Avaya Solution & Interoperability Test Lab

Configuring the Avaya G860 Media Gateway to Provide Connectivity between the Public Switched Telephone Network (PSTN), Avaya Meeting Exchange Enterprise S6200 Conferencing Server and Avaya SIP Enablement Services – Issue 1.0

Abstract

These Application Notes describe a compliance tested solution comprising the Avaya Meeting Exchange Enterprise S6200 Conferencing Server communicating directly with the Avaya G860 Media Gateway and via Avaya SIP Enablement Services. The Avaya G860 Media Gateway is utilized to enable connectivity between the Avaya Meeting Exchange Enterprise S6200 Conferencing Server and the Public Switched Telephone Network. This configuration provides a rich set of conferencing options available on the Avaya Meeting Exchange Enterprise S6200 Conferencing Server to participants associated with the Public Switched Telephone Network.

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

TABLE OF CONTENT

1. Introduction

These Application Notes describe a compliance tested solution comprised of the Avaya Meeting Exchange Enterprise S6200 Conferencing Server (MX), Avaya SIP Enablement Services (SES) and the Avaya G860 Media Gateway. The Avaya G860 Media Gateway is utilized to enable connectivity between Avaya Meeting Exchange Enterprise S6200 Conferencing Server and the Public Switched Telephone Network (PSTN). The end to end signalling connectivity between Avaya Meeting Exchange Enterprise S6200 Conferencing Server and the PSTN, either via the Avaya G860 Media Gateway directly or via Avaya SIP Enablement Services is shown in **Figure 1.**

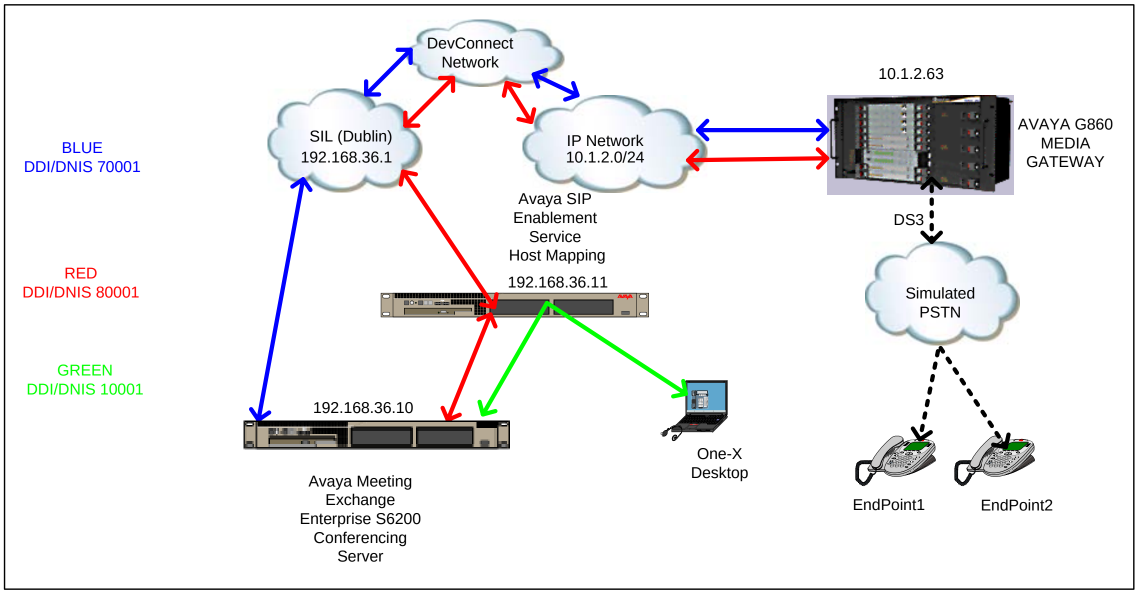


Figure 1: Network Configuration

Note: Figure 1 has two call flows configured (G860-MXS6200) and (G860-SES-MXS6200).These can be configured individually or both together depending on the requirement. Extra steps required to configure G860-SES-MXS6200 are described in Step 3.1.2b, Section 3.3 and Step 4.6.3.

Signalling connectivity between the Public Switched Telephone Network and the Avaya Meeting Exchange Enterprise S6200 Conferencing Server traversed the following path.

- T1 ISDN-PRI (D-channel on channel 24) multiplexed over a DS3 from the PSTN to the Avaya G860 Media Gateway (Black Dotted Line)
- SIP/(UDP, TLS and TCP) between the Avaya G860 Media Gateway and the Avaya Meeting Exchange Enterprise S6200 Conferencing Server (Blue Line)
- SIP/(UDP, TLS and TCP) between the Avaya G860 Media Gateway and Avaya Meeting Exchange Enterprise S6200 Conferencing Server via Avaya SIP Enablement Services host mapping (Red Line)
- SIP/(UDP, TLS and TCP) between Avaya One-X Desktop to Avaya Meeting Exchange Enterprise S6200 Conferencing Server via SIP Enablement Services host mapping (Green Line)

1.1. Avaya Meeting Exchange Enterprise S6200 Conferencing Server

The Avaya Meeting Exchange Enterprise S6200 Conferencing Server is SIP-based with call signalling and Media Server capability for voice conferencing. Avaya's Conferencing Applications include reservation-less, attended, event and mobile to support various IP network implementations. The following capabilities are supported by Avaya Meeting Exchange Enterprise S6200 Conferencing Server:

- RFC 2833 DTMF support
- In-band DTMF support
- Up to 3200-user and 140-operator conferences
- Support for up to four digitally recorded music sources
- Supports codecs G.711 PCMU , G.711 PCMA, iLBC, wbPCMU, wbPCMA and iSAC

1.2. Avaya G860 Media Gateway

The Avaya G860 Media Gateway allows customers to consolidate facilities and reduce communications costs by concentrating Public Switch Telephone Network traffic over DS3 facilities. For high call traffic applications such as conferencing, using a DS3 interface can provide a higher density, lower cost solution compared with DS1 facilities. The Avaya G860 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 Avaya G860 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 Avaya G860 Media Gateway using SNMP. An EMS client communicates with the EMS server from a Microsoft Windows based Personal Computer.

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 Bridge Talk (BT)	5.1.0.0.12
Avaya G860 Media Gateway Chassis Type Software Version Board Type EMS Server EMS Client	 M5k10 5.2.73 Tp6310Ds3 5.2.60 5.2.60
Avaya SIP Enablement Service (SES)	5.1.1 build 415.1
Avaya Meeting Exchange Enterprise S6200 Conferencing Server	5.1 build 161

Table 1: Hardware and Software Versions

3. Configure the Avaya Meeting Exchange Enterprise S6200 Conferencing Server

This section describes the steps for configuring the Avaya Meeting Exchange Enterprise S6200 Conferencing Server to interoperate with the Public Switched Telephone Network either directly via the Avaya G860 Media Gateway or via the Avaya SES (see **Figure 1**).

3.1. Configure the Avaya Meeting Exchange Enterprise S6200 Conferencing Server

The following steps describe the administrative procedures for configuring the Avaya Meeting Exchange Enterprise S6200 Conferencing Server:

- **System.cfg**
- **telnumToUri.tab**

Step	Description
3.1.1	Log in to the Avaya Meeting Exchange Enterprise S6200 Conferencing Server console (PuTTY) using ssh to access the Command Line Interface (CLI) with the appropriate credentials.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.1.2</td><td>Configure settings that enable SIP connectivity between the Avaya Meeting Exchange Enterprise S6200 Conferencing Server and other devices by editing the system.cfg file as follows:

Edit /usr/ipcb/config/system.cfg
Add MXS6200 IP address
IPAddress=(“192.168.36.10”)
Add a line to populate the From Header Field in SIP INVITE messages
MyListener=sip:6000@192.168.36.10
Note: The user field 6000, defined for this SIP URI must conform to RFC 3261. For consistency, it is selected to match the user field provisioned for the respContact entry (see below).
Add a line to provide SIP Device Contact address to use for acknowledging SIP messages from the Avaya Meeting Exchange Enterprise S6200 Conferencing Server:
respContact=<sip:6000@192.168.36.10:5061;transport=tls>
Note: Configure the following if using TCP and UDP
respContact=<sip:6000@192.168.13.101:5060;transport=tcp>
respContact=<sip:6000@192.168.13.101:5060;transport=udp>
Add the following lines to set the Min-SE timer to 900 seconds in SIP INVITE messages from the Avaya Meeting Exchange Enterprise S6200 Conferencing Server:
sessionRefreshTimerValue= 900
minSETimerValue= 900
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 Enterprise S6200 Conferencing Server, e.g., the Avaya G860 Media Gateway. This setting is necessary to enable Dial-Out from the Avaya Meeting Exchange Enterprise S6200 Conferencing Server to the Public Switched Telephone Network via the Avaya G860 Media Gateway.
</td></tr>
</table>

Step	Description
	a) • To enable Dial-Out from the Avaya Meeting Exchange Enterprise S6200 Conferencing Server to the Public Switched Telephone Network via the Avaya G860 Media Gateway, edit the **telnumToUri.tab** file as follows: • Edit **/usr/ipcb/config/telnumToUri.tab** file with a text editor, • Add a line to the file to route outbound calls from the Avaya Meeting Exchange Enterprise S6200 Conferencing Server to the Avaya G860 Media Gateway: *** sip:$1@10.1.2.63:5061;transport=tls G860** **b)** • To enable Dial-Out from the Avaya Meeting Exchange Enterprise S6200 Conferencing Server to the PSTN via the **Avaya SES** to the Avaya G860 Media Gateway, edit the **telnumToUri.tab** file as follows: • Edit **/usr/ipcb/config/telnumToUri.tab** file with a text editor. • Add a line to the file to route outbound calls from the Avaya Meeting Exchange Enterprise S6200 Conferencing Server to the Avaya SES to G860 Media Gateway: *** sip:$1@192.168.36.11:5061;transport=tls SES**

3.2. CBUTIL Utility

The following steps provide examples of how to provision Scan Flow (Scheduled and Demand) conference call functions by utilizing the cbutil utility on the Avaya Meeting Exchange Enterprise S6200 Conferencing Server.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.2.1</td><td>To map DNIS entries, run the cbutil utility on MX as follows:
If not already logged on, log in to the Avaya Meeting Exchange Enterprise S6200 Conferencing Server with an ssh connection using PuTTY with the appropriate credentials.
Enable Dial-In access (via passcode) to conferences provisioned on the Avaya Meeting Exchange Enterprise S6200 Conferencing Server as follows:
Add a DNIS entry for a scan call function corresponding to DID 70001 by entering the following command at the command prompt:
<pre>
cbutil add <dnis> <rg> <msg> <ps> <ucps> <func> [-l <ln> -c
<cn>], where the variables for add command is defined as follows:
            o <dnis>   DNIS
            o <rg>   Reservation Group
            o <msg>   Annunciator message number
            o <ps>   Prompt Set number (0-20)
            o <ucps>   Use Conference Prompt Set (y/n)
            o <func>   One of:
                        DIRECT/SCAN/ENTER/HANGUP/AUTOVL/FLEX
            o -l <"ln"> Optional line name to associate with
            caller
            o -c <"cn"> Optional company name to associate with
            caller
</pre>
In this sample configuration:
<pre>
S6200App->cbutil add 70001 0 1 1 n scan
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.
</pre></td></tr>
<tr><td>3.2.2</td><td>At the command prompt, enter cbutil list to verify the DNIS entries provisioned.
<pre>
[sroot@MX-G860 config]# cbutil list
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.

