Avaya Solution & Interoperability Test Lab

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

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between the Global Crossing SIP Trunking solution and an Avaya IP telephony solution. The Avaya solution consists of Avaya SIP Enablement Services, Avaya Communication Manager, and various Avaya SIP, H.323, digital and analog endpoints.

Headquartered in Hamilton, Bermuda, Global Crossing Limited provides Internet Protocol (IP) and legacy telecommunications services worldwide. Enterprise customers with an Avaya IP telephony 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: DID, Toll Free and VTERM.

Global Crossing is a member of the Avaya Dev*Connect* Service Provider program. Information in these Application Notes has been obtained through Dev*Connect* compliance testing and additional technical discussions. Testing was conducted via the Dev*Connect* Program at the Avaya Solution and Interoperability Test Lab.

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1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between the Global Crossing SIP Trunking solution and an Avaya IP telephony solution. The Avaya solution consists of Avaya SIP Enablement Services, Avaya Communication Manager, and various Avaya SIP, H.323, digital and analog endpoints.
Global Crossing is an international service provider offering a wide variety of telecommunications and data services such as:

- Domestic and international long-distance traffic
- Direct-inward-dialing transport
- Toll-free enhanced routing services via traditional TDM or VoIP interconnections
- Converged IP services, IP virtual private network (VPN) services, Internet access services
- Frame Relay and Asynchronous Transfer Mode (ATM) services

Customers using this Avaya IP telephony solution with Global Crossing's VoIP Network services are able to place and receive PSTN calls via a dedicated broadband Internet connection using the SIP protocol. This converged network solution is an alternative to more traditional PSTN trunks such as T1 or ISDN PRI.

Global Crossing can connect directly to an IP phone system as well as an external router. Global Crossing's VoIP Network service offerings are as follows:

- **DID** – IP Inbound to customer
- **Toll Free** – IP Inbound to customer
- **VTERM** – IP Outbound from customer
- **On-Net** – Enterprise site to site Calling

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Figure 1 illustrates an example Avaya IP telephony solution connected to Global Crossing's SIP Trunking solution. This is the configuration used during the Dev*Connect* compliance testing process.

The Avaya IP telephony solution used to create a simulated customer site contained:

- Avaya S8710 Server pair with an Avaya G650 Media Gateway.
- Avaya SIP Enablement Services (SES) software operating on an Avaya S8500B server platform.
- Avaya 9630 IP telephone (configured for the SIP protocol)
- Avaya 9640 IP telephone (configured for the H.323 protocol)
- Avaya 4610 IP telephone (configured to use either the SIP or H.323 protocol).
- Avaya 6416 digital and 6210 analog phones.
- Avaya one-X Desktop SIP Softphone

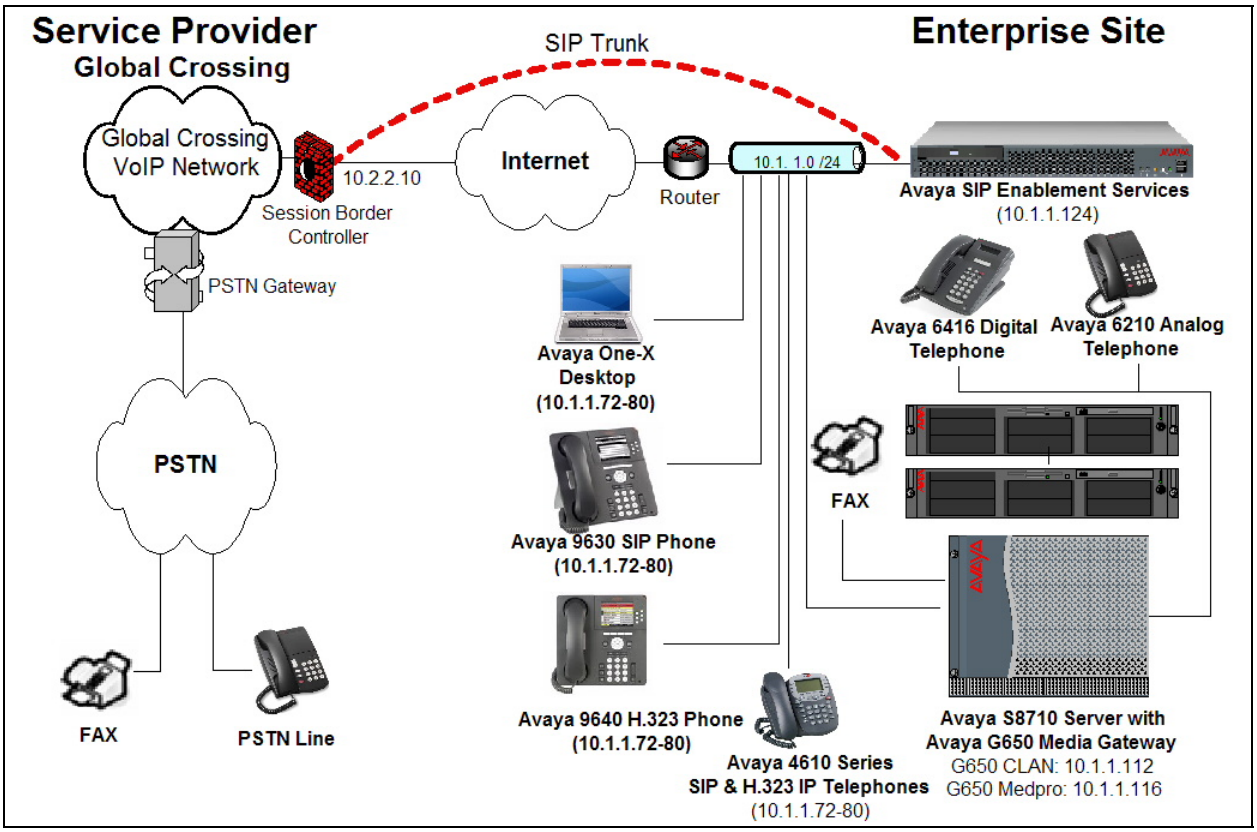

Figure 1: Avaya IP Telephony Network using Global Crossing SIP Trunking Service

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1.1 Call Flows

To better understand how calls are routed between the PSTN 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 Avaya Communication Manager.

1. A user on the PSTN dials a Global Crossing provided DID number assigned to an Avaya Communication Manager telephone at the enterprise site. The PSTN routes the call to the Global Crossing network (as the local service provider). Global Crossing then routes the DID number to the assigned customer.

2. Based on the DID number, Global Crossing offers the call to Avaya SES using SIP signaling messages sent over the converged access facility. Note that the assignment of the DID number and the address of the Avaya SES server was previously established during the ordering and provisioning of the service.

3. Avaya SES routes the call to the Avaya S8710 Server running Avaya Communication Manager over a SIP trunk between the elements.

4. Avaya Communication Manager terminates the call to the directly connected non SIP Phone, (i.e. Analog, Digital, or H.323 IP Phones), as shown in step 4.

- or –

4a. Inbound calls destined for a SIP extension at the enterprise are routed to Avaya Communication Manager which then transmits the appropriate SIP signaling via Avaya SES to the SIP telephone (as shown by the 4a arrow.)

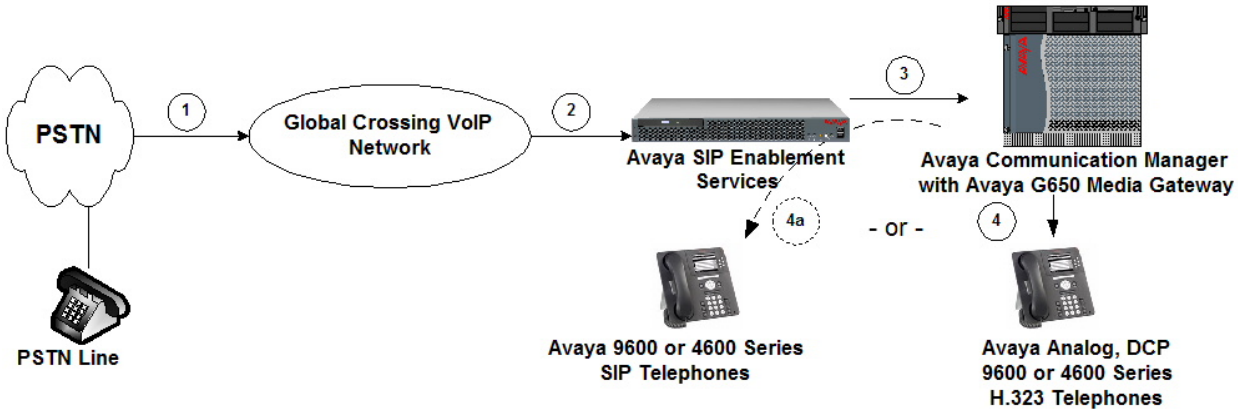

Figure 2: Incoming PSTN Calls to Avaya Communication Manager

Appendix A illustrates an example of a SIP INVITE message sent by Global Crossing for an incoming DID call.

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The second call scenario illustrated in **Figure 3** is an outgoing call from an Avaya telephone at the enterprise site to the PSTN via the SIP trunk to Global Crossing.

1. An Avaya H.323, analog or digital telephone served by Avaya Communication Manager originates a call to a user on the PSTN.

- or-

1a. An Avaya SIP telephone originates a call that is routed via Avaya SES (as shown by the 1a arrow) to Avaya Communication Manager.

2. The call request is handled by Avaya Communication Manager where origination treatment such as class of service restrictions and automatic route selection is performed. Avaya Communication Manager selects the SIP trunk and sends the SIP signaling messages to Avaya SIP Enablement Services.

