

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 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.

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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)

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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	M5k10
Software Version	5.2.73
Board Type	Tp6310Ds3
EMS Server	5.2.60
EMS Client	5.2.60
Avaya SIP Enablement Service (SES)	5.1.1 build 415.1
Avaya Meeting Exchange Enterprise S6200 Conferencing	5.1 build 161
Server	

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.

Step	Description						
3.1.2	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></sip:6000@192.168.36.10:5061;transport=tls> 						
	<i>Note: Configure the following if using TCP and UDP</i>						
	respContact= <sip:6000@192.168.13.101:5060;transport=tcp></sip:6000@192.168.13.101:5060;transport=tcp>						
	respContact= <sip:6000@192.168.13.101:5060;transport=udp></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.						

Step	Descr	iption
	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:
	b)	* sip:\$1@10.1.2.63:5061;transport=tls G860
	•	 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.

Step	Description							
3.2.1	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:							
	chutil add (dnis) (rg) (msg) (ps) (ucps) (func) [-] (ln) -c							
	<pre><cn>], where the variables for add command is defined as follows:</cn></pre>							
	o < dnis > DNIS							
	o < rg > Reservation Group							
	o < msg > Annunciator message number							
	o <ucps> Use Conference Prompt Set (v/n)</ucps>							
	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							
	In this sample configuration:							
	S6200App-> cbutil add 70001 0 1 1 n scan cbutil							
	Copyright 2004 Avaya, Inc. All rights reserved.							
3.2.2	At the command prompt, enter cbutil list to verify the DNIS entries provisioned.							
	[areatewy CP60 gapfig]# abutil ligt							
	[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							
	U U 20001 0 247 1 או פריאוז האיזיהס אוע-פרפ_							
	G860 0							

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 . List Hosts 						
	Commands Host Type SES Version Edit Map Go-To Test-Link Delete 192.168.36.11 SES combined home-edge SES-5.1.1.0-415.1						
	 Migrate Home/Edge 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 Ist Host Address Map - Microsoft Internet Explorer Address @ https://192.166.36.11/cg-bin/medmin/dg/editeddressmap/listmap?id=18cmd=Continue Search Web + W MSN + S + S + S + S + S + S + S + S + S +						
	Integrated Manage SIP Server Manage						
	Top Setup List Host Address Map Users Address Map Priorities Address Map Priorities Host Adgregator Edit Delete to_G860 Conferences Edit Delete to_G860 Emergency Contacts Edit Delete To_MX Edit Delete To_MX Edit Delete sip:\$(user)@192.168.36.10:5061;transport=tls Add Another Map Add Another Contact Delete Group Edit Delete To_MX Edit Delete sip:\$(user)@192.168.36.10:5061;transport=tls Add Another Map Add Another Contact Delete Group Edit Delete To_MX Edit Delete sip:\$(user)@192.168.36.10:5061;transport=tls Add Map In New Group Add Another Contact Delete Group Itist Migrate Home/Edge Add Map In New Group						

Step	Description						
Step 3.3.2	Description Adding the MXS6200 Users Address Map Priorities Adjunct Systems Aggregator Certificate Management Conferences Emergency Contacts Export/Import to ProVision Hosts IM logs Communication Manager Extensions Servers Communication Manager Extensions Server Configuration SIP Phone Settings Survivable Call Processors System Status Trace Logger Trusted Hosts Add List) Con Edit Edit	ferenc: mands Delete Delete	ing Server a	nd G860 as ' Trusted by Host 192.168.36.11 192.168.36.11	Comment G860 MX	
	List						

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. 					
	Avaya Bridge Talk login Sign-In: Password: Bridge: Operator: Next available OK Exit					

Step	Description					
3.4.2	Provision a dial from the Avaya • From the	list that is Meeting E Avaya Br	utilized fo xchange idge Talk	or Dial-Out (Enterprise Se Menu Bar, o	e.g., Blast dial a 5200 Conference click Fast Dial -	and Fast dial) ing Server. →New .
	avaya Bridge Taik - T	92.168.36.12 U	Uliadam Ula	5/02/09 10:20:04		
	File view Line Conference	Past Dial Tools	window Help			
		Edit	S /	Music Options Dura) 💡 📲 🐯 (
	Access Conference Display	Hold Dial		Music Options Purge	: Set Transfer retrieve L	Jpdate / Help
	C Cont Name TP	G				
	2 0	2				
	3 0					
	4 0					
	5 0					
	6 0					
	7 0					
	8 0		R Pending	0110110		
	9 0			Queue		
	10 0		Line	Name Progr	Company Phone	Caller ID PIN
	11 0					
	12 0					
	13 0					

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. 					
	Conferee List	PT ne: Test1 Option ▼ Display As Entered	al Access Code: 1000000	Direct	y to Conf	
	Name	Company	Moderator	Q&A Priority	Telephone	
	Ep1	Avaya			50302	
			(Save Canc	el <u>P</u> rint H	

Step	Description
3.4.4	Provision a conference with Auto Blast enabled.
	• From the Avaya Bridge Talk Menu Bar, click View \rightarrow Conference
	Scheduler.
	File View Line Conference FastDial Tools Window Help
	fain Enter Queue Image: A state of the
	Help Requests
	Acces V ToolBar Fastdial help reQuests Line Music Options Purge Set Transfer retrieVe Update ? Help
	Cc Conference Navigator 🕨 Conf ID Confirm # Mod Code Conferee C Music Source
	Update 0 Off
	Reports 0 Off
	Conference Scheduler
	6 0 Off
	7 0 Off
	8 U Off 9 0 Off
	10 0 Off
	11 0 Off 12 0 Dff
3.4.5	From the Conference Scheduler window, click File \rightarrow Schedule Conference .
	🗳 Conference Scheduler
	Eile Edit View Window Help
	Schedule Conference

