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Avaya Solution & Interoperability Test Lab

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

Abstract

These Application Notes describe the steps for configuring SIP trunking between the Global Crossing VoIP Network and an Avaya IP Telephony Network consisting of Avaya SIP Enablement Services, Avaya Communication Manager, and Avaya SIP, H.323, digital and analog endpoints. Enterprise customers with an Avaya SIP-based network can connect to the Global Crossing VoIP Network over the Internet and access the PSTN by subscribing to Global Crossing Enterprise VoIP Services: VoIP Outbound, VoIP Toll Free, and VoIP Local Service. The Global Crossing VoIP Network and the Avaya SIP Network communicate over the Internet using the Session Initiation Protocol (SIP). This solution allows enterprise customers with a converged network to reduce long distance and interconnection costs. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the Developer*Connection* Service Provider Program at the Avaya Solution and Interoperability Test Lab.

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GlobalCrossing.doc

1. Introduction

These Application Notes describe the steps for configuring SIP trunking between the Global Crossing VoIP Network and an Avaya IP Telephony Network consisting of Avaya SIP Enablement Services[1], Avaya Communication Manager, and Avaya SIP, H.323, digital and analog endpoints. Enterprise customers with an Avaya SIP-based network can connect to the Global Crossing VoIP Network over the Internet and access the public switched telephone network (PSTN) by subscribing to Global Crossing Enterprise VoIP Services: VoIP Outbound, VoIP Toll Free, and VoIP Local Service. The Global Crossing VoIP Network and the Avaya SIP Network communicate over the Internet using the Session Initiation Protocol (SIP). This solution allows enterprise customers with a converged network to reduce long distance and interconnection costs.

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

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

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

Figure 1 illustrates an enterprise site with an Avaya SIP-based network, including Avaya SIP Enablement Services, an Avaya S8300 Media Server with a G350 Media Gateway[2] running Avaya Communication Manager, Avaya Intuity Audix 770 integrated in the Avaya S8300 Media Server, and Avaya SIP, H.323, digital, and analog endpoints. The enterprise site is connected to the Global Crossing VoIP Network over the Internet and communicates using SIP. The Global Crossing VoIP Network consists of an Acme Packet Net-Net Session Director, a Sonus Networks PSX Softswitch, and a Sonus Networks GSX Gateway. The Net-Net Session Director is a session border controller, which sends and receives all SIP signaling messages to/from Avaya SIP Enablement Services. Although not included in **Figure 1**, the enterprise site could also have an alternate route to the PSTN using traditional trunks.

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[1] In release 2.1 or earlier, the product was referred to as Avaya Converged Communications Server.
[2] This solution is compatible with other Avaya Media Server and Media Gateway platforms running Avaya Communication Manager.

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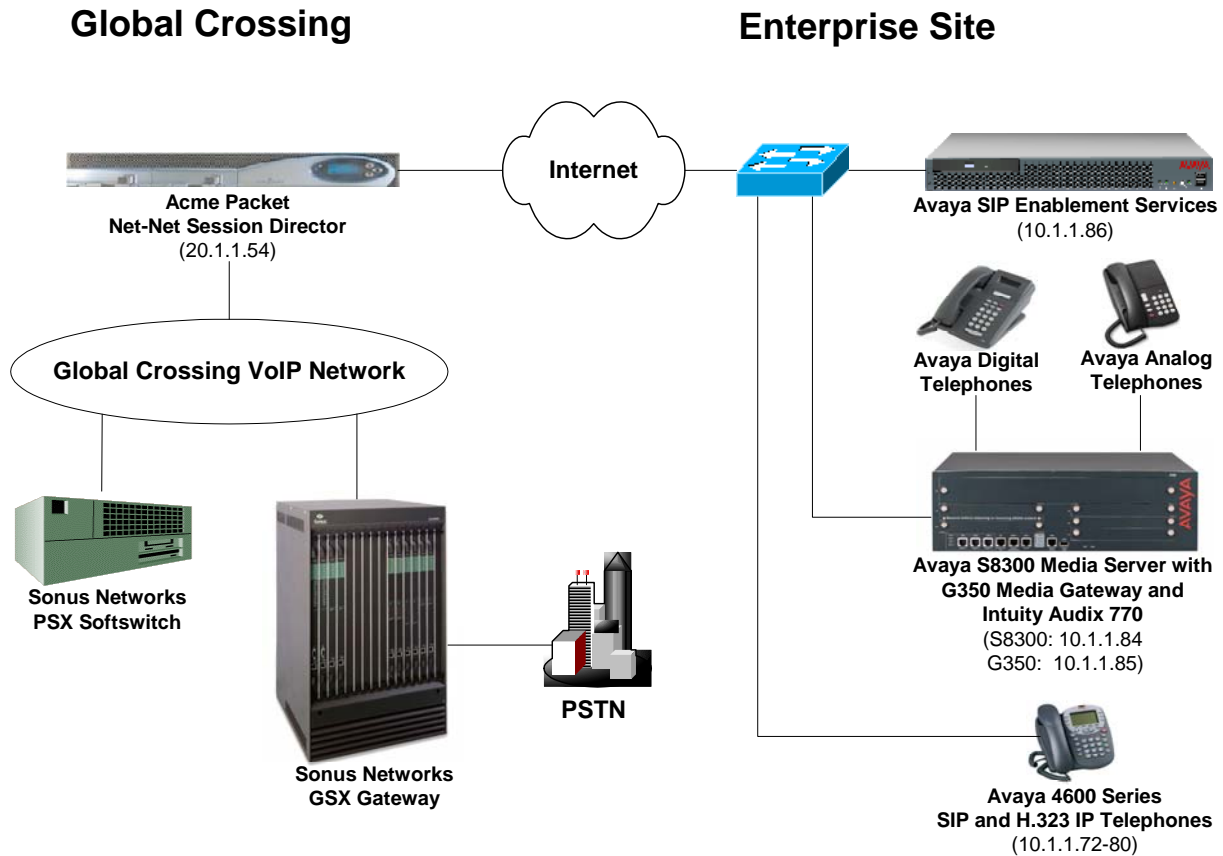