3. Avaya SIP Enablement Services routes the call to Global Crossing.

4. Global Crossing completes the call to the PSTN.

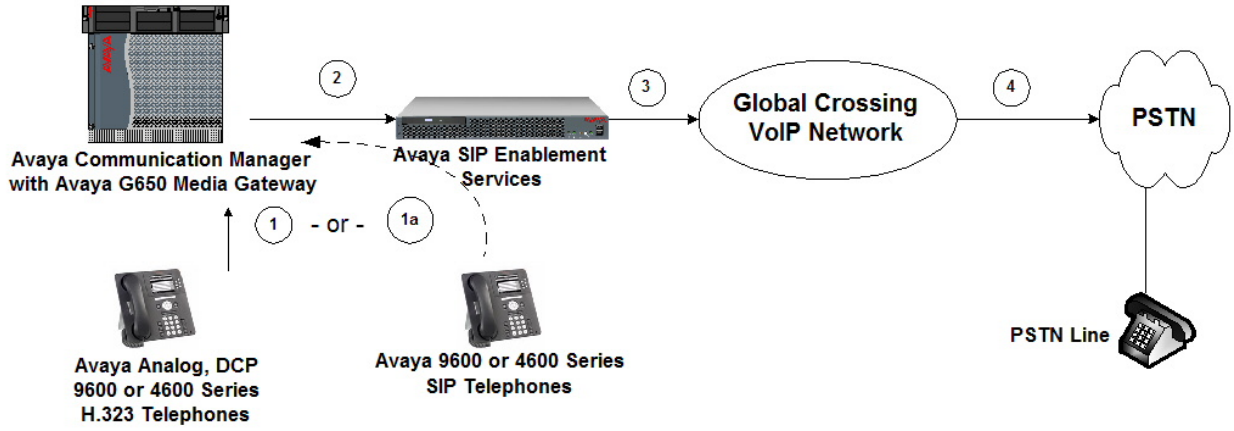

Figure 3: Outgoing Calls from Avaya Communication Manager to the PSTN

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2. Equipment and Software Validated

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

Avaya IP Telephony Solution Components	
Avaya S8710 Server with an Avaya G650 Media Gateway	Communication Manager 5.0 R015x.00.0.825.4 Update: 5.0SP2 and 15348, See PSN #1842U
Avaya SIP Enablement Services on S8500B Server	SES-5.0.0.0-825.31
Avaya 9640 IP Telephone	R1.550 – H.323
Avaya 9630 IP Telephone	R2.0.3 – SIP
Avaya 4621 IP Telephone	R2.2.2 – SIP
Avaya one-X Desktop Edition SIP endpoint	R2.1 – SIP – Build 78
Avaya 4610 IP Telephone	R2.8.8.7 – H.323
Avaya 6416 Digital Telephone	n/a
Avaya 6210 Analog Telephone	n/a
Global Crossing Global SIP Trunking Solution Components	
PSTN Gateway: Sonus HD NBS	V06.04.10R005
SBC: ACME Net-Net 4250	4.1.1 Patch 56a
Application Server: Sonus PSX	V06.04.11R000

Table 1: Equipment and Software Tested

The specific configuration above was used for the Global Crossing compatibility testing. Note that this solution will be compatible with all other Avaya Server and Media Gateway platforms running similar versions of Avaya Communication Manager and Avaya SIP Enablement Services.

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3. Configure Avaya Communication Manager

This section describes the steps for configuring a SIP trunk on Avaya Communication Manager. The SIP trunk is established between Avaya Communication Manager and Avaya SIP Enablement Services (SES) server. This trunk will carry the SIP signaling sent to the Global Crossing SIP Trunking solution.

This SIP trunk also provides the trunking for SIP endpoint devices such as Avaya 9600 and 4600 Series SIP telephones using Avaya Communication Manager. Avaya SIP telephones register with Avaya SES but have calling privileges and features managed by Avaya Communication Manager. Avaya Communication Manager acts as a back-to-back SIP user agent when a SIP phone places or receives a call over a SIP trunk to a service provider.

Note the use of SIP endpoints is optional. The steps discussed in Sections 3.2 and 4.3 describing SIP endpoint administration may be omitted if SIP endpoints are not used. In the Avaya SIP architecture, the Avaya SES acts as a SIP proxy through which all incoming and outgoing SIP messages flow to Global Crossing's SIP Trunking solution. There is no direct SIP signaling path between Global Crossing and Avaya Communication Manager or Avaya SIP endpoints.

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

All outgoing calls to the PSTN are processed within Avaya Communication Manager and may be first subject to outbound features such as automatic route selection, digit manipulation and class of service restrictions. Once Avaya Communication Manager selects a SIP trunk, the SIP signaling is routed to the Avaya SES.

The dial plan for the configuration described in these Application Notes consists of 10-digit dialing for local and long-distance calls over the PSTN. In addition, Directory Assistance calls (411) and International calls (011+Country Code) were also supported. Avaya Communication Manager routes all calls using Automatic Route Selection (ARS), except for intra-switch calls.

Avaya Communication Manager configuration was performed using the System Access Terminal (SAT). The general installation of the S8710 Server, G650 Media Gateway and circuit packs such as the CLAN is presumed to have been previously completed and is not discussed here.

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3.1 SIP Trunk Configuration

Step 1: Confirm Necessary Optional Features

Log into the Avaya Communication Manager SAT interface and confirm that sufficient SIP trunk and Off PBX Telephone capacities are enabled. Use the **display system-parameters customer-options** command to determine these values as shown in **Figure 4**. The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity.

```
display system-parameters customer-options                          Page 1 of 10
                                OPTIONAL FEATURES

     G3 Version: V15
       Location: 1                            RFA System ID (SID): 1
       Platform: 8                            RFA Module ID (MID): 1

                                                             USED
                                Platform Maximum Ports: 44000 229
                                      Maximum Stations: 36000 53
                             Maximum XMOBILE Stations: 0     0
                     Maximum Off-PBX Telephones - EC500: 10    0
                     Maximum Off-PBX Telephones -   OPS: 36000 23
                     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

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On Page 2, verify that the number of **Maximum Administered SIP Trunks** supported by the system is sufficient for the combination of trunks to the Global Crossing network, SIP endpoints and any other SIP trunks used. Note that each SIP telephone on a call with Global Crossing uses two SIP trunks for the duration of the call.

```
display system-parameters customer-options                      Page   2 of  10
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 100   20
           Maximum Concurrently Registered IP Stations: 100   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: 5     0
                  Maximum Video Capable H.323 Stations: 10    0
                   Maximum Video Capable IP Softphones: 10    0
                       Maximum Administered SIP Trunks: 200   156

   Maximum Number of DS1 Boards with Echo Cancellation: 1     0
                             Maximum TN2501 VAL Boards: 1     1
                     Maximum G250/G350/G700 VAL Sources: 50    0
           Maximum TN2602 Boards with 80 VoIP Channels: 2     1
          Maximum TN2602 Boards with 320 VoIP Channels: 2     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 Form – Page 2

Step 2: Assign Node Names

In the **IP Node Names** form, assign the node name and IP address for the Avaya SES at the enterprise site as shown in **Figure 6**. In this case "SES" and "10.1.1.124" are being used, respectively. The SES node name will be used throughout the other configuration screens of Avaya Communication Manager.

In this example "CLAN" and "10.1.1.112" are the name and IP address assigned to the TN799DP card. The CLAN entry was previously created during the installation of the system. Note, in smaller gateways such as an Avaya G350 Media Gateway, the S8300 Server processor address (procr) is used as the SIP signaling interface instead of the CLAN interface.

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```
change node-names ip                                          Page   1 of   1
                                IP NODE NAMES
    Name              IP Address          Name              IP Address
CLAN                10 .1  .1  .112                           .   .   .
default             0  .0  .0  .0                            .   .   .
ipsi                10 .1  .1  .109                          .   .   .
medpro-hw11         10 .1  .1  .116                          .   .   .
procr               10 .1  .1  .104                          .   .   .
SES                 10 .1  .1  .124                          .   .   .
val1-tn2501ap       10 .1  .1  .122                          .   .   .
windowPC            10 .1  .1  .101                          .   .   .
                       .   .   .                             .   .   .
                       .   .   .                             .   .   .
( 8 of 8   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 6: IP Nodes Names Form

Step 3: Define IP Network Region

The **IP Network Region** form specifies the parameters used by the SIP trunk group serving the Avaya SES server (used to reach Global Crossing and any optional SIP endpoints). Note that these parameters also apply to any other elements (such as H.323 phones, MedPro cards, etc.) also assigned to this region. Use the **change ip-network-region** form command to set the following values:

- The **Authoritative Domain** field is configured to match the domain name configured on the Avaya SES. In this configuration, the domain name is *east.devcon.com*. This field is required for endpoints to call the public network.
- By default, **IP-IP Direct Audio** (both **Intra** and **Inter Region**) is enabled to allow audio traffic to be sent directly between endpoints without using media resources such as the TN2302AP IP Media Processor (MedPro) card. In the case of Global Crossing, **IP-IP Direct Audio** will be supported and these parameters will be enabled.
- The **Codec Set** is set to the number of the IP codec set to be used for calls within the IP network region. In this configuration, this codec set will apply to calls with Global Crossing as well as any IP telephone (H.323 or SIP) within the enterprise.

In this case, the SIP trunk is assigned to the same IP network region as the G650 Media Gateway, CLAN and MedPro cards. 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.

Note also that the **IP Network Region** form is used to set the QoS packet parameters that provides priority treatment for signaling and audio packets over other data traffic on Global Crossing's Global SIP Trunking solution. These parameters may need to be aligned with the specific values provided by Global Crossing.

