Avaya Solution & Interoperability Test Lab

Configuring the AudioCodes Mediant 5000 Media Gateway to Provide Connectivity between the PSTN and the Avaya Meeting Exchange S6800 Conferencing Server - Issue 1.0

Abstract

These Application Notes describe a compliance tested solution comprised of the Avaya Meeting Exchange S6800 Conferencing Server and the AudioCodes Mediant 5000 Media Gateway. The AudioCodes Mediant 5000 Media Gateway is utilized to enable connectivity between the Avaya Meeting Exchange S6800 Conferencing Server and the PSTN. This configuration provides a rich set of conferencing options available on the Avaya Meeting Exchange S6800 Conferencing Server to participants associated with the PSTN.

Information in these Application Notes has been obtained through Developer*Connection* compliance testing and additional technical discussions. Testing was conducted via the Developer*Connection* Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe a compliance tested solution comprised of the Avaya Meeting Exchange S6800 Conferencing Server and the AudioCodes Mediant 5000 Media Gateway. The AudioCodes Mediant 5000 Media Gateway is utilized to enable connectivity between the Avaya Meeting Exchange S6800 Conferencing Server and the PSTN. The end to end signaling connectivity between the Avaya Meeting Exchange S6800 Conferencing Server and the PSTN was as follows: SIP/UDP between Avaya Meeting Exchange and the AudioCodes Mediant 5000 Media Gateway and either T1 ISDN-PRI/DS0/DS1/DS3 or T1 CAS/DS0/DS1/DS3 (Channel Associated Signaling) between the AudioCodes Mediant 5000 Media Gateway and the PSTN. This configuration provides a rich set of conferencing options available on the Avaya Meeting Exchange S6800 Conferencing Server to participants associated with the PSTN.

Figure 1 illustrates the network configuration utilized for this compliance tested solution.

Signaling connectivity between the PSTN and the Avaya Meeting Exchange S6800 Conferencing Server traversed the following Path.

- T1 CAS (robbed-bit, e.g., 8k "robbed" from each of the 24 channels comprising the T1 signal) multiplexed over a DS3 from the PSTN to the AudioCodes Mediant 5000 Media Gateway (green dashed/dotted line).
- T1 ISDN-PRI (D-channel on channel 24) multiplexed over a DS3 from the PSTN to the AudioCodes Mediant 5000 Media Gateway (green dashed/dotted line).
- SIP/UDP between the AudioCodes Mediant 5000 Media Gateway and the Avaya Meeting Exchange S6200 Application Server (blue dashed line).

Media connectivity between the PSTN and the Avaya Meeting Exchange S6800 Conferencing Server traversed the following Path.

- T1 CAS (24 56k channels) multiplexed over a DS3 from the PSTN to the AudioCodes Mediant 5000 Media Gateway (green dashed/dotted line).
- T1 ISDN-PRI (23 B-channels) multiplexed over a DS3 from the PSTN to the AudioCodes Mediant 5000 Media Gateway (green dashed/dotted line).
- RTP/UDP between the AudioCodes Mediant 5000 Media Gateway and the Convedia CMS-6000 Media Server (red dotted line).

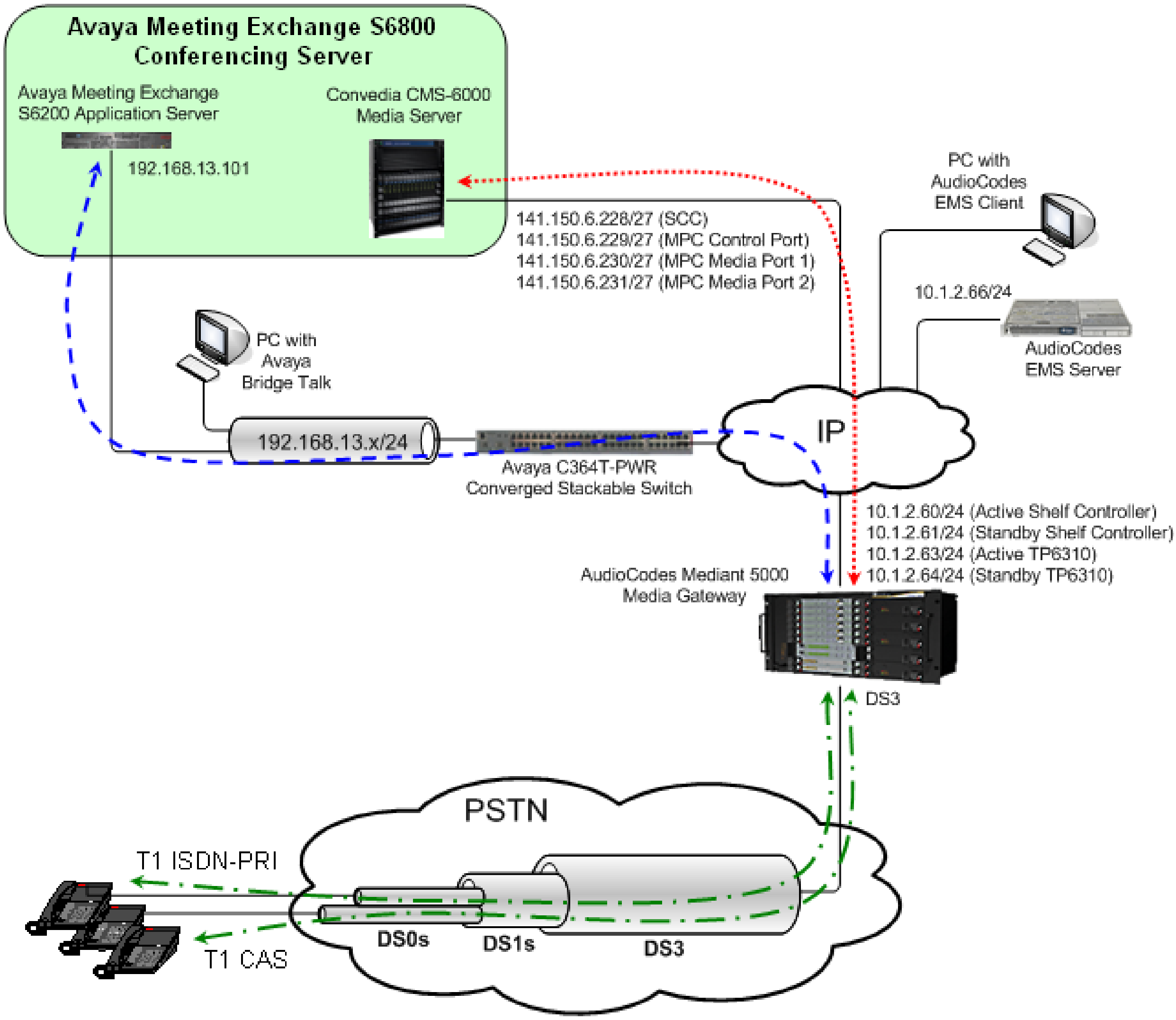


Figure 1: Network Configuration

1.1. Avaya Meeting Exchange S6800 Conferencing Server

The Avaya Meeting Exchange S6800 Conferencing Server is a SIP-based voice conferencing solution that extends Avaya's conferencing applications including reservation-less, attended, event, mobile to support various IP network implementations. The following capabilities are supported by the Avaya Meeting Exchange S6800 Conferencing Server:

- RFC 2833 DTMF support.
- In-band DTMF support.
- Up to 2016-user and 115-operator conferences.
- Support for up to four digitally recorded music sources.
- Support for one recorded music channel and up to four connection based (FDAPI) music channels.
- Any combination of G.711 a-law or u-law, G.729, G723, G726-16, G726-24, G726-32, or G726-40 codecs.

Figure 2 illustrates the configuration for the Avaya Meeting Exchange S6800 Conferencing Server, which is composed of the following:

- Up to four Avaya Meeting Exchange S6200 server(s) configured as Application Server(s), e.g., call signaling processes are managed by the S6200(s). For these Application Notes, one Avaya Meeting Exchange S6200 server is utilized as an Application Server.
- A Convedia CMS-6000 Media Server, containing the following cards:
    - One Media Processor Card (MPC).
    - One Shelf Control Card (SCC).
- Signaling between the Avaya Meeting Exchange Application Server(s) and the Convedia CMS-6000 Media Server is SIP.

Figure 2: Avaya Meeting Exchange S6800 Conferencing Server

1.2. AudioCodes Mediant 5000 Media Gateway

The AudioCodes Mediant 5000 Media Gateway provides a means for customers to consolidate facilities and reduce communications costs by concentrating PSTN traffic over DS3 facilities. For high call traffic applications such as conferencing servers, using DS3 facilities can provide a higher density, lower cost solution compared with DS1 facilities. The AudioCodes Mediant 5000 Media Gateway is a carrier class product that supports up to 8000 channels of SIP VoIP telephony. It uses N+1 redundancy of media gateway, Ethernet switch, shelf controller, and power supply modules to achieve high availability in mission critical applications.

The AudioCodes Mediant 5000 Media Gateway is shipped with an Element Management System (EMS) that is used for operations, administration, management, and provisioning functions. A Solaris based EMS server communicates with the AudioCodes Mediant 5000 Media Gateway using SNMP. An EMS client communicates with the EMS server from a Microsoft Windows based PC, and provides the graphical user interface.

2. Equipment and Software Validated

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

Equipment	Software
Avaya Meeting Exchange S6800 Conferencing Server • Avaya Meeting Exchange S6200 Application Server o Software version o IPCB build version • Convedia CMS-6000 Media Server o SCC2 (slot 1) o MPC2 (slot 2)	 40102h mx7_1.3.00-84 4.8.0.16 4.8.0.16
Avaya Bridge Talk	4.1.01b
Avaya C364T-PWR Converged Stackable Switch	4.5.14
AudioCodes Mediant 5000 Media Gateway • Chassis Type o Software Version • Board Type o TP Software Version o Flash Version o Firmware Version o Module Firmware Version	 M5k10 3.2.77 Tp6310Ds3 4.80.036.002 212 2 528
AudiCodes EMS Server	3.2.110
AudiCodes EMS Client	3.2.110

Table 1: Hardware and Software Versions

3. Configure the Avaya Meeting Exchange S6800 Conferencing Server

This section describes the steps for configuring the Avaya Meeting Exchange S6800 Conferencing Server to interoperate with the PSTN via the AudioCodes Mediant 5000 Media Gateway (see **Section 1**, **Figure 1**).

3.1. Configure the Avaya Meeting Exchange S6200 Application Server

The following steps describe the administrative procedures for configuring the Avaya Meeting Exchange S6200 Application Server to originate/terminate calls utilizing the Convedia CMS-6000 Media Server.

Step	Description
3.1	Log in to the Avaya Meeting Exchange S6200 Application Server console to access the Command Line Interface (CLI) with the appropriate credentials.

Step	Description

3.2	Configure settings that enable SIP connectivity between the Avaya Meeting Exchange S6200 Application Server and other SIP User Agent(s) by editing the **system.cfg** file as follows:
	• cd to **/usr/ipcb/config**
	• Edit the **system.cfg** file with a text editor, e.g., vi.
	• Add a line to identify the IP address of the Avaya Meeting Exchange S6200 Application Server (as defined in the /etc/hosts file):
	  o **IPAddress=192.168.13.101**
	• Add a line to populate the From Header Field in SIP INVITE messages from the Avaya Meeting Exchange S6200 Application Server:
	  o **MyListener=sip:001s6800@192.168.13.101**
	    Note: *The user field **001s6800**, defined for this SIP URI must conform to the RFC 3261. For consistency, it is selected to match the user field provisioned for the **respContact** entry (**see below**).*
	• Add a line to provide SIP User Agent(s) a Contact address to use for Acknowledging SIP messages from the Avaya Meeting Exchange S6200 Application Server:
	  o **respContact=<sip:001s6800@192.168.13.101:5060;transport=udp>**
	    Note: *The user field **001s6800**, defined for this SIP URI must conform to the RFC 3261 and is selected to uniquely identify this server. E.g., the user field **001s6800** will be inserted in the From header field of SIP INVITE messages from this Avaya Meeting Exchange S6200 Application Server. The intention is for **001s6800** to display on a participant's User Agent Client (UAC) when Dial-Out procedures from the Avaya Meeting Exchange S6200 Application Server are invoked. This allows end-user's to identify a call from this server.*
	• Add the following lines to set the Min-SE timer to **1800** seconds in SIP INVITE messages from the Avaya Meeting Exchange S6200 Application Server:
	  o **sessionRefreshTimerValue= 1800**
	  o **minSETimerValue= 1800**
	Note: *The values for the **sessionRefreshTimerValue** and the **minSETimerValue** are defined in seconds and should be provisioned to be greater than or equal to the value used by SIP User Agent(s) connecting to the Avaya Meeting Exchange S6200 Application Server, e.g., the AudioCodes Mediant 5000 Media Gateway. This setting is necessary to enable Dial-Out from the Avaya Meeting Exchange S6200 Application Server to the PSTN via the AudioCodes Mediant 5000 Media Gateway.*

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.3</td><td>To associate incoming calls to the Avaya Meeting Exchange S6200 Application Server with different call flows, edit the UriToTelnum.tab file to extract both Automatic Number Identification (ANI) and Direct Inward Dial (DID, also known as DDI in Europe) values as follows:

cd to /usr/ipcb/config
Edit the UriToTelnum.tab file with a text editor, e.g., vi.
Add a line to match the pattern of the To header field in SIP INVITE messages from the AudioCodes Mediant 5000 Media Gateway to the Avaya Meeting Exchange S6200 Application Server. If a match occurs, the DID is extracted from the To header field and the ANI is extracted from the From header field:

"*<sip:*@*" $2
Where the pattern "*<sip:*@*" matches:

To: <sip:777@192.168.13.101;user=phone> and $2 utilizes 777 (the variable contained in the second *) as the DID value for the call.
From: <sip:7325550501@10.1.2.63> and $2 utilizes 7325550501 (the variable contained in the second *) as the ANI for the call (see Step 6.9).

Enable an undefined caller to receive a prompt for operator assistance by administering for the condition of an unmatched SIP INVITE message by adding a wildcard entry as the last line in this file:

* $0
Note: Entries in this file are read sequentially, therefore, the line * $0 must be the last line in the file. Otherwise, all calls to the Avaya Meeting Exchange S6200 Application Server would match the wildcard and thus receive a prompt for operator assistance.

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

Step	Description
3.4	To enable Dial-Out from the Avaya Meeting Exchange S6200 Application Server to the PSTN via the AudioCodes Mediant 5000 Media Gateway, edit the **telnumToUri.tab** file as follows: • cd to **/usr/ipcb/config** • Edit the **telnumToUri.tab** file with a text editor, e.g., vi. • Add a line to the file to route outbound calls from the Avaya Meeting Exchange S6200 Application Server to the AudioCodes Mediant 5000 Media Gateway: o **50??? sip:$0@10.1.2.63:5060;transport=udp** Where the pattern **50???** matches any five digit number with a leading "**50**" and routes the call to the AudioCodes Mediant 5000 Media Gateway (**10.1.2.63**) via SIP/UDP. To enable SIP connectivity utilizing UDP, the entry contains: **5060** and **transport=udp**. The Avaya Meeting Exchange S6200 Application Server will substitute **$0** with the dialed number in outgoing SIP INVITE messages, e.g., if **50502** is dialed, the Avaya Meeting Exchange S6200 Application Server will send a SIP INVITE message with: **sip:50502@10.1.2.63:5060;transport=udp** in the SIP URI and To header field. ***Note**: Alternatively, routing to the AudioCodes Mediant 5000 Media Gateway could have been enabled with a wildcard entry:* • ***sip:$0@10.1.2.63:5060;transport=udp*** *Where ***** routes any dialed digits to the AudioCodes Mediant 5000 Media Gateway (**10.1.2.63**) via SIP/UDP*.

Step	Description
3.5	To configure the Avaya Meeting Exchange S6200 Application Server to utilize MPC resources on the Convedia CMS-6000 Media Server, edit the **processTable.cfg** file as follows: • cd to **/usr/ipcb/config** • Edit the **processTable.cfg** file with a text editor, e.g., vi. • Add an **ipAddress** for each corresponding **proccessName** in this file. ***Note**: The **processTable.cfg** for these Application Notes contains IP Addresses of **0.0.0.0**, where **0.0.0.0** is defined as a global IP address on the Avaya Meeting Exchange S6200 Application Server. Alternatively, the IP address of the Avaya Meeting Exchange S6200 Application Server (as defined in the /etc/hosts file) could have been entered in the **ipAddress** for each **proccessName**.*

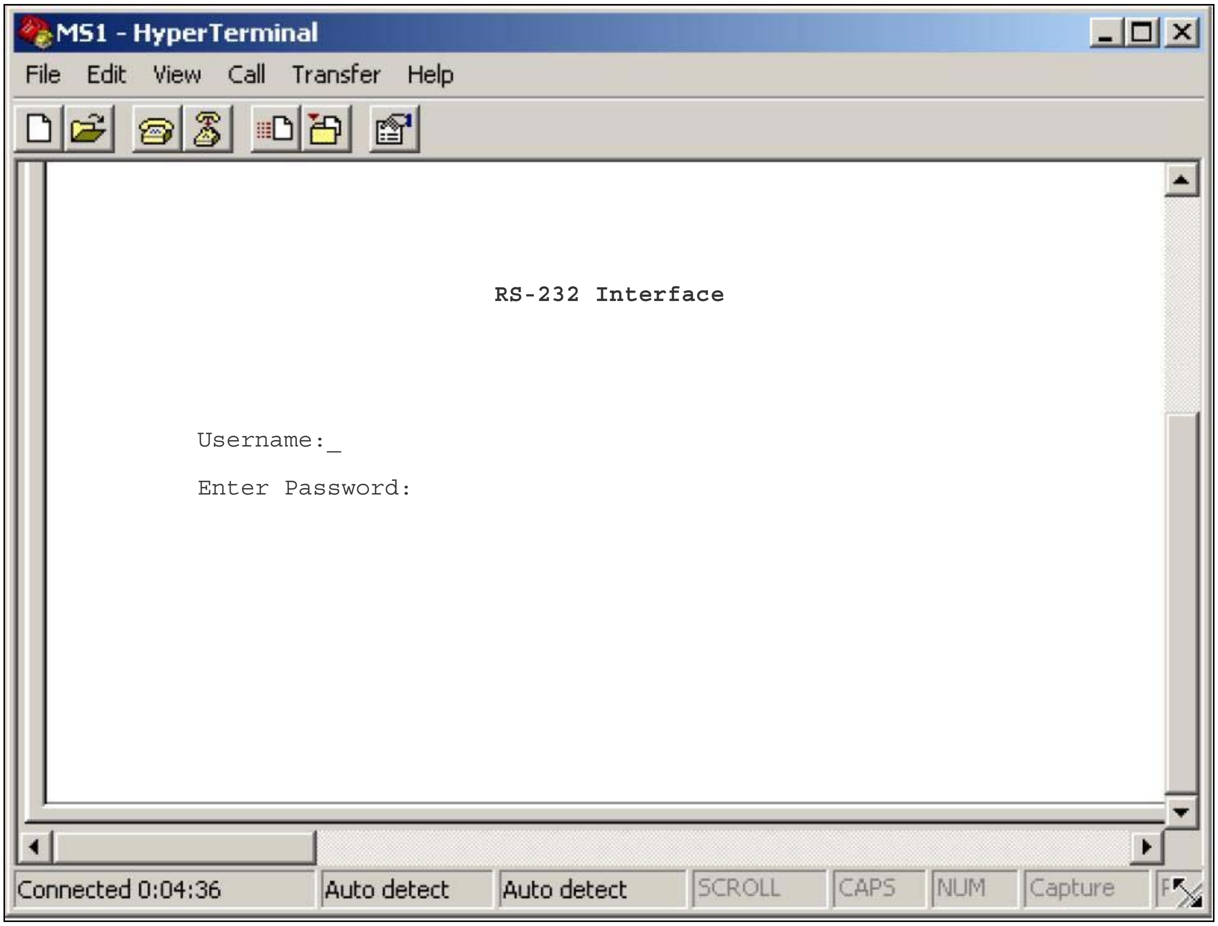
```
# processes file, enumerates the number of processes in the network.
# will have the name of the process   Key ID and the IP address
#
# The default configuration is a single MPC board system. There are
# two commented out entries for a second and third MPC board. If more
# than 1 board is needed for the system then uncomment out the appropriate
# line(s). The last thing on the line correlates to the _* entry in the
# mediaServerInterface.cfg. For example, for the 1st mediaServer line that
# ends with a 1. The _1 entries in the mediaServerInterface.cfg are used.
#
proccessName      ipcKeyNumber   ProcessExe                       ipAddress
route                                           ProcessArgs
initipcb          110            noexecute                        0.0.0.0
bridget700        100            noexecute                        0.0.0.0
dspEvents/msDispatcher,netEvents/sipAgent
commsProcess      111            /usr/dcb/bin/serverComms         0.0.0.0
sipAgent          101            /usr/dcb/bin/sipagent            0.0.0.0
dspEvents/msDispatcher,appEvents/bridget700
msDispatcher      102            /usr/dcb/bin/msdispatcher        0.0.0.0
netEvents/sipAgent,appEvents/bridget700,dspEvents/mediaServer
mediaServer       103            /usr/dcb/bin/convMS              0.0.0.0
appEvents/msDispatcher,netEvents/msDispatcher   1
#mediaServer      104            /usr/dcb/bin/convMS              0.0.0.0
appEvents/msDispatcher,netEvents/msDispatcher  2
#mediaServer      105            /usr/dcb/bin/convMS              0.0.0.0
appEvents/msDispatcher,netEvents/msDispatcher  3
```

3.2. Configure the Convedia CMS-6000 Media Server

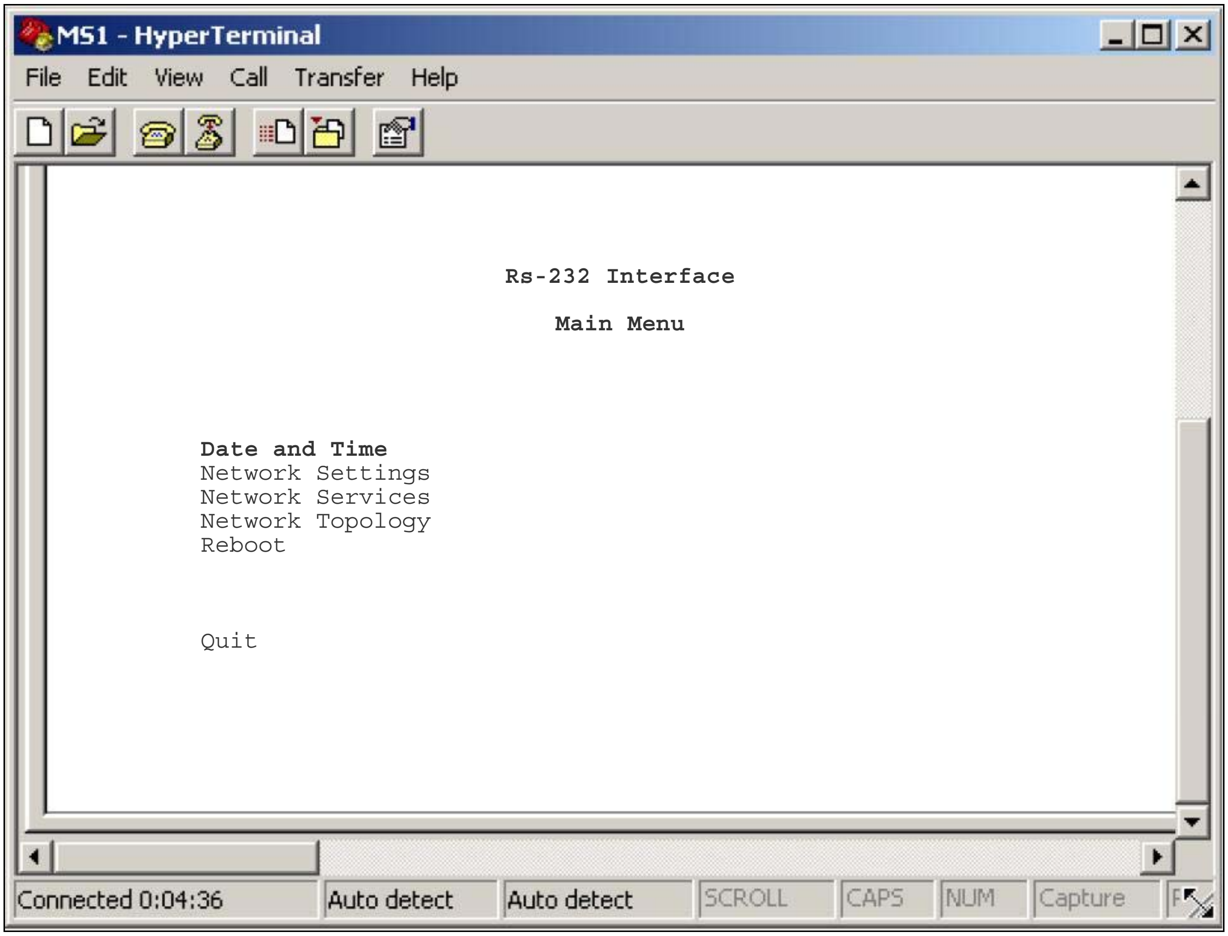
The following steps describe the administrative procedures for configuring the Convedia CMS-6000 Media Server to enable collaboration with the Avaya Meeting Exchange S6200 Application Server. For additional information regarding configuring the Convedia CMS-6000 Media Server, see **Section 8**, **Reference 2**.

Step	Description

3.6	Provision the SCC on the Convedia CMS-6000 Media Server as follows:
	• Establish an RS-232 connection from a services PC to the Convedia CMS-6000 Media Server by connecting a serial cable to the front of the SCC card (slot 1).
	• Start a terminal server application, e.g., HyperTerminal on the services PC with the following settings:
	  o Speed: 9600 bps.
	  o Data bits: 8 bits.
	  o Parity: No parity.
	  o Stop bit: 1 bit.
	  o Flow control: none.
	• Wait for the system to establish the connection, or press <Enter>.