Figure 1: Avaya IP Telephony Network connected to the Global Crossing VoIP Network

1.1. SIP Call Flows

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

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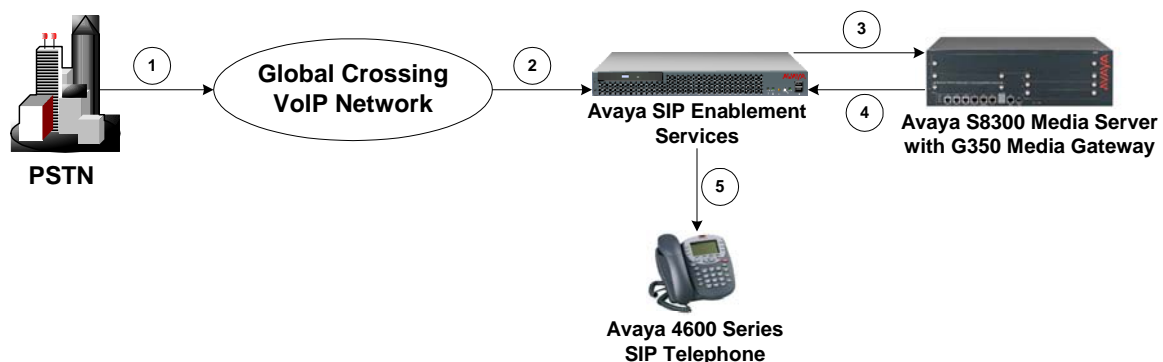5. Avaya SIP Enablement Services terminates the call to the Avaya SIP telephone.

Figure 2: PSTN Call to the Avaya SIP Network

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

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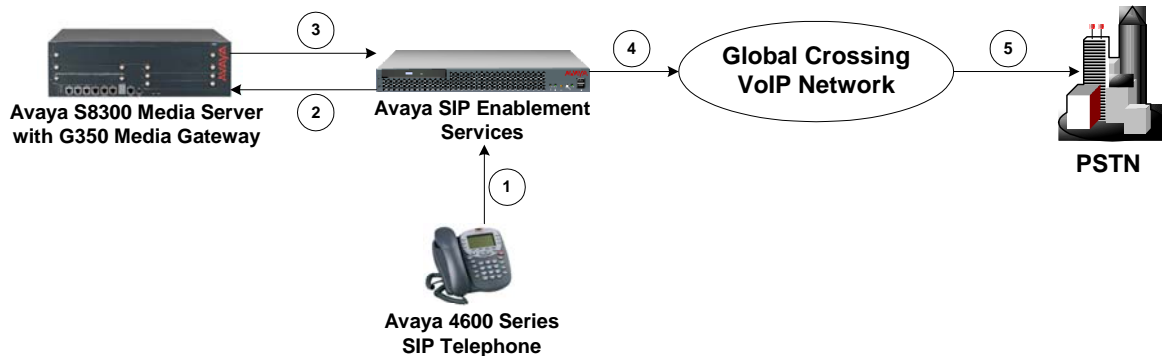5. Global Crossing routes the call to the PSTN.

