li>• Click <b>Save</b>.</li> </ul>

**New Dial List**

Name:  Optional Access Code:  ☒ Directly to Conf

Conferee List

☒ Display As Entered

Name	Company	Moderator	Q&A Prio...	Telephone
Cisco-1234		<input type="checkbox"/>		917325551234
Cisco-1235		<input type="checkbox"/>		917325551235
PSTN-34001		<input type="checkbox"/>		34001

Step	Description
3.4.2	<p>Schedule conferences that utilize the call branding for a direct call flow provisioned in <b>Step 3.3.1</b> as follows. From the Menu Bar, click <b>View → Conference Scheduler</b>. From the <b>Conference Scheduler</b> window that is displayed, click <b>File → Schedule Conference</b>. From the <b>Schedule Conference</b> window that is displayed, administer settings as follows:</p> <ul style="list-style-type: none"> <li>• Enter a unique passcode in the <b>Conferee Code</b> field to allow access to this conference.</li> <li>• Enter a unique passcode in the <b>Moderator Code</b> field to allow access to this conference with moderator/host privileges. Note, to enable access to this conference without entering a passcode, define a <b>Moderator Code</b> that aligns with the provisioning for a direct call flow (see <b>Step 3.3.1</b>).</li> <li>• Enter a descriptive name for this conference in the <b>Conference Name</b> field.</li> <li>• Administer settings to enable a blast dial by setting the <b>Auto Blast</b> field to <b>Manual</b> and selecting the dial list provisioned in <b>Step 3.4.1</b> in the <b>Dial List</b> field. <ul style="list-style-type: none"> <li>◦ Select a dial list by clicking <b>Dial List</b>.</li> <li>◦ <i>[Not Shown] Select a dial list from the <b>Create, Select or Edit Dial List</b> window that is displayed.</i></li> </ul> </li> <li>• Refer to [3] for definitions regarding the remaining fields on this screen.</li> <li>• Click <b>OK</b>.</li> </ul>


## 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


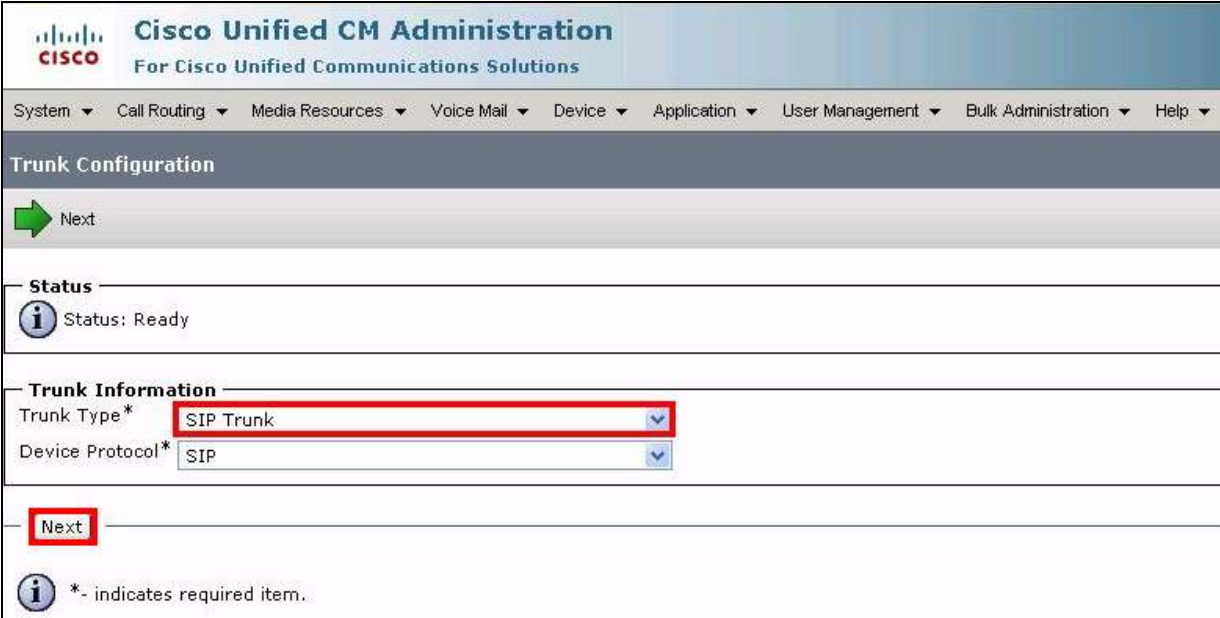
This section describes the steps for configuring SIP connectivity between Cisco UCM and Avaya Meeting Exchange.

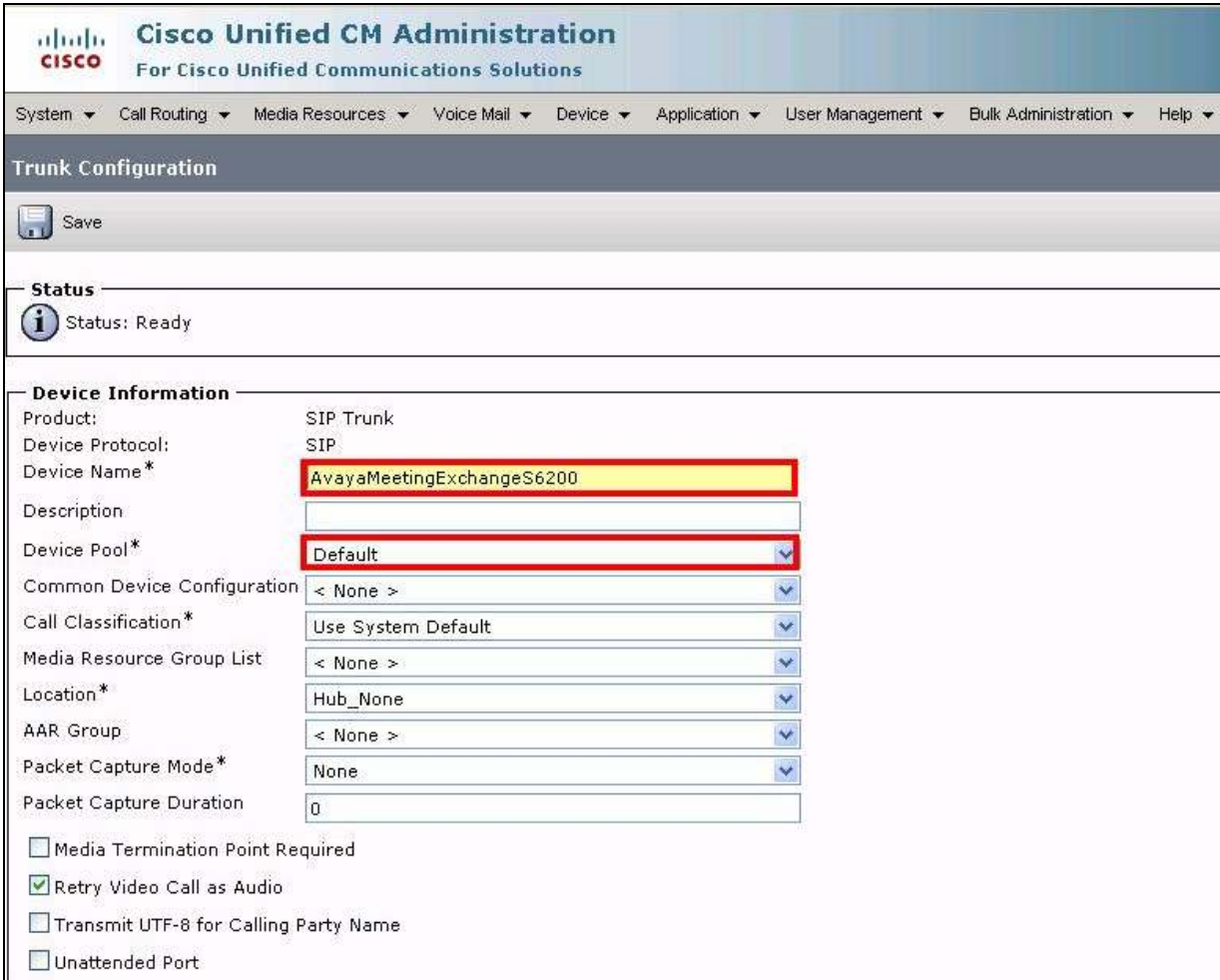
Step	Description
4.1.1	<p>To enable SIP connectivity with Avaya Meeting Exchange utilizing TCP, configure a <b>SIP Trunk Security Profile</b> as follows:</p> <ul style="list-style-type: none"><li>From the Cisco UCM main menu, select <b>System</b> → <b>Security Profile</b> → <b>SIP Trunk Security Profile</b>.</li><li>[<i>Not Shown</i>] Click <b>Add New</b> to create a new <b>SIP Trunk Security Profile</b>.</li><li>Provision settings as displayed and click <b>Save</b>.</li></ul> <p><i>Note: To enable SIP connectivity to Avaya Meeting Exchange utilizing UDP, set the <b>Outgoing Transport Type</b> field to UDP.</i></p>

Step	Description
4.1.2a	<p>To enable SIP connectivity with Avaya Meeting Exchange, configure a <b>SIP Profile</b> as follows:</p> <ul style="list-style-type: none"> <li>• From the Cisco UCM main menu, select <b>Device → Device Settings → SIP Profile</b>.</li> <li>• <i>[Not Shown]</i> Click <b>Add New</b> to create a new <b>SIP Profile</b>.</li> <li>• Provision settings under <b>SIP Profile Information</b> as displayed and scroll down.</li> </ul> 

