



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between the Priority Netherlands SIP Trunking Standard Service and an Avaya Aura™ SIP Telephony Solution – Issue 1.0

Abstract

These Application Notes describe the steps to configure SIP trunking between the Priority Netherlands SIP trunking standard service and an Avaya Aura™ SIP telephony solution consisting of Avaya Aura™ Communication Manager, Avaya Aura™ SIP Enablement Services, and various Avaya telephones.

Enterprise customers with this Avaya SIP-based solution can connect via dedicated Internet access to use Priority as a service provider to complete PSTN calls. This allows customers with a converged network to lower PSTN telecommunication costs, obtain local number presence without offices in each geographic area, and easily manage their network services using web-based tools.

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

1. Introduction

These Application Notes describe the steps for configuring Session Initiation Protocol (SIP) trunking between the Priority Netherlands SIP trunking standard service and an Avaya Aura™ SIP telephony solution consisting of Avaya Aura™ SIP Enablement Services (SES), Avaya Aura™ Communication Manager and Avaya IP telephones using H.323 and SIP protocols as endpoints.

SIP is a standards-based communications approach designed to provide a common framework to support multimedia communication. RFC 3261 [8] is the primary specification governing this protocol. In the configuration described in these Application Notes, SIP is used as the signaling protocol between the Avaya components and the network service offered by Priority Netherlands. SIP manages the establishment and termination of connections and the transfer of related information such as the desired codec, calling party identity, etc.

1.1. Priority Netherlands SIP Trunking Standard Service - IP Trunking Overview

The Priority Netherlands SIP trunking standard service with IP trunking used within these Application Notes serves as an interface between Avaya Telephones and ISDN, GSM and analog endpoints communicating with the Priority Netherlands network.

The Priority SIP Trunking standard service uses UDP as transport protocol.

Priority Netherlands SIP Trunking service offers end-customers the opportunity to minimize costs for PSTN calls. The Priority Netherlands SIP Trunking service usually includes the trunks, low cost per calls, data network access to the Priority Media PoP and a CPE (Customer Premises Equipment) router with NAT Traversal feature enabled on it. There is no need for equipment other than a SIP Enabled Avaya Aura™ Communication Manager as described in this document. There are special cases described later in this document where some modifications in the architecture are needed in order to use some features either in the Priority Network or in the Avaya Aura™ Communications Manager solution. Contact a Priority or Avaya Netherlands representative for more information.

1.2. Support

Priority Netherlands: **Phone:** 08000212131
Phone from out of Netherlands: +31 8000212131
Fax: 0881212714
Fax from out of Netherlands: +31 881212714
Email: csc@priority.nl
from January 1st 2010: csc@upcbusiness.nl
Webpage: www.priority.nl
from January 1st 2010: www.upc.nl/upcbusiness

2. SIP Trunking Configuration

Figure 1 illustrates a sample Avaya SIP telephony solution connected to Priority Netherlands SIP trunking service using SIP trunking. This is the configuration that was used during the DevConnect compliance testing.

The simulated customer configuration included:

- Two Avaya S8730 Media Servers with an Avaya G650 Media Gateway. The S8730 served as the processor for Avaya Aura™ Communication Manager.
- Avaya Aura™ SIP Enablement Services operating on a SES Combined Home/Edge server (S8500C).
- Avaya 9600-Series IP telephones using H.323 software
- Avaya 9600-Series IP telephones using SIP software
- Avaya Digital phones
- Avaya Analog phones
- Fax machines

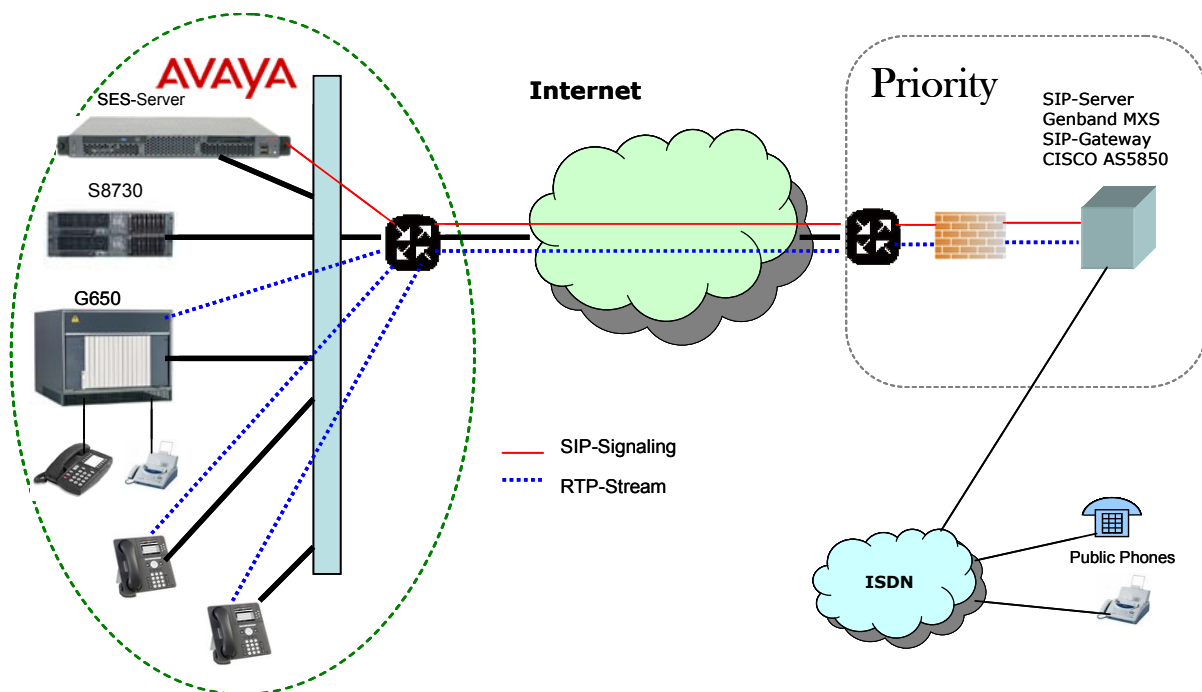


Figure 1: Avaya Aura™ SIP Telephony Solution using Priority Netherlands SIP trunking service

2.1. Call Flows

To better understand how calls are routed between the Priority Netherlands network and the enterprise site shown in **Figure 1** using SIP trunks, two call flows are described in this section.

The first call scenario illustrated in **Figure 2** is a PSTN call to the enterprise site terminating on a telephone supported by Communication Manager.

1. A user on the Priority Netherlands network dials a Priority Netherlands SIP trunking service provided number that is assigned to a Communication Manager telephone at the enterprise site.
2. Based on the DID number, the Priority Netherlands SIP trunking service offers the call to SES using SIP signaling messages sent over the managed access facility. Note that the assignment of the number and the address of the SES server were previously established during the ordering and provisioning of the service.
3. SES routes the call to the Avaya S8730 Media Server running Communication Manager over a SIP trunk.
4. Communication Manager terminates the call to an Avaya H.323/Analog/Digital telephone as shown in **Figure 2**.
- or -
- 4.a. Inbound calls destined for a SIP Telephone at the enterprise are routed to Communication Manager. Communication Manager, acting as a SIP back-to-back user agent, then transmits the appropriate SIP signaling via SES to the telephone (as shown by the 4a arrow.)

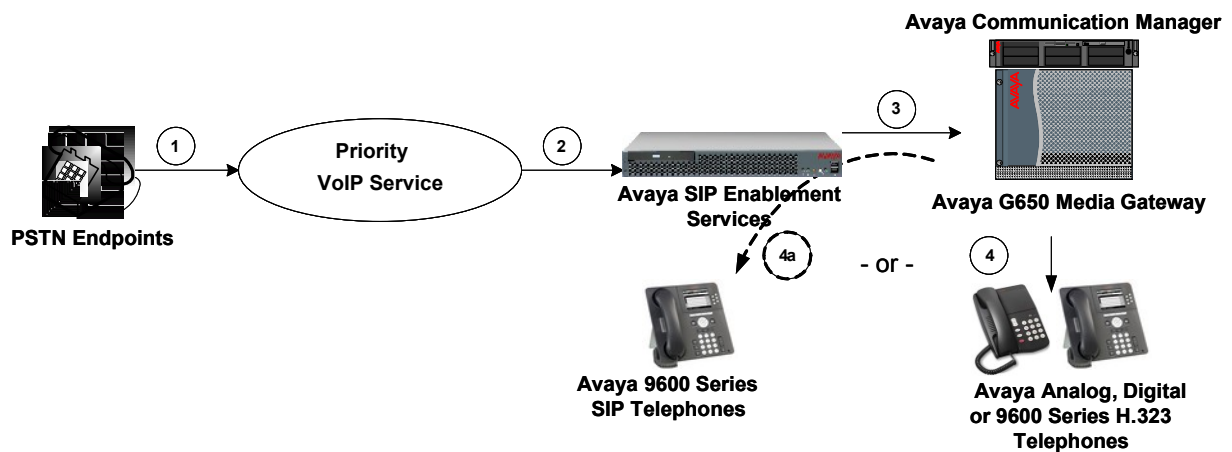


Figure 2: Incoming Priority Netherlands Network Calls to Avaya Aura™ Communication Manager

The second call scenario illustrated in **Figure 3** is an outgoing call from an Avaya telephone at the enterprise site to the Priority Netherlands network via the SIP trunk to the Priority Netherlands SIP trunking service.

1. An Avaya H.323, analog or Digital telephone served by Communication Manager originates a call to a user on the Priority Netherlands network.
- or -
- 1a. An Avaya SIP telephone originates a call that is routed via SES (as shown by the 1a arrow) to Communication Manager.
2. The call request is handled by Communication Manager where origination treatment such as Class of Restrictions (COR) and Automatic Route Selection (ARS) is performed. Communication Manager selects the SIP trunk and sends the SIP signaling messages to SES.
3. SES routes the call to the Priority Netherlands SIP trunking service.
4. The Priority Netherlands SIP trunking service completes the call to the Priority Netherlands network.

