

Avaya Solution & Interoperability Test Lab

Configuring Secure SIP Connectivity Utilizing Transport Layer Security (TLS) Between Avaya Communication Manager and the Avaya Meeting Exchange S6100 Conferencing Server Via Avaya SIP Enablement Services -Issue 1.0

Abstract

These Application Notes present the procedures for configuring secure SIP connectivity between Avaya Communication Manager and the Avaya Meeting Exchange S6100 Conferencing Server via Avaya SIP Enablement Services. Secure SIP connectivity is enabled by utilizing Transport Layer Security (TLS) authentication and encryption standards, thus providing customers a secure, standards based solution. This configuration leverages the flexibility offered by Avaya Communication Manager and the scalability provided by Avaya SIP Enablement Services to support a rich set of conferencing options available from Avaya Meeting Exchange.

1. Introduction

These Application Notes present the procedures for configuring secure SIP connectivity between Avaya Communication Manager and the Avaya Meeting Exchange S6100 Conferencing Server via Avaya SIP Enablement Services. Secure SIP connectivity is enabled by utilizing Transport Layer Security (TLS) authentication and encryption standards, thus providing customers a secure, standards based solution. This configuration leverages the flexibility offered by Avaya Communication Manager and the scalability provided by Avaya SIP Enablement Services to support a rich set of conferencing options available from Avaya Meeting Exchange.

The following call flows for accessing a conference on Avaya Meeting Exchange have been verified:

- DirectCallFlow, where conference participants Dial-In and enter a conference as moderator, without entering a passcode.
- BasicCallFlow, where conference participants Dial-In and enter a conference via a passcode.

The following features have been verified for adding participants to an active conference:

- Blast Dial, where a moderator on a conference call can enter a touchtone command (*9 in these Application Notes) to invoke a Blast Dial to a pre-provisioned list of one or more participants. The participants have the option of joining the conference call.
- Originator Dial-Out, where a moderator on a conference call can Dial-Out and add a participant to the conference call.

These Application Notes provide the administrative steps for configuring:

- Connectivity between Avaya Communication Manager and Avaya SIP Enablement Services via secure SIP trunking utilizing TLS (see Figure 1).
- Connectivity between Avaya SIP Enablement Services and Avaya Meeting Exchange via secure SIP trunking utilizing TLS (see **Figure 1**).



Figure 1: Network Configuration

1.1. Dial-Out from Avaya Meeting Exchange

The following figure shows how secure SIP trunking between Avaya SIP Enablement Services and Avaya Communication Manager is utilized to enable Dial-Out from Avaya Meeting Exchange to Avaya Communication Manager **Via** Avaya SIP Enablement Services. Since this configuration is configured for TLS, the SIP messages below (captured from a log file on Avaya SIP Enablement Services) are intended to illustrate the call flow.

- A SIP INVITE Message is sent From Avaya Meeting Exchange To Avaya SIP Enablement Services utilizing TLS (see red dashed line in Figure 2).
- The SIP **INVITE** Message is then sent to Avaya Communication Manager **Via** Avaya SIP Enablement Services utilizing TLS (see blue dotted line in **Figure 2**).



Figure 2: Dial-Out from Avaya Meeting Exchange

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1.2. Dial-In to Avaya Meeting Exchange

The following figure shows how secure SIP trunking between Avaya SIP Enablement Services and Avaya Meeting Exchange is utilized to enable Dial-In to Avaya Meeting Exchange from Avaya Communication Manager **Via** Avaya SIP Enablement Services. Since this configuration is configured for TLS, the SIP messages below (captured from a log file on Avaya SIP Enablement Services) are intended to illustrate the call flow.

- A SIP INVITE Message is sent From Avaya Communication Manager To Avaya SIP Enablement Services utilizing TLS (see red dashed line in Figure 3).
- The SIP **INVITE** Message is then sent to Avaya Meeting Exchange **Via** Avaya SIP Enablement Services utilizing TLS (see blue dotted line in **Figure 3**).



Figure 3: Dial-In to Avaya Meeting Exchange

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2. Equipment and Software Validated

The following equipment and software versions were used for the configuration:

Equipment	Software		
Avaya S8710 Media Servers	Avaya Communication Manager 3.1		
	(R013x.01.0.628.6)		
Avaya G650 Media Gateway			
• Avaya TN2312BP (IPSI)	HW12 FW031		
• Avaya TN799DP (C-LAN)	HW01 FW017		
• Avaya TN2302AP (MEDPRO)	HW20 FW112		
Avaya Meeting Exchange S6100 Conferencing	2.0.22.2		
Server			
Avaya SIP Enablement Services	SES-3.1.1.0-114.0		
Avaya C364T-PWR Converged Stackable	4.5.14		
Switch			
Avaya 4620 IP Telephones	2.3 (H.323)		
Avaya 4602 IP Telephones	2.2 (SIP)		
Avaya 6408D+ Digital Telephones			
Analog Telephones			

Table 1: Hardware and Software Versions

3. Avaya Communication Manager Configuration

This section describes the steps for configuring Avaya Communication Manager to interoperate with Avaya SIP Enablement Services via secure SIP trunking utilizing TLS.

The following configuration of Avaya Communication Manager is provisioned using the System Access Terminal (SAT). After completion of the configuration in this section, perform a **save translation** command to make the changes permanent.

Step	Description							
3.1	Verify feature licensing.							
	Issue the command "display system-parameters customer-options" and proceed to Page 2.							
	Verify that the number of SIP trunks supported by the system i	s suff	icient f	for the n	umber of			
	SIP trunks needed							
	Note: Each SIP call between two SIP endpoints (whether inter	nal oi	· ortorv	al) real	ires two			
	SIP trunks for the duration of the call. For these Application N	latas	Avava	Mootine	nies ino Frehange			
	is treated as an external SIP endpoint Thus a call from a SIP	toloni	hong to	Avava	Mooting			
	is irealed as an external SIF enapoint. Thus, a call form a SIF	leiepi	ione io	Avaya	Meeting			
	Exchange will use two SIP trunks. A call between a non-SIP te		ne ana	Avaya	Meeting			
	Exchange will use only one SIP trunk. The license file installed	t on th	ie syste	em contr	rols the			
	maximum permitted. If a required feature is not enabled or the	re is i	nsuffic	ient cap	pacity,			
	contact an authorized Avaya sales representative to make the a	ippro	priate d	changes	•			
	display system-parameters customer-options		Page	2 of	10			
	OPTIONAL FEATURES							
	IP PORT CAPACITIES		USED					
	Maximum Administered H.323 Trunks:	1000	0					
	Maximum Concurrently Registered IP Stations:	100	0					
	Maximum Administered Remote Office Trunks:	0	0					
	Maximum Concurrently Registered Remote Office Stations:	0	0					
	Maximum Concurrently Registered IP eCons:	0	0					
	Max Concur Registered Unauthenticated H.323 Stations:	0	0					
	Maximum Video Capable H.323 Stations: 0 0							
	Maximum Video Capable IP Softphones: 0 0							
	Maximum Administered SIP Trunks:	1000	0					
	Maximum Number of DS1 Boards with Echo Cancellation:	0	0					
	Maximum TN2501 VAL Boards:	1	0					
	Maximum G250/G350/G700 VAL Sources:	0	0					
	Maximum TN2602 Boards with 80 VoIP Channels:	0	0					
	Maximum TN2602 Boards with 320 VoIP Channels:	0	0					
	Maximum Number of Expanded Meet-me Conference Ports:	0	0					

