



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between the Global Crossing VoIP Network and an Avaya IP Telephony Network – Issue 1.0

Abstract

These Application Notes describe the steps for configuring SIP trunking between the Global Crossing VoIP network and an Avaya IP Telephony network consisting of Avaya Communication Manager, Avaya SIP Enablement Services, and various Avaya telephones.

Enterprise customers with an Avaya SIP-based network can connect to the Global Crossing VoIP Network over the Internet and access the PSTN by subscribing to Global Crossing Enterprise VoIP services: VoIP Outbound, VoIP Toll Free, VoIP Local and VoIP On-Net Plus. The Global Crossing VoIP Network and the Avaya SIP network communicate over the Internet using the Session Initiation Protocol (SIP). This solution allows enterprise customers with a converged network to reduce long distance and interconnection costs.

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

1. Introduction

These Application Notes describe the steps for configuring SIP trunking between the Global Crossing VoIP network and an Avaya IP Telephony network consisting of Avaya Communication Manager, Avaya SIP Enablement Services¹, and various Avaya telephones. Enterprise customers with an Avaya SIP-based network can connect to the Global Crossing VoIP Network over the Internet and access the Public Switched Telephone Network (PSTN) by subscribing to Global Crossing Enterprise VoIP Services: VoIP Outbound, VoIP Toll Free, VoIP Local or the VoIP On-Net Plus. The Global Crossing VoIP network and the Avaya SIP network communicate over the Internet using the Session Initiation Protocol (SIP). This solution allows enterprise customers with a converged network to reduce long distance and interconnection costs.

SIP is a signaling protocol designed to provide a common framework for session establishment, modification, and termination for supporting multimedia communications including voice and video. In converged communications, SIP acts as a trunking protocol, enabling the direct interconnection of independent systems with a SIP network interface.

The Global Crossing Enterprise VoIP services covered by this solution include:

- **VoIP Outbound:** Routes long distance and international calls from enterprise customers. The IP voice traffic is transported across Global Crossing's private VoIP platform for off network TDM completion via the PSTN.
- **VoIP Toll Free:** Routes toll free calls between the enterprise customer's network and the PSTN.
- **VoIP Local Service:** Provides local origination for nationwide local numbers through a single connection. This service enables enterprises to extend a local presence to other service areas covered by Global Crossing. Customers are offered local numbers that can be dialed as local calls from any service area covered by Global Crossing. Calls are then routed across Global Crossing's VoIP network to any enterprise location providing a nationwide local presence.
- **VoIP On-Net Plus** – Routes calls via a Virtual Private Network (VPN) to various customer locations using a private dial plan and either IP or TDM access.

Figure 1 illustrates an enterprise site with an Avaya SIP-based network, including Avaya SIP Enablement Services, an Avaya S8300 Server with an Avaya G350 Media Gateway² running Avaya Communication Manager, Avaya Intuity Audix 770 integrated in the Avaya S8300 Server, and Avaya SIP, H.323, digital, and analog endpoints. The enterprise site is connected to the Global Crossing VoIP Network over the Internet and communicates using SIP. Other enterprise sites that are part of the Global Crossing On-Net Plus VPN are also shown. The Global Crossing VoIP Network consists of an Acme Packet Net-Net Session Director, a Sonus Networks PSX Softswitch, and a Sonus Networks GSX Gateway. The Net-Net Session Director is a session border controller, which sends and receives all SIP signaling messages to/from

¹ In release 2.1 or earlier, the product was referred to as Avaya Converged Communications Server.

² This solution is compatible with all other Avaya S8xxx Servers and Media Gateway platforms running Avaya Communication Manager.

Avaya SIP Enablement Services. Although not shown in **Figure 1**, the enterprise site would normally also have alternate routes to the PSTN using traditional trunks.

Note: Security devices, such as firewall and network address translation (NAT) devices, were not included in the configuration. These Application Notes focused on SIP trunking interoperability. However, it is recommended that enterprise customers deploy security devices in a production environment.

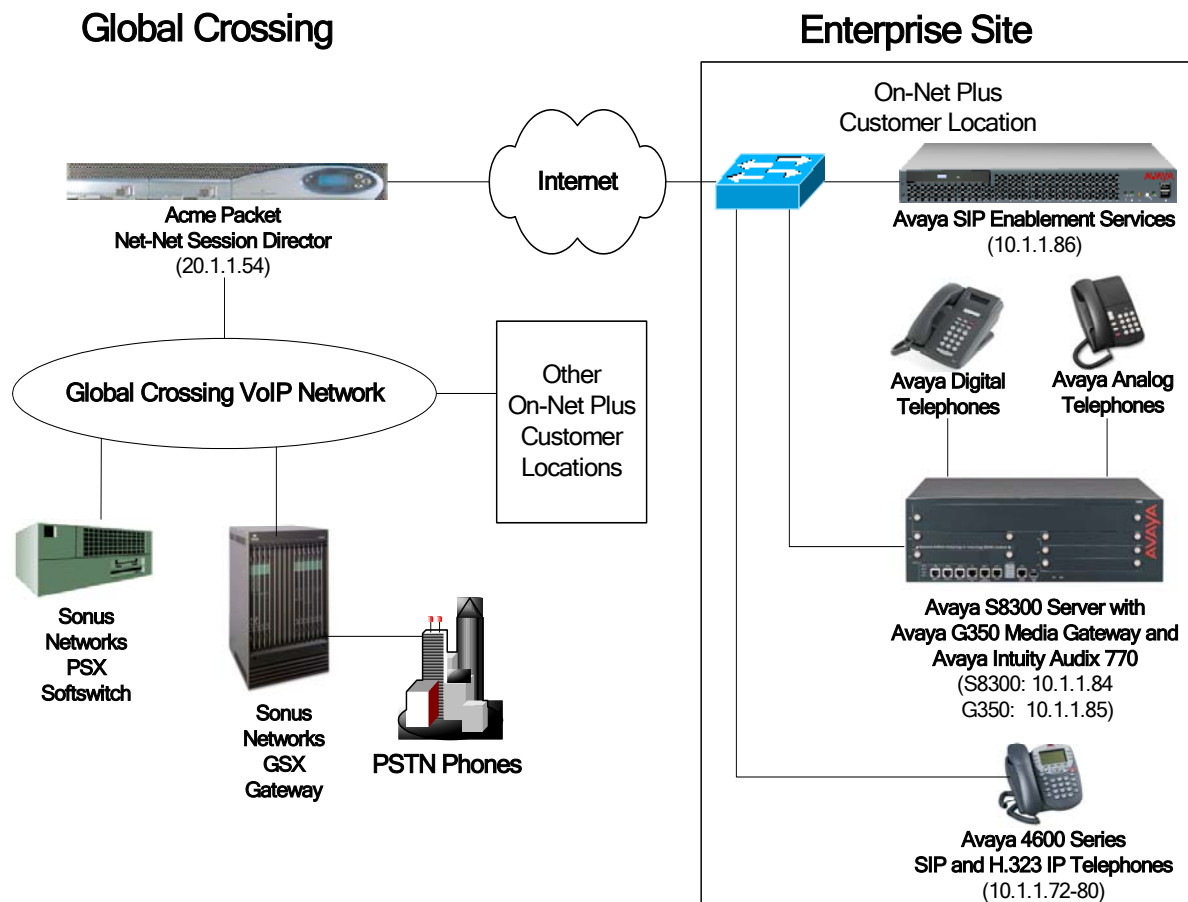


Figure 1: Avaya IP Telephony Network connected to the Global Crossing VoIP Network

1.1. SIP Call Flows

To better understand how calls are routed between the PSTN and the Avaya SIP network shown in **Figure 1**, two call flows are described in this section. The first call flow is a PSTN call to the enterprise site and the second call flow is an outgoing call from the enterprise site to the PSTN. In both cases, the call transits the Global Crossing VoIP Network. **Figure 2** illustrates the call flow for a call originated from the PSTN and terminated at the enterprise site.

1. A user on the PSTN dials a DID number assigned to an Avaya SIP telephone at the enterprise site. The enterprise site subscribes to Global Crossing Enterprise VoIP Services so the call is routed through the Global Crossing VoIP network.
2. Based on the DID number, Global Crossing routes the call to the enterprise site via SIP trunking. Global Crossing sends SIP signaling messages to Avaya SIP Enablement Services at the enterprise site. See Appendix A for an example of a SIP INVITE message sent by Global Crossing.
3. Avaya SIP Enablement Services routes the call to the Avaya S8300 Server running Avaya Communication Manager over a SIP trunk.
4. Since the call is destined for an Avaya SIP telephone, Avaya Communication Manager routes the call back to Avaya SIP Enablement Services over a SIP trunk. If the destination of the call is an H.323, digital or analog endpoint, Avaya Communication Manager terminates the call directly to an endpoint and steps 4 and 5 is not required.
5. Avaya SIP Enablement Services terminates the call to the Avaya SIP telephone.