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

2. Equipment and Software Validated

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

Hardware Component	Version
Avaya S8300 Media Server with a G350 Media Gateway and Intuity Audix 770 (IA770)	Communication Manager 3.0 (R013x.00.0.340.3)
Avaya SIP Enablement Services	3.0.0.0-31.0
Avaya 4600 Series SIP Telephone	2.2
Avaya SIP Softphone	Release 2 Build 43
Avaya 4600 Series H.323 IP Telephones	2.1
Acme Packet Net-Net Session Director	1.2.1p35
Sonus Networks PSX Softswitch	6.07
Sonus Networks GSX Gateway	6.0.9

3. Configure Avaya Communication Manager

This section describes the steps for configuring a SIP trunk and off-PBX stations[3] (OPS) on Avaya Communication Manager. The SIP trunk is established between Avaya Communication Manager and Avaya SIP Enablement Services. An off-PBX station (OPS) is configured for each Avaya 4600 Series SIP Telephone registered with Avaya SIP Enablement Services. Refer to [2] for additional information on configuring an off-PBX station. All incoming calls from Global Crossing are received by Avaya SIP Enablement Services and routed to Avaya Communication Manager over a SIP trunk for termination services. All outgoing calls to the PSTN are routed through Avaya Communication Manager for origination services. Avaya Communication Manager then routes the call to Avaya SIP Enablement Services, which in turn routes the call to the PSTN through the Global Crossing VoIP network. Note that Avaya SIP Enablement Services provides the SIP interface to the Global Crossing VoIP Network.

The dial plan for the configuration described in these Application Notes consisted of 10-digit dialing for local and long-distance calls over the PSTN. In addition, Operator calls (0), Directory Assistance calls (411) and International calls (011 Country Code) were also supported. Avaya Communication Manager routed all calls using Auto Route Selection (ARS), except for intra-switch calls. Configuring ARS is beyond the scope of these Application Notes and the reader should refer to [1] for additional information.

Avaya Communication Manager configuration was performed using the System Access Terminal (SAT). The IP network properties of the Avaya S8300 Media Server were configured via its Maintenance web interface using an Internet browser (not shown here). Using the SAT, verify that the Off-PBX Telephones (OPS) and SIP Trunks features are enabled on the **System-Parameters Customer-Options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative. On Page 1, verify that the number of OPS stations allowed in the system is sufficient.

```
display system-parameters customer-options                    Page   1 of  10
                            OPTIONAL FEATURES
    G3 Version: V13
      Location: 1                            RFA System ID (SID): 1
      Platform: 13                           RFA Module ID (MID): 1

                                                      USED
                          Platform Maximum Ports: 900    36
                                Maximum Stations: 40     6
                        Maximum XMOBILE Stations: 0      0
               Maximum Off-PBX Telephones - EC500: 50    0
               Maximum Off-PBX Telephones -   OPS: 50    3
               Maximum Off-PBX Telephones - SCCAN: 0     0
      (NOTE: You must logoff & login to effect the permission changes.)
```

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

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

On Page 2 of the **System-Parameters Customer-Options** form, verify that the number of SIP trunks supported by the system is sufficient.

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

IP PORT CAPACITIES                                                  USED
                     Maximum Administered H.323 Trunks: 100   6
            Maximum Concurrently Registered IP Stations: 40    1
               Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0      0
              Maximum Concurrently Registered IP eCons: 0      0
  Max Concur Registered Unauthenticated H.323 Stations: 0      0
                   Maximum Video Capable H.323 Stations: 0     0
                   Maximum Video Capable IP Softphones: 0      0
                      Maximum Administered SIP Trunks: 100     24

   Maximum Number of DS1 Boards with Echo Cancellation: 0      0
                           Maximum TN2501 VAL Boards: 0        0
                    Maximum G250/G350/G700 VAL Sources: 1      0
              Maximum TN2602 Boards with 80 VoIP Channels: 0   0
             Maximum TN2602 Boards with 320 VoIP Channels: 0   0
      Maximum Number of Expanded Meet-me Conference Ports: 0    0