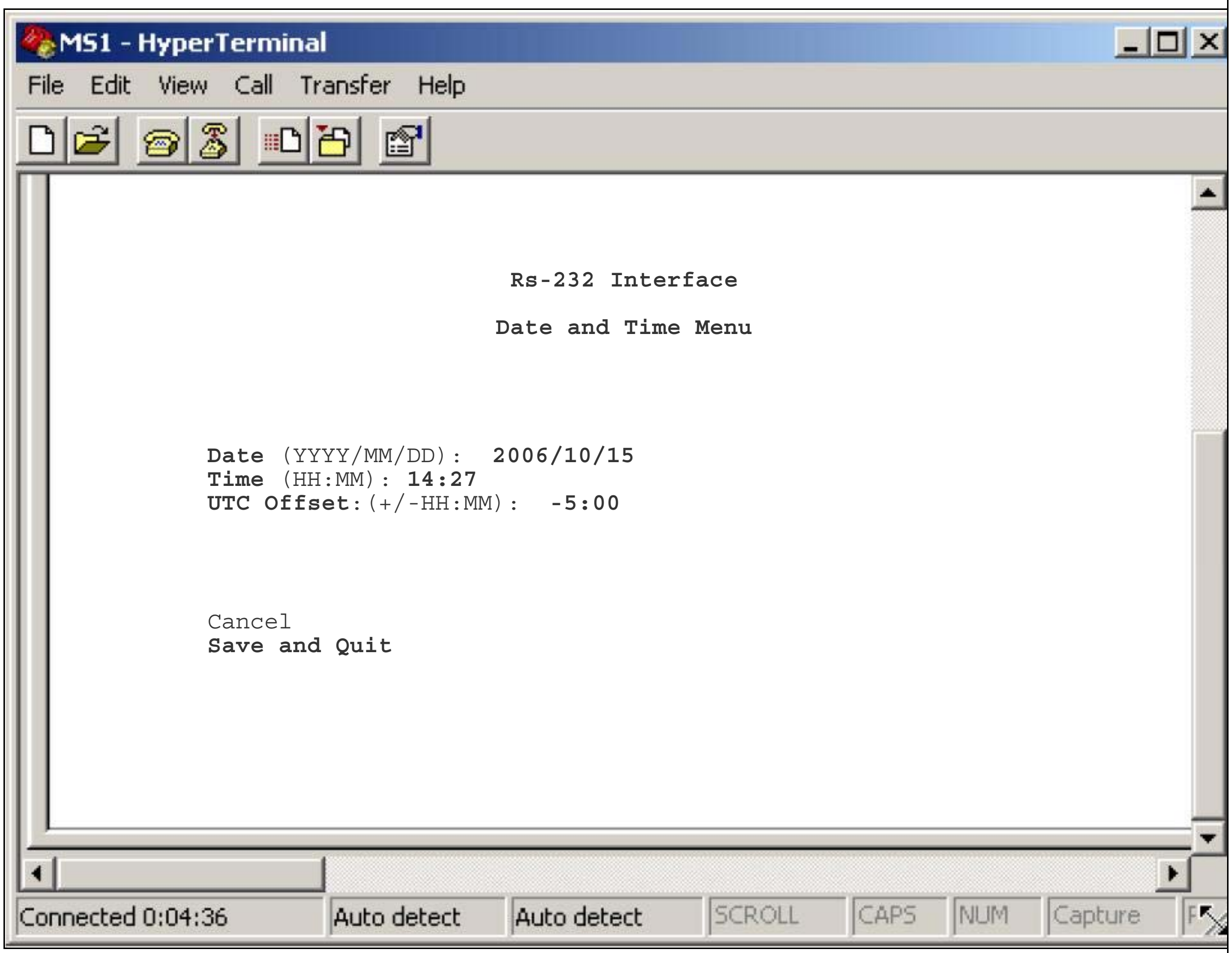
Step	Description
3.7	From the **RS-232 Interface** login screen that is displayed, log in to the Convedia CMS-6000 Media Server craft interface with the appropriate credentials. 

Step	Description
3.8	From the **RS-232 Interface Main Menu** screen that is displayed, select **Date and Time** and press <Enter>.



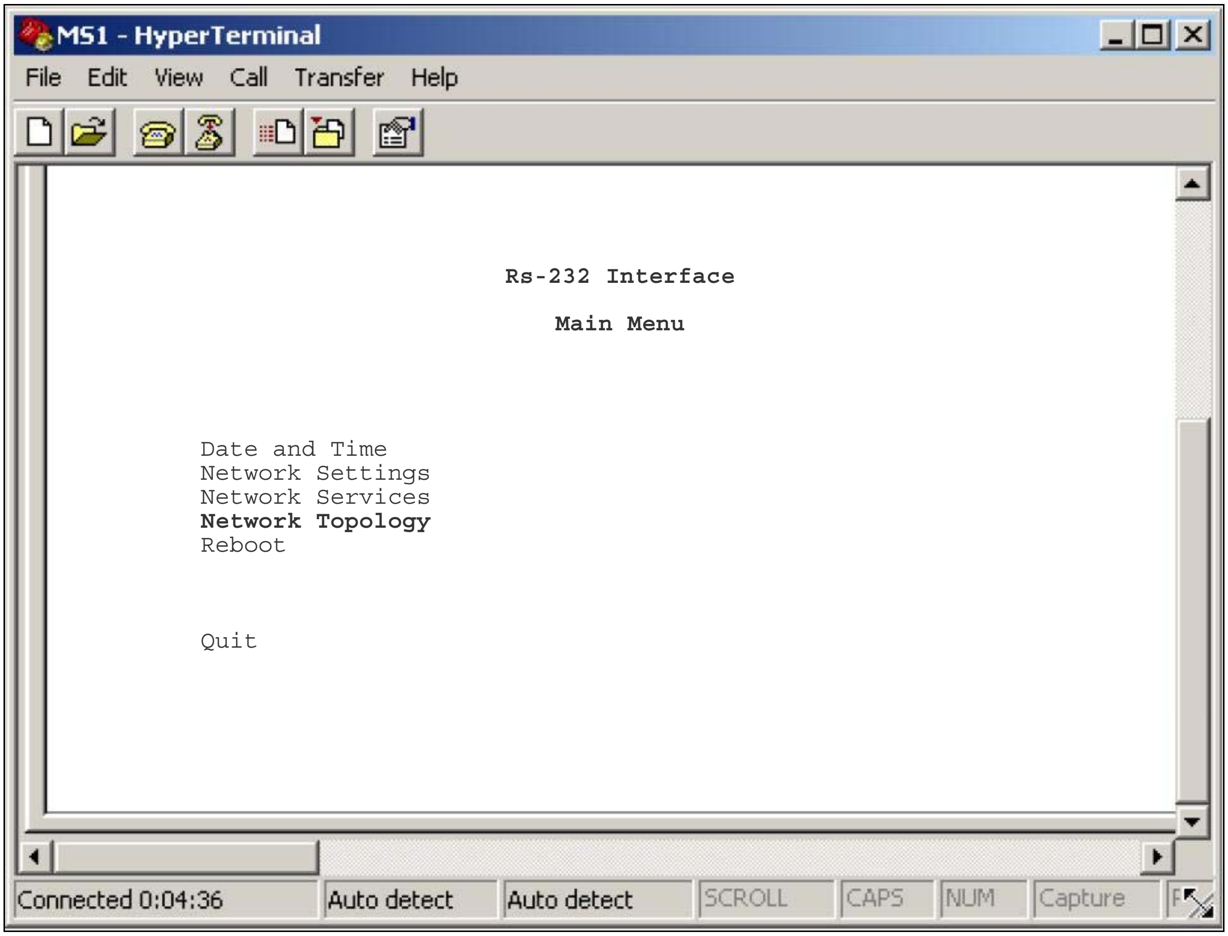
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.9</td><td>From the RS-232 Interface Date and Time Menu that is displayed, configure settings for the date and time as follows.

Set the Date to the current date.
Set the Time to the current time.
Set the UTC Offset to compensate for the location of the Convedia CMS-6000 Media Server relative to the Universal Time Clock (UTC) or Greenwich Mean Time (GMT).
Note: The UTC Offset is derived from the location of Convedia CMS-6000 Media Server relative to the UTC/GMT. Format is +|–hh:mm, where + represents the number of hours ahead of UTC, – is the number of hours behind UTC. For example, Moscow is +3:00, London is +0:00, New York is –5:00 and Los Angeles is –8:00.
Save the settings by using <Tab> to navigate down to Save and Quit and press <Enter>.



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

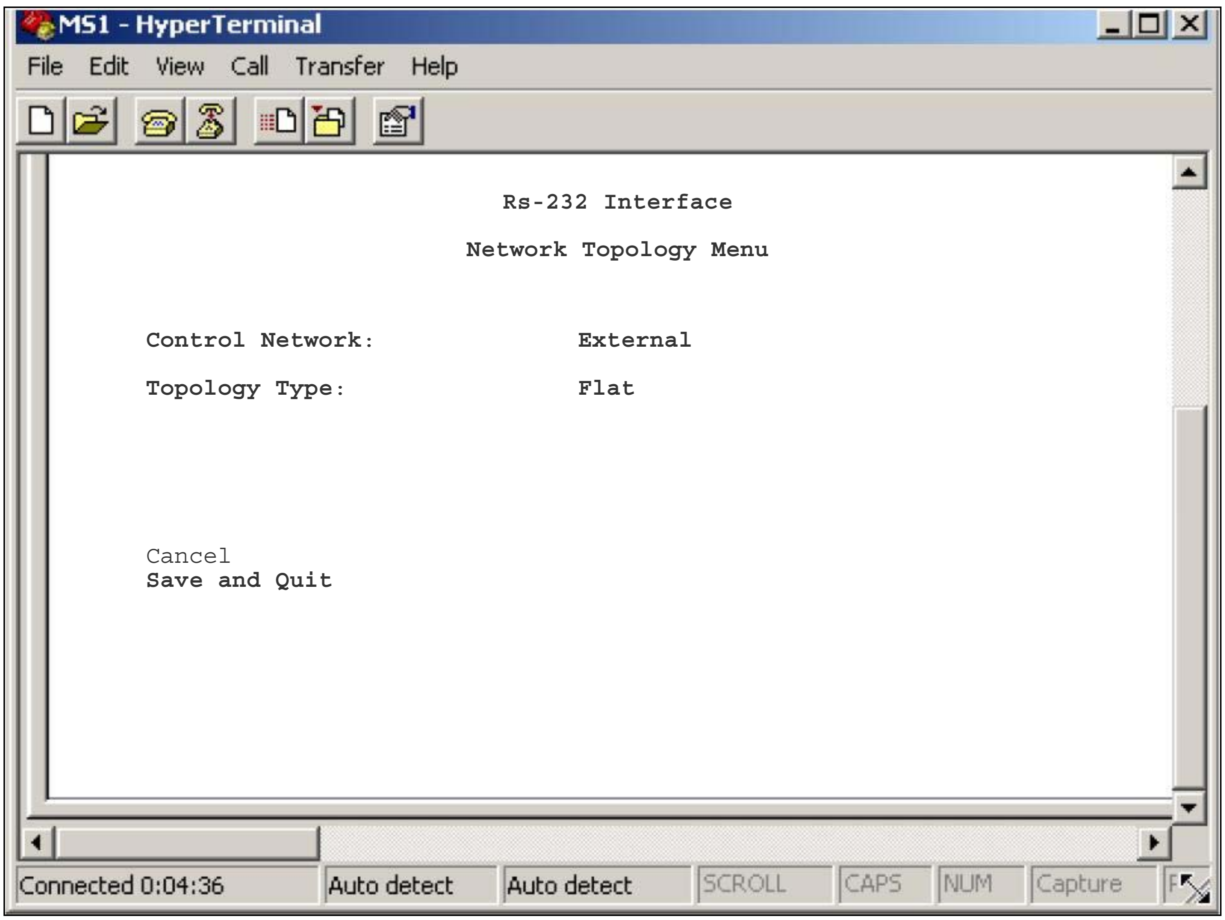
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.10</td><td>From the RS-232 Interface Main Menu screen that is displayed, select Network Topology and press <Enter>.



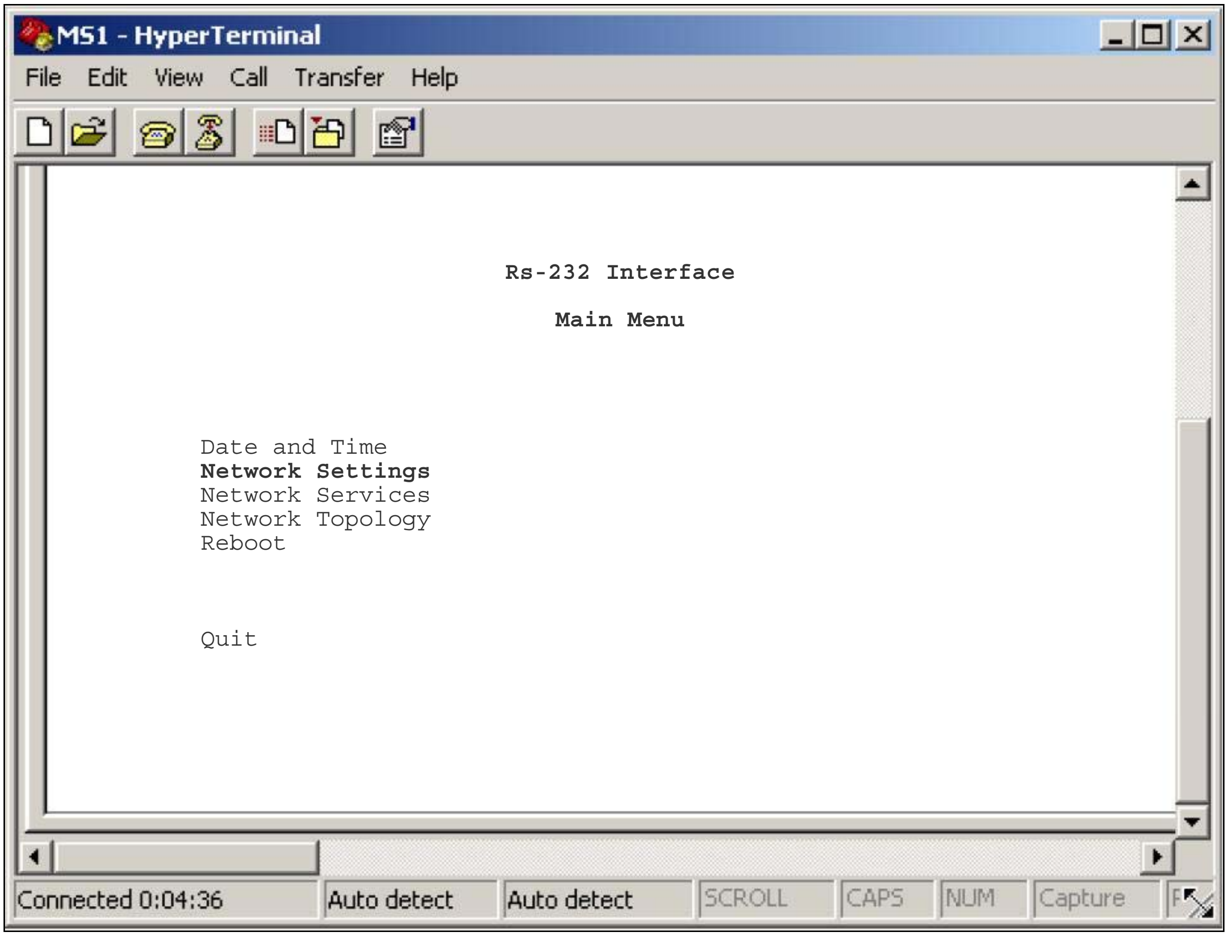
</td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.11</td><td>From the RS-232 Interface Network Topology Menu that is displayed, configure the network topology as follows.

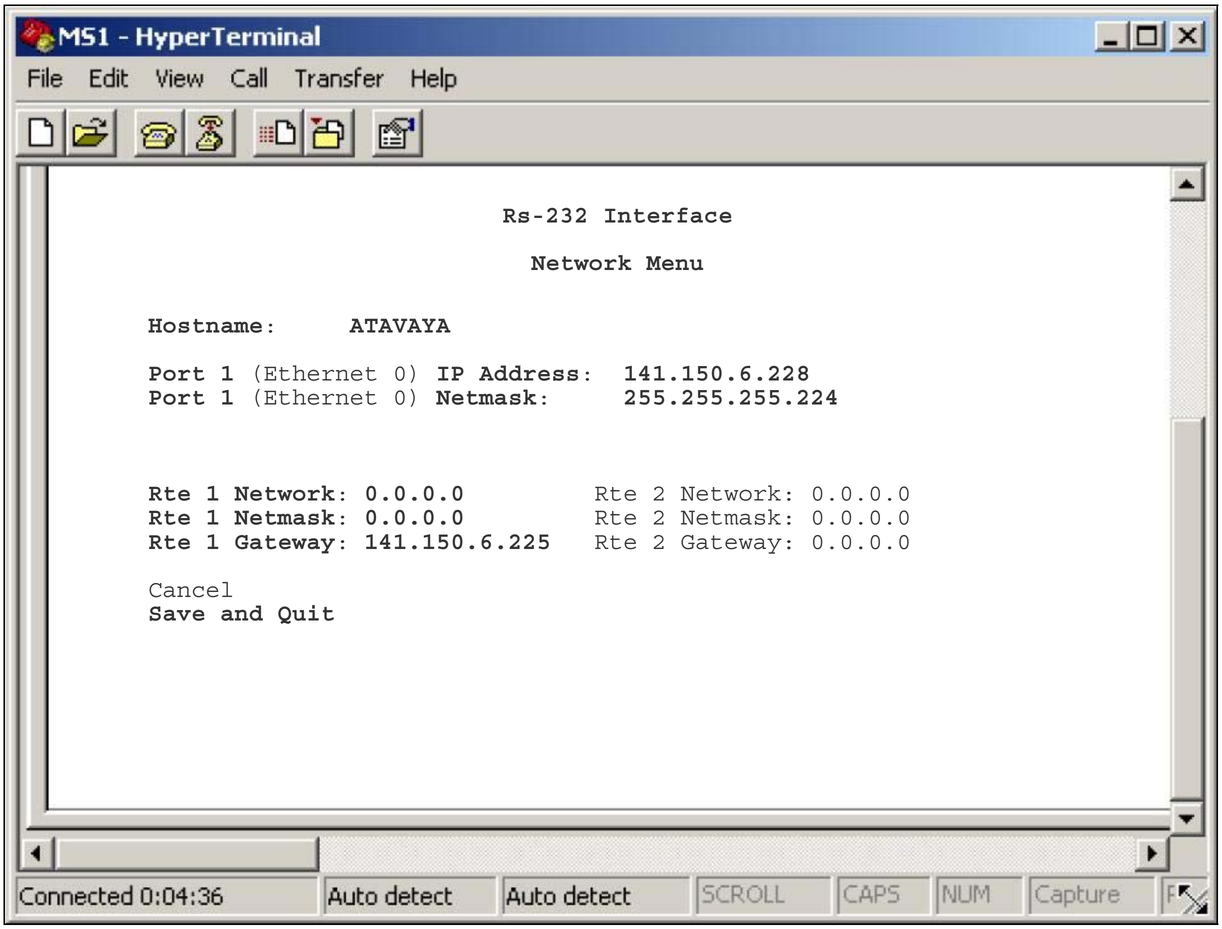
Set the Control Network to External by using the spacebar to toggle between values and press <Enter> to accept the value.
Note: An External Control Network is where MPC control interfaces have IP addresses on the external control subnet. The control agent communicates directly with an MPC through its control interface.
Set the Topology Type to Flat by using the spacebar to toggle between values and press <Enter> to accept the value.
Note: A Flat Topology Type is where control and media share a single network segment.
Save the settings by using <Tab> to navigate down to Save and Quit and press <Enter>.

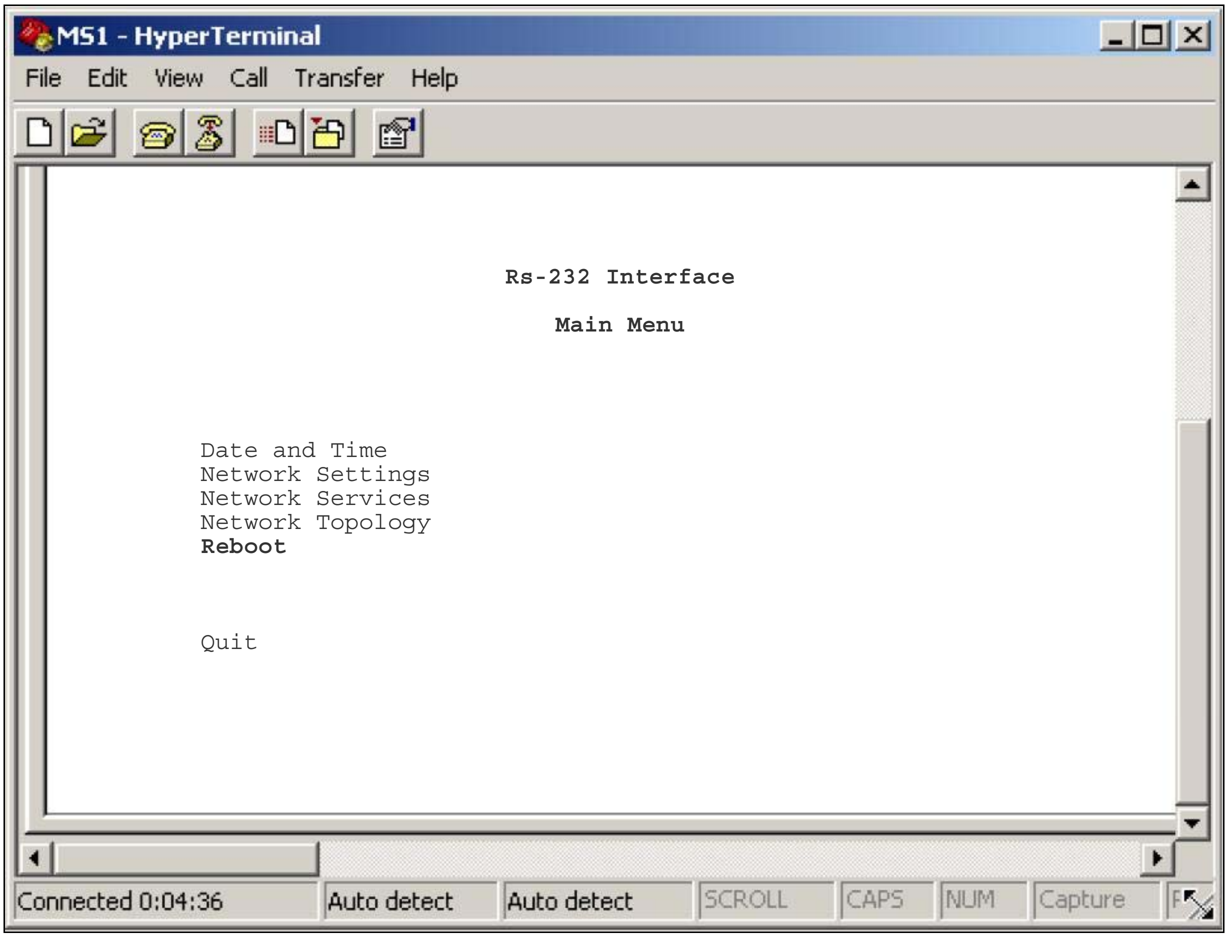


</td></tr>
</table>

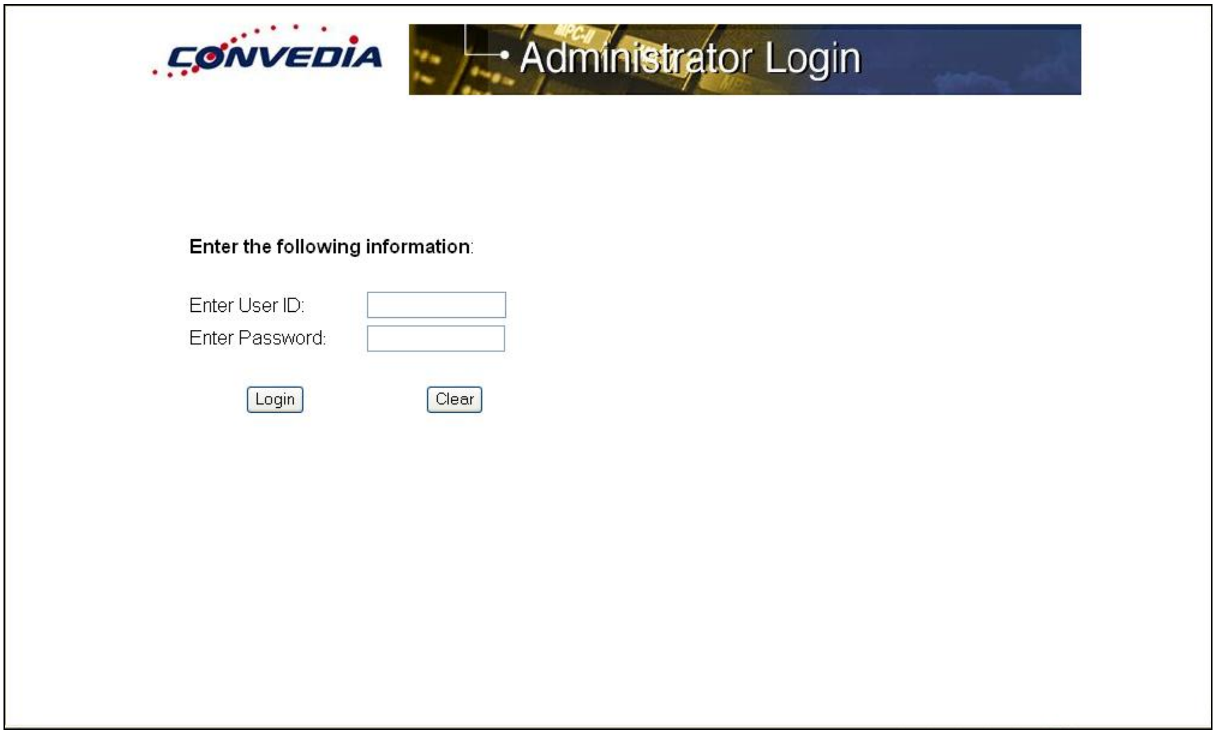
Step	Description
3.12	From the **RS-232 Interface Main Menu** screen that is displayed, select **Network Settings** and press <Enter>. 

Step	Description

3.13	From the **RS-232 Interface Network Menu** that is displayed, configure network settings as follows.
	• Administer network parameters used for control and management traffic on the Convedia CMS-6000 Media Server by specifying:
	  o A **Hostname** for the Convedia CMS-6000 Media Server.
	  o An **IP Address** and **Netmask** for **Port 1**.
	• Administer routing parameters used for remote control or management networks on the Convedia CMS-6000 Media Server by specifying:
	  o A **Network** IP address, **Netmask** and **Gateway** for **Rte 1**.
	    ***Note**: To indicate the default gateway, leave the **Network** IP address and **Netmask** blank (**0.0.0.0**). The **Gateway** must be on a directly connected network.*
	• Save the settings by using <Tab> to navigate down to **Save and Quit** and press <Enter>.
	

Step	Description
3.14	From the **RS-232 Interface Main Menu** screen that is displayed, preserve the configuration administered in the previous steps by rebooting the Convedia CMS-6000 Media Server. • Select **Reboot** and press <Enter>. o [***Not Shown***] *A confirmation message displays to confirm the reboot.* o [***Not Shown***] *Use the <Tab> key to toggle to the **YES** option and press <Enter>.* o [***Not Shown***] *Use the spacebar to toggle to the Choose the **Restart with Current Configuration** option.* o [***Not Shown***] *A confirmation message displays to confirm the reboot.* o [***Not Shown***] *Use the <Tab> key to toggle to the **YES** option and press <Enter>.* • The media server restarts and the network settings are enabled. 

Step	Description
3.15	Administer settings for Convedia CMS-6000 Media Server via the web GUI as follows: • Open a web browser and enter the following URL: **http://<IP address of Convedia CMS-6000 Media Server >** • Log in to the Convedia CMS-6000 Media Server with the appropriate credentials.