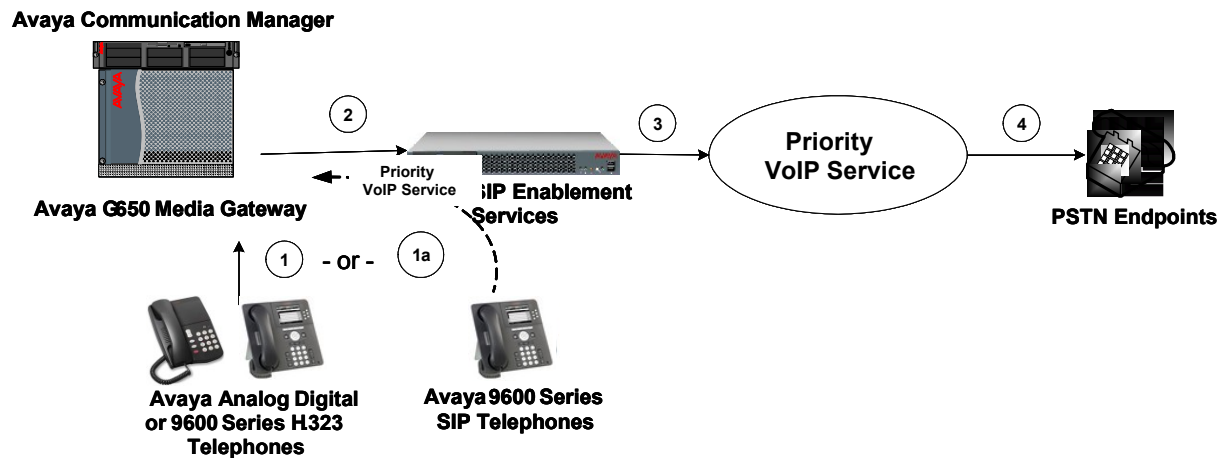


Figure 3: Outgoing Calls from Avaya Aura™ Communication Manager to the Priority Netherlands Network

Appendix A illustrates examples of the SIP INVITE messages sent on the SIP trunk to begin each call.

3. Equipment and Software Validated

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

Avaya SIP Telephony Solution Components	
Component	Software Version
Avaya Aura™ Communication Manager on S8730 Media Server	5.2 (R015x.02.0.947.3) <i>Patch: 02.0.947.3-17579</i>
Avaya G650 Media Gateway TN2312BP IP Server Interface TN799DP Control-LAN TN2602AP IP Media Processor TN799DP Control LAN TN793CP Analog Line TN2214CP Digital Line	HW28 FW044 HW16 FW035 HW02 FW049 HW01 FW034 HW07 FW010 HW08 FW015
Avaya SIP Enablement Services on S8500C	5.2 (SES05.2-02.0.947.3b) <i>Patch: SES-02.0.974.3-SP2a</i>
Avaya 9630 H.323 IP Telephones	3.0 <i>Application file: ha96xxua3_0_02.bin</i> <i>Boot file: hb96xxua3_00.bin</i>
Avaya 9630 SIP IP Telephones	2.4 <i>Application file: SIP96xx_2_4_1_0.bin</i> <i>Boot file: hb96xxua1_50.bin</i>
Avaya 1616 IP Telephones	1.2 <i>Application file: ha1616ua1_2000.bin</i> <i>Bm32a2_1.hex</i> <i>Boot file: hb1616ua1_2000.bin</i>
Avaya one-X® Communicator	R1.040-SP4-17370
Avaya 2420 Digital Telephones	n / a
Avaya 6221 Analog Telephones	n / a
Priority Netherlands SIP Trunking Service Components	
Component	Version
Session Border Controller Genband MSX	Software Release 4.3m7
Media Gateway Cisco AS5850	Software Release 12.4(25)

Table 1: Equipment and Software Used in the Tested Configuration

The specific configuration above was used for the Priority Netherlands trunking service interoperability testing. Note that this solution is extensible to other Avaya Server and Media Gateway platforms running similar versions of Communication Manager and SIP Enablement Services.

4. Configure Avaya Aura™ Communication Manager

This section describes the steps for configuring a SIP trunk on Communication Manager with the necessary signaling and media gateway characteristics for the SIP trunk connection with the Priority Netherlands SIP trunking standard service (UDP).

In the Avaya SIP trunk architecture, the SES acts as a SIP proxy through which all incoming and outgoing SIP messages flow to the Priority Netherlands SIP trunking service. There is no direct SIP signaling path between the Priority Netherlands SIP trunking service and Communication Manager or Avaya SIP endpoints.

The SIP trunk configuration also allows a shared trunking solution that supports SIP endpoint devices (such as Avaya 9600 Series SIP telephones) using Communication Manager in the recommended Avaya Off-PBX Station (OPS) configuration. In this configuration, the SIP endpoints are registered with SES but have calling privileges and features managed by Communication Manager. Communication Manager acts as a Back-to-Back SIP user agent for the SIP endpoint, when a call involves a SIP trunk to the Priority Netherlands SIP Trunk Service.

For incoming calls, SES uses media server address maps to direct the incoming SIP messages to the appropriate Communication Manager. Once the message arrives at the Communication Manager, further incoming call treatment such as incoming digit translations, class of service restrictions, etc., may be performed.

All outgoing calls to the Priority Netherlands network are processed within Communication Manager and are subject to outbound features such as automatic route selection (ARS), digit manipulation and class of service restrictions (COS). Once Communication Manager selects a SIP trunk, the SIP signaling is routed to SES. Within SES, host address maps direct the outbound SIP messages to the Priority Netherlands SIP trunking SIP Application Server. Note that SES acts as a SIP proxy through which all incoming and outgoing SIP messages flow to the Priority Netherlands SIP trunk service.

The Communication Manager configuration was performed using the System Access Terminal (SAT). The initial installation and configuration of the S8730 Servers, G650 Media Gateway and associated media modules are presumed to have been previously completed and are not discussed within these Application Notes.

4.1. SIP Trunk Configuration

4.1.1 Verify System Capacity

Using the SAT, verify that sufficient SIP Trunk capacity exists by displaying the **System-Parameters Customer-Options** form shown in **Figure 4** and **Figure 5**. The Communication Manager license file controls the customer options. Contact an authorized Avaya sales representative for assistance if insufficient capacity exists or a required feature is not enabled.

On Page 1 of the **System-Parameters Customer-Options** form, verify that the number of OPS stations available is sufficient for the number of SIP telephones to be used.

display system-parameters customer-options		Page 1 of 11
OPTIONAL FEATURES		
G3 Version: V15	Software Package: Standard	
Location: 2	RFA System ID (SID): 1	
Platform: 6	RFA Module ID (MID): 1	
		USED
	Platform Maximum Ports: 48000	74
	Maximum Stations: 36000	14
	Maximum XMOBILE Stations: 0	0
	Maximum Off-PBX Telephones - EC500: 500	1
	Maximum Off-PBX Telephones - OPS: 500	6
	Maximum Off-PBX Telephones - PBFMC: 0	0
	Maximum Off-PBX Telephones - PVFMC: 0	0
	Maximum Off-PBX Telephones - SCCAN: 0	0
(NOTE: You must logoff & login to effect the permission changes.)		

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

On Page 2, verify that the number of available **SIP Trunks** is sufficient for the combination of trunks to the Priority Netherlands SIP trunking service, trunks for SIP endpoints and any other SIP trunking applications.

Note that each call from a non-SIP endpoint to the Priority Netherlands SIP trunking service uses one SIP trunk for the duration of the call. However, due to the SIP OPS configuration, each SIP endpoint on a call with the Priority Netherlands SIP trunking service requires two SIP trunks for the duration of the call.

display system-parameters customer-options		Page 2 of 11
OPTIONAL FEATURES		
IP PORT CAPACITIES		USED
Maximum Administered H.323 Trunks: 100		5
Maximum Concurrently Registered IP Stations: 18000		4
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 Stations: 0		0
Maximum Video Capable IP Softphones: 0		0
Maximum Administered SIP Trunks: 100		55
Maximum Administered Ad-hoc Video Conferencing Ports: 0		0
Maximum Number of DS1 Boards with Echo Cancellation: 0		0
Maximum TN2501 VAL Boards: 128		0
Maximum Media Gateway VAL Sources: 0		0
Maximum TN2602 Boards with 80 VoIP Channels: 128		0
Maximum TN2602 Boards with 320 VoIP Channels: 128		1
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 Form – Page 2

4.2. Configure Special Applications

Using the SAT, verify the Special Applications by displaying the **system-parameters special-applications** form shown in **Figure 6**. The Communication Manager license file controls the Special-applications. Contact an authorized Avaya sales representative for assistance if a required feature is not enabled.

On Page 8 of the **system-parameters special-applications** form, verify that the feature Overwrite Calling Identity (SA8972) is enabled. When the SA8972 feature is enabled, the outgoing INVITE for a forwarded call (CFU, CFB, CFNA) contains the number of the forwarding station in the FROM header and P-Asserted-Identity. This number is considered valid by the Priority Netherlands Service and the call will be established. Otherwise the forwarded call will have the original calling party number, and when the Priority service checks the number, the service considers the number invalid and changes the number to the default number of the SIP trunk.

The following should be noted:

- On the display of the forwarded-to party is indicated the number of the forwarding station instead of the number of the calling station.
- The SA8972 feature does not apply to EC500. On the display of the called party (EC500 destination, such as a cell phone) is indicated the default number of the SIP trunk instead of the number of the calling station.

display system-parameters special-applications	Page 8 of 9
SPECIAL APPLICATIONS	
(SA8972) - Overwrite Calling Identity? y	
(SA8983) - Replace CPN when calls forwarded off-net? n	
(SA8986) - Simultaneous Ring of Attendants? n	
(SA8993) - Increase Tenant Partitions? n	
(SA9004) - Multi-Location Call Routing for IP-DECT? n	
(SA9009) - Duplicated Processor Ethernet for Sip? n	
(SA9014) - PIN Release of TOD Lock? n	
(SA9016) - G.726 Support on TN2302? n	
(SA9018) - Support 127 Bridge Appearances per Call-appr? n	
(SA9020) - IGAR triggered by IP network performance? n	
(SA9025) - ISDN Subaddress Treatment? n	
(SA9028) - Station TBCT for Heavily Trunked Systems? n	
(SA9035) - Increased Intercom Groups? n	
(SA9050) - Increased TOD Routing Tables/Partition Grp Num? n	
(SA9062) - Permissive Use CMS and EDP? n	
(SA9065) - Crisis Alert to Stations by Location? n	
(SA9070) - Increased Logged-In ACD Agents for XL? n	
(SA9073) - Use Called Party Location For LWC Time/Date? n	

Figure 6: System-parameters Special-applications Form – Page 8

4.2.1 Determine Node Names

In the **IP Node Names** form, view (or assign) the node names to be used in this configuration using the **change node-names ip** command shown in **Figure 7**.