        (NOTE: You must logoff & login to effect the permission changes.)
```

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

On the **System-Parameters Features** form, set the **Trunk-to-Trunk Transfer** field to *all* to allow calls to be transferred from the enterprise site to an endpoint on the PSTN.   Otherwise, leave the field set to *none*. The SIP call flows described in Section 1.1 did not require trunk-to-trunk transfer to be enabled.

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

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

Figure 6: System-Parameters Features Form

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

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

Figure 7: IP Nodes Names Form

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Avaya SIP Enablement Services. In this configuration, the domain name is *devcon.com*. This field is required for endpoints to call the public network. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between SIP endpoints without using media resources in the Avaya G350 Media Gateway. For calls involving Global Crossing, shuffling is only activated when the call originates or terminates on an Avaya 4600 Series SIP Telephone. In addition, DTMF transmission using RFC 2833 is also required for shuffling among SIP devices as shown in **Figure 10**. The **IP Network Region** form also specifies the **IP Codec Set** to be used for local calls and calls routed over the SIP trunk to Avaya SIP Enablement Services. This codec set is used when its corresponding network region (i.e., IP Network Region '1') is specified in the SIP signaling group as shown in **Figure 10**.

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

Figure 8: IP Network Region Form

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

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

                         IP Codec Set

    Codec Set: 1

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

Figure 9: IP Codec Set Form

Prior to configuring a SIP trunk group for communication with Avaya SIP Enablement Services, a SIP signaling group must be configured. Configure the **Signaling Group** form shown in **Figure 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.
- Specify the Avaya S8300 Media Server and the Avaya SIP Enablement Services Server as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form shown in **Figure 7**. For other media server platforms, the near (local) end of the SIP signaling group may be a C-LAN board instead of the media server.
- Ensure that the recommended TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields. If the **Far-end Network Region** field is different from the near-end network region, the preferred codec will be selected from the IP codec set assigned for the inter-region connectivity for the pair of network regions. Although the same network region (Network Region 1) was used for local and PSTN calls in this configuration, different network regions can be used in the field.
- Enter the domain name of Avaya SIP Enablement Services in the **Far-end Domain** field. In this configuration, the domain name is *devcon.com*. This domain is specified in the Uniform Resource Identifier (URI) of the "SIP To Address" in the INVITE message. Mis-configuring this field may prevent calls from being successfully established to other SIP endpoints or to the PSTN.

- If calls to/from SIP endpoints are to be shuffled, then the **Direct IP-IP Audio Connections** field must be set to 'y'.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*. Avaya Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

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

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


   Near-end Node Name: procr                 Far-end Node Name: CCS
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1
        Far-end Domain: devcon.com


                                         Bypass If IP Threshold Exceeded? n

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

Figure 10: Signaling Group Form

Configure the **Trunk Group** form as shown in **Figure 11**. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. For a call between the PSTN and a SIP endpoint, two trunk members are used for the duration of the call. For a call between the PSTN and a non-SIP endpoint, one trunk member is used for the duration of the call. Configure the other fields in bold and accept the default values for the remaining fields.

```
add trunk-group 1                                             Page   1 of  19
                              TRUNK GROUP

Group Number: 1                    Group Type: sip          CDR Reports: y
  Group Name: To CCS                        COR: 1     TN: 1        TAC: 101
    Direction: two-way        Outgoing Display? n
 Dial Access? n               Busy Threshold: 255      Night Service:
Queue Length: 0
Service Type: tie                 Auth Code? n
                                                        Signaling Group: 1
                                                      Number of Members: 24
TRUNK PARAMETERS

     Unicode Name? y


                                       Redirect On OPTIM Failure: 5000

          SCCAN? n                                    Digital Loss Group: 18
```

Figure 11: Trunk Group Form – Page 1

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

```
add trunk-group 1                                           Page   2 of  19
TRUNK FEATURES
         ACA Assignment? n              Measured: none
                                                      Maintenance Tests? y


                  Numbering Format: public

                                          Replace Unavailable Numbers? n
```

Figure 12: Trunk Group Form – Page 2

Configure the **Public/Unknown Numbering Format** form to send the calling party number to the far-end. Add an entry so that local stations with a 5-digit extension beginning with '7' and whose calls are routed over SIP trunk group '1' have the number sent to the far-end for display purposes. In the example shown in **Figure 13**, a CPN prefix is prepended to the 5-digit extension so that a 10-digit calling party number (e.g., extension 70000 is converted to 7327770000) is sent to the far-end.

```
change public-unknown-numbering 0                            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  7      1        73277          10
```

Figure 13: Public Unknown Format Form

The first step in configuring an off-PBX station (OPS) for the Avaya SIP telephones registered with Avaya SIP Enablement Services is to add an Admin WithOut Hardware (AWOH) station as shown in **Figure 14**. The recommended station type is 6408D+. For an AWOH station, the **Port** field is set to 'X' to indicate that there is no hardware associated with the port assignment. The Class of Restrictions (COR) and Class of Service (COS) assigned to the station should be configured with the appropriate call restrictions. The station is also configured with voicemail coverage using the Intuity Audix 770, an option integrated into the Avaya S8300 Media Server. For additional information on configuring Intuity Audix 770, refer to [4]. Repeat this step for each SIP endpoint at the enterprise site.

```
add station 70000                                           Page   1 of   4
                               STATION

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

                                         Media Complex Ext:
                                             IP SoftPhone? n
```

Figure 14: Admin Without Hardware (AWOH) Station – Page 1

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

```
add station 70000                                           Page   2 of   4
                               STATION
FEATURE OPTIONS
          LWC Reception: spe          Auto Select Any Idle Appearance? n
         LWC Activation? y                    Coverage Msg Retrieval? y
 LWC Log External Calls? n                              Auto Answer: none
            CDR Privacy? n                         Data Restriction? n
    Redirect Notification? y            Idle Appearance Preference? n
 Per Button Ring Control? n            Bridged Idle Line Preference? n
   Bridged Call Alerting? n                Restrict Last Appearance? y
 Active Station Ringing: single      Conf/Trans on Primary Appearance? n

       H.320 Conversion? n       Per Station CPN - Send Calling Number? y
     Service Link Mode: as-needed
      Multimedia Mode: enhanced
  MWI Served User Type: qsig-mwi              Display Client Redirection? n
                                        Select Last Used Appearance? n
                                         Coverage After Forwarding? s


 Remote Softphone Emergency Calls: as-on-local Direct IP-IP Audio Connections? y
 Emergency Location Ext: 70000     Always Use? n      IP Audio Hairpinning? y
```

Figure 15: Admin Without Hardware (AWOH) Station – Page 2

On Page 3 of the station form, configure the appropriate number of call appearances for the SIP telephone. For example, the Avaya 4610 SIP Telephone was configured to support three call appearances as shown in **Figure 16**.