<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.16</td><td>
Administer settings for Audio Codec(s) on the Convedia CMS-6000 Media Server as follows:

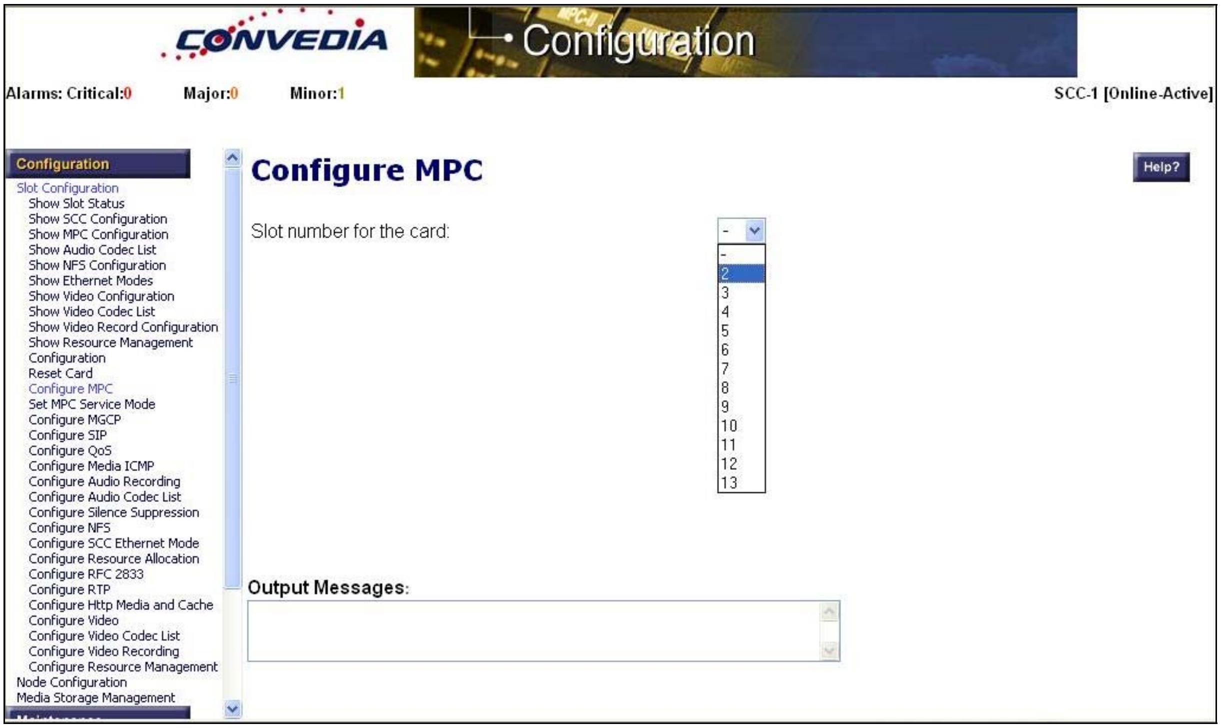
Click Configuration → Slot Configuration → Configure Audio Codec List.
Select either the Slot Number for the MPC card or all (MPC cards) to which this Audio Codec List will be applied.
Click Execute.

Note: Audio Codecs in the Audio Codec List are prioritized from First codec to Tenth codec.

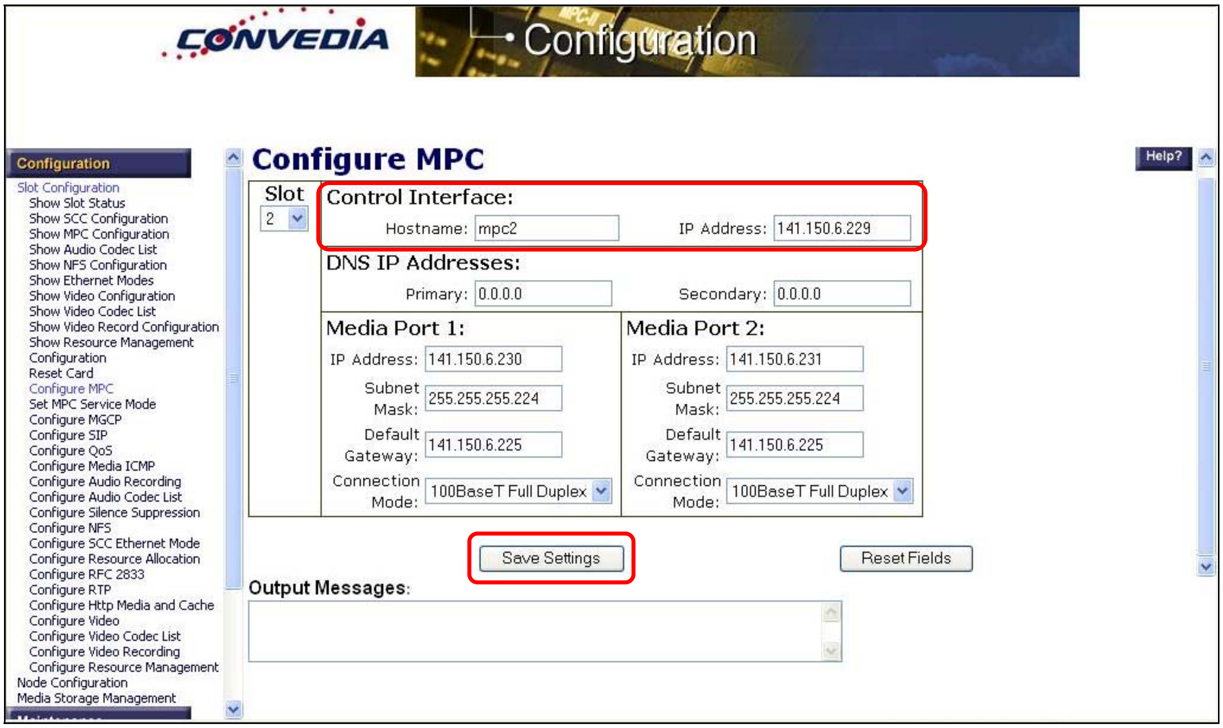
</td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.17</td><td>Administer settings for MPC(s) on the Convedia CMS-6000 Media Server as follows:

Click Configuration → Slot Configuration → Configure MPC.
Select the Slot Number for the MPC. For these Application Notes, the MPC was placed in Slot number 2.



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

Step	Description
3.18	Configure the MPC in slot 2 on the Convedia CMS-6000 Media Server as displayed: • Enter a **Hostname** and **IP Address** for the **Control Interface**. • Enter **IP Address**, **Subnet Mask**, **Connection Mode** and **Default Gateway** information for **Media Ports 1** and **2**. • Click on the **Save Settings** button when finished. ***Note***: *Repeat from **Step 3.17** to configure each MPC on the Convedia CMS-6000 Media Server. For these Application Notes, there is only one MPC.* 

3.3. Network File System

The following steps describe the administrative procedures to enable Network File System (NFS) sharing between the Avaya Meeting Exchange S6200 Application Server and the Convedia CMS-6000 Media Server. In this configuration, the Avaya Meeting Exchange S6200 Application Server will function as the NFS server. This will allow playback of audio conference(s) recorded on the Convedia CMS-6000 Media Server from the Avaya Meeting Exchange S6200 Application Server.

3.3.1. Configure NFS on the Avaya Meeting Exchange S6200 Application Server

The following steps describe the administrative procedures to provision NFS on the Avaya Meeting Exchange S6200 Application Server.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.19</td><td>Log in to the Avaya Meeting Exchange S6200 Application Server console to access the CLI with the appropriate credentials.</td></tr>
<tr><td>3.20</td><td>The NFS server communicates with the control interface on the Convedia CMS-6000 Media Server MPC. To resolve the IP address for the control interface on the Convedia CMS-6000 Media Server MPC, edit the hosts file as follows:

cd to /etc
Edit the hosts file with a text editor, e.g., vi.
Add a line to the file to resolve the IP address of the control interface to the Convedia CMS-6000 Media Server MPC in slot 2:
141.150.6.229 mpc2
Where 141.150.6.229 and mpc2 are the IP address and hostname of the control interface assigned to the Convedia CMS-6000 Media Server MPC in Step 3.18.
</td></tr>
<tr><td>3.21</td><td>To allow the Convedia CMS-6000 Media Server MPC to mount the /usr3/ipcb directory on the Avaya Meeting Exchange S6200 Application Server, edit the dfstab file as follows:

cd to /etc/dfs
Edit the dfstab file with a text editor, e.g., vi.
Add a line to the file to assign read/write (rw) privileges to the directory /usr3/ipcb for the Convedia CMS-6000 Media Server:
/usr/sbin/share -F nfs -o rw=mpc2 /usr3/ipcb
Where mpc2 is the hostname assigned to the Convedia CMS-6000 Media Server MPC in Step 3.20.
</td></tr>
</table>

Step	Description

3.22	To configure the Avaya Meeting Exchange S6200 Application Server as an NFS server, edit the **mediaServerInterface.cfg** file as follows:
	• cd to **/usr/ipcb/config**
	• Edit the **mediaServerInterface.cfg** file with a text editor, e.g., vi.
	• Add a line to the file to assign the Avaya Meeting Exchange Application Server as the NFS server:
	  o **NFSServerIPAddress=192.168.13.101**
	  Where **192.168.13.101** is the IP address assigned to the Avaya Meeting Exchange Application Server.
	• Add a line to the file to assign the Convedia CMS-6000 Media Server as a media server:
	  o **MediaServerIP_1=141.150.6.229**
	  Where **141.150.6.229** is the IP address of the control interface assigned to the Convedia CMS-6000 Media Server MPC in **Step 3.18**.
	    ***Note**: Multiple MPC cards on the Convedia CMS-6000 Media Server would each require an entry in the **mediaServerInterface.cfg** file. The requirement for successive entries is to increment the MediaServerIP_X variable by 1, e.g., MediaServerIP_2 would correspond to a second MPC, MediaServerIP_3 to a third, etc..*
	• Add a line to the file to assign a port to the Convedia CMS-6000 Media Server:
	  o **MediaServerInterfaceSipListenPort_1=5050**
	    ***Note**: Multiple MPC cards on the Convedia CMS-6000 Media Server would each require an entry for a unique port in the **mediaServerInterface.cfg** file. The requirement for the successive port entries are to decrease the port number by ten for each MPC card, e.g., the port number for a second MPC would be 5040, a third MPC would have a port entry of 5030, etc..*

	# This file contains the configuration information for the # Media Server Interface. This information includes the # IP Address for the NFS Server (where recordings are stored), # the IP address of the Media Server(may be more than 1), and # the udp port that the Media Server Interface code should # listen for SIP responses. # **# NFS Server** **NFSServerIPAddress=192.168.13.101** # **# MPC 1 on Convedia CMS-6000 Media Server (Control Port)** **MediaServerIP_1=141.150.6.229** **MediaServerInterfaceSipListenPort_1=5050**

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.23</td><td>From the /usr3 directory on the Avaya Meeting Exchange S6200 Application Server, verify the following symbolic link is present: confrp -> /usr3/ipcb/usr3/confrp.
<pre>S6200App->pwd
/usr3
S6200App->ls -l
total 4
drwxr-xr-x   3 root   dcb   1024 Jan 17 04:20 BACKUPS
lrwxrwxrwx   1 root   sys   22 Nov 30 19:01 confrp -> /usr3/ipcb/usr3/confrp
drwxr-xr-x   5 root   sys   96 Jun 29 2006 ipcb
drwxrwxrwx   20 root   root   1024 Nov 6 19:03 runtime
drwxrwxr-x   2 root   dcb   96 Oct 5 2005 savedroster</pre></td></tr>
<tr><td>3.24</td><td>Reboot the Avaya Meeting Exchange S6200 Application Server for changes to take effect.

Note: Rebooting the Avaya Meeting Exchange S6200 Application Server is service impacting.
<pre>[S6800]> init 6</pre></td></tr>
</table>

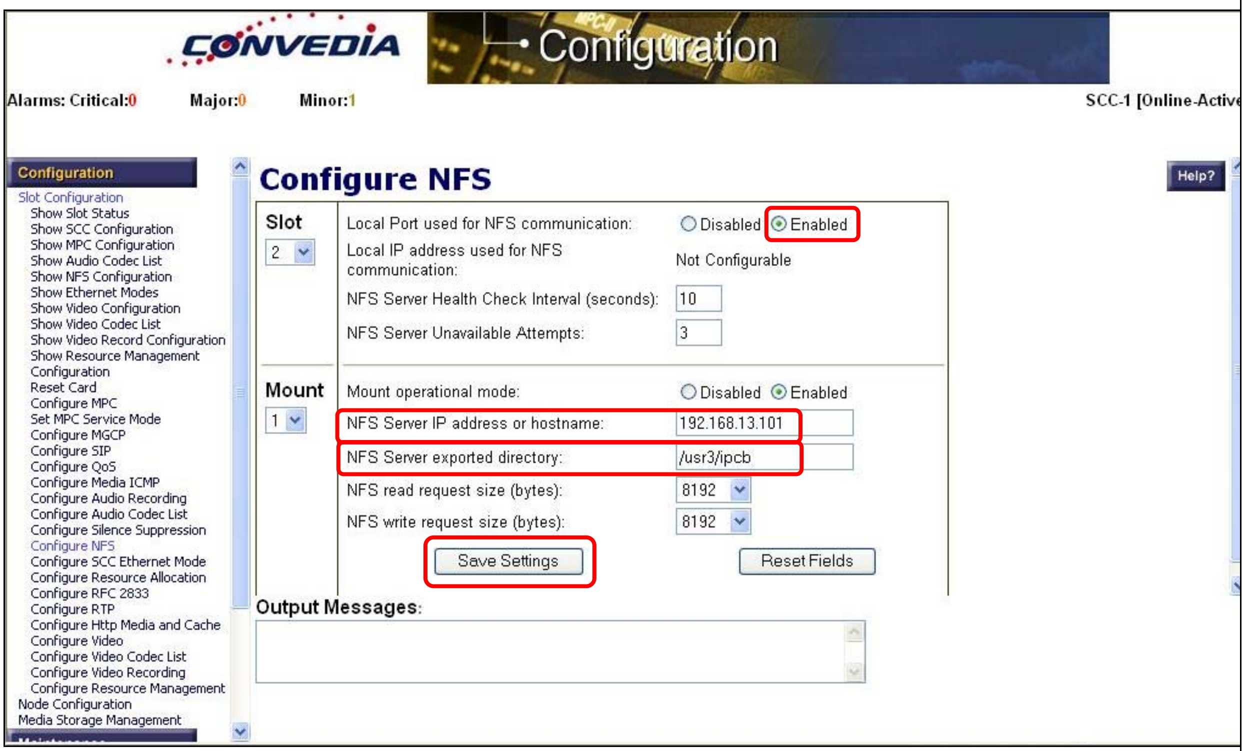
3.3.2. Configure NFS on the Convedia CMS-6000 Media Server

The following steps describe the administrative procedures to provision NFS on the Convedia CMS-6000 Media Server.

Step	Description
3.25	Administer settings for NFS on the Convedia CMS-6000 Media Server MPC(s) via the web GUI as follows: • Click **Configuration → Slot Configuration → Configure NFS**. • Select the **Slot Number for the** MPC to administer settings for NFS. For these Application Notes, the MPC was placed in **Slot number 2**. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.26</td><td>Configure NFS parameters for the MPC in slot 2 on the Convedia CMS-6000 Media Server as displayed:
• Select Enabled for the Local Port used for NFS communication to enable NFS on this MPC.
• Enter the IP address for the NFS server provisioned in Step 3.22 in the NFS Server IP address or hostname field.
• Enter /usr3/ipcb (see Step 3.21) in the NFS Server exported directory field.
• Remaining fields are default settings.
• Click on the Save Settings button when finished.

Note: Repeat from Step 3.25 to Configure NFS for each MPC on the Convedia CMS-6000 Media Server. For these Application Notes, there is only one MPC.


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

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.27</td><td>
Reset the Convedia CMS-6000 Media Server MPC in slot 2 for changes to take effect as follows:

- Click **Configuration → Reset Card**.
- Select the slot number for the MPC to reset. For these Application Notes, the MPC was placed in slot number **2**.
- Select **Forced** for the **Type of reset operation**.
- Click **Execute**.

Note*: If there is only one MPC in the Convedia CMS-6000 Media Server chassis, resetting the MPC is service impacting. If more than one MPC is present, resetting a single MPC would not be service impacting, as all traffic on the MPC being reset would fail over to an active MPC.*

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

3.4. CBUTIL Utility

The following steps provide examples of how to provision DIRECT and SCAN call functions by utilizing the cbutil utility on the Avaya Meeting Exchange S6200 Application Server. DID values (obtained from procedures in **Step 3.3**) are associated with call functions to access conferences provisioned on the Avaya Meeting Exchange S6200 Application Server.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td rowspan="3">3.28</td><td>To map DID values obtained in Step 3.3 to DNIS entries, run the cbutil utility as follows:

If not already logged on, log in to the Avaya Meeting Exchange S6200 Application Server console to access the CLI with the appropriate credentials.
At the command prompt enter tcsh to set the UNIX shell on the Avaya Meeting Exchange S6200 Application Server.
At the command prompt run the cbutil utility to verify DNIS entries provisioned on the Avaya Meeting Exchange S6200 Application Server.

Note: A command line utility, cbutil enables administrators to assign a specific annunciator message, line name, company name, system function, reservation group and prompt sets to a maximum of 30,000 DNIS or DID entries. The Avaya Meeting Exchange S6200 Application Server parses these entries in numerically ascending order, with the wildcard character “?” last in a series. For example, 129? follows 1299. The last entry in the table consists entirely of wildcard characters.</td></tr>
<tr><td><pre>S6200App->cbutil
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.

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

Step	Description

3.29 Enable Dial-In access (via passcode) to conferences provisioned on the Avaya Meeting Exchange S6200 Application Server as follows:

- Add a DNIS entry for a **scan call function** corresponding to DID **710** by entering the following command at the command prompt:
  cbutil add <**dnis**> <**rg**> <**msg**> <**ps**> <**ucps**> <**func**> [**-l** <**ln**> **-c** <**cn**>], where the variables for add command is defined as follows:
  - <**dnis**> DNIS
  - <**rg**> Reservation Group
  - <**msg**> Annunciator message number
  - <**ps**> Prompt Set number (0-20)
  - <**ucps**> Use Conference Prompt Set (y/n)
  - <**func**> One of: DIRECT/SCAN/ENTER/HANGUP/AUTOVL/FLEX
  - **-l** <"**ln**"> Optional line name to associate with caller
  - **-c** <"**cn**"> Optional company name to associate with caller

```
S6200App->cbutil add 710 0 1 1 n scan
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.
```

3.30 Enable Dial-In access (as moderator, without entering a passcode) to conferences provisioned on the Avaya Meeting Exchange S6200 Application Server by adding a DNIS entry for a **direct call function** corresponding to DID **777**.

```
S6200App->cbutil add 777 0 301 1 n direct
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.
```

3.31 At the command prompt enter **cbutil list** to verify the DNIS entries provisioned in **Step 3.29** and **Step 3.30** were provisioned and entered correctly.

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

```
S6200App->cbutil list
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.

DNIS             Grp Msg PS  CP Function Line Name            Company Name
---------------- --- --- --- -- -------- -------------------- -----------------
710              0   1   1   N  SCAN
777              0   301 1   N  DIRECT
???              0   208 1   N  ENTER
```

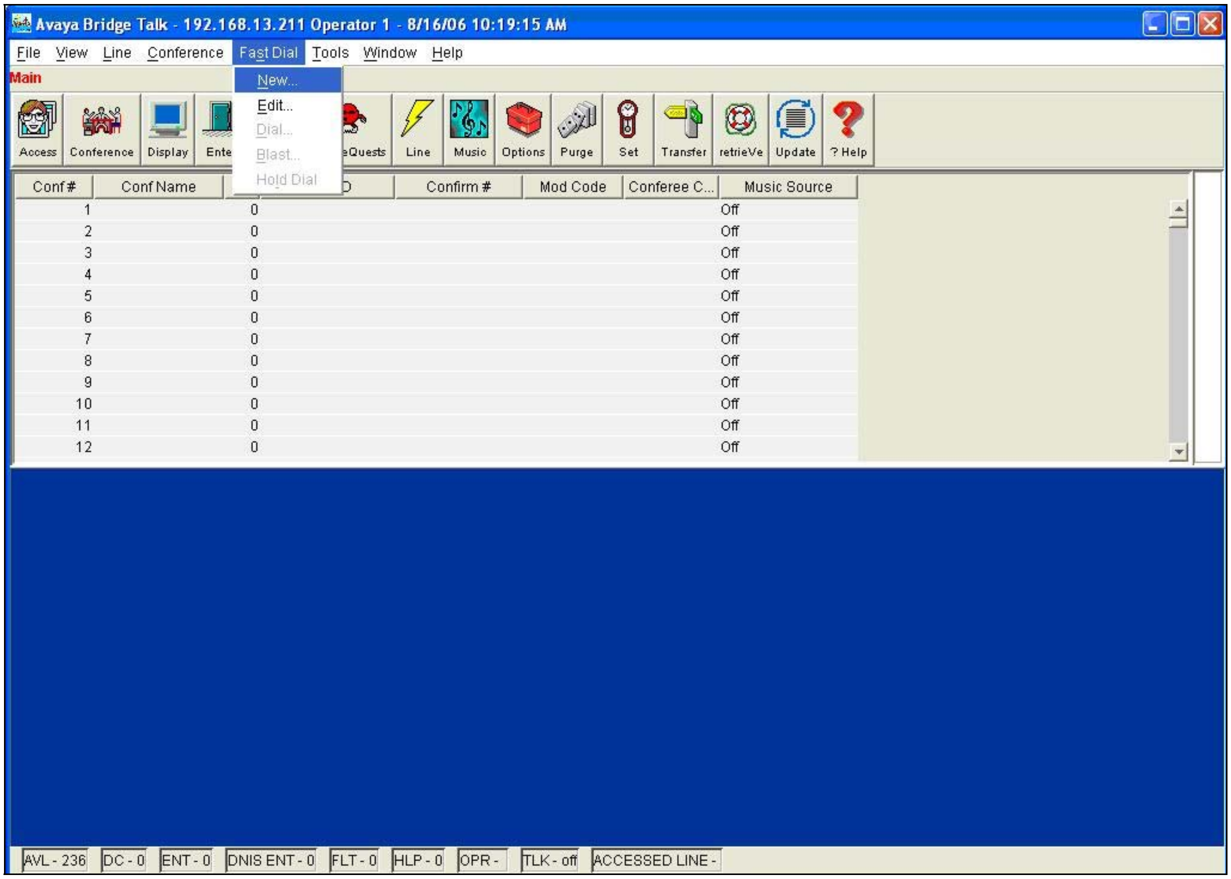
3.5. Bridge Talk

The following steps utilize the Avaya Bridge Talk application to provision a sample conference on the Avaya Meeting Exchange S6200 Application Server. This sample conference is utilized in conjunction with the DIRECT and SCAN call functions provisioned in **Section 3.4** to enable both Dial-In and Dial-Out access to audio conferencing for endpoints on the PSTN.

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

Step	Description
3.32	Invoke the Avaya Bridge Talk application as follows: • [*Not Shown*] *Double-click on the desktop icon from a PC loaded with the Avaya Bridge Talk application and with network connectivity to the Avaya Meeting Exchange S6200 Application Server*. • Enter the IP address of the Avaya Meeting Exchange S6200 Application Server (**192.168.13.101**) in the **Bridge** field. • Enter the appropriate credentials in the **Sign-In** and **Password** fields. 

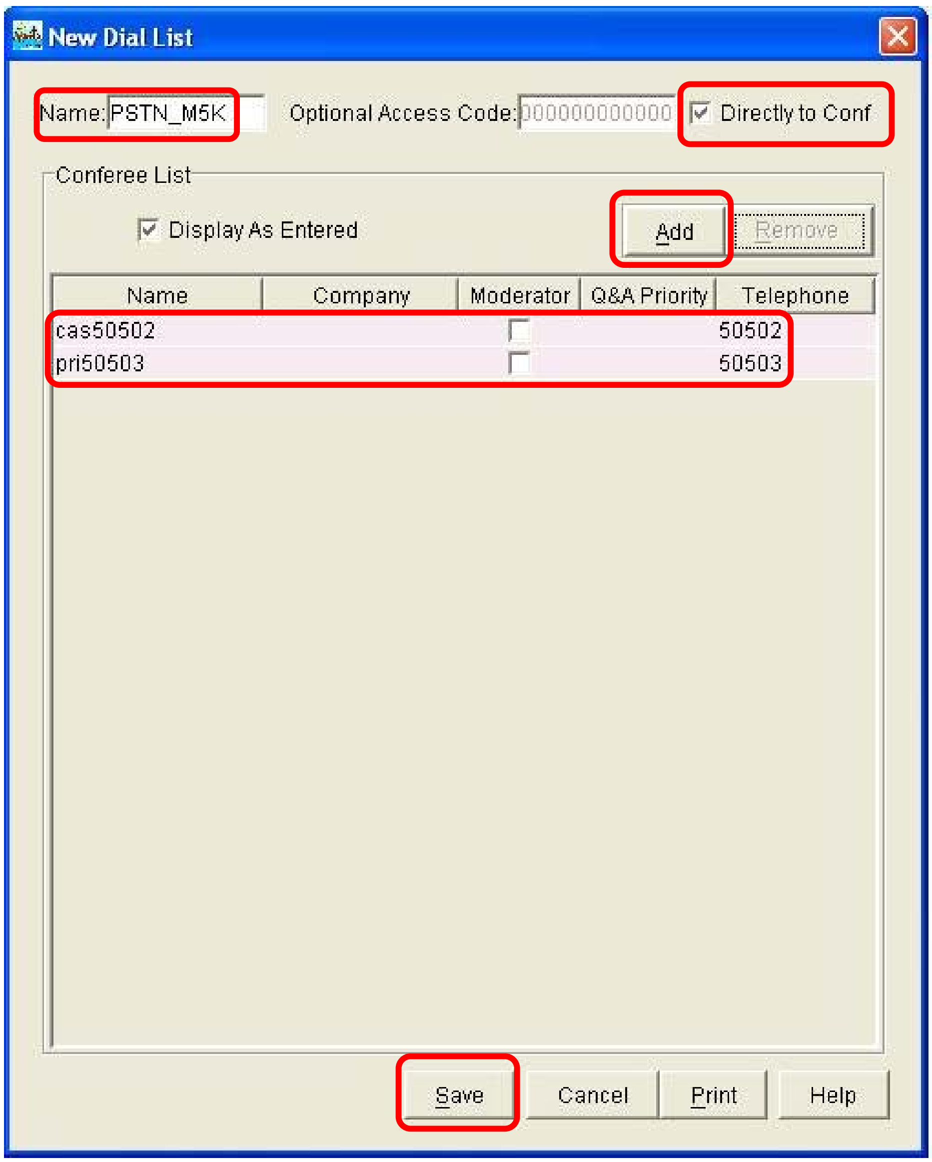
Step	Description
3.33	Provision a dial list that is utilized for Dial-Out (e.g., Blast dial and Fast Dial) from the Avaya Meeting Exchange S6200 Application Server. From the Avaya Bridge Talk Menu Bar, click **Fast Dial → New**.