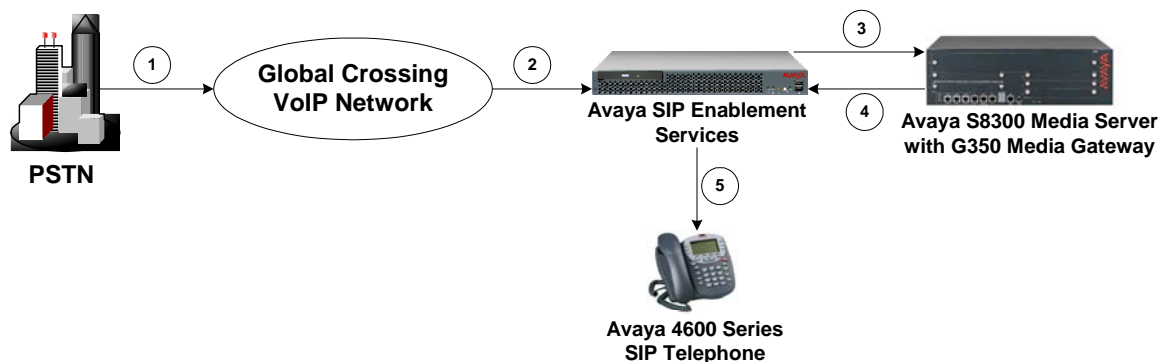


Figure 2: PSTN Call to the Avaya SIP Network

Figure 3 illustrates the call flow for an outgoing call from an Avaya SIP telephone on the Avaya SIP network at the enterprise site to the PSTN.

1. An Avaya SIP telephone originates a call to a user on the PSTN. The call request is delivered to Avaya SIP Enablement Services. If the originator is an H.323, digital or analog endpoint, the call request is sent to Avaya SIP Enablement Services from Avaya Communication Manager.
2. Avaya SIP Enablement Services routes the call over the SIP trunk to the Avaya S8300 Server running Avaya Communication Manager for origination services. This allows Avaya Communication Manager to apply the appropriate call restrictions to the endpoint, handle call routing, and track the status of the SIP telephone, which is an off-PBX station.
3. After applying the origination services, Avaya Communication Manager routes the call back to Avaya SIP Enablement Services over a SIP trunk.
4. Avaya SIP Enablement Services routes the call to the Global Crossing VoIP Network.
5. Global Crossing routes the call to the PSTN.

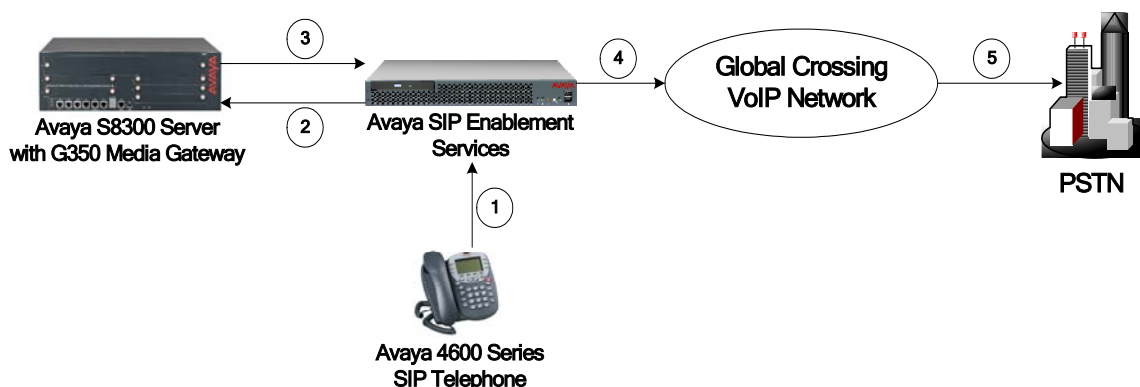


Figure 3: Outgoing Call from the Avaya SIP Network to the PSTN

2. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Hardware Component	Version
Avaya S8300 Server with a G350 Media Gateway and Intuity Audix 770 (IA770)	Avaya Communication Manager 3.1.2 (R013x.01.2.632.1 – update 12866)
Avaya SIP Enablement Services	3.1.0.0-018.0 (with update 10013)
Avaya 4600 Series SIP Telephone	2.2.2
Avaya one-X Desktop Edition	Release 2.1
Avaya 4600 Series H.323 IP Telephones	2.7
Acme Packet Net-Net Session Director	2.1.1 p38
Sonus Networks PSX Softswitch	6.15
Sonus Networks GSX Gateway	6.22

3. Configure Avaya Communication Manager

This section describes the steps for configuring a SIP trunk and SIP Off-PBX stations³ (OPS) on Avaya Communication Manager. The SIP trunk is established between Avaya Communication Manager and Avaya SIP Enablement Services. A SIP Off-PBX station is configured for each Avaya 4600 Series SIP Telephone registered with Avaya SIP Enablement Services. Refer to [2] for additional information on configuring an off-PBX station. All incoming calls from Global Crossing are received by Avaya SIP Enablement Services and routed to Avaya Communication Manager over a SIP trunk for termination services. All outgoing calls to the PSTN are routed through Avaya Communication Manager for origination services. Avaya Communication Manager then routes the call to Avaya SIP Enablement Services, which in turn routes the call to the PSTN through the Global Crossing VoIP network. Note that Avaya SIP Enablement Services provides the SIP interface to the Global Crossing VoIP Network.

The dial plan for the configuration described in these Application Notes is shown in the table below. It illustrates PSTN calling for 10-digit dialing for local and long-distance calls, operator calls (0), and international calls (011). A private dial plan for the On-Net Plus service between two private network locations is also included. Avaya Communication Manager routes all calls using Auto Route Selection (ARS) for PSTN calls and Automatic Alternate Routing (AAR) for the On-Net Plus private network calls. Configuring ARS/AAR is beyond the scope of these Application Notes and the reader should refer to [1] for additional information.

Call Type	Direction	Digits in SIP INVITE message	Destination
Incoming DID	From Global Crossing	602-555-xxxx	Extension 7xxxx
Outbound PSTN Long Distance	To Global Crossing	1 + 10 digits	PSTN phone
Outbound PSTN International	To Global Crossing	011 + international number	International PSTN phone
Outbound PSTN Operator	To Global Crossing	0 or 0 + 10 digits	Operator or Calling Card
Incoming On-Net Plus Private Network Call	From Global Crossing	227-xxxx	Extension 7xxxx
Outbound On-Net Plus Private Network Location	To Global Crossing	116-xxxx	Private Network Location

Table 1 - Dial Plan Used

The Avaya Communication Manager configuration was performed using the System Access Terminal (SAT). The IP network properties of the Avaya S8300 Server were configured via its Maintenance web interface using an Internet browser (not shown).

³ Avaya SIP telephones are configured as off-PBX stations (OPS) on Avaya Communication Manager. Off-PBX stations are not directly controlled by Avaya Communication Manager, but are allowed to access Avaya Communication Manager features. Avaya SIP telephones register with Avaya SIP Enablement Services.

Using the SAT, verify that the Off-PBX telephone and SIP trunk features are enabled using the “display system-parameters customer-options” command. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative.

On Page 1, verify that the number of OPS stations allowed in the system is sufficient.

display system-parameters customer-options		Page 1 of 10
OPTIONAL FEATURES		
G3 Version: V13		
Location: 1	RFA System ID (SID): 1	
Platform: 13	RFA Module ID (MID): 1	
	USED	
Platform Maximum Ports: 900	36	
Maximum Stations: 40	6	
Maximum XMOBILE Stations: 0	0	
Maximum Off-PBX Telephones - EC500: 50	0	
Maximum Off-PBX Telephones - OPS: 50	3	
Maximum Off-PBX Telephones - SCCAN: 0	0	
(NOTE: You must logoff & login to effect the permission changes.)		

Figure 4: System-Parameters Customer-Options – Page 1

On Page 2, verify that the number of SIP trunks supported by the system is sufficient.

display system-parameters customer-options		Page 2 of 10
OPTIONAL FEATURES		
IP PORT CAPACITIES	USED	
Maximum Administered H.323 Trunks: 100	6	
Maximum Concurrently Registered IP Stations: 40	1	
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: 100	24	
Maximum Number of DS1 Boards with Echo Cancellation: 0	0	
Maximum TN2501 VAL Boards: 0	0	
Maximum G250/G350/G700 VAL Sources: 1	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	
(NOTE: You must logoff & login to effect the permission changes.)		

Figure 5: System-Parameters Customer-Options – Page 2

Using the “change system-parameters features” command, set the **Trunk-to-Trunk Transfer** field to *all* to allow calls to be transferred from the enterprise site to an endpoint on the PSTN. Otherwise, leave the field set to *none*. The SIP call flows described in Section 1.1 do not require trunk-to-trunk transfer to be enabled.

```

change system-parameters features                                     Page 1 of 16
      FEATURE-RELATED SYSTEM PARAMETERS
      Self Station Display Enabled? n
      Trunk-to-Trunk Transfer: all
Automatic Callback - No Answer Timeout Interval (rings): 3
      Call Park Timeout Interval (minutes): 10
      Off-Premises Tone Detect Timeout Interval (seconds): 20
      AAR/ARS Dial Tone Required? y
      Music/Tone on Hold: none
      Music (or Silence) on Transferred Trunk Calls? no
      DID/Tie/ISDN/SIP Intercept Treatment: attd
Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
      Automatic Circuit Assurance (ACA) Enabled? n

      Abbreviated Dial Programming by Assigned Lists? n
      Auto Abbreviated/Delayed Transition Interval (rings): 2
      Protocol for Caller ID Analog Terminals: Bellcore
Display Calling Number for Room to Room Caller ID Calls? n
  
```

Figure 6: System-Parameters Features Form

In the **IP Node Names** form, assign an IP address and host name for Avaya SIP Enablement Services at the enterprise site. The host name will be used throughout the other configuration screens of Avaya Communication Manager.

```

change node-names ip                                               Page 1 of 1
      IP NODE NAMES
      Name      IP Address      Name      IP Address
CCS          10 .1 .1 .86      . . .
default        0 .0 .0 .0       . . .
msgserver      10 .1 .1 .73      . . .
procr        10 .1 .1 .84      . . .
      . . .
      . . .
( 4 of 4 administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
  