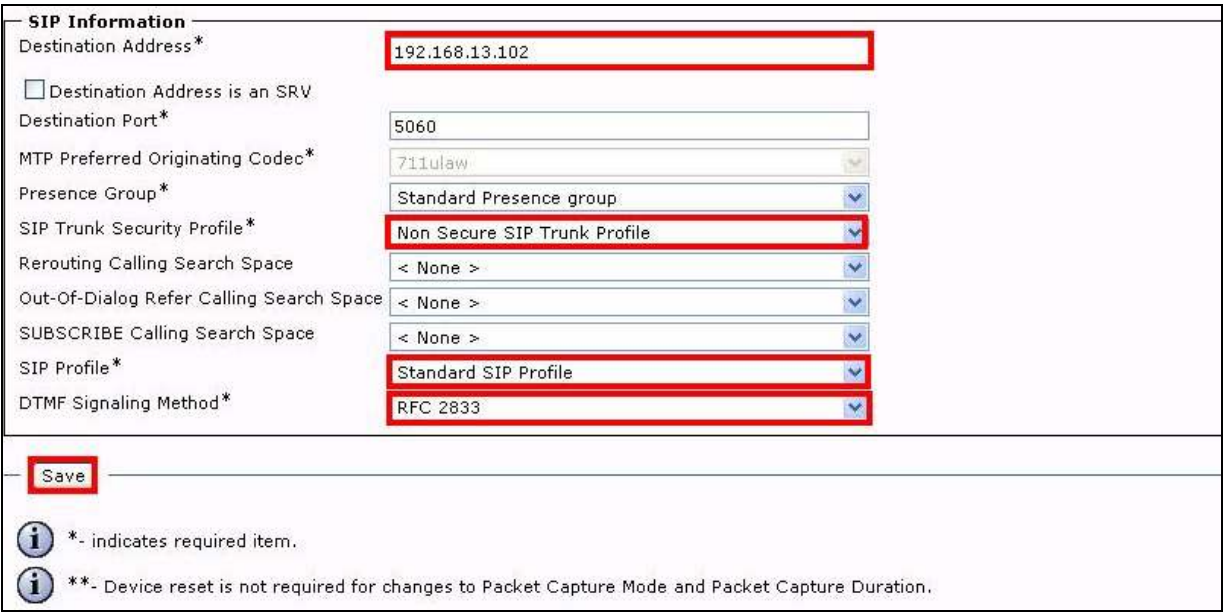




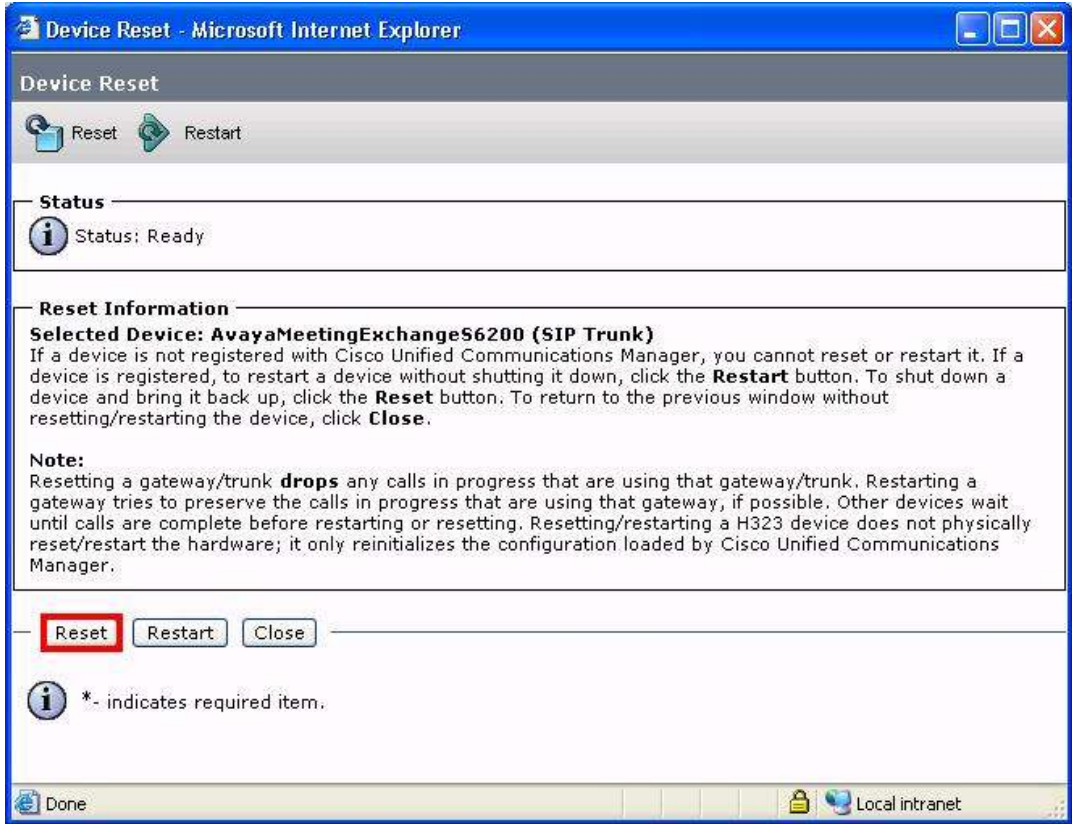
Step	Description																																																						
4.1.2b	<p>Use default settings under <b>Parameters used in Phone</b> as displayed and click <b>Save</b>.</p> <div> <p><b>Parameters used in Phone</b></p> <table> <tr><td>Timer Invite Expires (seconds)*</td><td>180</td></tr> <tr><td>Timer Register Delta (seconds)*</td><td>5</td></tr> <tr><td>Timer Register Expires (seconds)*</td><td>3600</td></tr> <tr><td>Timer T1 (msec)*</td><td>500</td></tr> <tr><td>Timer T2 (msec)*</td><td>4000</td></tr> <tr><td>Retry INVITE*</td><td>6</td></tr> <tr><td>Retry Non-INVITE*</td><td>10</td></tr> <tr><td>Start Media Port*</td><td>16384</td></tr> <tr><td>Stop Media Port*</td><td>32766</td></tr> <tr><td>Call Pickup URI*</td><td>x-cisco-serviceuri-pickup</td></tr> <tr><td>Call Pickup Group Other URI*</td><td>x-cisco-serviceuri-opickup</td></tr> <tr><td>Call Pickup Group URI*</td><td>x-cisco-serviceuri-gpickup</td></tr> <tr><td>Meet Me Service URI*</td><td>x-cisco-serviceuri-meetme</td></tr> <tr><td>User Info*</td><td>None</td></tr> <tr><td>DTMF DB Level*</td><td>Nominal</td></tr> <tr><td>Call Hold Ring Back*</td><td>Off</td></tr> <tr><td>Anonymous Call Block*</td><td>Off</td></tr> <tr><td>Caller ID Blocking*</td><td>Off</td></tr> <tr><td>Do Not Disturb Control*</td><td>User</td></tr> <tr><td>Telnet Level for 7940 and 7960*</td><td>Disabled</td></tr> <tr><td>Timer Keep Alive Expires (seconds)*</td><td>120</td></tr> <tr><td>Timer Subscribe Expires (seconds)*</td><td>120</td></tr> <tr><td>Timer Subscribe Delta (seconds)*</td><td>5</td></tr> <tr><td>Maximum Redirections*</td><td>70</td></tr> <tr><td>Off Hook To First Digit Timer (milliseconds)*</td><td>15000</td></tr> <tr><td>Call Forward URI*</td><td>x-cisco-serviceuri-cfwdall</td></tr> <tr><td>Abbreviated Dial URI*</td><td>x-cisco-serviceuri-abbrdial</td></tr> </table> <p> <input checked="" type="checkbox"/> Conference Join Enabled  <input type="checkbox"/> RFC 2543 Hold  <input checked="" type="checkbox"/> Semi Attended Transfer  <input type="checkbox"/> Enable VAD  <input type="checkbox"/> Stutter Message Waiting  <input type="checkbox"/> Call Stats </p> <p><b>Save</b></p> <p> *- indicates required item.</p> </div>	Timer Invite Expires (seconds)*	180	Timer Register Delta (seconds)*	5	Timer Register Expires (seconds)*	3600	Timer T1 (msec)*	500	Timer T2 (msec)*	4000	Retry INVITE*	6	Retry Non-INVITE*	10	Start Media Port*	16384	Stop Media Port*	32766	Call Pickup URI*	x-cisco-serviceuri-pickup	Call Pickup Group Other URI*	x-cisco-serviceuri-opickup	Call Pickup Group URI*	x-cisco-serviceuri-gpickup	Meet Me Service URI*	x-cisco-serviceuri-meetme	User Info*	None	DTMF DB Level*	Nominal	Call Hold Ring Back*	Off	Anonymous Call Block*	Off	Caller ID Blocking*	Off	Do Not Disturb Control*	User	Telnet Level for 7940 and 7960*	Disabled	Timer Keep Alive Expires (seconds)*	120	Timer Subscribe Expires (seconds)*	120	Timer Subscribe Delta (seconds)*	5	Maximum Redirections*	70	Off Hook To First Digit Timer (milliseconds)*	15000	Call Forward URI*	x-cisco-serviceuri-cfwdall	Abbreviated Dial URI*	x-cisco-serviceuri-abbrdial
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Call Forward URI*	x-cisco-serviceuri-cfwdall																																																						
Abbreviated Dial URI*	x-cisco-serviceuri-abbrdial																																																						

Step	Description
4.1.3	<p>To enable SIP connectivity with Avaya Meeting Exchange, configure a <b>SIP Trunk</b> as follows:</p> <ul style="list-style-type: none"> <li>From the Cisco UCM main menu, select <b>Device → Trunk</b>.</li> <li>Click <b>Add New</b> to create a new <b>SIP Trunk</b>.</li> </ul> 
4.1.4	<p>Select <b>SIP Trunk</b> from the drop-down list for the <b>Trunk Type</b> field. Accept the default setting for the <b>Device Protocol</b> field and click <b>Next</b>.</p> 

Step	Description
4.1.5a	<p>Provision settings under <b>Device Information</b> as displayed and scroll down. The <b>Location</b> field specifies the total bandwidth that is available for calls between this location and the central location, or hub. Using the default setting <b>Hub_None</b> specifies unlimited available bandwidth.</p>  <p>The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes links for System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled 'Trunk Configuration' and features a 'Save' button. Below this, the 'Status' section shows 'Status: Ready'. The 'Device Information' section contains the following fields:</p> <ul style="list-style-type: none"> <li>Product: SIP Trunk</li> <li>Device Protocol: SIP</li> <li>Device Name*: AvayaMeetingExchangeS6200</li> <li>Description: (empty text box)</li> <li>Device Pool*: Default</li> <li>Common Device Configuration: &lt; None &gt;</li> <li>Call Classification*: Use System Default</li> <li>Media Resource Group List: &lt; None &gt;</li> <li>Location*: Hub_None</li> <li>AAR Group: &lt; None &gt;</li> <li>Packet Capture Mode*: None</li> <li>Packet Capture Duration: 0</li> </ul> <p>At the bottom of the Device Information section, there are four checkboxes:</p> <ul style="list-style-type: none"> <li><input type="checkbox"/> Media Termination Point Required</li> <li><input checked="" type="checkbox"/> Retry Video Call as Audio</li> <li><input type="checkbox"/> Transmit UTF-8 for Calling Party Name</li> <li><input type="checkbox"/> Unattended Port</li> </ul>

Step	Description
4.1.5b	<p>Use default settings as displayed and scroll down.</p> <div data-bbox="297 338 1513 1045"> <div> <b>Multilevel Precedence and Preemption (MLPP) Information</b> </div> <div> MLPP Domain &lt; None &gt; </div> <div> <b>Call Routing Information</b> </div> <div> <b>Inbound Calls</b> </div> <div> <div>Significant Digits*</div> <div>All</div> </div> <div> <div>Connected Line ID Presentation*</div> <div>Default</div> </div> <div> <div>Connected Name Presentation*</div> <div>Default</div> </div> <div> <div>Calling Search Space</div> <div>&lt; None &gt;</div> </div> <div> <div>AAR Calling Search Space</div> <div>&lt; None &gt;</div> </div> <div> <div>Prefix DN</div> <div></div> </div> <div> <input type="checkbox"/> Redirecting Diversion Header Delivery - Inbound </div> <div> <b>Outbound Calls</b> </div> <div> <div>Calling Party Selection*</div> <div>Originator</div> </div> <div> <div>Calling Line ID Presentation*</div> <div>Default</div> </div> <div> <div>Calling Name Presentation*</div> <div>Default</div> </div> <div> <div>Caller ID DN</div> <div></div> </div> <div> <div>Caller Name</div> <div></div> </div> <div> <input type="checkbox"/> Redirecting Diversion Header Delivery - Outbound </div> </div>