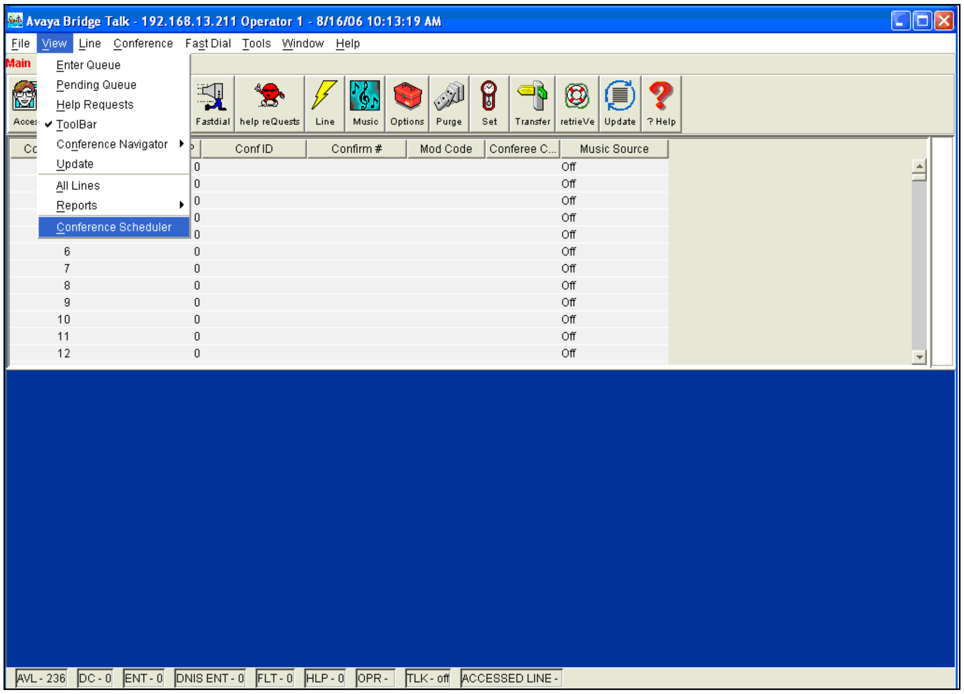
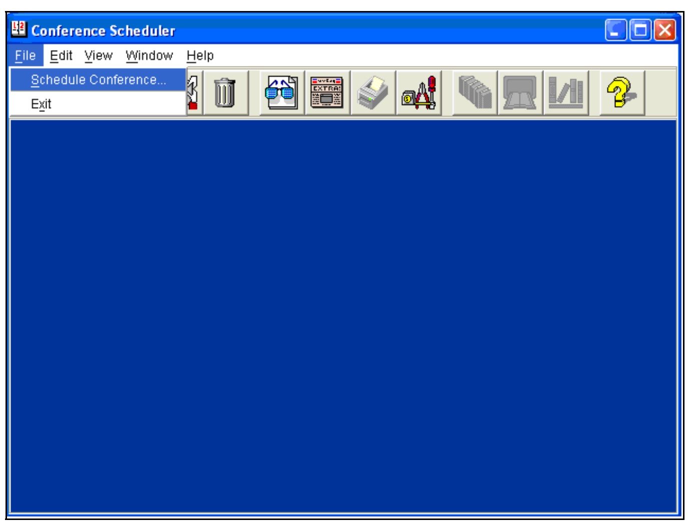
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.34</td><td>

From the **New Dial List** window that is displayed:

- Enter a descriptive label in the **Name** field.
- Enable conference participants on the dial list to enter the conference without a passcode by checking the **Directly to Conf** box as displayed.
- Add entries to the dial list by clicking on the **Add** button for each participant.
  - [*Optional*] *Moderator privileges may be granted to a conference participant by checking the **Moderator** box.*
- See **Section 8**, **Reference 3** for provisioning the remaining fields in this screen.
- When finished, click on the **Save** button on the bottom of the screen.

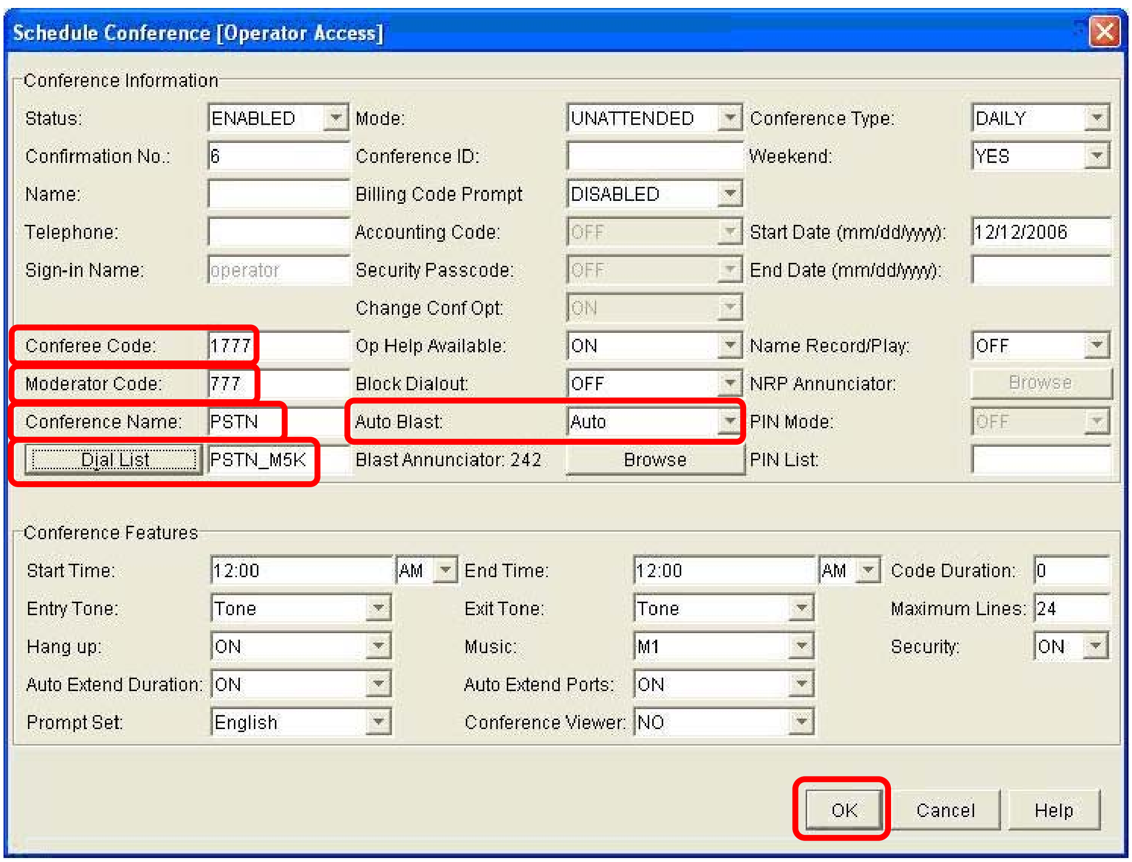


</td></tr>
</table>

Step	Description
3.35	Provision a conference with Auto Blast enabled. From the Avaya Bridge Talk Menu Bar, click **View → Conference Scheduler**. 
3.36	From the **Conference Scheduler** window that is displayed, click **File → Schedule Conference**. 

Step	Description

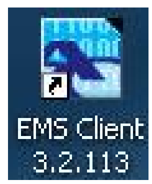
3.37	From the **Schedule Conference** window that is displayed, provision a conference as follows:
	• Enter a unique **Conferee Code** to allow participants access to this conference.
	• Enter a unique **Moderator Code** to allow participants access to this conference with moderator privileges. Enable moderator access without a passcode for this conference call by configuring the following:
	    o The **Moderator Code** "**777**" must have an associated **direct call function** provisioned for "**777**" (see **Step 3.30**).
	__Note__: This conference remains open for participants to enter as either moderator or participant by entering the appropriate code when prompted.
	• Enter a descriptive label in the **Conference Name** field.
	• Administer settings to enable an Auto Blast dial by setting **Auto Blast** to **Auto** and selecting the dial list provisioned in **Step 3.34**.
	    o [***Not Shown***] *Select a dial list by clicking on the **Dial List** button ➔ select a dial list from the **Create, Select or Edit Dial List** window that is displayed ➔ click on the **Select** button.*
	• See **Section 8**, **Reference 3** for provisioning the remaining fields in this screen.
	• When finished, click on the **OK** button on the bottom of the screen.



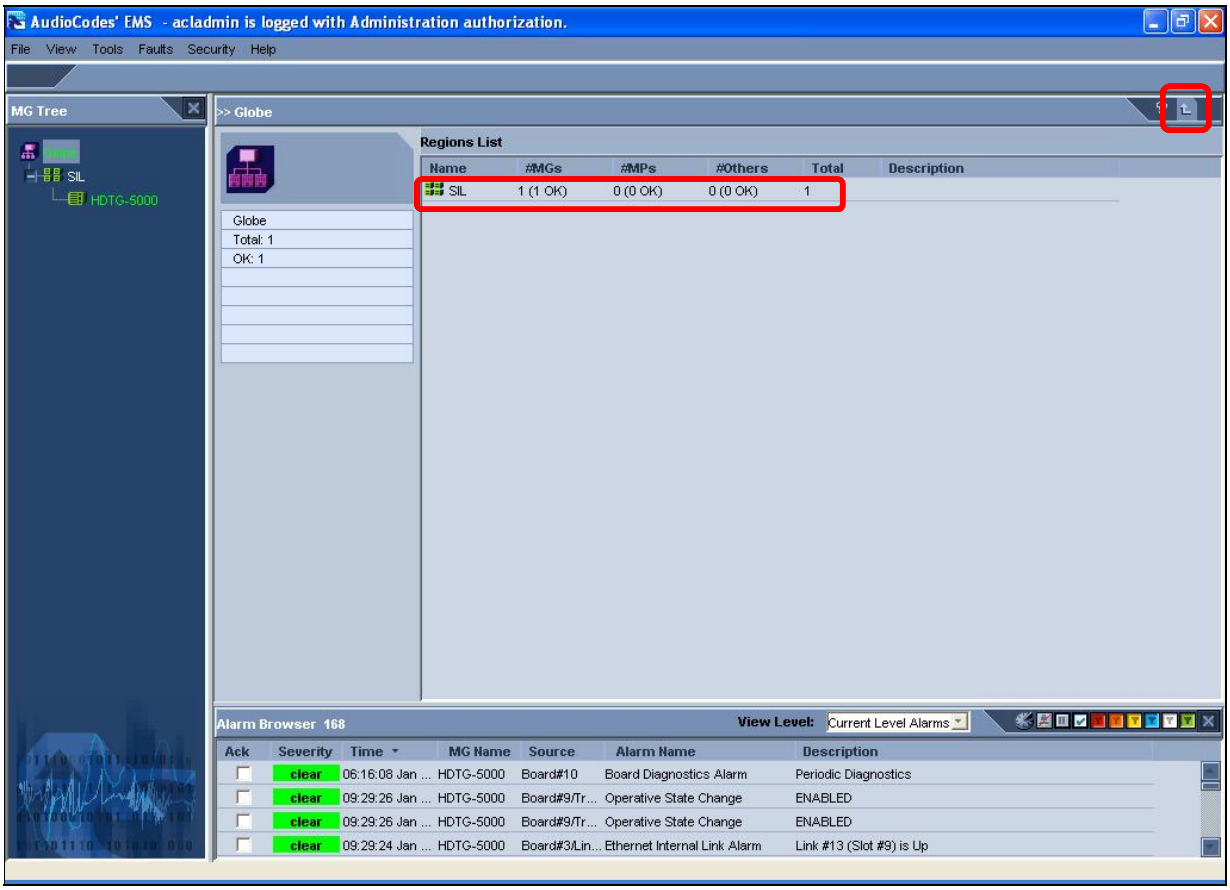
4. Configure the AudioCodes Mediant 5000 Media Gateway

The following sections describe the steps for configuring the SIP and PSTN trunks and call routing for the AudioCodes Mediant 5000 Media Gateway. This configuration will enable the AudioCodes Mediant 5000 Media Gateway to interoperate with both the Avaya Meeting Exchange S6800 Conferencing Server and the PSTN (see **Section 1**, **Figure 1**).

Configuration is performed using the EMS client GUI-based provisioning system, which is supported by the Microsoft Operating System. It is assumed that the AudioCodes Mediant 5000 Media Gateway, EMS server, and EMS client have already been installed (see **Section 8**, **Reference 4** and **Reference 5**).

Step	Description
4.1	Invoke the GUI provisioning system from a PC running the EMS client by double-clicking on the desktop icon as shown below. 

Step	Description
4.2	From the login screen that is displayed, enter the login, password and the IP address of the EMS server. 

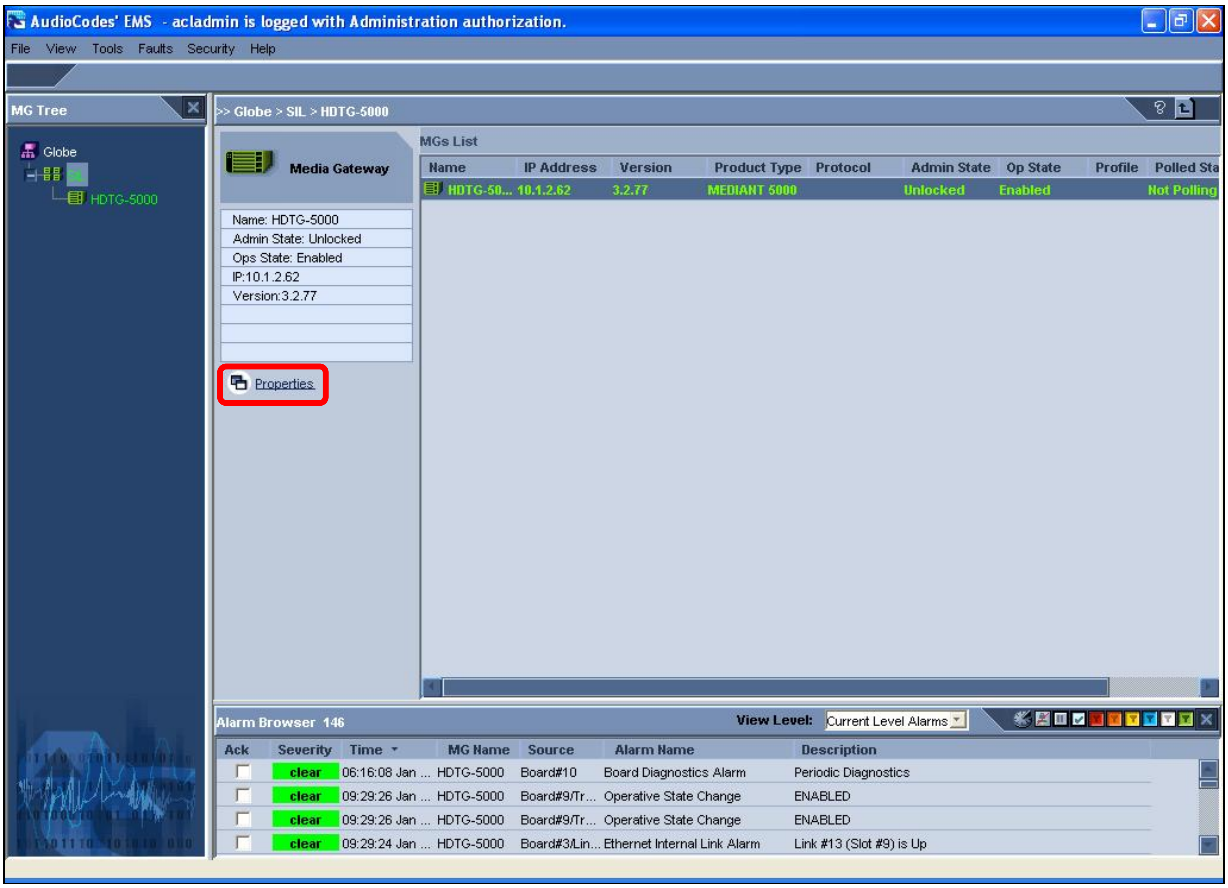
Step	Description
4.3	From the main GUI provisioning screen that is displayed, locate the **Regions List** pane where logical/geographical regions are presented. Double-click on the appropriate row entry. ***Note**: Media gateways, including AudioCodes Mediant 5000 Media Gateways reside in logical/geographical regions. The*  *icon shown on the right side of the screen can be clicked recursively to navigate from this screen or any successive screen back to a previous screen.* 

4.1. Configure the AudioCodes Mediant 5000 Media Gateway Properties

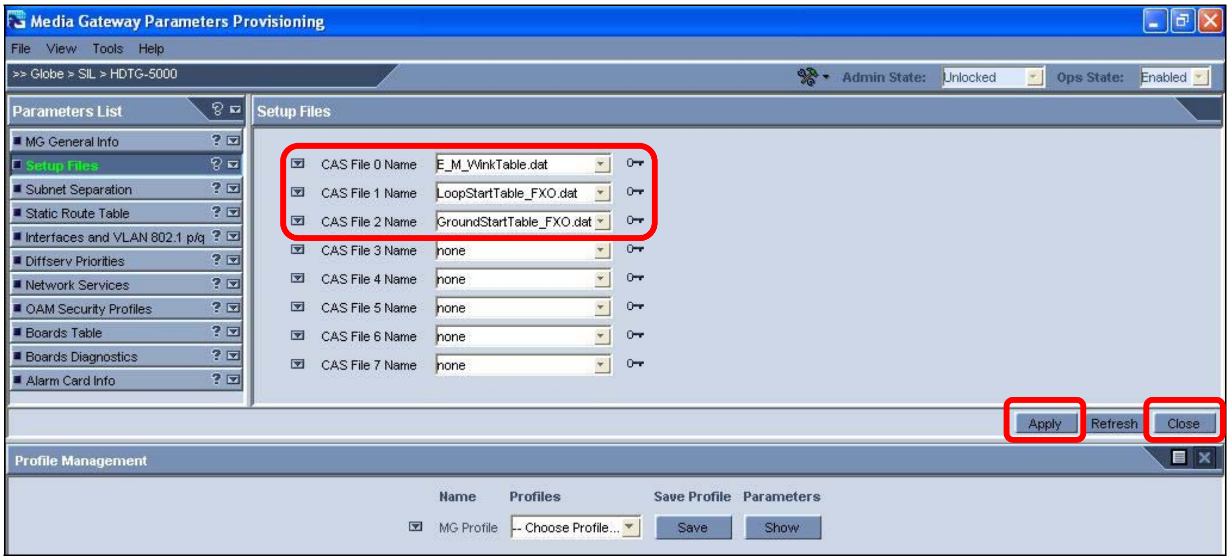
The following steps describe the administrative procedures for configuring system-wide parameters on the AudioCodes Mediant 5000 Media Gateway.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.4</td><td>
From the media gateway list in the MGs List pane that is displayed:

- Select the entry corresponding to the AudioCodes Mediant 5000 Media Gateway to be configured.
- Click on **Properties**.



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

Step	Description
4.5	From the **Media Gateway Parameters Provisioning** window that is displayed, administer CAS signaling files to enable T1 CAS connectivity to the PSTN as follows: • Click on **Setup Files** under **Parameters List**. • Select the appropriate **CAS File**(s) to enable interoperability with the PSTN. *__Note__: Up to eight files are supported on the AudioCodes Mediant 5000 Media Gateway*. • Click on **Apply** and then **Close**. 

4.2. Configure the TP6310 Board

The following steps describe the administrative procedures for configuring the active TP6310 board in the AudioCodes Mediant 5000 Media Gateway chassis. These procedures will administer settings for SIP and DS3 trunking, as well as the call routing rules associated with this TP6310 board to enable signaling/media connectivity between the Avaya Meeting Exchange S6800 Conferencing Server and the PSTN.

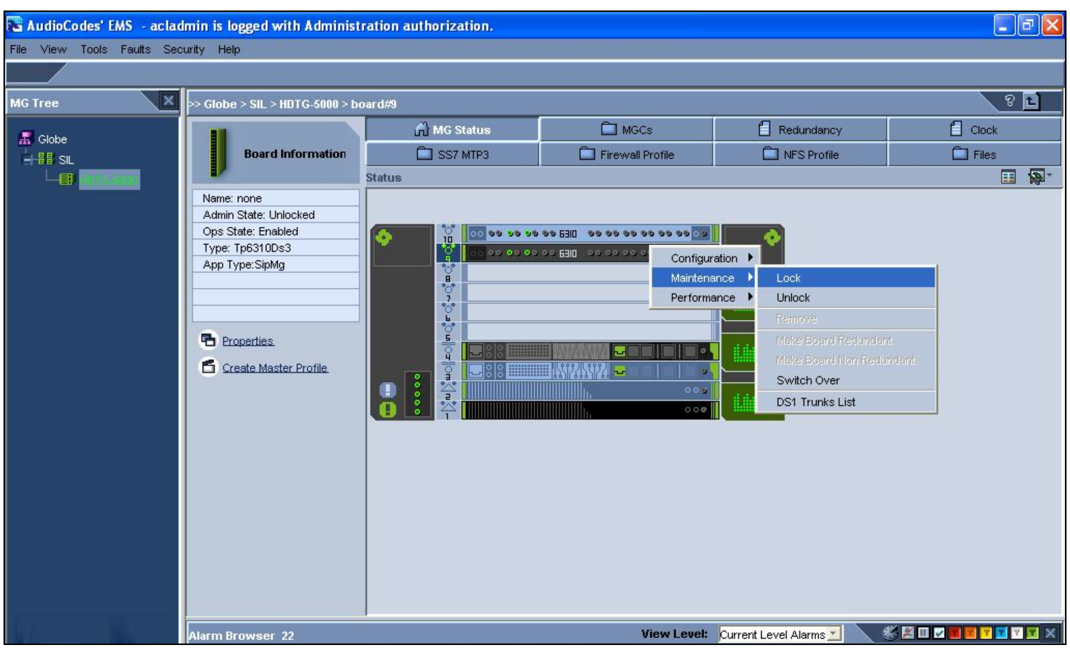
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.6</td><td>

Prior to making any change to the configuration of the TP6310 board, the board must be locked.

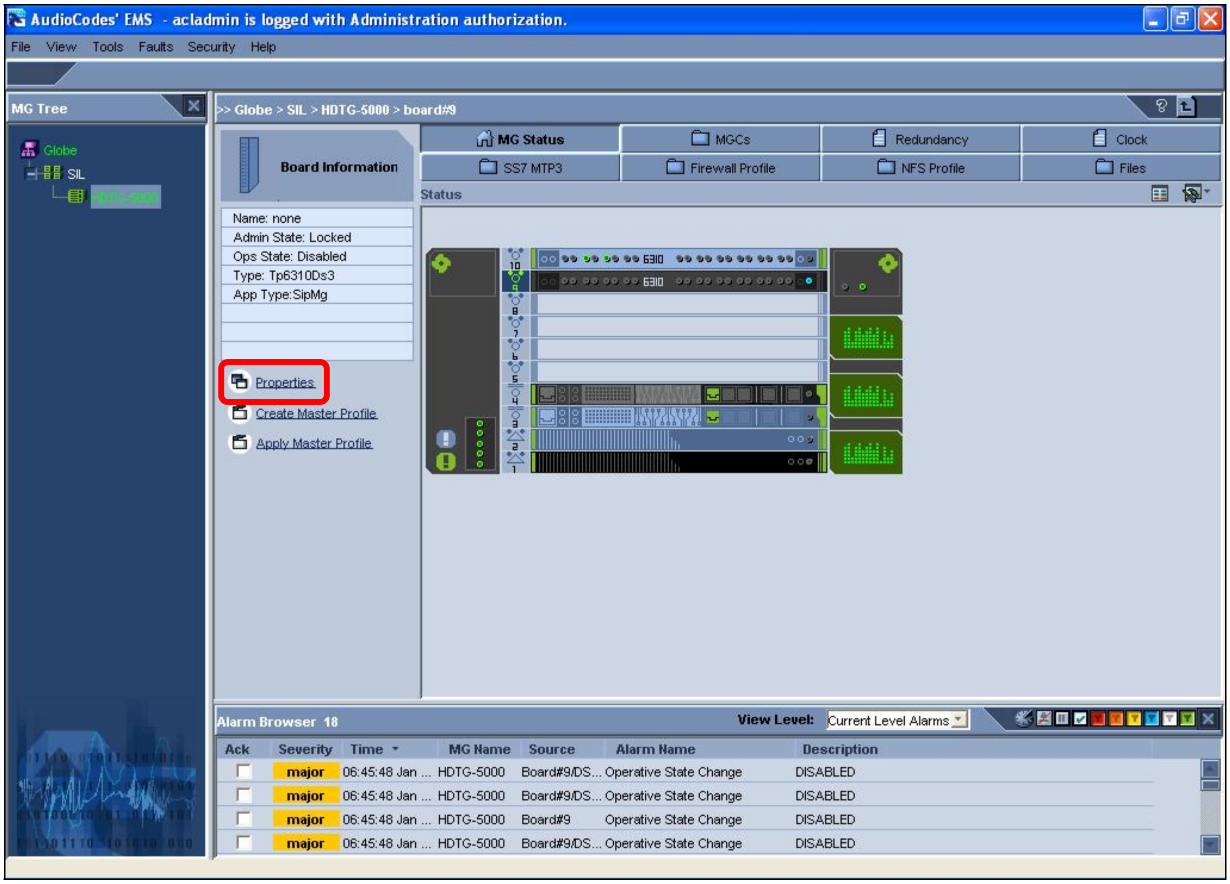
- On the MGs List pane (see **Step 4.4**), double-click on the row corresponding to the AudioCodes Mediant 5000 Media Gateway.