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```
change ip-network-region 1                                      Page   1 of  19
                              IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: east.devcon.com
    Name: Default System All
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? y
   UDP Port Max: 65531
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 7: IP Network Region Form

Step 4: Define IP Codecs and T.38 FAX

Open the IP Codec Set form using the ip-codec value specified in the **IP Network Region** form (**Figure 7**) and enter the audio codec type to be used for calls routed over the SIP trunk. The settings of the **IP Codec Set** form are shown in **Figure 8.** Note that the **IP Codec Set** form may include multiple codecs listed in priority order to allow the codec for the call to be negotiated during call establishment. In the configuration below, G.729a is the preferred codec followed by G.711mu.

```
change ip-codec-set 1                                           Page   1 of   2

                          IP Codec Set

    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.729A            n         2        20
 2: G.711MU           n         2        20
 3:
 4:
 5:
 6:
 7:
```

Figure 8: IP Codec Set Form

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The T.38 FAX relay standard will be used for FAX transmission. To enable T.38, go to page 2 of the IP Codec Set form and configure **t.38-standard** in the **FAX Mode** field as shown in **Figure 9** below.

```
change ip-codec-set 5                                          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 9: IP Codec Set Form – Page 2

Step 5: Configure the Signaling Groups

For interoperability with Global Crossing, a minimum of two signaling groups must be configured. One signaling group will be used for inbound and outbound PSTN calls to and from the Global Crossing gateway while the second signaling group is required for calls involving SIP stations – recall that SIP stations register with the SES and therefore require this dedicated SIP trunk to leverage the calling privileges and features managed by Avaya Communication Manager. For the purposes of this document, these groups will be referred to as the "PSTN Signaling and Trunk group" and the "OPS Signaling and Trunk group". The configuration steps below show how to configure both of these signaling groups.

Configure the PSTN **Signaling Group** form which will be used for PSTN calls shown in **Figure 10** 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.
- **Near-end Node Name** Specify the Avaya CLAN card (node name "CLAN"). This field value is taken from the **IP Node Names** form shown in **Figure 6**. For smaller media server platforms, the near (local) end of the SIP signaling group may be the S8300 media server processor (procr) rather than the CLAN.
- **Far-end Node Name** Specify the Avaya SIP Enablement Services (node name "SES"). This field value is taken from the **IP Node Names** form shown in **Figure 6**.
- 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 7**). 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

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region (Network Region 1) was used for local and PSTN calls; however, different network regions can be used in the field.

- For **Far-end Domain** field, enter the IP Address of Global Crossing's Session Border Controller (SBC). For outbound PSTN calls to Global Crossing, this field sets the domain in the Uniform Resource Identifier (URI) of the SIP "To" address in the outbound INVITE message. For inbound PSTN calls from Global Crossing, the domain part of the incoming INVITE's "From" header is compared to the **Far-end Domain** field of the configured signaling groups. The call is then routed to the signaling group that matches this field. Mis-configuring this field may prevent calls from being successfully established to other SIP endpoints or to the PSTN.
- The **Direct IP-IP Audio Connections** field is set to 'y'. Global Crossing supports the Avaya **Direct IP-IP Audio** feature. This feature can be disabled if desired.
- The **DTMF over IP** field should remain set to the default value of *rtp-payload*. This value enables Avaya Communication Manager to send DTMF transmissions using RFC 2833. [8]
- 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: CLAN                  Far-end Node Name: SES
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1
       Far-end Domain: 10.2.2.10

                                             Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
                                                       IP Audio Hairpinning? n
 Session Establishment Timer(min): 120
```

Figure 10: PSTN Signaling Group Form

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Now, configure the *OPS* **Signaling Group** form following the same steps used for the signaling group above with one exception. Set the **Far-end Domain** field to the local SES SIP domain, in this case the value used is *east.devcon.com* as shown in **Figure 11**:

```
add signaling-group 2                                          Page 1 of 1
                              SIGNALING GROUP

 Group Number: 2              Group Type: sip
                        Transport Method: tls



   Near-end Node Name: CLAN                  Far-end Node Name: SES
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1
       Far-end Domain: east.devcon.com

                                             Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
                                                       IP Audio Hairpinning? n
 Session Establishment Timer(min): 120
```

Figure 11: OPS Signaling Group Form

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Step 6: Configure the Trunk Groups
As described above in step 5, two trunks must also be configured. One trunk will be paired with the PSTN signaling group and the other with the OPS signaling group.

Configure the *PSTN* **Trunk Group** form as shown in **Figure 12** using the **add trunk-group** command. In this case the trunk group number chosen is 1. On Page 1 of this form:

- Set the **Group Type** field to *sip*.
- Choose a mnemonic **Group Name**.
- Specify an available trunk access code (**TAC**).
- Set the **Service Type** field to *public-ntwrk.*
- Specify the *PSTN* signaling group associated with this trunk group in the **Signaling Group** field as previously specified in **Figure 10**.
- Specify the **Number of Members** supported by this SIP trunk group.

```
add trunk-group 1                                            Page   1 of  21
                                TRUNK GROUP

Group Number: 1                   Group Type: sip            CDR Reports: y
  Group Name: Global Crossing PSTN Access     COR: 1         TN: 1        TAC: 101
   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: 10
```

Figure 12: PSTN Trunk Group Form – Page 1

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



                      Numbering Format: public
                                             Prepend '+' to Calling Number? n


                                               Replace Unavailable Numbers? n
```

Figure 13: PSTN Trunk Group Form – Page 3

On page 4 of the **Trunk Group** form:

- set the **Telephone Event Payload Type** to *101*. This is the value Global Crossing recommends for their service.

```
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
Service Provider Network Call Redirection? n
             Telephone Event Payload Type: 101
```

Figure 14: PSTN Trunk Group Form – Page 4

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Now, configure the *OPS* **Trunk Group** form as shown in **Figure 15** using the **add trunk-group** command. In this case the trunk group number chosen is 2. On Page 1 of this form:

- Set the **Group Type** field to *sip*.
- Choose a mnemonic **Group Name**.
- Specify an available trunk access code (**TAC**).
- Set the **Service Type** field to *tie*.
- Specify the *OPS* signaling group associated with this trunk group in the **Signaling Group** field as previously specified in **Figure 11**.
- Specify the **Number of Members** supported by this SIP trunk group.

```
add trunk-group 2                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 1                   Group Type: sip          CDR Reports: y
  Group Name: OPS                                   COR: 1        TN: 1        TAC: 102
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                          Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                      Signaling Group: 2
                                                    Number of Members: 10
```

Figure 15: OPS Trunk Group Form – Page 1

On Page 3 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 2                                           Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                       Maintenance Tests? y



                     Numbering Format: public
                                             Prepend '+' to Calling Number? n


                                               Replace Unavailable Numbers? n
```

Figure 16: PSTN Trunk Group Form – Page 3

Step 7: Configure Calling Party Number Information

Use the **change public-unknown-numbering** command to configure Avaya Communication Manager to send the full calling party number to the far-end.

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In this case, all stations with a 5-digit extension beginning with 6 should send the calling party number 732-856-xxxx when an outbound call uses SIP trunk group #1 (*this is the PSTN trunk group specified in* **Step 6**). This calling party number will be sent to the far-end in the SIP "From" header. Note that the number used here is just an example, in a real deployment the DID numbers are assigned by Global Crossing.

Figure 17 shows the use of the **change public-unknown-numbering** command to implement this rule.

```
change public-unknown-numbering 0                                Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                   Total                                    Total
Ext Ext    Trk       CPN            CPN Ext Ext    Trk       CPN            CPN
Len Code   Grp(s)    Prefix         Len Len Code   Grp(s)    Prefix         Len

 5  6       1         73285          10
```

Figure 17: Numbering Public/Unknown Format Form

Step 8: Automatic Route Selection for Outbound Calls

In these Application Notes, the Automatic Route Selection (ARS) feature will be used to route outbound calls via the SIP trunk to the Global Crossing SIP Trunking solution to a PTSN destination.

Use the **change dialplan analysis** command to add ***9*** as a feature access code (**fac**).

```
change dialplan analysis                                         Page   1 of  12
                            DIAL PLAN ANALYSIS TABLE
                                                                Percent Full:    3
      Dialed  Total  Call      Dialed  Total  Call      Dialed  Total  Call
      String  Length Type      String  Length Type      String  Length Type
        1       3    dac
        7       4    ext
        8       4    ext
        9       1    fac
        *       3    fac
        #       3    fac
```

Figure 18: Change Dialplan Analysis Form

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Use the **change feature-access-codes** command to specify ***9*** as the access code for outside dialing.

```
change feature-access-codes                                          Page   1 of   7
                              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: *03
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code:
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                  Automatic Callback Activation:        Deactivation:
Call Forwarding Activation Busy/DA: *10    All: *11    Deactivation: #10
                         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:
                    Contact Closure   Open Code:          Close Code:
                    Contact Closure  Pulse Code:
```

Figure 19: Feature Access Codes Form

Now use the **change ars analysis** command to configure the route pattern selection rule based upon the number dialed following the dialed digit "9". In this sample configuration, the PSTN numbers dialed are all in the form 1AAANNXXXXX (A= Area Code, N=[2-9], X=[0-9]). If the area code (AAA) is 732, the call is to be routed to a route pattern containing the SIP trunk groups used for Global Crossing. Note that further administration of ARS is beyond the scope of these Application Notes but discussed in References [1] and [2].

```
change ars analysis 173                                              Page   1 of   2
                            ARS DIGIT ANALYSIS TABLE
                                  Location:  all           Percent Full:    3


          Dialed           Total       Route    Call   Node  ANI
          String            Min  Max  Pattern   Type   Num   Reqd
    173                     11   11     1       fnpa           n
```

Figure 20: ARS Analysis Form

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Use the **change route-pattern** command to define the SIP trunk group included in the route pattern that ARS selects. In this configuration, route pattern 1 will be used to route calls to trunk group 1 (the SIP trunk created in **Step 6, Figure 12**).

```
change route-pattern 1                                          Page   1 of   3
                    Pattern Number: 1   Pattern Name: To PSTN
                              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: 1    0       1                                                       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 3 4 W     Request                                  Dgts Format
                                                           Subaddress
 1: y y y y y n  n            rest                                         none
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
 6: y y y y y n  n            rest                                         none
```

Figure 21: Route Pattern Form

Note that additional entries need to be added to the ARS table for additional numbering plan options such as 411 information or 911 emergency services.