Step	Description							
3.2	 3.2 Proceed to Page 3 on the system-parameters customer-options form and verify that the system is licensed to utilize Automatic Alternate Routing (AAR) without Feature Access Co (FAC). <i>Note:</i> AAR without FAC allows direct access to the AAR digit analysis table (see Step 3.9) upon matching a Dialed String in the dial plan analysis table (see Step 3.8). 							
	display system-parameters customer-options Page 3 of 10							
	OPTIONAL FEATURES							
	Abbreviated Dialing Enhanced List?nAudible Message Waiting?nAccess Security Gateway (ASG)?nAuthorization Codes?nAnalog Trunk Incoming Call ID?nBackup Cluster Automatic Takeover?nA/D Grp/Sys List Dialing Start at 01?nCAS Branch?nAnswer Supervision by Call Classifier?nCAS Main?nARS?YChange COR by FAC?nARS/AAR Partitioning?YComputer Telephony Adjunct Links?nARS/AAR Dialing without FAC?YCvg Of Calls Redirected Off-net?nASAI Link Core Capabilities?nDCS (Basic)?nAsync. Transfer Mode (ATM) PNC?nDCS with Rerouting?n							
	ATM WAN Spare Processor? n Digital Loss Plan Modification? n ATMS? n DS1 MSP? n Attendant Vectoring? n DS1 Echo Cancellation? n (NOTE: You must logoff & login to effect the permission changes.)							
	(NOTE: You must logoff & login to effect the permission changes.)							

Step	Description						
3.3	Configure an IP codec set.						
	Issue the command " change ip-codec-set <n></n> ", where n is the number of an available codec set. Configure an Audio Codec that is supported on Avaya Meeting Exchange. For these Application Notes, G.711MU is selected.						
	change ip-codec-set 1	Page	1 of	2			
	IP Codec Set						
	Codec Set: 1						
	Audio CodecSilence SuppressionFrames Per PktPacket Size(ms)1:G.711MUn2202:3:4:5:6:7:						

Step	Description
3.4	Configure an IP network region.
	 Issue the command "change ip-network-region <n>", where n is the number of an available IP network region and administer settings as per below.</n> Enter the number of the IP codec set provisioned in Step 3.3 in the Codec Set field. Configure the Authoritative Domain to match the configuration for the System Properties on Avaya SIP Enablement Services (see Step 5.3).
	change ip-network-region 1 Page 1 of 19
	IP NETWORK REGION
	Region: 1 Location: Authoritative Domain: avaya.com
	Mame: MEDIA PARAMETERS Intra-region IP-IP Direct Audio: yes Codec Set: 1 Inter-region IP-IP Direct Audio: yes UDP Port Min: 2048 IP Audio Hairpinning? n UDP Port Max: 3327 DIFFSERV/TOS PARAMETERS RTCP Reporting Enabled? y Call Control PHB Value: 46 RTCP MONITOR SERVER PARAMETERS Audio PHB Value: 46 Use Default Server Parameters? y Video PHB Value: 26 802.1P/Q PARAMETERS Call Control 802.1p Priority: 6 Audio 802.1p Priority: 6 Video 802.1p Priority: 5 AUDIO RESOURCE RESERVATION PARAMETERS H.323 IP ENDPOINTS RSVP Enabled? n H.323 Link Bounce Recovery? y Idle Traffic Interval (sec): 5 Keep-Alive Count: 5
3.5	 Configure IP node names. Issue the command "change node-names ip" and administer settings as per below. Add a node Name and IP Address for Avaya SIP Enablement Services (SES). Verify that node names and IP addresses are configured for the C-LAN and MEDPRO boards.
	change node-names ip Page 1 of 1
	IP NODE NAMES Name IP Address CLAN-1A02 192.168.11 .10 MEDPRO-1A03 192.168.11 .11 SES 192.168.11 .20

Step	Description						
3.6	Configure a SIP signaling group.						
	 Issue the command "add signaling-group <n>", where n is the number of an unallocated signaling group and administer settings as per below.</n> To enable secure SIP connectivity utilizing TLS, configure the following: Set the Group Type to sip. Set the Transport Method to tls. Set the Far-end Listen Port to 5061. Leave the Near-end Listen Port at the default value (5061). Enter the IP node name of the C-LAN displayed in Step 3.5 in the Near-end Node Name field. Enter the IP node name of Avaya SIP Enablement Services provisioned in Step 3.5 in the Far-end Node Name field. Enter the number of the IP network region provisioned in Step 3.4 in the Far-end Network Region field. Set the Direct IP-IP Audio Connections field to y to enable direct IP-to-IP audio connectivity, the following must be administered: [Not Shown] Direct IP-to-IP audio connectivity must be enabled at the system-level on Page 16 of the system-parameters features form by setting the parameter Direct IP-IP Audio Connectivity must be enabled on the station form by setting the Direct IP-IP Audio Connections field to y. 						
	add signaling-group 1 Page 1 of 1						
	SIGNALING GROUP						
	Group Number: 1 Group Type: sip Transport Method: tls						
	Near-end Node Name: CLAN-1A02Far-end Node Name: SESNear-end Listen Port: 5061Far-end Listen Port: 5061Far-end Domain:						
	Bypass If IP Threshold Exceeded? n						
	DTMF over IP: rtp-payload Direct IP-IP Audio Connections? y IP Audio Hairpinning? n						
	Session Establishment Timer(min): 120						