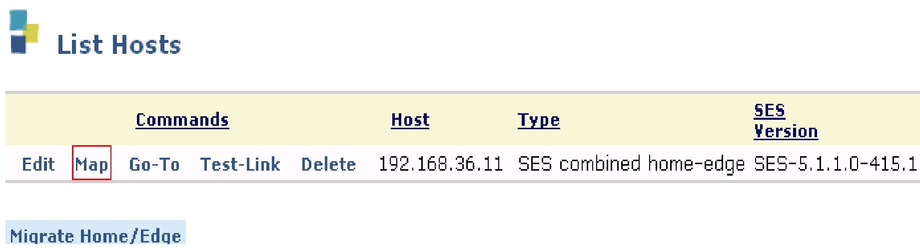
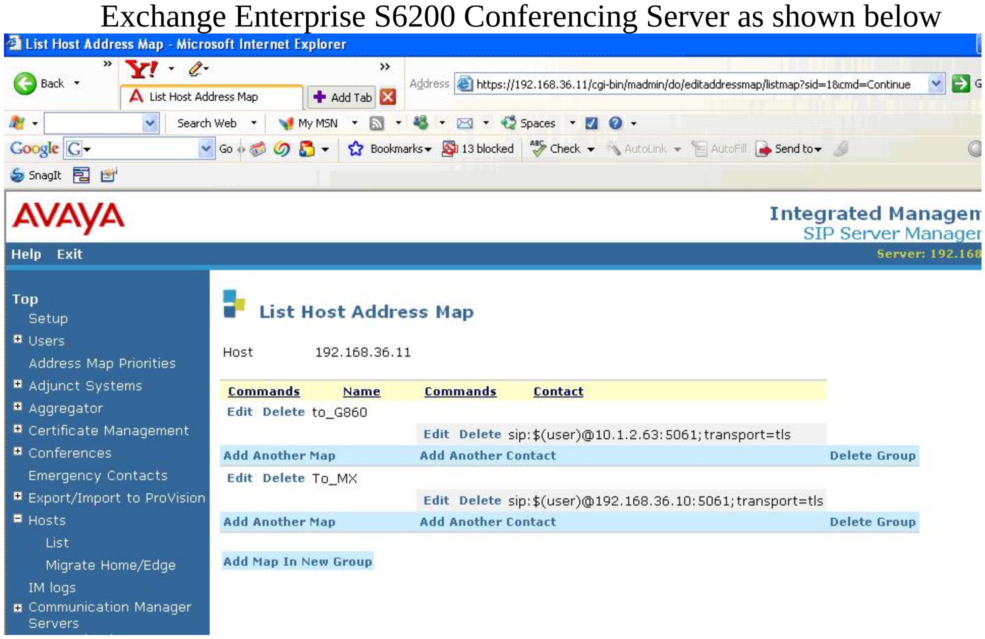
DNIS   Grp Msg PS CP Function On Failure Line Name
Company Name   Room Start Room End

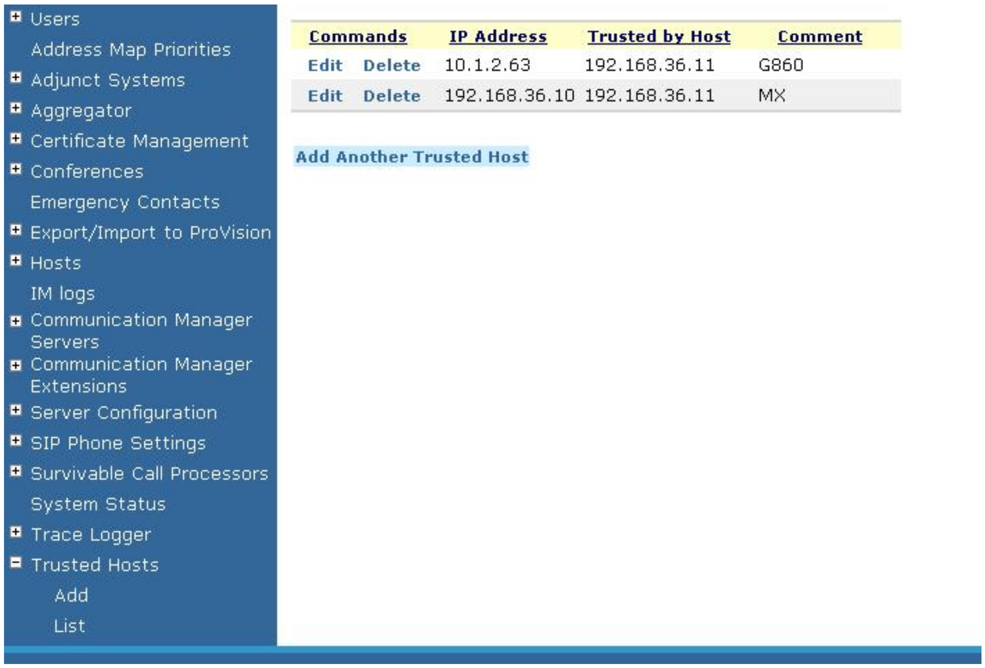
10001   0   247 1   N SCAN   ENTER   LocalMX-
OneX   0   0
70001   0   247 1   N SCAN   ENTER   MX-G860
0   0
80001   0   247 1   N SCAN   ENTER   MX-SES-
G860   0
</pre></td></tr>
</table>

3.3. Configure Avaya SIP Enablement Services (SES)

The following steps describe the administrative procedures for configuring host mapping in Avaya SIP Enablement Services to enable call routing between Avaya Meeting Exchange Enterprise S6200 Conferencing Server and Avaya G860 Media Gateway.

- **SES-MXS6200 for dial in calls to MX**
- **SES-G860 for dial out calls from MX**
- **Adding Trusted Hosts**

Step	Description
3.3.1	Configure host mapping in SES as follows: • Open the SES web page http://192.168.36.11/admin, enter the appropriate credentials, and click on **Launch SES Administration Interface**. • Click on **Hosts Tab-->List Hosts** tab and Click on **Map** .  • Add the host map from Avaya SIP Enablement Services to the Avaya G860 Media Gateway by clicking **Add Map in New Group** and **Add another Contact** • Repeat the same for Avaya SIP Enablement Services to Meeting Exchange Enterprise S6200 Conferencing Server as shown below 

Step	Description
3.3.2	Adding the MXS6200 Conferencing Server and G860 as Trusted Hosts 

3.4. Bridge Talk

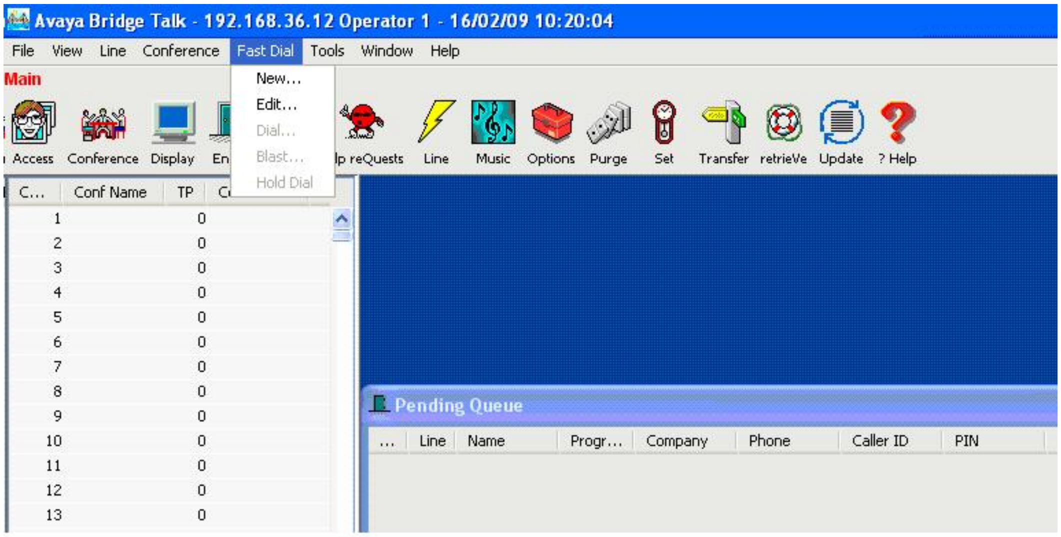
The following steps utilize the Avaya Bridge Talk application to provision a sample conference on the Avaya Meeting Exchange Enterprise S6200 Conferencing Server. This sample conference enables both Dial-In and Dial-Out access to audio conferencing for endpoints on the Public Switched Telephone Network.

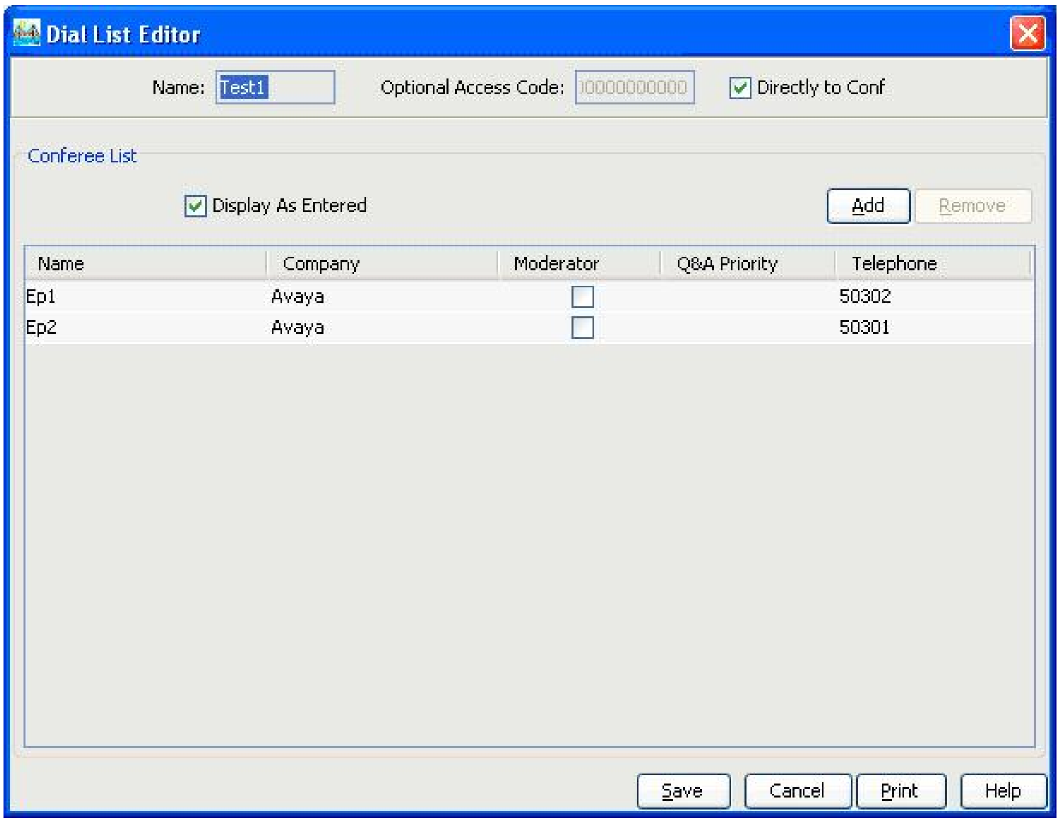
***Note**: If any of the features displayed in the Avaya Bridge Talk screen captures are not present, contact an authorized Avaya sales representative to make the appropriate changes*.

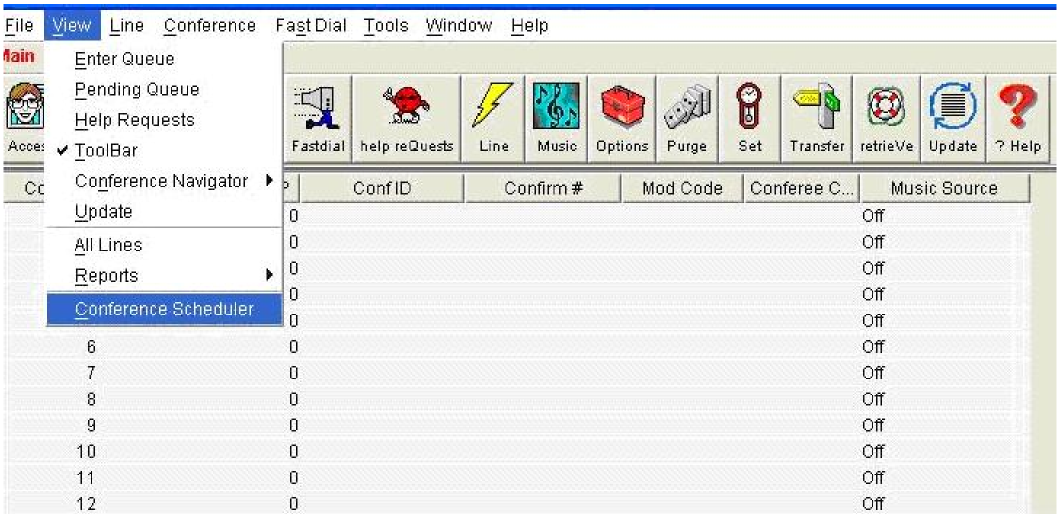
- **Logging into Bridge Talk**
- **Creating Dial List (Manual/Blast dial)**
- **Scheduling Conference**