Step 9: Configure Incoming Digit Translation

This step configures the settings necessary to map incoming DID calls to the proper extension(s).

The incoming digits sent in the INVITE message from Global Crossing are manipulated as necessary to route calls to the proper extension on Avaya Communication Manager. Note that this step assumes that the Global Crossing subscriber has already received their DID numbers from Global Crossing.

In the examples used in these Application Notes, the incoming DID numbers provided by Global Crossing do not have a direct correlation to the internal extensions assigned within Avaya Communication Manager. Thus all incoming called number digits are deleted and replaced by the assigned extension number.

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To create a fully mapped extension number as shown in **Figure 22**:

- Open the **Incoming Call Handling Treatment** form for the *PSTN* SIP trunk group configured in **Figure 12**, in this case Trunk Group 1.
- For each extension assigned a DID number from Global Crossing, enter **10** into the **Called Len** and **Del** fields, and the entire **10 digit DID number** into the **Called Number** field.
- Enter the desired Avaya Communication Manager extension number into the **Insert** field.

```
change inc-call-handling-trmt trunk-group 1                    Page   1 of  30
                      INCOMING CALL HANDLING TREATMENT
 Service/      Called    Called      Del Insert
 Feature        Len      Number
 public-ntwrk    10 6023574046       10  60003
```

Figure 22: Incoming Call Handling Treatment – Full Extension Mapping

If the customer's extension numbering aligns with the DID numbers (i.e., the final DID digits match the extension), it is not necessary to define an entry for each DID number. Assuming a PBX dial plan that used the 5 digit extensions 60000 thru 61999 and assuming Global Crossing provided DID numbers of 732-626-0000 thru 9999, the incoming number translation would be done similar to **Figure 23**. Note that the Called Number entry in this case represents the common matching portion applicable to all incoming numbers. Thus 732626 matches all numbers in the assigned DID block from Global Crossing.

```
change inc-call-handling-trmt trunk-group 1                    Page   1 of  30
                              INCOMING CALL HANDLING TREATMENT
        Service/       Called    Called        Del  Insert
        Feature         Len      Number
        public-ntwrk    10      732626         5
```

Figure 23: Incoming Call Handling Treatment – Simple Extension Mapping

Step 10: Save Avaya Communication Manager Changes

Enter "save translation" to make the changes permanent.

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3.2 SIP Endpoint Configuration

This section describes the administration of SIP telephones and requires the preceding SIP Trunk configuration to have been completed. SIP telephones are optional and not required to use the Global Crossing SIP Trunking solution.

Step 1: **Assign a Station**

The first step in adding an off-PBX station (OPS) for Avaya SIP telephones registered with Avaya SIP Enablement Services is to assign a station as shown in **Figure 24**. This example uses an Avaya one-X 9630 Deskphone.

Using the **add station** command from the SAT:

- Set the station **Type** to the value "9630".
- Enter a **Name** for the station that will be displayed.
- The **Security Code** is 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 60004                                          Page   1 of   5
                                     STATION

Extension: 60004                        Lock Messages? n               BCC: 0
     Type: 9630                         Security Code:                 TN: 1
     Port: IP                          Coverage Path 1:                COR: 1
     Name: 9630 SIP                   Coverage Path 2:                 COS: 1
                                      Hunt-to Station:
STATION OPTIONS
                                          Time of Day Lock Table:
              Loss Group: 19         Personalized Ringing Pattern: 1
                                               Message Lamp Ext: 60004
            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


                                           Customizable Labels? y
```

Figure 24: Station Administration – Page 1

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On Page 2 of the **Station** form,

- Set the **Restrict Last Appearance** value to 'n' on phones that have 3 or fewer call appearances to maintain proper SIP conference and transfer operation.
  Setting the **Restrict Last Appearance** value to 'y' reserves the last call appearance for outbound calls. Certain SIP conference and transfer features will not function properly if a third appearance is not available for incoming calls.

```
add station 60004                                              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? n
  Active Station Ringing: single
                                                          EMU Login Allowed? n
        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed
         Multimedia Mode: enhanced
    MWI Served User Type: sip-adjunct           Display Client Redirection? n
                                              Select Last Used Appearance? n
                                                Coverage After Forwarding? s
                                                  Multimedia Early Answer? n
                                              Direct IP-IP Audio Connections? y
  Emergency Location Ext: 60004         Always Use? n IP Audio Hairpinning? n
```

Figure 25: Station Administration – Page 2

On Page 4 of the **Station** form, configure at least 3 call appearances under the Button Assignments section for the SIP telephone as shown in **Figure 26**.

```
add station 60004                                              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:
```

Figure 26: Station Administration – Page 4

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A similar number of call appearances should be configured on the SIP Telephone which is beyond the scope of these Application Notes. The parameters to administer call appearances (and many other settings) are described in Reference [7].

Step 2: **Configure Off-PBX Station Mapping**

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 Avaya SIP Enablement Services, which will then route the call to the SIP telephone.

On the **Off-PBX-Telephone Station-Mapping** form shown in **Figure 27**:

- Specify the **Station Extension** of the SIP endpoint.
- 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 on Avaya SIP Enablement Services also match the extensions of the corresponding stations on Avaya Communication Manager. However, this is not a requirement.
- Set the **Trunk Selection** field to *2*, which is the number assigned to the *OPS* SIP trunk group used to route the call to the SIP station. This trunk group number was previously defined in **Figure 15**.
- Set the **Configuration Set** value. In these Application Notes, Configuration Set 1 uses the default values of the Configuration Set form.

```
change off-pbx-telephone station-mapping 60004                  Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station     Application   Dial   Phone Number      Trunk       Configuration
 Extension                 Prefix                   Selection   Set
 60004         OPS              - 60004              2          1
```

Figure 27: 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 60000                  Page   2 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Call       Mapping      Calls       Bridged
 Extension     Limit      Mode         Allowed     Calls
 60000         3          both           all         both
```

Figure 28: Stations with Off-PBX Telephone Integration – Page 2

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Step 3: Repeat for each SIP Phone
Repeat Steps 1 and 2 for each SIP phone to be added.

Step 4: Save Avaya Communication Manager Changes
Enter "save translation" to make the changes permanent.

3.3 Configuration of Non G.729a SIP Endpoints

As discussed in **Section 3.1**, **Step 4**, the preferred codec in this configuration is G.729a. However, the Avaya 4600 Series SIP phones do not support the G.729a codec. Instead, they support G.729b. Because of this G.729 codec incompatibility, the voice path could be disrupted when the phone is configured to Shuffle or use the Direct-IP feature. To avoid this, separate network regions can be configured for these phones. Inter region codec values can be established between the network region that interfaces Global Crossing' SIP Trunking devices and the network region that the phones reside in such that a non G.729 codec type is used. The details for this configuration are beyond the scope of this document, please refer to the Avaya Communication Manager Network Region Configuration Guide [9].

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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. It is assumed that Avaya SIP Enablement Services software and the license file have already been installed on Avaya SIP Enablement Services. During the software installation, the **initial_setup** 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 [4].

This section is divided into two parts: **Section 4.1** provides the steps necessary to configure a SIP trunk to Global Crossing's SIP Trunking solution. **Section 4.2** provides the steps necessary to complete the administration for optional SIP endpoints.

4.1 SIP Trunking to Global Crossing

Step 1: Log in to Avaya SIP Enablement Services

Access the SES Administration web interface, by entering **http://<ip-addr>/admin** as the URL in an Internet browser, where <*ip-addr*> is the IP address of Avaya SIP Enablement Services server.

Log in with the appropriate credentials and then select the *Launch Administration Web Interface* link from the main screen as shown in **Figure 29**.

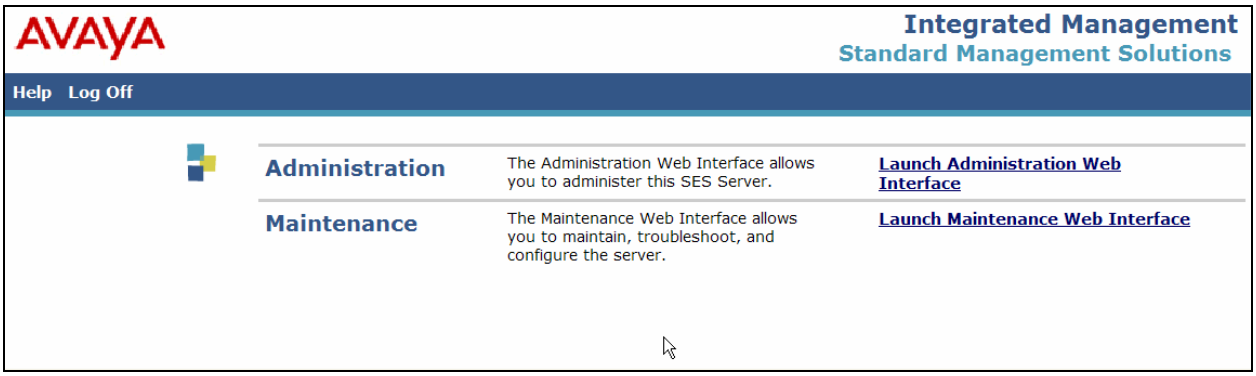

Figure 29 - Avaya SES Main Screen

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The SES administration home screen shown in **Figure 30** will be displayed.