Step	Description
4.1.5c	<p>Provision settings under <b>SIP Information</b> as displayed.</p> <ul style="list-style-type: none"> <li>• Enter the IP address of Avaya Meeting Exchange in the <b>Destination Address</b> field.</li> <li>• Select the SIP Trunk Security Profile provisioned in <b>Step 4.1.1</b> from the drop-down list for the <b>SIP Trunk Security Profile</b> field.</li> <li>• Select the SIP Profile provisioned in <b>Step 4.1.2</b> from the drop-down list for the <b>SIP Profile</b> field.</li> <li>• Select <b>RFC 2833</b> from the drop-down list for the <b>DTMF Signaling Method</b> field.</li> <li>• Click <b>Save</b>.</li> </ul> 
4.1.6	<p>From the pop-up window, click <b>OK</b> and reset the trunk by clicking <b>Reset</b>,  [<i>Not Shown, located at the bottom of the SIP Trunk page</i>].</p> 

Step	Description
4.1.7	<p>From the pop-up window, click <b>Reset</b>.</p> 



## 4.2. Configure Call Routing

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

Step	Description
4.2.1a	<p>To enable routing from Cisco UCM to Avaya Meeting Exchange utilizing the SIP trunk provisioned in <b>Section 4.1</b>, configure a <b>Route Pattern</b> as follows:</p> <ul style="list-style-type: none"> <li>From the Cisco UCM main menu, select <b>Call Routing → Route/Hunt → Route Pattern</b>.</li> <li>[<i>Not Shown</i>] Click <b>Add New</b> to create a new <b>Route Pattern</b>.</li> <li>Provision settings under <b>Pattern Definition</b> as displayed and scroll down. <ul style="list-style-type: none"> <li>Enter a pattern in the <b>Route Pattern</b> field that corresponds to the call branding for direct and scan call flows provisioned on Avaya Meeting exchange in <b>Section 3.3</b>. Note that “X” is a wildcard and represents any digit 0 through 9.</li> <li>Select the SIP trunk group provisioned in <b>Section 4.1</b> from the drop-down list for the <b>Gateway/Route List</b> field.</li> <li>Verify that the <b>Provide Outside Dial Tone</b> field is not selected.</li> </ul> </li> </ul>

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

**Route Pattern Configuration**

Save

**Status**  
 Status: Ready

**Pattern Definition**

Route Pattern\* SXX

Route Partition < None >

Description ToAvayaMeetingExchangeS6200

Numbering Plan -- Not Selected --

Route Filter < None >

MLPP Precedence\* Default

Gateway/Route List\* AvayaMeetingExchangeS6200 (Edit)

Route Option  
☒ Route this pattern  
☐ Block this pattern No Error

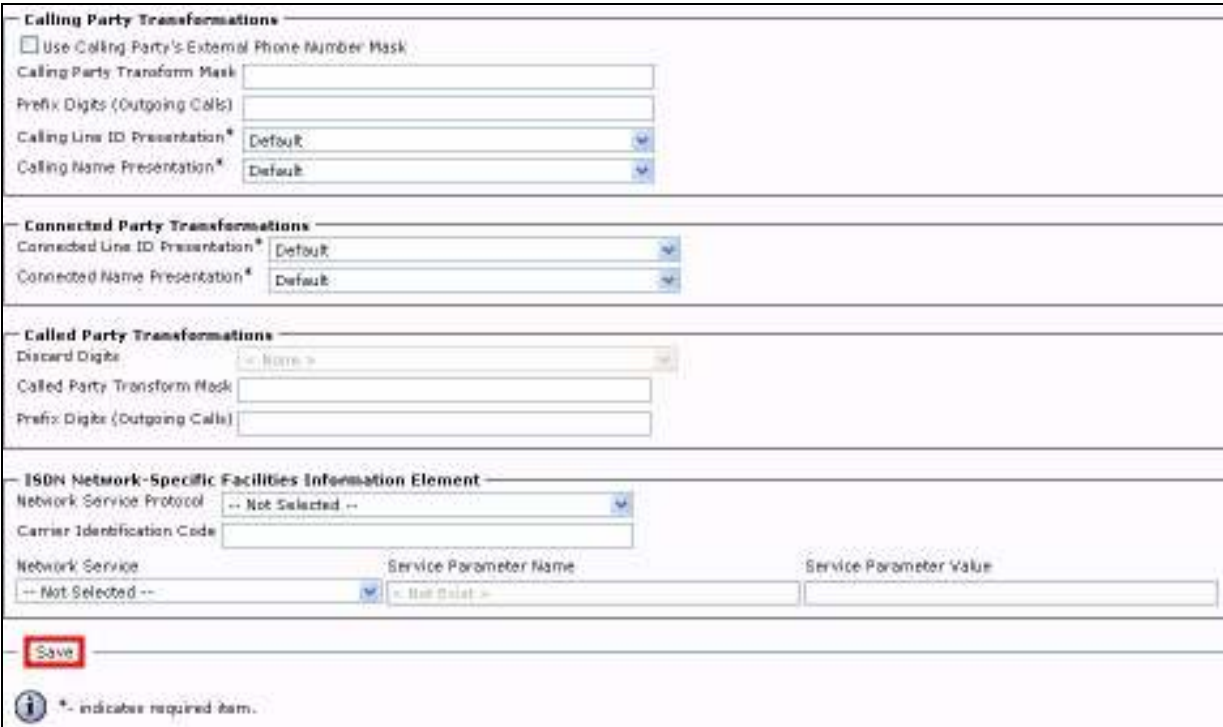
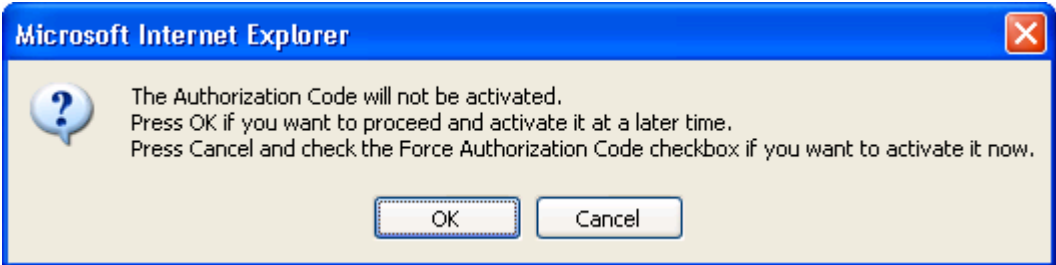
Call Classification\* OffNet

☐ Allow Device Override ☐ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority

☐ Require Forced Authorization Code

Authorization Level\* 0

☐ Require Client Matter Code

Step	Description
4.2.1b	<p>Use default settings as displayed and click <b>Save</b>.</p> 
4.2.2	<p>The <b>Require Forced Authorization Code</b> option was not enabled in <b>Step 4.2.1</b>, click <b>OK</b>.</p> 



## 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

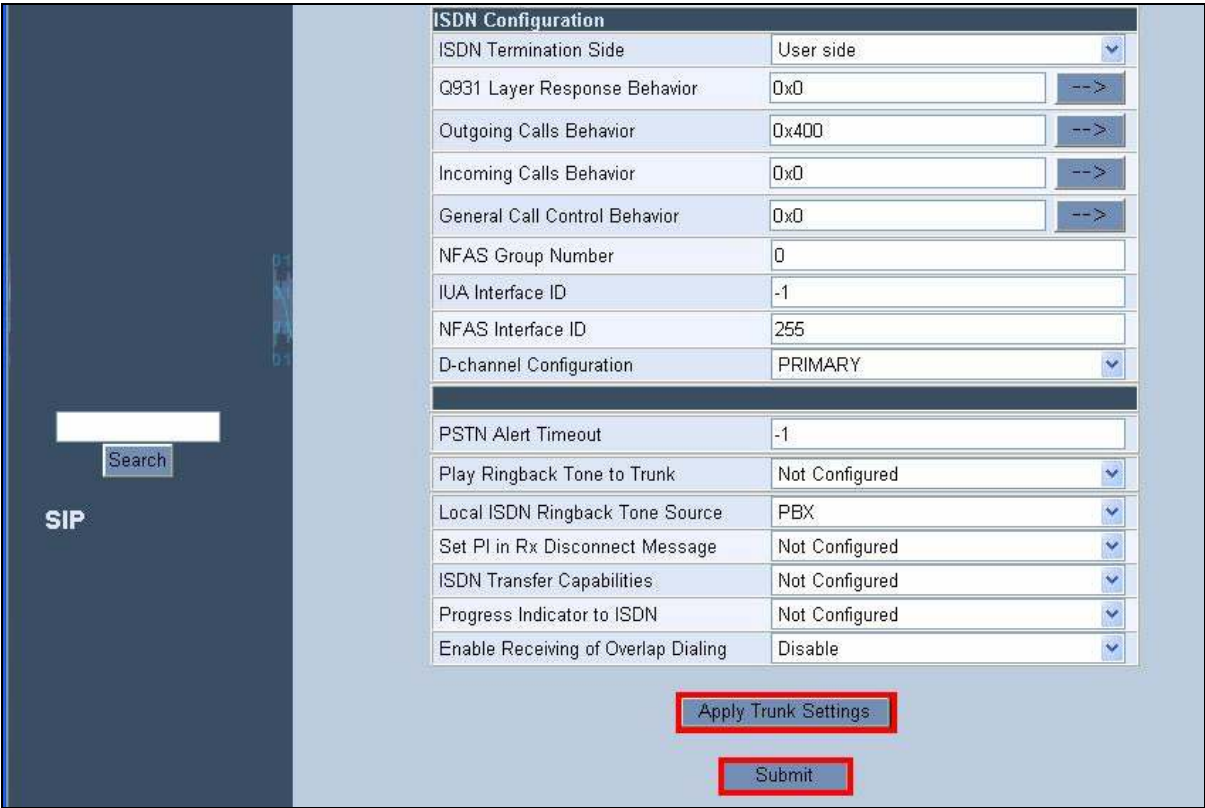
This section describes the steps for configuring T1 connectivity between the AudioCodes Mediant 2000 and the PSTN.