```

Figure 7: IP Nodes Names Form

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Avaya SIP Enablement Services. In this configuration, the domain name is *devcon.com*. This field is required for endpoints to call the public network. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP and SIP endpoints without using media resources in the Avaya G350 Media Gateway. For calls involving Global Crossing, shuffling is only activated when the call originates or terminates on an Avaya 4600 Series SIP Telephone. In addition, for shuffling to occur, the **DTMF over IP** field must be set to *rtp-payload* (to send DTMF transmission using RFC 2833) as shown in **Figure 11**. The IP Network Region form also specifies the **IP Codec Set** to be used for local calls and calls routed over the SIP trunk to Avaya SIP Enablement Services. This codec set is used when its corresponding network region (i.e., IP Network Region '1') is specified in the SIP signaling group as shown in **Figure 11**.

change ip-network-region 1		Page 1 of 19
IP NETWORK REGION		
Region: 1		
Location:	Authoritative Domain: devcon.com	
Name:		
		Intra-region IP-IP Direct Audio: yes
		Inter-region IP-IP Direct Audio: yes
MEDIA PARAMETERS		IP Audio Hairpinning? y
Codec Set: 1		
UDP Port Min: 2048		
UDP Port Max: 3028		RTCP Reporting Enabled? y
DIFFSERV/TOS PARAMETERS		RTCP MONITOR SERVER PARAMETERS
Call Control PHB Value: 46		Use Default Server Parameters? y
Audio PHB Value: 46		
Video PHB Value: 26		
802.1P/Q PARAMETERS		
Call Control 802.1p Priority: 6		
Audio 802.1p Priority: 6		AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS		RSVP Enabled? n
H.323 Link Bounce Recovery? y		
Idle Traffic Interval (sec): 20		
Keep-Alive Interval (sec): 5		
Keep-Alive Count: 5		

Figure 8: IP Network Region Form

In the **IP Codec Set** form, select the audio codec type used for calls routed over the SIP trunk. The form is accessed via the “change ip-codec-set 1” command. Note that IP codec set ‘1’ was specified in IP Network Region ‘1’ shown in **Figure 8**. The default settings of the **IP Codec Set** form when using G.711MU are shown below. Note, the **IP Codec Set** form may specify multiple codecs, such as G.729 or G.729B, to allow the codec for the call to be negotiated during call establishment.

```
change ip-codec-set 1
```

Page 1 of 2

IP Codec Set

Codec Set: 1

	Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size(ms)
1:	G.711MU	n	2	20
2:				
3:				

Figure 9: IP Codec Set Form Audio Codec

Fax using the T.38 mode codec is specified on Page 2 of the IP Codec Set form by setting **Mode** to *t.38-standard*.

```
change ip-codec-set 1
```

Page 2 of 2

IP Codec Set

Allow Direct-IP Multimedia? n

	Mode	Redundancy
FAX	t.38-standard	0
Modem	off	0
TDD/TTY	off	3
Clear-channel	n	0

Figure 10: IP Codec Set Form FAX Setting

Prior to configuring a SIP trunk group for communication with Avaya SIP Enablement Services, a SIP signaling group must be configured. Configure the **Signaling Group** form shown in **Figure 11** as follows:

- Set the **Group Type** field to *sip*.
- The **Transport Method** field will default to *tls* (Transport Layer Security). TLS is the only link protocol that is supported for SIP trunking with Avaya SIP Enablement Services.
- Specify the Avaya S8300 Server and the Avaya SIP Enablement Services Server as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form shown in **Figure 7**. For other Avaya S8XXX server platforms, the near (local) end of the SIP signaling group may be a C-LAN board instead.
- Ensure that the recommended TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields. If the **Far-end Network Region** field is different from the near-end network region, the preferred codec will be selected from the IP codec set assigned for the inter-region connectivity for the pair of network regions. Although the same network region (Network Region 1) was used for local and PSTN calls in this configuration, different network regions can be used in the field.
- Enter the domain name of Avaya SIP Enablement Services in the **Far-end Domain** field. In this configuration, the domain name is *devcon.com*. This domain is specified in the Uniform Resource Identifier (URI) of the “SIP To Address” in the INVITE message. Configuring this field incorrectly may prevent calls from being successfully established to other SIP endpoints or to the PSTN.
- If calls to/from SIP endpoints are to be shuffled, then the **Direct IP-IP Audio Connections** field must be set to ‘y’.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*. Avaya Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

add signaling-group 1		Page 1 of 1
SIGNALING GROUP		
Group Number: 1	Group Type: sip Transport Method: tls	
Near-end Node Name: procr Near-end Listen Port: 5061	Far-end Node Name: CCS Far-end Listen Port: 5061 Far-end Network Region: 1	
Far-end Domain: devcon.com		
Bypass If IP Threshold Exceeded? n		
DTMF over IP: rtp-payload		Direct IP-IP Audio Connections? y IP Audio Hairpinning? y
Session Establishment Timer(min): 120		

Figure 11: Signaling Group Form

Configure the **Trunk Group** form as shown in **Figure 12**. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the number of members allocated to this SIP trunk group in the **Number of Members** field. The **Number of Members** is the number of simultaneous connections this trunk groups can handle. For a call between the PSTN and a SIP endpoint, two trunk members are used for the duration of the call. For a call between the PSTN and a non-SIP endpoint, one trunk member is used for the duration of the call. Configure the other fields in bold and accept the default values for the remaining fields.

add trunk-group 1		Page 1 of 19	
TRUNK GROUP			
Group Number: 1	Group Type: sip	CDR Reports: y	
Group Name: To CCS	COR: 1	TN: 1	TAC: 101
Direction: two-way	Outgoing Display? n	Night Service:	
Dial Access? n	Busy Threshold: 255		
Queue Length: 0			
Service Type: tie	Auth Code? n		
		Signaling Group: 1	
		Number of Members: 24	
TRUNK PARAMETERS			
Unicode Name? y			
Redirect On OPTIM Failure: 5000			
SCCAN? n			
Digital Loss Group: 18			

Figure 12: Trunk Group Form – Page 1

On Page 2 of the Trunk Group form, set the **Numbering Format** field to *public*. This field specifies the format of the calling party number sent to the far-end.

add trunk-group 1		Page 2 of 19	
TRUNK FEATURES			
ACA Assignment? n	Measured: none	Maintenance Tests? y	
Numbering Format: public			
Replace Unavailable Numbers? n			

Figure 13: Trunk Group Form – Page 2

Configure the **Public/Unknown Numbering Format** form to send the calling party number to the far-end. Add an entry so that local stations with a 5-digit extension beginning with '7' and whose calls are routed over SIP trunk group '1' have the number sent to the far-end for display purposes. In the example shown in **Figure 14**, a CPN prefix is pre-pended to the 5-digit extension so that a 10-digit calling party number (e.g., extension 70000 is converted to 7327770000) is sent to the far-end.

change public-unknown-numbering 0									
NUMBERING - PUBLIC/UNKNOWN FORMAT									
Total									
Ext	Ext	Trk	CPN	CPN	Ext	Ext	Trk	CPN	Total
Len	Code	Grp(s)	Prefix	Len	Len	Code	Grp(s)	Prefix	CPN
5	7	1	73277	10					CPN
									Len

Figure 14: Public/Unknown Numbering Format Form

Configure the **Incoming Call Handling Treatment** form to translate the incoming digits received in the INVITE message to route to the desired extension number. Using the “**change inc-call-handling-trmt trunk-group**” command, translations for both incoming DID numbers and incoming On-Net Plus numbers must be entered. The first entry in **Figure 15** below translates incoming DID numbers in the range 602-555-xxxx to the corresponding the 5 digit extension 7xxxx by deleting the leading 7 digits and inserting the digit 7. The second entry performs the corresponding translation for the On-Net Plus calls made to this location using 227-xxxx as the private network dialing plan.

change inc-call-handling-trmt trunk-group 1				
INCOMING CALL HANDLING TREATMENT				
Service/	Called	Called	Del	Insert
Feature	Len	Number		
tie	10	602555	7	7
tie	7	227	2	

Figure 15: Incoming Call Handling Treatment Form

The first step in configuring an Off-PBX station for the Avaya SIP telephones registered with Avaya SIP Enablement Services is to add an Admin WithOut Hardware (AWOH) station as shown in **Figure 16**. The recommended station type is 6408D+. For an AWOH station, the **Port** field is set to 'X' to indicate that there is no hardware associated with the port assignment. The Class of Restrictions (COR) and Class of Service (COS) assigned to the station should be configured with the appropriate call restrictions. The station is also configured with voicemail coverage using the Intuity Audix 770, an option integrated into the Avaya S8300 Server. Repeat this step for each SIP endpoint at the enterprise site.

add station 70000		Page 1 of 4
STATION		
Extension: 70000	Lock Messages? n	BCC: 0
Type: 6408D+	Security Code:	TN: 1
Port: X	Coverage Path 1: 1	COR: 1
Name: SIP70000	Coverage Path 2:	COS: 1
	Hunt-to Station:	
STATION OPTIONS		
Loss Group: 2	Personalized Ringing Pattern: 1	
Data Module? n	Message Lamp Ext: 70000	
Speakerphone: 2-way	Mute Button Enabled? y	
Display Language: English		
	Media Complex Ext:	
	IP SoftPhone? n	