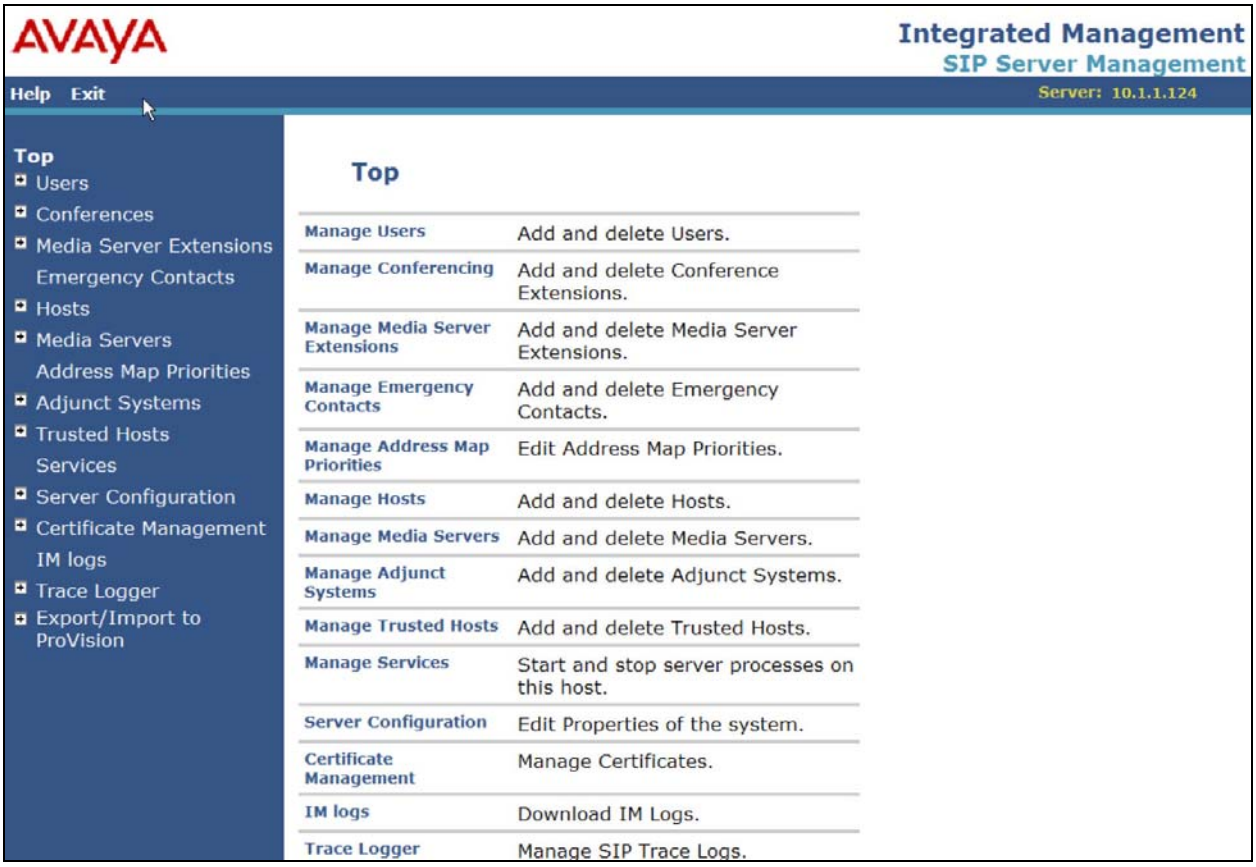

Figure 30: Avaya SES Administration Home Page

Step 2: Verify System Properties

From the left pane of the Administration web interface, expand the **Server Configuration** option and select **System Properties**. This screen displays the SES version and the network properties entered via the **initial_setup** script during the installation process.

In the **System Properties** screen,

- Verify the **SIP Domain** name assigned to Avaya SIP Enablement Services.
- Verify the **License Host** field. This is the host name, the fully qualified domain name, or the IP address of the SES that is running the WebLM application and has the associated license file installed. This entry should be set to **localhost** unless the WebLM server is not co-resident with this server.
- If changes were necessary in the **System Properties** screen, click the **Update** button.

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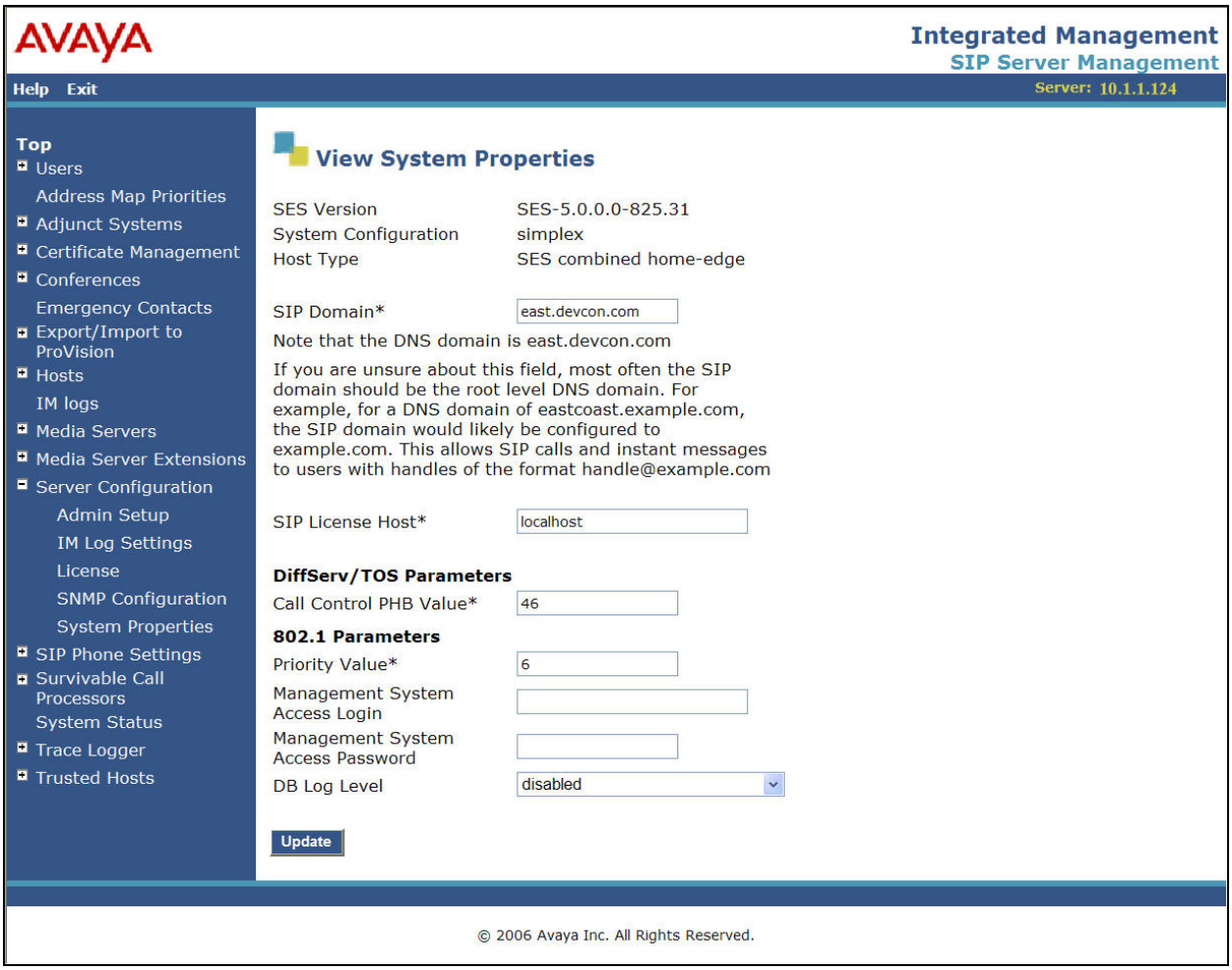

Figure 31: System Properties

Step 3: Verify the Avaya SES Host Information

Verify the Avaya SES Host information using the **Edit Host** page. In these Application Notes the Avaya SES **Host Type** is a combined ***home/edge***. This means that both the Global Crossing SIP Trunking Solution and Avaya Communication Manager are contacting the same SES. In a separate **home/edge** configuration, Avaya Communication Manager would contact the **home** Avaya SES server while the Global Crossing SIP Trunking Solution would contact the Avaya SES **edge** server. Display the **Edit Host** page (**Figure 32)** by following the **Hosts** link in the left navigation pane. Click 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**, **TCP** and **TLS** checkboxes are enabled as **Listen Protocols**.
- Verify that **TLS** is selected as the **Link Protocol**.
- Default values for the remaining fields may be used.
- Click the **Update** button only if changes are necessary. Otherwise exit the **Edit Host** page by selecting the **Top** link on the left navigation bar.