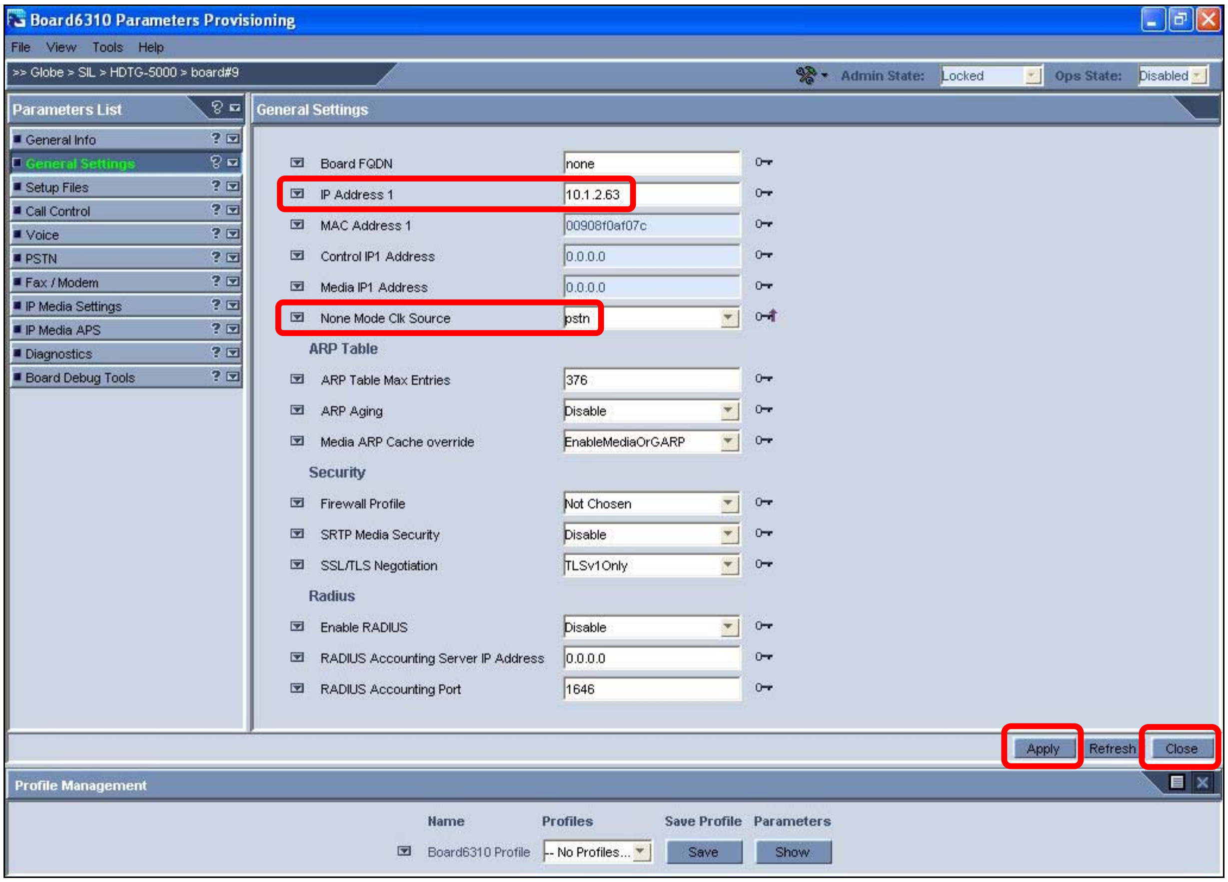
  ***Note**: The **MG Status** tab will be highlighted and the **Status** pane will open, depicting a replica of the front panel of the AudioCodes Mediant 5000 Media Gateway chassis. Board slots are numbered from 1 to 10 from bottom to top on the left side. For these Application Notes, installed boards include: the TP6310 DS3 boards (slots 9 and 10), Ethernet switch boards (slots 3 and 4), and shelf controller boards (slots 1 and 2).*
- Click on the active TP6310 board shown in black, and use mouse button to select **Maintenance ➔ Lock**.
- [***Not Shown***] *To confirm Lock, click **Yes** in the confirmation window that is displayed.*

***Note**: If there is a single TP6310 board in the AudioCodes Mediant 5000 chassis, locking this board removes it from service and is service impacting.*



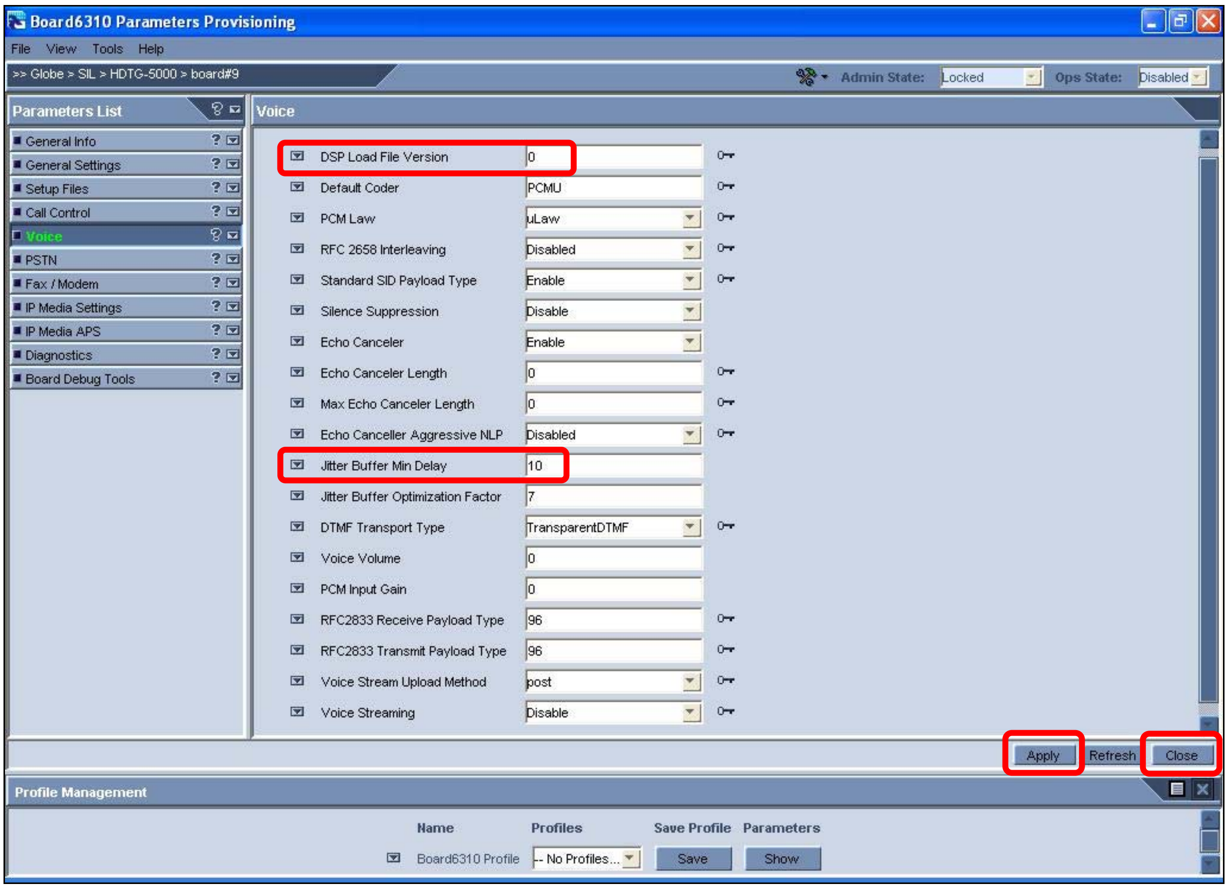
</td></tr>
</table>

Step	Description
4.7	Administer settings on the locked TP6310 board as follows: • Select the locked TP6310 board in the **Status** pane. *__Note__: A locked board is indicated by a blue "locking pin" on its right hand side (see __slot 9__).* • Select the **Properties** link. 

Step	Description
4.8	From the **Board6310 Parameters Provisioning** window that is displayed: • Click on **General Settings** under **Parameters List**. • Set **IP Address 1** to the IP address for this board (see **Section 1**, **Figure 1**). • Select **pstn** for the **None Mode Clk Source**. • Remaining fields are default settings. • Click on **Apply** and then **Close**. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.9</td><td>
From the Board6310 Parameters Provisioning window that is displayed:

Click on Voice under Parameters List.
Set DSP Load File Version to 0.
Set the Jitter Buffer Min Delay to 10 milliseconds.
Note: The jitter buffer is administered to align with the network configuration utilized for these Application Notes, e.g., VoIP traffic will be on an internal enterprise network with low delay characteristics.
Remaining fields are default settings.
Click on Apply and then Close.

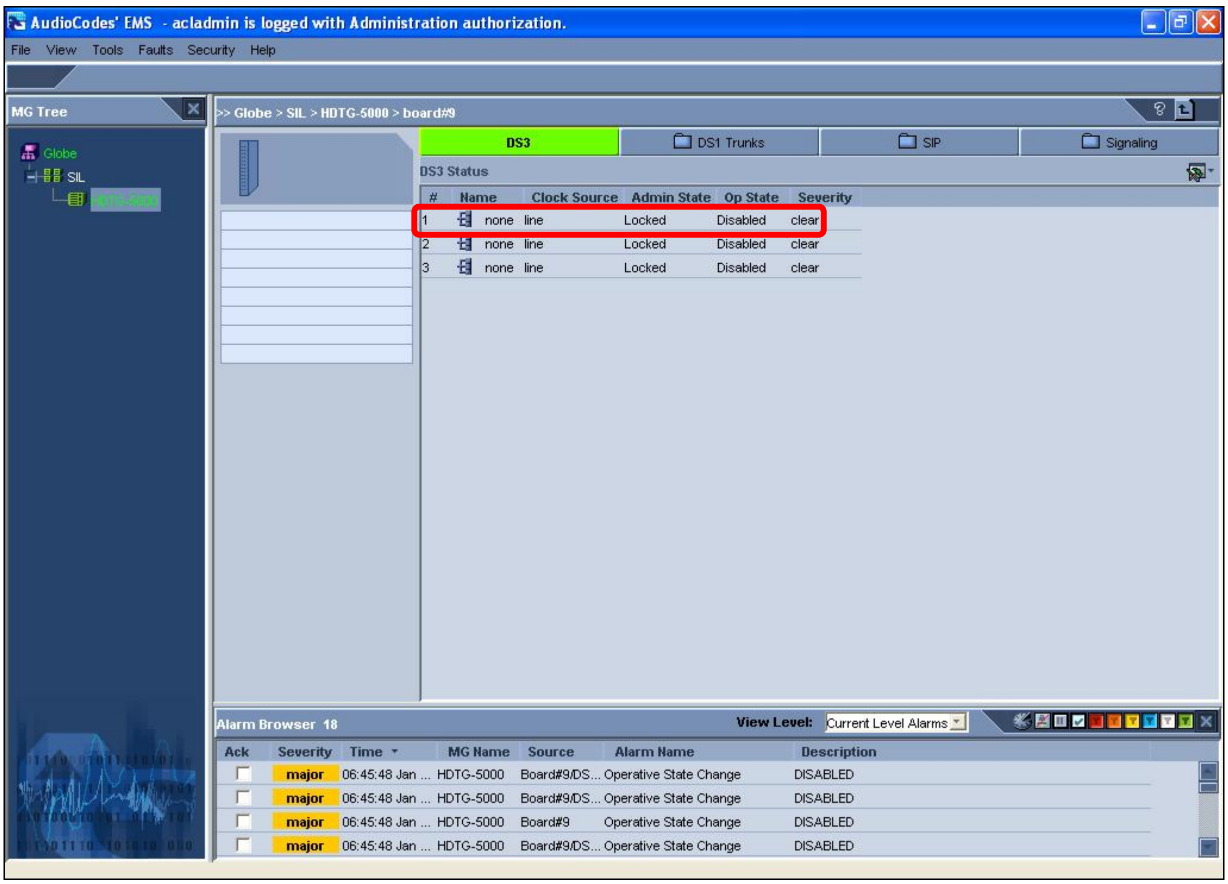


</td></tr>
</table>

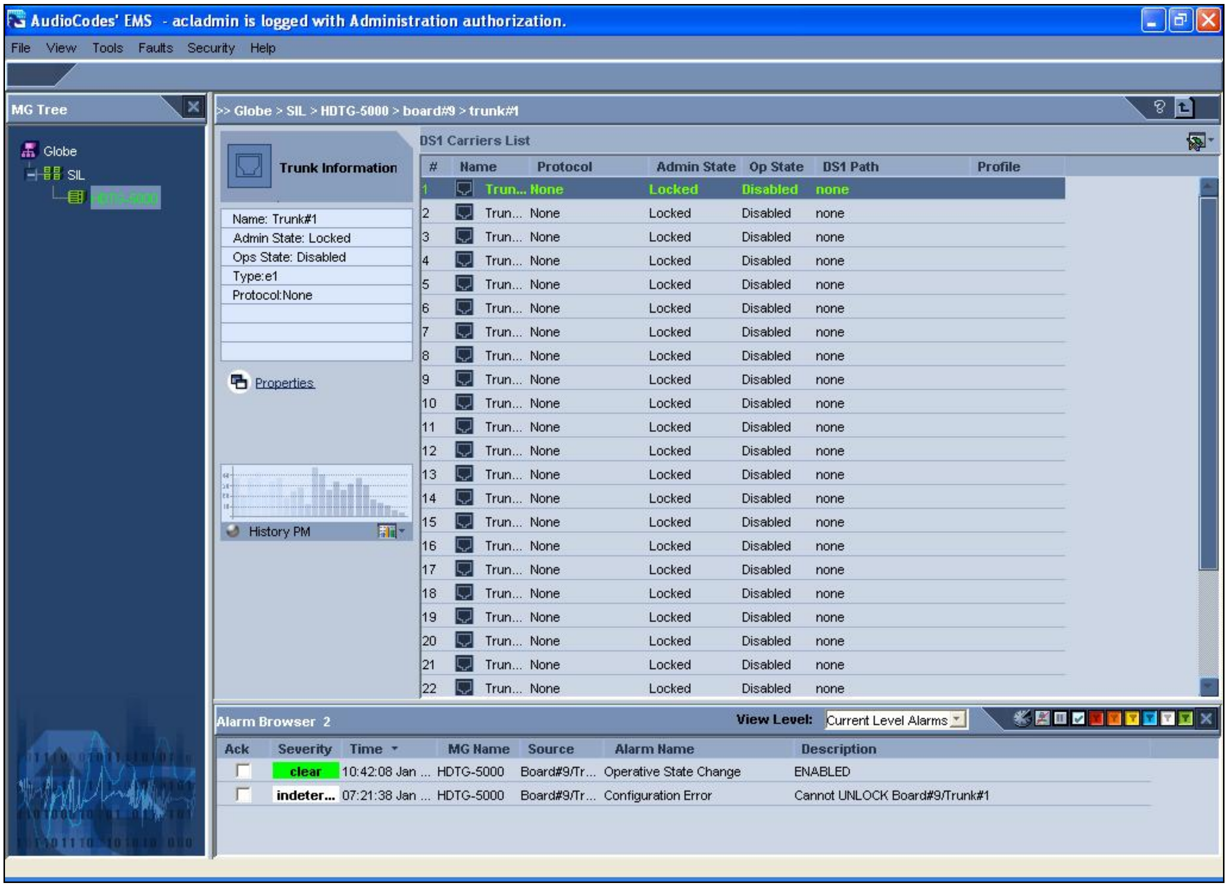
4.3. Configure DS3/DS1 Trunking

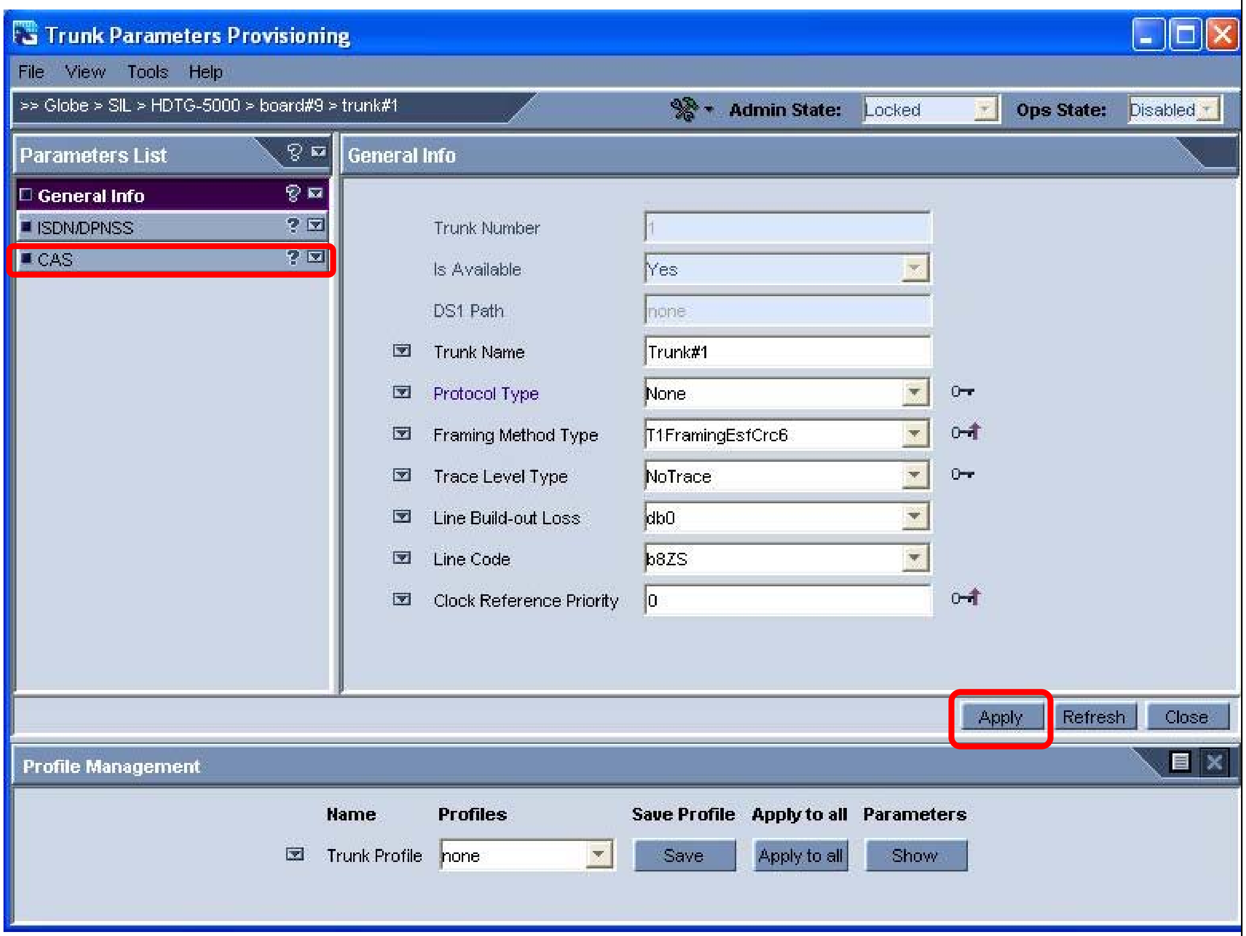
The following steps describe the administrative procedures for configuring the DS3 and constituent DS1 trunking between the AudioCodes Mediant 5000 Media Gateway and the PSTN.

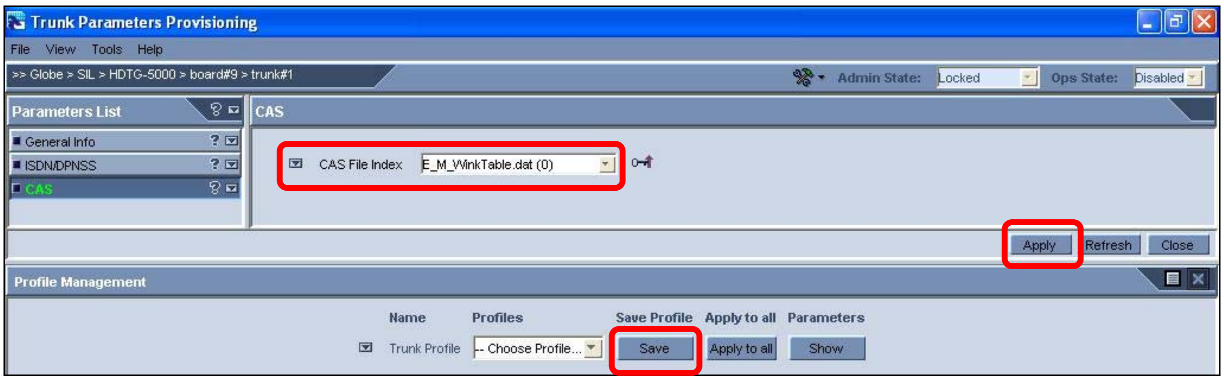
Step	Description
4.10	Administer settings for a DS3 trunk to enable connectivity to the PSTN as follows: • [*Not Shown*] *Double-click on the locked TP6310 board in the Status pane (see **Step 4.7**).* • Click on the **DS3** tab. • From the **DS3 Status** pane that is displayed, double-click on the DS3 for which the DS1 channel interface parameters are to be defined. ***Note****: The **DS3 Status** pane displays the status of each of the 3 DS3 interfaces on this board.*

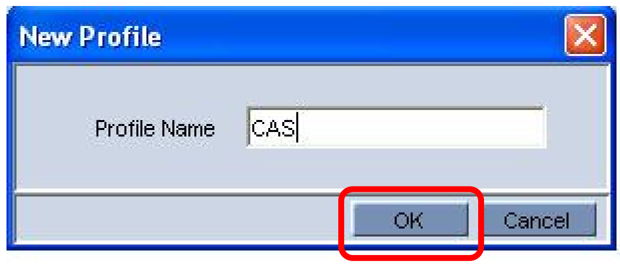


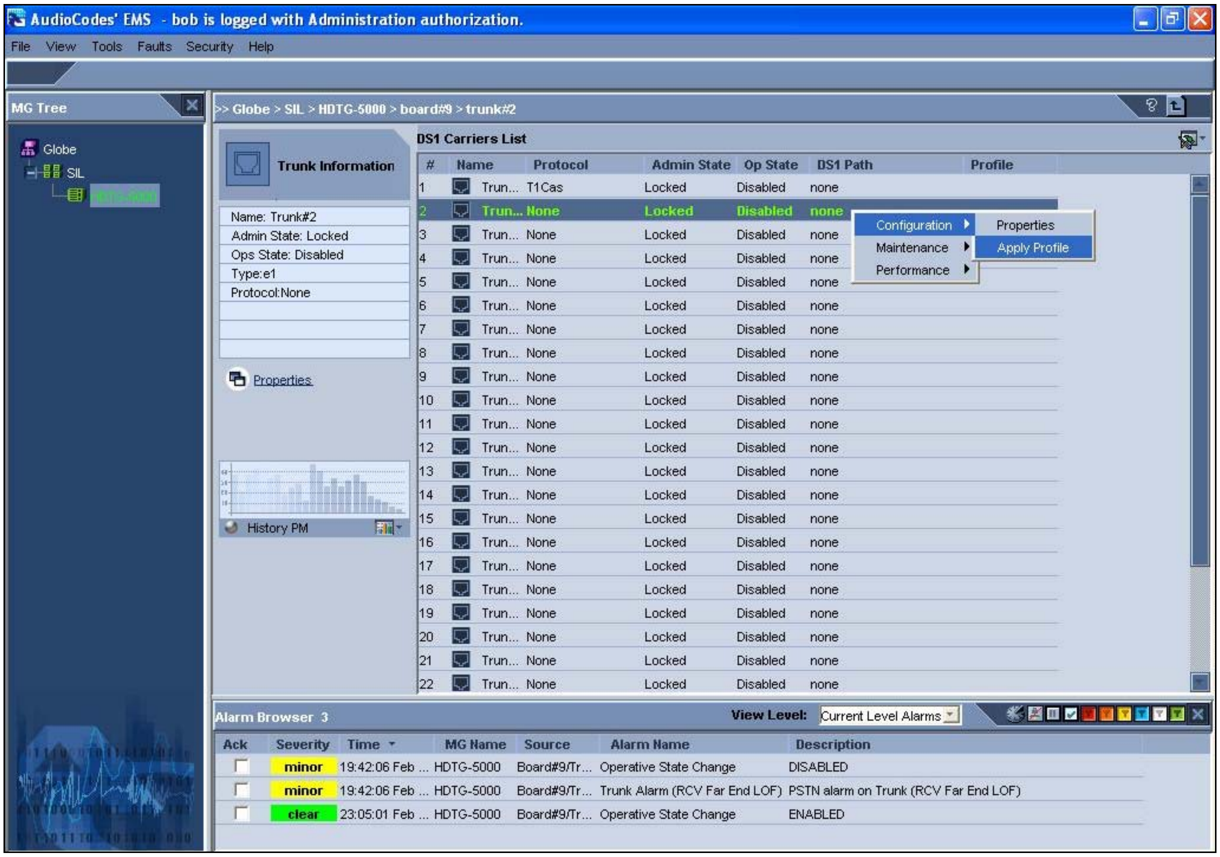
Step	Description
4.11	From the **DS1 Carriers List** pane that is displayed, provision a DS1 on this DS3 interface by double-clicking on its entry in the list.

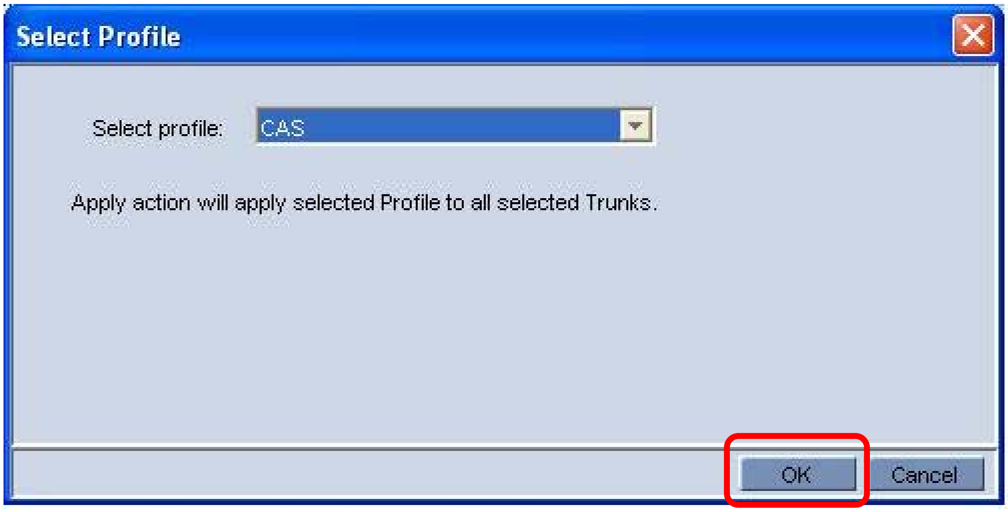
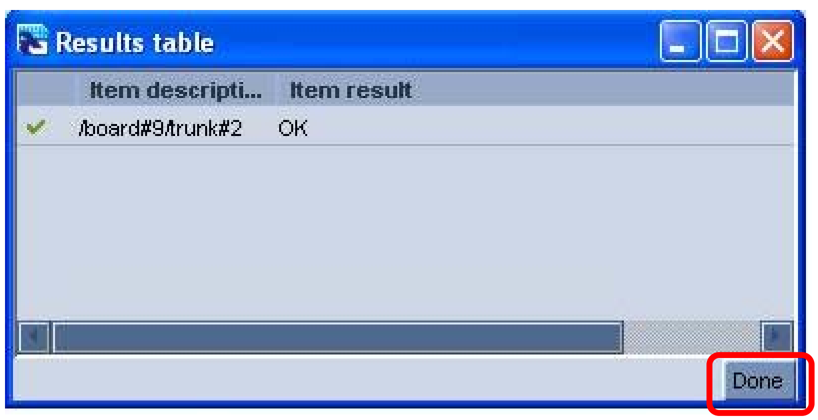


Step	Description
4.12	From the **General Info** pane, in the **Trunk Parameters Provisioning** window that is displayed: • Administer settings to enable connectivity with the PSTN. ***Note**: Obtain configuration details regarding the setting required for this connection to the PSTN from the service provider. The entries for this trunk correspond to a T1 CAS connection between the AudioCodes Mediant 5000 Media Gateway and the PSTN.* • Click on **Apply**. • Click on **CAS** under **Parameters List**. 