Step	Description	
3.7	Configure a SIP trunk group.	
	 Issue the command "add trunk-group <n>", where n is group and administer settings as per below.</n> Set the Group Type to sip, which is consistent w Step 3.6. Set the Trunk Access Code (TAC) to a number the plan (see Step 3.8). Set the Service Type to tie. Enter the number of the signaling group provision Group field. Specify the Number of Members supported by the Step 3.1, each SIP call between two SIP endpoint requires two SIP trunks for the duration of the call Meeting Exchange is treated as an external SIP ertelephone to Avaya Meeting Exchange will use two SIP telephone and Avaya Meeting Exchange will 	the number of an unallocated trunk ith the signaling group provisioned in nat is consistent with the existing dial ned in Step 3.6 in the Signaling his SIP trunk group. As mentioned in the source of the second second second second second to the second second second second second second to SIP trunk group. As mentioned in the source of the second second second second the second second second second second second second the second second second second second second second the second second second second second second second second the second second second second second second second second second second the second sec
	add trunk-group 1	Page 1 of 21
	TRUNK GROUP	
	Group Number: 1 Group Name: SES SIP Direction: two-way Dial Access? n Queue Length: 0 Service Type: tie Group Type: sip COR: 1 Outgoing Display? n Auth Code? n	CDR Reports: y TN: 1 TAC: 101 Night Service: Signaling Group: 1
		Number of Members: 50

3.1. Call Routing

The following steps show procedures to enable call routing from Avaya Communication Manager to Avaya SIP Enablement Services. For these Application Notes, AAR is utilized (in conjunction with a route pattern) to route calls over the secure SIP trunk group provisioned in **Step 3.7**.

Step	Description							
3.8	Configure the dial plan analysis table.							
	Issue the command "change dialplan analysis" and add an entry in the table to treat any digit							
	string of 3 digits in Total Length with a leading Dialed String of 4 as a Call Type of aar .							
	0 0			C	U			
	change dialpl	an anal	ysis				Page 1 of 12	
					ANAT VOTO	ייד די די די די		
				DIAL PLAN	ANALISIS	IABLE	Percent Full: 1	
	Dialed	Total	Call	Dialed	Total C	all	Dialed Total Call	
	String	Length	Type	String	Length T	ype	String Length Type	
	0	1	attd					
	1	3	dac					
	2	5	ext					
	3	5	ext					
	4	3	aar					
	5	3	aar					
	б	3	ext					
	7	4	ext					
	7	5	ext					
	8	1	fac					
	9	1	fac					
	*	3	fac					
	#	3	fac					

Step	Description							
3.9	Configure the AAR digit and	alysis table.						
	Issue the command "change aar analysis" and administer settings as per below. Add entries in							
	the table to send the following	ng Dialed S	trings to I	Koute P	attern	I. 		C 4
	• Dialed String 401 is (45)	used by Av	aya Meetii	ig Exch	ange to	r BasicCal	IF IOW (S	ee Step
	 Dialed String 444 is 	used by Av	ava Meetii	o Exch	ange fo	r DirectC a	liFlow (see Sten
	4.7).	useu by IIV	aya Meetii			Directed		see step
	change aar analysis					Page	1 of	2
		AAR DI	GIT ANALYS	SIS TABI	ΞE			
						Percent	Full:	1
	Dialed String	Total Min Max	Route Pattern	Call Type	Node Num	ANI Reqd		
	401	3 3	1	aar		n		
	412	3 3	12	aar		n n		
	413	3 3 3 3	13 1	aar		n		
3.10	Configure a route pattern							
0.10	Comigure a route pattern.							
	Issue the command "change	route-nati	ern <n>"</n>	where i	n is the	number of	the rout	e nattern to
	he administered Add an ent	ry in the tel	n < n >	o tho tru	in 15 the	un provisio	ned in S	ton 37
	be administered. Add an ent	i y in the tat			ilik gio	up provisio		tep 5.7.
	change route-pattern 1					Pa	age 10	of 3
	Pattern Number: 1 Pattern Name: SES SIP							
	Grp FRL NPA Pfx Hop T	oll No. In	nserted	ure sie	f 11		DCS/	IXC
	No Mrk Lmt L	ist Del D	igits				QSIG	
		Dgts					Intw	
		0					n	user
	2.						n	user
	4:						n	user
	5:						n	user
	6:						n	user
			TE Conti	o/Footi	INO DAR	M No Nur	boring	
	0 1 2 3 4 W Reque	st	CIE SELVIC	.e/react	ILC FAN	Dats For	mat.	LIAIC
					S	ubaddress		
	1: уууууп п	rest					:	none
	2: уууууп п	rest					1	none
	3: yyyyyn n	rest					1	none
	4. yyyyyn n	rest					1	none
	5. yyyyyn n 6: vyyyyn n	rest						none
		TCBC					-	

4. Avaya Meeting Exchange Configuration

This section describes the steps for configuring Avaya Meeting Exchange to interoperate with Avaya SIP Enablement Services via secure SIP connectivity utilizing TLS.

Step Description **4.1** Verify licensing. Avaya Meeting Exchange uses Avaya Web License Manager (WebLM) for licensing. WebLM is a Web-based license manager that runs on both Microsoft Windows and UNIX systems. The WebLM server provides a Web User Interface (UI) for license administration that can be accessed from a standard web browser over a secure SSL link. Open a web browser and enter the following URL: https://<IP Address of Avaya Meeting Exchange>/WebLM Log in to the WebLM server with the appropriate credentials and verify Avaya Meeting Exchange is licensed for Meeting Exchange Groupware Edition Ports. *Note*: Each conference participant in a conference on Avaya Meeting Exchange requires one port for the duration of the conference call. The license file installed on the system controls the maximum permitted. If there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes. AVAYA Web License Manager (WebLM v4.0) Install License You are here: Licensed products > Meeting Exchange Licensed Products Groupware Edition Meeting Exchange Change Password License installed on: Jun 30, 2006 2:17:28 PM EDT Manage Users <u>View Peak Usage</u> Logout License Acquisition Status License acquisition enabled: Yes Currently failed over: No Licensed Features Feature (Keyword) Licensed Acquired Number of Meeting Exchange Groupware Edition Ports (VALUE_MXGE_PORTS) 300 0 permanent Number of Meeting Exchange Groupware Edition G.711 CODEX Supported (VALUE_MXGE_G711_CODEX) 300 0 permanent Acquired Licenses