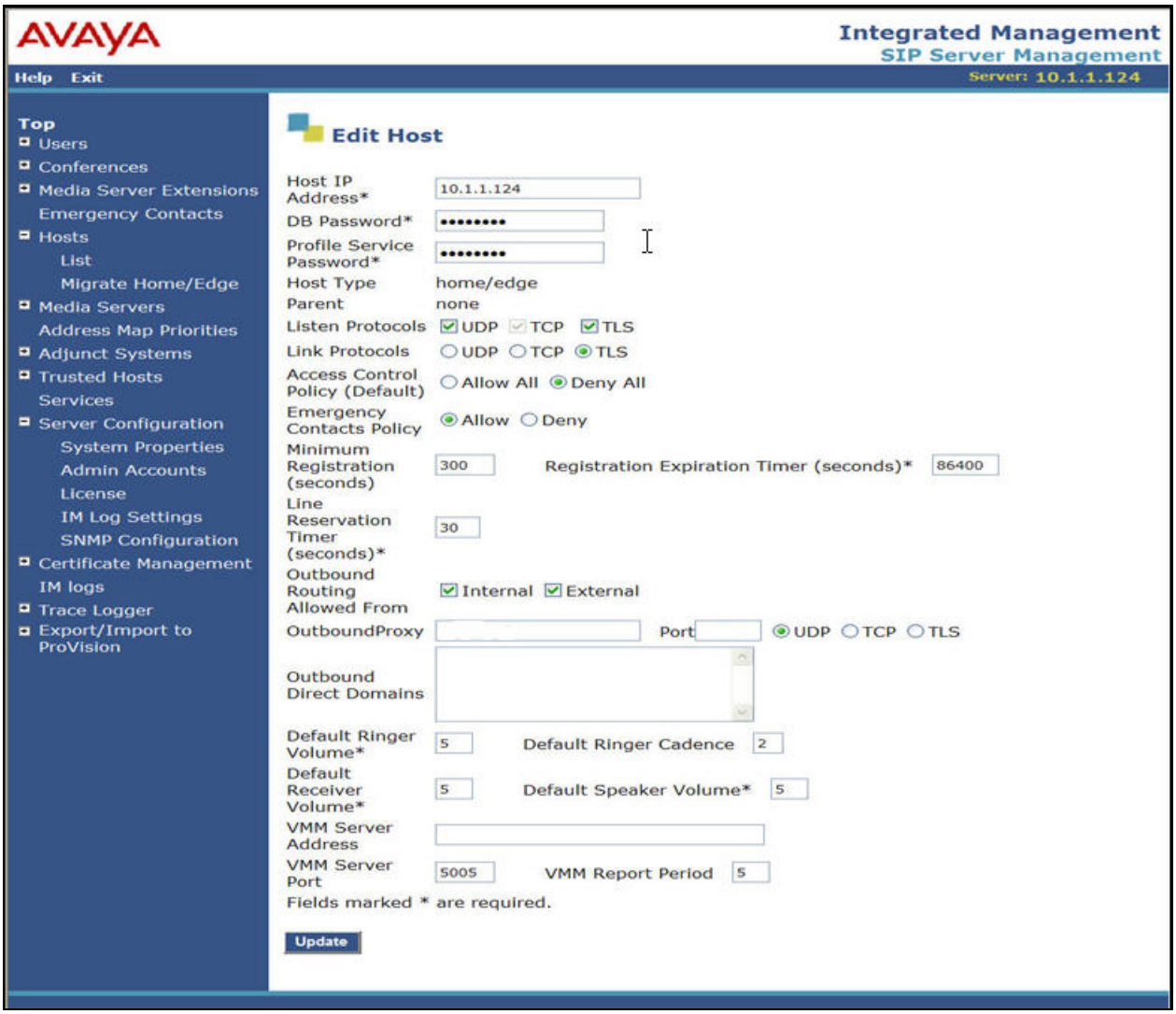

Figure 32: Edit Host

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Step 4: **Add Avaya Communication Manager as Media Server**

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

In the **Add Media Server** screen, enter the following information:

- A descriptive name in the **Media Server Interface** field.
- Select IP address of the home SES server in the **Host** field as specified in **Figure 32**.
- 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 CLAN board in the **SIP Trunk IP Address** field. (Note: This may be the IP address of the media server processor in smaller Avaya Communication Manager configurations such as an Avaya S8300 Server using an Avaya G350 Media Gateway.)
- Enter the IP address of a Communication Manager LAN interface for the **Media Server Admin Address** field and the admin login and password for the **Media Server Admin Login** and **Media Server Admin Password** fields.
- After completing the **Add Media Server** screen, click on the **Add** button.

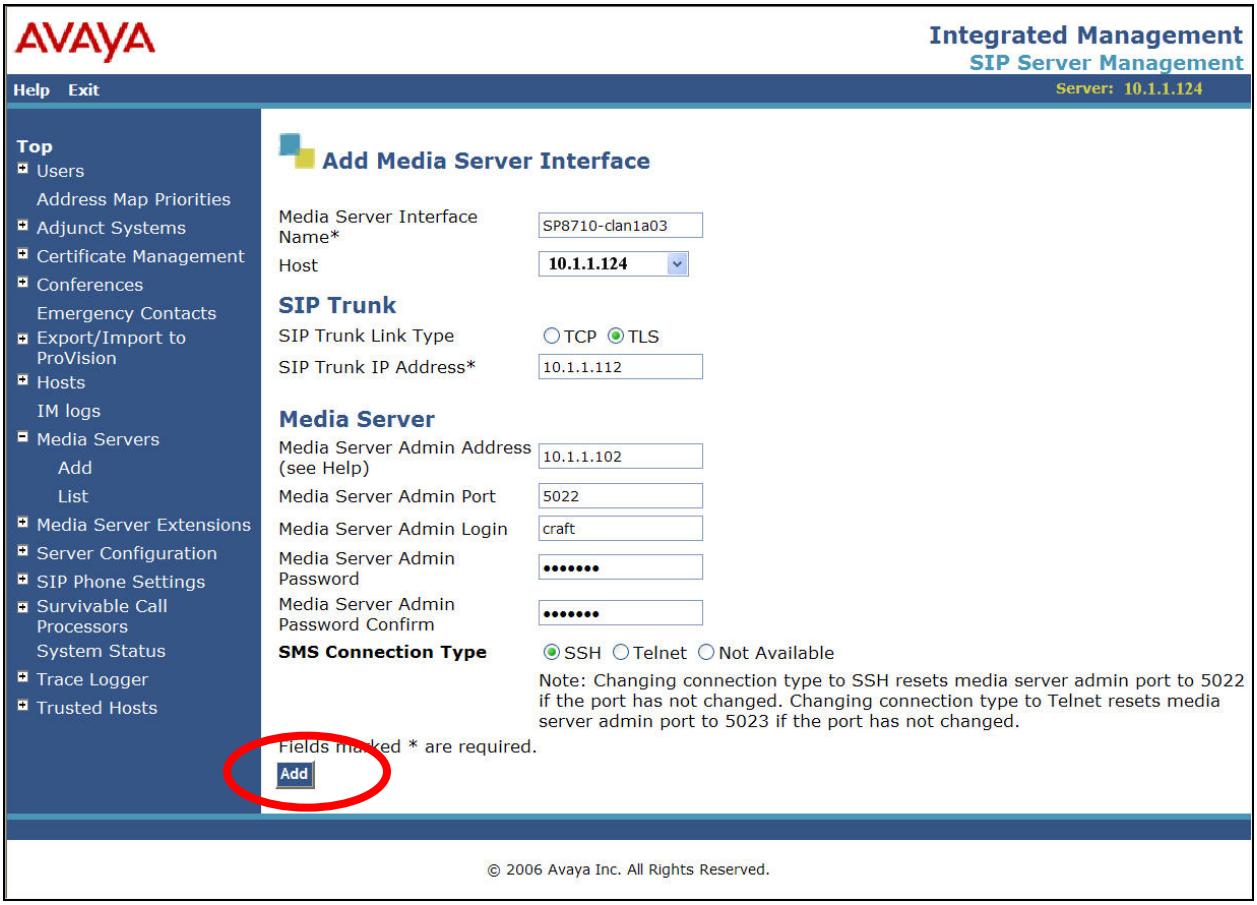

Figure 33: Add Media Server

Step 5: Specify Address Maps to Media Servers
Incoming calls arriving at Avaya SIP Enablement Services are routed to the appropriate Avaya Communication Manager for termination services. This routing is specified in a Media Server Address Map configured on Avaya SIP Enablement Services.

This routing compares the Uniform Resource Identifier (URI) of an incoming INVITE message to the pattern configured in the Media Server Address Map**,** and if there is a match, the call is routed to the designated Avaya Communication Manager. The URI usually takes the form of *sip:user@domain*, where *domain* can be a domain name or an IP address. Patterns must be specific enough to uniquely route incoming calls to the proper destination if there are multiple Avaya Communication Manager systems supported by the Avaya SES server.

In these Application Notes, only incoming calls from the PSTN require a media server address map entry. Calls originated by Avaya SIP telephones configured as OPS are automatically routed to the proper Avaya Communication Manager by the assignment of an Avaya Media Server extension to that phone. Address map definitions for SIP endpoints not assigned a media server extension and connections to multiple service providers are beyond the scope of these Application Notes.

For Global Crossing's SIP Trunking solution, the *user* portion of the SIP URI will contain the 10 digit value specified for the incoming direct inward dialed telephone number. An example of a SIP URI in an INVITE message received from Global Crossing would be:

sip:6023574046@10.1.1.124;user=phone;npdi=yes

The user portion in this case is the 10 digit DID number "6023574046". The strategy used to define the media server address maps will be to create a pattern that matches the DID numbers assigned to the customer by Global Crossing. The SES will forward the messages with matching patterns to the appropriate CLAN interface.

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.
- Click on the **Map** link associated with the appropriate media server to display the **List Media Server Address Map** screen.
- Click on the **Add Map In New Group** link. The screen shown in **Figure 34** is displayed.
- Enter a descriptive name in the **Name** field
- Enter the regular expression to be used for the pattern matching in the **Pattern** field. In this configuration, the DID numbers provided by Global Crossing are 602-357-4046 thru 4049. The pattern specification (without the double quotes) for DID numbers assigned is: "^sip:602357404[6789]". This means that URIs beginning with "sip: "602357404" followed by the digits 6, 7, 8 or 9 will match the pattern and be routed to the interface defined for S8710-CLAN. Appendix B provides a detailed description of the syntax for address map patterns.
- Click the **Add** button once the form is completed.