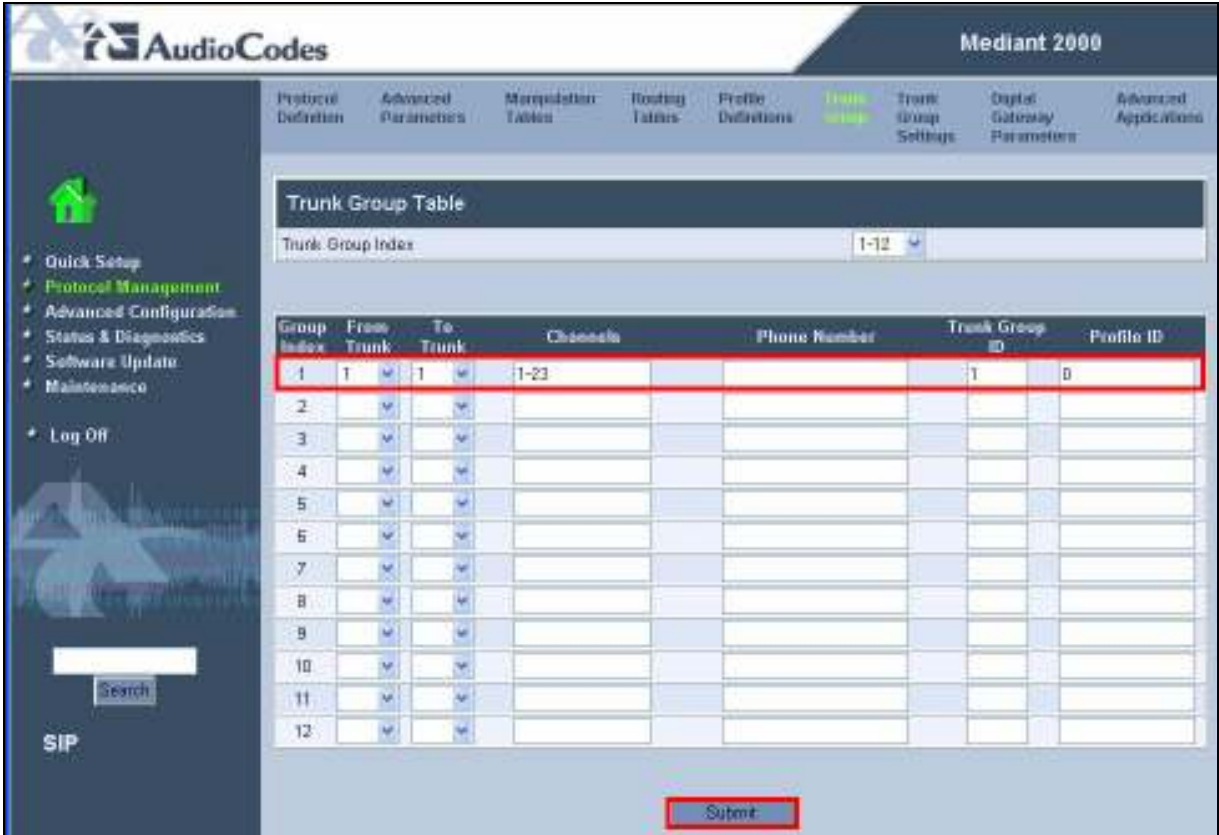
Step	Description
5.1.1a	<p>To enable connectivity to the PSTN, administer <b>Trunk Settings</b> as follows:</p> <ul style="list-style-type: none"> <li>From the embedded web server, select <b>Advanced Configuration</b> → <b>PSTN Settings</b> → <b>Trunk Settings</b>.</li> <li>Select the appropriate trunk to provision. For this sample configuration, T1 connectivity to the PSTN uses trunk 1.</li> <li>[<i>Not Shown</i>] Click the <b>Stop Trunk</b> button to modify the selected trunk's parameters. <ul style="list-style-type: none"> <li>The status of the <b>Trunk Configuration State</b> parameter changes to <b>Inactive</b>.</li> </ul> </li> <li>Configure <b>Trunk Settings</b> for this interface to enable T1 ISDN-PRI connectivity to the PSTN according to requirements defined by the PSTN service provider.</li> <li>Scroll down.</li> </ul>

The screenshot displays the AudioCodes Mediant 2000 web interface. The top navigation bar includes links for Network Settings, Media Settings, PSTN Settings (highlighted), SS7 Configuration, TDM Bus Settings, Configuration File, Regional Settings, Security Settings, and Management Settings. The left sidebar contains a home icon and a list of menu items: Quick Setup, Protocol Management, Advanced Configuration (highlighted in green), Status & Diagnostics, Software Update, Maintenance, and Log Off. The main content area is titled 'Trunk Settings' and shows a table for 'Trunk Configuration' with the following details:

Trunk Configuration	
Trunk ID	1
Trunk Configuration State	Inactive
Protocol Type	T1 5ESS 10 ISDN
Clock Master	Recovered
Auto Clock Trunk Priority	0
Line Code	B8ZS
Line Build Out Loss	0 dB
Trace Level	No Trace
Line Build Out Overwrite	OFF
Framing Method	T1 FRAMING ESF CRC6

At the top of the configuration area, there is a 'Trunk Status' bar showing 8 trunks. Trunk 1 is highlighted in yellow, and the others are red. Above this bar, there are tabs for 'CAS State Machines' and 'Cas State Machines'.

Step	Description
5.1.1b	<p>Provision settings as required.</p> <ul style="list-style-type: none"> <li>Click <b>Submit</b>.</li> <li>Click <b>Apply Trunk Settings</b>.</li> </ul>  <p>The screenshot shows a web interface for configuring ISDN settings. On the left is a dark blue sidebar with a 'SIP' label and a 'Search' button. The main area is light blue and contains a table of configuration settings. The table is divided into two sections. The first section, titled 'ISDN Configuration', includes settings for 'ISDN Termination Side' (User side), 'Q931 Layer Response Behavior' (0x0), 'Outgoing Calls Behavior' (0x400), 'Incoming Calls Behavior' (0x0), 'General Call Control Behavior' (0x0), 'NFAS Group Number' (0), 'IUA Interface ID' (-1), 'NFAS Interface ID' (255), and 'D-channel Configuration' (PRIMARY). The second section includes 'PSTN Alert Timeout' (-1), 'Play Ringback Tone to Trunk' (Not Configured), 'Local ISDN Ringback Tone Source' (PBX), 'Set PI in Rx Disconnect Message' (Not Configured), 'ISDN Transfer Capabilities' (Not Configured), 'Progress Indicator to ISDN' (Not Configured), and 'Enable Receiving of Overlap Dialing' (Disable). At the bottom of the configuration area, the 'Apply Trunk Settings' and 'Submit' buttons are highlighted with red boxes.</p>

Step	Description
5.1.2	<p>To assign common rules for routing IP-to-PSTN calls, administer a <b>Trunk Group</b> as follows:</p> <ul style="list-style-type: none"> <li>From the embedded web server, select, <b>Protocol Management → Trunk Group</b>.</li> <li>Select the appropriate trunk from the <b>Group Index</b> column. For this sample configuration, T1 connectivity to the PSTN uses trunk 1.</li> <li>Set the <b>From Trunk</b> and <b>To Trunk</b> fields to the starting and ending number of the trunk to the PSTN.</li> <li>Set the <b>Channels</b> 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.</li> <li>Set the <b>Trunk Group ID</b> field to an available number.</li> <li>Click <b>Submit</b>.</li> </ul> <p><i>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.</i></p> 

Step	Description
5.1.3	<p>To determine the method in which new calls are assigned to B-channels within a trunk group, administer <b>Trunk Group Settings</b> as follows:</p> <ul style="list-style-type: none"> <li>From the embedded web server, select <b>Protocol Management → Trunk Group Settings</b>.</li> <li>Add an entry that corresponds to trunk group provisioned in <b>Step 5.1.2</b> as displayed. <ul style="list-style-type: none"> <li>Set the <b>Trunk Group ID</b> field to the Trunk Group ID assigned to the trunk group provisioned in <b>Step 5.1.2</b>.</li> <li>Set the <b>Channel Select Mode</b> 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 <b>Ascending</b> mode, while call origination from the PSTN selects B-channels in a descending fashion.</li> </ul> <p><i>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).</i></p> </li> <li>Click <b>Submit</b>.</li> </ul>

The screenshot displays the 'Trunk Group Settings' page in the AudioCodes Mediant 2000 web interface. The page includes a navigation menu on the left with options like Quick Setup, Protocol Management, and Advanced Configuration. The main content area shows a table for configuring trunk groups. The first row is highlighted with a red border, indicating the current configuration. The table has columns for Trunk Group ID, Channel Select Mode, and Registration Mode. The first row shows Trunk Group ID 1, Channel Select Mode set to Ascending, and Registration Mode set to a dropdown menu. The table has 12 rows in total. A 'Submit' button is visible at the bottom right of the table area.

Trunk Group ID	Channel Select Mode	Registration Mode
1	Ascending	
2		
3		
4		
5		
6		
7		
8		
9		
10		
11		
12		

## 5.2. Configure SIP Connectivity

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

Step	Description
5.2.1	<p>To enable IP connectivity to the AudioCodes Mediant 2000, administer <b>IP Settings</b> as follows:</p> <ul style="list-style-type: none"> <li>From the embedded web server, select <b>Advanced Configuration → Network Settings → IP Settings</b>.</li> <li>Select the appropriate IP network configuration from the drop-down list for the <b>IP Networking Mode</b> field. For this sample configuration, Media, Control and OAM are on the same network.</li> <li>Set the <b>IP Address</b>, <b>Subnet Mask</b> and <b>Default Gateway Address</b> fields accordingly.</li> <li>Use default settings for remaining fields.</li> <li>Click <b>Submit</b>.</li> </ul>

**AudioCodes Mediant 2000**

Network Settings | Media Settings | PSTN Settings | SS7 Configuration | TDM Bus Settings | Configuration File | Regional Settings | Security Settings | Management Settings

IP Settings | Application Settings | Routing Table | VLAN Settings | SCTP Settings

**IP Settings**

IP Networking Mode: Single IP Network

IP Address: 192.168.11.111

Subnet Mask: 255.255.255.0

Default Gateway Address: 192.168.11.1

**DNS Settings**

DNS Primary Server IP: 10.2.1.2

DNS Secondary Server IP:

**DHCP Settings**

Enable DHCP: Disable

**NAT Settings**

! NAT IP Address: 0.0.0.0

**Differential Services**

Network QoS: 48

Media Premium QoS: 46

Control Premium QoS: 40

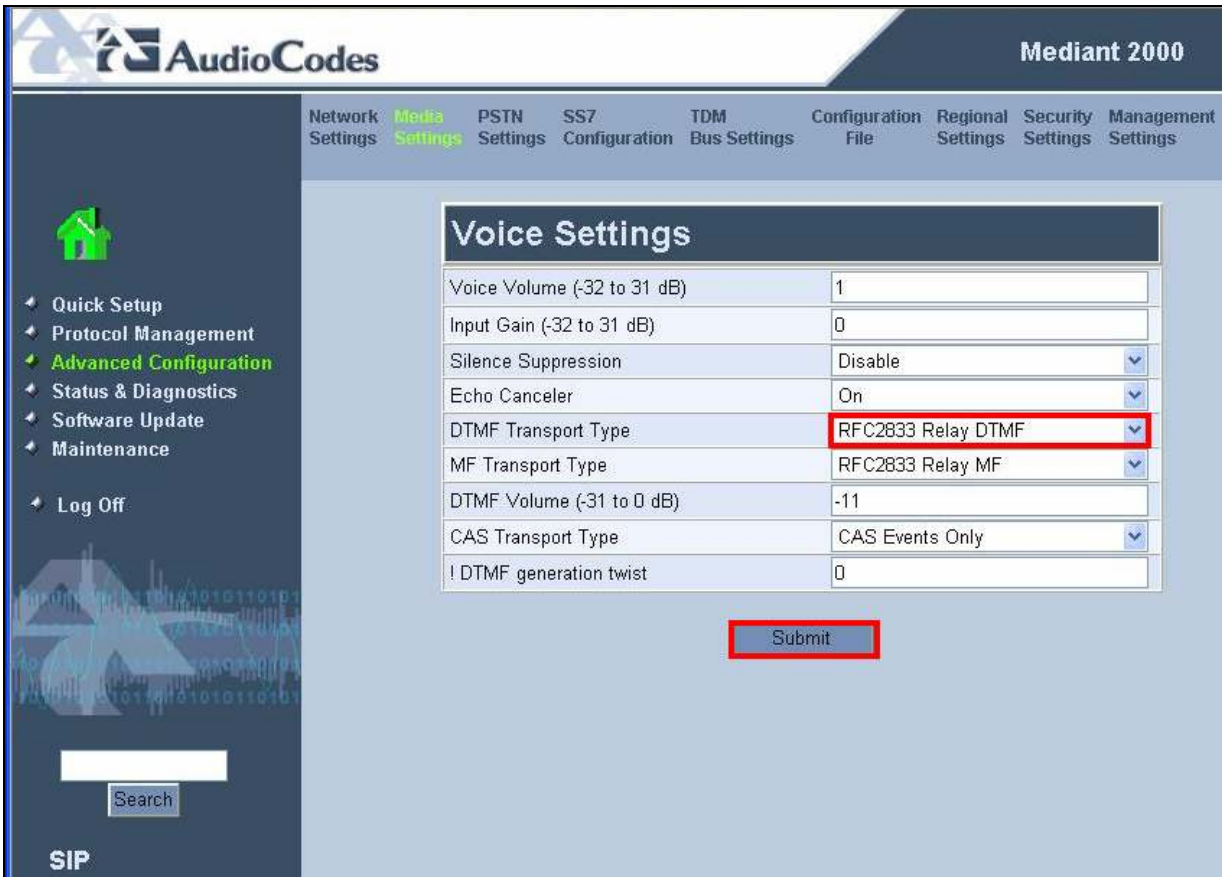
Gold QoS: 26

Bronze QoS: 10

**Submit**

When changing 'IP Networking Mode', click 'Submit' then 'Reset' (with the 'Burn To FLASH' selected).



Step	Description																				
5.2.2	<p>Administer <b>Media Settings</b> as displayed.</p> <ul style="list-style-type: none"> <li>Click on <b>Advanced Configuration → Media Settings → Voice Settings</b>.</li> <li>Select <b>RFC2833 Relay DTMF</b> from the drop-down list for the <b>DTMF Transport Type</b> field.</li> <li>Click <b>Submit</b>.</li> </ul>  <p>The screenshot displays the AudioCodes Mediant 2000 web interface. The top navigation bar includes links for Network Settings, Media Settings (highlighted), PSTN Settings, SS7 Configuration, TDM Bus Settings, Configuration File, Regional Settings, Security Settings, and Management Settings. The left sidebar contains a home icon and a list of menu items: Quick Setup, Protocol Management, Advanced Configuration (highlighted), Status &amp; Diagnostics, Software Update, Maintenance, and Log Off. The main content area is titled 'Voice Settings' and contains a table of configuration parameters:</p> <table border="1"> <thead> <tr> <th>Parameter</th> <th>Value</th> </tr> </thead> <tbody> <tr> <td>Voice Volume (-32 to 31 dB)</td> <td>1</td> </tr> <tr> <td>Input Gain (-32 to 31 dB)</td> <td>0</td> </tr> <tr> <td>Silence Suppression</td> <td>Disable</td> </tr> <tr> <td>Echo Canceler</td> <td>On</td> </tr> <tr> <td>DTMF Transport Type</td> <td>RFC2833 Relay DTMF</td> </tr> <tr> <td>MF Transport Type</td> <td>RFC2833 Relay MF</td> </tr> <tr> <td>DTMF Volume (-31 to 0 dB)</td> <td>-11</td> </tr> <tr> <td>CAS Transport Type</td> <td>CAS Events Only</td> </tr> <tr> <td>! DTMF generation twist</td> <td>0</td> </tr> </tbody> </table> <p>At the bottom right of the settings table, there is a 'Submit' button.</p>	Parameter	Value	Voice Volume (-32 to 31 dB)	1	Input Gain (-32 to 31 dB)	0	Silence Suppression	Disable	Echo Canceler	On	DTMF Transport Type	RFC2833 Relay DTMF	MF Transport Type	RFC2833 Relay MF	DTMF Volume (-31 to 0 dB)	-11	CAS Transport Type	CAS Events Only	! DTMF generation twist	0
Parameter	Value																				
Voice Volume (-32 to 31 dB)	1																				
Input Gain (-32 to 31 dB)	0																				
Silence Suppression	Disable																				
Echo Canceler	On																				
DTMF Transport Type	RFC2833 Relay DTMF																				
MF Transport Type	RFC2833 Relay MF																				
DTMF Volume (-31 to 0 dB)	-11																				
CAS Transport Type	CAS Events Only																				
! DTMF generation twist	0																				

Step	Description
5.2.3a	<p>To enable SIP connectivity with the Avaya Meeting Exchange, administer <b>General Parameters</b> as follows:</p> <ul style="list-style-type: none"> <li>From the embedded web server, select <b>Protocol Management</b> → <b>Protocol Definition</b> → <b>General Parameters</b>.</li> <li>Set the <b>SIP Transport Type</b>, <b>SIP TCP Local Port</b> and <b>SIP Destination Port</b> fields to enable SIP connectivity with the Avaya Meeting Exchange utilizing TCP. <ul style="list-style-type: none"> <li><i>Note: To enable SIP connectivity with Avaya Meeting Exchange utilizing UDP, set the <b>SIP Transport Type</b> and <b>SIP UDP Local Port</b> fields for UDP.</i></li> </ul> </li> <li>Use default settings for remaining fields.</li> <li>Scroll down.</li> </ul>

Parameter	Value
PRACK Mode	Supported
Channel Select Mode	Cyclic Ascending
Enable Early Media	Enable
183 Message Behavior	Progress
Session-Expires Time	0
Minimum Session-Expires	90
Session Expires Method	Re-Invite
Asserted Identity Mode	Disabled
Fax Signaling Method	T.38 Relay
I Detect Fax on Answer Tone	Initiate T.38 on Preamble
SIP Transport Type	TCP
SIP UDP Local Port	5060
SIP TCP Local Port	5060
SIP TLS Local Port	5061
Enable SIPs	Disable
Enable TCP Connection Reuse	Enable
TCP Timeout	0
SIP Destination Port	5060
Use "user=phone" in SIP URL	Yes
Use "user=phone" in From Header	No
Use Tel URI for Asserted Identity	Disable
Tel to IP No Answer Timeout	180
Enable Remote Party ID	Disable