Figure 16: Admin Without Hardware Station – Page 1

On Page 2 of the station form, verify that the **Per Station CPN – Send Calling Number** field is set to 'y' or blank to allow calling party number information to be sent to the far-end when placing outgoing calls from this station. The default value for this field is blank.

add station 70000		Page 2 of 4
STATION		
FEATURE OPTIONS		
LWC Reception: spe	Auto Select Any Idle Appearance? n	
LWC Activation? y	Coverage Msg Retrieval? y	
LWC Log External Calls? n	Auto Answer: none	
CDR Privacy? n	Data Restriction? n	
Redirect Notification? y	Idle Appearance Preference? n	
Per Button Ring Control? n	Bridged Idle Line Preference? n	
Bridged Call Alerting? n	Restrict Last Appearance? y	
Active Station Ringing: single	Conf/Trans on Primary Appearance? n	
H.320 Conversion? n	Per Station CPN - Send Calling Number? y	
Service Link Mode: as-needed		
Multimedia Mode: enhanced		
MWI Served User Type: qsig-mwi	Display Client Redirection? n	
	Select Last Used Appearance? n	
	Coverage After Forwarding? s	
Remote Softphone Emergency Calls: as-on-local	Direct IP-IP Audio Connections? y	
Emergency Location Ext: 70000	Always Use? n	IP Audio Hairpinning? y

Figure 17: Admin Without Hardware Station – Page 2

On Page 3 of the station form, configure the appropriate number of call appearances for the SIP telephone. For example, the Avaya 4610 SIP Telephone was configured to support three call appearances as shown in **Figure 18**.

add station 70000		Page 3 of 4
STATION		
SITE DATA		
Room:		Headset? n
Jack:		Speaker? n
Cable:		Mounting: d
Floor:		Cord Length: 0
Building:		Set Color:
ABBREVIATED DIALING		
List1:	List2:	List3:
BUTTON ASSIGNMENTS		
1: call-appr	5:	
2: call-appr	6:	
3: call-appr	7:	
4:	8:	

Figure 18: Admin Without Hardware Station – Page 3

The second step of configuring an Off-PBX station is to configure the Stations with Off-PBX Telephone Integration form so that calls destined for a SIP telephone at the enterprise site are routed to Avaya SIP Enablement Services, which will then terminate the call to the SIP telephone. On this form, specify the extension of the SIP endpoint and set the **Application** field to *OPS*. The **Phone Number** field is set to the digits to be sent over the SIP trunk. In this case, the SIP telephone extensions configured on Avaya SIP Enablement Services also match the extensions of the corresponding AWOH stations on Avaya Communication Manager. However, this is not a requirement. Finally, the **Trunk Selection** field is set to '1', the SIP trunk group number. This field specifies the trunk group used to route the call. Another option for routing a call over a SIP trunk group is to use Auto Alternate Routing (AAR) or Auto Route Selection (ARS) routing instead. In this case, the **Trunk Selection** field would be set to *aar* or *ars*. Configuration of other AAR or ARS forms would also be required. Refer to [1] for information on routing calls using AAR or ARS. Repeat this step for each SIP endpoint at the enterprise site.

change off-pbx-telephone station-mapping 70000						Page 1 of 2
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION						
Station Extension	Application	Dial Prefix	Phone Number	Trunk Selection	Configuration Set	
70000	OPS	-	70000	1	1	
		-				

Figure 19: Stations with Off-PBX Telephone Integration – Page 1

On Page 2, set the **Call Limit** field to the maximum number of calls that may be active simultaneously at the station. In this example, the call limit is set to '3', which corresponds to the number of call appearances configured on the station form. Accept the default values for the other fields.

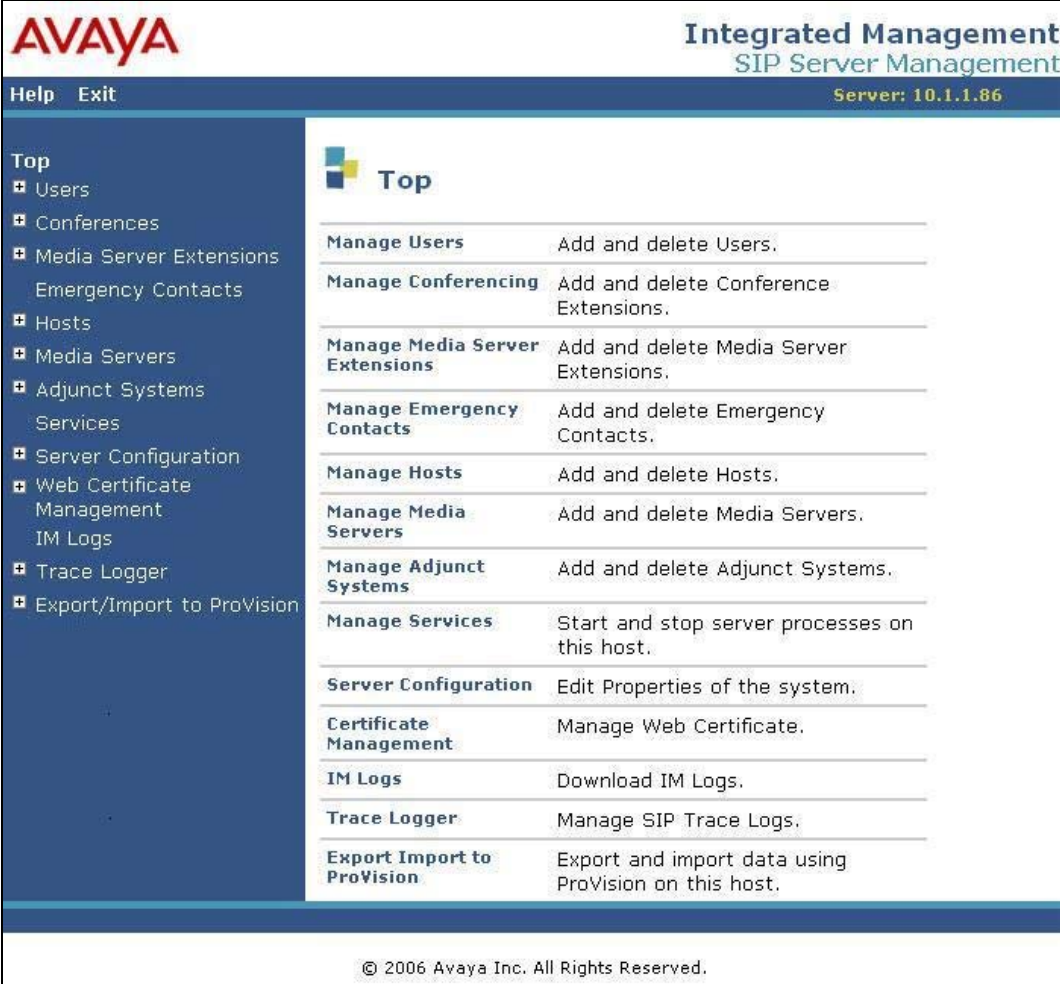
change off-pbx-telephone station-mapping 70000				Page 2 of 2
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION				
Station Extension	Call Limit	Mapping Mode	Calls Allowed	Bridged Calls
70000	3	both	all	both

Figure 20: Stations with Off-PBX Telephone Integration – Page 2

4. Configure Avaya SIP Enablement Services

This section covers the administration of Avaya SIP Enablement Services (SES). Avaya SIP Enablement Services is configured via an Internet browser using the Administration web interface. To access the Administration web interface, enter `http://<ip-addr>/admin` as the URL in an Internet browser, where `<ip-addr>` is the IP address of Avaya SIP Enablement Services. Log in with the appropriate credentials and then select the *Launch Administration Web Interface* link from the **Interface** screen. The main screen shown in **Figure 21** is displayed. After configuring the system, click on the **Update** link that will appear in the left pane for the changes to take effect.

Note: It is assumed that the Avaya SIP Enablement Services software and the license file have already been installed on Avaya SIP Enablement Services. During the software installation, the install script is run on the Linux shell of the server to specify the IP network properties of the server along with other parameters. For additional information on these installation tasks, refer to [3].



Top	
Manage Users	Add and delete Users.
Manage Conferencing	Add and delete Conference Extensions.
Manage Media Server Extensions	Add and delete Media Server Extensions.
Manage Emergency Contacts	Add and delete Emergency Contacts.
Manage Hosts	Add and delete Hosts.
Manage Media Servers	Add and delete Media Servers.
Manage Adjunct Systems	Add and delete Adjunct Systems.
Manage Services	Start and stop server processes on this host.
Server Configuration	Edit Properties of the system.
Certificate Management	Manage Web Certificate.
IM Logs	Download IM Logs.
Trace Logger	Manage SIP Trace Logs.
Export Import to ProVision	Export and import data using ProVision on this host.

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Figure 21: SES Main Screen

From the left pane of the Administration web interface, expand the **Server Configuration** option and select **System Properties**. This screen displays the SES version information and the network properties entered via the install script during the installation process. In the **System Properties** screen, enter the domain name assigned to Avaya SIP Enablement Services and the License Host. For the **License Host** field, enter the host name, the fully qualified domain name, or the IP address of the SIP proxy server that is running the WebLM application and has the associated license file installed. This entry should always be **localhost** unless the WebLM server is not co-resident with this server. After configuring the **System Properties** screen, click the **Update** button.