SES and **192.168.193.110** are used below as the **Name** and **IP Address** of the SES server at the enterprise site.

```
change node-names ip
```

		IP NODE NAMES
Name	IP Address	
Gateway001	192.168.193.1	
clan-cab-a-2	192.168.193.106	
clan-cabinet-a	192.168.193.104	
default	0.0.0.0	
medpro-cabineta	192.168.193.105	
medpro2caba	192.168.193.107	
procr	0.0.0.0	
ses	192.168.193.110	

(10 of 10 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

4.2.2 Define IP Codec Sets

Priority Netherlands SIP trunking service supports the codecs G711A, G729 and T38 (Fax).

- Define a codec set (**Codec Set 1** below) to support voice and Fax calls with the Priority Netherlands SIP trunking service. G.729 and G.711A are the codecs to be used.
- The only accepted packet size for Priority Netherlands SIP Trunking service is 20 ms and 2 Frames per packet.

Use the **change ip-codec-set** command (**Figure 8 and 8a**) to define the codec set. On Page 2 of **ip-codec-set** form, the **Fax Mode** must be **t.38-standard**, and the **Modem** and **TTD/TTY Mode** fields must be **off**.

```
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.711A	n	2	20
2: G.729AB	n	2	20
3: G.729	n	2	20
4:			
5:			
6:			
7:			

Media Encryption

1: **none**

2:

3:

Figure 8: IP Codec Set Forms for Voice

```
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	0
Clear-channel	n	0

Figure 8a: IP Codec Set Forms for Fax

4.2.3 Verify Near End IP Network Region

The **IP Network Region** form defines the parameters associated with the SIP trunk group serving the SES proxy (used to reach Priority Netherlands and any optional SIP endpoints). In the **IP Network Region** form shown in **Figure 9**:

- The **Authoritative Domain** field is configured to match the domain name configured on the SES server. In this configuration, the domain name is "*localdomain.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 endpoints without using media resources in the Avaya G650 Media Gateway.
- The **Codec Set** is set to the number of the ip-codec-set ("1" for voice and fax) to be used for calls within IP network region 1.¹

change ip-network-region 1		Page 1 of 19
IP NETWORK REGION		
Region: 1		
Location: 1	Authoritative Domain: localdomain.com	
Name: CLAN1		
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: 3329		
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): 20		
Keep-Alive Interval (sec): 5		
Keep-Alive Count: 5		

Figure 9: IP Network Region Form

¹ In this case, the SIP trunk is assigned to the same ip-network-region as the G650 media gateway. If multiple network regions are used, page 3 of each ip-network-region form must be used to specify the codec set for inter-region communications.

4.2.4 Define SIP Trunk Groups

Configure the **Signaling Group** form shown as follows, shown in **Figure 10a** and **Figure 10b**:

- 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 on a SIP trunk between Communication Manager and SES.
- Specify the Avaya G650 Media Gateway CLAN board (node name “clan-cabinet-a”) and the SES Server (node name “ses”) as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** fields, respectively. These field values are taken from the **IP Node Names** form shown in Figure 7².
- Ensure that the recommended TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- Enter the IP Network Region value assigned in the ip-network-region form (**Figure 9**). Note that 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. In this case, the same IP network region (Network Region 1) was used for local and Priority Netherlands network calls; however, different network regions can be used in the field.
- Leave the **Far-end-Domain** field blank. When an incoming SIP call is placed to the SES, SES will look for a trunk to route the call to Communication Manager. This is done by looking at the domain/IP address of the SIP URI in the “FROM”-header and then by looking for a match on one of the SIP signaling groups configured on Communication Manager. When the Far-end-Domain field of a signaling group is left blank, all calls with no valid match for the Domain part of the SIP URI, will use this signaling group. (Note: This domain is specified in the Uniform Resource Identifier (URI) of the “SIP To Address” in the INVITE message. Misconfiguring this field may prevent calls from being successfully established to other SIP endpoints or to the PSTN.)
- If calls to/from 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*. This value is used to have Communication Manager send DTMF transmissions using RFC 2833.
- The **Alternate Route Timer** field is set to *20* for establishing calls to international and mobile extensions.

² The near (local) end of the SIP signaling group may be a server “processor ethernet” instead of a C-LAN board.

```

change signaling-group 1

                                SIGNALING GROUP

Group Number: 1                  Group Type: sip
                                Transport Method: tls

IMS Enabled? n
IP Video? n


Near-end Node Name: clan-cabinet-a    Far-end Node Name: ses
Near-end Listen Port: 5061            Far-end Listen Port: 5061
Far-end Network Region: 1

Far-end Domain:

                                Bypass If IP Threshold Exceeded? n

DTMF over IP: rtp-payload          Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3    IP Audio Hairpinning? n
Enable Layer 3 Test? n                Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n Alternate Route Timer(sec): 6

```

Figure 10a: Signaling Group Form SIP endpoints and Incoming Calls from Priority SIP trunk

```

change signaling-group 40

                                SIGNALING GROUP

Group Number: 40                  Group Type: sip
                                Transport Method: tls

IMS Enabled? n
IP Video? n


Near-end Node Name: clan-cabinet-a    Far-end Node Name: ses
Near-end Listen Port: 5061            Far-end Listen Port: 5061
Far-end Network Region: 1

Far-end Domain:

                                Bypass If IP Threshold Exceeded? n

DTMF over IP: rtp-payload          Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3    IP Audio Hairpinning? n
Enable Layer 3 Test? n                Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n Alternate Route Timer(sec): 20

```

Figure 10b: Signaling Group Form Outgoing Calls to Priority SIP trunk

Configure the **Trunk Group** form as shown in **Figure 11a** and **Figure 11b** using the **add trunk-group** command. **Trunk Group** number 1 is defined for SIP endpoints, Incoming Calls from Priority SIP trunking service and **Trunk Group** number 40 is defined for using Priority Netherlands SIP trunking service in this configuration.

On Page 1 of the **Trunk Group** form:

- Set the **Group Type** field to **sip**.
- Choose a descriptive **Group Name** such as **SES Trunk** and **SIP_Priority**.
- Specify an available trunk access code (**TAC**) such as ***01** and ***09**.
- Set the **Service Type** field to **public-ntwrk** in Trunk Group 1 and 40.
- Enter **1** and **40** as the **Signaling Group** numbers. These values were previously determined during the **Signaling Group** configuration specified in **Figure 10a** and **Figure 10b**.
- Specify the **Number of Members** supported by each SIP trunk group.
Note that one trunk member is required for each call between a non-SIP endpoint and the Priority Netherlands SIP trunking service. Calls involving a SIP endpoint and the Priority Netherlands SIP trunking service will use two trunk members for the duration of the call, one from trunk group 1 and one from trunk group 40.
- Set the **Direction** field in trunk group 40 to **outgoing**. Trunk group 40 is used for outgoing calls to Priority Netherlands SIP trunking service.

```

add trunk-group 1                                     Page 1 of 21
                                     TRUNK GROUP
Group Number: 1                                     Group Type: sip          CDR Reports: y
  Group Name: SES Trunk                             COR: 1          TN: 1          TAC: *01
    Direction: two-way                               Outgoing Display? n
    Dial Access? n                                   Night Service:
    Queue Length: 0
  Service Type: public-ntwrk                         Auth Code? n
                                                    Signaling Group: 1
                                                    Number of Members: 30
  
```

Figure 11a: Trunk Group Form – Page 1 SIP endpoints and Incoming Calls from Priority SIP trunk

```

add trunk-group 40                                     Page 1 of 21
                                     TRUNK GROUP
Group Number: 40                                     Group Type: sip          CDR Reports: y
  Group Name: SIP_Priority                           COR: 1          TN: 1          TAC: *09
    Direction: outgoing                               Outgoing Display? n
    Dial Access? n                                   Night Service:
    Queue Length: 0
  Service Type: public-ntwrk                         Auth Code? n
                                                    Signaling Group: 40
                                                    Number of Members: 30
  
```

Figure 11b: Trunk Group Form – Page 1 outgoing calls to Priority SIP trunk

On Page 3 of the **Trunk Group** form (**Figure 12a** and **Figure 12b**):

- 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 3 of 21
TRUNK FEATURES
  ACA Assignment? n      Measured: none
                          Maintenance Tests? y

  Numbering Format: public
                          UI Treatment: service-provider

                          Replace Restricted Numbers? n
                          Replace Unavailable Numbers? n

Show ANSWERED BY on Display? y
```

Figure 12a: Trunk Group Form – Page 3 SIP endpoints and Incoming Calls from Priority SIP trunk

```
add trunk-group 40                                     Page 3 of 21
TRUNK FEATURES
  ACA Assignment? n      Measured: none
                          Maintenance Tests? y

  Numbering Format: public
                          UI Treatment: service-provider

                          Replace Restricted Numbers? n
                          Replace Unavailable Numbers? n