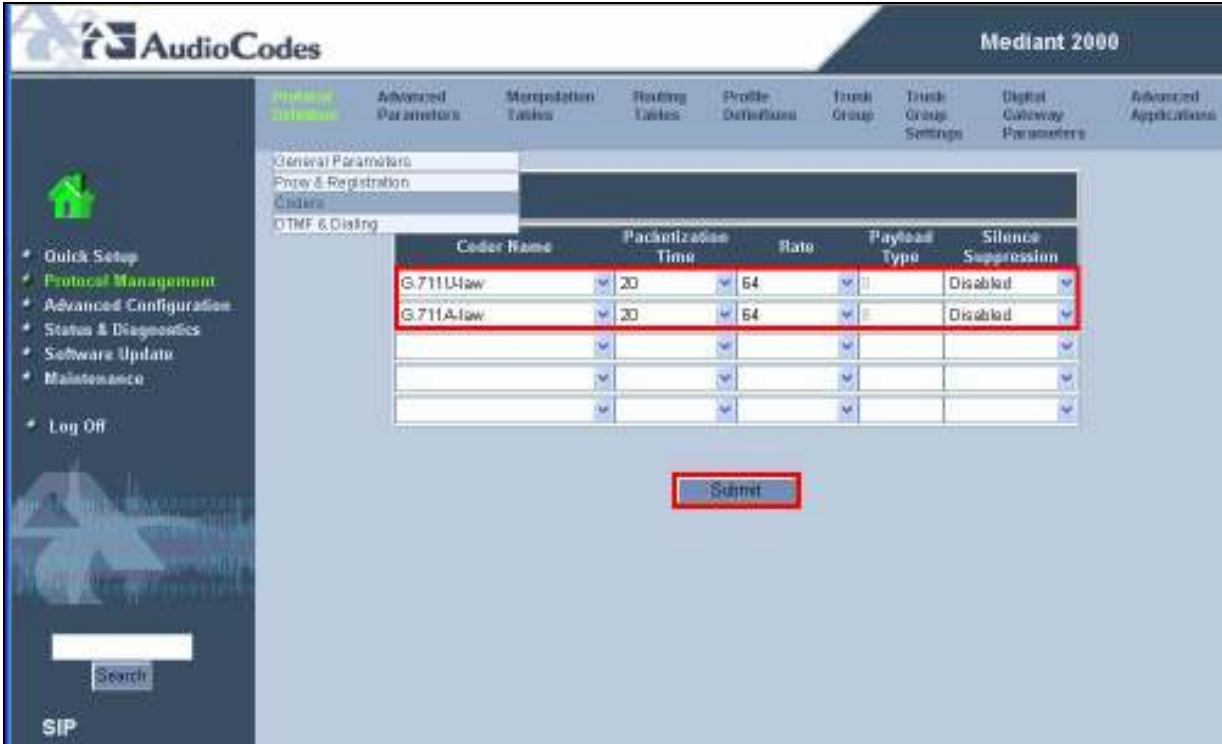
Step	Description																																														
5.2.3b	<p>Use default settings and click <b>Submit</b>.</p>  <p>The screenshot displays a configuration window with a list of SIP-related settings. Most settings are set to their default values, such as 'Yes' for 'Add Number Plan and Type to Remote Party ID Header', 'Disable' for 'Enable History-Info Header', and 'None' for 'Multiple Packetization Time Format'. The 'Submit' button at the bottom right is highlighted with a red rectangular box.</p> <table border="1"> <thead> <tr> <th>Setting</th> <th>Value</th> </tr> </thead> <tbody> <tr><td>Add Number Plan and Type to Remote Party ID Header</td><td>Yes</td></tr> <tr><td>Enable History-Info Header</td><td>Disable</td></tr> <tr><td>Use Source Number as Display Name</td><td>No</td></tr> <tr><td>Use Display Name as Source Number</td><td>No</td></tr> <tr><td>Play Ringback Tone to IP</td><td>Don't Play</td></tr> <tr><td>Play Ringback Tone to Tel</td><td>Play According to Early Me</td></tr> <tr><td>Use Ttp information</td><td>Disable</td></tr> <tr><td>Enable GRUU</td><td>Disable</td></tr> <tr><td>User-Agent Information</td><td></td></tr> <tr><td>SDP Session Owner</td><td>AudiocodesGW</td></tr> <tr><td>Play Busy Tone to Tel</td><td>Don't Play</td></tr> <tr><td>Subject</td><td></td></tr> <tr><td>Multiple Packetization Time Format</td><td>None</td></tr> <tr><td>Enable Reason Header</td><td>Enable</td></tr> <tr><td>Enable Semi-Attended Transfer</td><td>Disable</td></tr> <tr><td>3xx Behavior</td><td>Forward</td></tr> <tr><td>Enable P-Charging Vector</td><td>Disable</td></tr> <tr><td>Enable VoiceMail URI</td><td>Disable</td></tr> <tr><td colspan="2"><b>Retransmission Parameters</b></td></tr> <tr><td>SIP T1 Retransmission Timer (msec)</td><td>500</td></tr> <tr><td>SIP T2 Retransmission Timer (msec)</td><td>4000</td></tr> <tr><td>SIP Maximum RTX</td><td>7</td></tr> </tbody> </table>	Setting	Value	Add Number Plan and Type to Remote Party ID Header	Yes	Enable History-Info Header	Disable	Use Source Number as Display Name	No	Use Display Name as Source Number	No	Play Ringback Tone to IP	Don't Play	Play Ringback Tone to Tel	Play According to Early Me	Use Ttp information	Disable	Enable GRUU	Disable	User-Agent Information		SDP Session Owner	AudiocodesGW	Play Busy Tone to Tel	Don't Play	Subject		Multiple Packetization Time Format	None	Enable Reason Header	Enable	Enable Semi-Attended Transfer	Disable	3xx Behavior	Forward	Enable P-Charging Vector	Disable	Enable VoiceMail URI	Disable	<b>Retransmission Parameters</b>		SIP T1 Retransmission Timer (msec)	500	SIP T2 Retransmission Timer (msec)	4000	SIP Maximum RTX	7
Setting	Value																																														
Add Number Plan and Type to Remote Party ID Header	Yes																																														
Enable History-Info Header	Disable																																														
Use Source Number as Display Name	No																																														
Use Display Name as Source Number	No																																														
Play Ringback Tone to IP	Don't Play																																														
Play Ringback Tone to Tel	Play According to Early Me																																														
Use Ttp information	Disable																																														
Enable GRUU	Disable																																														
User-Agent Information																																															
SDP Session Owner	AudiocodesGW																																														
Play Busy Tone to Tel	Don't Play																																														
Subject																																															
Multiple Packetization Time Format	None																																														
Enable Reason Header	Enable																																														
Enable Semi-Attended Transfer	Disable																																														
3xx Behavior	Forward																																														
Enable P-Charging Vector	Disable																																														
Enable VoiceMail URI	Disable																																														
<b>Retransmission Parameters</b>																																															
SIP T1 Retransmission Timer (msec)	500																																														
SIP T2 Retransmission Timer (msec)	4000																																														
SIP Maximum RTX	7																																														

Step	Description
5.2.4	<p>Administer settings for <b>Proxy &amp; Registration</b> as follows:</p> <ul style="list-style-type: none"> <li>From the embedded web server, select <b>Protocol Management → Protocol Definition → Proxy &amp; Registration</b>.</li> <li>Select <b>Don't Use Proxy</b> from the drop-down list for the <b>Enable Proxy</b> field.  <i>Note: SIP connectivity between the AudioCodes Mediant 2000 and Avaya Meeting Exchange is direct.</i></li> <li>Use default settings for remaining fields.</li> <li>Click <b>Submit</b>.</li> </ul>

The screenshot displays the AudioCodes Mediant 2000 web interface. The top navigation bar includes links for General Parameters, Advanced Parameters, Manipulation Tables, Floating Tables, Profile Definitions, Trunk Group, Trunk Group Settings, Digital Gateway Parameters, and Advanced Applications. The left sidebar contains a home icon and links for Quick Setup, Protocol Management (highlighted), Advanced Configuration, Status & Diagnostics, Software Update, Maintenance, and Log Off. The main content area is titled 'Registration' and contains a form with the following fields:

Field	Value
Enable Proxy	Don't Use Proxy
Enable Registration	Disable
Gateway Name	
Gateway Registration Name	
DNS Query Type	A-Record
Proxy DNS Query Type	A-Record
Use Gateway Name for OPTIONS	No
Number of RTX Before Hot-Swap	3
User Name	
Password	*****
Nonce	0a123bcd
Authentication Mode	Per Gateway
Challenge Caching Mode	None
Mutual Authentication Mode	Optional

At the bottom of the form, there are buttons for 'Register', 'Un-Register', and 'Submit' (highlighted with a red box).

Step	Description
5.2.5	<p>Administer codec preferences and attributes for the SIP trunk between the AudioCodes Mediant 2000 and Avaya Meeting Exchange as follows:</p> <ul style="list-style-type: none"> <li>From the embedded web server, select <b>Protocol Management → Protocol Definition → Coders</b>.</li> <li>Add entries for codecs that are supported on the Convedia CMS-6000 Media Server (see <b>Step 3.16</b>) as displayed. <ul style="list-style-type: none"> <li>Select a codec from the drop-down list for the <b>Coder Name</b> field that is compatible with the codecs supported on Avaya Meeting Exchange.</li> <li>Use default settings for remaining fields.</li> </ul> </li> <li>Click <b>Submit</b>.</li> </ul> <p><i><b>Note:</b> 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.</i></p> 

### 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	<p>Administer call routing rules that are applied to calls originating from the PSTN to Avaya Meeting Exchange by adding <b>Tel to IP Group Routing</b> rules as follows:</p> <ul style="list-style-type: none"> <li>From the embedded web server, select <b>Protocol Management → Routing Tables → Tel to IP Group Routing</b>.</li> <li>Select <b>Route calls before manipulation</b> from the drop-down list for the <b>Tel to IP Routing Mode</b> field. Add an entry to enable call origination from the PSTN to Avaya Meeting Exchange as displayed. <ul style="list-style-type: none"> <li>Enter a rule in the <b>Dest. Phone Prefix</b> 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.</li> <li>Enter a rule in the <b>Source Phone Prefix</b> field to match the calling party number for calls from the PSTN.</li> <li>Enter the IP address of Avaya Meeting Exchange in the <b>Dest. IP Address</b> field.</li> </ul> </li> <li>Click <b>Submit</b>.</li> </ul>

Step	Description
5.3.2	<p>Administer call routing rules that are applied to calls originating from Avaya Meeting Exchange to the PSTN by adding <b>IP to Hunt Group Routing</b> rules as follows:</p> <ul style="list-style-type: none"> <li>From the embedded web server, select <b>Protocol Management → Routing Tables → IP to Hunt Group Routing</b>.</li> <li>Select <b>Route calls before manipulation</b> from the drop-down list for the <b>IP To Tel Routing Mode</b> 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. <ul style="list-style-type: none"> <li>Enter a rule in the <b>Dest. Phone Prefix</b> 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.</li> <li>Enter a rule in the <b>Source Phone Prefix</b> field to match the calling party number for calls from the Avaya Meeting Exchange.</li> <li>Enter a rule in the <b>Source IP Address</b> field to match the IP address of Avaya Meeting Exchange.</li> <li>Enter the Trunk Group ID for the T1 ISDN-PRI trunk group provisioned in <b>Section 5.1</b> in the <b>Trunk Group ID</b> field.</li> </ul> </li> <li>Click <b>Submit</b>.</li> </ul>

The screenshot shows the AudioCodes Mediant 2000 web interface. The main navigation bar includes links for Protocol Definition, Advanced Parameters, Maintenance Tables, Routing Tables (highlighted), Profile Definition, Trunk Group, Trunk Group Settings, Digital Gateway Parameters, and Advanced Applications. A left sidebar contains a home icon and links for Quick Setup, Protocol Management (highlighted), Advanced Configuration, Status & Diagnostics, Software Update, Maintenance, and Log Off. The main content area is titled 'IP to Trunk Group Routing Table'. It features a 'Routing Index' dropdown set to '1-12' and a 'IP To Tel Routing Mode' dropdown set to 'Route calls before manipulation'. Below this is a table with columns: Dest. Phone Prefix, Source Phone Prefix, Source IP Address, Trunk Group ID, and Profile ID. The first row (index 1) is highlighted with a red border and contains the values: \*, \*, \*, 1, and 0. The table has 12 rows in total. At the bottom right of the table area is a 'Submit' button.