AVAYA Integrated Management SIP Server Management
 Help Exit Server: 10.1.1.86

Top

- Users
- Conferences
- Media Server Extensions
 - Emergency Contacts
- Hosts
- Media Servers
- Adjunct Systems
 - Services
- Server Configuration**
 - System Properties**
 - Admin Accounts
 - License
 - IM Log Settings
 - SNMP Configuration
- Web Certificate Management
 - IM Logs
- Trace Logger
- Export/Import to ProVision

Edit System Properties

SES Version SES-3.1.0.0-018.0
 System Configuration simplex
 Host Type home/edge

SIP Domain*
 Note that the DNS domain is: devcon.com
 If you are unsure about this field, most often the SIP domain should be the root level DNS domain. For example, for a DNS domain of eastcoast.example.com, the SIP domain would likely be configured to example.com. This allows SIP calls and instant messages to users with handles of the format handle@example.com

License Host*

Network Properties

Local IP 10.1.1.86
 Local Name k2.devcon.com
 Logical IP 10.1.1.86
 Logical Name k2.devcon.com
 Gateway IP Address 10.1.1.65

Redundant Properties

Management Device SAMP

Fields marked * are required.

Update

Figure 22: System Properties

After setting up the domain in the System Properties screen, create a host computer entry for Avaya SIP Enablement Services. The following example shows the **Edit Host** screen since the host had already been added to the system. The **Edit Host** screen is accessible by clicking on the **Hosts** link in the left pane and then clicking on the **Edit** option under the **Commands** section of the subsequent page that is displayed. Enter the host name of this server in the **Host IP Address** field. The value for **DB Password** was specified while running the install script during the system installation. Next, configure the **Host Type** field. In this example, the host server was configured as a *home/edge* so that no additional proxy servers may exist in the enterprise network. The default values for the other fields may be used as shown in **Figure 23**. Click the **Update** button.

AVAYA

Help Exit

Top

- Users
- Conferences
- Media Server Extensions
 - Emergency Contacts
- Hosts
- Media Servers
- Adjunct Systems
 - Services
- Server Configuration
- Certificate Management
 - IM Logs
- Trace Logger
- Export/Import to ProVision

Edit Host

Host IP Address* 10.1.1.86

DB Password

Profile Service Password

Host Type home/edge

Parent none

Listen Protocols ☒ UDP ☒ TCP ☒ TLS

Link Protocols ☐ UDP ☐ TCP ☒ TLS

Presence Access Policy (Default) ☐ Allow All ☒ Deny All

Emergency Contacts Policy ☒ Allow ☐ Deny

Minimum Registration (seconds) 300 Registration Expiration Timer (seconds)* 86400

Line Reservation Timer (seconds)* 30

Outbound Routing Allowed ☒ Internal ☒ External

From

OutboundProxy Port ☐ UDP ☐ TCP ☐ TLS

Outbound Direct Domains

Default Ringer Volume* 5 Default Ringer Cadence* 2

Default Receiver Volume* 5 Default Speaker Volume* 5

VMM Server Address

VMM Server Port 5005 VMM Report Period 5

Fields marked * are required.

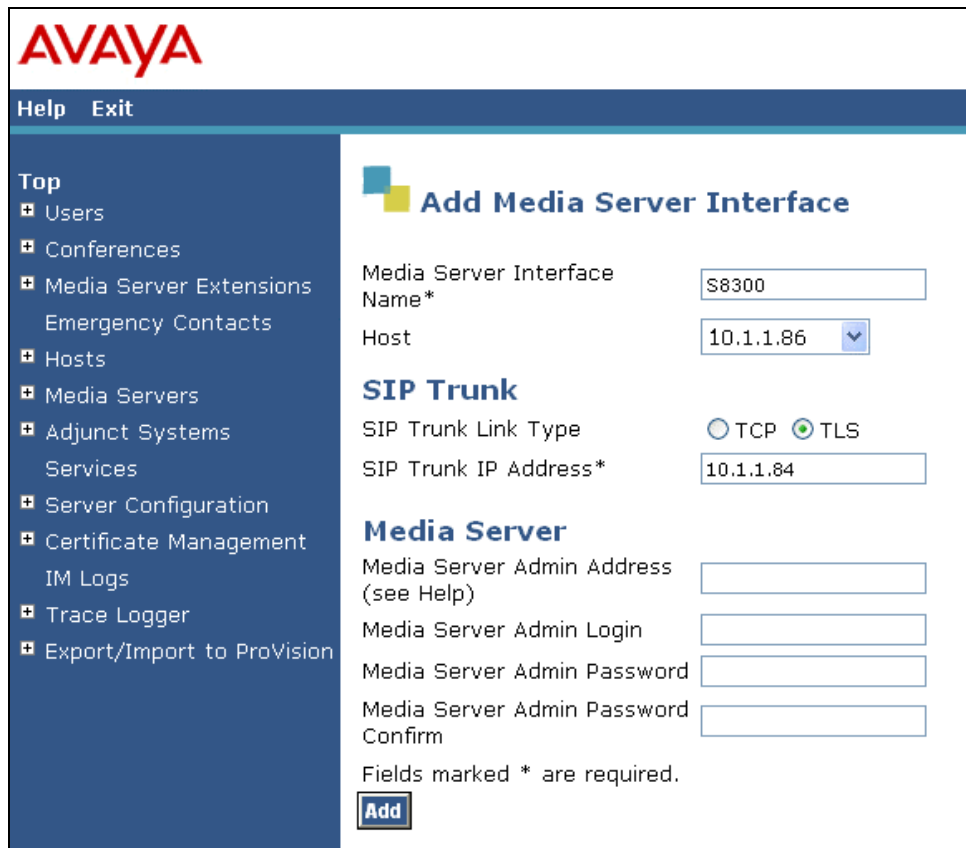
Update

Figure 23: Edit Host

Under the **Media Servers** option in the Administration web interface, select **Add** to add the Avaya S8300 Server in the enterprise site since a SIP trunk is required between Avaya Communication Manager and Avaya SIP Enablement Services. In the **Add Media Server** screen, enter the following information:

- A descriptive name in the **Media Server Interface** field (e.g., S8300).
- Select the home server in the **Host** field.
- Select **TLS** (Transport Link Security) for the **Link Type**. TLS provides encryption at the transport layer. TLS is the only link protocol that is supported for SIP trunking with Avaya Communication Manager.
- Enter the IP address of the Avaya S8300 Server in the **SIP Trunk IP Address** field. Depending on the platform of the Avaya S8XXX server and gateway, this field may be set to the IP address of a C-LAN board.

After completing the **Add Media Server** screen, click on the **Add** button.



The screenshot displays the Avaya Administration web interface. At the top is the Avaya logo. Below it is a navigation menu with options like 'Help', 'Exit', 'Top', 'Users', 'Conferences', 'Media Server Extensions', 'Emergency Contacts', 'Hosts', 'Media Servers', 'Adjunct Systems', 'Services', 'Server Configuration', 'Certificate Management', 'IM Logs', 'Trace Logger', and 'Export/Import to ProVision'. The main content area is titled 'Add Media Server Interface'. It contains several form fields: 'Media Server Interface Name*' with the value 'S8300', 'Host' with a dropdown menu showing '10.1.1.86', 'SIP Trunk Link Type' with radio buttons for 'TCP' and 'TLS' (where 'TLS' is selected), and 'SIP Trunk IP Address*' with the value '10.1.1.84'. Below these are four text input fields for 'Media Server Admin Address (see Help)', 'Media Server Admin Login', 'Media Server Admin Password', and 'Media Server Admin Password Confirm'. At the bottom of the form is an 'Add' button and a note: 'Fields marked * are required.'

Figure 24: Add Media Server

Incoming calls arriving at Avaya SIP Enablement Services are routed to Avaya Communication Manager for termination services. Calls to be routed to Avaya Communication Manager are specified in a **Media Server Address Map** configured on Avaya SIP Enablement Services. The Uniform Resource Identifier (URI) of an incoming INVITE message is compared to the pattern configured in the **Media Server Address Map**, and if there is a match, the call is routed to Avaya Communication Manager. The URI usually takes the form of *sip:user@domain*, where *domain* can be a domain name or an IP address. In this example, *user* is actually the telephone number of the phone. An example of a URI would be *sip:6025554069@20.1.1.54*. Only incoming calls from the PSTN require a media server address map. By default, all calls originated from an Avaya SIP telephone are routed through Avaya Communication Manager for origination services because the Avaya SIP telephones are assigned a media server extension.

To configure a Media Server Address Map, select **Media Servers** in the left pane of the Administration web interface. This will display the List Media Servers screen (not shown). Click on the **Map** link associated with the appropriate media server to display the List Media Server Address Map screen and click on the **Add Map In New Group** link. The screen shown in **Figure 25** is displayed. The Host field will be displayed with the name of the media server. Provide a descriptive name in the **Name** field and enter the regular expression to be used for the pattern matching in the **Pattern** field. In this configuration, the pattern specification matches a URI that begins with *sip:602555* followed by four digits. Note that DID numbers beginning with 602555 were assigned to endpoints at the enterprise site. The **Replace URI** checkbox should remain checked. See Appendix B for a more detailed description of the syntax for address map patterns. Click the **Add** button.