```
add station 70000                                          Page   3 of   4
                              STATION
 SITE DATA
      Room:                                      Headset? n
      Jack:                                      Speaker? n
     Cable:                                      Mounting: d
     Floor:                                   Cord Length: 0
   Building:                                     Set Color:


ABBREVIATED DIALING
    List1:                    List2:                    List3:

BUTTON ASSIGNMENTS
 1: call-appr                        5:
 2: call-appr                        6:
 3: call-appr                        7:
 4:                                  8:
```

Figure 16: Admin Without Station (AWOH) Station – Page 3

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

```
change off-pbx-telephone station-mapping 70000           Page   1 of   2
                STATIONS WITH OFF-PBX TELEPHONE INTEGRATION


 Station       Application   Dial  Phone Number    Trunk      Configuration
 Extension                   Prefix                Selection  Set
 70000         OPS            - 70000              1          1
                               -
```

Figure 17: Stations with Off-PBX Telephone Integration – Page 1

On Page 2, set the **Call Limit** field to the maximum number of calls that may be active simultaneously at the station. In this example, the call limit is set to '3', which corresponds to the number of call appearances configured on the station form. Accept the default values for the other fields.
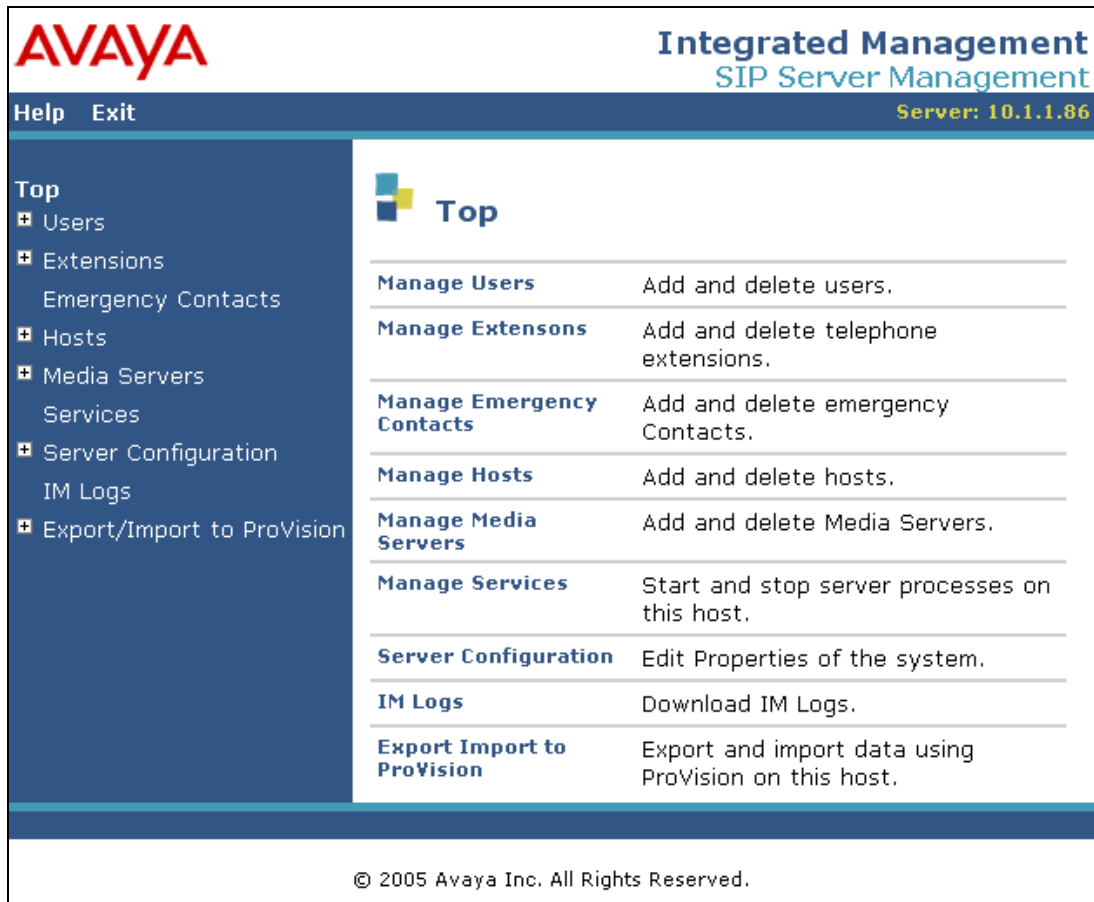
```
change off-pbx-telephone station-mapping 70000               Page   2 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Call        Mapping      Calls       Bridged
 Extension     Limit       Mode         Allowed     Calls
 70000           3         both         all         both
```

Figure 18: Stations with Off-PBX Telephone Integration – Page 2

4. Configure Avaya SIP Enablement Services