Step	Description									
3.4.6	From the Sch	edule Confer	ence	window th	hat i	is displa	iyed, p	rovision a	a cont	ference
	as follows:					-				
	• Enter	a unique Con	feree	e Code to a	llov	w partic	ipants	access to	this	
	confe	rence.								
	• Enter	a unique Moc	lerat	or Code to	o all	ow part	icipant	s access	to this	S
	confe	rence with mo	derat	tor privileg	ges.	Enable	moder	ator acces	ss wit	h a
	passee	ode for this co	onfere	ence call by	y co	onfigurir	ng the f	following	:	
		Note: This c	confe	rence rema	ins	open fo	or parti	cipants to	o ente	er as
		either mode	rator	or partici	pan	t by ente	ering th	he approp	oriate	code
		when promp	oted.		•		C*			
	• Enter a descriptive fabel in the Conference Name field.									
	• Admi	nister settings	to er	hable an AI	uto	Blast d	ial by s	setting Ai	ito	
	Blast/	Manual deper	iding	on this tes	st.	1.1.		1. D: 1	r : 1	
	0	[Not Shown	Sel	ect a dial l	ist t	by clicki	ng on i	the Dial I	List b	utton,
		select a dial	list f	rom the Ci	reat	e, Selec	t or Ea	lit Dial L		ndow
		that is displa	ayed,	and click	on	the Sele	e ct butt	<i>on</i> (10 ve	erify	Dial
	XX 71	out and Blas	st D18	ul out).			- 1			
	• when	finished, clic	k on	the save b	utto	on on the	e botto	m of the	scree	a.
		ation			í		1			1000
	Status:	ENABLED	Mod	e: 			Conference Type:		DAILY	×
	Confirmation No.:	1	Cont	erence ID:			weekena:		YES	×
	Name:		Billin	g Code Prompt	DISA	BLED 📉			-	
	Telephone:		Acco	unting Code:	OFF	<u>×</u>	Start Date	(dd/mm/yyyy)		
	Sign-in Name:	*	Secu	irity Passcode:	OFF	<u>×</u>	End Date (dd/mm/yyyy):	_	
	Res Group:	0	Char	nge Cont Opt:	ON	<u>×</u>] 1. –	11-1	(
	Conferee Code:	111111	Oph	leip Available:	ON	×	Name Reco	ord/Play:	OFF	×
	Moderator Code:	222222	Block	(Dialout:	OFF	~	NRP Annur	nciator:	Bro	Wse
	Conference Name:	test1	Auto	Blast:	Manu	Jal 🔽	PIN Mode:		OFF	~
	Dial List	Test1	Blast	: Annunciator:242		Browse	PIN List:			
	Conference Featur	es		i		-		4	1	
	Start Time:	00:00	(77.8	End Time:		00:00		Code Duration	ı: 0	
	Entry Tone:	Tone & Messag	ie 💌	Exit Tone:		Tone & Mes	sage 💌	Maximum Line:	s: 10	
	Hang up:	ON	×	Music:		M1	~	Security:	ON	~
	Auto Extend Duration: OFF			Auto Extend Port	:\$;	OFF	~			
	Prompt Set:	English	×	Conference View	er:	NO	Y			1
				(<u>S</u> av	/e Car	ncel	Prev De	xt 🛛	Help

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. <i>Note:</i> 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									
	Carl Codes' EMS - fred	<mark>is logged with Administra</mark> urity Help	ition authorizati	on.						×
	MG Tree	>> Globe > SITL						A 💿	← - → -	t ?
	Globe		MGs List							
		Region	Name EII G860	IP Address 10.1.2.62	Version 5.2.73	Product Type MEDIANT 5000	Protocol	Admin State Unlocked	Op State Enabled	Master
	- 🖽 G860	Name: SITL								
		Total:1	_							
		#MPs:0/0 OK	-							
		#Others:0/0 OK								
			_							
			-1							
		Alarm Browser 350			U	fiew Level: Curren	t Level Alarms 🗾	2 🐇 🖻		T T X
	10 Mar. 1	Ack Severity T	ime * MG	Name Sourc	e Alarn	n Name	Description	<i>8</i>		10.00
	ALL ALL ALL	Clear 03	03:19 Mar G860		Operati	onal Info	Backup proces	ss was completed		.
	The second second second	major 07:	15:26 Mar G860	Board#9	WTru Trunk A	larm (XMT AIS)	PSTN alarm on	Trunk (XMT AIS)		
	The AMERICAN STREET	major 07:	15:26 Mar G860	Board#9	MTru Trunk A	larm (XMT AIS)	PSTN alarm on	Trunk (XMT AIS)		
	A HAVE TO PARTY	major 07:	15:26 Mar G860	Board#9	9/Tru Trunk A	larm (XMT AIS)	PSTN alarm on	Trunk (XMT AIS)		1
	A11001110 10 00000		45-00 14-0 0000	Description		0/4T AICS	DOTNI ALAMA AN	T		
		P								

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.

Step	Description							
4.1.1	From the med • Select be con • Click	lia gateway the entry co figured. on Propert	list in the I orrespondin ies .	MGs List p ng to the A	oane that is vaya G860	s displayed) Media Ga	: iteway 1	to
	RadioCodes' EMS - fred i File View Tools Faults Sec	is logged with Administra urty Help	ation authorization.					
	MG Tree	>> Globe > SITL				A 📾	+ - ⇒ - 1.	. ?
	E Globe	-	MGs List					
		Region	Name IP Ad	dress Version	Product Type Prot	tocol Admin State	Op State	Master
	- El ceeo	Name: SITL Totak:1 #M65:11 OK #Others:0:0 OK						
		Alarm Browser 350		Vie	w Level: Current Level A	Alarms 🗾 🛛 🗹 🖾		
	Land Bank	Clear 03	:03:19 Mar G860	Operationa	al Info Back	kup process was completed		
	TILLO, UTOIT TOKOT I	major 07	:15:26 Mar G860	Board#9/Tru Trunk Alan	m (XMT AIS) PSTN	N alarm on Trunk (XMT AIS)		
	12 million and	major 07	15:26 Mar G860	Board#9/Tru Trunk Alan Board#9/Tru Trunk Alan	m (XMTAIS) PSTN m (XMTAIS) PSTN	N alarm on Trunk (XMT AIS) N alarm on Trunk (XMT AIS)		
	111101110 101010 010		46.00 No. 0000	Description Toronto Alex		N down on Townh Old IT Allon		

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 maki	ng any chan	ge to the conf	figuration of	the TP6310 bc	oard, the					
	board must be	e locked.									
	• On the MGs List pane, double-click on the row corresponding to the										
	Avaya	a G860 Med	ia Gateway.								
	0	• <i>Note: The MG Status tab will be highlighted and the Status</i>									
		pane will o	open, depictin	ıg a replica o	f the front pan	el of the					
		Avaya G8	60 Media Gat	teway chassis	s. Board slots o	are					
		numbered	from 1 to 10 j	from bottom	to top on the le	eft side.					
	For these Application Notes, installed gateway boards										
	include the TP6310 DS3 boards in slots 8, 9, and 10.										
	• Click	on the activ	e TP6310 boa	ard shown in	black, and use	mouse					
	buttor	n to selec	t Maintenano	ce Lock.		-					
	• [Not S	Shown] To c	onfirm Lock,	click Yes in 1	the confirmation	on window					
	that is	s displayed.									
	<i>Note:</i> If there is a single TP6310 board in the Avaya G860 Media Gateway,										
	locking this b	oard remov	es it from serv	vice and is se	rvice impactin	ıg.					
	R AudioCodes' EMS - actadmin is logged with Administration authorization.										
	MG Trine X by Globar 5 Stl > HDTC: 5880 > hoard#9										
	A Globe		MG Status	Clock							
		Board Information	Status	Firewall Profile	NFS Profile	Files					
		Name: none Admin State: Unlocked									
		Ops State: Enabled Type: Tp6310Ds3		55 530 55 55 55 55 55 55 55 55 55 55 55 55 55							
		App Type:SipMg	100 B	Maintenar Performer	nce F Lock						
			-		Femove-						
		S Create Master Profile.			Meebord Conservant						
				000 000	DS1 Trunks List						
		Alarm Browser 22		View Level:	Current Level Alarms 🗾 🔪 🚳 🗷						



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.