AVAYA

Help Exit

Top

- Users
- Conferences
- Media Server Extensions
 - Emergency Contacts
- Hosts
- Media Servers
- Adjunct Systems
- Services
- Server Configuration
- Certificate Management
- IM Logs
- Trace Logger
- Export/Import to ProVision
- Update

Add Media Server Address Map

Host S8300

Name* DID-602555xxxx

Pattern* ^sip:602555[0-9]{4}

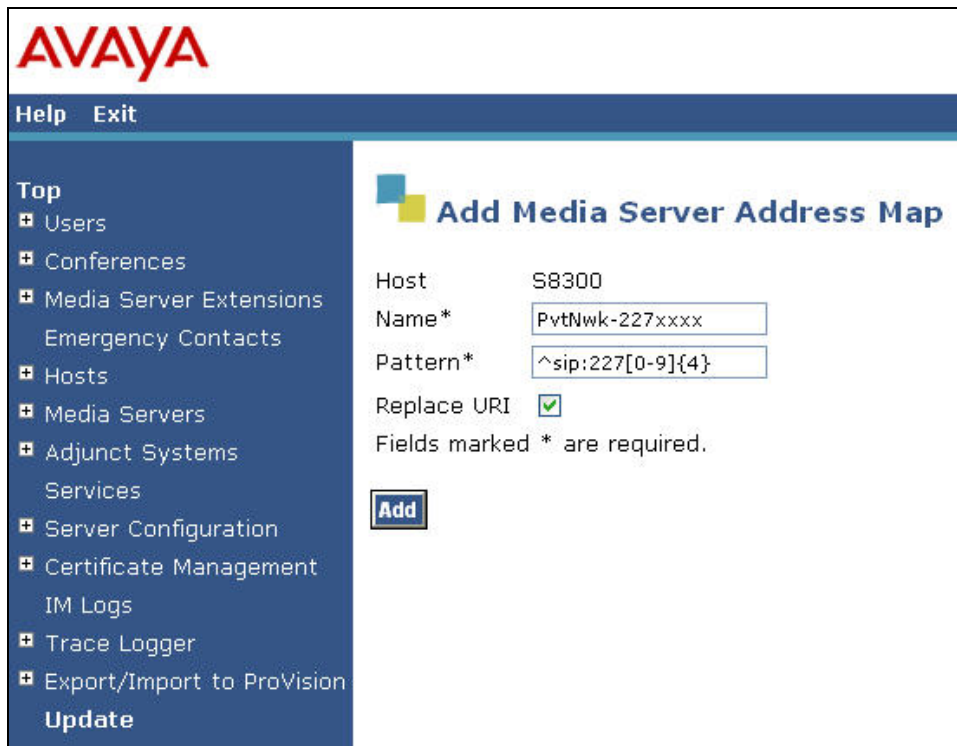
Replace URI ☒

Fields marked * are required.

Add

Figure 25: Media Server Address Map for PSTN DID Numbers

A second **Media Server Address Map** was similarly configured to route incoming calls from other On-Net Plus private network locations. In this case the pattern routes incoming addresses matching 227xxxx to Avaya Communication Manager on the Avaya S8300 server.



The screenshot displays the Avaya web interface for configuring a Media Server Address Map. The interface features a top navigation bar with the Avaya logo and 'Help' and 'Exit' links. A left-hand menu lists various system components, including Users, Conferences, Media Server Extensions, Emergency Contacts, Hosts, Media Servers, Adjunct Systems, Services, Server Configuration, Certificate Management, IM Logs, Trace Logger, and Export/Import to ProVision, with an 'Update' button at the bottom. The main content area is titled 'Add Media Server Address Map' and contains the following configuration fields:

- Host:** S8300
- Name*:** PvtNwk-227xxxx
- Pattern*:** ^sip:227[0-9]{4}
- Replace URI:** ☒

Below these fields, a note states: 'Fields marked * are required.' An 'Add' button is positioned at the bottom of the configuration section.

Figure 26: Media Server Address Map for Private Network Numbers

After the **Media Server Address Map** is added, the **Media Server Contact** is created automatically. For the **Media Server Address Map** added in **Figure 25**, the following contact string was created:

sip:\$(user)@10.1.1.84:5061;transport=tls

The contact specifies the IP address of the Avaya S8300 Server and the transport protocol used to send SIP signaling messages. The user in the original request URI is substituted for *\$(user)*.

After configuring the media server address maps, the List Media Server Address Map screen appears as shown in **Figure 27**.

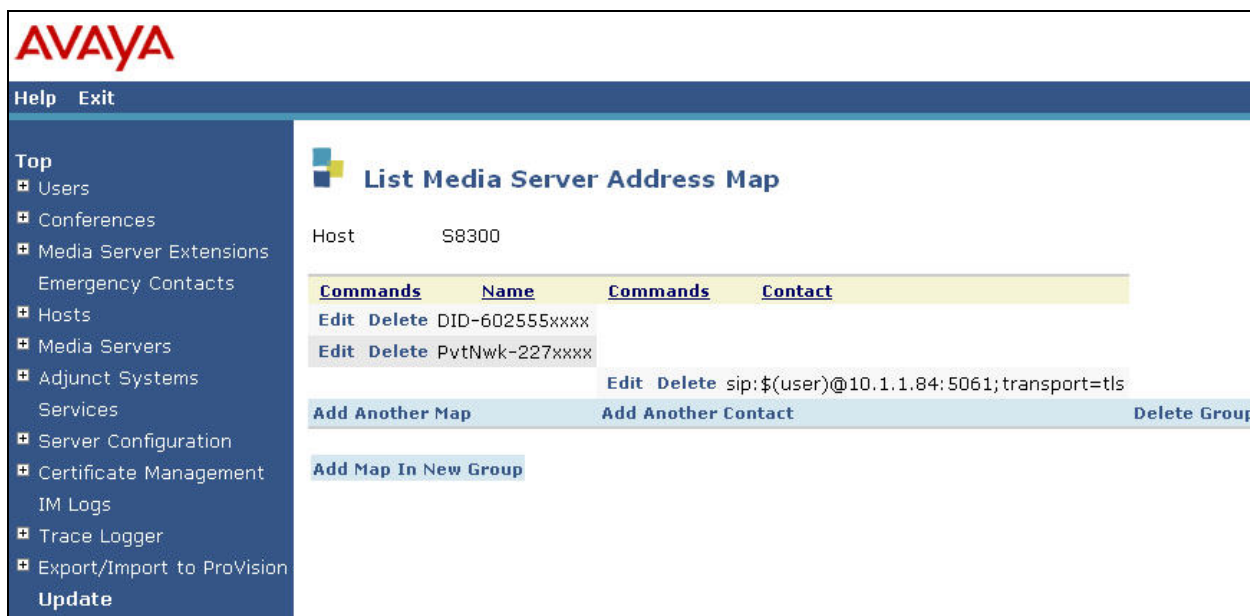


Figure 27: List Media Server Address Map

All calls destined for the Global Crossing VoIP services are routed through Avaya SIP Enablement Services guided by host address maps. In these Application Notes, host address maps for North American long distance, operator and On-Net Plus private network dialing calls are created.

The host address map for “1 plus 10 digit” long distance dialing is shown in **Figure 28**. To access the Add Host Address Map screen, select the **Hosts** link in the left pane of the Administration web interface and then click on the **Map** link associated with the appropriate host (e.g., k2.devcon.com). The List Host Address Map screen is displayed. From this screen, click the **Add Map In New Group** link to display the screen shown in **Figure 28**. Configure a descriptive name for the map and specify an appropriate pattern for the call type. In this example, the pattern is used for “1 plus 10 digit” calls. The **Replace URI** checkbox should be checked. Click the **Add** button.



The screenshot displays the Avaya Administration web interface. The top header features the Avaya logo and navigation links for 'Help' and 'Exit'. A left-hand sidebar contains a tree view of system components, including 'Top', 'Users', 'Conferences', 'Media Server Extensions', 'Emergency Contacts', 'Hosts', 'Media Servers', 'Adjunct Systems', 'Services', 'Server Configuration', 'Certificate Management', 'IM Logs', 'Trace Logger', 'Export/Import to ProVision', and 'Update'. The main content area is titled 'Add Host Address Map' and contains the following fields: 'Host' (pre-filled with 'k2.devcon.com'), 'Name*' (text input with '1plus10-LD'), 'Pattern*' (text input with '^sip:1[0-9]{10}'), and 'Replace URI' (checkbox, checked). A note below the fields states 'Fields marked * are required.' An 'Add' button is located at the bottom left of the form area.

Figure 28: Add Host Address Map Entry for Long Distance

The host address map for outbound operator and international PSTN calling is shown in **Figure 29**.



The screenshot shows the Avaya web interface for adding a host address map. The top navigation bar includes the Avaya logo and 'Help' and 'Exit' links. A left sidebar contains a tree view of system components, with 'Hosts' selected. The main content area is titled 'Add Host Address Map' and contains the following fields:

- Host: k2.devcon.com
- Name*: 0and0plus
- Pattern*: ^sip:0
- Replace URI: ☒

Below the fields is a note: 'Fields marked * are required.' and an 'Add' button.

Figure 29: Add Host Address Map Entry for Operator and International Dialing

The host address map to reach a private network location is shown in **Figure 30**.



The screenshot shows the Avaya web interface for adding a host address map. The top header features the Avaya logo and 'Help Exit' links. A left-hand navigation menu lists various system management options, including 'Top', 'Users', 'Conferences', 'Media Server Extensions', 'Emergency Contacts', 'Hosts' (with sub-options like 'Update All', 'List', and 'Migrate Home/Edge'), 'Media Servers', 'Adjunct Systems', 'Services', 'Server Configuration', 'Certificate Management', 'IM Logs', 'Trace Logger', and 'Export/Import to ProVision'. The main content area is titled 'Add Host Address Map' and contains the following fields and controls:

- Host:** k2.devcon.com
- Name*:** PvtNwk-116xxxx
- Pattern*:** ^sip:116[0-9]{4}
- Replace URI:** ☒
- Fields marked * are required.**
- Add:** A button to submit the form.