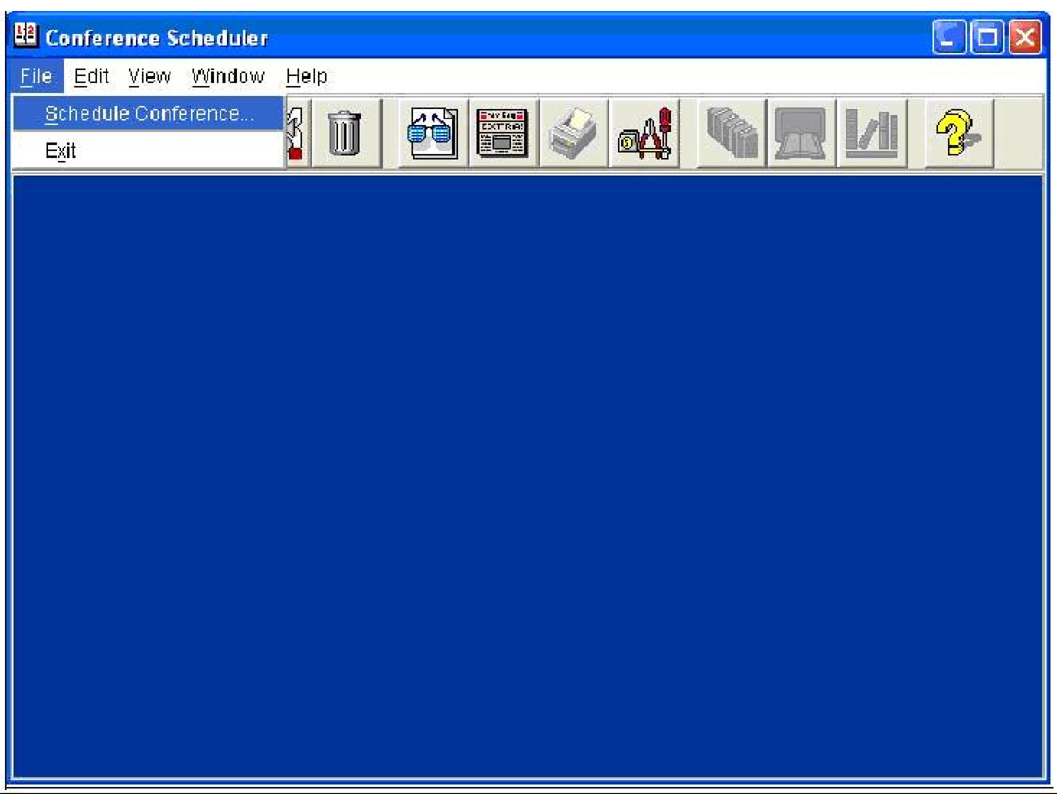
Step	Description
3.4.1	Invoke the Avaya Bridge Talk application as follows: • [*Not Shown*] *Double-click on the desktop icon from a Personal Computer loaded with the Avaya Bridge Talk application and with network connectivity to the Avaya Meeting Exchange Enterprise S6200 Conferencing Server*. • Enter the IP address of the Avaya Meeting Exchange Enterprise S6200 Conferencing Server (**192.168.36.10**) in the **Bridge** field. • Enter the appropriate credentials in the **Sign-In** and **Password** fields. 

..

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

Step	Description
3.4.3	From the **New Dial List→Dial List Editor** 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 and enter **Name**, **Company** and **Telephone** number for dial out for each participant. [*Optional*] *Moderator privileges may be granted to a conference participant by checking the* ***Moderator*** *box.* • When finished, click on the **Save** button on the bottom of the screen. 

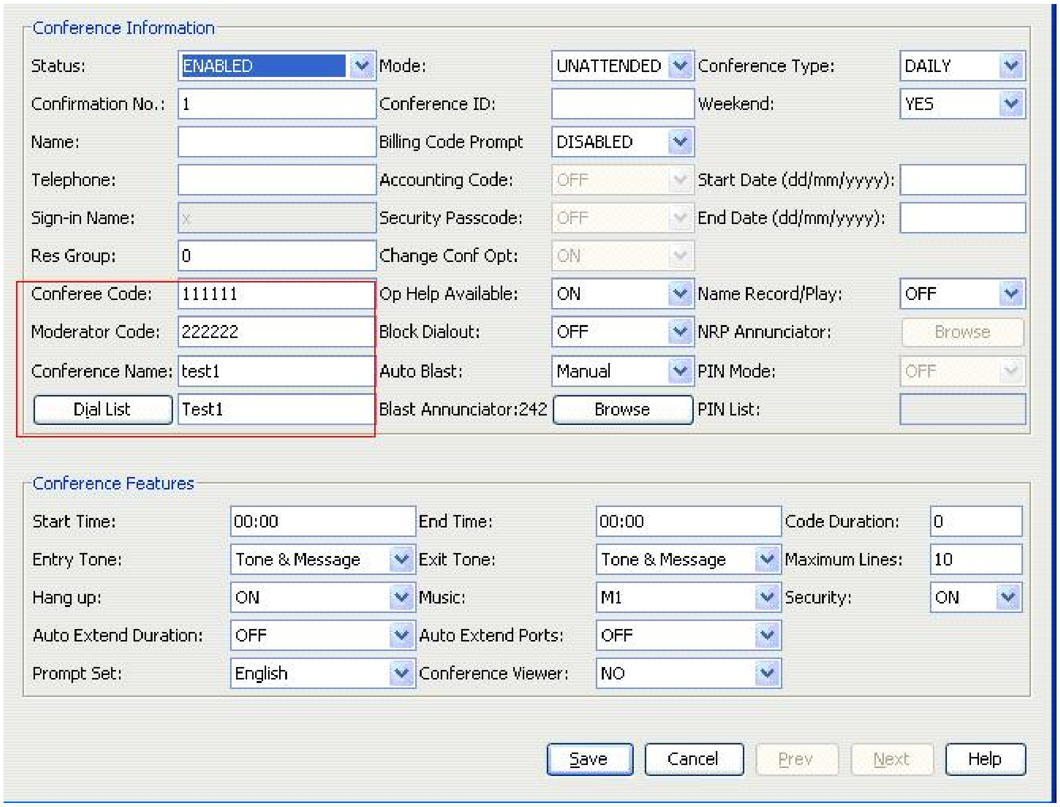
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.4.4</td><td>Provision a conference with Auto Blast enabled.
From the Avaya Bridge Talk Menu Bar, click View →Conference Scheduler.


</td></tr>
<tr><td>3.4.5</td><td>From the Conference Scheduler window, click File →Schedule Conference.


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

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.4.6</td><td>
From the Schedule Conference window that is displayed, provision a conference as follows:

Enter a unique Conferee Code to allow participants access to this conference.
Enter a unique Moderator Code to allow participants access to this conference with moderator privileges. Enable moderator access with a passcode for this conference call by configuring the following:
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/Manual depending on this test.
[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, and click on the Select button (To verify Dial out and Blast Dial out).
When finished, click on the Save button on the bottom of the screen.

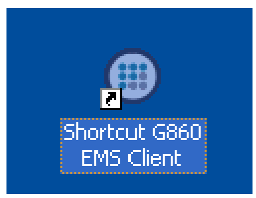


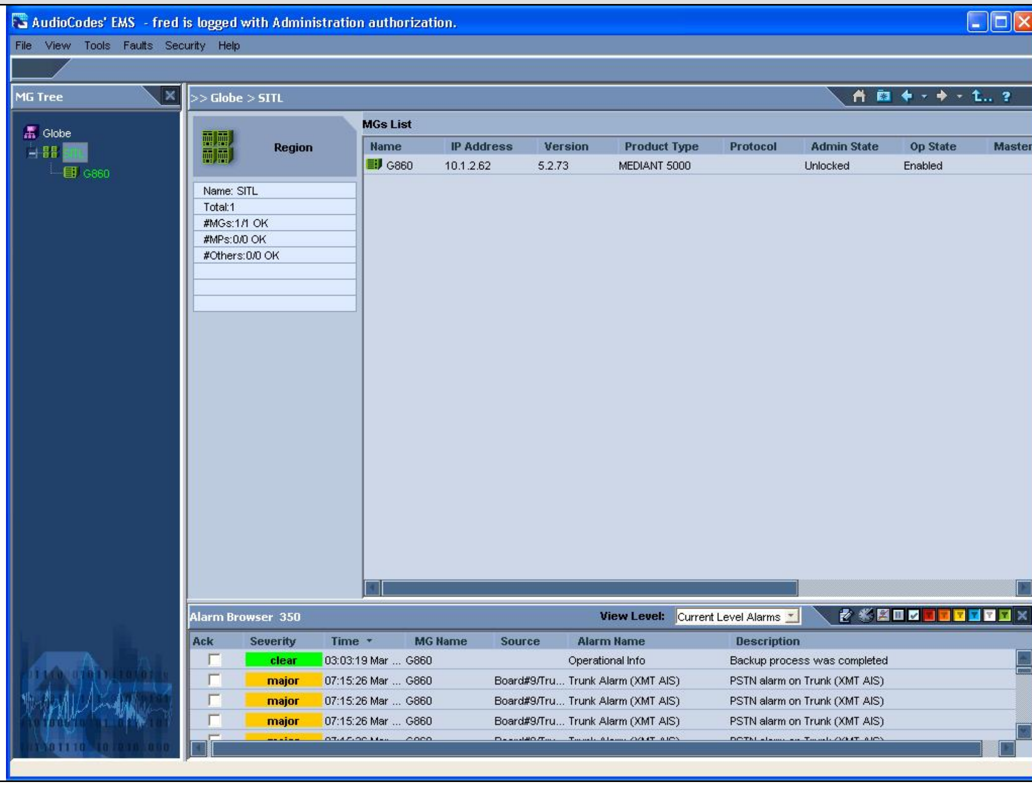
</td></tr>
</table>

4. Configure the Avaya G860 Media Gateway

The following sections describe the steps for configuring the SIP and Public Switched Telephone Network trunks and call routing for the Avaya G860 Media Gateway. This configuration will enable the Avaya G860 Media Gateway to interoperate with both the Avaya Meeting Exchange Enterprise S6200 Conferencing Server both directly and with Avaya SIP Enablement Services (see **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 Avaya G860 Media Gateway, EMS server, and EMS client have already been installed.

- **Logging into Gateway using EMS Client**

Step	Description
4.1	Invoke the GUI provisioning system from a Personal Computer running the EMS client by double-clicking on the desktop icon as shown below. From the login screen that is displayed, enter the login, password and the IP address of the EMS server. 
4.2	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 Avaya G860 Media Gateway 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.*

Step	Description
	

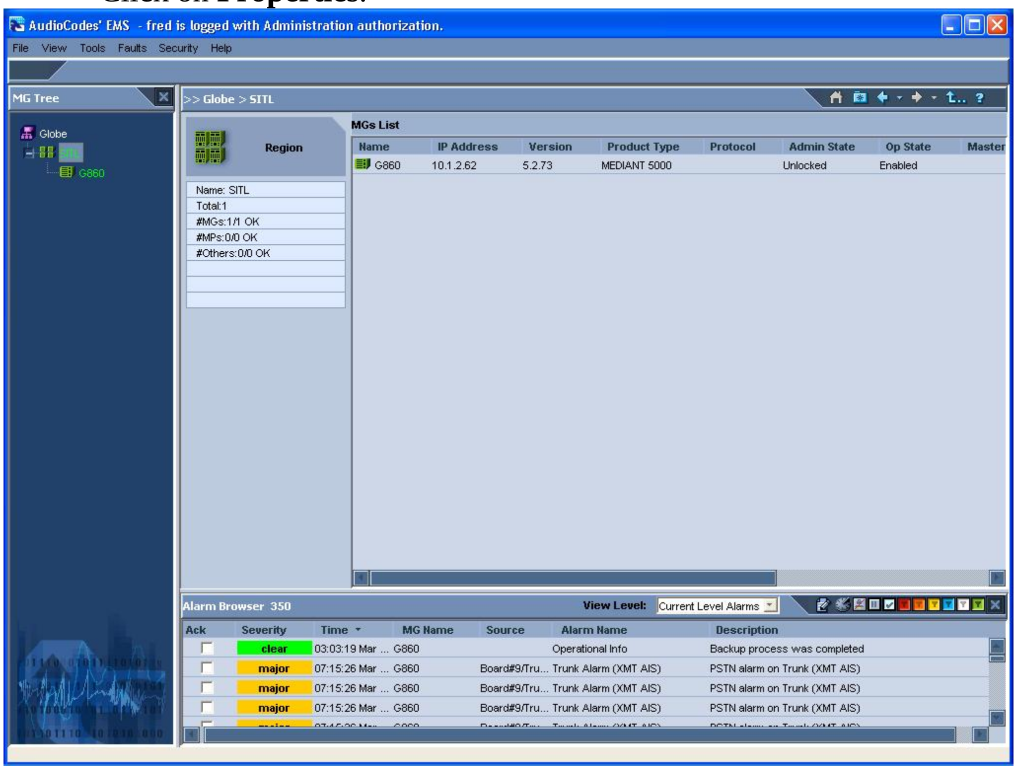
4.1. Configure the Avaya G860 Media Gateway Properties

The following steps describe the administrative procedures for configuring system-wide parameters on the Avaya G860 Media Gateway.

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

From the media gateway list in the **MGs List** pane that is displayed:

- Select the entry corresponding to the Avaya G860 Media Gateway to be configured.
- Click on **Properties**.