Step	Description						
4.2	Administer settings for Avaya Meeting Exchange as follows:						
	• Open a web browser and enter the following URL:						
	https:// <ip address="" avaya="" exchange="" meeting="" of="">/mx</ip>						
	• Log in to Avaya Meeting Exchange with the appropriate credentials.						
4.3	Configure settings that enable secure SIP connectivity between Avaya Meeting Exchange and						
	other SIP User Agents by administering SIP Agent parameters as follows:						
	• Click Configuration from the S6100 web interface toolbar.						
	• Click SIP Agent under Bridge Configuration .						
	• Add a SIP Address for Avava Meeting Exchange. To enable secure SIP connectivity						
	utilizing TLS, the entry for the SIP Address must contain sips , 5061 and						
	transport=tls. Avaya Meeting Exchange uses this field to populate the From header in						
	SIP INVITE messages from Avaya Meeting Exchange.						
	• Add a Contact address to provide User Agents a Contact to use for acknowledging SIP						
	messages from Avaya Meeting Exchange. To enable secure SIP connectivity utilizing						
	TLS, the entry for the Contact address must contain 5061 and transport=tls .						
	• Leave all remaining required fields at the default settings.						
	• When finished click the Submit button						
	Meeting Exchange Express Edition						
	AVAYA Install Engineer						
	Help Log Out Installation Configuration Provisioning						
	System Configuration						
	System Address Data Retention Times						
	Directories sips:S610009192.168.13.212:5061;trans						
	Blast Dial Controls Playback Controls						
	Bridge Configuration						
	Media Server * Ethernet VLAN Value 10 SIP Agent III IIII						
	SIP Proxies Contact <sip:s6100@192.168.13.212:5061;trans port=tls></sip:s6100@192.168.13.212:5061;trans 						
	URI to Service Map						
	TelNum to URI Map Suctor Dependent						
	Prompt Sets						
	Conference Maps						
	Participant Key Map						
	Report Configuration						
	Reports						

Step	Description	
4.4	To associate incomin Service Map entry f Click URI to Click the Ad	ng calls to Avaya Meeting Exchange with a BasicCallFlow, add a URI to follows: • Service Map under System Maps. • d button.
	AVAYA	Meeting Exchange Express Edition Install Engineer
	Help Log Out Installation	Configuration Provisioning
	System Configuration System Address Data Retention Times Directories Blast Dial Controls Playback Controls Bridge Configuration Media Server SIP Agent SIP Proxies System Maps URI to Service Map TelNum to URI Map System Prompts Prompt Sets Conference Maps Moderator Key Map Participant Key Map Report Configuration Reports	URI to Service Map

Step	Description		
4.5	Configure a URI to Service Map Parameter for a BasicCallFlow as follows:		
	• Leave the Order f	ield at the default value.	
	Note: Avaya Meeting Exchange parses all URI to Service Map entries, searching for		
	pattern matches in descending order, terminating the search once a pattern is matched.		
	For these Application Notes, order is irrelevant as the patterns for BasicCallFlow and		
	DirectCallFlow (see Step 4.7) are mutually exclusive.		
	• Add a URI Patter	n to match the pattern of incoming Request URIs in SIP INVITE	
	messages from Av	aya SIP Enablement Services.	
	sin•401@1	92 168 13 212.5061.transnort-tls SIP/2 0. The URI Pattern below	
	is configure	ed to match sin:401@*, which will match sin:401@, then any string	
	following t	he @.	
	• Set the Call Flow	to BasicCallFlow to allow conference participants to enter a	
	conference by ente	ring a passcode.	
	• Enter a descriptive	name for the Service Name field.	
	• When finished, cli	ck the Add button.	
	🗿 Meeting Exchange Exp	oress Edition - Microsoft Internet Explorer	
	Add URI to	Service Map Parameter	
	* Order	1	
		sip:4010*	
	* URI Pattern		
	* Call Flow	BasicCallFlow	
	Greeting	greeting	
	* Service Name	"401" from Avava CM via Avava SES	
	Add Cancel *	Required Fields	
		54	
	<		
	E Done	🔒 🧐 Local intranet 🛒	

Step	Description				
4.6	To associate incomi Service Map entry • Click URI t Note • Click the Ac	ing calls to Avaya Me follows: o Service Map under : The entry for a Bas Id button.	eeting Exchange r System Maps . <i>icCallFlow</i> was	with a DirectC configured in S	allFlow, add a URI to Step 4.5.
	Αναγα	Meeting Exc	hange Express Enstall Engineer	Edition	
	Help Log Out Installation	Configuration Provisioning	-		
	System Configuration System Address Data Retention Times Directories Blast Dial Controls Playback Controls Bridge Configuration Media Server SIP Agent SIP Proxies System Maps URI to Service Map TelNum to URI Map System Prompts Prompt Sets Conference Maps Moderator Key Map	URI to Service Map Order URI Pattern 1 sip:401@*	Call Flow BasicCallFlow	Greeting greeting	Service Name "401" from Avaya CM via Avay SES
	Participant Key Map Report Configuration Reports	Add dit Delete Move U	IP Move Down tal: 1 Rows/Page: 10	Refresh	

Step	Description		
4.7	Configure a URI to Service Map Parameter for a DirectCallFlow as follows:		
	• Leave the Order	field at the default value.	
	Note: Avaya Meeting Exchange parses all URI to Service Map entries, searching for		
	pattern matches in descending order, terminating the search once a pattern is matched. For these Application Notes, order is irrelevant as the patterns for BasicCallFlow (see		
	Step 4.5) and Dir	ectCallFlow are mutually exclusive.	
	• Add a URI Patte	rn to match the pattern of incoming Request URIs in SIP INVITE	
	messages from A	vaya SIP Enablement Services.	
	o For examp	102 168 13 212:5061:transport-tls SID/2 0 The LIDI Dettorn below	
	sip.444@	red to match sin·444@* which will match sin·444@ then any string	
	following	the @	
	• Set the Call Flow	to DirectCallFlow to allow conference participants to enter a	
	conference as mo	derator, without entering a passcode.	
	• Enter a descriptiv	e name for the Service Name field.	
	• When finished, cl	ick the Add button.	
	Meeting Exchange Exchange Exchange	opress Edition - Microsoft Internet Explorer 🗧 🗖 🔀	
	Add URI to	Service Map Parameter	
	* Order	2	
		cin.1118t	
	* URI Pattern	510.4440	
	* coll them	DiseaseOutElaur	
	* Call Flow	DirectCaliFlow	
	Greeting	greeting	
	* Service Name	"444" from Avaya CM via Avaya SES	
	Add Cancel *	Required Fields	
	100		
	E Done	🗎 😼 Local intranet 🛒	
	e Done	🖃 🥞 Locai Intranet	