Figure 30: Add Host Address Map Entry for a Private Network Location

The contact URL to reach the Global Crossing network used by above Host Address Maps must be explicitly entered. From the List Host Address Map screen, click on the **Add Another Contact** link associated with the address maps added in **Figure 28** through **Figure 30**. In this screen, the **Contact** field specifies the Global Crossing Network as next hop for the SIP call and it is configured as:

sip:\$(user)@20.1.1.54:5060;transport=udp

The contact specifies the IP address of the Acme Net-Net Session Director in the Global Crossing VoIP Network and the transport protocol used to send SIP signaling messages. The transport protocol must be coordinated with Global Crossing. The user in the original request URI is substituted for *\$(user)*. Click the **Add** button when completed.

AVAYA

Help Exit

Top

- 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 k2.devcon.com

Handle PvtNwk-116xxxx

Contact* sip:\$(user)@20.1.1.54:5060;transport=udp

Fields marked * are required.

Add

Figure 31: Add Host Contact

After configuring the host address maps, the **List Host Address Map** screen appears as shown in **Figure 32**.

AVAYA Integrated M SIP Server

Help Exit

Top

- 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

List Host Address Map

Host k2.devcon.com

Commands	Name	Commands	Contact
Edit Delete	0and0plus		
Edit Delete	1plus10-LD		
Edit Delete	PvtNwk-116xxxx		
		Edit Delete	sip:\$(user)@20.1.1.54:5060;transport=udp

Add Another Map Add Another Contact Delete Group

Figure 32: List Host Address Map

Add a user for each Avaya 4600 Series SIP Telephone registering with Avaya SIP Enablement Services. In the Add User screen, enter the extension of the SIP endpoint in the **Primary Handle** field. Enter a user password in the **Password** and **Confirm Password** fields. This password must match the one configured on the SIP telephone. The password entered in these two fields must match. In the **Host** field, select the Avaya SIP Enablement Services server hosting the domain (*devcon.com*) for this user. Enter the user's name into the **First Name** and **Last Name** fields. To associate a media server extension with this user, select the **Add Media Server Extension** checkbox. Calls from this user will always be routed through Avaya Communication Manager over the SIP trunk for origination services. The **Add Media Server Extension** screen shown in **Figure 34** will be displayed after adding this user's profile by clicking on the **Add** button.

AVAYA

Help Exit

Top

- Users
 - List
 - Add
 - Search
 - Edit
 - Delete
 - Password
 - Default Profile
 - Registered 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

Add User

Primary Handle* 70000

User ID 70000

Password*

Confirm Password*

Host* 10.1.1.86

First Name* John

Last Name* Doe

Address 1

Address 2

Office

City

State

Country

Zip

Add Media Server Extension ☒

Fields marked * are required.

Add

Figure 33: Add User

In the **Add Media Server Extension** screen, enter the **Extension** configured on the media server, shown in **Figure 16**, for the previously added user. Usually, the media server extension and the user extension are the same (recommended). Select the **Media Server** assigned to this extension. Click on the **Add** button. To commit the configuration changes, click on the **Update** link in the left pane.

Figure 34: Add Media Server Extension

The last step is to configure the Acme Packet Net-Net Session Director as a trusted host on Avaya SIP Enablement Services. Note that the Acme Packet Net-Net Session Director sends all SIP signaling messages from the Global Crossing VoIP Network. Global Crossing provides the Session Director as the SIP proxy interface. As a trusted host, Avaya SIP Enablement Services will not challenge incoming requests from the Acme Packet Net-Net Session Director. For example, with this trusted host configuration, Avaya SIP Enablement Services will not challenge the BYE SIP message sent by Global Crossing to disconnect the call. This is required because Global Crossing will not respond to the challenge resulting in the call not terminating immediately on the Avaya network side. Instead the call will be dropped after the expiration of a timeout period. If multiple SIP proxies are used, then each SIP proxy must be added as a trusted host. To configure a trusted host, use the *trustedhost* command in the Linux shell of Avaya SIP Enablement Services. As shown in **Figure 35**, the *trustedhost -a* command adds the Acme Packet Net-Net Session Director with IP address 20.1.1.54 as a trusted host of Avaya SIP Enablement Services with a host name of 10.1.1.86. The *trustedhost -L* command allows the list of trusted hosts to be displayed and the *trustedhost -d* command allows a trusted host to be deleted from the system. After configuring a trusted host, the user must go to the Administration web interface and click on the **Update** link for the changes to take effect. This is required even though the trusted host was configured via the Linux shell.

```
admin@k2> trustedhost -a 20.1.1.54 -n 10.1.1.86
20.1.1.54 is added to trusted host list.

admin@k2> trustedhost -L
Third party trusted hosts.
-----+-----+-----+
Trusted Host | CCS Host Name | Comment
-----+-----+-----+
20.1.1.54    | 10.1.1.86     | GlobalCrossing
```

Figure 35: Configuring a Trusted Host

Important Note: After making any configuration changes on Avaya SIP Enablement Services, the user must click on the **Update** link in the Administration web interface for the changes to take effect.

5. Global Crossing VoIP Network Configuration

The following table contains the configuration information, provided by Global Crossing, which was used during the interoperability compliance testing.

Service / Feature Information	Test Configuration
Global Crossing Enterprise VoIP Service(s): <ul style="list-style-type: none">▪ VoIP Outbound▪ VoIP Toll Free▪ VoIP Local Service▪ VoIP On-Net Plus Service	All Global Crossing Enterprise VoIP Services were tested in the Application Notes.
Global Crossing Codec(s): <ul style="list-style-type: none">▪ G.711mu-law▪ G.711a-law▪ G.729▪ G.729B▪ RFC2833 DTMF▪ T.38 for facsimile	The network configuration described in these Application Notes was tested with all the codecs (payload types) listed in the left column, except for G.711 a-law.
Global Crossing Dial Plan	10-Digit North American, International (011 Country Code), and 7 digit private networking calls were supported by the test configuration. The numbering plan and routing choices are affected by the services used and will be customer dependent.
Global Crossing Direct Inward Dialing Numbers	Direct inward dialing numbers should be assigned to the endpoints at the enterprise site. This allows calls to be delivered from the PSTN. In this configuration, direct inward dialing numbers beginning with area code 602-555 were assigned to the SIP, H.323, digital, and analog endpoints in the enterprise network.

Service / Feature Information	Test Configuration
Global Crossing Proxy IP Address	The IP address of the Acme Packet Net-Net Session Director in the Global Crossing network was 20.1.1.54 and used to configure the host address maps in Avaya SIP Enablement Services.
Customer IP Address of Avaya SIP Enablement Services	The IP address of Avaya SIP Enablement Services in the enterprise network was 10.1.1.86. Global Crossing used this IP address for routing calls destined to the listed directory numbers assigned to the enterprise site. For a redundant system, the logical IP address is used.
SIP Transport Protocol and Port	SIP signaling was transported between Avaya SIP Enablement Services and Global Crossing using UDP and port 5060.

6. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between the Global Crossing VoIP network and an Avaya SIP-based network, including Avaya SIP Enablement Services. This section covers the general test approach and the test results.

6.1. General Test Approach

An enterprise site containing an Avaya SIP-based network was interconnected to the Global Crossing VoIP network using SIP trunking. This allowed the enterprise site to access the PSTN through the Global Crossing VoIP network. The following features and functionality were covered during the SIP trunking interoperability compliance test:

- Incoming calls to the Avaya SIP network from the PSTN. PSTN calls were routed through the Global Crossing VoIP network.
- Outgoing calls from the Avaya SIP network to the PSTN through the Global Crossing VoIP network.
- Incoming and outgoing calls to another private network location using a 7 digit private network numbering plan.
- Calls to/from SIP, H.323, digital and analog endpoints in the Avaya enterprise network.
- Various call types including: local, long distance, international, toll free, operator and directory assistance, and private networking calls.
- Calls using G.711 and G.729 codecs.
- DTMF transmission using RFC 2833.
- Fax calls to/from a group 3 fax machine to a PSTN connected fax machine using the T.38 mode.

- Voicemail coverage and retrieval for endpoints at the enterprise site.
- Direct IP-to-IP media (also known as “shuffling” which allows SIP endpoints to send audio (RTP) packets directly to each other without using media resources on the Avaya Media Gateway).

6.2. Test Results

All test cases passed. The Avaya SIP-based network was successful in establishing various call types, such as long distance and international calls, to the PSTN through the Global Crossing VoIP network. Calls were established using SIP trunking over the Internet.

7. Verification Steps

This section provides verification steps that may be performed in the field to verify that the SIP, H.323, digital and analog endpoints can access the PSTN through the Global Crossing VoIP network.

1. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call can remain active for more than 35 seconds. This time period is included to verify that proper routing of the SIP messaging has satisfied SIP protocol timers.
2. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active for more than 35 seconds.
3. Verify that the user on the PSTN can terminate an active call by hanging up.
4. Verify that an endpoint at the enterprise site can terminate an active call by hanging up.
5. If shuffling is enabled, verify that a call originated or terminated on an Avaya 4600 Series SIP Telephone is shuffled. To determine if the call is shuffled, identify the trunk member active on the call by running the **status trunk <group>** command on the SAT of Avaya Communication Manager. Next, run the **status trunk group/member** command and check the **Audio Connection** field. If the call is shuffled, the field should be set to *ip-direct*; otherwise, the field would be set to *ip-tdm*.