Show ANSWERED BY on Display? y
```

Figure 12b: Trunk Group Form – Page 3 Outgoing Calls to Priority SIP trunk

On Page 4 of the **Trunk Group** form (**Figure 13a** and **Figure 13b**):

- Set the **Send Diversion Header** field to **y**. This field allows the Diversion header to be included in the SIP INVITE. For a forwarded call, it contains the number of the call forwarding station.
- Set the **Support Request History** field to **y**. This field specifies the call forwarding history.
- Set the **Telephone Event Payload Type** field to **101**. This field specifies the payload for DTMF signaling (RFC2833).
- Set **Overwrite Calling Identity** (field appears, if SA8972 Overwrite Calling Identity is enabled) field to **y**. With this field enabled, the outgoing INVITE for a forwarded call (CFU, CFB, CFNA) contains the number of the forwarding station in the FROM header and P-Asserted-Identity instead of the number of the calling station.

add trunk-group 1	Page 4 of 21
PROTOCOL VARIATIONS	
Mark Users as Phone? n	
Prepend '+' to Calling Number? n	
Send Transferring Party Information? n	
Network Call Redirection? n	
Send Diversion Header? y	
Support Request History? y	
Telephone Event Payload Type: 101	
Overwrite Calling Identity? n	

Figure 13a: Trunk Group Form – Page 4 SIP endpoints and Incoming Calls from Priority SIP trunk

add trunk-group 40	Page 4 of 21
PROTOCOL VARIATIONS	
Mark Users as Phone? n	
Prepend '+' to Calling Number? n	
Send Transferring Party Information? n	
Network Call Redirection? n	
Send Diversion Header? y	
Support Request History? y	
Telephone Event Payload Type: 101	
Overwrite Calling Identity? y	

Figure 13b: Trunk Group Form – Page 4 outgoing calls to Priority SIP trunk

4.2.5 Configure Calling Party Number Information for Outgoing Calls

Configure the **public-unknown-numbering** form to send the correct calling party number to the Priority Netherlands SIP trunking service for calls originating from Communication Manager. The calling party number (sent in the SIP “From” header) must match the value expected by the Priority Netherlands SIP trunking service. If the calling party number is not matched, the call will be rejected.

In these Application Notes, the extensions on Communication Manager use a 4 digit dialing plan with extensions between 6000 and 6059 that match the DID numbers assigned by the Priority Netherlands SIP trunking service. Note this requirement may vary with other numbering plans provided by Priority Netherlands.

Figure 14 shows the use of the **change public-unknown-numbering** command to send the correct calling party number. The entries below indicate that 4-digit extension numbers beginning with 6 will send the corresponding digits via trunk group 40.

change public-unknown-numbering 0					Page	1	of	2
NUMBERING - PUBLIC/UNKNOWN FORMAT								
Total					Total			
Ext	Ext	Trk	CPN	CPN				
Len	Code	Grp(s)	Prefix	Len				
4	6	40	3110340	11	Total Administered: 7			
					Maximum Entries: 9999			

Figure 14: Public/Unknown Numbering Form

4.2.6 Configure Call Routing

4.2.6.1 Outbound Calls

In these Application Notes, the Automatic Route Selection feature is used to route calls via the SIP trunk to the Priority Netherlands SIP trunking service, which in turn completes the calls to the Priority Netherlands network destinations.

Use the **change dialplan analysis** command to add **0** as a feature access code (**fac**).

change dialplan analysis						Page 1 of 12		
DIAL PLAN ANALYSIS TABLE								
						Percent Full: 0		
Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type
0	1	fac						
1	1	fac						
2	3	ext						
3	4	ext						
4	3	ext						
6	4	ext						
7	3	ext						
*	3	dac						
*5	3	fac						
*6	3	fac						
#	3	dac						
#5	3	fac						
#6	3	fac						

Figure 15: Dialplan Analysis Form

Use the **change feature-access-codes** command to specify **0** as the access code for external dialing (ARS ACCESS CODE).

change feature-access-codes		Page 1 of 8	
FEATURE ACCESS CODE (FAC)			
Abbreviated Dialing List1 Access Code:			
Abbreviated Dialing List2 Access Code:			
Abbreviated Dialing List3 Access Code:			
Abbreviated Dial - Prgm Group List Access Code:			
Announcement Access Code:			
Answer Back Access Code:			
Attendant Access Code:			
Auto Alternate Routing (AAR) Access Code:			
Auto Route Selection (ARS) - Access Code 1: 0		Access Code 2: 1	
Automatic Callback Activation:		Deactivation:	
Call Forwarding Activation Busy/DA: All: *10		Deactivation: #10	
Call Forwarding Enhanced Status: Act:		Deactivation:	
Call Park Access Code:			
Call Pickup Access Code:			
CAS Remote Hold/Answer Hold-Unhold Access Code:			
CDR Account Code Access Code:			
Change COR Access Code:			
Change Coverage Access Code:			
Conditional Call Extend Activation:		Deactivation:	
Contact Closure Open Code:		Close Code:	

Figure 16: Dialplan Analysis Form

Now use the **change ars analysis** command to configure the route pattern selection rule based upon the number dialed following the dialed digit “0”. Note that further administration of ARS is beyond the scope of these Application Notes but is discussed in References [1] and [2].

change ars analysis 0							Page 1 of 2	
ARS DIGIT ANALYSIS REPORT								
Location: all					Percent Full: 0			
Dialed	Total		Route	Call	Node	ANI		
String	Min	Max	Pattern	Type	Num	Reqd		
00	7	18	40	pubu		n		
06	5	11	41	pubu		n		
08	5	12	41	pubu		n		

Figure 17: ARS Analysis Form

Use the **change route-pattern** command to define the SIP trunk groups included in the route pattern that ARS selects

- Set the **Inserted Digits** field to **p** for international calls (**Figure 18a**). The Priority Netherlands SIP trunking service need for an outbound call the called number in the TO header in the canonical address format.
- Set the **Inserted Digits** field to **p31** for national calls (**Figure 18b**). The Priority Netherlands SIP trunking service need for an outbound call the called number in the TO header in the canonical address format.

change route-pattern 40										Page 1 of 3	
					Pattern Number: 2		Pattern Name: int			Priority	
					SCCAN? n		Secure SIP? n				
Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted		DCS/	IXC	
No			Mrk	Lmt	List	Del	Digits		QSIG		
							Dgts		Intw		
1:	40	1				3	p		n	user	
2:									n	user	
3:									n	user	
4:									n	user	
5:									n	user	
6:									n	user	
BCC		VALUE		TSC	CA-TSC		ITC		BCIE	Service/Feature	PARM
0		1 2 M 4 W			Request					No. Numbering	LAR
									Dgts Format		
									Subaddress		
1:	y	y	y	y	y	n	rest			none	
2:	y	n	n	n	n	n	rest			none	
3:	y	n	n	n	n	n	rest			none	
4:	y	n	n	n	n	n	rest			none	
5:	y	y	y	y	y	n	rest			none	
6:	y	y	y	y	y	n	rest			none	

Figure 18a: Route Pattern Form - International calls

change route-pattern 41										Page	1 of 3	
Pattern Number: 2										Pattern Name: nat		Priority
SCCAN? n										Secure SIP? n		
Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted			DCS/	IXC	
No			Mrk	Lmt	List	Del	Digits			QSIG		
							Dgts			Intw		
1:	40	1					2	p31		n	user	
2:											n	user
3:											n	user
4:											n	user
5:											n	user
6:											n	user
BCC VALUE				TSC	CA-TSC	ITC BCIE Service/Feature				PARM	No. Numbering	LAR
0	1	2	M	4	W	Request				Dgts Format		
										Subaddress		
1:	y	y	y	y	y	n	rest					none
2:	y	n	n	n	n	n	rest					none
3:	y	n	n	n	n	n	rest					none
4:	y	n	n	n	n	n	rest					none
5:	y	y	y	y	y	n	rest					none
6:	y	y	y	y	y	n	rest					none

Figure 18b: Route Pattern Form - National calls

4.2.6.2 Incoming Calls

This step configures the mapping of incoming DID numbers to the proper extensions.

In this configuration, the Priority Netherlands SIP trunking service sends exactly 11 digits in the incoming INVITE message. The last four incoming digits received are matched to the intended extension. Therefore the incoming call handling treatment is used to modify the number in order to reach the desired extension. The example below illustrates the technique to assign the incoming DID numbers beginning with 31103406020 to the corresponding extension 6020.

Use the **change inc-call-handling-trmt trunk-group** command to administer this assignment. For this example shown in **Figure 19**:

- Enter **11** into the **Number Len** field to match the length of the incoming digits.
- Enter **3110304** into the **Number Digits** field as the digit pattern to be matched.
- Enter **7** into the **Del** field as the number of digits that should be deleted from the beginning of the incoming digits.

change inc-call-handling-trmt trunk-group 1										Page	1 of 30
INCOMING CALL HANDLING TREATMENT											
Service/	Number	Number	Del Insert								
Feature	Len	Digits									
public-ntwrk	11	3110340	7								

Figure 19: Incoming Call Handling Treatment – Simple Extension Mapping

4.2.7 Save Avaya Aura™ Communication Manager Changes

Enter the **save translation** command to make the changes permanent.

4.3. SIP Endpoint Configuration

This section describes the administration of optional SIP endpoints and requires the preceding SIP Trunk configuration been completed beforehand. SIP endpoints are optional, but not required, to use the Priority Netherlands SIP Trunk Service.

Step 1:

The first step in adding an off-PBX station (OPS) for Avaya SIP telephones registered with SES is to assign a station as shown in **Figure 20, 21 and 22**.

Using the **add station** command from SAT using the station's desired extension as parameter:

- Set the station **Type** to **9630SIP** value.
- The **Port** field will default to a system specified value.
- Enter a **Name** for the station that will be displayed.
- As **Security Code** was "123456" entered but normally it could be left blank for SIP OPS extensions.

The remaining fields are configured per normal station administration that is beyond the scope of these Application Notes. Note that the Class of Restrictions (**COR**) and Class of Service (**COS**) will govern the features and call restrictions that apply to this station.

add station 6050		Page 1 of 5
STATION		
Extension: 6050	Lock Messages? n	BCC: 0
Type: 9630SIP	Security Code: 123456	TN: 1
Port: S00005	Coverage Path 1:	COR: 1
Name: Prio SIP	Coverage Path 2:	COS: 1
	Hunt-to Station:	
STATION OPTIONS		
Loss Group: 19	Time of Day Lock Table:	
	Personalized Ringing Pattern: 1	
	Message Lamp Ext: 200	
Speakerphone: 2-way	Mute Button Enabled? y	
Display Language: english	Button Modules: 0	
Survivable GK Node Name:		
Survivable COR: internal	Media Complex Ext:	
Survivable Trunk Dest? y	IP SoftPhone? n	
	IP Video? n	
	Customizable Labels? y	

Figure 20: Add 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. The **Direct IP-IP Audio Connections?** field should be set to **y**. If there are shuffling problems for this station, set the **Direct IP-IP Audio Connections?** to **n**.

add station 6050		Page 2 of 5
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		
	EMU Login Allowed? n	
H.320 Conversion? n	Per Station CPN - Send Calling Number? y	
Service Link Mode: as-needed	EC500 State: disabled	
Multimedia Mode: enhanced	Audible Message Waiting? n	
MWI Served User Type:	Display Client Redirection? n	
AUDIX Name:	Select Last Used Appearance? n	
	Coverage After Forwarding? s	
	Direct IP-IP Audio Connections? y	
Emergency Location Ext: 200	Always Use? n IP Audio Hairpinning? n	

Figure 21: Add Station – Page 2

On Page 4 of the station form, configure the appropriate number of call appearances for the SIP telephone.

add station 6050		Page 4 of 5
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:	
voice-mail Number:		

Figure 22: Add Station– Page 3

Step 2:

The second step of configuring an off-PBX station is to configure the **Off-PBX Telephone** form so that calls destined for a SIP telephone at the enterprise site are routed to SES, which will then direct the call to the intended SIP telephone.

On the **Off-PBX-Telephone Station-Mapping** form shown in **Figure 23**:

- Specify the extension of the SIP endpoint, as shown in **Figure 20**.
- Set the **Application** field to **OPS**.
- Set the **Phone Number** field to the digits to be sent over the SIP trunk. In this case, the SIP telephone extensions configured for SES also match the extensions of the corresponding stations on Communication Manager
- Set the **Trunk Selection** field to **1**, the SIP trunk group number assigned in **Figure 11a**.

change off-pbx-telephone station-mapping 6050						Page 1 of 2
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION						
Station Extension	Application	Dial Prefix	CC	Phone Number	Trunk Selection	Configuration Set
6050	OPS	-		6050	1	1

Figure 23: 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 6050						Page 2 of 2
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION						
Station Extension	Appl Name	Call Limit	Mapping Mode	Calls Allowed	Bridged Calls	Location
6050	OPS	3	both	all	none	

Figure 24: Stations with Off-PBX Telephone Integration – Page 2

Step 3:

Repeat steps 1 and 2 for each SIP Telephone to be added.

Step 4:

Perform a **save translation** to make the changes permanent.

5. Configure Avaya SIP Enablement Services

This section covers the administration of SIP Enablement Services (SES). SES is configured via an Internet browser using **SIP Server Management** screens. It is assumed that SES software and the license file have already been installed. 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 including DNS server address(es). For additional information on these installation tasks, refer to Reference [4].

This section is divided into two parts: section 5.1 provides the steps necessary to configure SIP trunking to the Priority Netherlands SIP trunking service. Section 5.2 provides the steps necessary to complete the administration for optional SIP endpoints in this Avaya SIP telephony solution. SIP endpoints are not necessary to use the Priority Netherlands SIP trunking service.

5.1. SIP Trunking to the Priority Netherlands SIP Trunking Service

The “standard” Priority Netherlands SIP Trunking service uses UDP as the transport protocol.

Log in to Avaya SIP Enablement Services.

Access the SES **SIP Server Management** pages by entering **https://<ip-addr>/admin** as the URL in an Internet browser, where **<ip-addr>** is the IP address of the SES server.

Log in with the appropriate credentials and then select the **Administration --> SIP Enablement Services** link from the main page as shown in **Figure 25**.

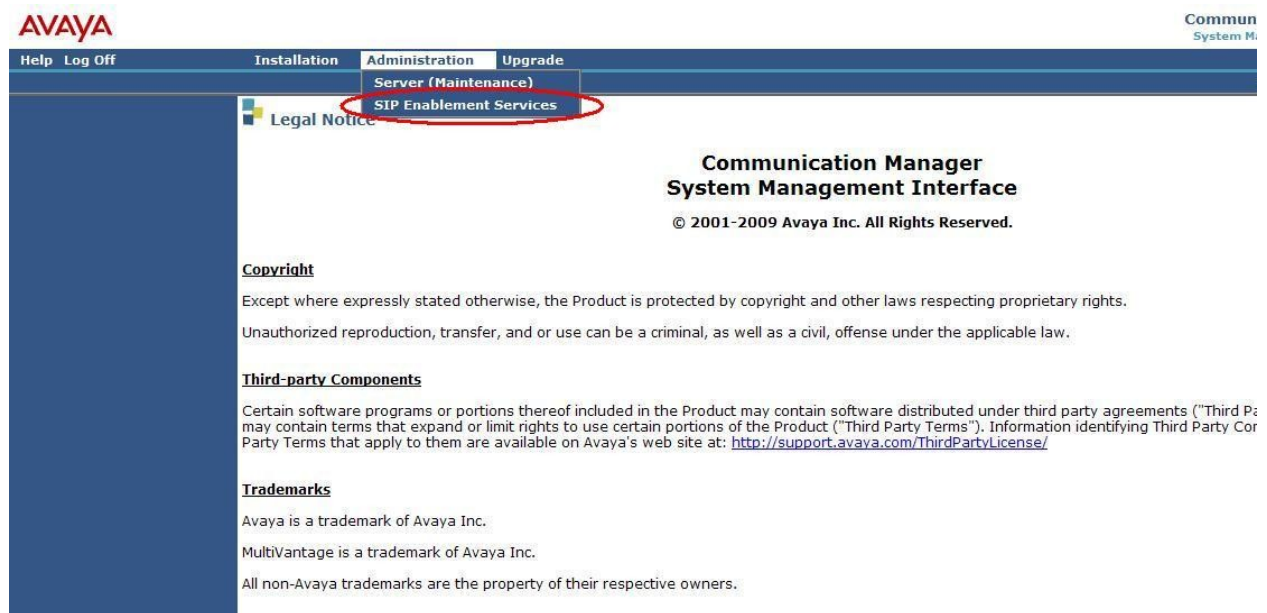


Figure 25: Avaya Aura™ SES Main Page

The SES administration home page shown in **Figure 26** is displayed.

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Manage Address Map Priorities	Adjust Address Map Priorities.
Manage Adjunct Systems	Add and delete Adjunct Systems.
Manage Event Aggregators	Add/Delete Event Aggregators.
Certificate Management	Manage Certificates.
Manage Conferencing	Add and delete Conference Extensions.
Manage Emergency Contacts	Add and delete Emergency Contacts.
Export Import to ProVision	Export and import data using ProVision on this host.
Manage Hosts	Add and delete Hosts.
IM logs	Download IM Logs.
Manage Communication Manager Servers	Add and delete Communication Manager Servers.
Manage Communication Manager Extensions	Add and delete Communication Manager Extensions.
Server Configuration	View Properties of the system.
Manage SIP Phone Settings	Add/Delete Phone Settings
Manage Survivable Call Processors	Add and delete Survivable Call Processors.
System Status	View System Status.
Trace Logger	Manage SIP Trace Logs.
Manage Trusted Hosts	Add and delete Trusted Hosts.

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Figure 26: SES Administration Home Page

5.1.1 Verify System Properties

From the left pane of any **SIP Server Management** page, expand the **Server Configuration** option and select **System Properties**. This page displays the **SES Version** and the **Network Properties** entered via the install script during the installation process.

In the **Edit System Properties** page note the **SIP Domain** entered. The **SIP Domain** **localdomain.com** is used in these Application Notes (see **Figure 9**).