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. 									
	🖼 Board6310 Parameters Provisi	oning								
	File View Tools Help									
	ss Globe's Sic s Hord-Solid s boardwa			* Admin State: Locked Ops State: Disabled						
	Parameters List 8 🕫	General Settings								
	General Info Constraints Constraints Constraints	Board EODN	none	0-						
	Setup Files ? 🗹	D Address 1	10.1.2.63	0-						
	Call Control ? 🗹	MAC Address 1	00908t0at07c	07						
	Voice ? 🗹			0						
	Fax / Modem ? 🗹	Madia ID1 Address	0.000	(m						
	IP Media Settings ? 😒	Mount Printers	0.0.0.0	-						
	IP Media APS ? 🖸	APD Table	psin							
	Diagnostics Z	ARP Table	070	~						
		ARP Table Wax primes	Disable v							
		Made APD Cooke supplie	Enclusion Concernent							
		Committe								
		Security	hum							
			Not Chusen							
		SRIP Media Security	pisable *							
		SSL/ILS Negotiation	JILSVIONY							
		Radius								
		Enable RADIUS	Disable							
		RADIUS Accounting Server IP Address	0.0.0	0						
		RADIUS Accounting Port	1646	0-						
				Apply Refresh Close						
	Profile Management									
		Name P Board6310 Profile	rofiles Save Profile No Profiles 🔨 Save	Parameters						

From the	 Click on Voice under Parameters Frovisioning 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. 									
Board6310 F File View Tools 35 Globa 5 SLI> H Parameters List 6 General Setting 8 Setue Files 6 Call Control 9 Viaios 9 PSTN 9 Fax: Modem 9 P Media Setting 9 P Media APS 9 Diagnostics 9 Board Debug To	Parameters Provisionin PHole DIG-5000 > board#5 2 0 Vaic 2 0 V	B Image: Second Secon	0 PCMU JLaw Disabled Enable Disable Finable 0 0 0 Disabled 10 7 TransparentDTMF 0		Admin State: _ocked _ Ope State: _isable =					
Profile Managen	nent	PCM Input Gain RFC2833 Receive Payloo: Type RFC2833 Transmt Payloo: Type Voice Stream Upload Method Voice Streaming	0 96 96 post Disable Profiles Sa	× ×	0 O O Refresh 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

Step	Description											
4.3.1	Administer se	ttings for	a DS3 trunk to	enable conne	ctivity to the l	Public						
	Switched Tele	ephone No	etwork as follo	ws:								
	• [Not S	hown] Do	ouble-click on t	the locked TP6	310 board in	the Status						
	pane.	-										
	Click	on the DS	S3 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											
	<i>Note:</i> The D55 Status pane aisplays the status of each of the 3 DS3 interfaces on this hoard											
	interfaces on	this board	<i>d</i> .									
	File View Tools Faults Ser	Imin is logged with	Administration authorization.	-	-40							
		unity Help										
	MG Tree	>> Globe > SIL > HDT	G-5000 > board#9			8 t						
	Globe		DS3	DS1 Trunks	C SIP	Signaling						
			DS3 Status			P						
			1 B none line	Source Admin State Op State Locked Disabled	clear							
			2 🖞 none line	Locked Disabled	clear							
			3 🗄 none line	Locked Disabled	clear							
	6	Alarm Browser 18		View Lo	evel: Current Level Alarms 🗾							
	The state of the	Ack Severity	Time * MGName Sourc	e Alarm Name	Description							
	A A A A A A A A A A A A A A A A A A A	major (16:45:48 Jan HDTG-5000 Board#3	NDS Operative State Change	DISABLED							
	A WARE CAN WE A	T major 0	06:45:48 Jan HDTG-5000 Board#	Operative State Change	DISABLED							
	A 12011 Ideapartit Inte	T major 0	06:45:48 Jan HDTG-5000 Board#9	DS Operative State Change	DISABLED							

Step	Description								
4.3.2	From the DS	1 Carriers Li	st pane i	hat is d	lisplayed,	, pro	vision a DS	S1 on this	5
	DS3 interface	by double-cl	icking of	n its ent	try in the	list.			
		-)			j				
	The Audio Control Diff. And	1. J	and the strengthere					(m)	
	File View Tools Faults Ser	curity Help	JII dutiivrizativii.						
	MG Tree	>> Globe > SITL > G860 > bo	oard#9				A	🔯 🔶 - 🔶 - 1.	. ?
	Clobe		DS1 Carriers List						\$
		DS1 Carriers List	# Name	Protocol	Admin State	Op State	e DS1 Path	Master Profile	
	L-0)		1 🛡 Trun	T15Ess10lsdn	Unlocked	Enabled	none		
		-	2 🔽 Trun	T15Ess10lsdn	Unlocked	Enabled	none		
			3 💭 Truni	T15Ess10lsdn	Unlocked	Enabled	none		
			4 🔽 Trun	T15Ess10lsdn	Unlocked	Disabled	none		
			5 🔽 Trun	T15Ess10lsdn	Unlocked	Disabled	none		
			b 🔽 Trun	T15Ess10lsdn	Unlocked	Disabled	none		-
				115Ess10isan	Unlocked	Disabled	none		
				T15Ess10isdn	Unlocked	Disabled	none		-
			10 Trun	T15Ess10isuri	Unlocked	Disabled	none		
			11 True	T15Ess10isuri	Unlocked	Disabled	none		-
			12 Trun	T15Eee10ledn	Liniocked	Disabled	none		
			13 Trun	T15Ess10lsdn	Unlocked	Disabled	none		
			14 Trun	T15Ess10lsdn	Liniocked	Disabled	none		
		40-	15 Trun	T15Ess10lsdn	Unlocked	Disabled	none		
			16 🔽 Trun	T15Ess10lsdn	Unlocked	Disabled	none		
		🥑 History PM 🛛 📊 -	17 🗖 Trun	T15Ess10lsdn	Unlocked	Disabled	none		
			18 🜄 Trun	T15Ess10lsdn	Unlocked	Disabled	none		
			19 🛄 Trun	T15Ess10lsdn	Unlocked	Disabled	none		_
			20 🔽 Trun	T15Ess10lsdn	Unlocked	Disabled	none		
			21 👿 Trun	T15Ess10lsdn	Unlocked	Disabled	none		
			22 🗾 Trun	T15Ess10lsdn	Unlocked	Disabled	none		
		Alarm Browser 350			View Level:	Current Let	vel Alarms 🗾 📃 🛃	S 🕮 🖬 🔽 🔳 🔽 🔽	7 7 ×
	Lun Mas	Ack Severity Time	e 🔹 MG Name	Source	Alarm Name		Description		
	A DAY MAN	major 07:15	:26 Mar G860	Board#9/Tru.	Trunk Alarm (XMT Als	5)	PSTN alarm on Trunk (XMT A	IS)	
	PUTION OF DET LIGHT AND	major 07:15:	:26 Mar G860	Board#9/Tru.	Trunk Alarm (XMT Als	S)	PSTN alarm on Trunk (XMT A	IS)	
	Malal A Langeron	major 07:15:	26 Mar G860	Board#9/Tru.	Trunk Alarm (XMT Als	S)	PSTN alarm on Trunk (XMT A	IS)	
	A HAVE A PARTY	major 07:15:	26 Mar G860	Board#9/Tru.	Trunk Alarm (XMT Als	5) I	PSTN alarm on Trunk (XMT A	IS)	
	LANDERING IN AND AND			D	T 01 2204T 616		DOTNI -I TI. (2/44T A		
1									