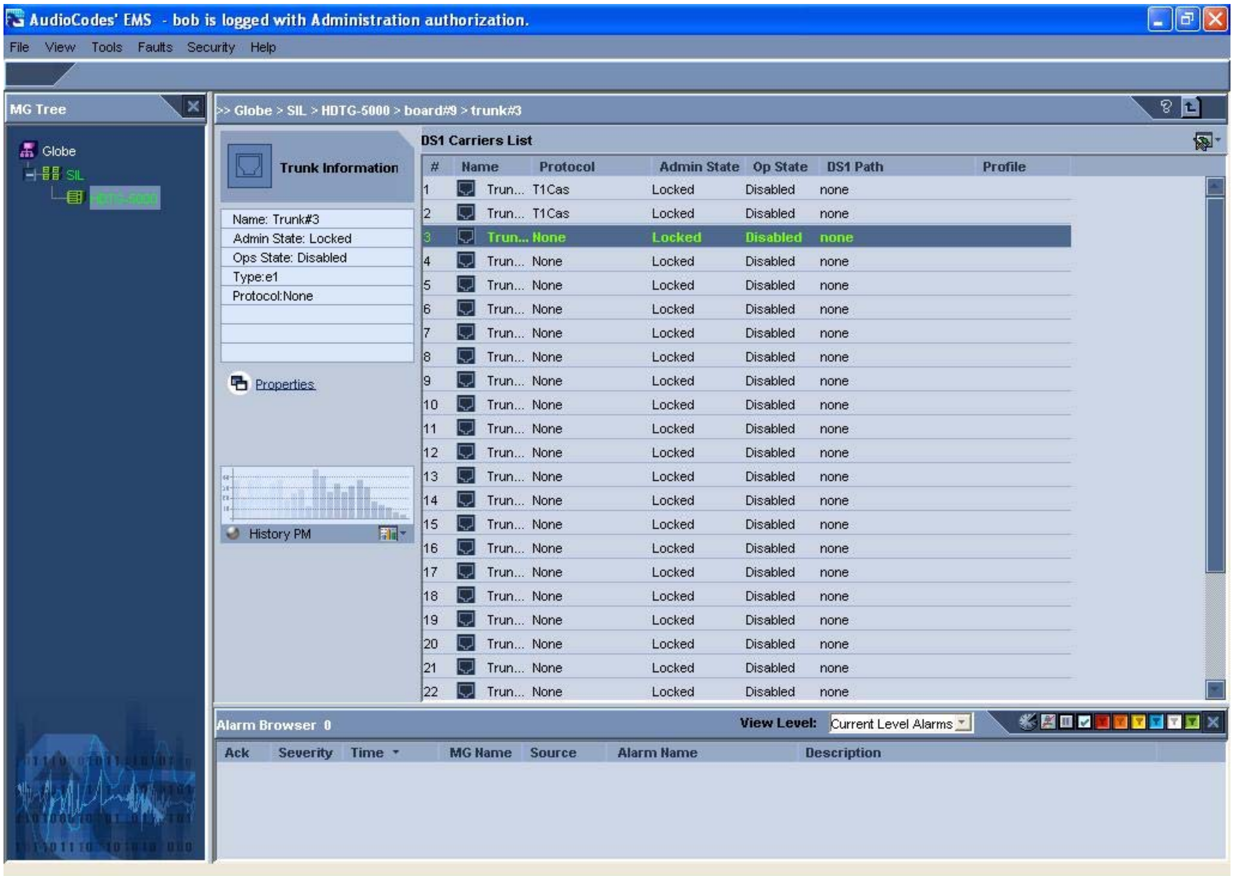
Step	Description
4.13	From the **CAS** pane that is displayed:. • Select a CAS setup file entered in **Step 4.5** that is supported by the PSTN. • Click on **Apply**. • From the **Profile Management** pane, select **Save** to save the DS1 administered in **Step 4.12** and **Step 4.13**. *__Note__: The Profile Management pane can be used to define a configuration profile that can be applied to one or many DS1 interfaces, saving configuration steps and also reducing the chance for data entry error.* 

Step	Description
4.14	In the **New Profile** dialogue box that is displayed: • Fill in a descriptive name for the profile administered in **Step 4.12** and **Step 4.13**. • Click on **OK**. 

Step	Description
4.15	From the **DS1 Carriers List** pane that is displayed, select a range of DS1 interfaces by clicking on the appropriate entry(s) to which a previously saved profile will be applied and right click on the mouse, selecting **Apply Profile**. 

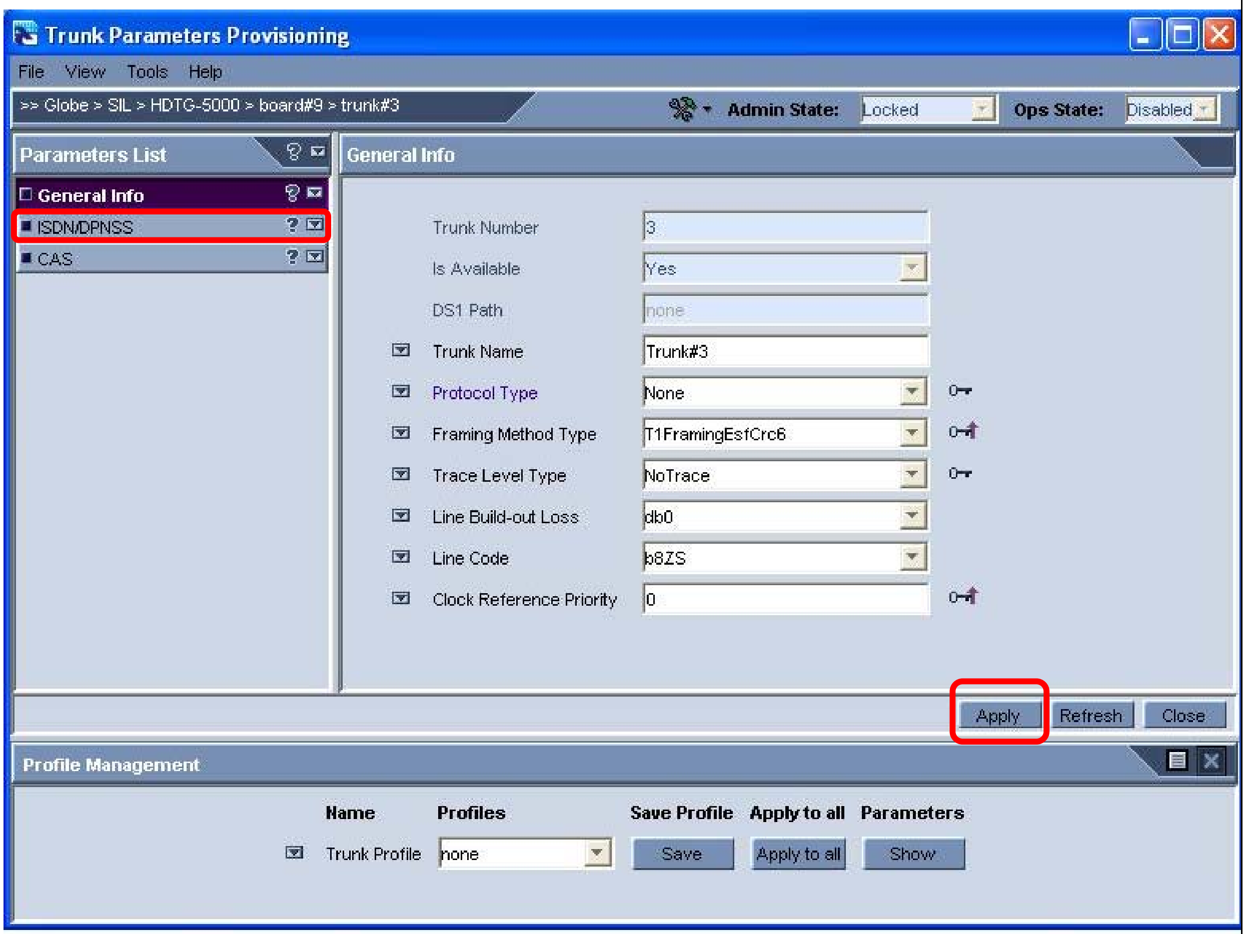
Step	Description
4.16	In the **Select Profile** dialog box that is displayed: • Select the **DS1** profile saved in **Step 4.14**. • Click on **OK**. 
4.17	In the **Results Table** window that is displayed, indicating a successful application of this profile, click on **Done** to close this window. 

Step	Description
4.18	From the **DS1 Carriers List** pane that is displayed, provision the third DS1 on this DS3 interface by double-clicking on its entry in the list.



<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.19</td><td>From the General Info pane, in the Trunk Parameters Provisioning window that is displayed:

Administer settings to enable connectivity with the PSTN.
Note: Obtain configuration details regarding the setting required for this connection to the PSTN from the service provider. The entries for this trunk correspond to a T1 ISDN-PRI connection between the AudioCodes Mediant 5000 Media Gateway and the PSTN.
Click on Apply.
Click on ISDN/DPNSS under Parameters List.



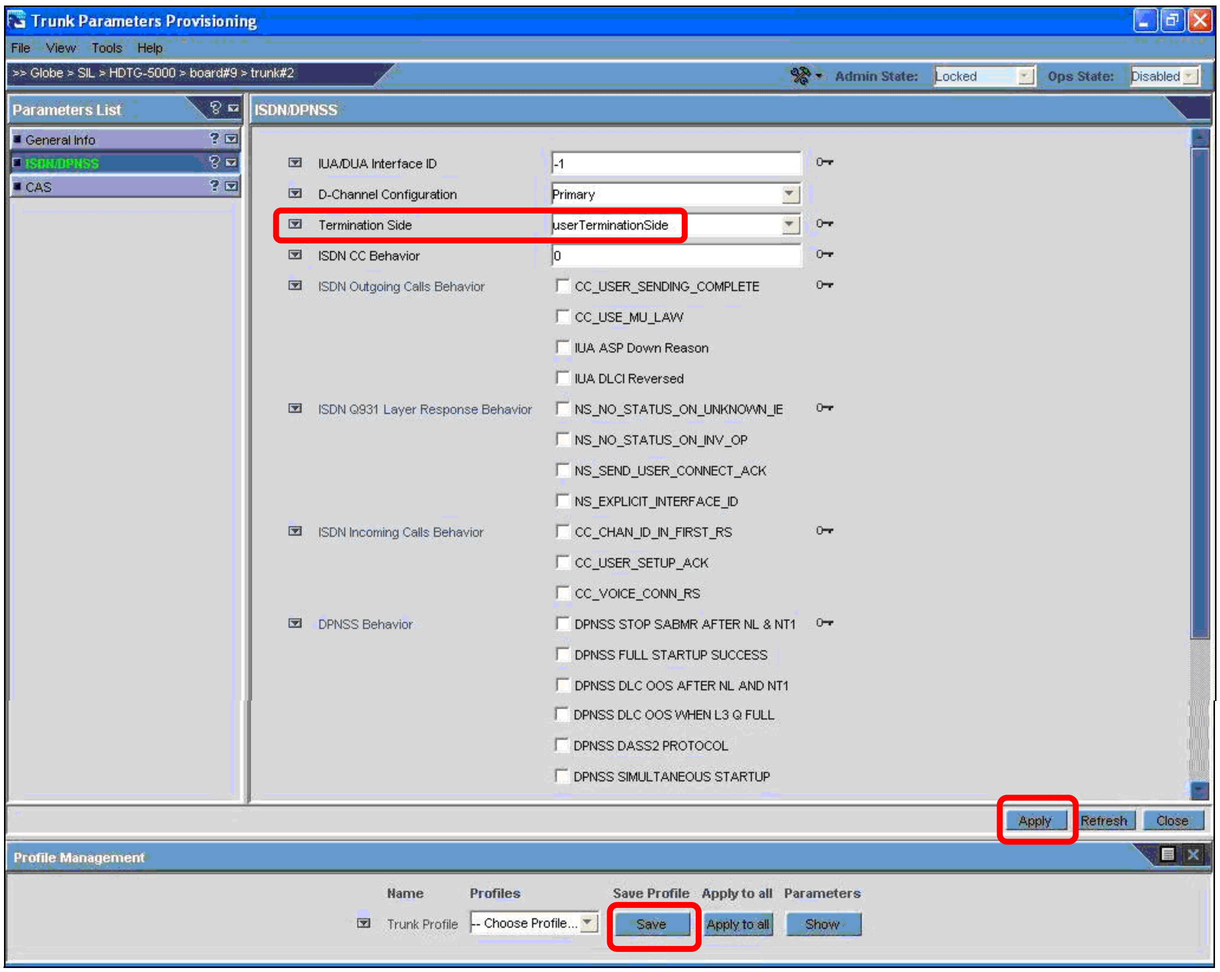
</td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.20</td><td>

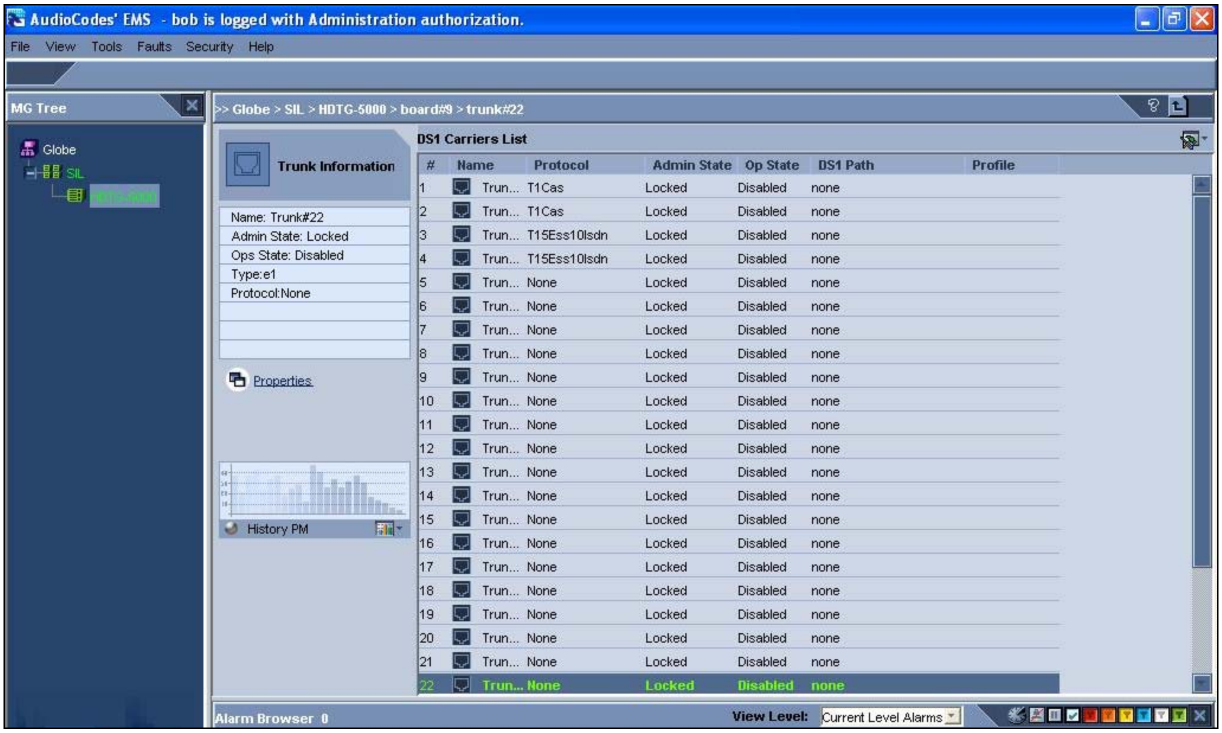
From the **ISDN/DPNSS** pane that is displayed:.

- Select the appropriate value for the **Termination Side**, usually **userTerminationSide** if the PSTN connection is to a service provider.
- Click on **Apply**.
- From the **Profile Management** pane, select **Save** to save the DS1 administered in **Step 4.19** and **Step 4.20** with a descriptive name.

Note: *The Profile Management pane can be used to define a configuration profile that can be applied to many DS1 interfaces, saving configuration steps.*

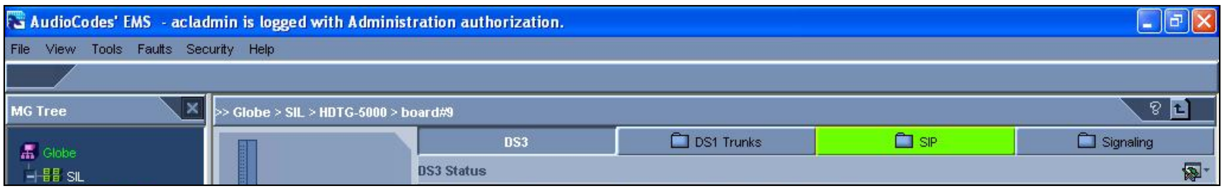
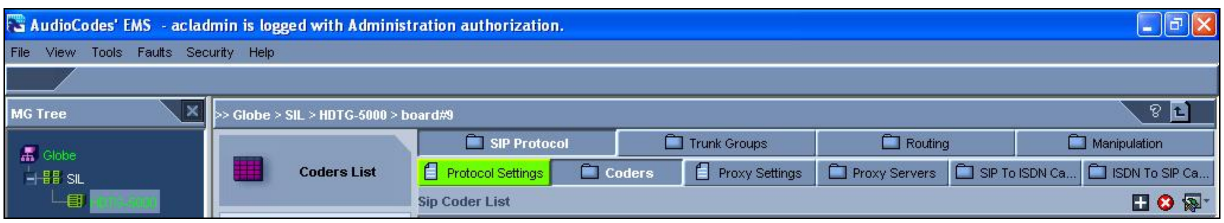


</td></tr>
</table>

Step	Description
4.21	Repeat **Step 4.15** and **Step 4.16** to apply the DS1 configuration saved in **Step 4.20** to the fourth DS1 on this DS3. The resultant **DS1 Carriers List** is shown below. 

4.4. Configure SIP and T1 Trunking

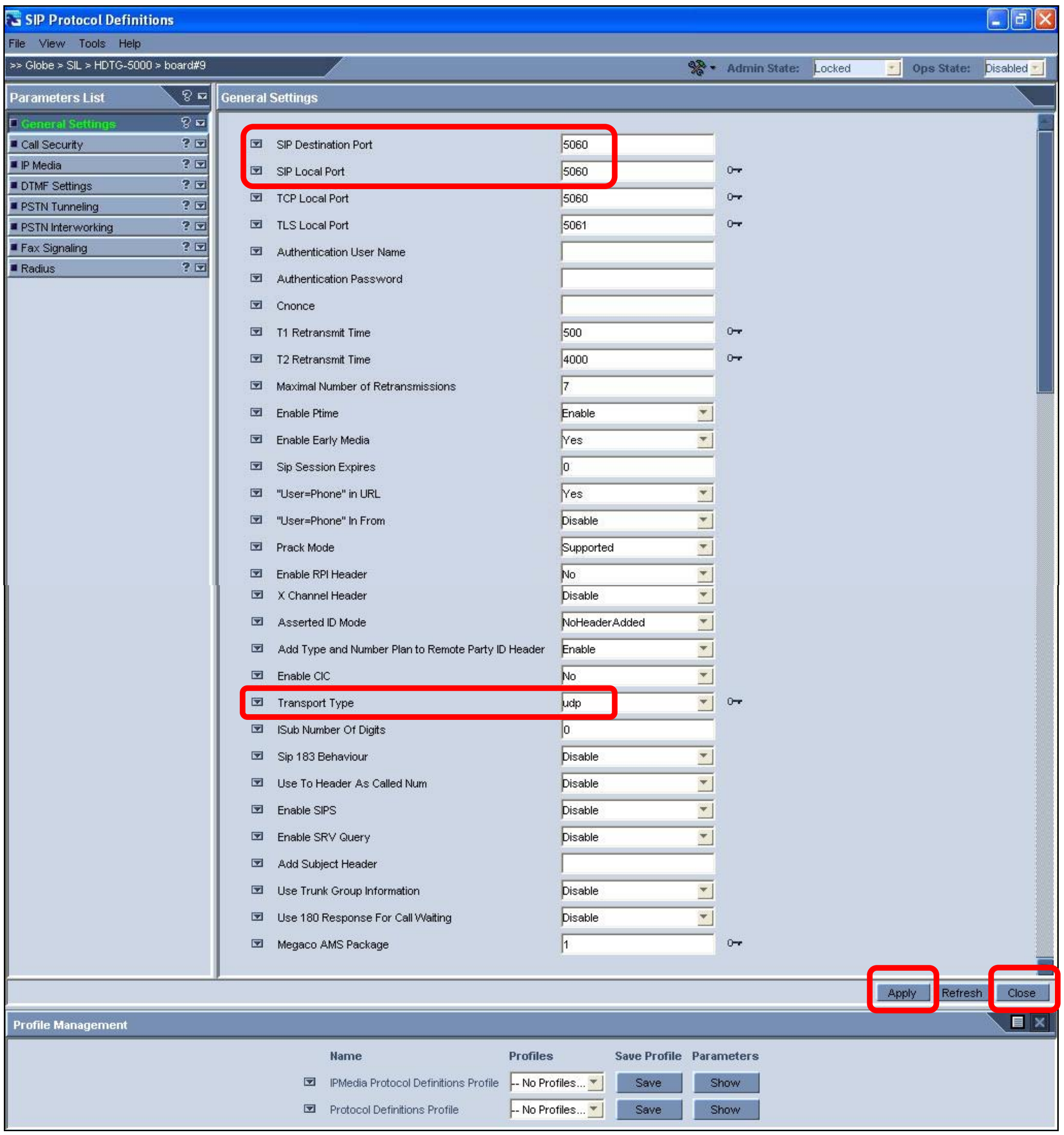
The following steps describe the administrative procedures for configuring SIP and T1 trunking between the AudioCodes Mediant 5000 Media Gateway and the Avaya Meeting Exchange S6200 Application Server.

Step	Description
4.22	Administer settings for SIP trunking to enable connectivity with the Avaya Meeting Exchange S6200 Application Server as follows: • [*Not Shown*] *Click on the* *icon to navigate back to the screen displayed below*. • Click on the **SIP** tab. 
4.23	Click on the **SIP Protocol** tab; then click on the **Protocol Settings** tab. 

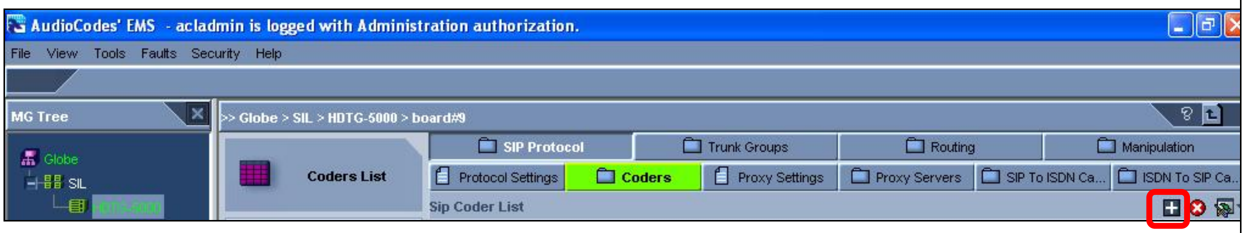
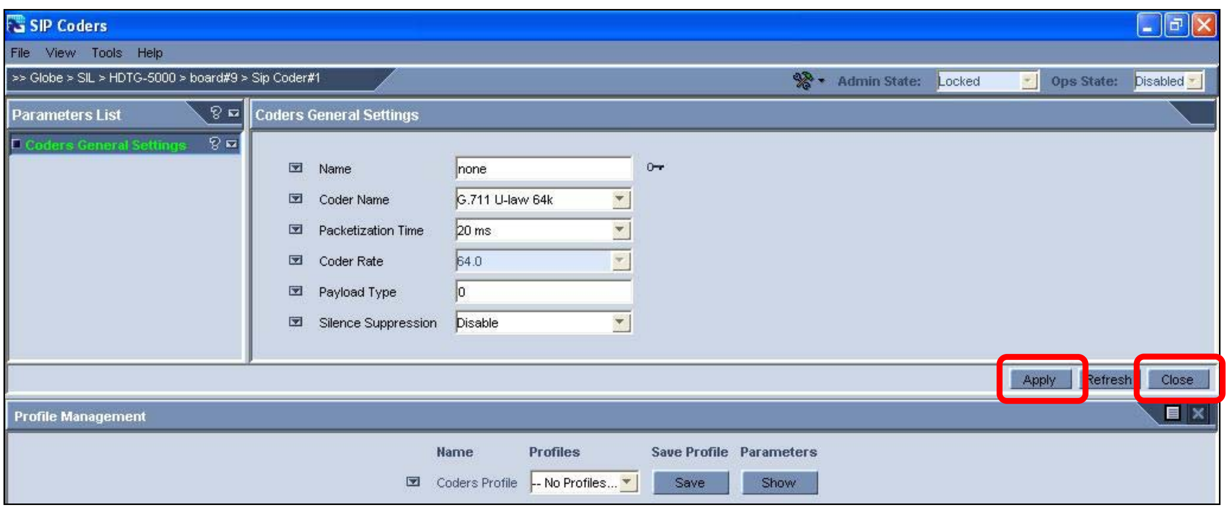
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.24</td><td>

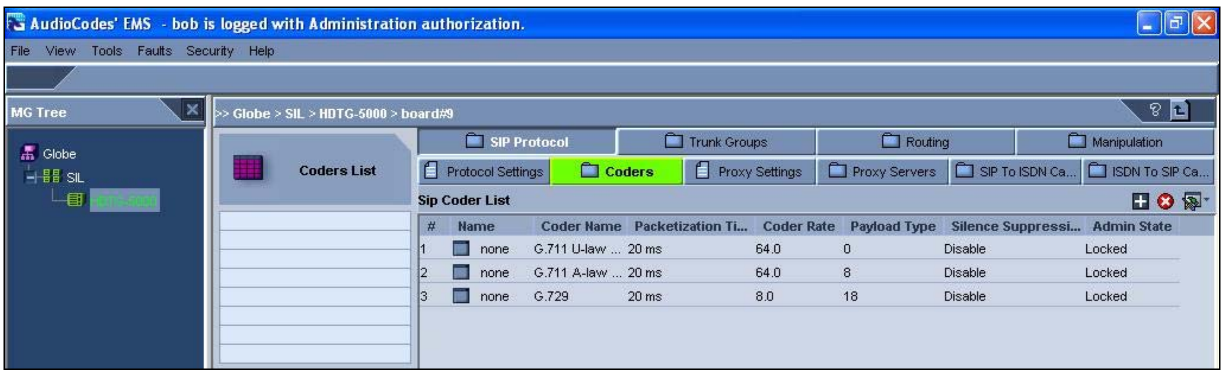
From the **General Settings** pane, in the **SIP Protocol Definitions** window that is displayed, administer settings to enable SIP connectivity with the Avaya Meeting Exchange S6200 Application Server as follows:

- Set the **SIP Destination Port**, **SIP Local Port** and **Transport Type** to enable SIP/UDP connectivity with the Avaya Meeting Exchange S6200 Application Server (see **Step 3.2** and **Step 3.4**).
- Remaining fields are default settings.
- Click on **Apply** and then **Close**.



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

Step	Description
4.25	Click on the **Coders** tab, under **SIP Protocol** to administer the codec preferences for this SIP trunk between the AudioCodes Mediant 5000 Media Gateway and the Avaya Meeting Exchange S6200 Application Server. From the **Sip Coder List** pane that is displayed, click on the icon to add codec(s), ordered sequentially from most to least preferred. 
4.26	From the **Coders General Settings** pane, in the **SIP Coders** window that is displayed: • Add a codec that is supported on the Convedia CMS-6000 Media Server (see **Step 3.16**). In this case, administer settings for **G.711 U-law 64k**. • Remaining fields are default settings. • Click on **Apply** and then **Close**. 