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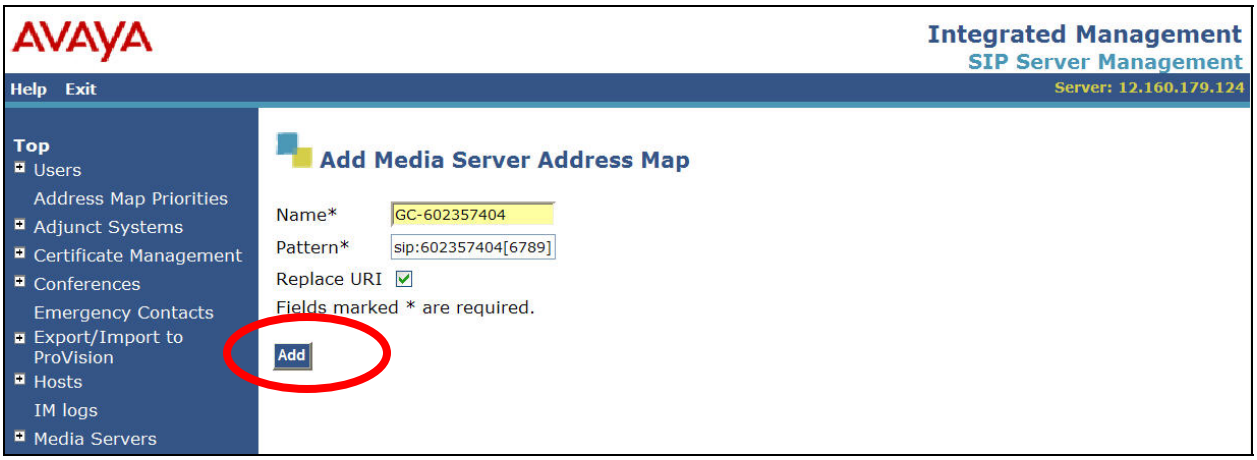

Figure 34: Media Server Address Map

After configuring the media server address map, the **List Media Server Address Map** screen appears as shown in **Figure 35**.

Figure 35: List Media Server Address Map

Note that after the first **Media Server Address Map** is added, the **Media Server Contact** is created automatically. For the **Media Server Address Map** added in **Figure 34**, the following contact was created:

sips:$(user)@10.1.1.112:5061;transport=tls

The contact specifies the IP address of the CLAN and the transport protocol used to send SIP signaling messages. The incoming DID number sent in the user part of the original request URI is substituted for $(user).

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Step 6: Specify the Global Crossing SIP Proxy as a Trusted Host

The final step to complete the SIP trunk administration on Avaya SES is to designate the IP address of the Global Crossing SIP Proxy as a trusted host. As a trusted host, Avaya SES will not issue SIP authentication challenges for incoming requests from the designated IP address.[1] If multiple SIP proxies are used, the IP address of each SIP proxy must be added as a trusted host.

To configure a trusted host:

- Select **Trusted Hosts** in the left pane of the Administration web interface. Then Select the **Add Trusted Hosts** option. The screen shown in **Figure 36** is displayed.
- Enter the IP address of the Global Crossing SIP Proxy in the **IP Address** field.
- Select the SES IP address from the **Host** drop down menu.
- Enter a description in the **Comment** field.

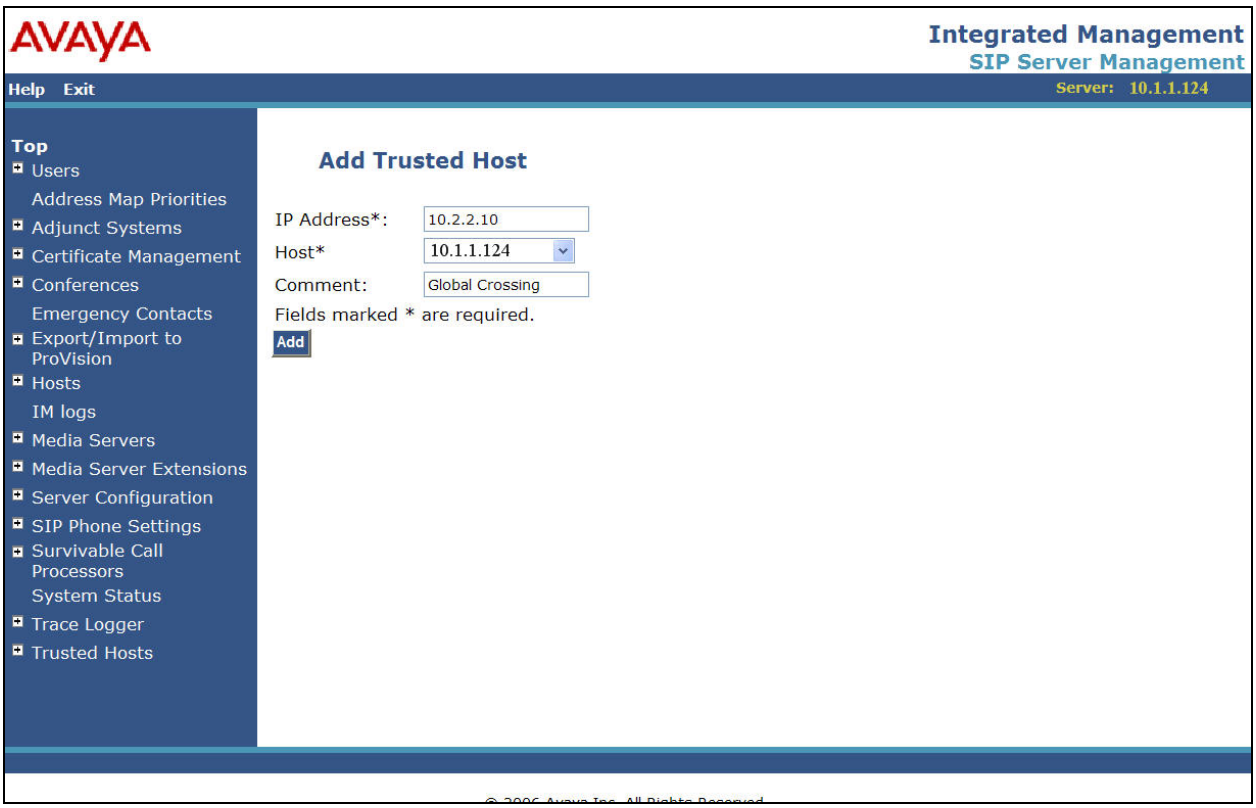

Figure 36: Configuring a Trusted Host

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

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4.3 Configuration for 9600-Series SIP Telephones

This section provides very basic instructions for completing the administration necessary to support the optional Avaya SIP telephones. Additional features such as the use of mnemonic addressing and instant messaging are also supported by Avaya SES but are beyond the scope of these Application Notes.

Step 1: Add a SIP User

Create the SIP user record as follows:

- In Avaya SES administration, expand the **Users** link in the left side blue navigation bar and click on the **Add** link.
- 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 will be used when logging into the user's SIP telephone.
- In the **Host** field, select the Avaya SIP Enablement Services server hosting the domain (*10.1.1.124*) for this user. Enter the **First Name** and **Last Name** of the user.
- 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.
- Press the **Add** button. This will cause a confirmation screen to appear.
- Press **Continue** on the confirmation screen.

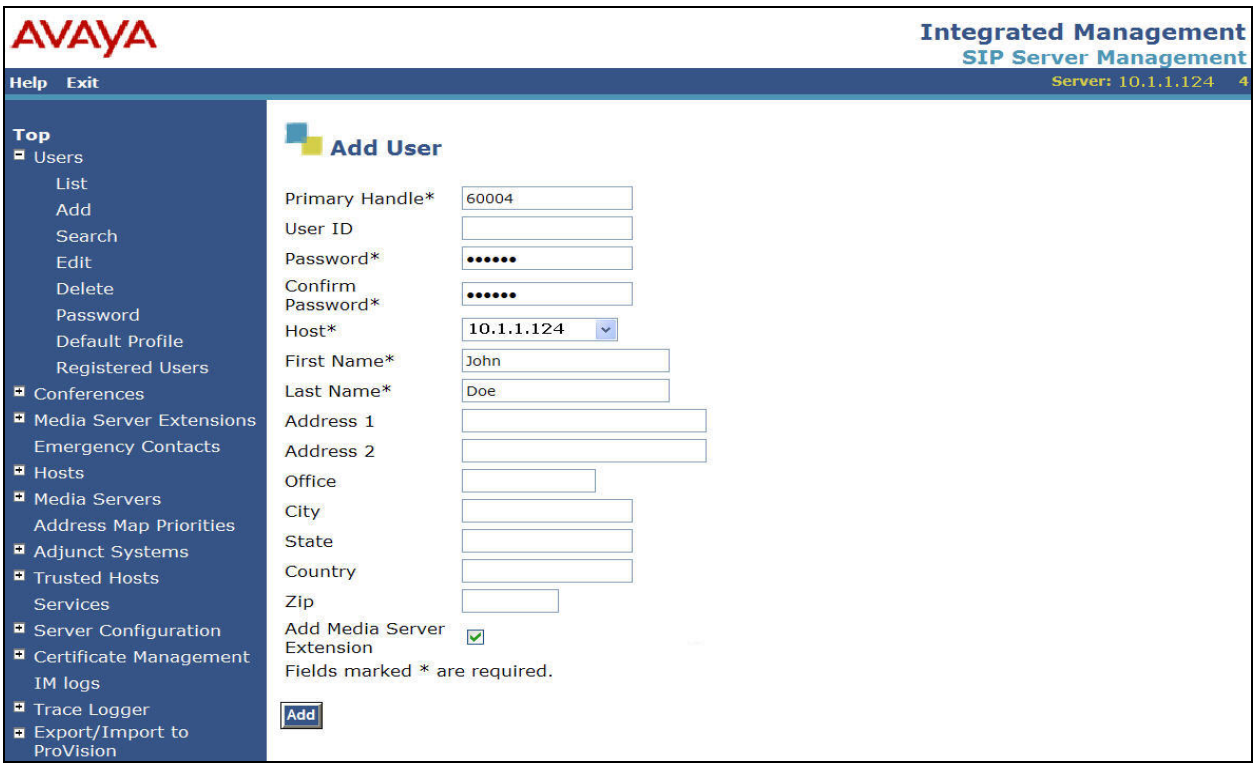

Figure 37: Add User

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Step 2: Specify Corresponding Avaya Communication Manager Extension
The SIP phone handle must now be associated with the corresponding extension on Avaya Communication Manager. After adding the User above, the **Add Media Server Extension** screen will appear.