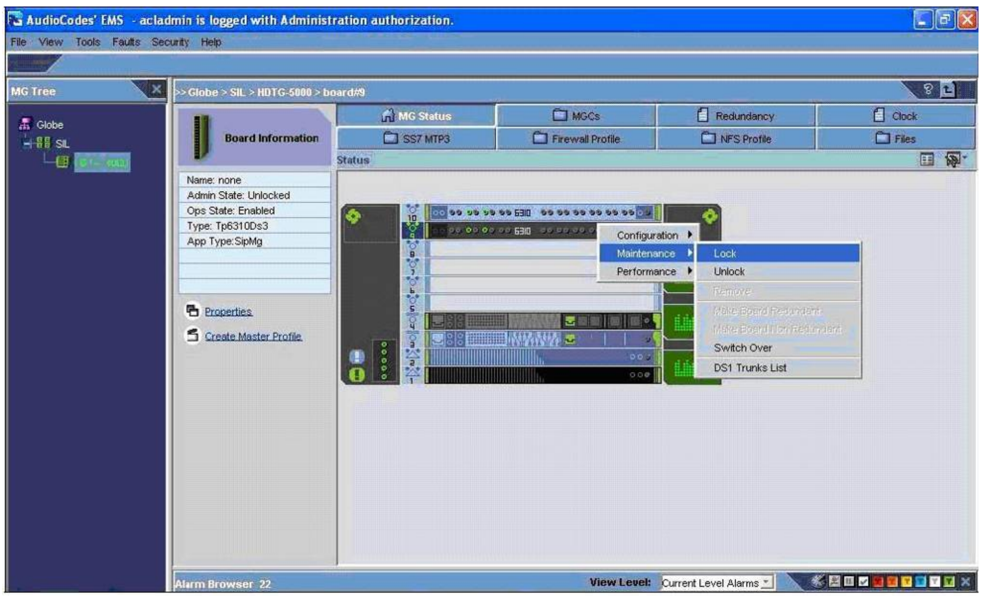


</td></tr>
</table>

4.2. Configure the TP6310 Board

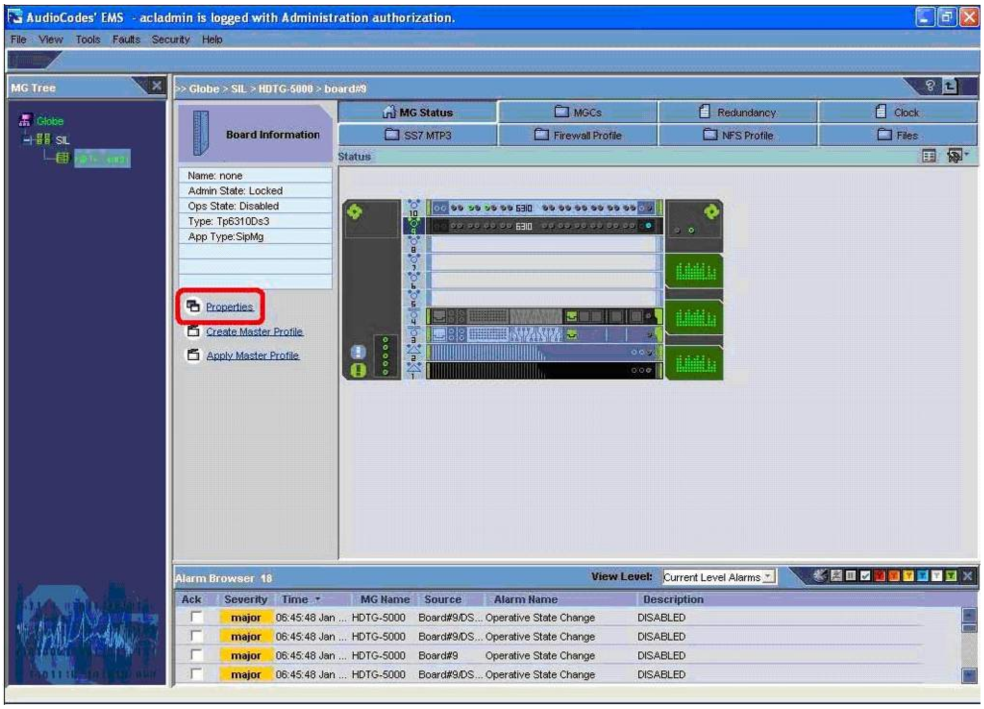
The following steps describe the administrative procedures for configuring the active TP6310 board in the Avaya G860 Media Gateway chassis. These procedures will administer settings for SIP and DS3 trunking, as well as the call routing rules associated with this TP6310 board to enable signalling/media connectivity between the Avaya Meeting Exchange Enterprise S6200 Conferencing Server and the Public Switch Telephone Network.

- **Board IP Address**
- **Board Voice configuration**

Step	Description
4.2.1	Prior to making any change to the configuration of the TP6310 board, the board must be locked. • On the MGs List pane, double-click on the row corresponding to the Avaya G860 Media Gateway. o ***Note****: The* ***MG Status*** *tab will be highlighted and the* ***Status*** *pane will open, depicting a replica of the front panel of the* Avaya G860 Media Gateway *chassis. Board slots are numbered from 1 to 10 from bottom to top on the left side. For these Application Notes, installed gateway boards include the TP6310 DS3 boards in slots 8, 9, and 10.* • 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 Avaya G860 Media Gateway, locking this board removes it from service and is service impacting*. 

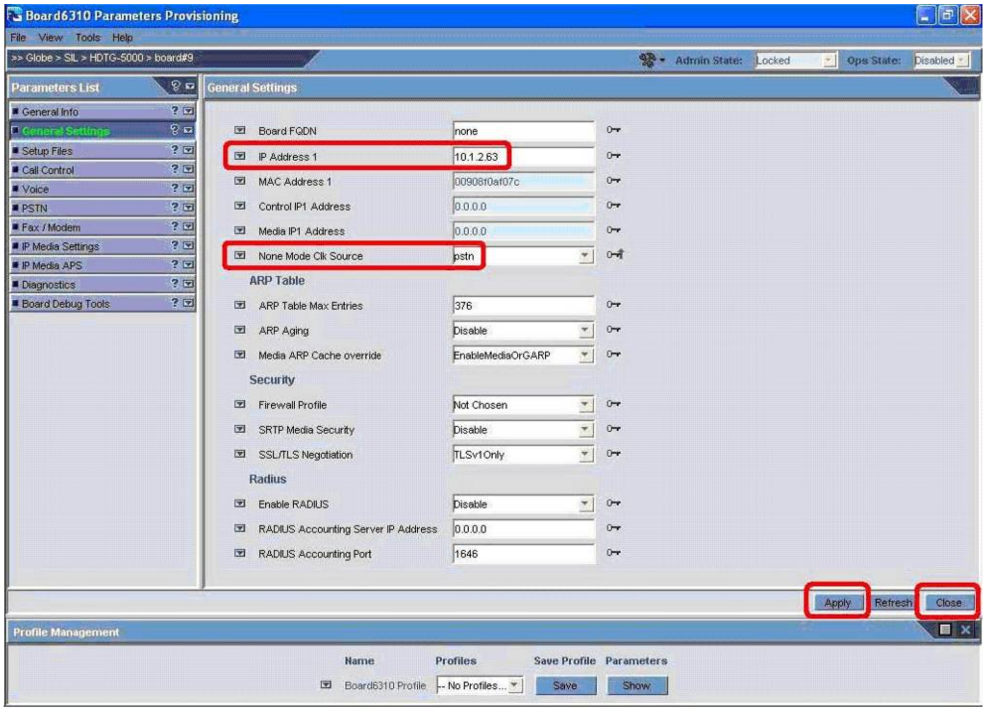
<table>
<tr>
<td>4.2.2</td>
<td>
Administer settings on the locked TP6310 board as follows:

- 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.



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

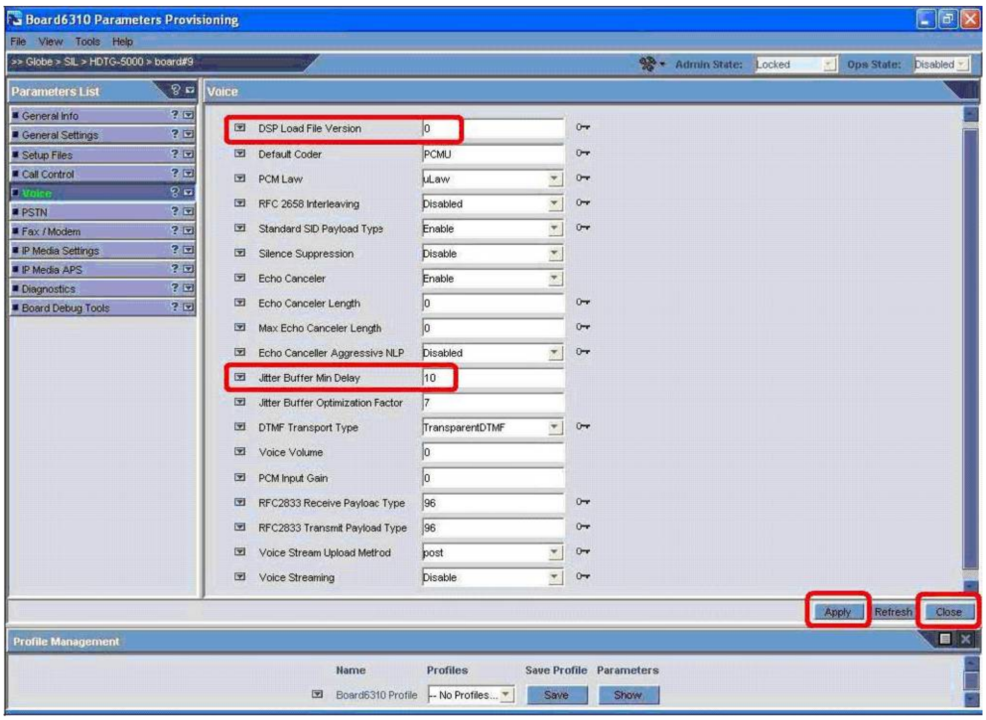
4.2.3	From the **Board6310 Parameters Provisioning** window that is displayed: • Click on **General Settings** under **Parameters List**. • Set **IP Address 1** to the IP address for this board • Select **pstn** for the **None Mode Clk Source**. • Remaining fields are default settings. • Click on **Apply**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



4.2.4

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**.



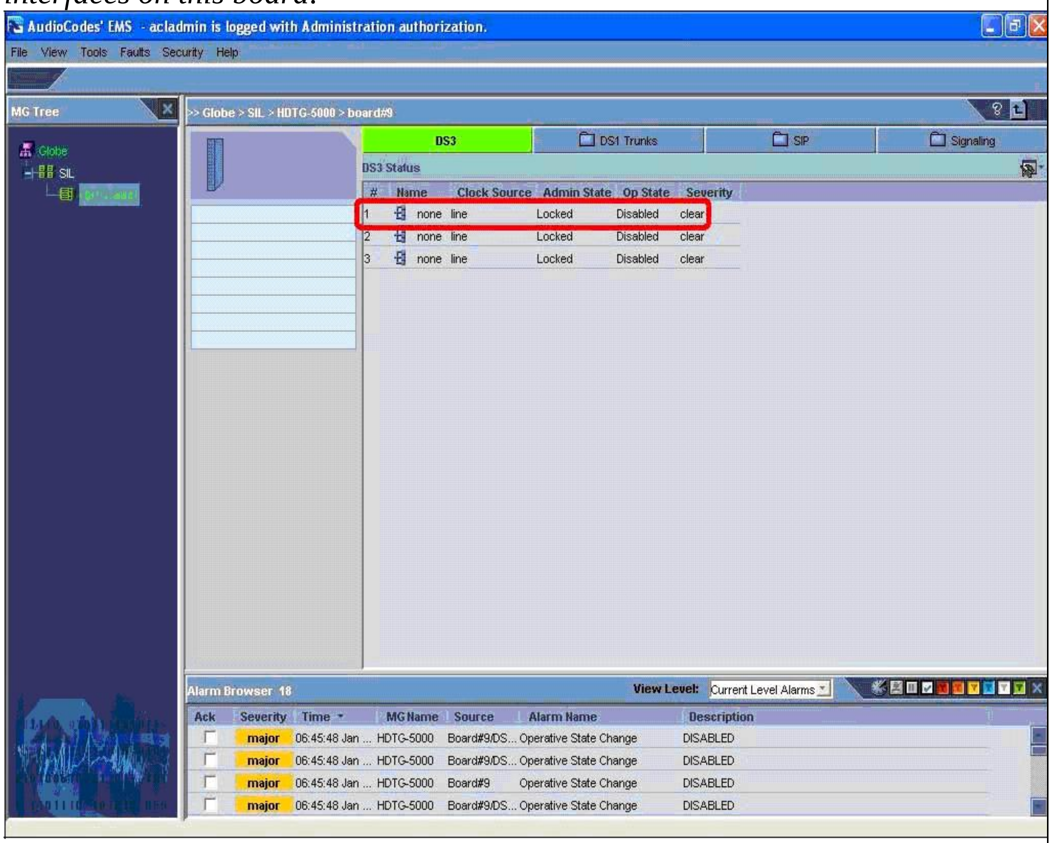
4.3. Configure DS3/DS1 Trunking

The following steps describe the administrative procedures for configuring the DS3 and constituent DS1 trunking between the Avaya G860 Media Gateway and the Public Switched Telephone Network.