Step	Description
4.27	Repeat **Step 4.25** and **Step 4.26** to add a codec that is supported on the Convedia CMS-6000 Media Server (see **Step 3.16**) with the following parameters: • Administer settings for **G.711 A-law 64k**. • Remaining fields are default settings.
4.28	Repeat **Step 4.25** and **Step 4.26** to add a codec that is supported on the Convedia CMS-6000 Media Server (see **Step 3.16**) with the following parameters: • Administer settings for **G.729**. • Remaining fields are default settings. The resultant **Sip Coder List** is shown below. 

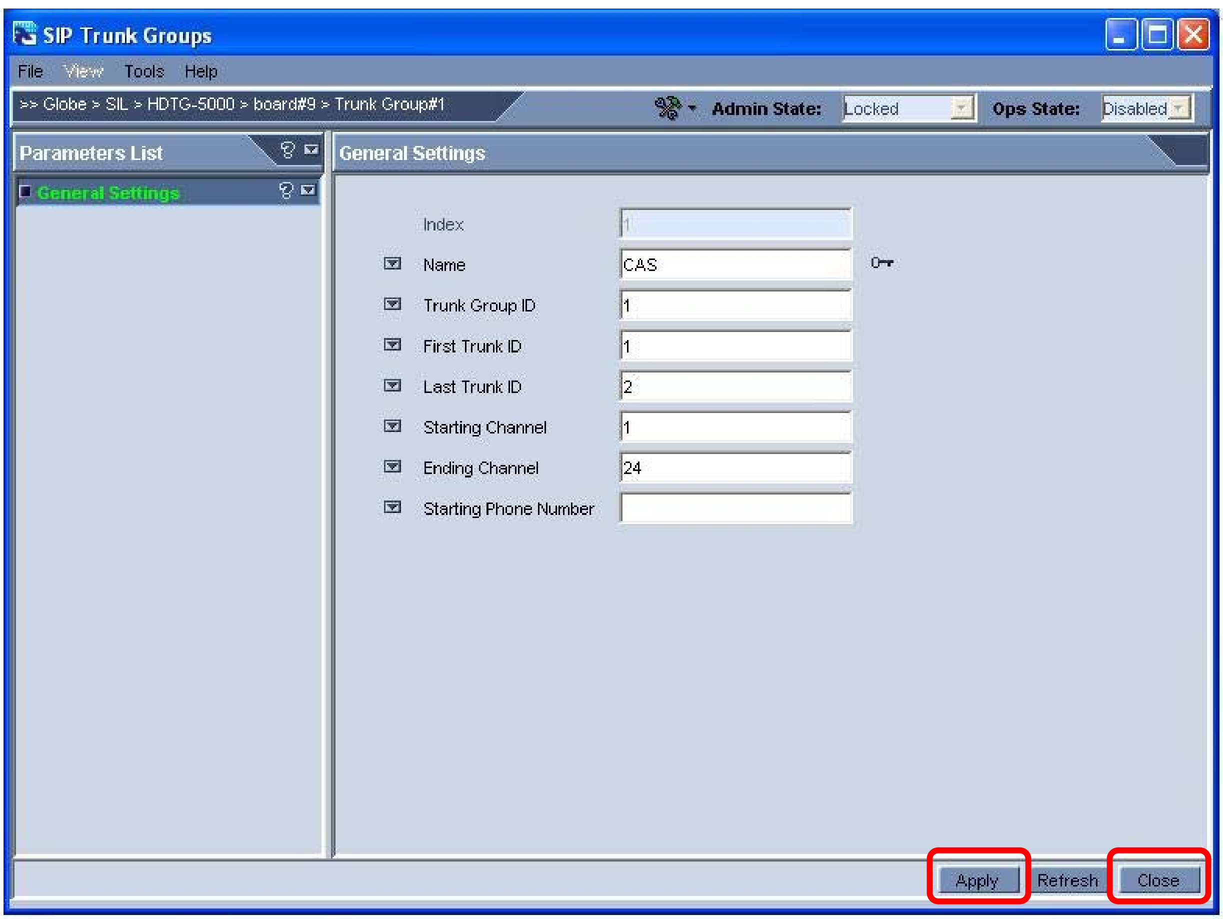
4.5. Configure B-channels

The following steps describe the administrative procedures for assigning profiles to B-channels. These profiles are logical entities referred to as trunk group(s) that are used for routing IP to telephone calls with common rules, e.g., methods in which new calls are assigned to B-channels within each trunk group.

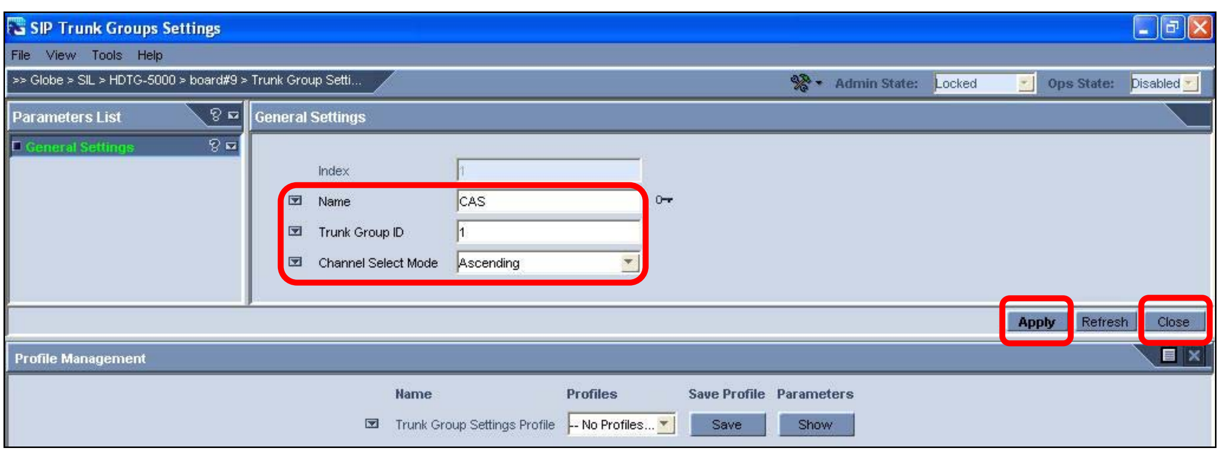
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.29</td><td>Administer settings to assign profiles to the AudioCodes Mediant 5000 Media Gateway’s T1 B-channels as follows:

- Click on the **Trunk Groups** tab.
- Click on the **Trunk Group** tab.
- From the **Sip Trunk Group List** pane that is displayed, click on the icon to add trunk group(s).

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

Step	Description
4.30	From the **SIP Trunk Groups** window that is displayed, administer settings for CAS trunking between the AudioCodes Mediant 5000 Media Gateway and the PSTN as follows: • Enter a descriptive label in the **Name** field. • Set the **Trunk Group ID** to **1**. • Set the **First Trunk ID** to **1** (first T1 in the first T3) and the **Last Trunk ID** to **2**; thus, logically provisioning this trunk with 48 B-channels. *Note: Logically provisioning more B-channels than are carried in a single DS1 enables redundancy over multiple DS1 interfaces.* • Set the **Starting Channel** to **1** (first B-channel in each T1) and **Ending Channel** to **24** (last B-channel in each T1). • The **Starting Phone Number** field is optional. The logical numbers defined in this field are used when an incoming PSTN/PBX call doesn't contain the calling number or called number. In this case, the entry in the **Starting Phone Number** field is used to replace them. • Click on **Apply** and then **Close**. 

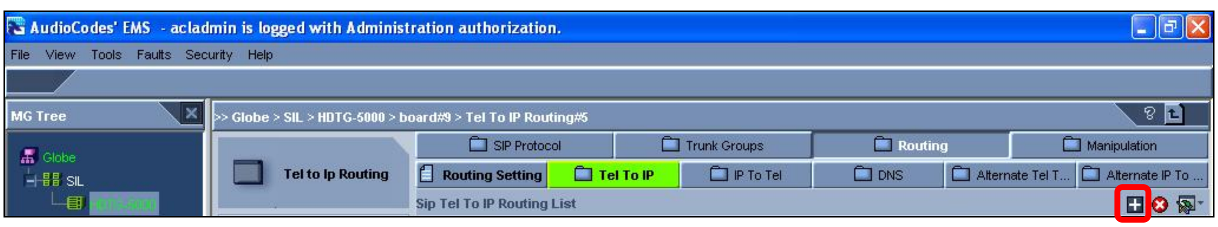
Step	Description
4.31	Repeat **Step 4.29** and **Step 4.30** to administer settings for ISDN-PRI trunking between the AudioCodes Mediant 5000 Media Gateway and the PSTN with the following parameters: • Enter **PRI** in the **Name** field. • Set the **Trunk Group ID** to **2**. • Set the **First Trunk ID** to **3** (first T1 in the first T3) and the **Last Trunk ID** to **4**; thus, logically provisioning this trunk with 46 B-channels. • Set the **Starting Channel** to **1** (first B-channel in each T1) and **Ending Channel** to **23** (last B-channel in each T1). • Leave the **Starting Phone Number** field blank. The resultant **Sip Trunk Group List** is shown below. 
4.32	Administer settings that are used to determine the method in which new calls are assigned to B-channels within each trunk group as follows: • Click on the **Trunk Groups** tab. • Click on the **Trunk Group Settings** tab. • From the **Sip Trunk Group Settings List** pane that is displayed, click on the ⊞ icon to add trunk group setting(s). 

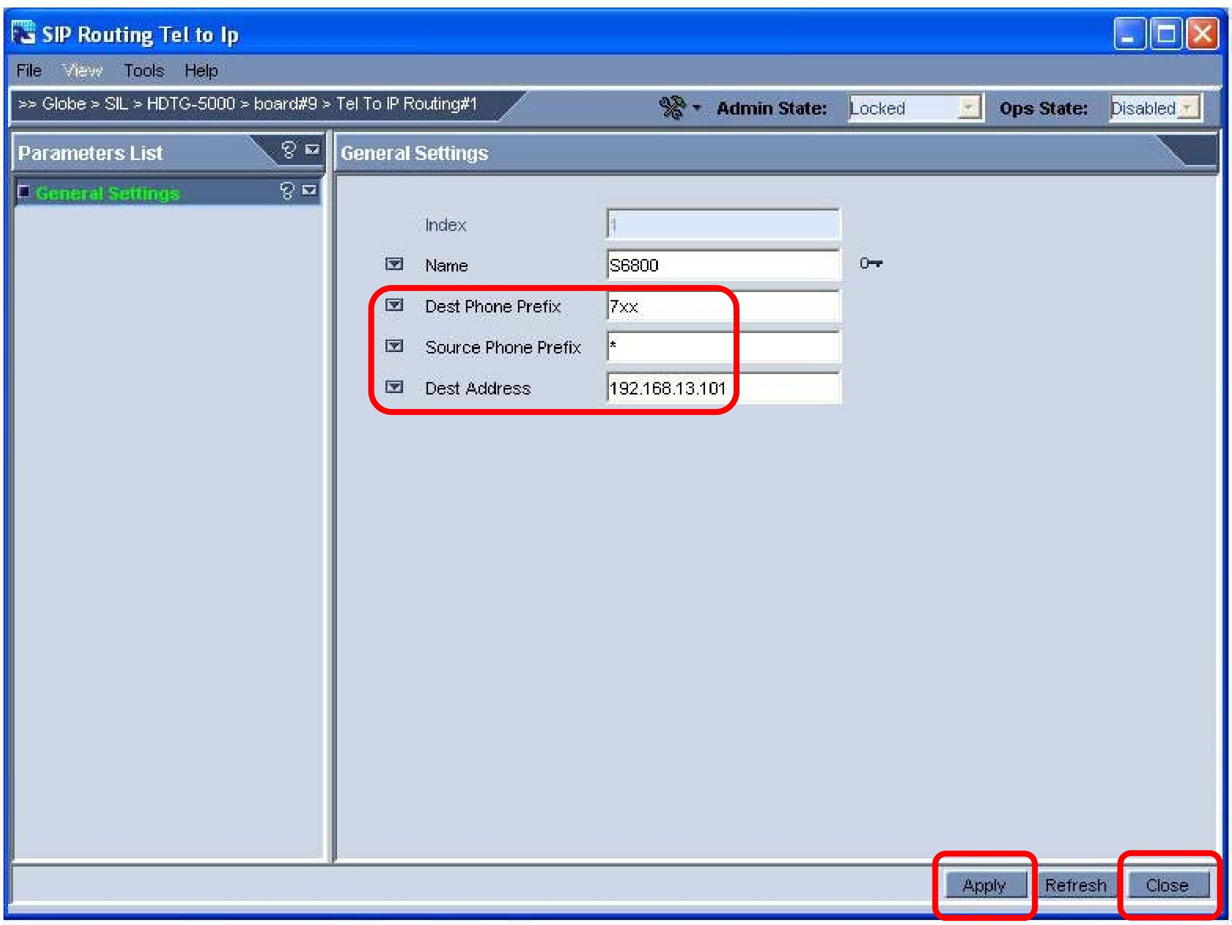
Step	Description
4.33	In the SIP Trunk Groups Settings window that is displayed, administer settings to determine the method in which new calls are assigned to B-channels within the CAS trunk group provisioned is **Step 4.30** as follows: • Enter a descriptive label in the **Name** field. • Set the **Trunk Group ID** to correspond to the Trunk Group ID assigned to the trunk provisioned in **Step 4.30**. • Set the **Channel Select Mode** to determine the method in which new calls are assigned to B-channels within a trunk group. For these Application Notes, this trunk group is administered to select B-channels in **Ascending** mode, while the PSTN selects B-channels in a descending fashion. *__Note__: To reduce the probability of glare (glare occurs when both sides of a trunk group select the same B-channel for call initiation) on this trunk, the network needs to be administered so both sides of the interface select B-channels from opposite ends of the trunk group. This is called linear hunting, ascending or descending. For example, on a 24B (or 23B+D for ISDN-PRI) trunk group, the user side could be administered to select B-channels starting at channel 1 (ascending) while the network side (PSTN) would be administered to start selecting B-channels at channel 24 (or 23 for ISDN-PRI).* • Click on **Apply** and then **Close**. *__Note__: This channel selection pattern, in combination with the logical trunk provisioning in __Step 4.30__ enable ascending channel selection over 48 B-channels spread over two physical DS1 connections between the AudioCodes Mediant 5000 Media Gateway and the PSTN. Thus, if one DS1 goes out of service, service will not be impacted for call origination from the AudioCodes Mediant 5000 Media Gateway.* 

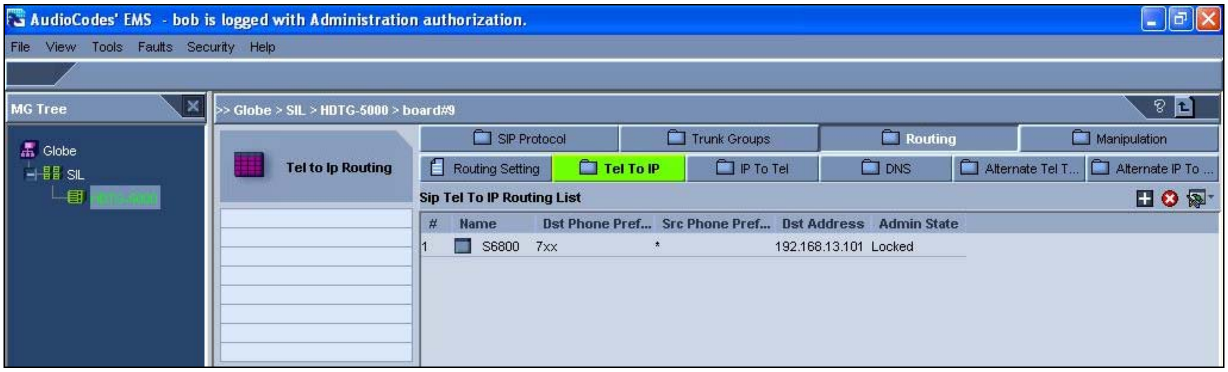
Step	Description
4.34	Repeat **Step 4.32** and **Step 4.33** to administer settings to determine the method in which new calls are assigned to B-channels within the ISDN-PRI trunk group provisioned in **Step 4.31** with the following parameters: • Enter **PRI** in the **Name** field. • Set the **Trunk Group ID** to correspond to the Trunk Group ID assigned to the trunk provisioned in **Step 4.31**. • Set the **Channel Select Mode** to **Ascending**. ***Note****: This channel selection pattern, in combination with the logical trunk provisioning in* ***Step 4.31*** *enable ascending channel selection over 46 B-channels spread over two physical DS1 connections between the AudioCodes Mediant 5000 Media Gateway and the PSTN. Thus, if one DS1 goes out of service, service will not be impacted for call origination from the AudioCodes Mediant 5000 Media Gateway.* The resultant **Sip Trunk Group Settings List** is shown below. 

4.6. Administer Call Routing Rules

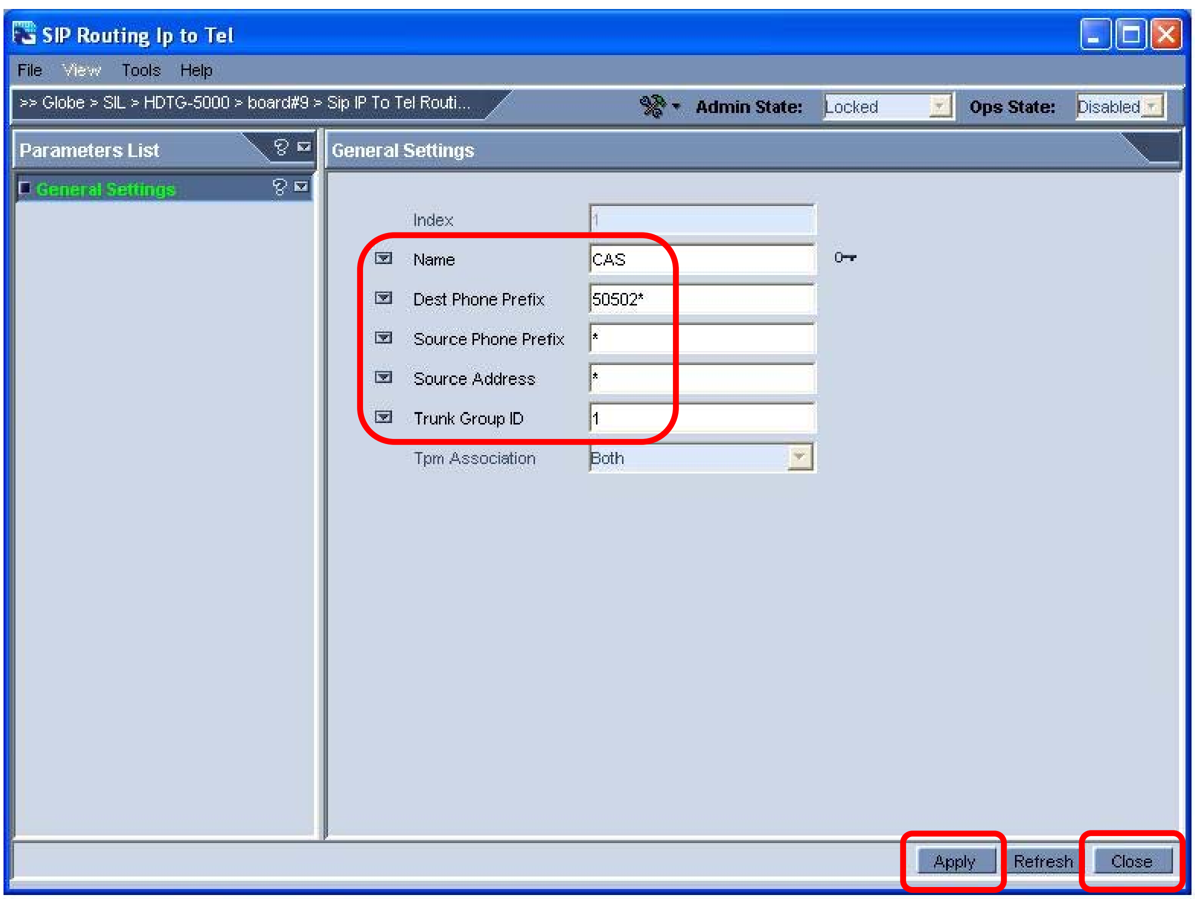
The following steps describe the administrative procedures for administering call routing rules on the AudioCodes Mediant 5000 Media Gateway to enable call origination/termination between the Avaya Meeting Exchange S6800 Conferencing Server and the PSTN.

Step	Description
4.35	Administer call routing rule(s) that are applied to calls originating from the PSTN to the Avaya Meeting Exchange S6800 Conferencing Server by adding Tel To IP routing rule(s) to as follows: • Click on the **Routing** tab. • Click on the **Tel To IP** tab. • From the **Sip Tel To IP Routing List** pane that is displayed, click on the  icon to add routing rule(s). ***Note**: The **Tel To IP** routing table is used to route incoming Tel calls from the PSTN to IP addresses. This routing table associates a called/calling telephone number's prefix with a destination IP address or with an FQDN (Fully Qualified Domain Name). When a call is routed through the AudioCodes Mediant 5000 Media Gateway, the called and calling numbers are compared to the list of prefixes on the IP Routing Table (up to 50 prefixes can be configured). Calls that match these prefixes are sent to the corresponding IP address or FQDN. If the number dialed does not match these prefixes, the call is not made.* 

Step	Description
4.36	From the **SIP Routing Tel to IP** window that is displayed, administer settings to enable Dial-In to the Avaya Meeting Exchange S6800 Conferencing Server from the PSTN as follows: • Enter a descriptive label in the **Name** field. • Enter a rule in the **Dest Phone Prefix** field that matches the pattern of incoming calls to the Avaya Meeting Exchange S6800 Conferencing Server from the PSTN. For these Application Notes, all calls to the Avaya Meeting Exchange S6800 Conferencing Server from the PSTN are three digits in length with a leading digit of **7**. The rule **7xx** is utilized, where **x** is a wildcard and will match any single digit. • Enter an ***** in the **Source Phone Prefix** field to allow routing for any source telephone number Dialing-In to the Avaya Meeting Exchange S6800 Conferencing Server from the PSTN. • Enter the IP address of the Avaya Meeting Exchange S6200 Application Server (see **Step 3.2**) in the **Dest Address** field. • Click on **Apply** and then **Close**. 

Step	Description
4.37	Only one Tel To IP routing rule is used for these Application Notes. If more than one rule is required, repeat **Step 4.35** and **Step 4.36** to add additional rule(s). The resultant **Sip Tel To IP Routing List** is shown below. 

Step	Description
4.38	Administer call routing rule(s) that are applied to calls originating from the Avaya Meeting Exchange S6800 Conferencing Server to the PSTN by adding IP To Tel routing rule(s) to as follows: • Click on the **Routing** tab. • Click on the **IP To Tel** tab. • From the **Sip IP To Tel Routing List** pane that is displayed, click on the  icon to add routing rule(s). ***Note****: The* ***IP to Tel*** *routing table is used to route incoming IP calls to groups of B-channels referred to as trunk group(s) provisioned in* ***Steps 4.29 – 4.31****. Calls are assigned to trunk groups according to any combination of the following three options (or using each independently):* • *Destination phone prefix.* • *Source phone prefix.* • *Source IP address.* *The call is then sent to the AudioCodes Mediant 5000 Media Gateway channels assigned to that trunk group. The specific channel, within a trunk group, that is assigned to accept the call is determined according to the trunk group's channel selection mode which is defined in the Trunk Group Settings Table provisioned in* ***Steps 4.32 – 4.34****.* 

Step	Description
4.39	From the **SIP Routing IP to Tel** window that is displayed, administer settings to enable Dial-Out from the Avaya Meeting Exchange S6800 Conferencing Server to the PSTN over a CAS trunk as follows: • Enter a descriptive label in the **Name** field. • Enter a rule in the **Dest Phone Prefix** field that matches the pattern of outgoing calls from the Avaya Meeting Exchange S6800 Conferencing Server to the PSTN. For these Application Notes, all calls from the Avaya Meeting Exchange S6800 Conferencing Server to the PSTN via CAS trunking are placed to telephone numbers with the leading digits 50502. The rule **50502*** is utilized, where ***** is a wildcard and will match any remaining digit(s). • Enter an ***** in the **Source Phone Prefix** and **Source Address** fields to allow routing for any party Dialing-Out from the Avaya Meeting Exchange S6800 Conferencing Server to the PSTN. • Enter the Trunk Group ID for the CAS trunk group provisioned in **Step 4.30** in the **Trunk Group ID** field. • Click on **Apply** and then **Close**. 

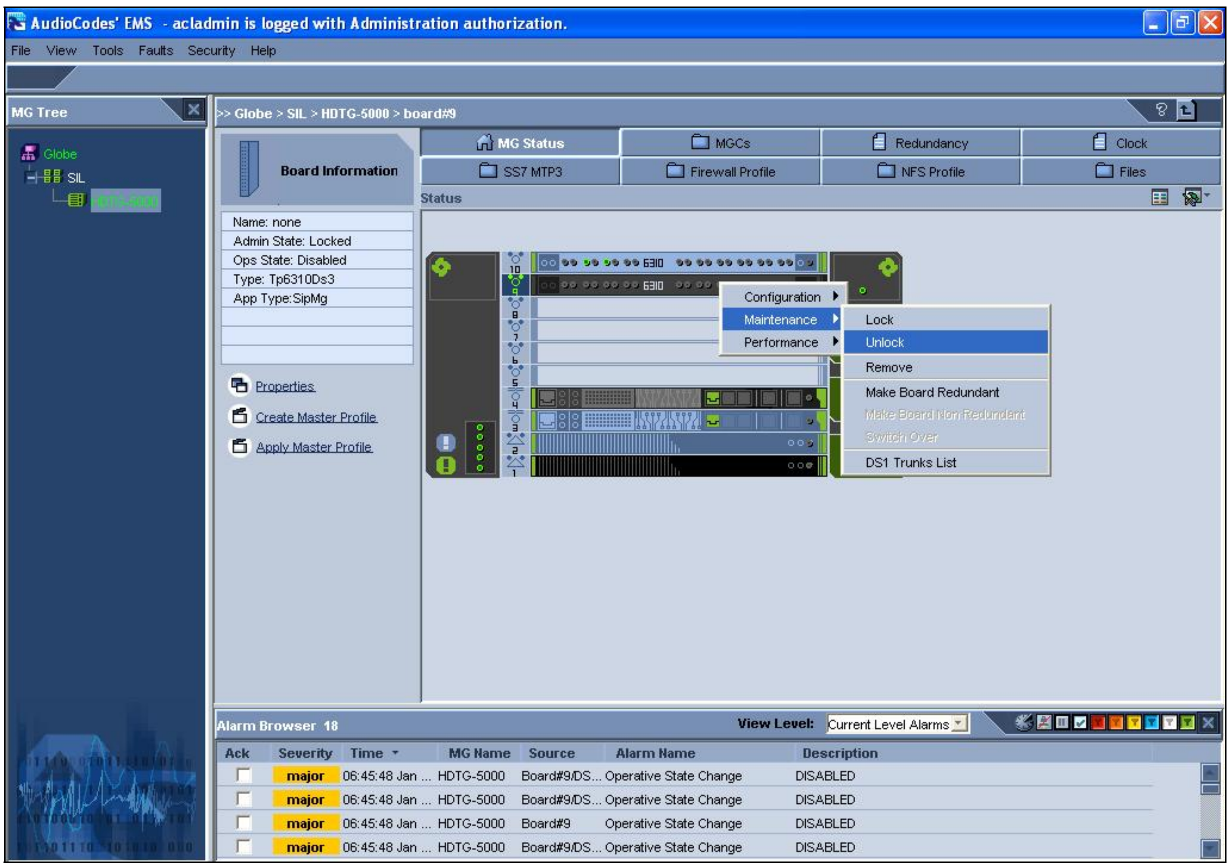
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.40</td><td>
Repeat Step 4.38 and Step 4.39 to administer settings to enable Dial-Out from the Avaya Meeting Exchange S6800 Conferencing Server to the PSTN over an ISDN-PRI trunk with the following parameters:

Enter PRI in the Name field.
Enter a rule in the Dest Phone Prefix field that matches the pattern of outgoing calls from the Avaya Meeting Exchange S6800 Conferencing Server to the PSTN. For these Application Notes, all calls from the Avaya Meeting Exchange S6800 Conferencing Server to the PSTN via ISDN-PRI trunking are placed to telephone numbers with the leading digits 50503. The rule 50503* is utilized, where * is a wildcard and will match any remaining digit(s).
Enter an * in the Source Phone Prefix and Source Address fields to allow routing for any party Dialing-Out from the Avaya Meeting Exchange S6800 Conferencing Server to the PSTN.
Enter the Trunk Group ID for the ISDN-PRI trunk group provisioned in Step 4.31 in the Trunk Group ID field.