Descrip	otion											
From th	ne Gene	ral In	fo par	e, in the Tru	nk Parameter	s Pro	visionir	ng wii	ndow			
that is d	lisplaye	d: Cor	nfigure	e as required	by service prov	ider		U				
•	Admini	ster se	ettings	to enable con	nnectivity with	the P	ublic Sv	vitch				
r	Telepho	one Ne	etwork		5							
	<i>Note:</i> Obtain configuration details regarding the setting											
		reauir	ed for	this connect	ion to the Publi	c Swi	tched Te	eleph	one			
		Netwo	ork fro	m the service	provider. The	entrie	s for thi	is trui	ık			
		corres	pond	to a T1 PRI c	onnection betw	veen ti	he Avay	a G8(50			
		Media	Gate	wav and the l	Public Switched	l Tele	phone N	Vetwo	rk.			
							r · · · · · · ·					
Trunk P	arameters P	Provisioni	ng					1				
File View	Tools Help											
>> Globe > Sl	IL > HDTG-5000	D > board#9	> trunk#1		🛞 ★ Admin State:	Locked	💽 Ops	State:	Disabled 📩			
Parameters	s List	<u>ଚ</u> ଜ	General	Info								
🗖 General In	nfo	<u>م</u> 8										
ISDN/DPNS	S	? 🗹		Trunk Number	1							
• CAS		* M		Is Available	Yes	<u> </u>						
				DS1 Path	none	REAL PROPERTY						
				Trunk Name	Trunk#1							
				Protocol Type	None	•	0					
				Framing Method Type	T1FramingEsfCrc6	<u>•</u>	0 1					
			9	Trace Level Type	NoTrace	*	0					
				Line Build-out Loss	db0	<u>•</u>						
				Line Code	þ8ZS	<u>*</u>						
				Clock Reference Priority	0		0 -4					
1			1				-	-				
-							Apply	Refresh	Close			
Profile Man	agement											
		đ	Name	Profiles	Save Profile Apply to all	Paramet	ers					
			Trunk Profile	none 🗾	Save Apply to all	Show	2					

Step	Description						
4.3.4	From the DS this DS3 inter	I Carriers Li	st pan le-clic	e that is d king on it	isplayed, s entry in	provision the list.	the third DS1 on
	RadioCodes' EMS - fred File View Tools Faults Se	is logged with Administratio curity Help	m authorizati	ion.			
	MG Tree	>> Globe > SITL > G860 > bo	oard#9				fi 🖻 🕈 * 🕈 * 1 ?
1	📕 Globe	-	DS1 Carriers	List			<u>ها</u> -
		DS1 Carriers List	# Nar	me Protocol	Admin State	Op State DS1 Pa	th Master Profile
				Trunk T15Ess10isdn	Unlocked	Enabled none	
			3	Trunk T15Ess10lsdn	Unlocked	Enabled none	
			4 👿	Trunk T15Ess10lsdn	Unlocked	Disabled none	
			5 🗾	Trunk T15Ess10lsdn	Unlocked	Disabled none	
			6 🗾	Trunk T15Ess10lsdn	Unlocked	Disabled none	
			7 🗾	Trunk T15Ess10lsdn	Unlocked	Disabled none	
			8 🜄	Trunk T15Ess10lsdn	Unlocked	Disabled none	
			9 🗾	Trunk T15Ess10lsdn	Unlocked	Disabled none	
			10 🗾	Trunk T15Ess10lsdn	Unlocked	Disabled none	
			11 🗾	Trunk T15Ess10lsdn	Unlocked	Disabled none	
			12 📃	Trunk T15Ess10lsdn	Unlocked	Disabled none	
			13 📃	Trunk T15Ess10lsdn	Unlocked	Disabled none	
			14 🜄	Trunk T15Ess10lsdn	Unlocked	Disabled none	
		11 21	15 🗾	Trunk T15Ess10lsdn	Unlocked	Disabled none	
		H	16 📃	Trunk T15Ess10lsdn	Unlocked	Disabled none	
		III History PM	17 🗾	Trunk T15Ess10lsdn	Unlocked	Disabled none	
			18 🜄	Trunk T15Ess10lsdn	Unlocked	Disabled none	
			19 🗾	Trunk T15Ess10lsdn	Unlocked	Disabled none	
			20 📃	Trunk T15Ess10lsdn	Unlocked	Disabled none	
			21	Trunk T15Ess10lsdn	Unlocked	Disabled none	
			22	Trunk T15Ess10lsdn	Unlocked	Disabled none	
		Alarm Browser 350			View Level:	Current Level Alarms 🗾	
	L 1	Ack Severity Time	e ▼ MG	Name Source	Alarm Name	Description	
	A A A	major 07:15:	26 Mar G860	Board#9/Tru.	Trunk Alarm (XMT AIS)) PSTN alarm on 1	Frunk (XMT AIS)
		major 07:15:	26 Mar G860	Board#9/Tru.	Trunk Alarm (XMT AIS)) PSTN alarm on 1	Frunk (XMT AIS)
	Mar Anna	major 07:15:	26 Mar G860	Board#9/Tru.	Trunk Alarm (XMT AIS)) PSTN alarm on 1	Frunk (XMT AIS)
	the fact in the second	major 07:15:	26 Mar G860	Board#9/Tru.	Trunk Alarm (XMT AIS)) PSTN alarm on 1	Frunk (XMT AIS)
	100001110 10.000.000	1 014C		Decode0/7-1	Touch along Other all?	DOTN slama and	
		P					