Step	Description		
4.9	Configure a TelNum to URI Map Parameter as follows:		
	• Leave the Order field at the default value.		
	Note: Avaya Meeting Exchange parses all TelNum to URI Map entries, searching for		
	pattern matches in descending order, terminating the search once a pattern is matched		
	For these Application Notes, order is irrelevant as there is only one entry in the table.		
	• Add a Telephone Number Pattern and SIP URI Pattern to allow for Dial-Out from		
	Avaya Meeting Exchange.		
	Note: The configuration for these Application Notes sends all Dial-Out traffic (* = $match all$) to Avaya SIP Engligement Services (102,168,11,20). To englise secure SIP		
	match all) to Avaya SIF Enablement Services (192.100.11.20). 10 enable secure SIF connectivity utilizing TLS for Dial Out, the SIP URI Pattern must contain 5061 and		
	transnort_tls in the entry Avava Meeting Fychange will substitute "\$1" with the dialed		
	number in outgoing SIP INVITE messages. e.g., if 31001 is dialed. Avava Meeting		
	Exchange will send a SIP INVITE message with:		
	sip:31001@192.168.11.20:5061;transport=tls in the SIP URI and To header field.		
	• Enter a descriptive label for the Comment field		
	• When finished, click the Add button.		
	Meeting Exchange Express Edition - Microsoft Internet Explorer		
	Add TelNum to URI Map Parameter		
	* Order 1		
	* Telephone		
	Number Pattern		
	sip:\$10192.168.11.20:5061;transport=tls		
	SIP UKI Patterii		
	To Avaya SIP Enablement Services		
	Add Cancel * Required Fields		

Step	Description		
4.10	0 Following all updates to Avaya Meeting Exchange configured via the web browser, reboot		
	Avaya Meeting Exchange as follows:		
	• Log in to the Avaya Meeting Exchange Server console with the appropriate credentials.		
	• At the command prompt, enter the command init 6 .		
	<i>Note: Rebooting Avaya Meeting Exchange is service impacting.</i>		
	[S6100]> init 6		

4.1. Provision Accounts

The following steps provide examples of how to provision accounts on Avaya Meeting Exchange. Accounts are utilized in conjunction with the Call Flows provisioned in **Step 4.5** and **Step 4.7**.

Step	Description		
4.11	To utilize the DirectCallFlow provisioned in Step 4.7 , administer an Account CSV file as		
	follows:		
	• Log in to the Avaya Meeting Exchange Server console with the appropriate credentials.		
	• Create an Account CSV file with the format of the myAccount.csv shown below.		
	Note: The myAccount.csv file is correlated to the URI Pattern provisioned in Step 4.7		
	via the def_modpass_code entry. The myAccount.csv file is also correlated to the		
	myBlastDial.csv file provisioned in Step 4.12 via the import_tag entry.		
	[SE100]> ast /ugr/tmp/ggyEilog/mulggoupt ggy		
	account_note,def_confpass_code, def_modpass_code ,mx_conf_size,mx_confdur_mins, import _		
	<pre>tag,disabled_ind,logon_password,contact_name,contact_phone,contact_email,import_tag,</pre>		
	conf_profile_id,message_profile_id "DirectDial 444","1444"," 444 ","250","30"," 444 Tag ","f","444","CSV Account		
	444","1234551444","csv@account444.com","CSV_Company_5","5",""		
	• Write the myAccount.csv file to the database by running the bulk-loader.sh utility as		
	follows:		
	 cd to /usr/crystal/bulkloader. 		
	• At the command prompt, enter the command shown below.		
	[S6100]> sh bulk-loader.sh -A/usr/tmp/csvFiles/myAccount.csv		
	com.avaya.crystal.common.Logger.LogDir set to: /usr/crystal/config//logs		
	Log configuration file [/usr/crystal/config/CrystalLog.xml] loaDING.		
	Write Account File :All 1 row(s) were successfull		

Step	Description		
4.12	To enable the Blast Dial feature, administer a Blast Dial CSV file as follows:		
	• Create a Blast Dial CSV file with the format of the myBlastDial.csv shown below.		
	Note: The myBlastDial.csv file is correlated to the myAccount.csv file provisioned in		
	Step 4.11 via the reservation_import_tag entry. The contact_phone variable is the		
	number dialed when the Blast Dial feature is invoked.		
	[S6100]> cat /usr/tmp/csvFiles/myBlastDial.csv		
	<pre>reservation_import_tag,contact_name,contact_phone,contact_email,person_import_tag "444 Tag" "BlastDialContact4" "31001" "csy@blastdialcontact4 com" "PersonImportTag4"</pre>		
	"444_Tag", "BlastDialContact5", "32001", "csv@blastdialcontact5.com", "PersonImportTag5"		
	"444_Tag", "BlastDialContact6", "32002", "csv@blastdialcontact6.com", "PersonImportTag6"		
	"444_Tag", "BlastDialContact7", "33002", "csv@blastdialcontact7.com", "PersonImportTag7"		
	• Write the myBlastDial.csv file to the database by running the bulk-loader.sh utility as		
	follows:		
	o cd to /usr/crystal/bulkloader.		
	• At the command prompt, enter the command shown below.		
	[S6100]> sh bulk-loader.sh -B/usr/tmp/csvFiles/myBlastDial.csv		
	com.avaya.crystal.common.Logger.LogDir not set, setting log location to default		
	<pre>com.avaya.crystal.common.Logger.LogDir set to: /usr/crystal/config//logs Log configuration file [/usr/crystal/config/CrystalLog.xml] loaDING. Log configuration file [/usr/crystal/config/CrystalLog.xml] was loaded.</pre>		
	Write BlastDial File :All 4 row(s) were successfull		

5. Avaya SIP Enablement Services Configuration

This section describes the steps for configuring Avaya SIP Enablement Services to enable secure SIP connectivity between Avaya Communication Manager and Avaya Meeting Exchange utilizing TLS.