	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	Trunk Group ID	Profile ID
1	*	*	*	1	0
2					
3					
4					
5					
6					
7					
8					
9					
10					
11					
12					

## 6. Verification Steps

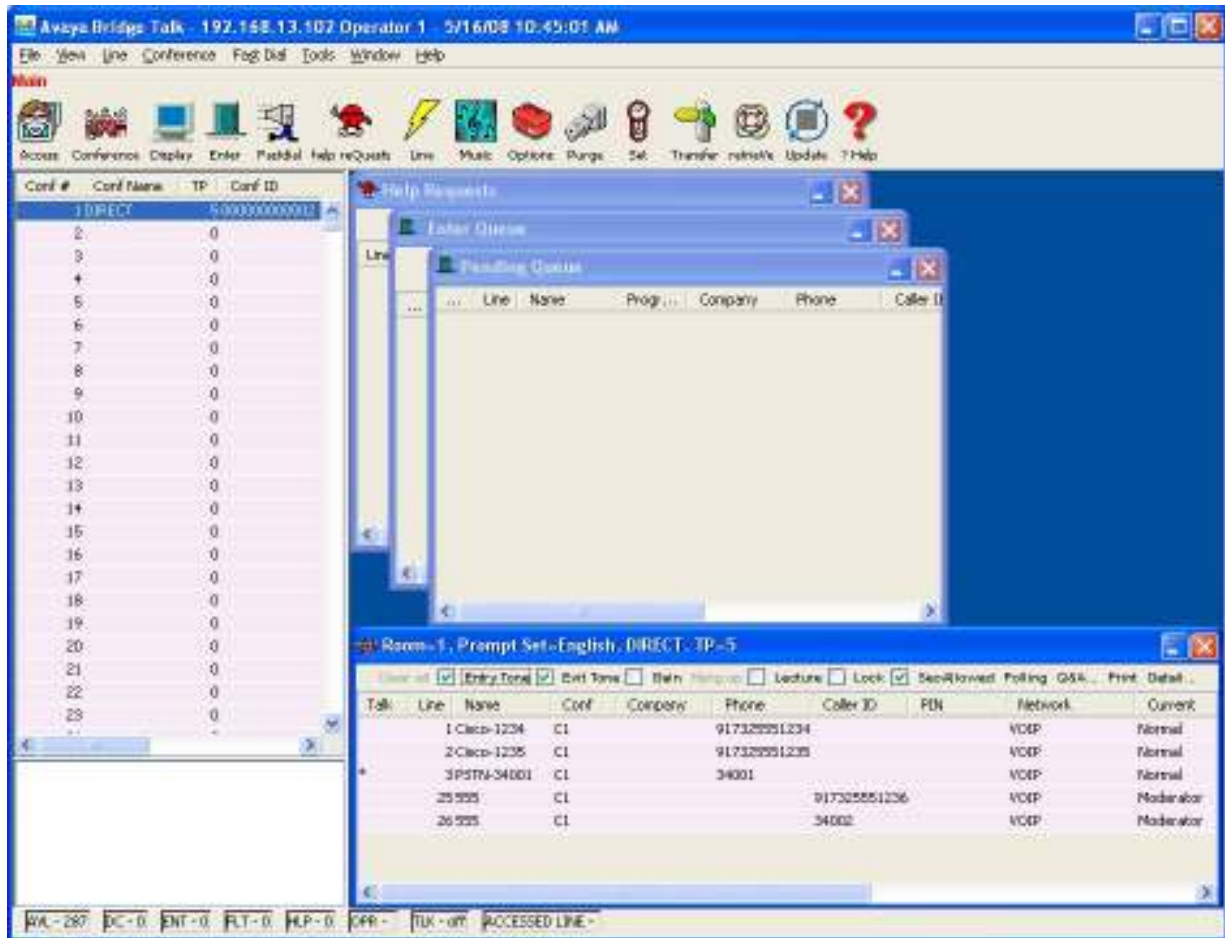
The following steps were used to verify the administrative steps presented in these Application Notes and are applicable for similar configurations in the field.

Step	Description
6.1	<p>Verify conferencing related processes are running on Avaya Meeting Exchange. From the Avaya Meeting Exchange CLI, enter “<b>service mx-bridge status</b>” at the command prompt and verify that a Process ID (PID) is present for all processes.</p> <pre> S6200-&gt; 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 sglxecd with 5 children           </pre>
6.2	<p>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 <b>Section 4.1</b> 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 <b>555</b> to enter the conference provisioned in <b>Section 3.4</b> as moderator via the call branding for a direct call flow provisioned in <b>Step 3.3.1</b>.</p>
6.3	<p>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 <b>Section 5.1</b> and <b>Section 5.2</b> 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 <b>555</b> to enter the conference provisioned in <b>Section 3.4</b> as moderator via the call branding for a direct call flow provisioned in <b>Step 3.3.1</b>.</p>

Step	Description
6.4	<p>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 <b>Section 4.1</b>, <b>Section 5.1</b> and <b>Section 5.2</b> 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 <b>Step 6.2</b>, or <b>Step 6.3</b>), enter the appropriate touchtone command to initiate the blast dial feature as provisioned in <b>Section 3.4</b>. 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.</p>



Step	Description
6.5	<p>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.</p> <ul style="list-style-type: none"><li>• If not already logged on, log in to the Avaya Bridge Talk application with the appropriate credentials.</li><li>• From the Conference Navigator, double-click the appropriate entry to open the corresponding Conference Room.</li><li>• Verify conference participants are added/removed from conferences by observing the Conference Navigator and/or Conference Room window.</li></ul> <p><i>Note: The screen capture below displays the conference that was initiated in Step 6.2, Step 6.3 and Step 6.4.</i></p>

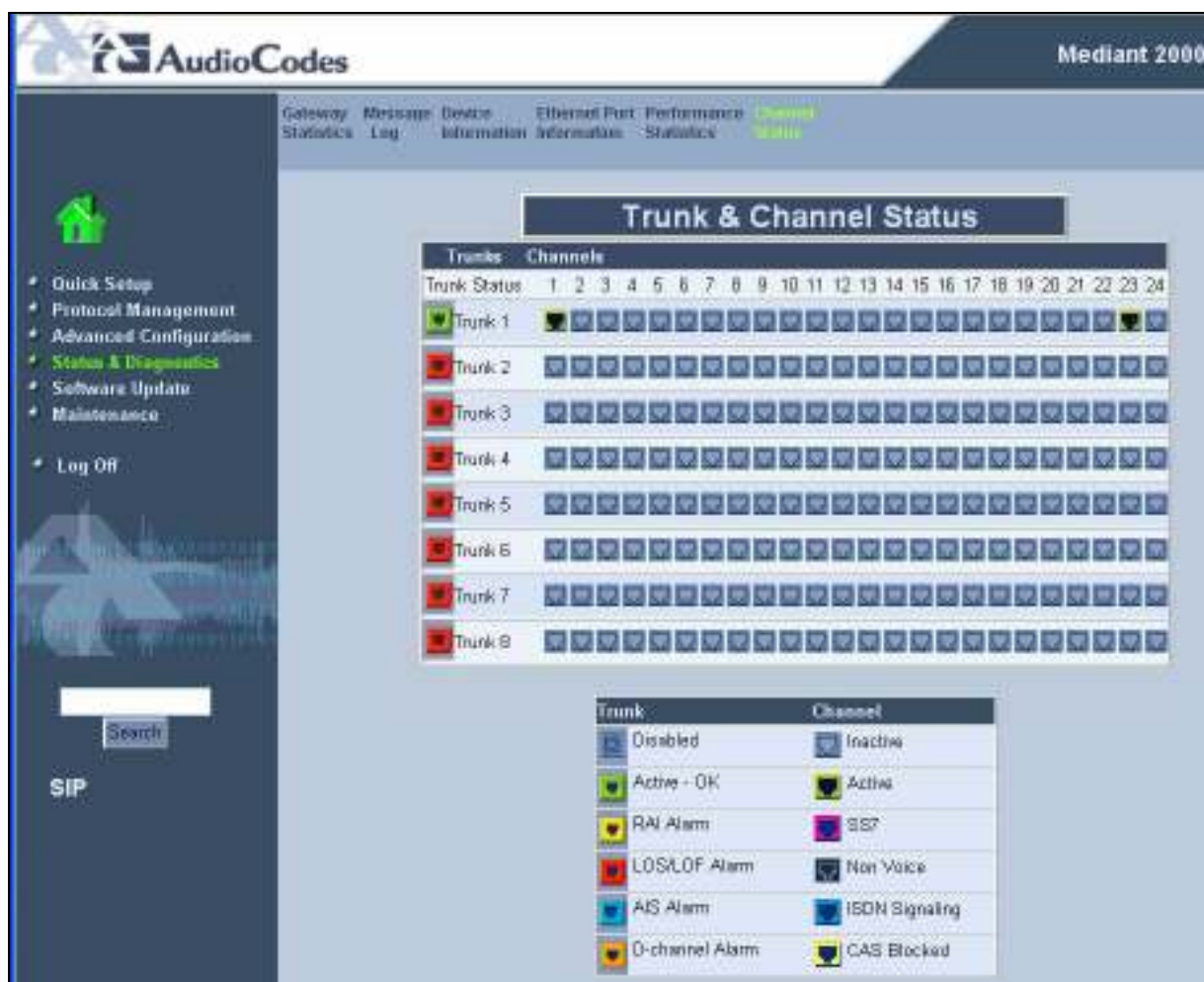


The screenshot displays the Avaya Bridge Talk application interface. The main window is titled "Avaya Bridge Talk - 192.168.13.107 Operator 1 - 5/16/08 10:45:01 AM". It features a menu bar (File, View, Line, Conference, Fag Dial, Tools, Window, Help) and a toolbar with various icons. The left pane shows a list of conference rooms with columns for Conf #, Conf Name, TP, and Conf ID. The right pane shows a detailed view of a selected conference room, titled "Room-1, Prompt Set-English, DIRECT-TP-5". This view includes a table of participants and their roles.

Talk	Line	Name	Conf	Company	Phone	Caller ID	PIN	Network	Current
1	Clacp-1234	C1			917329951234			VOIP	Normal
2	Clacp-1235	C1			917329951235			VOIP	Normal
3	SPSTW-34001	C1			34001			VOIP	Normal
25	5555	C1				917325551236		VOIP	Moderator
26	5555	C1				34002		VOIP	Moderator



Step	Description
6.6	<p>Verify <b>Channel Status</b> on the AudioCodes Mediant 2000 as follows:</p> <ul style="list-style-type: none"> <li>From the embedded web server, select <b>Status &amp; Diagnostics</b> → <b>Channel Status</b>.  <i>Note: The <b>Trunk &amp; Channel Status</b> displays <b>Active - OK</b> for T1 ISDN-PRI trunk group provisioned in <b>Section 5.1</b>.</i></li> <li>This screen capture displays the channel selection pattern for the <b>Active</b> channels that are associated with the conference that was initiated in <b>Step 6.3</b> and <b>Step 6.4</b>.  <i>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 <b>Channel 23 on Trunk 1</b> is selected by the PSTN for call origination to Avaya Meeting Exchange. Conversely, <b>Channel 1 on Trunk 1</b> is selected by the AudioCodes Mediant 2000 for call origination to the PSTN.</i></li> </ul>



Step	Description
6.7	<p>Below is a SIP call flow of the scenario that was initiated in <b>Step 6.2</b>, <b>Step 6.3</b> and <b>Step 6.4</b>. This trace is intended to display the provisioning presented in these Application Notes.</p> <ul style="list-style-type: none"><li>• Cisco UCM (<b>60.1.1.9</b>) sends a SIP INVITE message to Avaya Meeting Exchange (<b>192.168.13.102</b>). Avaya Meeting Exchange extracts the DID (<b>555</b>) using the provisioning in <b>Step 3.2.1</b> and places the call in conference using the call branding provisioned in <b>Section 3.3</b>.</li><li>• The AudioCodes Mediant 2000 (<b>192.168.11.111</b>) sends a SIP INVITE message to Avaya Meeting Exchange (<b>192.168.13.102</b>). Avaya Meeting Exchange extracts the DID (<b>555</b>) using the provisioning in <b>Step 3.2.1</b> and places the call in conference using the call branding provisioned in <b>Section 3.3</b>.</li><li>• Avaya Meeting Exchange sends SIP INVITE messages to both Cisco UCM and the AudioCodes Mediant 2000 using the provisioning in <b>Step 3.2.2</b>.</li></ul>