8. Support

For technical support on Global Crossing Enterprise VoIP Services, contact the Global Crossing Customer Center at 866-456-2259 (866-GLOBALX).

9. Conclusion

These Application Notes describe the configuration steps required to connect an enterprise site consisting of an Avaya SIP-based network to the Global Crossing VoIP network. This allows enterprise customers to reduce long distance and interconnection costs by accessing the PSTN through the Global Crossing VoIP network. Enterprise customers subscribing to Global Crossing’s VoIP services, including VoIP Outbound, VoIP Toll Free, VoIP Local and On-Net Plus, can receive and place local, long distance, international, directory assistance, operator and toll free calls.

10. References

This section references the Avaya documentation relevant to these Application Notes. The following Avaya product documentation is available at <http://support.avaya.com>.

- [1] *Administrator Guide for Avaya Communication Manager*, May 2006, Issue 2.1, Document Number 03-300509.
- [2] *Avaya Extension to Cellular and Off-PBX Station (OPS) Installation and Administration Guide Release 3.1*, February 2006, Issue 9, Document Number 210-100-500.
- [3] *Installing and Administering SIP Enablement Services*, March 2007, Issue 2.1, Document Number 03-600768.
- [4] *SIP Support in Release 3.1 of Avaya Communication Manager Running on the Avaya S8300, S8500, S8500B, S8700, and S8710 Media Server*, March 2007, Issue 6.1, Document Number 555-245-206.

Additional information about the Global Crossing Enterprise VoIP Services is available at <http://www.globalcrossing.com>.

APPENDIX A: Sample SIP INVITE Messages

This section displays the format of the SIP INVITE messages sent by the Global Crossing VoIP Network and the Avaya SIP Network at the enterprise site. Customers may use these INVITE messages for comparison and troubleshooting purposes. Differences in these messages may indicate different configuration options selected.

Sample SIP INVITE Message from Global Crossing VoIP Network:

```
Session Initiation Protocol
Request-Line: INVITE sip:6025554069;npdi=yes@10.1.1.86:5060 SIP/2.0
Method: INVITE
Resent Packet: False
Message Header
Via: SIP/2.0/UDP 20.1.1.54:5060;branch=z9hG4bK2iij4m2048ulnbghj0g0.1sr
From: <sip:7325551638@20.1.1.54:5060>;tag=SD18n4901-gK020b71b5
    SIP from address: sip:7325551638@20.1.1.54:5060
    SIP tag: SD18n4901-gK020b71b5
To: <sip:6025554069@10.1.1.86:5060>
    SIP to address: sip:6025554069@10.1.1.86:5060
Call-ID: SD18n4901-1078adf431bdef166f2bab28e134a6bb-v3000i1
CSeq: 23091 INVITE
Max-Forwards: 69
Allow: OPTIONS, INVITE, ACK, CANCEL, BYE, REFER, INFO, SUBSCRIBE, NOTIFY, PRACK, UPDATE
Accept: multipart/mixed, application/sdp, application/isup, application/dtmf,
application/dtmf-relay
Contact: <sip:7325551638@20.1.1.54:5060;transport=udp>
Content-Length: 288
Content-Disposition: session; handling=required
Content-Type: application/sdp
Message body
Session Description Protocol
    Session Description Protocol Version (v): 0
    Owner/Creator, Session Id (o): Sonus_UAC 13795 29028 IN IP4 20.1.1.54
        Owner Username: Sonus_UAC
        Session ID: 13795
        Session Version: 29028
        Owner Network Type: IN
        Owner Address Type: IP4
        Owner Address: 20.1.1.54
    Session Name (s): SIP Media Capabilities
    Connection Information (c): IN IP4 20.1.1.54
        Connection Network Type: IN
        Connection Address Type: IP4
        Connection Address: 20.1.1.54
    Time Description, active time (t): 0 0
        Session Start Time: 0
        Session Stop Time: 0
    Media Description, name and address (m): audio 10362 RTP/AVP 18 0 8 101
        Media Type: audio
        Media Port: 10362
        Media Proto: RTP/AVP
        Media Format: ITU-T G.729
        Media Format: ITU-T G.711 PCMU
        Media Format: ITU-T G.711 PCMA
        Media Format: 101
    Media Attribute (a): rtpmap:18 G729/8000
        Media Attribute Fieldname: rtpmap
        Media Attribute Value: 18 G729/8000
    Media Attribute (a): rtpmap:0 PCMU/8000
```

```

Media Attribute Fieldname: rtpmap
Media Attribute Value: 0 PCMU/8000
Media Attribute (a): rtpmap:8 PCMA/8000
Media Attribute Fieldname: rtpmap
Media Attribute Value: 8 PCMA/8000
Media Attribute (a): rtpmap:101 telephone-event/8000
Media Attribute Fieldname: rtpmap
Media Attribute Value: 101 telephone-event/8000
Media Attribute (a): fmp:101 0-15
Media Attribute Fieldname: fmp
Media Attribute Value: 101 0-15
Media Attribute (a): sendrecv
Media Attribute (a): maxptime:20
Media Attribute Fieldname: maxptime
Media Attribute Value: 20

```

Sample SIP INVITE Message from Avaya SIP Enablement Services to Global Crossing:

```

Session Initiation Protocol
Request-Line: INVITE sip:7325551638@20.1.1.54:5060;transport=udp SIP/2.0
Method: INVITE
Resent Packet: False
Message Header
Call-ID: 0a993e9d18da12a3e43176d700
CSeq: 1 INVITE
From: "70000" <sip:70000@10.1.1.86>;tag=0a993e9d18da1293e43176d700
SIP Display info: "70000"
SIP from address: sip:70000@10.1.1.86
SIP tag: 0a993e9d18da1293e43176d700
Record-Route:
<sip:10.1.1.86:5060;transport=udp;lr>,<sip:10.1.1.84:5061;lr;transport=tls>
To: sip:7325551638@devcon.com
SIP to address: sip:7325551638@devcon.com
Via: SIP/2.0/UDP
10.1.1.86:5060;branch=z9hG4bK03030316161693939348dd.0,SIP/2.0/TLS
10.1.1.84;psrrposn=2;branch=z9hG4bK0a993e9d18da12b3e43176d700,SIP/2.0/UDP
10.1.1.72:5060;psrrposn=1;branch=z9hG4bK1f0b47acb
Content-Length: 160
Content-Type: application/sdp
Contact: "70000" <sip:7327770000@10.1.1.84:5061;transport=tls>
Max-Forwards: 67
User-Agent: Avaya CM/R013x.00.0.340.3
Allow: INVITE,CANCEL,BYE,ACK,PRACK,SUBSCRIBE,NOTIFY,REFER,OPTIONS
Session-Expires: 240;Refresher=uac
Min-SE: 240
Supported: 100rel,timer,replaces,join
P-Asserted-Identity:"SIP70000"<sip:7327770000@devcon.com:5061>
Message body
Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): - 1 1 IN IP4 10.1.1.84
Owner Username: -
Session ID: 1
Session Version: 1
Owner Network Type: IN
Owner Address Type: IP4
Owner Address: 10.1.1.84
Session Name (s): -
Connection Information (c): IN IP4 10.1.1.85
Connection Network Type: IN
Connection Address Type: IP4

```

Connection Address: 10.1.1.85
Time Description, active time (t): 0 0
Session Start Time: 0
Session Stop Time: 0
Media Description, name and address (m): audio 3026 RTP/AVP 0 127
Media Type: audio
Media Port: 3026
Media Proto: RTP/AVP
Media Format: ITU-T G.711 PCMU
Media Format: 127
Media Attribute (a): rtpmap:0 PCMU/8000
Media Attribute Fieldname: rtpmap
Media Attribute Value: 0 PCMU/8000
Media Attribute (a): rtpmap:127 telephone-event/8000
Media Attribute Fieldname: rtpmap
Media Attribute Value: 127 telephone-event/8000
Data (2 bytes)

APPENDIX B: Specifying Pattern Strings in Address Maps

The syntax for the pattern matching used within Avaya SES is a Linux regular expression used to match against the URI string found in the SIP INVITE message.

Regular expressions are a way to describe text through pattern matching. The regular expression is a string containing a combination of normal text characters, which match themselves, and special *metacharacters*, which may represent items like quantity, location or types of characters.

In the pattern matching string used in the Avaya SES are the following:

- Normal text characters and numbers match themselves.
- Common metacharacters used are:
 - A period `.` matches any character once (and only once).
 - An asterisk `*` matches zero or more of the preceding characters.
 - Square brackets enclose a list of any character to be matched. Ranges are designated by using a hyphen. Thus the expression `[12345]` or `[1-5]` both describe a pattern that will match any single digit between 1 and 5.
 - Curly brackets containing an integer 'n' indicate that the preceding character must be matched exactly 'n' times. Thus `5{3}` matches '555' and `[0-9]{10}` indicates any 10 digit number.
 - The circumflex character `^` as the first character in the pattern indicates that the string must begin with the character following the circumflex.

Putting these constructs together as used in this document, the pattern to match the SIP INVITE string for any valid "1+ 10 digit" number in the North American Dial Plan would be:

`^sip:1[0-9]{10}`

This reads as: "Strings that begin with exactly "**sip:1**" and having any 10 digits following will match.

A typical INVITE request below uses the shaded portion to illustrate the matching pattern.

INVITE **sip:17325551638**@20.1.1.54:5060;transport=udp SIP/2.0

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