Step	Description			
5.1	Administer settin	Administer settings for Avaya SIP Enablement Services as follows:		
	• Open a w	eb browser and e	nter the following URL:	
	https://<	LP address of Av	aya SIP Enablement Servi	ces>/admin
	• Log III to	Avaya SIF Ellau	iement Services with the app	propriate credentials.
	Αναγα			Integrated Management Standard Management Solutions
	Help			bandara management bolations
		•	ogon	
			Logon]
5.2	Click Launch A	dministration W	eb Interface.	
	AVAYA			Integrated Management Standard Management Solutions
	Help Log Off			
	•	Administration	The Administration Web Interface allows you to administer this SES Server.	Launch Administration Web Interface
		Maintenance	The Maintenance Web Interface allows you to maintain, troubleshoot, and configure the server.	<u>Launch Maintenance Web</u> Interface

Step	Description				
5.3	Verify the System Properties for Avaya SIP Enablement Services as follows.				
	From the Administr	ration Web Inter	face:		
	• Click the +	sign to expand th	ne options under	er Server Configuration.	
	Click System	m Properties.			
	• Verify the S	SIP Domain mate	ches the authori	ritative domain configured for the IP	
	network reg	ion on Avaya Co	ommunication N	Manager in Step 3.4.	
					_
	AVAYA			Integrated Manageme SIP Server Manageme	nt
	Help Exit			Server: 192.168.11	.20
	T				
	Op Users	Edit System P	roperties		
	• Conferences	SES Version	SES-3.1.1.0-114.0		
	Media Server Extensions	System Configuration	simplex		
	Emergency Contacts	Host Type	home/edge		
	 Media Servers 	SID Domain*	24272.000		
	• Adjunct Systems	Note that the DNS doma	in is: avava.com		
	Services	If you are unsure about	this field, most often the	e SIP	
	Server Configuration Suctom Properties	domain should be the root level DNS domain. For example, for a DNS domain of east-coast example.com, the STP			
	Admin Accounts	domain would likely be configured to example.com. This			
	License	of the format handle@ex	ample.com	with handles	
	IM Log Settings		1232		
	SNMP Configuration	License Host*	SES		
	IM Loos	Network Properties			
	▪ Trace Logger	Local IP	192.168.11.20		12
	Export/Import to ProVision	Local Name	SES.avaya.com		
		Logical Name	SES.avaya.com		
		Gateway IP Address	192.168.11.1		
		Redundant Properties			
		Management Device	SAMP		
		Fields marked * are requi	red.		
		Update			

5.1. Enable Dial-Out from Avaya Meeting Exchange

The following steps enable secure SIP trunking between Avaya SIP Enablement Services and Avaya Communication Manager. This will allow Dial-Out from Avaya Meeting Exchange to Avaya Communication Manager via Avaya SIP Enablement Services (see **Figure 2**).



Description		
The Add Media Server Interfac	ce page is displayed.	
 To enable secure SIP connectivity to Avaya Communication Manager, provision SIP Trunk parameters as follows: Enter a descriptive name for Media Server Interface Name field. Set the SIP Trunk Link Type to TLS, consistent with the configuration for the signaling group provisioned on Avaya Communication Manager in Step 3.6. Enter the IP address of the C-LAN on Avaya Communication Manager (see Step 3.5) in the SIP Trunk IP Address field. Click the Add button when finished. [Not Shown] Click the Continue button on the confirmation page. 		
AVAYA		
Help Exit	7	
Top Setup • Users	Add Media Server	Interface
 Conferences Media Server Extensions 	Media Server Interface Name* Host	AvayaCM01
Emergency Contacts Hosts	SIP Trunk	
Media Servers List	SIP Trunk Link Type SIP Trunk IP Address*	O TCP O TLS 192.168.11.10
Add Adjunct Systems Services Server Configuration	Media Server Media Server Admin Address (see Help)	
 Certificate Management IM Logs 	Media Server Admin Login Media Server Admin Password	
 Trace Logger Export/Import to ProVision 	Media Server Admin Password Confirm Fields marked * are required.	
	Add	

Step	Description			
5.6	To route SIP traffic to Avaya Communication Manager, provision a Media Server Address			
	Map for the corresponding med	ia server configured in Step 5.5 by clicking Map.		
	AVAYA	Αναγα		
	Help Exit			
	Top ¤ Users	List Media Servers		
		<u>Commands</u> <u>Interface</u> <u>Host</u>		
	Media Server Extensions Emergency Contacts	Edit Extensions Map Test-Link Delete AvayaCM01 192.168.11.20		
	 Hosts 			
	E Media Servers	Add Another Media Server Interface		
	List			
	Add			
	Adjunct Systems			
	Services			
	Server Configuration			
	Certificate Management			
	IM Logs			
	Trace Logger			
	Export/Import to ProVision			
	Update			

Step	Description	Description		
5.7	Click Add Map In New Gro	lick Add Map In New Group.		
	AVAYA			
	Help Exit			
	Top ■ Users	List Media Server Address Map		
	Conferences Media Server Ext	Host AvayaCM01		
	Emergency Cont	acts No address map entries.		
	 Hosts Media Servers 	Add Map In New Group		
	List			
	Add			
	🖪 Adjunct Systems			
	Services			
	💻 Server Configura	tion		
	📱 Certificate Mana	gement		
	IM Logs			
	💻 Trace Logger			
	📮 Export/Import to	ProVision		
	Update			

Step	Description
5.8	The Add Media Server Address Map page is displayed.
	 To match the pattern of incoming SIP INVITE messages (from Avaya Meeting Exchange) destined for Avaya Communication Manager, configure settings for the Media Server Address Map as follows: Enter a descriptive name for the Name field. Enter a Pattern that corresponds to the following: The dial plan configuration for station extensions on Avaya Communication Manager (for these Application Notes, station extensions on Avaya Communication Manager are 5 digits in length with a leading 3, see Step 3.8 and Figure 1). Note: The URI usually takes the form sip:user@domain, where domain can be a domain name or an IP address. For these Application Notes, user is actually the telephone number of the phone. An example of a URI sent by a SIP endpoint to Avaya SIP Enablement Services would be sip:31001@192.168.11.20. The Pattern ^sip:[3][0-9][4] matches the string sip:3 (if it occurs at the beginning of the URI), followed by 4 more digits, each in the range 0 through 9. To replace the URI with the contact displayed in Step 5.9, select Replace URI. Click the Add button when finished. [Not Shown] Click the Continue button on the confirmation page.
	Αναγα
	Help Exit
	Top □ Users □ Conferences ■ Media Server Extensions Emergency Contacts ■ Hosts ■ Hosts ■ Media Servers List Add ■ Adjunct Systems Services ■ Server Configuration □ Certificate Management IM Logs ■ Trace Logger ■ Export/Import to Provision Update
	 Hergency contacts Hosts Media Servers List Add Adjunct Systems Services Server Configuration Certificate Management IM Logs Trace Logger Export/Import to ProVision Update



5.2. Enable Dial-In to Avaya Meeting Exchange

The following steps enable secure SIP trunking between Avaya SIP Enablement Services and Avaya Meeting Exchange. This will allow Dial-In to Avaya Meeting Exchange from Avaya Communication Manager via Avaya SIP Enablement Services (see **Figure 3**).