The screenshot displays the Avaya SIP Server Management interface. The left sidebar contains a navigation menu with the following items: Top, Users, Address Map Priorities, Adjunct Systems, Aggregator, Certificate Management, Conferences, Emergency Contacts, Export/Import to ProVision, Hosts, IM logs, Communication Manager Servers, Communication Manager Extensions, Server Configuration (expanded), Admin Setup, IM Log Settings, License, SNMP Configuration, System Properties, SIP Phone Settings, Survivable Call Processors, System Status, Trace Logger, and Trusted Hosts. The main content area is titled 'View System Properties' and contains the following information:

SES Version	SES-5.2.0.0-947.3b
System Configuration	Simplex
Host Type	SES combined home-edge
SIP Domain*	<input type="text" value="localdomain.com"/>

Note that the DNS domain is olymp.org

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

SIP License Host*	<input type="text" value="192.168.193.110"/>
-------------------	--

DiffServ/TOS Parameters

Call Control PHB Value*	<input type="text" value="46"/>
-------------------------	---------------------------------

802.1 Parameters

Priority Value*	<input type="text" value="6"/>
Management System Access Login	<input type="text" value="craft"/>
Management System Access Password	<input type="password" value="....."/>
DB Log Level	<input type="text" value="disabled"/>

Figure 27: System Properties

5.1.2 Verify the Avaya SES Host Information

Verify the SES Host information using the **Edit Host** page. In these Application Notes the SES **Host Type** is a combined **home/edge**. This means that both the Priority Netherlands SIP trunking service and Communication Manager are contacting the same SES.

Display the **Edit Host** page (**Figure 28**) by following the **Hosts** link in the left navigation pane and then clicking on the **Edit** option under the **Commands** section of the **List Hosts** screen.

On the **Edit Host** screen:

- Verify that the IP address of this combined SES Home/Edge server is in the **Host IP Address** field.
- Do not modify the **DB Password** or **Profile Service Password** fields. If these fields are changed, exit the form without using the **Update** button. These values must match the values entered during the SES installation; incorrect changes may disable the SES.
- Verify that the **UDP** and **TLS** checkboxes are enabled as **Listen Protocols**.
- Verify that **TLS** is selected as the **Link Protocol**.
- Ensure that the Priority Netherlands trunking service SIP Application Server (e.g., *Priority-sip.net*) is not in the **Outbound Proxy** or **Outbound Direct Domains** fields.
- Default values for the remaining fields may be used.
- After completing the **Edit Host** screen, click on the **Update** button.

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Edit Host

Host IP Address* 192.168.193.110

Profile Service Password*

Host Type SES combined home-edge

Parent none

Listen Protocols ☒ UDP ☒ TCP ☒ TLS

Link Protocols ☐ UDP ☐ TCP ☒ TLS

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

Emergency Contacts Policy ☒ Allow ☐ Deny

Minimum Registration (seconds) 900 Registration Expiration Timer (seconds)* 86400

Subscription 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 28: Edit Host

5.1.3 Add Communication Manager Server Interfaces

Under the **Communication Manager Servers** option in the Administration web interface, select **Add** to add the Communication Manager Server in the enterprise site. This will create the SES side of the SIP trunk previously created in Communication Manager.

In the **Add Communication Manager Server Interface** screen, enter the following information:

- A descriptive name in the **Communication Manager Server Interface** field (e.g., **192.168.193.100**).
- Select the SES home/edge IP address 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 between SES and Communication Manager.
- Enter the IP address of the C-LAN board in the **SIP Trunk IP Address** field.³
- After completing the **Add Communication Manager Server** screen, click on the **Update** button.

The screenshot shows the Avaya Integrated Management SIP Server Manager web interface. The top header includes the Avaya logo and the text 'Integrated Management SIP Server Manager This Server'. A navigation menu on the left lists various system components. The main content area is titled 'Edit Communication Manager Server Interface' and contains the following fields and options:

- Communication Manager Server Interface Name***: Text input field containing '192.168.193.100'.
- Host**: Text input field containing '192.168.193.110'.
- SIP Trunk** section:
 - SIP Trunk Link Type**: Radio buttons for TCP and TLS (selected).
 - SIP Trunk IP Address***: Text input field containing '192.168.193.104'.
- Communication Manager Server** section:
 - Communication Manager Server Admin Address*** (see Help): Text input field containing '192.168.193.100'.
 - Communication Manager Server Admin Port***: Text input field containing '5022'.
 - Communication Manager Server Admin Login***: Text input field containing 'init'.
 - Communication Manager Server Admin Password***: Password input field with masked characters.
 - Communication Manager Server Admin Password Confirm***: Password input field with masked characters.
- SMS Connection Type**: Radio buttons for SSH (selected), Telnet, and Not Available.

A note at the bottom states: 'Note: If the Communication Manager Server connection type is changed and the admin port value is not also changed, changing connection type to SSH will change the admin port to 5022 when Add or Update is clicked and connection type to Telnet will change admin port to 5023 when Add or Update is clicked.' Below the fields, it says 'Fields marked * are required.' and there is an 'Update' button.

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Figure 29: Add Communication Manager Server Interface

³ Depending on the platform of the server, this field may be set to the IP address of a C-LAN board or a server's "processor ethernet".

5.1.4 Configure Call Routing

SES functions as a SIP proxy server for the SIP trunking with the Priority Netherlands SIP trunking service. In this role, for outbound calls the SES must direct SIP messages originating from Communication Manager to the Priority Netherlands SIP Application Server. In a similar manner for incoming DID calls, the SES must route messages received from the Priority Netherlands SIP Application Server to the proper signaling interface on Communication Manager.

In these Application Notes, the SIP message routing will be done for both outbound and inbound calls using Address Maps that examine some or all of the *called number* (using a Pattern) and route to a specific predetermined destination (called a Contact).

The *called number* is contained within the *user* part of the Uniform Resource Identifier (URI) of an incoming SIP INVITE message. The URI usually takes the form of *sip:user@domain*, where *domain* can be a fully qualified domain name or an IP address. The *user* part for SIP trunking in these Application Notes will only contain digits⁴.

The Address Map Patterns are specified using Linux regular expression syntax. Patterns are generally designed to match a collection of *called numbers* that require identical SIP message routing. However, each Pattern must also be specific enough to direct each unique *called number* to the proper signaling Contact. The Address Map Patterns must also be mutually exclusive (non-overlapping) from all other Address Map Patterns used in the SES to ensure proper operation.

Appendix B provides a detailed description of the Linux regular expression syntax used within the address map patterns.

5.1.4.1 Outbound PSTN Calls

SIP signaling for outbound calls is directed to SES based upon rules configured for Automatic Route Selection in Communication Manager. The choice of trunk group (made within ARS) determines the codec that will be used for voice (or fax or modem) calls. For outbound calls, no further differentiation of voice vs. fax / modem is required within the SES address maps.

Outbound calls require SIP signaling messages to be routed to the Priority Netherlands SIP trunking SIP Application Server using Host Address Maps within SES. Calls matching Host Address Map Patterns will be directed to the Priority Netherlands SIP trunking SIP Application Server contained within the corresponding Contact information.

⁴ SIP does permit mnemonic addressing such as “sip:john.doe@customer.com”. However, this convention is not used in these Application Notes for SIP Trunking. Further discussion of this topic is beyond the scope of this document.

More specific pattern matches would be used if necessary to selectively route SIP traffic to different destinations (such as multiple service providers serving different geographic regions). Also note that a user dialed access code (such as “9” or “0” to place a PSTN or ISDN call) has been previously deleted (by ARS) prior to seizing the outbound SIP trunk.

5.1.4.1.1 Outbound Routing - Host Maps

Begin the outbound routing configuration by navigating to the **Add Host Address Map** pages.

- From any **SIP Server Management** page, expand the **Hosts** link and choose the **List** link causing the **List Hosts** page to appear.
- Select the **Map** link on the **List Host** page causing the **List Host Address Map** page to appear (**Figure 31**).

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List Hosts

Showing 1 to 1 of 1 Hosts

Commands		Host	Type	SES Version			
Edit	Map	Go-To	Test-Link	Delete	192.168.193.110	SES combined home-edge	SES-5.2.0.0-947.3b

Migrate Home/Edge

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Figure 30: Accessing Host Address Maps from List Hosts Screen

- Select the **Add Map In New Group** link or **Edit** link on the **List Host Address Map** page (Figure 31). This will display the **Add (or Edit) Host Address Map** screen (Figure 32).

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 - Add
 - List

List Host Address Map

Host 192.168.193.110

Commands	Name	Commands	Contact
Edit Delete	out_0800_sip		
Edit Delete	out_Ffm		
Edit Delete	out_avaya_sip		
Edit Delete	out_schwenk		
		Edit Delete	sip:\${user}@212.144.52.8:5060;transport=udp
Add Another Map		Add Another Contact	Delete Group
Edit Delete	out_sp_ger		
Edit Delete	out_spain_19		
Edit Delete	out_spain_81		
Edit Delete	out_spain_90		
Edit Delete	out_spain_91		
Edit Delete	out_spain_92		
		Edit Delete	sip:\${user}@62.7.40.48:5060;transport=udp
Add Another Map		Add Another Contact	Delete Group
Edit Delete	Priority_NL		
		Edit Delete	sip:\${user}@213.160.252.154:5060;transport=udp
Add Another Map		Add Another Contact	Delete Group
Add Map In New Group			

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Figure 31: Adding a Host Address Map Group

The configuration of the **Host Address Map** for a call to a PSTN number in Priority Netherlands network is shown in **Figure 32**.

- Enter a descriptive **Name** for the map, such as **Priority_NL**.
- Enter the appropriate pattern for the call type. In this example, the pattern is used for calls into Priority Netherlands network: **^sip:[+]**.
- Select the **Replace URI** checkbox.
- Click the **Update** button.

The screenshot shows the Avaya Management System interface. At the top is the Avaya logo. Below it is a navigation sidebar with a 'Top' link and a list of categories: Users, Address Map Priorities, Adjunct Systems, Aggregator, Certificate Management, Conferences, Emergency Contacts, Export/Import to ProVision, Hosts (with sub-items: List, Migrate Home/Edge, IM logs), Communication Manager Servers, Communication Manager Extensions, Server Configuration, SIP Phone Settings, Survivable Call Processors, System Status, Trace Logger, and Trusted Hosts. The main content area is titled 'Edit Host Map Entry' and contains the following form fields: 'Name*' with the value 'Priority_NL', 'Pattern*' with the value '^sip:[+]', and 'Replace URI' with a checked checkbox. Below these fields is a note: 'Fields marked * are required.' and an 'Update' button. At the bottom right of the interface, there is a copyright notice: '© 2006 Avaya Inc. All Rights Reserved.'

Figure 32: Address Map for a PSTN number in Priority network

Note that in these Application Notes some other Maps were used, that are not explained here in detail.

5.1.4.1.2 Outbound Routing–Priority SIP Trunking Service Gateway

The next step is to enter the **Contact** information for the Priority Netherlands SIP trunking SIP Application Server. For these Application Notes, the test environment IP address 213.160.252.154 was used. The correct IP Address will be provided by your Priority representative to ensure the solution has correct routing and redundancy options.

To enter the Priority Netherlands SIP trunking SIP Application Server information:

- Access the **Host Address Map** page by expanding the **Hosts** link in the left pane of any **SIP Server Management** page, selecting **List** and then clicking on the **Map** link associated with the appropriate **Host**.

The **List Host Address Map** page is displayed as shown in **Figure 33**. Notice that the **Contact** information is blank when first creating this entry.

Note: Should an entry already exist due to prior administration, the entry should be edited or deleted instead of using **Add Another Contact**.

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List Host Address Map

Host 192.168.193.110

Commands	Name	Commands	Contact
Edit Delete	out_0800_sip		
Edit Delete	out_Ffm		
Edit Delete	out_avaya_sip		
Edit Delete	out_schwenk		
		Edit Delete	sip:\$(user)@212.144.52.8:5060;transport=udp
Add Another Map		Add Another Contact	Delete Group
Edit Delete	out_sp_ger		
Edit Delete	out_spain_19		
Edit Delete	out_spain_81		
Edit Delete	out_spain_90		
Edit Delete	out_spain_91		
Edit Delete	out_spain_92		
		Edit Delete	sip:\$(user)@62.7.40.48:5060;transport=udp
Add Another Map		Add Another Contact	Delete Group
Edit Delete	Priority_NL		
		Edit Delete	sip:\$(user)@213.160.252.154:5060;transport=udp
Add Another Map		Add Another Contact	Delete Group
Add Map In New Group			

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Figure 33: List Host Address Map

- Click on the **Add Another Contact / Edit Host Contact** link to open the **Add Host Contact / Edit Host Contact** page (**Figure 34**).
- In the **Add/Edit Host Contact** page, the **Contact** field specifies the destination for the call. In these Application Notes the Contact field is entered as:

sip:\$(user)@213.160.252.154;transport=udp

Note: the IP address is assigned and provided by Priority Netherlands.

- Click the **Submit** button when completed.



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Edit Host Contact

Host

192.168.193.110

Contact

sip:\$(user)@213.160.252.154:5060;transpo

Fields marked * are required.

Submit

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Figure 34: Edit (/Add) Host Contact Entry

5.1.4.2 Inbound Direct Inward Dialed Calls

SIP messages for incoming calls from the Priority Netherlands SIP trunking service are sent to the SES server. SES then routes these messages to the appropriate Communication Manager **Signaling Group** using an **SES Communication Manager Server Address Map**.

In these Application Notes, the incoming calls from the Priority Netherlands SIP trunking service use **Communication Manager Server Address Map Patterns** matching the 11-digit *called number* in the *user* part of the SIP URI.

An example of a SIP URI in an INVITE message received from the Priority Netherlands SIP trunking service for the number **31103406050** is:

sip:31103406050@192.168.193.110

The *user* part in this case is the 11-digit number “**31103406050**”.

To configure the **Communication Manager Server Address Map** for voice calls:

- Expand the **Communication Manager Servers** link in the left navigation menu of any **SIP Server Management** page. Select **List** to display the **List Communication Manager Servers** page as shown in **Figure 35**.
- Click on the **Map** link of the **Commands** to display the **List Communication Manager Server Address Map** (shown in **Figure 36**) screen associated with this signaling interface.

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List Communication Manager Servers

Commands					Interface	Host
Edit	Extensions	Map	Test-Link	Delete	192.168.193.100	192.168.193.110
Add Another Communication Manager Server Interface						

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Figure 35: List Communication Manager Servers

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List Communication Manager Server Address Map

Commands	Name	Commands	Contact
Edit Delete	inc_arcor		
Edit Delete	inc_bt_spain		
Edit Delete	inc_priority_1		
		Edit Delete	sip:\${user}@192.168.193.104:5061;transport=tls
Add Another Map		Add Another Contact Delete Group	
Add Map In New Group			

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Figure 36: List Communication Manager Server Address Map

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- Click on the **Add Map In New Group** (/Edit in the Command column) link. The page shown in **Figure 37** is displayed.
- Enter a descriptive name in the **Name** field, such as **inc_priority_1**.
- Enter the **Address Map Pattern** for incoming calls into the **Pattern** field.
In this case, the numbers provided by Priority Netherlands SIP trunking for voice extensions are 31103406000 thru 31103406059. The pattern specification for voice DID numbers is:
^sip:3110340[0-9]

This means that URIs beginning with “sip:3110340” followed by the single digit 0, 1, 2, 3, 4, 5, 6, 7, 8 or 9 will match the pattern and be routed to the CLAN controlled by the S8730 Server.

- Click the **Update** button once the form is completed.

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Edit Communication Manager Map Entry

Name*

Pattern*

Fields marked * are required.

Update

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Figure 37: Incoming Calls - Communication Manager Server Address Map

After configuring the server address map, the **List Communication Manager Server Address Map** page appears as shown in **Figure 38**.

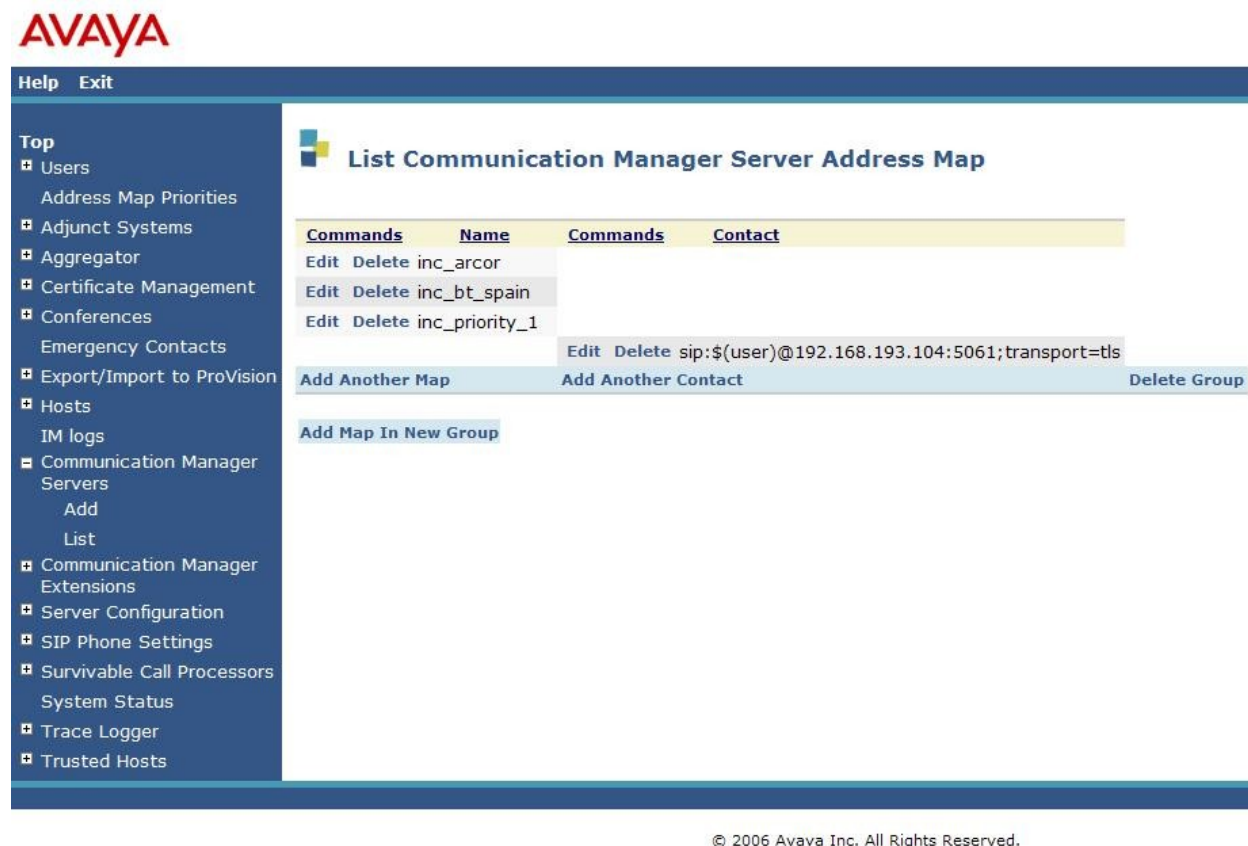


Figure 38: List Communication Manager Server Address Map

Note that after the first **Communication Manager Server Address Map** is created, a corresponding **Contact** entry is created automatically.

sip:\$(user)@192.168.193.104:5061;transport=tls

This **Contact** entry contains the IP address of a CLAN to contact the Avaya S8730 Server, the port (5061) and the transport protocol (TLS) to be used to send SIP signaling messages. The incoming digits sent in the *user* part of the original request URI will replace the \$(user) string when the message is sent to the **Contact**.

5.1.5 Specify the Priority Netherlands SIP trunking SIP Application Server as a Trusted Host

The final step to complete the SIP trunk administration on SES is to designate the IP address of the Priority Netherlands SIP Trunk Service as a trusted host. As a trusted host, SES will not issue SIP authentication challenges for incoming requests from the designated IP address.⁵

If multiple IP addresses are used to access the Priority Netherlands SIP Trunk Service, each IP Address must be added as a trusted host.

To configure a trusted host:

- Expand the **Trusted Hosts** link in the left navigation menu of any **SIP Server Management** page. Select **List** to display the **List Trusted Hosts** page as shown in **Figure 39**.
- Click on the **Add Another Trusted Host** link. The page shown in **Figure 40** is displayed. The **Host** field displays the ip address of the SIP server
- Enter the IP address of the Priority Netherlands SIP Trunk Service in the **IP Address** field (**Figure 40**).
- Enter a descriptive name in the **Comment** field such as **Priority NL**.
- Click the **Update** button once the form is completed.

⁵ Note, if the trusted host step is not done, authentication challenges to incoming SIP messages (such as INVITEs and BYEs) will be issued but not responded to. This may cause call setup to fail, active calls be disconnected after timeout periods, and/or SIP protocol errors.

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List Trusted Hosts

Commands		IP Address	Trusted by Host	Comment	Perform Origination Processing
Edit	Delete	192.168.193.100	192.168.193.110	CM	<input type="checkbox"/>
Edit	Delete	212.144.52.8	192.168.193.110	SIP Arcor	<input type="checkbox"/>
Edit	Delete	213.160.252.154	192.168.193.110	Priority NL	<input type="checkbox"/>
Edit	Delete	62.7.40.48	192.168.193.110	SIP BT-Spain	<input type="checkbox"/>

[Add Another Trusted Host](#)

Figure 39: List Trusted Hosts Map

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Edit Trusted Host

IP Address*:

213.160.252.154

Host:

192.168.193.110

Comment:

Priority NL

Perform Origination Processing:

☐

Fields marked * are required.

Update

Figure 40: Edit Trusted Hosts Map

5.2. Configuration for SIP Telephones

This section provides basic instructions for completing the administration to use Avaya 9600 Series SIP telephones with the described configuration. Additional features, such as the use of mnemonic addressing, are beyond the scope of these Application Notes.

As noted previously, SIP telephones are optional; it is not necessary to have SIP telephones to use SIP trunking to the Priority Netherlands SIP trunking service.

The steps below are repeated for each SIP telephone provisioned.

5.2.1 Add a SIP User

Add a SIP User to the SES as follows:

- In **SES SIP Server Management**, expand the **Users** link in the left side blue navigation bar and click on the **Add** link.
- In the **Add User** page (**Figure 41**) enter the extension number for the SIP telephone in the **Primary Handle** field.
- Enter a user password in the **Password** and **Confirm Password** fields. This password will be used when logging into the user's SIP telephone.
- In the **Host** field, select the SES server hosting the domain for this user.
- Enter the **First Name** and **Last Name** of the user.
- Select the **Add Communication Manager Server Extension** checkbox. This associates a Communication Manager extension with this SIP User. Calls from this user will be provided features and routing via Communication Manager.
- Click the **Add** button. This will cause a confirmation screen to appear.
- Click **Continue** on the confirmation screen.

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Add User

Primary Handle*

6050

User ID

6050

Password*

••••••••

Confirm Password*

••••••••

Host*

192.168.193.110

First Name*

Prio

Last Name*

SIP

Address 1

Address 2

Office

City

State

Country

Zip

Survivable Call Processor

none

Add Communication Manager Extension

☒

Fields marked * are required.

Add

Figure 41: Add User

5.2.2 Specify Corresponding Communication Manager Extension

The SIP phone handle must now be associated with the corresponding extension on Communication Manager.

If the **Add Communication Manager Server Extension** checkbox was not selected, complete this step:

- In the **Add Communication Manager Extension** page, enter the extension number (e.g., **6050**) corresponding to the extension previously configured on Communication Manager **Figure 20** in section 4.3. The **Communication Manager Extension** and the SIP Primary Handle are usually the same, but are not required to be.
- Select the **Communication Manager Server** interface assigned to this extension.
- Click the **Add** button.

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Add Communication Manager Extension

Extension

Communication Manager Server

Fields marked * are required.

Add

Figure 42: Add Communication Manager Server Extension

6. Priority Netherlands SIP Trunking Service Configuration

In order to use Priority Netherlands SIP trunking service, a customer must order service from Priority Netherlands SIP trunking using their sales processes. The process can be started by contacting a Priority Netherlands SIP trunking sales representative.

Priority Netherlands SIP trunking provided the information contained in **Table 2** to complete the configuration in these Application Notes. This information was necessary to complete the Avaya Communication Manager and Avaya SIP Enablement Services administration discussed in the previous sections.

Required Information	Values Used In These Application Notes
Priority Netherlands SIP trunking service SIP Application Server	213.160.252.154
Assigned DID Numbers	31103406000 to 31103406059
Codecs Supported	G.729, G.711A, T38

Table 2: Priority Netherlands SIP Trunking Service Configuration Information

7. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between the Priority Netherlands SIP trunking standard service and an Avaya SIP telephony solution using SIP trunking. This section covers the general test approach and the test results.

7.1. General Test Approach

A simulated enterprise site consisting of an Avaya SIP telephony solution supporting SIP trunking was connected to a laboratory version of the Priority Netherlands SIP trunking service via simulated managed access facilities. The enterprise site was configured as if using the generally available service provided by Priority Netherlands.

The following features and functionality were covered during the SIP trunking interoperability compliance test. All calls involved various Avaya telephones, the Priority Netherlands SIP trunking service and various ISDN telephones.

- Outgoing calls to ISDN PSTN telephones
- Incoming calls to Priority Netherlands SIP trunking provided DID numbers from PSTN telephones
- Calls using Avaya 9600 Series IP Telephones with the H.323 firmware configurations
- Calls using Avaya 9600 Series IP Telephones with the SIP firmware configurations
- G.711A and G.729 codecs for voice calls
- G.711A, G.729 and T38 codecs for fax calling
- DTMF tone transmission using RFC 2833 with successful Voice Mail navigation
- Telephone features such as hold, transfer and conference
- Direct IP-to-IP media (also known as “shuffling”)
- Call forwarding to internal and external destinations
- Extension to Cellular (EC500)
- Calling Party Number and Called Party Number presentation and restriction for incoming and outgoing calls
- Simultaneous Calls
- Long Calls

7.2. Test Results

Interoperability testing of the sample configuration was completed with successful results.

The following issues described in **Table 3** were observed.

Item	Issue Observed	Discussion / Workaround
Keep alive	Priority does not send OPTIONS for keep alive	
Outgoing calls to ISDN or mobile numbers	Time between INVITE and SESSION PROGRESS is 14 seconds	Set the field <i>Alternate Route Timer (sec)</i> in the Signaling Group to 20
Outgoing calls to international or mobile numbers	Priority may not send a “RINGING” message	This is the case with most international and mobile calls but the enterprise might see a 180 Ringing for a call to a fixed PSTN number in the Netherlands. This depends on the way the call is presented. If there is an inband media path from the ISDN side, Priority will send a 183 Session Progress with SDP and if this is not the case, Priority will send a 180 ringing without SDP.
Outbound call to unassigned number	no announcement is heard (only busy tone)	
Incoming Calling Number Restriction	“Anonymous” in PAI instead of calling number	Incoming Calling Number Restriction is ok.
Call forwarding off Net	Default Number of SIP trunk on display of forwarded-to endpoint	If SA8972 enabled, the Number of forwarding station will appear on the display of the forwarded-to endpoint.
PSTN call to station with EC500. Call answered at host station	Calling number on EC500 display is the default number of the SIP trunk, not calling number (changed by Priority, number not valid for this trunk)	

Table 3: Summary of observations identified during Interoperability Testing

8. Verification Steps

This section provides the verification steps that may be performed to verify operation of the Avaya SIP telephony solution with the Priority Netherlands SIP trunking service.

- Verify outbound voice calls can be placed to ISDN PSTN phones and that the calls remain connected for 3 minutes.
- Verify incoming voice calls can be received from ISDN PSTN phones and that the calls remain connected for 3 minutes.
- Verify that calls are properly disconnected when either end disconnects first.
- Using the **list trace station** command, verify that voice calls are using the expected codec and SIP trunk group (for both outbound and incoming calls).
- Verify outbound and inbound fax calls can be placed to and from ISDN fax machines.
- Using the **list trace station** command verify that fax calls are using the expected codec and trunk group (for inbound and outbound calls).
- If Avaya 9600 Series SIP telephones are used with G.711 codecs and Direct IP-to-IP is enabled, verify that the RTP media path is reconfigured once a stable call is established with an ISDN PSTN telephone. Verify by using the **list trace station** command to observe that the IP address of the SIP phone and Priority SIP Application Server are the media endpoints after the call is established.

9. Conclusion

These Application Notes describe the configuration steps required to connect customers using an Avaya SIP telephony solution to the Priority Netherlands SIP trunking standard service using SIP trunking. The Priority Netherlands SIP trunking service is a Voice over IP solution for customers ranging from small businesses to large enterprises. SIP trunking uses the Session Initiation Protocol to connect private company networks to the public telephone network via converged IP access. It provides businesses a flexible, cost-saving alternative to current TDM-based telephony trunk lines.

10. References

This section references documentation relevant to these Application Notes.

The Avaya product documentation is available at <http://support.avaya.com>.

- [1] *Administering Avaya Aura™ Communication Manager*, May 2009, Issue 5.0, Release 5.2, Document Number 03-300509.
- [2] *Avaya Aura™ Communication Manager Feature Description and Implementation*, May 2009, Issue 7, Release 5.2, Document Number 555-245-205
- [3] *Avaya Extension to Cellular and Off-PBX Station (OPS) Installation and Administration Guide Release 3.0*, June 2005, Issue 9, Version 6.0, Document Number 210-100-500.
- [4] *Installing, Administering, Maintaining and Troubleshooting Avaya Aura™ SIP Enablement Services*, May 2009, Issue 0.0, Document Number 03-600768
- [5] *SIP Support in Avaya Aura™ Communication Manager Running on S8xxx Servers*, May 2009, Issue 9, Document Number 555-245-206.
- [6] *Avaya one-X Deskphone Edition for 9600 Series IP Telephones Administrator Guide Release 3.0*, May 2009, Issue 7, Document Number 16-300698
- [7] *Avaya one-X Deskphone Edition for 9600 Series SIP IP Telephones Administrator Guide Release 2.0*, December 2007, Issue 2, Document Number 16-601944

Several Internet Engineering Task Force (IETF) standards track RFC documents were referenced within these Application Notes. The RFC documents may be obtained at: <http://www.rfc-editor.org/rfcsearch.html>.

- [8] RFC 3261 - *SIP (Session Initiation Protocol)*, June 2002, Proposed Standard
- [9] RFC 2833 - *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, May 2000, Proposed Standard

APPENDIX A: Sample SIP INVITE Messages

This section displays the format of the SIP INVITE messages sent by Priority Netherlands SIP trunking and the Avaya SIP network at the enterprise site. These INVITE messages may be used for comparison and troubleshooting purposes. Differences in these messages may indicate that different configuration options were selected.

Sample SIP INVITE Message from Priority Netherlands SIP trunking to Avaya SIP Enablement Services:

```
Session Initiation Protocol
  Request-Line: INVITE sip:31103406030@213.160.12.140:5060 SIP/2.0
    Method: INVITE
    Request-URI: sip:31103406030@213.160.12.140:5060
      Request-URI User Part: 31103406030
      Request-URI Host Part: 213.160.12.140
      Request-URI Host Port: 5060
    [Resent Packet: False]
  Message Header
    Max-Forwards: 68
    Session-Expires: 3600;refresher=uac
    Supported: timer, 100rel
    To: <sip:31103406030@213.160.12.140:5060>
    From: "496975009497" <sip:496975009497@213.160.252.154>;tag=3465096628-521391
    P-Asserted-Identity:<sip:496975009497@213.160.249.236>
    Privacy: none
    Call-ID: 75-3465096628-521385@MSCLAB.admin.prioritytelecom.net
    CSeq: 1 INVITE
    Allow: INVITE, BYE, OPTIONS, CANCEL, ACK, REGISTER, NOTIFY, INFO, REFER, SUBSCRIBE,
PRACK, UPDATE
    Via: SIP/2.0/UDP 213.160.252.154:5060;branch=z9hG4bKf60b3d7b09d06f59fba64411c05133ad
    Contact: <sip:496975009497@213.160.252.154:5060>
    Expires: 180
    Call-Info: <sip:213.160.252.154>;method="NOTIFY;Event=telephone-event;Duration=1000"
    Allow-Events: telephone-event
    Content-Type: application/sdp
    Content-Length: 345
  Message Body
    Session Description Protocol
      Session Description Protocol Version (v): 0
      Owner/Creator, Session Id (o): MSCLAB 6150 669 IN IP4 213.160.252.154
      Session Name (s): sip call
      Connection Information (c): IN IP4 213.160.252.155
      Time Description, active time (t): 0 0
      Media Description, name and address (m): audio 20150 RTP/AVP 8 0 18 125 100 101
      Media Attribute (a): rtpmap:8 PCMA/8000
      Media Attribute (a): rtpmap:0 PCMU/8000
      Media Attribute (a): rtpmap:18 G729/8000
      Media Attribute (a): fmp:18 annexb=yes
      Media Attribute (a): rtpmap:125 X-CCD/8000
      Media Attribute (a): rtpmap:100 X-NSE/8000
      Media Attribute (a): fmp:100 192-194
      Media Attribute (a): rtpmap:101 telephone-event/8000
      Media Attribute (a): fmp:101 0-16
```

Sample SIP INVITE Message from Avaya SIP Enablement Services to Priority Netherlands SIP trunking:

```
Session Initiation Protocol
Request-Line: INVITE sip:+496975009497@213.160.252.154:5060;lr SIP/2.0
Method: INVITE
Request-URI: sip:+496975009497@213.160.252.154:5060;lr
    Request-URI User Part: +496975009497
    Request-URI Host Part: 213.160.252.154
    Request-URI Host Port: 5060
[Resent Packet: False]
Message Header
Accept-Language: en
Call-ID: 06861be4ac5de17d7d4ac35f4a00
CSeq: 1 INVITE
From: "Prio DCP" <sip:31103406030@localdomain.com>;tag=06861be4ac5de17c7d4ac35f4a00
Record-Route: <sip:213.160.12.140:5060;lr>
Record-Route: <sip:213.160.12.135:5061;transport=tls;lr>
To: "+496975009497" <sip:+496975009497@213.160.12.140:5060>
Via: SIP/2.0/UDP 213.160.12.140:5060;branch=z9hG4bK03030363636383833ae1.0
Contact: "Prio DCP" <sip:31103406030@213.160.12.135:5061;transport=tls>
Max-Forwards: 70
User-Agent: Avaya CM/R015x.02.0.947.3
Allow: INVITE, CANCEL, BYE, ACK, PRACK, SUBSCRIBE, NOTIFY, REFER, OPTIONS, INFO, PUBLISH
Supported: timer, replaces, join, histinfo, 100rel
Alert-Info: <cid:internal@invalid.unknown.domain>;avaya-cm-alert-type=internal
Min-SE: 1200
Session-Expires: 1200;refresher=uac
P-Asserted-Identity: "Prio DCP" <sip:31103406030@localdomain.com:5061>
P-Charging-Vector: icid-value="AAS:1042-be6168001dec54ac34a7d7b4a5f"
History-Info: <sip:+496975009497@192.168.193.110>;index=1,"+496975009497"
<sip:+496975009497@192.168.193.110>;index=1.1
Content-Type: application/sdp
Content-Length: 239
Message Body
Session Description Protocol
    Session Description Protocol Version (v): 0
    Owner/Creator, Session Id (o): - 1 1 IN IP4 213.160.12.135
    Session Name (s): -
    Connection Information (c): IN IP4 213.160.12.141
    Bandwidth Information (b): AS:64
    Time Description, active time (t): 0 0
    Media Description, name and address (m): audio 2360 RTP/AVP 8 18 0 101
    Media Attribute (a): rtpmap:8 PCMA/8000
    Media Attribute (a): rtpmap:18 G729/8000
    Media Attribute (a): fmp:18 annexb=yes
    Media Attribute (a): rtpmap:0 PCMU/8000
    Media Attribute (a): rtpmap:101 telephone-event/8000
```


APPENDIX B: Specifying Pattern Strings in Address Maps

The syntax for the pattern matching used within the 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 character(s).

In the pattern matching string used in the Avaya Aura™ SES:

- 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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