This section covers the administration of Avaya SIP Enablement Services (SES). Avaya SIP Enablement Services is configured via an Internet browser using the Administration web interface. To access the Administration web interface, enter http://<ip-addr>/admin as the URL in an Internet browser, where *<ip-addr>* is the IP address of Avaya SIP Enablement Services. Log in with the appropriate credentials and then select the *Launch Administration Web Interface* link from the **Interface** screen. The main screen shown in **Figure 19** is displayed. After configuring the system, click on the **Update** link that will appear in the left pane for the changes to take effect. See **Figure 20**.

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

Figure 19: SES Main Screen

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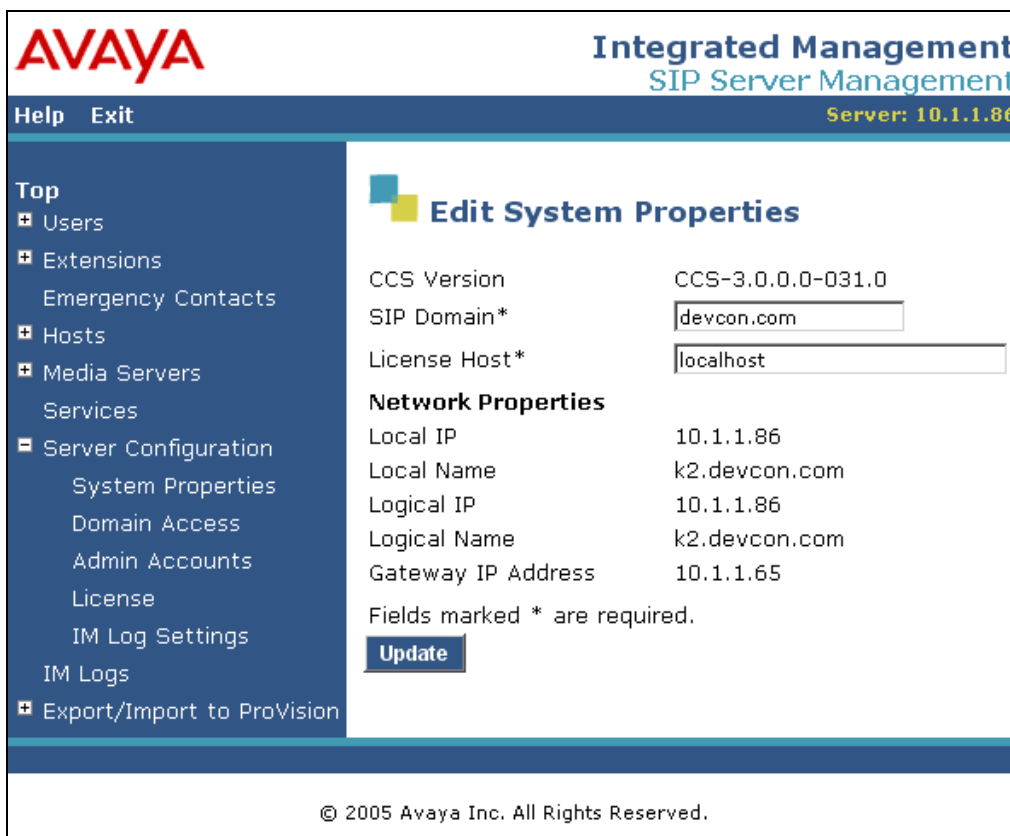

Figure 20: System Properties

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

The default values for the other fields may be used as shown in Figure 21. Click the **Update** button.

Figure 21: Host

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

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

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

Figure 22: Add Media Server

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

To configure a **Media Server Address Map**, select **Media Servers** in the left pane of the Administration web interface. This will display the **List Media Servers** screen. Click on the **Map** link associated with the appropriate media server to display the **List Media Server Address Map** screen and click on the **Add Map In New Group** link. The screen shown in **Figure 23** is displayed. The Host field will be displayed with the name of the media server. Provide a descriptive name in the **Name** field and enter the regular expression to be used for the pattern matching in the **Pattern** field. In this configuration, the pattern specification matches a URI that begins with `sip:602555` followed by four digits. Note that DID numbers beginning with 602555 were assigned to endpoints at the enterprise site. See Appendix B for a more detailed description of the syntax for address map patterns. Click the **Add** button.
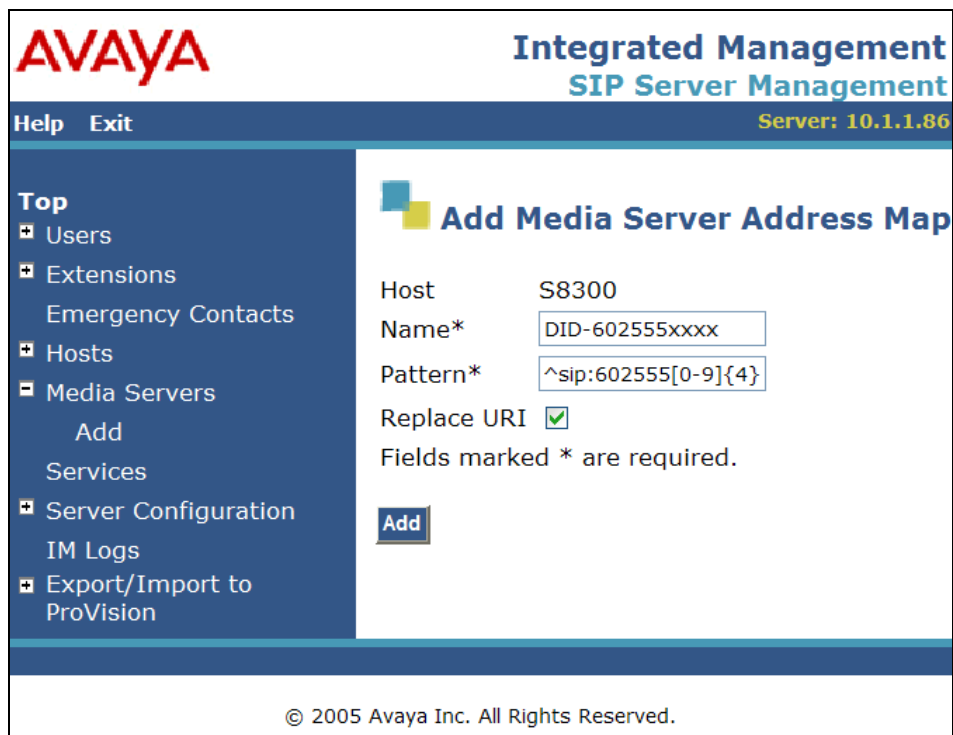
Figure 23: Media Server Address Map

After the **Media Server Address Map** is added, the first **Media Server Contact** is created automatically. For the **Media Server Address Map** added in **Figure 23**, the following contact was created:

```
sips:$(user)@10.1.1.84:5061;transport=tls
```

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

After configuring the media server address map, the **List Media Server Address Map** screen appears as shown in
Figure 24.

Figure 24: List Media Server Address Map

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All calls destined for the PSTN are routed through Avaya SIP Enablement Services using host address maps. In this configuration, host address maps for the following call types are created. These call types include: calls to area code 732, directory assistance calls, international calls, operator calls, and toll free calls.

As an example, the host address map for calls to area code 732 is shown in **Figure 25**. To access the **Add Host Address Map** screen, select the **Hosts** link in the left pane of the Administration web interface and then click on the **Map** link associated with the appropriate host (e.g., k2.devcon.com). The **List Host Address Map** screen is displayed. From this screen, click the **Add Map In New Group** link to display the screen shown in **Figure 25**. Configure a descriptive name for the map and specify an appropriate pattern for the call type. In this example, the pattern is used for calls to area code 732. By default, the **Replace URI** checkbox is selected. Click the **Add** button.

Figure 25: Edit Host Map Entry

From the **List Host Address Map**, click on the **Add Another Contact** link associated with the address map added in **Figure 25**. In this screen, the **Contact** field specifies the destination for the call and it is configured as:

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

The contact specifies the IP address of the Acme Net-Net Session Director in the Global Crossing VoIP Network and the transport protocol used to send SIP signaling messages. The

transport protocol must be coordinated with Global Crossing. The user in the original request URI is substituted for `$(user)`. Click the **Add** button when completed.

Figure 26: Add Host Contact

After configuring the host address maps, the **List Host Address Map** screen appears as shown in **Figure** 27.
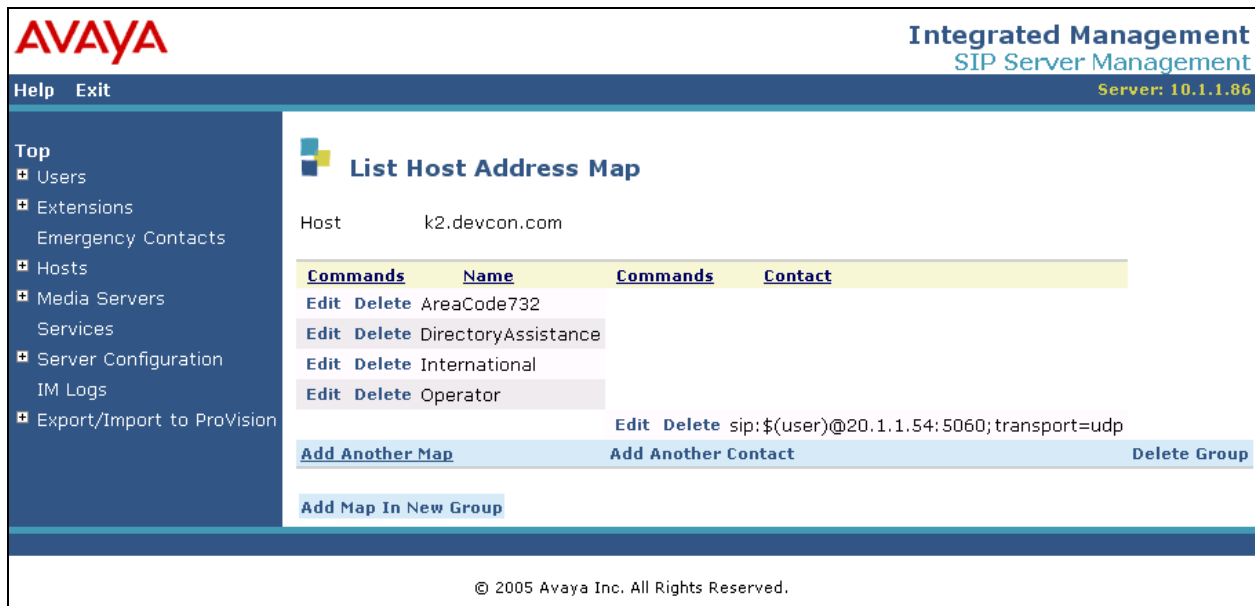
Figure 27:List Host Address Map

Add a user for each Avaya 4600 Series SIP Telephone registering with Avaya SIP Enablement Services. In the **Add User** screen, enter the extension of the SIP endpoint in the **Primary Handle** field. Enter a user password in the **Password** and **Confirm Password** fields. This password must match the one configured on the SIP telephone. The password entered in these two fields must match. In the **Host** field, select the Avaya SIP Enablement Services server hosting the domain (*devcon.com*) for this user. Enter the 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. The **Add Media Server Extension** screen shown in **Figure 29** will be displayed after adding this user's profile by clicking on the **Add** button.