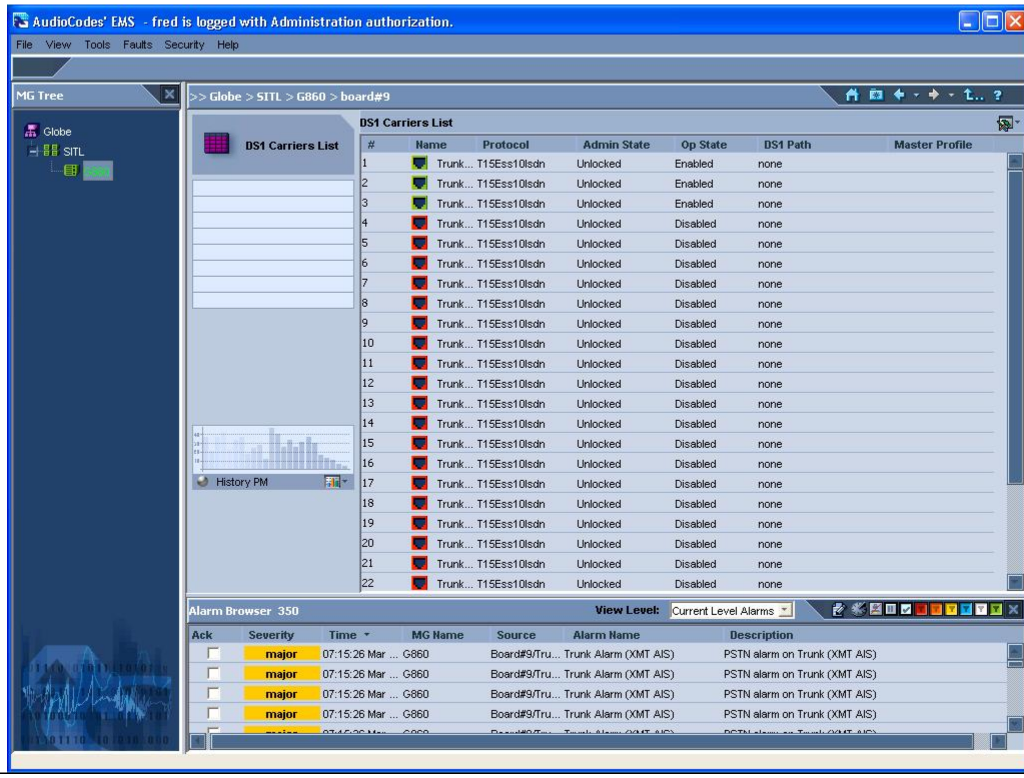
- **Configuring DS3 and DS1**

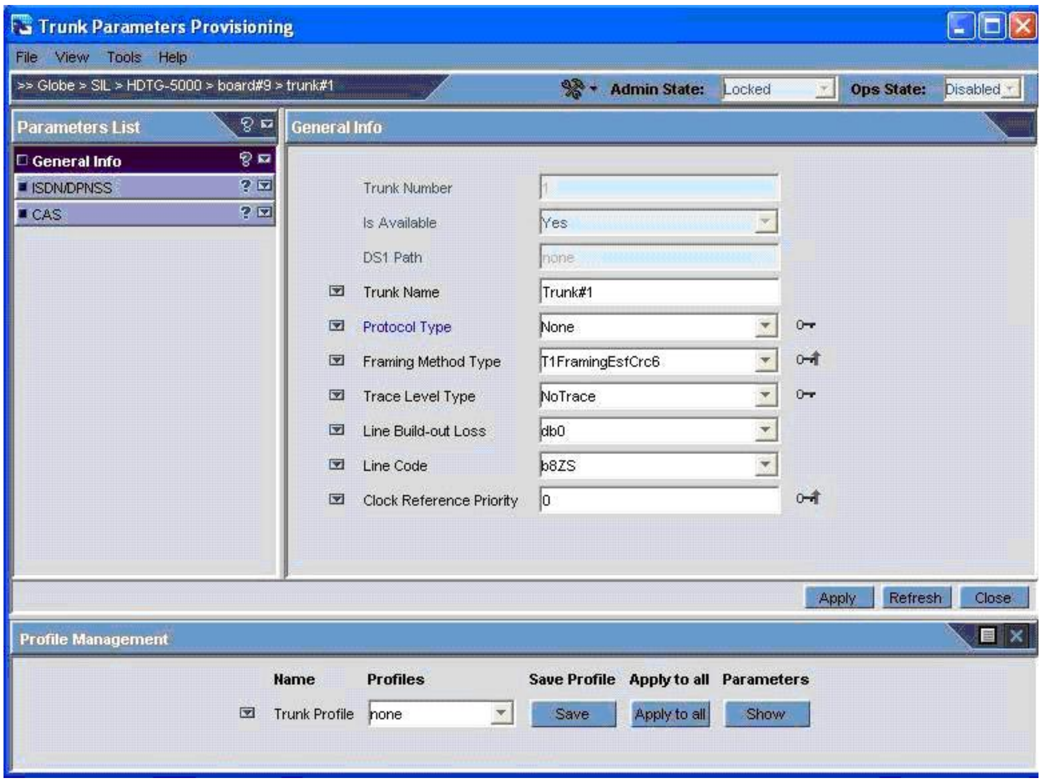
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.3.1</td><td>Administer settings for a DS3 trunk to enable connectivity to the Public Switched Telephone Network as follows:

[Not Shown] Double-click on the locked TP6310 board in the Status pane .
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.

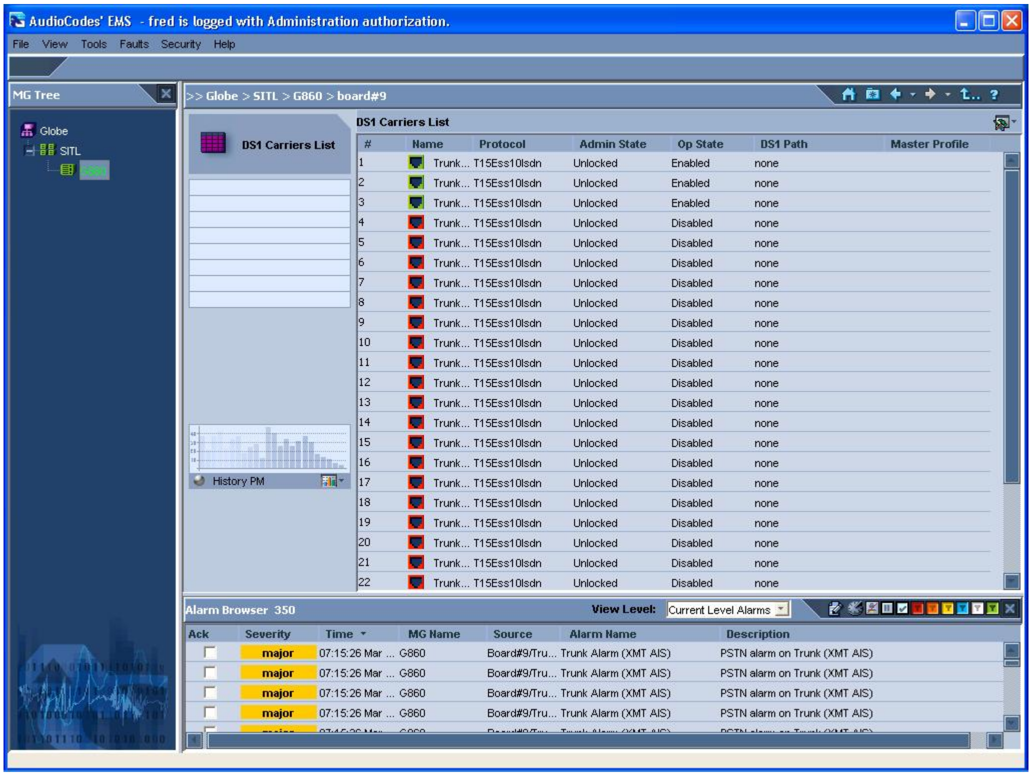


</td></tr>
</table>

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

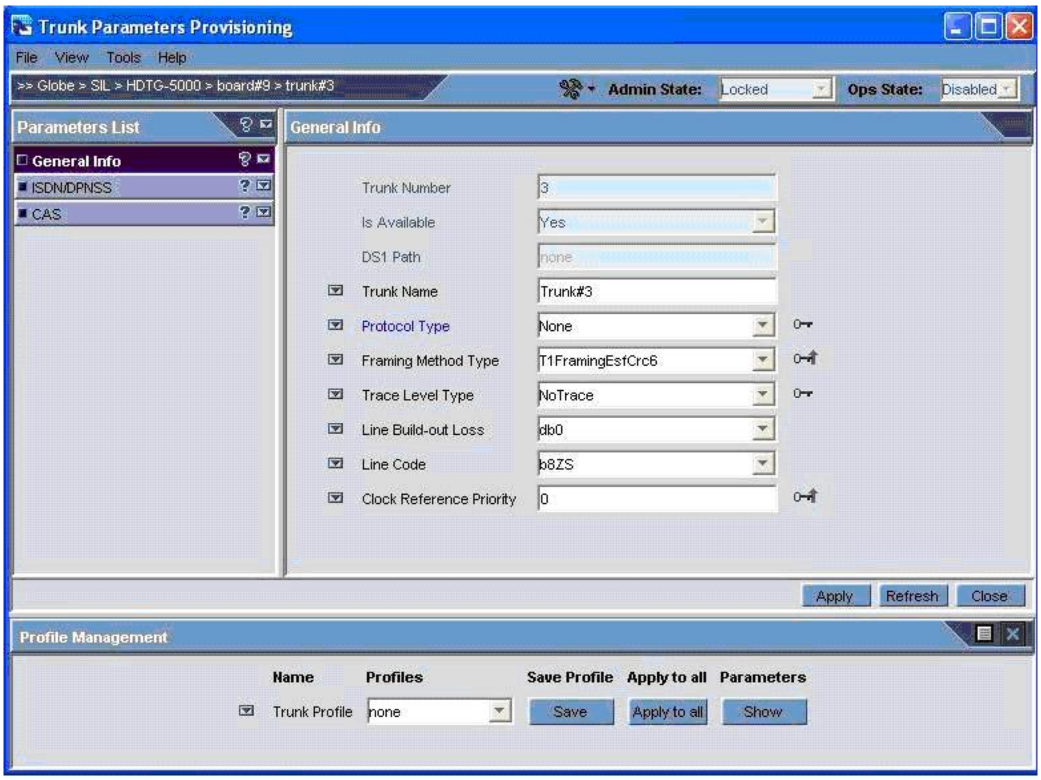
Step	Description
4.3.3	From the **General Info** pane, in the **Trunk Parameters Provisioning** window that is displayed: Configure as required by service provider • Administer settings to enable connectivity with the Public Switch Telephone Network. ***Note****: Obtain configuration details regarding the setting required for this connection to the Public Switched Telephone Network from the service provider. The entries for this trunk correspond to a T1 PRI connection between the Avaya G860 Media Gateway and the Public Switched Telephone Network.* 

Step	Description
4.3.4	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.