Step	Description					
4.3.5	From the General Info pa	ne, in the Tru	nk Parameter	s Pro	visioning w	vindow
	that is displayed:					
	Administer settings	s to enable cor	nnectivity with	the P	ublic Switch	1
	Telephone Networ	k.				
	Note: Obtain confi	guration detail	ils regarding th	ie sett	ing required	d for
	this connection to t	the Public Swi	tch Telephone	Netwo	ork from the	service
	provider. The entri	es for this tru	ik correspond i	to a T	1 ISDN-PR	I
	connection between	n the Avaya G	860 Media Gat	teway	and the Put	blic
	Switch Telephone I	Network.		Ţ		
	• Click on Apply.					
	Click on ISDN/DP	NSS under Pa	arameters List	t.		
	🔀 Trunk Parameters Provisioning					
	File View Tools Help					
	>> Globe > SIL > HDTG-5000 > board#9 > trunk#3		% ▼ Admin State:	Locked	Ops State:	Disabled
	Parameters List 8 Genera	al Info				
	General Info 8 2	Trunk Number	3			
	CAS	is Available	Ves	-		
		DS1 Path	none			
		Trunk Name	Trunk#3			
		Protocol Type	None	*	0	
		Framing Method Type	T1FramingEsfCrc6	*	0-41	
		Trace Level Type	NoTrace	-	0-+	
		Line Build-out Loss	db0	*		
		I Line Code	b8ZS	*		
		Clock Reference Priority	0		0- - f	
					Apply Retres	h Close
	Profile Management					
	Name	Profiles	Save Profile Apply to all	Paramet	ers	
	Trunk Profi	ile hone 🗾	Save Apply to all	Show		

Step	Description						
4.3.6	From the ISDN	/DI	PNSS pane that	t is displayed:			
	• Select the appropriate value for the Termination Side , usually userTerminationSide if the Public Switch Telephone Network						
	Clicker			provider.			
	Click of	1 AJ	ppiy.	, 1	. C	1 D01 1	
	From th	e P	rofile Manage	ement pane, sele	ct Save to sa	ve the DS1, and	
	assign a	nai	ne to the profi	le.	_		
	1	Not	e: The Profile	Management pai	ne can be use	ed to define a	
		conj	figuration prof	file that can be a	pplied to mai	ny DS1 interfaces,	
		savi	ng configurati	on steps.			
				-			
	Trunk Parameters Provisionin	8					
	File View Tools Help >> Globe > SIL > HDTG-5000 > board#9 >	trunk#2		9	Admin State: Locked	Ops State: Disabled	
	Parameters List 🛛 😵 🛱	ISDN/DP	NSS				
	General Info		III MIII Interfere ID	T.	-		
	CAS ? 🗹		D-Channel Configuration	Primary			
			Termination Side	userTerminationSide 🝸	0		
			ISDN CC Behavior	0	0-+		
			ISDN Outgoing Calls Behavior		0-+		
				CC_USE_MU_LAW			
				UA DLCI Reversed			
		T	ISDN Q931 Layer Response Behavior	T NS_NO_STATUS_ON_UNKNOWN_IE	0-+		
				T NS_NO_STATUS_ON_INV_OP			
				NS_SEND_USER_CONNECT_ACK			
		191	1000 Incoming Colle Debauter	Secondaria Interface_ID			
			ISDN Incoming Cails Denavior	CC_USER SETUP ACK			
				CC_VOICE_CONN_RS			
			DPNSS Behavior	C DPNSS STOP SABMR AFTER NL & NT1	0		
				C DPNSS FULL STARTUP SUCCESS			
				DPNSS DLC OOS AFTER NL AND NT1		ang d	
				T DPNSS SIMULTANEOUS STARTUP			
						Apply Refresh Close	
	Profile Management						
			Name Profiles	Save Profile Apply to all Pa	arameters		
			Trunk Profile - Choose P	Profile T Save Apply to all	Show		