Step Description	n	
5.11 The Add H	ost page is displayed	1.
To enable s	ecure SIP connectivi	ity for this host, provision as follows:
• Ente	er the password assig	gned to the database at installation for the DB Password field.
• Ente	er a password which	uniquely identifies Avaya SIP Enablement Services for intra-
and	inter-proxy commun	nication for the Profile Service Password field.
• Sele	ect TLS from the ava	ilable Link Protocols, which is consistent with the SIP agent
con	figuration defined for	r Avaya Meeting Exchange in Step 4.3.
• Lea	ve all remaining requ	aired fields at the default settings.
• Clic	k the Add button wh	ien finished.
	o [Not Shown] Cli	ck the Continue button on the confirmation page.
	• [Not Shown] To	apply the administration, click on Update on the left side of the
	page.	
	AVAYA	
	Help Exit	
	Ton	
	Setup	Add Host
	 Users Conferences 	Host IP 192.168.11.20
	• Media Server Extensions	DB Password*
	Emergency Contacts Hosts	Profile Service
	List	Host Type home/edge 🛩
	Add Migrate Home/Edge	Parent NONE
	Media Servers	
	 Adjunct Systems Services 	Access Control
	Server Configuration	Emergency Ollow Openv
	 Certificate Management IM Logs 	Contacts Policy Contacts Policy
	Trace Logger	Registration 300 Registration Expiration Timer (seconds)* 86400 (seconds)
	Export/Import to Provision	Line Reservation Timer (seconds) 30
		* Outbound
		Routing Allowed VInternal External From
		OutboundProxy Port OUDP OTCP OTLS
		Outbound Direct
		Domains
		Default Ringer 5 Default Ringer Cadence* 2
		Default Receiver 5 Default Speaker Volume* 5
		VMM Server
		VMM Server 5005 VMM Report Period 5
		Fields marked * are required.



AVAYA	
Help Exit	
Top Setup • Users	List Host Address Map
 Conferences Media Server Extensions 	Host 192.168.11.20
Emergency Contacts Hosts	Add Map Ip New Group
List Migrate Home/Edge	
Media Servers Adjunct Systems Convictors	
Services Server Configuration	
Certificate Management IM Logs	
 Trace Logger Export/Import to ProVision 	

Step	Description	
5.14	The Add Host Address Map page is displ	ayed.
	To match the pattern of incoming SIP INV Exchange, configure settings for the Host A • Enter a descriptive name for the Na • Enter a Pattern that corresponds to Exchange in Step 4.5 and Step 4.7. <i>Note: The Pattern ^sip:[4][</i> <i>beginning of the URI), follo</i> • To replace the URI with the contact • Click the Add button when finished • [Not Shown] Click the Contact	 TTE messages destined for Avaya Meeting Address Map as follows: ame field. the call flows provisioned for Avaya Meeting
	Help Exit	
	 Top Setup Users Conferences Media Server Extensions Emergency Contacts Hosts List Migrate Home/Edge Media Servers Adjunct Systems Services Server Configuration Certificate Management IM Logs Trace Logger Export/Import to ProVision 	Add Host Address Map Host 192.168.11.20 Name* toS6100 Pattern* ^sip:[4][0-9]{2} Replace URI Image: The second s

Αναγα	The second function of the second sec				
Help Exit					
Top Setup	🚦 List H	lost Addr	ess Map		
 Users Conferences 	Host	192.168.11.2	20		
• Media Server Extensions	Commands	Name	Commands	<u>Contact</u>	
Emergency Contacts	Edit Delete	toS6100			
= Hosts	Add Another	Мар	Add Another	Contact	Delet
Update All					
List	Add Map In N	ew Group			
Migrate Home/Edge					
Media Servers					
Adjunct Systems					
Services					
Server Configuration					
Certificate Management					
Irace Logger					
Export/Import to Provision					

Step	Description		
5.16	The Add Host Contact page is displayed.		
	 To enable secure SIP connersip:\$(user)@192.168.13.21 Note: The IP address, port a agent configuration for Ava Enablement Services substituin the incoming SIP INVITE Click the Add button when [Not Shown] Click the 	ctivity to Avaya Meeting Exchange, enter 2:5061;transport=tls in the Contact field. number and transport protocol are consistent with the SIP ya Meeting Exchange defined in Step 4.3 . Avaya SIP tutes " \$(user) " with the user field (i.e., the dialed number) E message. finished. the Continue button on the confirmation page.	
	AVAYA Help Exit		
	 Top Setup Users Conferences Media Server Extensions Emergency Contacts Hosts Update All List Migrate Home/Edge Media Servers Adjunct Systems Services Server Configuration Certificate Management IM Logs Trace Logger Export/Import to ProVision Update 	Add Host Contact Host 192.168.11.20 Handle toS6100 Contact* \$(user)@192.168.13.212:5061;transport=tls Fields marked * are required.	



Step	Description					
5.18	8 Add Avaya Meeting Exchange as a tru	Add Avaya Meeting Exchange as a trusted host on Avaya SIP Enablement Services.				
	 All SIP user agents, proxies and/or gateways to which calls can be routed should be administered as trusted hosts on Avaya SIP Enablement Services. This permits call setup and termination by remote parties to be handled without authentication challenges to a trusted host. This is provisioned at the Avaya SIP Enablement Services command line of the edge server (or as per these Application Notes, at the edge/home server, if only one server is used). Log in to the Avaya SIP Enablement Services console with the appropriate credentials. Add Avaya Meeting Exchange as a trustedhost by entering the following command: trustedhost -a trusted-host-IP-address -n trusting-SES-IP-address [-c 'comment text'] 					
	SES>trustedhost -a 192.168.13.212	-n 192.168.11.20 -c s	56100			
	 Verify trusted host entries by e SES> trustedhost -L 	ntering the following c	ommand: trustedhost -L			
	Third party trusted hosts. Trusted Host IP address SES :	Host IP address	Comment			
	192.168.13.212 192.16	8.11.20	s6100			
5.19	9 To apply the administration defined in on the web browser interface.	Step 5.18, click on Up	odate on the left side of the page			
	Update					

6. Verification Steps

The following steps can be used to verify the configuration described in these Application Notes.

Step	Description
6.1	Verify all members for the SIP trunk group provisioned in Step 3.7 are in-service/idle.
	 From a SAT session: Issue the command "status trunk <n>", where n is the number of the trunk group to verify.</n> Verify that all members in the trunk group are in-service/idle.