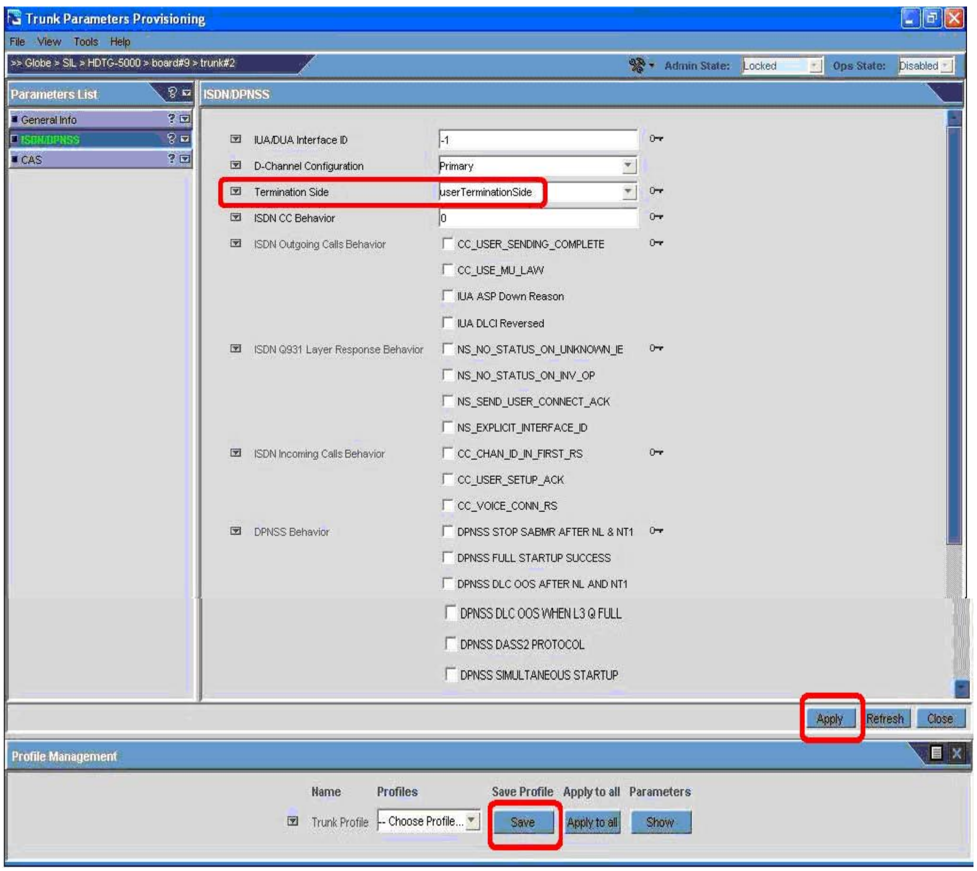


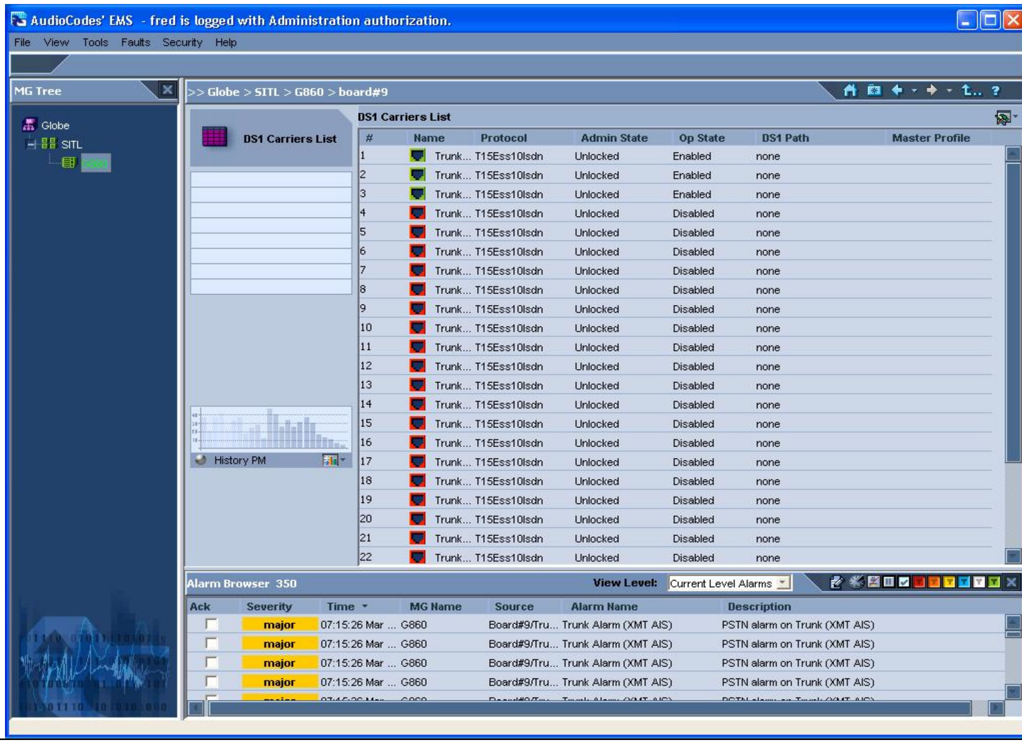
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.3.5</td><td>From the General Info pane, in the Trunk Parameters Provisioning window that is displayed:

Administer settings to enable connectivity with the Public Switch Telephone Network.
Note: Obtain configuration details regarding the setting required for this connection to the Public Switch Telephone Network from the service provider. The entries for this trunk correspond to a T1 ISDN-PRI connection between the Avaya G860 Media Gateway and the Public Switch Telephone Network.
Click on Apply.
Click on ISDN/DPNSS under Parameters List.



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

Step	Description
4.3.6	From the **ISDN/DPNSS** pane that is displayed: • Select the appropriate value for the **Termination Side**, usually **userTerminationSide** if the Public Switch Telephone Network connection is to a service provider. • Click on **Apply**. • From the **Profile Management** pane, select **Save** to save the DS1, and assign a name to the profile. *__Note__: The Profile Management pane can be used to define a configuration profile that can be applied to many DS1 interfaces, saving configuration steps.* 

Step	Description
4.3.7	Apply the DS1 configuration saved 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 Avaya G860 Media Gateway and the Avaya Meeting Exchange Enterprise S6200 Conferencing Server.

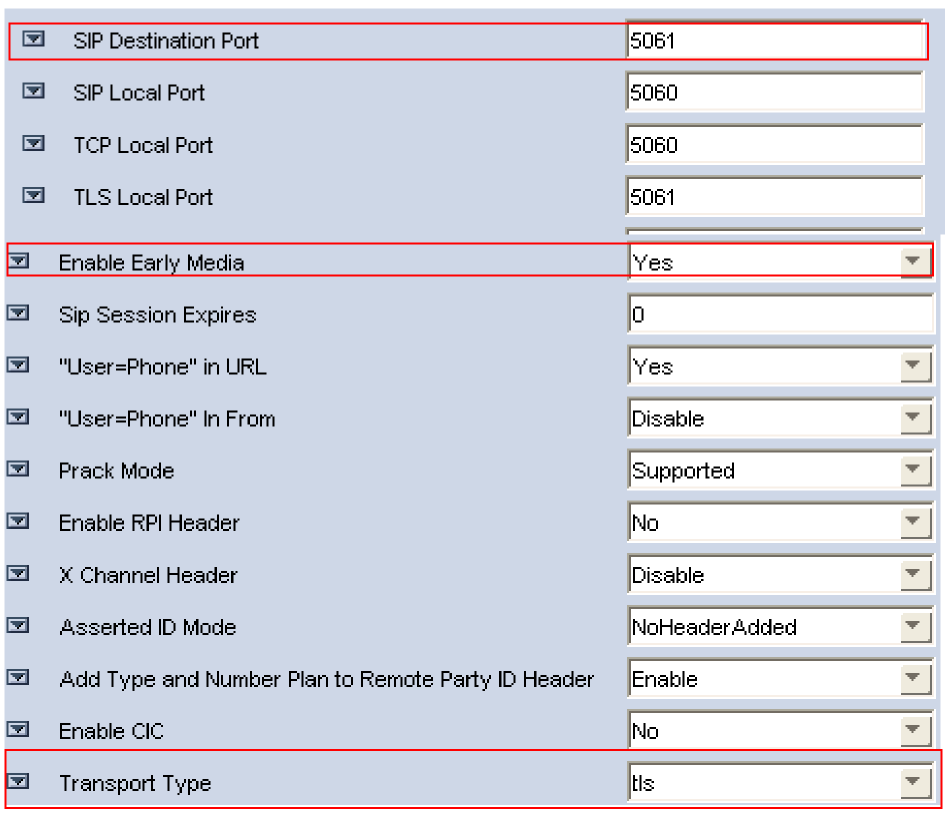
- Configuring Transport Protocol/Codecs between G860 and MXS6200

Step	Description
4.4.1	Double Click on Board 9, and administer settings for SIP trunking to enable connectivity with the Avaya Meeting Exchange Enterprise S6200 Conference Server as follows: • [*Not Shown*] *Click on the* *icon to navigate back to the screen displayed below.* • Click on the **SIP** tab. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.4.2</td><td>Click on the SIP Protocol tab; then click on the Protocol Settings tab.

</td></tr>
<tr><td>4.4.3</td><td>From the General Settings pane, in the SIP Protocol Definitions window, administer settings to enable SIP connectivity with the Avaya Meeting Exchange Enterprise S6200 Conferencing Server as follows:

- Set the SIP Destination Port, Enable Early Media and Transport Type to enable SIP-TLS/UDP/TCP connectivity with the Avaya Meeting Exchange Enterprise S6200 Conferencing Server as shown below:
- Remaining fields are default settings
- Click on Apply and then Close



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

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.4.4</td><td>Click on the Coders tab under SIP Protocol to administer the codec preferences for this SIP trunk between the Avaya G860 Media Gateway and the Avaya Meeting Exchange Enterprise S6200 Conferencing 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.

</td></tr>
<tr><td>4.4.5</td><td>Add a codec that is supported on the Avaya Meeting Exchange Enterprise S6200 with the following parameters:
• Administer settings for G.711 U-law 64k
• Remaining fields are default settings
Note: For testing other supported codecs, change the parameter as shown below.

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

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.

- **Configuring Trunk Groups**
- **Adding PRI trunk Group**

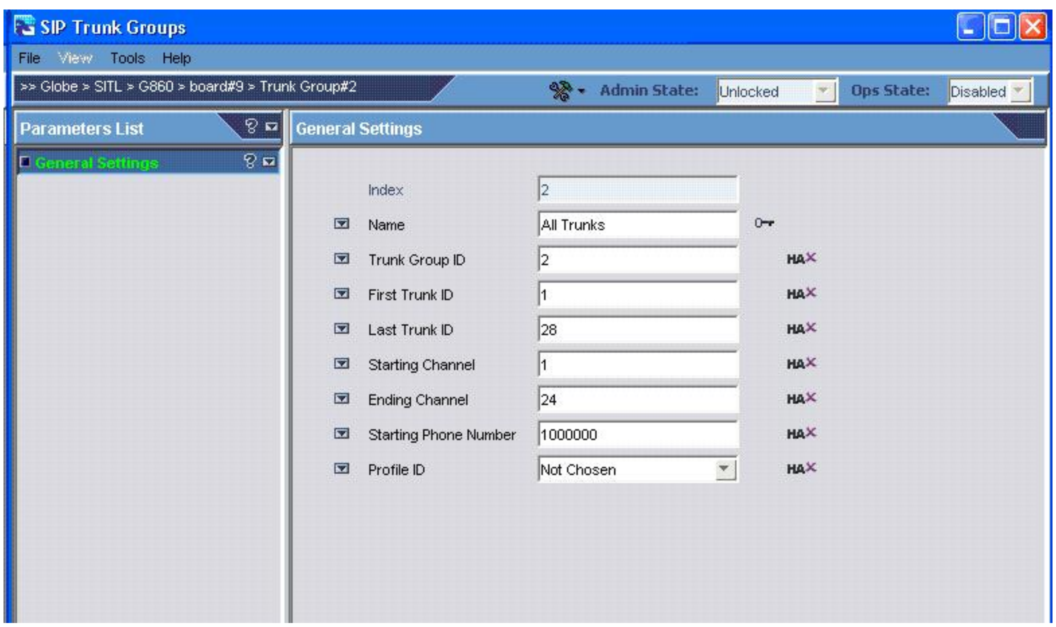
..

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.5.1</td><td>Administer settings to assign profiles to the Avaya G860 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>
<tr><td>4.5.2</td><td>Administer settings for ISDN-PRI trunking between the Avaya G860 Media Gateway and the Public Switched Telephone Network with the following parameters:

Enter All Trunks in the Name field.
Set the Trunk Group ID to 2.
Set the First Trunk ID to 1 (first T1 in the first T3) and the Last Trunk ID to 28; 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 24 (last B-channel in each T1).
Set the Starting Phone Number field to 1000000.

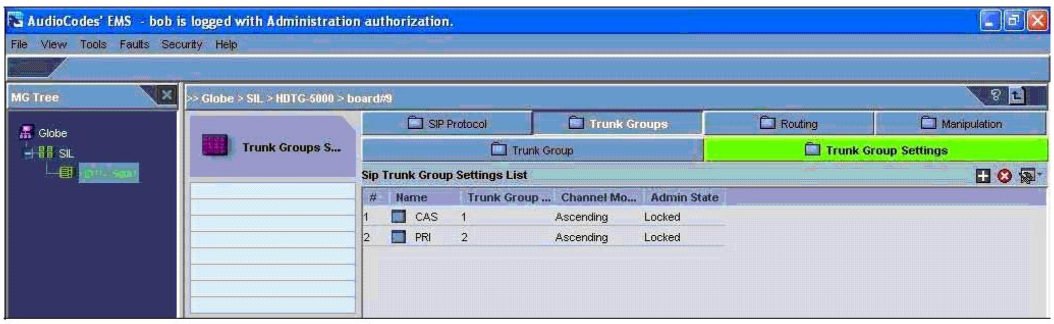


The resultant Sip Trunk Group List is shown below.</td></tr>
</table>

Step	Description
	
4.5.3	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). 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.5.4</td><td>Administer settings to determine the method in which new calls are assigned to B-channels within the ISDN-PRI trunk group provisioned 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
Set the Channel Select Mode to Ascending.