4.3.7 Apply the DS1 configuration saved to the fourth DS1 on this DS DS1 Carriers List is shown below. * AudioCodes' LMS - fred is logged with Administration authorization. File View Tools Fault's Security Help * Globe * Globe * State	3. The resultan
AndioCodes' PAS — Fred is logged with Administration authorization. File View Tools Faults Security Heip Mic Tree Colobe Colo	▲ ● ▲ ★ ★ ★ ★ ★ 1 ? Master Profile
MiG Tree >> Globe > STIL > G660 > board#9 Image: Still and Still a	A Da + + + t?
BS1 Carriers List BS1 Carriers List Mame Protocol Admin State Op State DS1 Path I © Trunk 115Ess10isch Urlocked Enabled none Si Trunk 115Ess10isch Urlocked Enabled none S © Trunk 115Ess10isch Urlocked Disabled none T © Trunk 115Ess10isch Urlocked Disabled none S © Trunk 115Ess10isch Urlocked Disabled none S © Trunk 115Ess10isch Urlocked Disabled none T © Trunk 115Ess10isch Urlocked Disabled none S © Trunk 115Ess10isch Urlocked	Master Profile
Image: Sint Sint Image: Sint Sint Image: Sint Sint Sint Sint Sint Sint Sint Sint	Master Profile
I I I I I I I I I I I I I I I I I	
2 Image: Trunk 115Ess10isch Unlocked Enabled none 3 Image: Trunk 115Ess10isch Unlocked Enabled none 4 Image: Trunk 115Ess10isch Unlocked Disabled none 5 Image: Trunk 115Ess10isch Unlocked Disabled none 6 Image: Trunk 115Ess10isch Unlocked Disabled none 7 Image: Trunk 115Ess10isch Unlocked Disabled none 9 Image: Trunk 115Ess10isch Unlocked Disabled none 9 Image: Trunk 115Ess10isch Unlocked Disabled none 10 Image: Trunk	
3 3 Trunk 115Ess10tsch Unlocked Enabled none 4 6 Trunk 115Ess10tsch Unlocked Disabled none 5 6 6 Trunk 115Ess10tsch Unlocked Disabled none 6 6 7 7 Trunk 115Ess10tsch Unlocked Disabled none 7 9 Trunk 115Ess10tsch Unlocked Disabled none 9 9 Trunk 115Ess10tsch Unlocked Disabled none 10 9 Trunk 115Ess10tsch Unlocked Disabled none 11 9 Trunk 115Ess10tsch Unlocked Disabled none 12 9 Trunk 115Ess10tsch Unlocked Disabled none 13 9 Trunk 115Ess10tsch Unlocked Disabled none 14 9 Trunk	
9 1 Turk 115Ess101sth Uhlocked Disabled none 9 1 Turk 115Ess101sth Uhlocked Disabled none 6 1 Turk 115Ess101sth Uhlocked Disabled none 7 1 Turk 115Ess101sth Uhlocked Disabled none 8 1 Turk 115Ess101sth Uhlocked Disabled none 9 1 Turk 115Ess101sth Uhlocked Disabled none 10 1 Turk 115Ess101sth Uhlocked Disabled none 11 1 Turk 115Ess101sth Uhlocked Disabled none 13 1 Turk 115Ess101sth Uhlocked Disabled none 13 1 Turk 115Ess101sth Uhlocked Disabled none 14 1 Turk 115Ess101sth Uhlocked Disabled none 15 1 Turk 115Ess101sth Uhlocked Disabled none 17 1 Turk 115Ess101sth Uhlocked Disabled non	
 Turk 1755s10isch Uhlocked Disabled none 	
 Possible Possible<	
P Induk	
P P	
History PM History PM Turk. T15Ess10isdn Unlocked Disabled none If Ses10isdn Unlocked If Ses10isdn Unlocked If Ses10isdn Unlo	
In the intervention of the interventinter of the intervention of the intervention of the intervention	
12 TrunkT15Ess10isdn Uhlocked Disabled none 13 TrunkT15Ess10isdn Uhlocked Disabled none 14 TrunkT15Ess10isdn Uhlocked Disabled none 15 TrunkT15Ess10isdn Uhlocked Disabled none 17 TrunkT15Ess10isdn Uhlocked Disabled none 18 TrunkT15Ess10isdn Uhlocked Disabled none 19 TrunkT15Ess10isdn Uhlocked Disabled none	
History PM Hestory PM Turk. 1755s10isch Unlocked Disabled none 13 Turk1755s10isch Unlocked Disabled none 15 Turk1755s10isch Unlocked Disabled none 17 Turk1755s10isch Unlocked Disabled none 17 Turk1755s10isch Unlocked Disabled none 18 Turk1755s10isch Unlocked Disabled none 19 Turk1755s10isch Unlocked Disabled none 19	
14 If Tunk 115Ess10listin Unlocked Disabled none 15 Trunk 115Ess10listin Unlocked Disabled none 16 Trunk 115Ess10listin Unlocked Disabled none 17 Trunk 115Ess10listin Unlocked Disabled none 18 Trunk 115Ess10listin Unlocked Disabled none 19 Trunk 115Ess10listin Unlocked Disabled none	
IS ITrunkT1SEss10isdn Unlocked Disabled none Isablet none	
History PM H	
History PM III I7 E Trunk T15Ess10isch Unlocked Disabled none 18 E Trunk T15Ess10isch Unlocked Disabled none 19 E Trunk T15Ess10isch Unlocked Disabled none	
18 👿 Trunk 115Ess10isdn Unlocked Disabled none 19 🐷 Trunk	
19 Trunk T15Ess10isch Unjested Disabled none	
Tranka Tracastrata Childred Disabled Hole	
20 🐷 Trunk T15Ess10lsdn Unlocked Disabled none	
21 👿 Trunk T15Ess10lsdn Unlocked Disabled none	
22 👿 Trunk T15Ess10lsdn Unlocked Disabled none	
Alarm Browser 350 View Level: Current Level Alarms 🝸	2 🐇 🗵 🖬 🖬 🖬 🔽 🔽 🗙
Ack Severity Time * MG Name Source Alarm Name Description	
major 07:15:28 Mar G880 Board#9/Tru Trunk Alarm (XMT ALS) PSTN alarm on Trunk	XMT AIS)
major 07:15:25 Mar G860 Board#9/Tru Trunk Alarm (XMT AIS) PSTN alarm on Trunk	XMT AIS)
major 07:15:26 Mar G860 Board#9/Tru Trunk Alarm (XMT AIS) PSTN alarm on Trunk	
major 07:15:26 Mar G860 Board#9/Tru Trunk Alarm (XMT AIS) PSTN elerm on Trunk	XMT AIS)
	XMT AIS) XMT AIS)
	XMT AIS) XMT AIS)

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 Licon to navigate back to the screen displayed below. Click on the SIP tab.
	AudioCodes' EMS - acladmin is logged with Administration authorization. Fie View Tools Faults Security Hele MG Tree Codobe>SIL > HDTG-5000 > board#9 2
	BS3 DS1 Trunks DSP Sgneling H ## SL DS3 Status Sgneling

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Step	Description	
4.4.2	Click on the SIP Protocol tab; then click on the	e Protocol Settings tab.
	AudioCodes' EMS - acladmin is logged with Administration authorization. File View Tools Fauts Scourty Heb MG Tree MG Tree Sche Globe > SL > HDTG-5000 > board#9 Coders List Coders List Sip Protocol Settings Coders Elist Sip Coder List	Routing Manputation ge Proxy Servers S PTo ESIN Ca
4.4.3	From the General Settings pane, in the SIP Pr	otocol Definitions
	 Window, administer settings to enable SIP conn Meeting Exchange Enterprise S6200 Conference Set the SIP Destination Port, Enable E Transport Type to enable SIP-TLS/UE the Avaya Meeting Exchange Enterprise Server as shown below: Remaining fields are default settings Click on Apply and then Close 	ectivity with the Avaya eing Server as follows: Early Media and DP/TCP connectivity with e S6200 Conferencing
	SIP Destination Port	5061
	SIP Local Port	5060
	TCP Local Port	5060
	TLS Local Port	5061
	🗷 Enable Early Media	Yes
	Sip Session Expires	0
	I "User=Phone" in URL	Yes
	"User=Phone" In From	Disable
	💌 Prack Mode	Supported 💌
	Enable RPI Header	No
	💌 X Channel Header	Disable
	Asserted ID Mode	NoHeaderAdded
	Add Type and Number Plan to Remote Party ID Header	Enable
	Enable CIC	No
	☑ Transport Type	tis

Step	Description
4.4.4	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 D icon to add codec(s), ordered sequentially from most to least preferred.
4.4.5	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. SIP Coders File View Tools Help SIP Coders General Settings Coders General Settings Coder Setting Settings Coder Name Coder Name Coder 1 Coder 1 Coder Setting Setting Settings Coder Name Coder Name
	Image: Coder Name G.711 U-law 64k Image: Packetization Time 20 ms Image: Coder Rate 84.0 Image: Payload Type 0 Image: Silence Suppression Disable