- In the **Add Media Server Extension** screen, enter the **Extension** configured on the media server, shown in **Figure 24**, for the OPS extension on Avaya Communication Manager previously defined in **Section 3.2**. Usually, the media server extension and the user extension are the same (recommended) but it is not required.
- Select the **Media Server** assigned to this extension.
- Click on the **Add** button.

Figure 38: Add Media Server Extension

Step 3: Repeat for Each SIP User
Repeat Steps 1 and 2 for each SIP user.

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4.4 Configuration for 4600-Series SIP Telephones

The steps outlined in **Section 4.3** can be followed for the configuration of Avaya 4600-Series SIP telephones with the additional considerations described below.

The Avaya 4600-Series SIP phones do not support the G.729a codec type. Instead, they support the G.729b codec type. If the G.729a codec type is chosen for calls between Global Crossing and the Avaya IP telephony solution, a codec incompatibility will be created. This codec incompatibility presents a problem in a "shuffling" or Direct IP scenario where the phone is asked to send RTP directly with Global Crossing's network.

To work around this codec incompatibility, the Avaya 4600-Series SIP phone can be placed in a separate IP Network Region. By doing this, shuffling can then be disabled when these phones communicate with other IP network regions. Specific configuration details are beyond the scope of this document, please refer to [1] for more details regarding additional IP Network Region configuration.

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5. Global Crossing Services Configuration

Service / Feature Information	Test Configuration
Global Crossing Enterprise VoIP Service(s): • **DID** – IP Inbound to customer • **Toll Free** – IP Inbound to customer • **VTERM** – IP Outbound from customer • **On-Net –** Site to Site	All Global Crossing Enterprise VoIP Services were tested in the Application Notes.
Global Crossing Codec(s): • G.711mu-law • G.711a-law • G.729A • RFC2833 DTMF • T.38 FAX	The network configuration described in these Application Notes was tested with all the codecs (payload types) listed in the left column.
Global Crossing Dial Plan	10-Digit North American and International (011 Country Code) 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 602-357 were assigned to the H.323, SIP, digital, and analog endpoints in the enterprise network.
Global Crossing Proxy IP Address	The IP address of the Acme Packet Net-Net Session Director in the Global Crossing network was 10.2.2.10.

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Customer IP Address	The IP address of the Avaya SES server in the enterprise network was 10.1.1.124. Global Crossing used this IP address for routing calls destined to the listed directory numbers assigned to the enterprise site.
SIP Transport Protocol and Port	SIP signaling was transported between Avaya SES and Global Crossing using UDP and port 5060.

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6. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between Global Crossing's SIP Trunking Solution and an Avaya IP Telephony Solution. This section covers the general test approach and the test results.

6.1. General Test Approach

A simulated enterprise site consisting of an Avaya IP telephony solution supporting SIP trunking was connected to the public Internet using a dedicated broadband connection. The enterprise site was configured to use the commercially available SIP Trunking solution provided by Global Crossing. This allowed the enterprise site to use SIP trunking for PSTN calling.

The following features and functionality were covered during the SIP trunking interoperability compliance test:

- Incoming calls to the enterprise site from the PSTN were routed to the DID numbers assigned by Global Crossing.
- Outgoing calls from the enterprise site were completed via Global Crossing to the PSTN destinations.
- Site to site calls through Global Crossing's On-Net Service.
- Calls using SIP, H.323, digital and analog endpoints supported by the Avaya IP telephony solution.
- Various call types including: local, long distance, international, toll free, emergency, and directory assistance calls.
- Calls using G.729a, G.711mu and G.711a codec.
- T.38 Fax calls.
- DTMF tone transmission using RFC 2833 with successful Voice Mail/Vector navigation
- Telephone features such as hold, transfer, conference.
- Direct IP-to-IP media (also known as "shuffling") with SIP/H323 telephones.

6.2. Test Results

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

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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 place outbound and receive inbound PSTN calls through Global Crossing.

1. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains 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.

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 customers using an Avaya Communication Manager and Avaya SIP Enablement Services telephony solution to a Global Crossing SIP Trunking solution. Global Crossing's SIP Trunking solution is a robust Voice over IP solution for customers ranging from small businesses to large enterprises. SIP trunks use the Session Initiation Protocol (SIP) to connect private company networks to the public telephone network via converged IP access. A Global Crossing SIP Trunking solution provides businesses a flexible, cost-saving alternative to traditional hardwired telephony trunk lines.

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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*, January 2008, Issue 4.0, Document Number 03-300509.

[2] *Feature Description and Implementation for Avaya Communication Manager*, January 2008, Issue 6, Document Number 555-245-205

[3] *Avaya Extension to Cellular and Off-PBX Station (OPS) Installation and Administration Guide Release 4.0*, Feb 2007, Issue 10, Document Number 210-100-700.

[4] *Installing and Administering SIP Enablement Services*, May 2007, Issue 4, Document Number 03-600768

[5] *SIP Support in Avaya Communication Manager Running on the Avaya S8300, S8400, S8500 series, and S8700 series Media Server*, January 2008, Document Number 555-245-206.

[6] *4600 Series IP Telephone LAN Administrator Guide*, October 2007, Document Number 555-233-507

[7] *Avaya one-X Deskphone Edition for 9600 Series IP Telephones Administrator Guide Release 2.0*, May 2008, Document Number 16-300698

[8] *Avaya one-X Deskphone SIP for 9600 Series IP Telephones Administrator Guide Release 2.0*, December 2007, 16-601944

[9] *Avaya Communication Manager Network Region Configuration Guide*, October 2005, 103244

The following RFCs are available at http://www.ietf.org/

[10] RFC 3261 *SIP: Session Initiation Protocol*

[11] RFC 2833 *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*

[12] RFC 4244, *An Extension to the Session Initiation Protocol (SIP) for Request History Information*

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APPENDIX A: Sample SIP INVITE Messages

This section displays the format of the SIP INVITE messages sent by Global Crossing 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 to Avaya SIP Enablement Services:

```
INVITE sip:+16023574045;npdi=yes@10.1.1.124:5060 SIP/2.0
Via: SIP/2.0/UDP 10.2.2.10:5060;branch=z9hG4bKvt4gjd207841ob0ed0c1.1
From: <sip:+17328521637@10.2.2.10>;tag=SD4bvc801-gK067e87f8
To: <sip:+16023574045@10.1.1.124>
Call-ID: SD4bvc801-332829c692d42ff31f87bd9b4b7e2caa-v3000i1
CSeq: 7652 INVITE
Max-Forwards: 69
Allow:INVITE,ACK,CANCEL,BYE,REGISTER,INFO,SUBSCRIBE,NOTIFY,PRACK,UPDATE,OPTI
ONS
Accept: application/sdp, application/isup, application/dtmf, application/dtmf-relay,  multipart/mixed
Contact: <sip:+17328521637@10.2.2.10:5060;transport=udp>
Remote-Party-ID: <sip:+17328521637@10.2.2.10:5060>;privacy=off
Supported: timer
Session-Expires: 64800
Min-SE: 64800
Content-Length: 260
Content-Disposition: session; handling=required
Content-Type: application/sdp

Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): Sonus_UAC 26697 19451 IN IP4 10.2.2.10
Session Name (s): SIP Media Capabilities
Connection Information (c): IN IP4 10.2.2.10
Time Description, active time (t): 0 0
Media Description, name and address (m): audio 50004 RTP/AVP 18 0 100
Media Attribute (a): rtpmap:18 G729/8000
Media Attribute (a): rtpmap:0 PCMU/8000
Media Attribute (a): rtpmap:100 telephone-event/8000
Media Attribute (a): fmtp:100 0-15
Media Attribute (a): sendrecv
Media Attribute (a): maxptime:20
```


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

```
INVITE sip:17328521637@10.2.2.10 SIP/2.0
Accept-Language: en
Call-ID: 0c8fb7d4314dd1f648158f7400
CSeq: 1 INVITE
From: "Ima IP4610" <sip:7133433752@east.devcon.com:5061>;tag=0c8fb7d4314dd1e648158f7400
Record-Route: <sip:10.1.1.124:5060;lr>,<sip:12.160.179.112:5061;lr;transport=tls>
To: "17328521637" <sip:17328521637@10.2.2.10>
Via: SIP/2.0/UDP 10.1.1.124:5060;branch=z9hG4bK03030336363683838329ec.0,SIP/2.0/TLS
12.160.179.112;psrrposn=2;received=12.160.179.112;branch=z9hG4bK0c8fb7d4314dd110648158f7400
Content-Length: 214
Content-Type: application/sdp
Contact: "Ima IP4610" <sip:7133433752@12.160.179.112:5061;transport=tls>
Max-Forwards: 69
User-Agent: Avaya CM/R015x.00.0.825.4
Allow:INVITE,CANCEL,BYE,ACK,PRACK,SUBSCRIBE,NOTIFY,REFER,OPTIONS,INFO,PUBLIS
H
Supported: 100rel,timer,replaces,join,histinfo
Alert-Info: <cid:internal@10.2.2.10>;avaya-cm-alert-type=internal
Min-SE: 600
Session-Expires: 600;refresher=uac
P-Asserted-Identity: "Ima IP4610" <sip:7133433752@east.devcon.com:5061>
History-Info: <sip:17328521637@10.2.2.10>;index=1,"17328521637"
<sip:17328521637@10.2.2.10>;index=1.1

Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): - 1 1 IN IP4 12.160.179.112
Session Name (s): -
Connection Information (c): IN IP4 12.160.179.114
Bandwidth Information (b): AS:64
Time Description, active time (t): 0 0
Media Description, name and address (m): audio 6676 RTP/AVP 18 0 101
Media Attribute (a): rtpmap:18 G729/8000
Media Attribute (a): fmtp:18 annexb=no
Media Attribute (a): rtpmap:0 PCMU/8000
Media Attribute (a): rtpmap:101 telephone-event/8000
```


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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 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 1+ 10 digit number 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@proxy-Global Crossing:5060;transport=udp SIP/2.0

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