Note: This channel selection pattern, in combination with the logical trunk provisioning enable ascending channel selection over 46 B-channels spread over two physical DS1 connections between the Avaya G860 Media Gateway and the Public Switched Telephone Network. Thus, if one DS1 goes out of service, service will not be impacted for call origination from the Avaya G860 Media Gateway.
The resultant Sip Trunk Group Settings List is shown below.


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

4.6. Administer Call Routing Rules

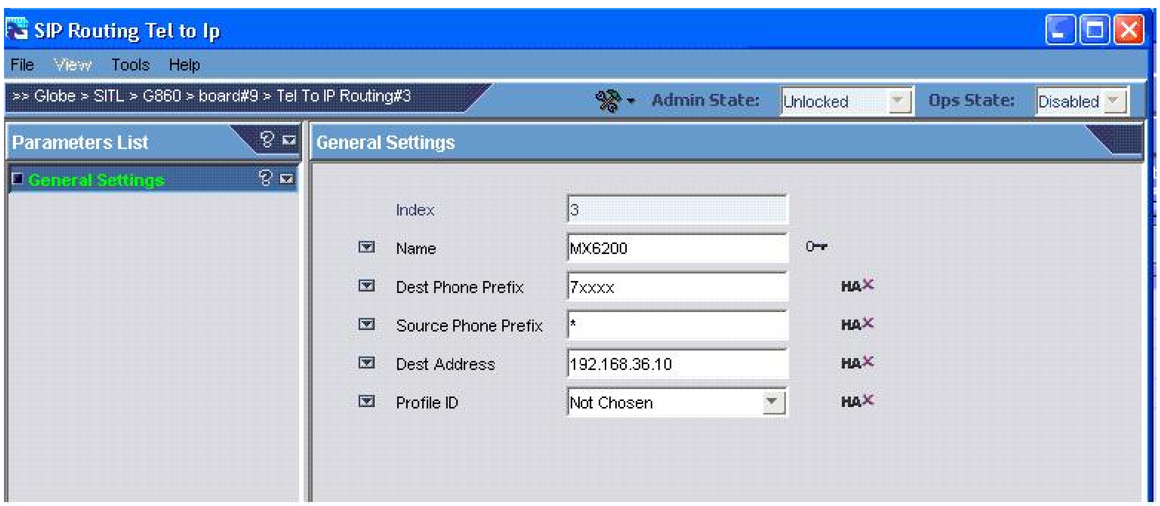
The following steps describe the administrative procedures for administering call routing rules on the Avaya G860 Media Gateway to enable call origination/termination between the Avaya Meeting Exchange Enterprise S6200 Conferencing Server and the Public Switched Telephone Network.

- **Configuring Tel To IP Routing (Avaya G860 Media Gateway to Avaya SIP Enablement Services and Avaya Meeting Exchange Enterprise S6200 Conferencing Server)**

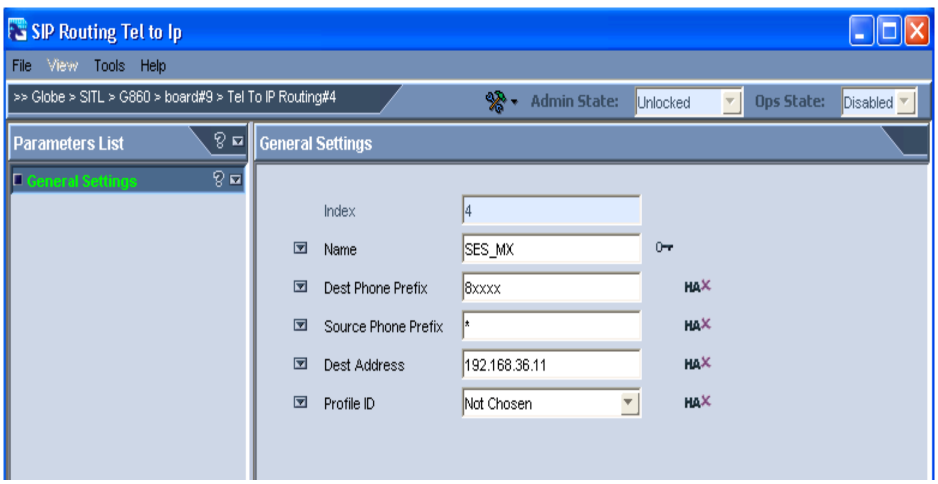
Step	Description
4.6.1	Administer call routing rule(s) that are applied to calls originating from the Public Switched Telephone Network to the Avaya Meeting Exchange Enterprise S6200 Conferencing Server by adding Tel To IP routing rule(s) as follows: • Click on the **SIP tab →Routing** tab. • Click on 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 Public Switched Telephone Network 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 Avaya G860 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.* 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.6.2</td><td>From the SIP Routing Tel to Ip window that is displayed, administer settings to enable Dial-In to the Avaya Meeting Exchange Enterprise S6200 Conferencing Server from the Public Switched Telephone Network 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 Enterprise S6200 Conferencing Server from the Public Switched Telephone Network. For these Application Notes, all calls to the Avaya Meeting Exchange Enterprise S6200 Conferencing Server from the Public Switched Telephone Network are five digits in length with a leading digit of 7. The rule 7xxxx 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 Enterprise S6200 Conferencing Server from the Public Switched Telephone Network.
Enter the IP address of the Avaya Meeting Exchange Enterprise S6200 Conferencing Server in the Dest Address field.
Click on Apply and then Close. (Not shown in this screen shot)



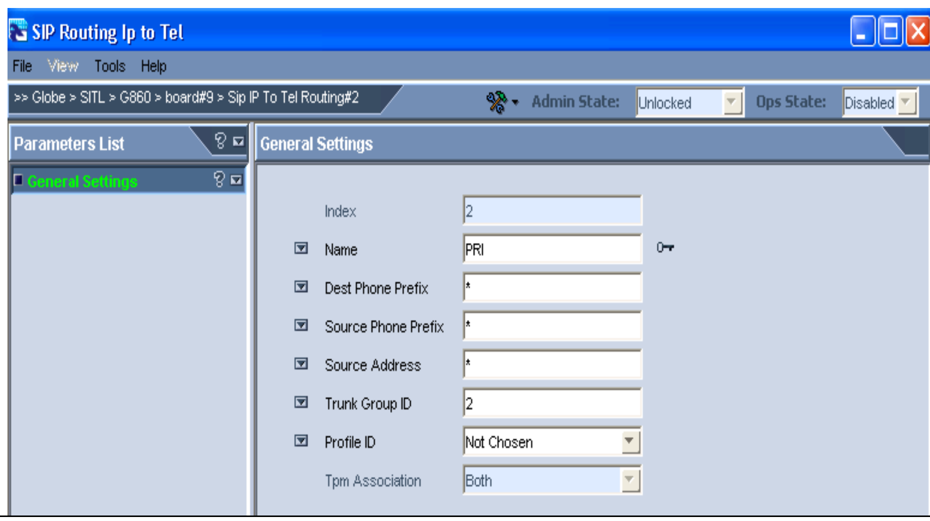
</td></tr>
</table>

Step	Description
4.6.3	From the **SIP Routing Tel to IP** window that is displayed, administer settings to enable dial-in to the Avaya Meeting Exchange Enterprise S6200 Conferencing Server from the Public Switched Telephone Network via Avaya SES 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 Enterprise S6200 Conferencing Server from the Public Switched Telephone Network. For these Application Notes, all calls to the Avaya Meeting Exchange Enterprise S6200 Conferencing Server from the Public Switch Telephone Network are five digits in length with a leading digit of **8**. The rule **8xxxx** 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 Enterprise S6200 Conferencing Server from the Public Switched Telephone Network. • Enter the IP address of the Avaya SES Server in the **Dest Address** field. • Click on **Apply** and then **Close**. 

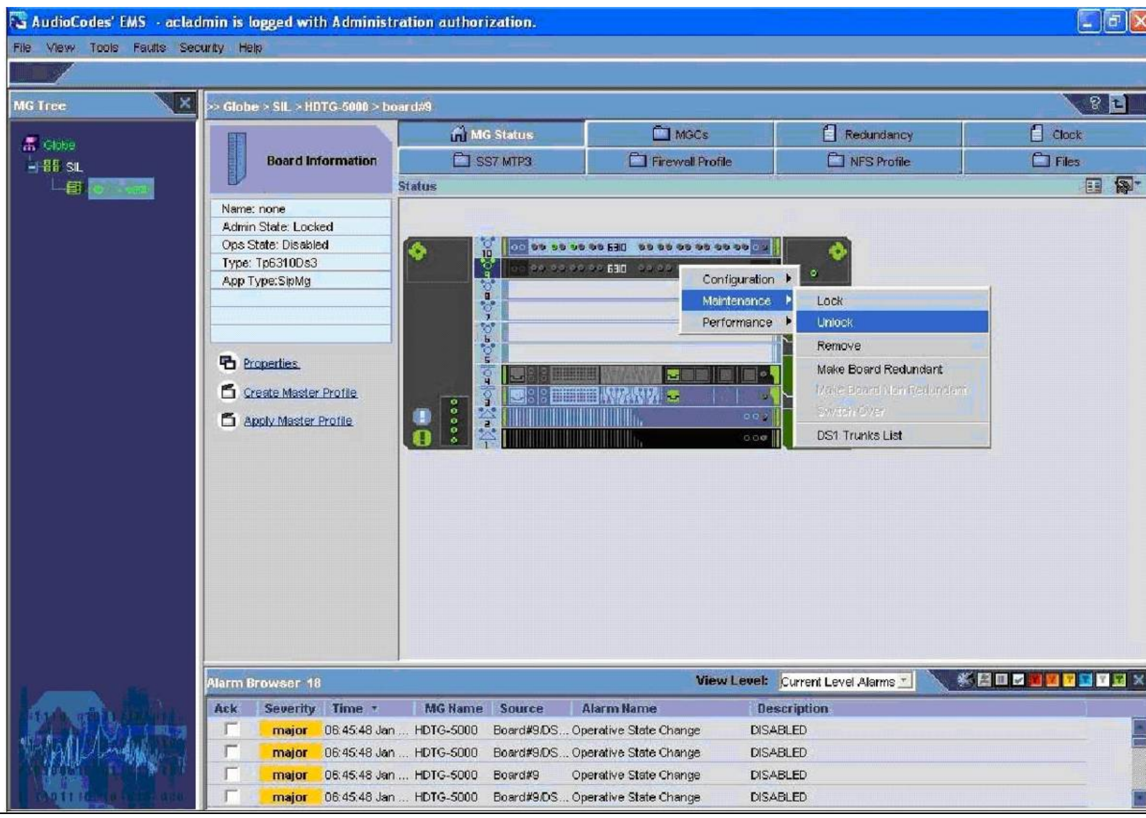
Step	Description
4.6.4	Administer call routing rule(s) that are applied to calls originating from the Avaya Meeting Exchange Enterprise S6200 Conferencing Server to the Public Switched Telephone Network by adding IP To Tel routing rule(s) as follows: • Click on the **Routing** tab. • Click on the **IP 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 provisioned groups of B-channels referred to as trunk group(s). 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 Avaya G860 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 provisioned Trunk Group Settings Table*. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.6.5</td><td>
From the SIP Routing IP to Tel window that is displayed, administer settings to enable dial-out from the Avaya Meeting Exchange Enterprise S6200 Conferencing Server and Avaya SIP Enablement Services to the Public Switched Telephone Network over a PRI 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 Enterprise S6200 Conferencing Server to the Public Switched Telephone Network. For these Application Notes, all calls from the Avaya Meeting Exchange Enterprise S6200 Conferencing Server to the Public Switched Telephone Network via PRI trunking are placed where * is a wildcard and will match any 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 Enterprise S6200 Conferencing Server to the Public Switched Telephone Network.
Enter the Trunk Group ID for the PRI trunk group provisioned in the Trunk Group ID field.
Click on Apply and then Close.



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

Step	Description
4.6.6	The board must be unlocked for the 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).* • 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.* 

4.7 Configuring In-band DTMF and RFC2833 (Out-Of-band) in the Avaya G860 Media Gateway

The following steps describe the configuration of In-Band DTMF and RFC2833 (Out-Of-Band) in the Avaya G860 Media gateway.