4.5. Configure B-channels

The following steps describe the administrative procedures for assigning profiles to Bchannels. 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

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Step	Description
4.5.1	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 M
	icon to add trunk group(s).
	🖫 AudioCodes' EMS bob is logged with Administration authorization.
	File View Tools Faults Security Help
	MG Tree X >> Globe > SIL > H0TG-5000 > board#9 8
	Globe
	H ## SL I Trunk Group Settings Sip Trunk Group List
452	Administer settings for ISDN DDI trunking between the Avera COCO Media
4.5.2	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 11) and Ending Channel to 24 (last B-channel in each T1)
	 Set the Starting Phone Number field to 1000000.
	SIP Trunk Groups
	File View Tools Help >> Globe > SITL > G860 > board#9 > Trunk Group#2
	Parameters List 8 🖬 General Settings
	General Settings 8 🖘
	Index 2
	I Ivame All Irunks ····
	First Trunk ID 1 HA×
	🖾 Last Trunk ID 28 HA×
	Starting Channel 1 HAX
	Ending Channel 24
	Starting Phone Number 1000000 HA×
	The resultant Sip Trunk Group List is shown below
<u> </u>	The resolution of the oroug and is shown below.

Step	Description						
	>> Globe > SITL > G860 > board#9						
		SIP Protocol		Trunk Groups	Routing		Manipulation
	SIP Trunk Groups	🔲 Trunk Group		🔲 Trunk G	Froup Settings	<u> </u>	Tel Profile
		Sip Trunk Group List					🖬 🔕 🖓 '
		# Name Trunk (Group ID	First Trunk ID	Last Trunk ID S	tarting Channel	Ending Channel
		All Iru 2		1	26 1		24
	-						
452	A 1 · · ·	41 4 1	. 1.	• 1	.1 1.	1 * 1	
4.5.3	Administer settin	igs that are used	to dei	ermine the	e method in v	which nev	w calls are
	assigned to B-ch	annels within each	ch tru	nk group a	s follows:		
	Click on	the Trunk Grou	ps tal	b .			
	Click on	the Trunk Gro u	n Set	tings tab			
	• Erom the	Sin Trunk Cro	un Se	ttings List	t nona that is	diaplaya	d aliak on
	• From the	SIP TTUIK GIO	up Se	tungs Lis	i pane mai is	suispiaye	a, click oli
	the 🖬 ic	on to add trunk g	roup	setting(s).			
	AudioCodes' EMS - bob is logged	with Administration authorization.					
	File View Tools Faults Security Help						
	MC Tree X School SH SHITTE 5000 Shored#0						
		SIP Proto	col	Trunk Groups	Routing	Manipulatio	on la
	Globe	Trunk Groups S	Trunk G	roup	Trunk (Group Settings	
		Sip Trunk Group Se	ttings List			B	0 5-

Step	Description	
4.5.4	Administer settings to determine the n B-channels within the ISDN-PRI trun parameters:	nethod in which new calls are assigned to k group provisioned with the following
	• Enter PRI in the Name field.	
	• Set the Trunk Group ID to c to the trunk provisioned	orrespond to the Trunk Group ID assigned
	Set the Channel Select Mode	to Ascending.
	Note: This channel selection pattern,	in combination with the logical trunk
	provisioning enable ascending chann	el selection over 46 B-channels spread over
	two physical DS1 connections betwee	n 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 originate	on from the Avaya G860 Media Gateway.
	The resultant Sip Trunk Group Sett	ings List is shown below.
	AudioCodes' EMS bob is logged with Administration authorization.	
	File View Tools Faults Security Help	- 1000
	MG Tree >> Globe > SiL > HDTG-5000 > board#9	8 E
	Globe	Trunk Groups
	Sin Trunk Groups S	Trunk Group Settings
	# Name Tro	Ink Group Channel Mo Admin State
	1 CAS 1	Ascending Locked
		, recently and the second s

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 II 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.				
	AudioCodes' EMS - actadmin is lagged with Administration authorization. File Mew Tools Faults File Mew Tools Faults MG Tree > Globe > StL > HDTG-500t > board#8 > Tel To IP Routing#5 P Image: Color StP Protocol Trunk Groups Routing Image: Color StP Protocol Trunk Groups Aternate PTo Image: Color Stp Tel To IP Routing List Image: Color Image: Color				

Description				
 6.2 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 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) 				
File View Tools Help >>> Globe > SITL > G860 > board#9 > Tel To IP Routing#3 Parameters List Image: Concret Settings Concret Settings Image: Concret Settings Image: Concret Settings Image: Concret Settings <	Admin State: Unlocked J J MX6200 Orr 7xxxx HAX 192.168.36.10 HAX Not Chosen HAX			
	From the SIP Routing Tel to Ip window Dial-In to the Avaya Meeting Exchange IP Public Switched Telephone Network as for Enter a descriptive label in the National State of the Avaya Meeting Exchanging to the Avaya Meeting Exchanging to the Avaya Meeting Exchanging to the Avaya Meeting Exchanging digit of 7. The rule 7xxxx match any single digit. Enter an * in the Source Phone Free telephone number dialing in to the Conferencing Server from the Public Server in the Dest A and Conferencing Server in the Dest A conferencies Server in the Dest A conference Server is the provide ID index is source Phone Prefix. Source Phone Prefix is source Phone Prefix is provide ID index is source Phone Prefix. Source Phone Prefix is prefile D			

Step	Description		
4.6.3	 5.3 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 Switched 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 Server from the Public Switched Telephone Prefix field to allow routing for any source telephone number dialing in to the Avaya SES Server in the Dest Address field. Click on Apply and then Close. 		
	SIP Routing Tel to Ip		
	File View Tools Help >> Globe > SITL > G860 > board#9 > Tel To IP Routing#4		
	Parameters List General Settings Index 4 Index 5ES_MX Index Max Index Baxxx Index Hax Index 192.168.36.11 Hax Profile ID Not Chosen Hax		

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 II 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. 			
	Fig. AudioCodes' EMS - bob is logged with Administration authorization.			
	MG Tree SIL > H0TG-5000 > board#9 8 E			
	All Cobe Ip To Tel Routing Routing Setting C Tel To P IP To Tel C DNS C Attende Tel T Attende P To			
	Sip IP To Tel Routing List			