Step	Description	1		
6.2	Verify the S	SIP trunk provisioned in Step 3.7 is utilized when a call from an Avava		
	Communics	ation Manager telephone dials in to Avava Meeting Exchange. This step also		
	varifies the conformating applications provisioned in Section 4			
	vermes the	conterencing applications provisioned in Section 4.		
	From a SAT	C session.		
	FIOII a SA			
	•]	Issue the command "list trace tac $\langle n \rangle$ ", where n is the TAC defined for the trunk group provisioned in Step 3.7.		
	• 1	From an endpoint associated with Avava Communication Manager dial 444 to		
		enter a conference as moderator via the DirectCallFlow scenario provisioned in		
		Section 4.		
	Note . The t	race below shows a SIP telephone dialing in to Avava Meeting Exchange via a		
	Dinact Call	Flow A SID tolophone was arbitrarily selected to place the call (Dial In) as the		
	DireciCalif	tow. A SIF telephone was arbitrarily selected to place the call (Diat-In), as the		
	configuratio	on presented in these Application Notes allows any station or trunk type (e.g., SIP,		
	H.323, Digi	tal or Analog) on Avaya Communication Manager access (both Dial-In and Dial-		
	Out) to Ava	ya Meeting Exchange via secure SIP connectivity.		
	list trace	tac 101 Page 1		
		LIST TRACE		
	time	data		
	16:13:40	Calling party station 31002 cid 0x5f		
	16:13:40	Calling Number & Name 31002 SIP 31002		
	16:13:40	active station 31002 cid 0x5f		
	16:13:40	G711MU ss:off ps:20 rn:1/1 192.168.12.13:34008 192.168.11.11:2512		
	16:13:40	xoip: fax:Relay modem:off tty:US 192.168.11.11:2512 uid:0x50020		
	16:13:40	dial 444 route:AAR		
	16:13:40	term trunk-group 1 cid 0x5f		
	16:13:40	dial 444 route:AAR		
	16:13:40	route-pattern 1 preference 1 cid 0x5f		
	16:13:40	seize trunk-group 1 member 40 cid 0x5t		
	16:13:40	Calling Number & Name NO-CPNumber SIP 31002		
	16:13:41	Proceed trunk-group 1 member 40 cid 0x51		
	16:13:41	active trunk-group 1 member 40 cid 0x5t		
	16:13:41	G/11MU ss:off ps:20 rn:1/1 192.168.13.212:42038 192.168.11.11:2516		
	16:13:41	xolp: tax:Relay modem:off tty:US 192.168.11.11:2516 uid:0x50028		
	16:13:41	G/11MU ss:off ps:20 rn:1/1 192.168.13.212:42038 192.168.12.13:34008		
	16:13:41	G/11MU ss:off ps:20 rn:1/1 192.168.12.13:34008 192.168.13.212:42038		

Step	Descriptio	n
6.3	Verify the	SIP trunk provisioned in Step 3.7 is utilized for Dial-Out calls from Avaya Meeting
	Exchange.	This step also verifies the conferencing applications provisioned in Section 4 .
	8	
	From a SA	T session:
	•	Issue the command " list trace tac <n></n> ", where n is the TAC defined for the trunk group provisioned in Step 3.7 .
	•	Enter the appropriate touchtone command (for these Application Notes *9) to invoke a Blast Dial as provisioned is Section 4 .
		L
	Note: The	trace below shows a call originating from Avava Meeting Exchange to a SIP
	telephone.	For brevity, only the trace for the SIP telephone is displayed.
	list trace	e tac 101 Page 1
		LIST TRACE
	time	data
	16:14:56	Calling party trunk-group 1 member 1 cid 0x2182
	16:14:56	Calling Number & Name NO-CPNumber NO-CPName
	16:14:56	active trunk-group 1 member 1 cid 0x2182
	16:14:56	G711MU ss:off ps:20 rn:1/1 192.168.13.212:42054 192.168.11.11:3116
	16:14:56	xoip: fax:Relay modem:off tty:US 192.168.11.11:3116 uid:0x50001
	16:14:56	dial 31001
	16:14:56	term station 31001 cid 0x2182
	16:14:57	active station 31001 cld 0x2182
	16:14:57	G711MU ss:off ps:20 rn:1/1 192.168.13.212:42054 192.168.12.11:34008
	10.14.5/	G/IIMU SS-OIL PS-20 IN-1/1 192.108.12.11.34008 192.108.13.212.42054

Step	Description
6.4	Verify direct IP-to-IP audio connectivity for the SIP telephone dialing in to Avaya Meeting
	Exchange.
	From a SAT session:
	• Issue the command "status trunk t/m (where t is the trunk group and m is the trunk group member obtained from the procedures in Step 6.2)".
	• The Audio Connection Type = ip-direct shows that direct IP-to-IP audio connectivity is enabled for this endpoint.
	<i>Note</i> : An <i>Audio Connection Type = ip-tdm</i> would indicate that direct IP-to-IP audio connectivity is <u>not</u> enabled for an endpoint. For brevity, the screen to verify direct IP-to-IP audio connectivity is displayed only for a SIP telephone.
	status trunk 1/40 Page 1 of 2
	TRUNK STATUS
	Trunk Group/Member: 0001/040Service State: in-service/activePort: T00040Maintenance Busy? noSignaling Group ID:Signaling Group ID:
	Connected Ports: T00032
	Port Near-end IP Addr : Port Far-end IP Addr : Port Signaling: 01A0217 192.168. 11. 10 : 5061 192.168. 11. 20 : 5061
	G.711MU Audio: 192.168.12.13:34008 192.168.13.212:42038 Video: Video Codec:
	Authentication Type: None Audio Connection Type: ip-direct

Step	Description
6.5	Verify that calls to and from Avaya Meeting Exchange are managed correctly, e.g., callers are
	added/removed from conferences.
	 This is verified by the following procedures: Log in to the Avaya Meeting Exchange Server console with the appropriate credentials. At the command prompt, enter the command: watch -t -n 5 -d "ipinfo -l egrep -ci active" This command provides a real time, continuous update of port utilization on Avaya Meeting Exchange.

7. Conclusion

These Application Notes provide administrators with the procedures to configure connectivity between Avaya Communication Manager and the Avaya Meeting Exchange S6100 Conferencing Server via Avaya SIP Enablement Services. This configuration utilizes secure SIP connectivity via TLS based on industry standards.

8. Additional References

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

- 1. Administrator Guide for Avaya Communication Manager, Issue 2.1, Doc ID 03-300509, May 2006.
- 2. *SIP Enablement Services Implementation Guide*, Issue 3, Doc ID 16-300140, February 2006.

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