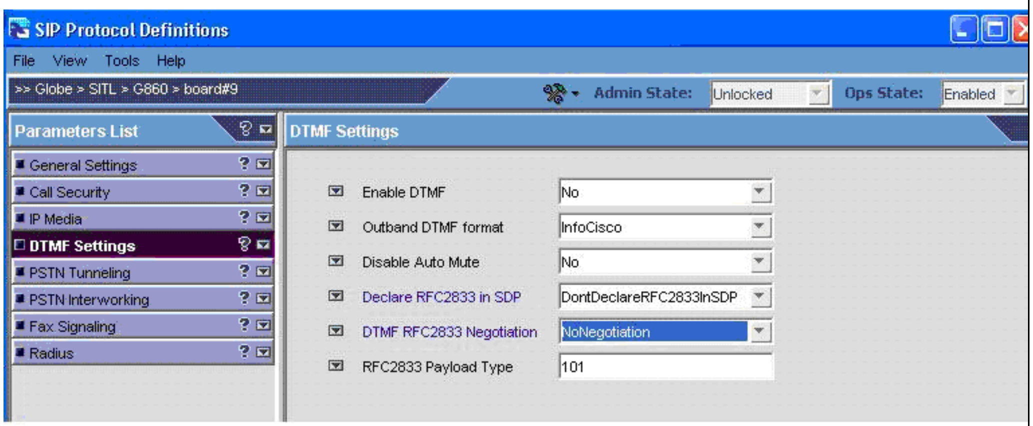
- **Board 9 DTMF settings**

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.7.1</td><td>

Board9->Protocol Settings->DTMF Settings

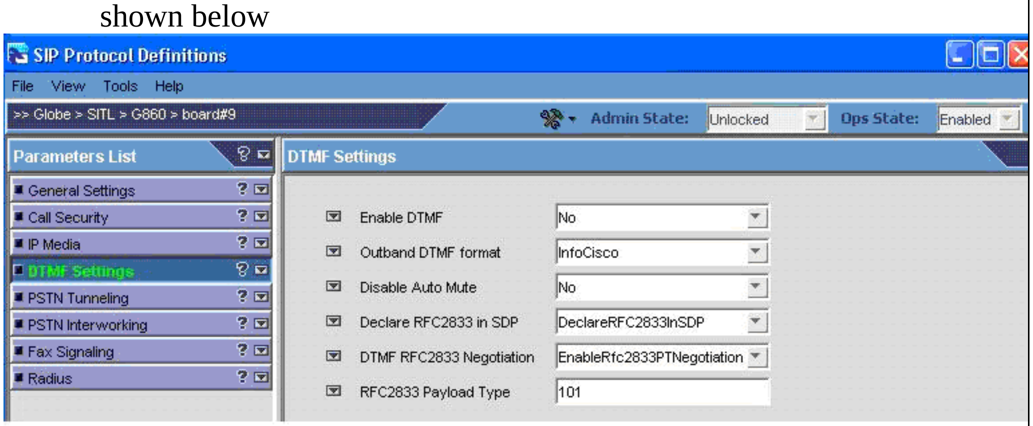
Configuring In Band DTMF

- **Declare RFC2833 in SDP**= “DontDeclareRFC2833inSDP”
- **DTMF RFC2833 Negotiation**=NoNegotiation” as shown below



Configuring RFC 2833 Out Of DTMF

- **Declare RFC2833 in SDP**= “DeclareRFC2833InSDP”
- **DTMF RFC2833 Negotiation**=”EnableRfc2833PTNegotiation” as shown below



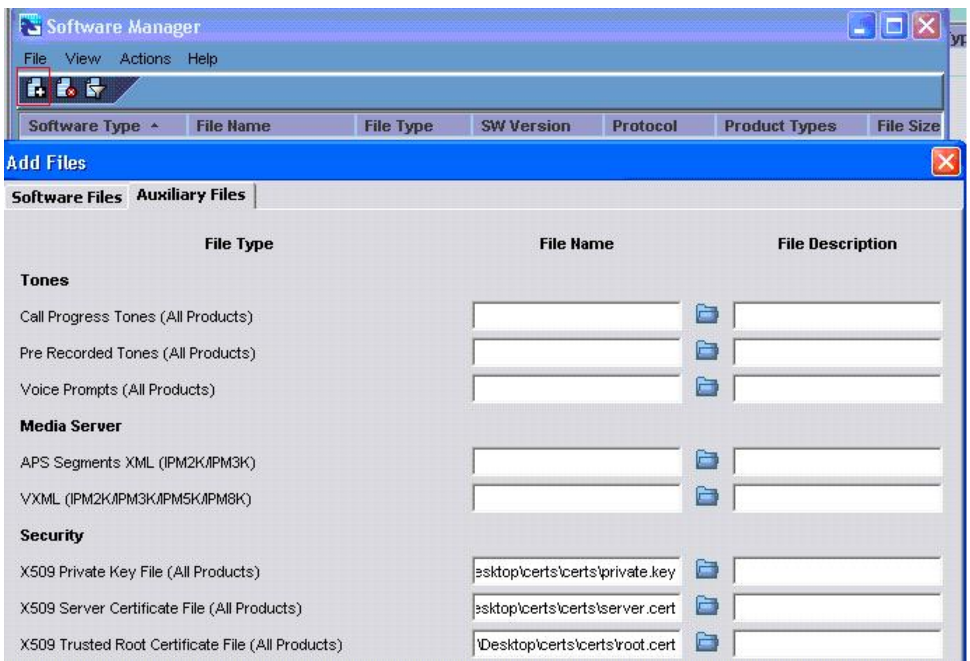
</td></tr>
</table>

4.8 Installing and Configuring Avaya Signed TLS Certificates on the Avaya G860 Media Gateway

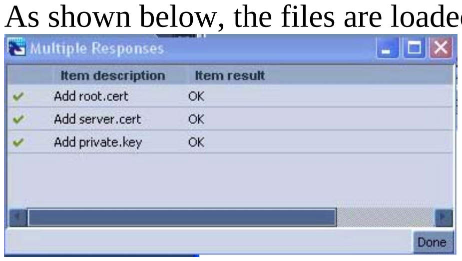
Installing and Configuring TLS certificates on the Avaya G860 Media Gateway enables it to communicate with Avaya SIP Enablement Services and Avaya Meeting Exchange Enterprise S6200 Conferencing Server using TLS.

- **Uploading files**
- **Configuring the certs**

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.8.1</td><td>
The following certificates are required: private.key, server.cert and root.cert. They can be obtained from an Avaya sales representative.
Using the EMS client, on the Menu bar go to Tools →Software Manager. Click on the '+" sign to add a file and click on the Auxiliary Files Tab. Under the Security section, browse and load the files by clicking on Apply as shown below



As shown below, the files are loaded on the Server.



To Configure the files on Avaya G860 Media Gateway
Click on the MG Status→Properties button and select MG
</td></tr>
</table>

Step	Description
	Security Settings as displayed. • Configure the certificates as show below. Press **Apply**. • Connect to the Active SC board via telnet, SSH or RS-232 console. Use Global IP address when connecting via telnet or SSH. • Login as the CLI user with administrative privileges.At the prompt type x509 and press **Enter**. • Wait until expiration date for all configured certificates is calculated and updated. • Now configure **Certification Expiration Date Reminder Days** and **Trusted Root Certificate Expiration Date Reminder Days**. Click **Apply**. • Go Back and click on the **MG Status** button in the EMS navigation bar. • Right click on the desired Media Gateway board and from the popup menu select **Maintenance →Lock** and again unlock the board.(This restarts the board and certificates are applied). • Now go back to the **MG Security Settings** make sure certificates are applied. **Note:** In this case certificates have been loaded and configured on all available boards 

5. Interoperability Compliance Testing

5.1. General Test Approach

The general test approach was to place calls between the Avaya Meeting Exchange Enterprise S6200 Conferencing Server and the Public Switched Telephone Network directly via the Avaya G860 Media Gateway and via Avaya SES utilizing the network configuration displayed in **Figure 1**. The main objectives were to verify the following:

- Dial-In Conferencing:
  - DNIS Scan 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
  - DNIS Scan call function and enter Moderator Code and autonomously invokes a Blast dial to a pre-provisioned dial list of 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.
- All the conference features using DTMF/Touchtone commands
  - *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 codec's were verified: G711MU, G.711ALaw,iLBC
- TLS, UDP and TCP connectivity between Avaya Meeting Exchange Enterprise S6200 Conferencing Server and Avaya G860 Media Gateway directly and via Avaya SES

- In-Band DTMF and RFC2833

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:

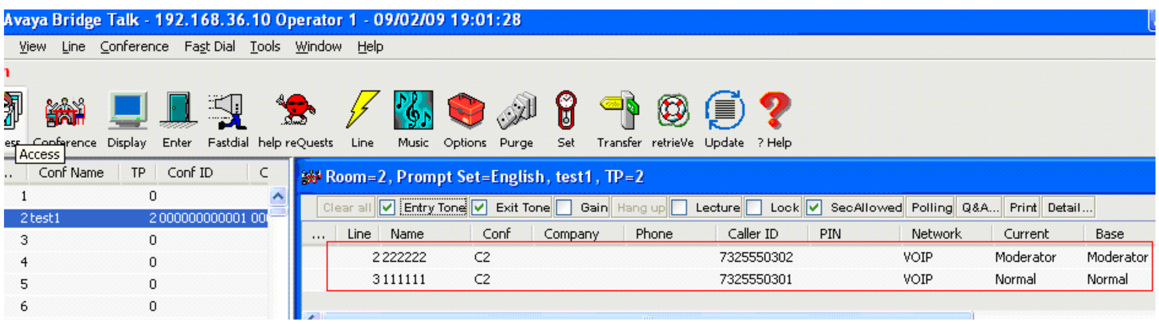
- The Avaya Meeting Exchange Enterprise S6200 Conferencing Server configuration
- Verify that the DS3 and DS1 trunks are up on the Avaya G860 Media Gateway by verifying the icons for those entries on the Trunk & Channel Status screen are green.
- Verify successful inbound and outbound calls between the Avaya Meeting Exchange Enterprise S6200 Conferencing Server and the Public Switched Telephone Network via Avaya SES

Step	Description

6.1	Verify all conferencing related processes are running on the Avaya Meeting Exchange Enterprise S6200 Conferencing Server as follows:
	• Log in to the Avaya Meeting Exchange Enterprise S6200 Conferencing 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.
	`[sroot@MX-G860 ~]# dcbps`
	`12803 ?        00:00:01 initdcb`
	`12920 ?        00:00:00 log`
	`12923 ?        00:00:00 bridgeTranslato`
	`12924 ?        00:00:00 netservices`
	`12931 ?        00:00:00 timer`
	`12932 ?        00:00:00 traffic`
	`12933 ?        00:00:00 chdbased`
	`12934 ?        00:01:09 startd`
	`12935 ?        00:00:00 cdr`
	`12936 ?        00:00:00 modapid`
	`12937 ?        00:00:00 schapid`
	`12938 ?        00:00:02 callhand`
	`12939 ?        00:00:00 initipcb`
	`12943 ?        00:00:00 sipagent`
	`12944 ?        00:00:00 msdispatcher`
	`12945 ?        00:00:00 serverComms`
	`12946 ?        00:06:59 softms`
	`12956 ?        00:00:21 softms`
	`12957 ?        00:09:33 softms`
	`12960 ?        00:08:41 softms`
	`12961 ?        00:04:02 softms`
	`12969 ?        00:10:44 softms`
	`13005 ?        00:00:00 cdrland`
	` 3207 ?        00:00:00 postmaster with 25 children`

6.1. Verify Call Routing

Step	Description
6.1.1	Verify end to end signalling/media connectivity between the Avaya Meeting Exchange Enterprise S6200 Conferencing Server and the Public Switched Telephone Network directly via the Avaya G860 Media Gateway and via Avaya SIP Enablement Services. This is accomplished by placing calls to and from the Avaya Meeting Exchange Enterprise S6200 Conferencing Server. This step utilizes the Avaya Bridge Talk application to verify calls to and from the Avaya Meeting Exchange Enterprise S6200 Conferencing Server are managed correctly, e.g., callers are added/removed from conferences. This step will also verify the conferencing applications provisioned • From an endpoint on the Public Switch Telephone Network, Dial **70001/80001** to enter a conference as **Moderator** (with passcode) while simultaneously invoking the associated Auto Blast dial feature for this conference • If not already logged on, log in to the Avaya Bridge Talk application with the appropriate credentials • **Double-Click on the** highlighted **Conf #** to open a **Conference Room** window • Verify conference participants are added/removed from conferences by observing the Conference Navigator and/or Conference Room windows.



7. Conclusion

These Application Notes presented a compliance-tested solution comprised of the Avaya Meeting Exchange Enterprise S6200 Conferencing Server and the Avaya G860 Media Gateway. This solution enables connectivity between the Avaya Meeting Exchange Enterprise S6200 Conferencing Server directly with Avaya G860 Media Gateway and via the Avaya SIP Enablement Services.

There is issue to note from testing.Codec G.711Alaw/Mu was used. iLBC codec was not supported in this test configuration

8. Additional References

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

- *Meeting Exchange S6200 5.1 Administration and Maintenance S6200/S6800*
- *Avaya Meeting Exchange Enterprise Groupware Edition Version 5.1 User's Guide for Bridge Talk*
- Avaya G860 Media Gateway 5.2 *Administration and Maintenance*
- *Avaya SIP Enablement Services 5.1.1 Administration and Maintenance*

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