Step	Description					
4.6.5	From the SIP Rout	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 des	criptive label in	n the Name fiel	d.		
	• Enter a rule	e in the Dest Ph	one Prefix fiel	ld that matches the	e pattern of outgoing	
	calls from t	he Avaya Meet	ting Exchange	Enterprise S6200	Conferencing Server	
	to the Publi	ic Switched Tel	ephone Netwo	rk. For these Appl	ication Notes, all calls	
	from the A	vaya Meeting E	Exchange Enter	prise S6200 Confe	erencing Server to the	
	Public Swi	tched Telephon	e Network via	PRI trunking are p	laced where * is a	
	wildcard ar	nd will match a	ny digit(s).			
	• Enter an *	in the Source I	Phone Prefix a	nd Source Addre	ss fields to allow	
	routing for	any party diali	ng out from the	Avaya Meeting E	xchange Enterprise	
	S6200 Con	ferencing Serve	er to the Public	Switched Telepho	ne Network.	
	• Enter the T	runk Group ID	for the PRI tru	nk group provision	ned in the Trunk	
	Group ID	Group ID field.				
	• Click on A	pply and then (Close.			
	SID Routing In to Tol					
	File View Tools Help			<u> </u>		
	>> Globe > SITL > G860 > board#9 > Sip IP To Tel Routing#2					
	Parameters List 8 General Settings					
	E General Settings 8 ₽					
	Index 2 ☑ Name ☑ Dest Phone Prefix					
		🗵 Source Phone Prefix	×			
	Source Address *					
	Trunk Group ID 2					
		Profile ID	Not Chosen 💌			
	Tpm Association Both					



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

Step	Description			
4.7.1	Board9->Protocol Settings->DTMF Settings			
	Configuring In Band 1 • Declare RFC • DTMF RFC2	DTMF 2833 in SDP= "Don 833 Negotiation=N	tDeclareRFC2833 oNegotiation" as s	inSDP" hown below
	🔀 SIP Protocol Definitions			
	File View Tools Help			
	>> Globe > SITL > G860 > board#9		% - Admin State: Unlocked	🝸 Ops State: Enabled 🍸
	Parameters List 🛛 🖇 🗖	DTMF Settings		
	General Settings ? Call Security ? P Media ? DTMF Settings ? PSTN Tunneling ? Fax Signaling ? Radius ? Configuring RFC 283 Declare RFC2	 Enable DTMF Outband DTMF format Disable Auto Mute Declare RFC2833 in SDP DTMF RFC2833 Negotiation RFC2833 Payload Type 3 Out Of DTMF 2833 in SDP= "Dec 833 Negotiation="1	No	P" Negotiation" as
	shown below	0		C
	🔀 SIP Protocol Definitions			
	File View Tools Help			
	>> Globe > SITL > G860 > board#9		🎇 🔻 Admin State: Unlocked	Dps State: Enabled 🗾
	Parameters List 🛛 🛛 🖾	DTMF Settings		
	General Settings ? Call Security ? Call Security ? P Media ? DTME Settings ? PSTN Tunneling ? PSTN Interworking ? Fax Signaling ? Radius ?	 Enable DTMF Outband DTMF format Disable Auto Mute Declare RFC2833 in SDP DTMF RFC2833 Negotiation RFC2833 Payload Type 	No InfoCisco No CeclareRFC2833InSDP LenableRfc2833PTNegotiation 101	

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

Step	Description				
4.8.1	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 \rightarrow 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				
	Software Manager				
	File View Actions Help				
	Software Type * File Name File Type SW Version Protocol Product Types File Size				
	Add Files				
	Software Files Auxiliary Files				
	File Type File Name File Description				
	Tones				
	Call Progress Tones (All Products)				
	Pre Recorded Tones (All Products)				
	Voice Prompts (All Products)				
	Media Server				
	APS Segments XML (IPM2K/IPM3K)				
	VXML (IPM2K/IPM3K/IPM5K/IPM5K)				
	Security				
	X509 Private Key File (All Products) esktop/certs/certs/private.key 📄				
	X509 Server Certificate File (All Products) =sktop\certs\server.cert				
	X509 Trusted Root Certificate File (All Products)				
	As shown below, the files are loaded on the Server.				
	• Click on the MG Status→Properties button and select MG				

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 conso Use Global IP address when connecting via telnet or SSH. Login as the CLI user with administrative privileges. At the prom 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) 	le. pt
 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
 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
 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 Media Gateway Parameters Provisioning 	S
 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 Media Gateway Parameters Provisioning Media Gateway Parameters Provisioning 	
 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 Media Gateway Parameters Provisioning File View Tools Help >> Globe> SITL > G660 W+ Admin State: Unlocked Ops State: Enabled 	
 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 Media Gateway Parameters Provisioning Media Gateway Parameters Provisioning Media Gateway Parameters List Media Gateway Pa	
 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 Media Gateway Parameters Provisioning Media Gateway Parameters Provisioning Certificate Expiration Date (YYYY-MM-DD H+tMM:SS) Media Gateway Parameters Parameters Parameters Parame	
 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 Media Gateway Parameters Provisioning Media Gateway Parameters Provisioning Certificate Expiration Date (YYYY-MM-DD H+MM:SS) Media Gateway Parameters Parameters Provisioning	
 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 Media Gateway Parameters Provisioning Media Gateway Parameters Provisioning Certificate Expiration Date Reminder Days Media Gateway Parameters Parameters Provisioning	

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:
 - o 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
 - o *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
 - o *7 (as moderator) toggle conference security on/off
 - *8 play the roster of participant name during conference
 - \circ *93X (where X is defined from 1 to 9) to invoke a subconference
 - o *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

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• 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 Avava Meeting Exchange Enterprise S6200				
	Conferencing Server console to access the CLI with the				
	appropriate gradentiale				
	 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 ? UU:UU:UU modapid				
	12937 ? 00:00:00 schapid				
	12930 : 00.00.02 callhand				
	12939: 00.00.00 mittped				
	12944 2 00:00:00 mgdispatcher				
	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				

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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. 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 Boom window Verify conference participants are added/removed from conferences by observing the Conference Navigator and/or Conference Room windows.
	Avaya Bridge Talk - 192.168.36.10 Operator 1 - 09/02/09 19:01:28 Wew Line Conference Fast Dial Tools Window Help Image: Conference Fast Dial Tools Window Help Image: Conference Fast Dial Tools Window Help Image: Conference Display Enter Fastdial help reQuests Line Music Options Purge Set Transfer retrieve Update ? Help Image: Conference Display Enter Fastdial help reQuests Line Music Options Purge Set Transfer retrieve Update ? Help Image: Conference Display Enter Fastdial help reQuests Line Music Options Purge Set Transfer retrieve Update ? Help Image: Conference Display Enter Fastdial help reQuests Line Music Options Purge Set Transfer retrieve Update ? Help Image: Conference Display Enter Fastdial help reQuests Line Music Options Purge Set Transfer retrieve Update ? Help Image: Conference Display Enter Fastdial help reQuests Line Music Options Purge Estit Tone Gain Hang up Lecture Look Image: Conference Polling Q&A Print Detail Image: Conference Co

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

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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