The resultant Sip IP To Tel Routing List is shown below.


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

Step	Description
4.41	The board must be unlocked for the above configuration to be applied to the TP6310 board, after which it is reset and enabled for service. • [*Not Shown*] *Click on the* icon *to navigate back to the screen displaying the locked TP6310 board (see* ***Step 4.6****).* • Click on the locked TP6310 board and use mouse button to select **Maintenance → Unlock**. • [***Not Shown***] *To confirm Unlock, click* ***Yes*** *in the confirmation window that is displayed.* ***Note****: The TP6310 board will reset and return to service after several minutes. The* ***Alarm Browser*** *pane at the bottom of the window will indicate the status of the board.*



5. Interoperability Compliance Testing

5.1. General Test Approach

The general test approach was to place calls between the Avaya Meeting Exchange S6800 Conferencing Server and the PSTN via the AudioCodes Mediant 5000 Media Gateway utilizing the network configuration displayed in **Section1**, **Figure 1**.

The main objectives were to verify the following:

- Dial-In Conferencing:
  - DNIS direct call function, where conference participants enter a conference as moderator, without entering a participant-access-code (passcode).
  - Scan call function, where conference participants enter a conference with a valid passcode.
- Dial-Out Conferencing:
  - Blast dial
    - Auto, where a conference participant enters a conference via a DNIS direct call function and autonomously invokes a Blast dial to a pre-provisioned dial list of one or more participants.
    - Manual, where a conference participant is already in a conference as moderator and invokes a Blast dial (by entering *92) to a pre-provisioned dial list of one or more participants.
  - Originator Dial-Out, where a conference participant is already in a conference as moderator and invokes a Dial-Out (by entering *1) to a single participant
  - Operator Fast Dial, where an operator can Dial-Out to a pre-provisioned dial list of one or more participants.
- Operator Dial-Out to establish an Audio Path.
- Operator Dial-In to establish an Audio Path.
- Dial-Out to an FDAPI channel for audio recording.
- Line Transfer invoked from Avaya Bridge Talk.
- Conference Transfer invoked from Avaya Bridge Talk.
- Touchtone commands {e.g.: *0 Request Help, *2 (as moderator) to start/stop conference recording, *3 to start/stop playback of conference recording, *5 (as moderator) toggle lecture on/off, *6 toggle mute on/off, *7 (as moderator) toggle conference security on/off, *8 play the roster of participant name during conference, *93X (where X is defined from 1 to 9) to invoke a subconference, *930 entered from a subconference to go back to the main conference, *93# entered from a subconference (as moderator) to bring all conference participants back to the main conference, ## (as moderator) to end the conference}.
- The following codecs were verified:
  - G711MU, G.711A, G.729.

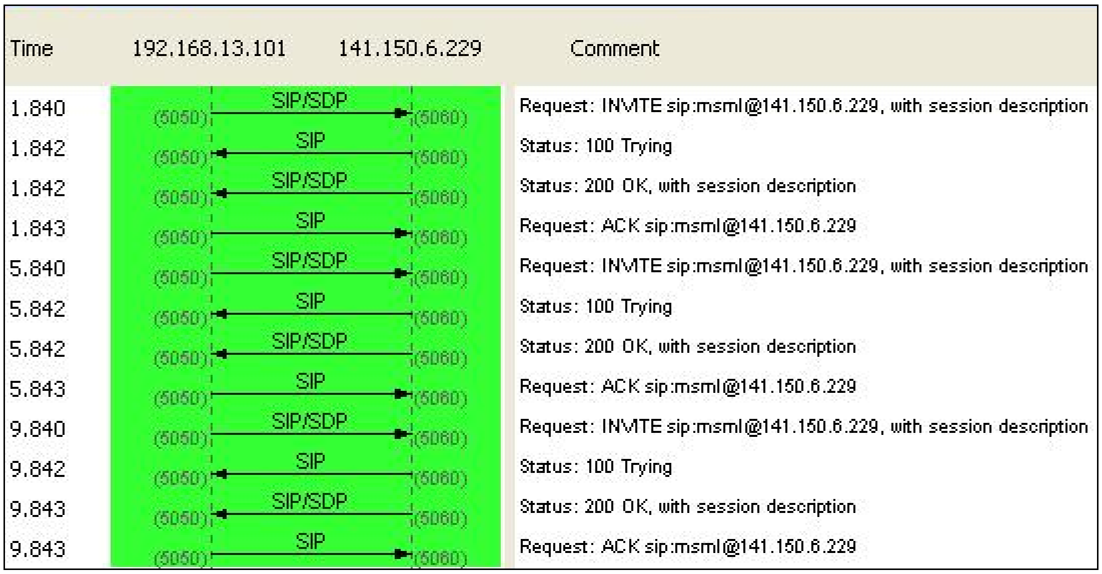
6. Verification Steps

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

- The Avaya Meeting Exchange S6800 Conferencing Server configuration as displayed in **Section 1**, **Figure 2** (verified in **Step 6.1** and **Step 6.2**).
- NFS between the Avaya Meeting Exchange S6200 Application Server and the Convedia CMS-6000 Media Server MPC (verified in **Step 6.3 - Step 6.5**).
- Bi-directional end to end layer-3 connectivity between the MPC in slot 2 on the Convedia CMS-6000 Media Server and the TP6310 board on the AudioCodes Mediant 5000 Media Gateway (verified in **Step 6.6**).
- Verify that the DS3 and DS1 trunks are up on the AudioCodes Mediant 5000 Media Gateway by verifying the icons for those entries on the Trunk & Channel Status screen are green (verified in **Step 6.10**).
- Verify successful inbound and outbound calls between the Avaya Meeting Exchange S6800 Conferencing Server and the PSTN (verified in **Step 6.7 - Step 6.12**).

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>6.1</td><td>Verify all conferencing related processes are running on the Avaya Meeting Exchange S6800 Conferencing Server as follows:
• Log in to the Avaya Meeting Exchange S6200 Application Server console to access the CLI with the appropriate credentials.
• cd to /usr/dcb/bin
• At the command prompt, run the script dcbps and confirm all processes are running by verifying an associated Process ID (PID) for each process.
Note: The process, convMS is running, verifying the Convedia CMS-6000 is functioning as a media server in the Avaya S6800 Conferencing Server architecture (see Section 1, Figure 2).</td></tr>
</table>

```
S6200App->dcbps
  1783   FP 101 ?        0:00 log
  1773   FP 144 ?        0:05 initdcb
  1784   FP 101 ?        0:00 bridgeTr
  1785   FP 105 ?        0:00 netservi
  1788   FP 129 ?        0:00 timer
  1789   FP 101 ?        0:00 traffic
  1790   FP 104 ?        0:00 chdbased
  1791   FP 101 ?        0:00 startd
  1792   FP 109 ?        0:00 cdr
  1793   FP 101 ?        0:00 modapid
  1794   FP 101 ?        0:00 schapid
  1795   FP 104 ?        0:00 callhand
  1796   FP 139 ?        0:00 initipcb
  1797   FP 139 ?        0:00 sipagent
  1798   FP 139 ?        0:00 msdispat
  1799   FP 139 ?        0:00 convMS
  1800   FP 139 ?        0:00 serverCo
  1556   TS  80 ?        0:00 sqlexecd with 5 children
```

Step	Description
6.2	Verify SIP connectivity between the Avaya Meeting Exchange S6800 Conferencing Server and the Convedia CMS-6000 Media Server. The call flow was captured from a mirrored port of the Avaya Meeting Exchange S6200 Application Server's Ethernet interface, utilizing a network protocol analyzer and shows the "keep alive" SIP message set that is exchanged between the Avaya Meeting Exchange S6200 Application Server (**192.168.13.101**) and the control port on the Convedia CMS-6000 Media Server MPC in slot 2 (**141.150.6.229**). 

Step	Description

6.3	Verify that the NFS server is mounted on the Convedia CMS-6000 Media Server MPC as follows:
	• Telnet to the Convedia SCC console (**141.150.6.228**, provisioned in **Step 3.13**) and log in to access the SCC CLI with the appropriate credentials.
	• From the Convedia SCC CLI command prompt:
	o [***Not Shown***] *Enter the command,* ***telnet mpc2*** *(the hostname for control interface on the MPC card in slot 2 provisioned in* ***Step 3.18****) and log in to the console to access the MPC CLI with the appropriate credentials*.
	• From the Convedia MPC CLI command prompt, change directory to **/mnt** and list files to verify the NFS server is mounted on this Convedia CMS-6000 Media Server MPC.
	
	`[mpc2]$ cd /mnt`
	`[mpc2]$ ls -l`
	`total 1`
	`lrwxrwxrwx   1 root       23 Jan 16 10:32 192.168.13.101 -> /mnt/pfa_192.168.13.101`
	`drwxrwxrwx   7 root      512 Dec 31  1999 flashdisk`
	`drwxrwxrwx  16 root      512 Dec 20  2005 nvramdisk`
	`drwxr-xr-x   5 root       96 Jun 29  2006 pfa_192.168.13.101`
	`drwxrwxrwx  14 root      512 Nov  6  2006 ramdisk`

6.4	Verify write privileges to the NFS server from the mount point on the Convedia CMS-6000 Media Server MPC as follows:
	• [***Not Shown***] *From /mnt, change directory to* ***pfa_192.168.13.101/usr3/confrp*** *and list files to verify the directory is empty*.
	• Create a file that does not already exist on the on the NFS server.
	• List the files in **pfa_192.168.13.101/usr3/confrp** and verify newly created file is present.
	
	`[mpc2]$ touch test.NFS`
	`[mpc2]$ ls -l`
	`-rw-r--r--   1 admin          0 Jan 16 15:11 test.NFS`

6.5	From the NFS server, verify the file created in **Step 6.4** from the mount point on the Convedia CMS-6000 Media Server MPC is present in **/usr3/ipcb/usr3/confrp**.
	
	`S6200App->pwd`
	`/usr3/ipcb/usr3/confrp`
	`S6200App->ls -l`
	`total 0`
	`-rw-r--r--    1 500      500               0 Jan 16 15:11 test.NFS`

Step	Description
6.6	Verify bi-directional end to end layer-3 connectivity between the MPC in slot 2 on the Convedia CMS-6000 Media Server and the TP6310 board on the AudioCodes Mediant 5000 Media Gateway using ping or another network diagnostic tool. Bi-directional end to end layer-3 connectivity between the MPC in slot 2 on the Convedia CMS-6000 Media Server and the TP6310 board on the AudioCodes Mediant 5000 Media Gateway implies a bi-directional audio path, e.g., layer-3 connectivity in one direction may imply one-way audio. Verify bi-directional layer-3 connectivity between the MPC in slot 2 on the Convedia CMS-6000 Media Server and the TP6310 board on the AudioCodes Mediant 5000 Media Gateway as follows: • From the MPC in slot 2 on the Convedia CMS-6000 Media Server, verify layer-3 connectivity to the TP6310 board on the AudioCodes Mediant 5000 Media Gateway. • From the TP6310 board on the AudioCodes Mediant 5000 Media Gateway verify layer-3 connectivity to the MPC in slot 2 on the Convedia CMS-6000 Media Server.

6.1. Verify Call Routing

The following steps utilize the network configuration displayed in **Section1**, **Figure 1** to verify the general test approach defined in **Section 6**.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>6.7</td><td>The purpose of this step (and Step 6.8) is to obtain a baseline for the number of ports created on the MPC in slot 2 on the Convedia CMS-6000 Media Server prior to the scenario invoked in Step 6.9. Verify port utilization on the Convedia CMS-6000 Media Server MPC in slot 2 via the web GUI as follows:

[Optional, Not Shown] Reset statistics for the MPC card in slot 2 as follows:

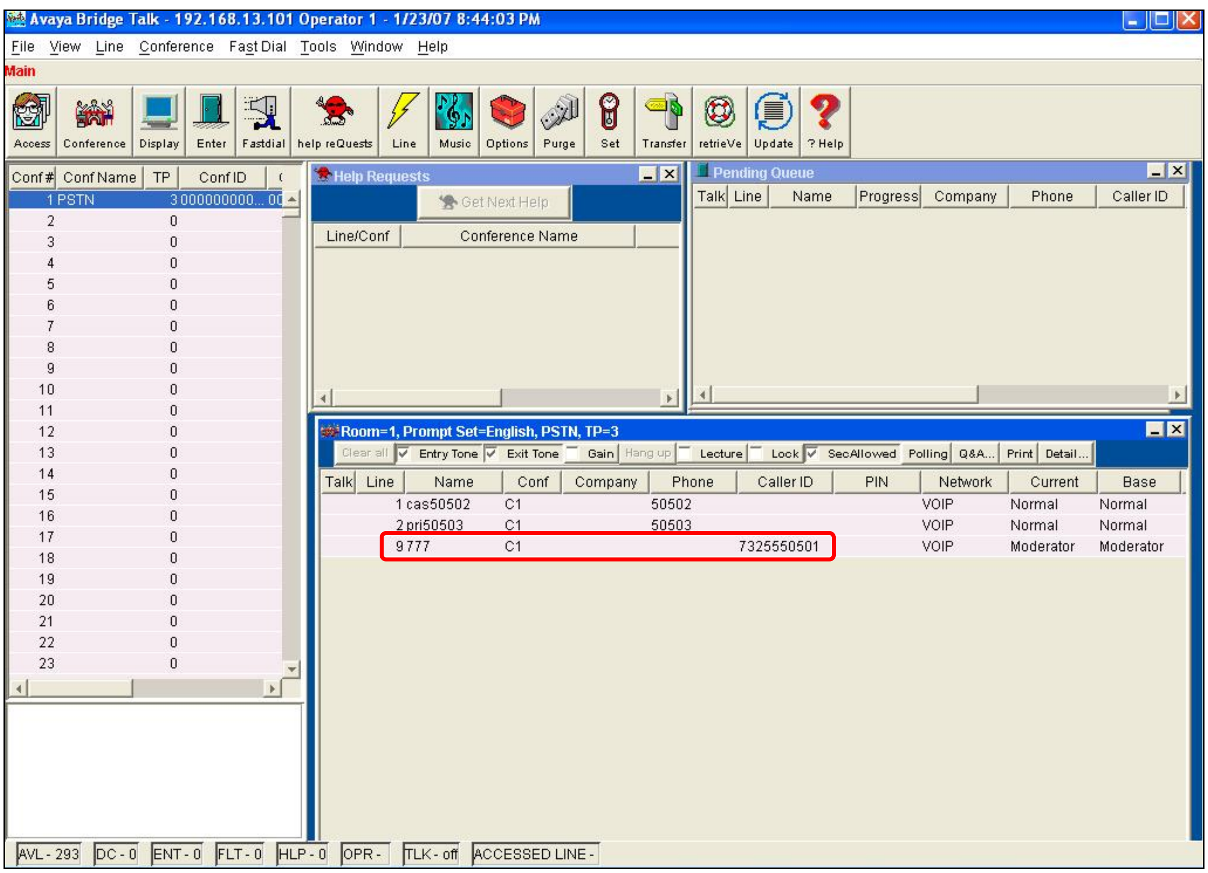
Click Configuration ➔ Performance Mgt ➔ Reset Statistics.
Select the Slot Number for the MPC. For these Application Notes, the MPC was placed in Slot number 2.
Click Execute and wait for the message Statistics for card in slot 2 have been reset to display in the Output Messages window.

Click Configuration ➔ Performance Mgt ➔ Show Real-Time Statistics.
Select the Slot Number for the MPC. For these Application Notes, the MPC was placed in Slot number 2.
Click Execute.

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

Step	Description
6.8	From the **Show Real-Time Statistics** screen that is displayed, note that the number of **Ports Created** for the MPC in slot 2 is **0**. 

Step	Description
6.9	Verify end to end signaling/media connectivity between the Avaya Meeting Exchange S6800 Conferencing Server and the PSTN via the AudioCodes Mediant 5000 Media Gateway. This is accomplished by placing calls to and from the Avaya Meeting Exchange S6800 Conferencing Server. This step utilizes the Avaya Bridge Talk application to verify calls to and from the Avaya Meeting Exchange S6800 Conferencing Server are managed correctly, e.g., callers are added/removed from conferences. This step will also verify the conferencing applications provisioned in **Section 3**. • From an endpoint on the PSTN, Dial **777** to enter a conference as **Moderator** (without passcode) while simultaneously invoking the associated Auto Blast dial feature for this conference (see **Step 3.37**). • If not already logged on, log in to the Avaya Bridge Talk application with the appropriate credentials. • **Double-Click on the** highlighted **Conf #** to open a **Conference Room** window. • Verify conference participants are added/removed from conferences by observing the Conference Navigator and/or Conference Room windows. ***Note****: The ANI extracted via the procedures in* ***Step 3.3*** *is displayed in the* ***Caller ID*** *field for the participant Dialing-In to this conference.*



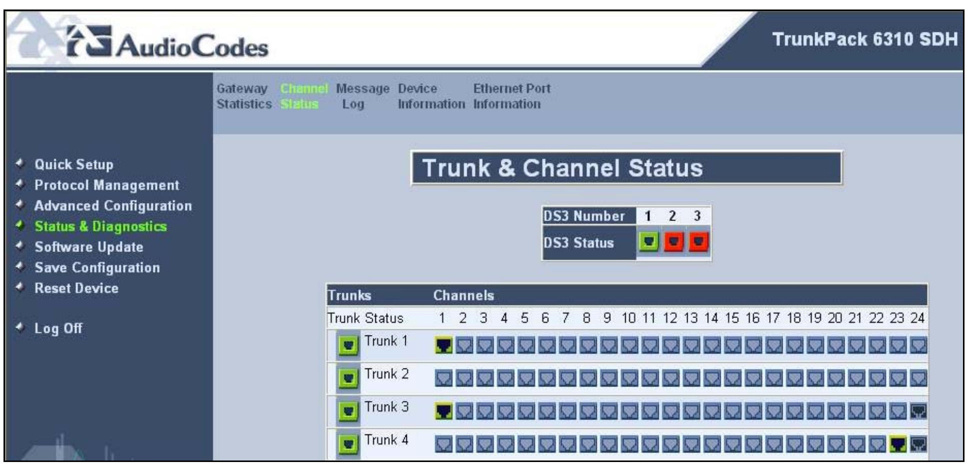
Step	Description
6.10	Verify ISDN **Trunk & Channel Status** on the AudioCodes Mediant 5000 Media Gateway as follows:

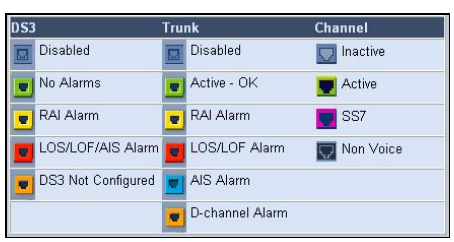
- Open a web browser and enter the following URL:
  http://<IP address of the Active TP6310 board on the AudioCodes Mediant 5000 Media Gateway >
- Log in to the TP6310 board on the AudioCodes Mediant 5000 Media Gateway with the appropriate credentials.
- Click on **Status & Diagnostics**.
- Click on **Channel Status**.

  Note: *The **Trunk & Channel Status** displays **No Alarms** for the DS3 and **Active – OK** for the constituent DS1s provisioned in **Section 4**.*

- This screen capture also depicts the channel selection pattern for the three **Active** channels on this trunk that are associated with the scenario invoked in **Step 6.9**.

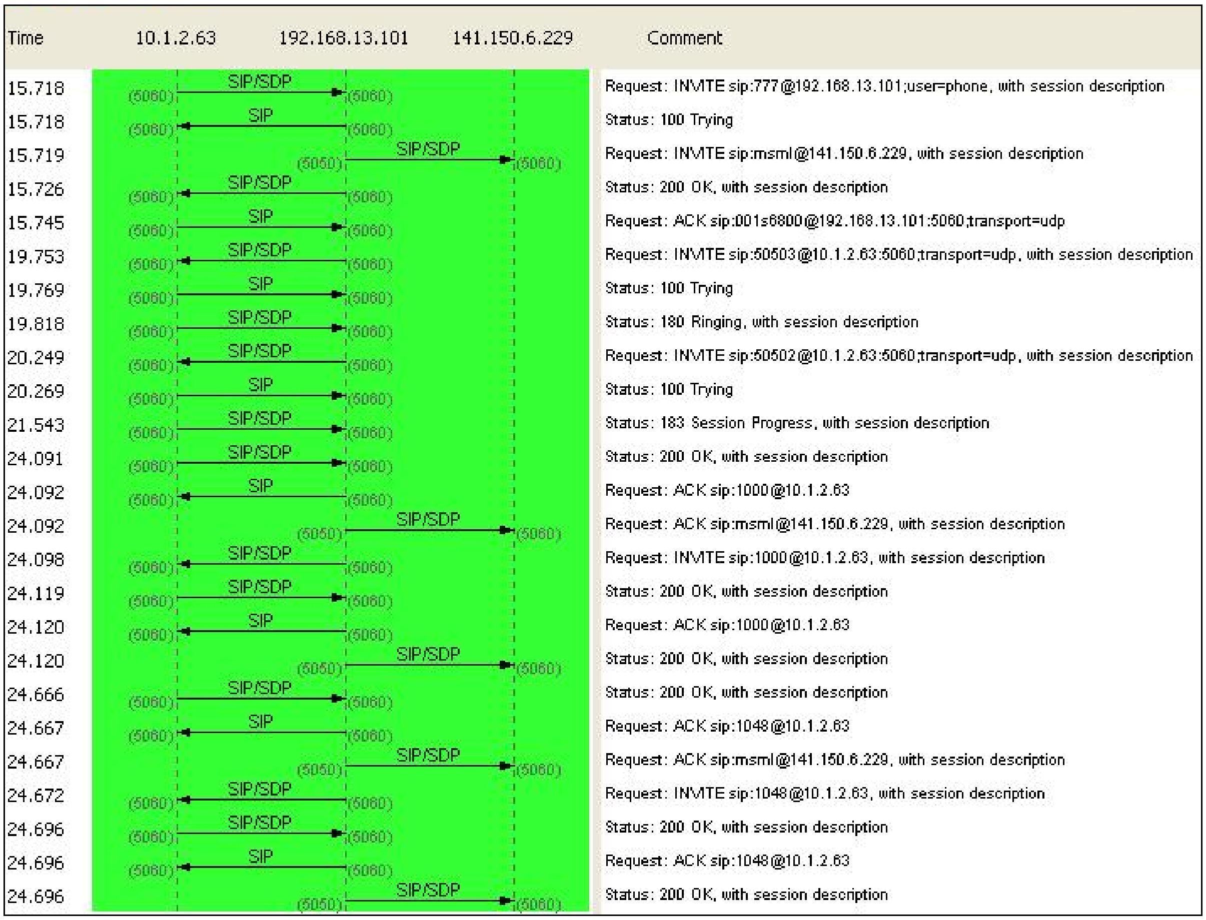
  Note: *The PSTN is administered to select channels in a descending pattern over the ISDN-PRI trunk between the PSTN and the AudioCodes Mediant 5000 Media Gateway. This display shows **Channel 23** on **Trunk 4** is selected by the PSTN for Dial-In to the Avaya Meeting Exchange S6800 Conferencing Server. **Channel 1** on **Trunk 1** is selected by the AudioCodes Mediant 5000 Media Gateway to Dial-Out to the PSTN over the CAS trunk (see **Step 4.33**). **Channel 1** on **Trunk 3** is selected by the AudioCodes Mediant 5000 Media Gateway to Dial-Out to the PSTN over the ISDN-PRI trunk (see **Step 4.34**).*





<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>6.11</td><td>The following SIP call flow displays the moderator Dial-In plus Auto Blast dial scenario invoked in Step 6.9. The call flow was captured from a mirrored port of the Avaya Meeting Exchange S6200 Application Server’s Ethernet interface, utilizing a network protocol analyzer and shows SIP signaling between:

The TP6310 board on the AudioCodes Mediant 5000 Media Gateway (10.1.2.63).
The Avaya Meeting Exchange S6200 Application Server (192.168.13.101).
The control port on the Convedia CMS-6000 Media Server MPC in slot 2 (141.150.6.229).


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

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>6.12</td><td>
Verify port utilization on the Convedia CMS-6000 Media Server MPC in slot 2 following the scenario invoked in Step 6.9 as follows:

From the Show Real-Time Statistics screen (opened via procedures in Step 6.7), click Execute.
Note that the number of Ports Created for the MPC in slot 2 is greater that the number of ports created prior to the scenario invoked in Step 6.9.

Note: This step (in conjunction with Step 6.7 and Step 6.8) validates that the Convedia CMS-6000 Media Server is functioning as a media server. The Avaya Meeting Exchange S6200 Application Server has the capability to function as a stand alone media server. Validating that ports were created on the Convedia CMS-6000 Media Server following a call scenario verifies the Avaya Meeting Exchange S6800 Conferencing Server configuration.

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

7. Conclusion

These Application Notes presented a compliance-tested solution comprised of the Avaya Meeting Exchange S6800 Conferencing Server and the AudioCodes Mediant 5000 Media Gateway. This solution enables connectivity between the Avaya Meeting Exchange S6800 Conferencing Server and the PSTN via the AudioCodes Mediant 5000 Media Gateway.

8. Additional References

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

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

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

4. *AudioCodes EMS User's Manual*, Version 3.2, Document # LTRT-91007.
5. *Element Management System (EMS) Server Installation, Operation & Maintenance Manual*, Version 3.2, Document # LTRT-94109.
6. *IPmedia 5000 Installation, Operation, Maintenance*, Version 3.2, Document # LTRT-89602.

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 Developer*Connection* Program at devconnect@avaya.com.