Time	60.1.1.9	192.168.13.102	192.168.11.111	Comment
11.353	(37831) → SIP → (5060)			Request: INVITE sip:555@192.168.13.102:5060
11.353	(37831) ← SIP ← (5060)			Status: 100 Trying
11.354	(37831) → SIP/SDP → (5060)			Status: 200 OK, with session description
11.361	(37831) → SIP/SDP → (5060)			Request: ACK sip:\$6200@192.168.13.102:5060;transport=top, with session description
21.570		(5060) ← SIP/SDP ← (53601)		Request: INVITE sip:555@192.168.13.102:user=phone, with session description
21.571		(5060) ← SIP ← (53601)		Status: 100 Trying
21.572		(5060) → SIP/SDP → (53601)		Status: 200 OK, with session description
21.593		(5060) ← SIP ← (53601)		Request: ACK sip:\$6200@192.168.13.102:5060;transport=top
31.273	(5060) → SIP/SDP → (43645)			Request: INVITE sip:917325551234@60.1.1.9:5060;transport=top, with session description
31.275	(5060) → SIP/SDP → (43645)			Request: INVITE sip:917325551235@60.1.1.9:5060;transport=top, with session description
31.275	(5060) → SIP → (43645)			Status: 100 Trying
31.277	(5060) → SIP → (43645)			Status: 100 Trying
31.277		(43645) → SIP/SDP → (5060)		Request: INVITE sip:34001@192.168.11.111:5060;transport=top, with session description
31.293		(43645) ← SIP ← (5060)		Status: 100 Trying
31.339		(43645) ← SIP ← (5060)		Status: 180 Ringing
31.407	(5060) → SIP → (43645)			Status: 180 Ringing
31.882	(5060) → SIP → (43645)			Status: 180 Ringing
33.164		(43645) ← SIP/SDP ← (5060)		Status: 200 OK, with session description
33.164		(43645) → SIP → (5060)		Request: ACK sip:1000@192.168.11.111;transport=top
37.512	(5060) → SIP/SDP → (43645)			Status: 183 Session Progress, with session description
37.513	(5060) → SIP/SDP → (43645)			Status: 200 OK, with session description
37.514	(5060) → SIP → (43645)			Request: ACK sip:917325551235@60.1.1.9:5060;transport=top
38.233	(5060) → SIP/SDP → (43645)			Status: 183 Session Progress, with session description
38.233	(5060) → SIP/SDP → (43645)			Status: 200 OK, with session description
38.234	(5060) ← SIP ← (43645)			Request: ACK sip:917325551234@60.1.1.9:5060;transport=top

## 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.

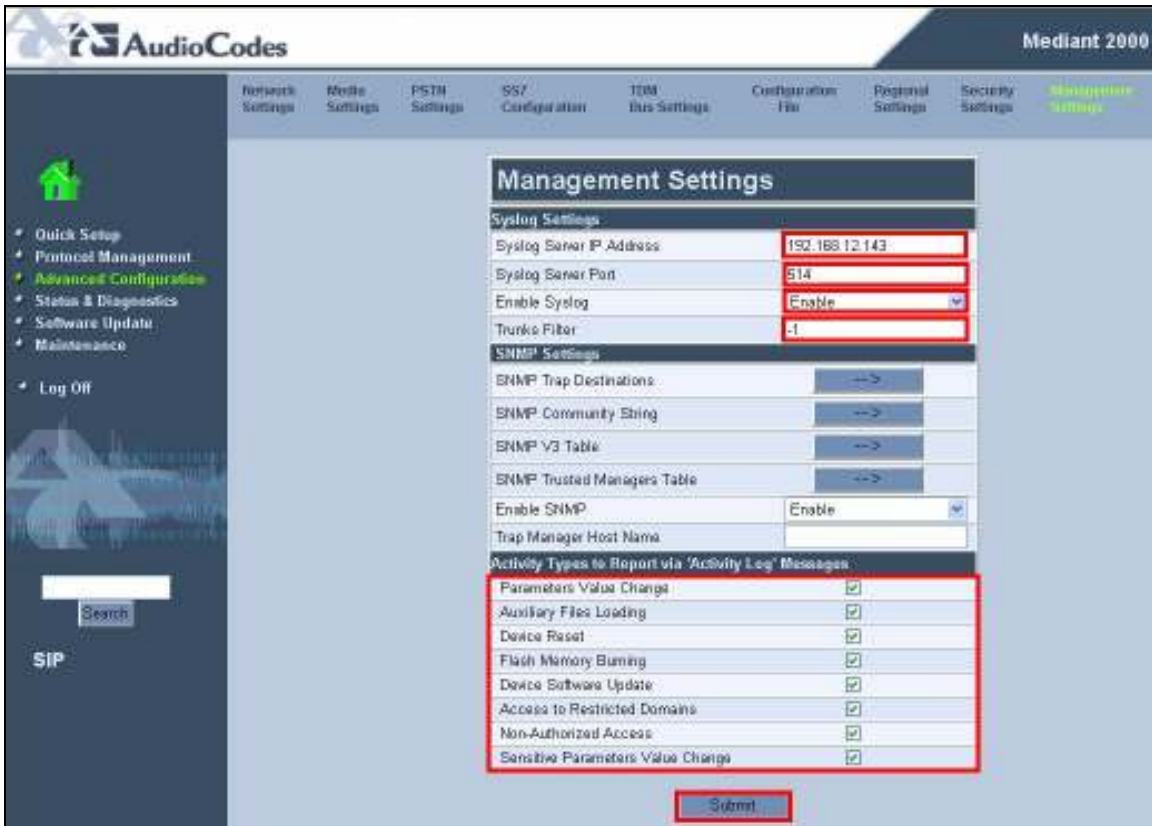
AudioCodes references are available at <http://www.audiocodes.com>.

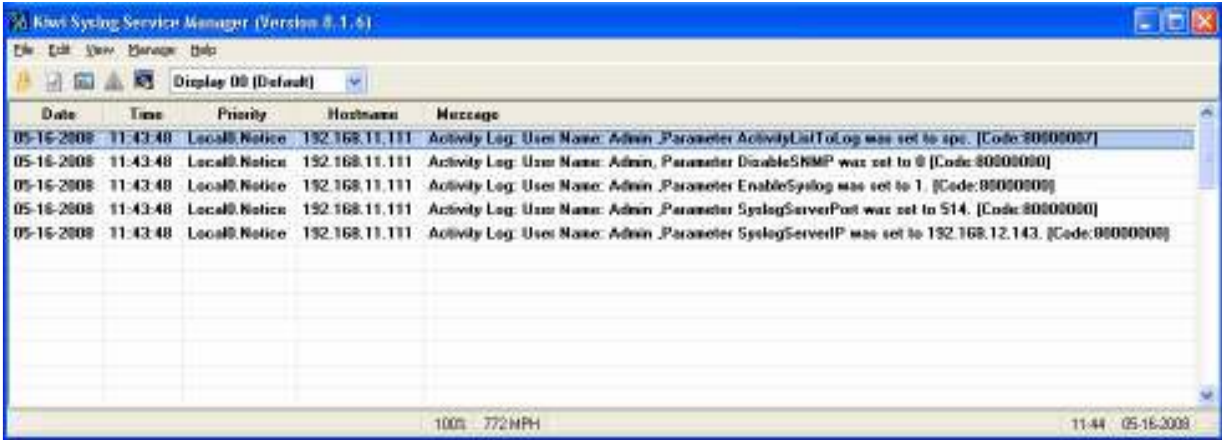
- [5] Mediant™ 2000 User's Manual version 5.2, Document # LTRT-69806, August 2007.
- [6] Mediant™ 2000 & TP-1610 & TP-260-UNI SIP User's Manual Ver 5.2, Document #: LTRT-68806, September 2007.

## 9. Appendix


### Appendix A - AudioCodes Mediant 2000 Syslog Configuration


Step	Description
A1	<p>Enable syslog functionality on the AudioCodes Mediant 2000 as follows:</p> <ul style="list-style-type: none"><li>From the embedded web server, select <b>Advanced Configuration</b> → <b>Management Settings</b>.</li><li>Set the <b>Syslog Server IP Address</b> and <b>Syslog Server Port</b> fields appropriately.</li><li>Select <b>Enable</b> from the drop-down list for the <b>Enable Syslog</b> field.</li><li>Set the <b>Trunks Filter</b> field to enable syslog reporting for the appropriate trunk(s). For this sample configuration, <b>-1</b> is used, which sets no filter, thus allowing reporting for all trunks to appear in the syslog.</li><li>Enable logging by selecting the appropriate fields under the <b>Activity Types to Report</b> via <b>'Activity Log' Messages</b> heading. If enabled, any action associated with the enabled field will be logged (see <b>Step A2</b> for a sample syslog regarding 'Activity Types').</li><li>Click <b>Submit</b>.</li></ul>



Step	Description
A2	<p>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 <b>Step A1</b>.</p> 



Step	Description
A3	<p>Configure the log level to capture messages for debug traces as follows:</p> <ul style="list-style-type: none"> <li>From the embedded web server, select <b>Protocol Management</b> → <b>Advanced Parameters</b> → <b>General Parameters</b>.</li> <li>Select the appropriate setting from the drop-down list for the <b>Debug Level</b> field. For this sample configuration, the <b>Debug Level</b> field is set to <b>4</b> (see <b>Step A4</b> and <b>Step A5</b> for sample logging regarding 'debug traces').</li> <li>[<i>Button Not Shown</i>] Click <b>Submit</b>.</li> </ul> <p><i>Note: It is recommended to provision the <b>Debug Level</b> field to the highest setting for debug traces. Setting the <b>Debug Level</b> field to 0 turns off logging for debug traces and will not affect the syslog of 'Activity Type' messages, as configured in <b>Step A1</b>.</i></p>  <p>The screenshot shows the AudioCodes Mediant 2000 web interface. The 'General Parameters' tab is selected. The 'Debug Level' field is set to 4, which is highlighted with a red box. Other fields include IP Security (Disable), Filter Calls to IP (Don't Filter), I Enable Digit Delivery to Tel (Disable), I Enable Digit Delivery to IP (Disable), RTP Only Mode (Disable), PSTN Alert Timeout (180), Disconnect on Broken Connection (Yes), Broken Connection Timeout [100 msec] (100), Disconnect Call on Silence Detection (No), Silence Detection Period [sec] (120), Silence Detection Method (Packets Count), Enable Fax Re-Routing (Disable), and CDR and Debug section with CDR Server IP Address, CDR Report Level (None), and Debug Level (4).</p>

Step	Description
A4	<p>With logging set at the appropriate level, messages will get logged to the embedded web server on the AudioCodes Mediant 2000.</p> <p><i>Note: This display is from the call originating from the PSTN to Avaya Meeting Exchange via the AudioCodes Mediant 2000 (see Step 6.3).</i></p> 





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