Figure 28: Add User

In the **Add Media Server Extension** screen, enter the **Extension** configured on the media server, shown in **Figure 14**, for the previously added user. Usually, the media server extension and the user extension are the same (recommended). Select the **Media Server** assigned to this extension. Click on the **Add** button. To commit the configuration changes, click on the **Update** link in the left pane.

Figure 29: Add Media Server Extension

The last step is to configure the Acme Packet Net-Net Session Director as a trusted host on Avaya SIP Enablement Services. Note that the Acme Packet Net-Net Session Director sends all SIP signaling messages from the Global Crossing VoIP Network. Global Crossing provides the Session Director as the SIP proxy interface. As a trusted host, Avaya SIP Enablement Services will not challenge incoming requests from the Acme Packet Net-Net Session Director. For example, with this trusted host configuration, Avaya SIP Enablement Services will not challenge the BYE SIP message sent by Global Crossing to disconnect the call. This is required because Global Crossing will not respond to the challenge resulting in the call not terminating immediately on the Avaya network side. Instead the call will be dropped after the expiration of a timeout period. If multiple SIP proxies are used, then each SIP proxy must be added as a trusted host. To configure a trusted host, use the trustedhost command in the Linux shell of Avaya SIP Enablement Services. As shown in **Figure 30**, the trustedhost -a command adds the Acme Packet Net-Net Session Director with IP address 20.1.1.54 as a trusted host of Avaya SIP Enablement Services with a host name of k2.devcon.com. The trustedhost -L command allows the list of trusted hosts to be displayed and the trustedhost -d command allows a trusted host to be deleted from the system. After configuring a trusted host, the user must go to the Administration web interface and click on the **Update** link for the changes to take effect. This is required even though the trusted host was configured via the Linux shell.

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

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

Figure 30: Configuring a Trusted Host

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

5. Global Crossing VoIP Network Configuration

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

Feature	Test Configuration
Select Global Crossing Enterprise VoIP Service(s): ▪ VoIP Outbound ▪ VoIP Toll Free ▪ VoIP Local Service	Select the Global Crossing Enterprise VoIP Service(s) required by the customer.
Specify Codec(s) Required: ▪ G.711mu-law ▪ G.711a-law ▪ G.729 ▪ G.729B ▪ RFC2833 DTMF (required)	The network configuration described in these Application Notes was tested with all the codecs (payload types) listed in the left column, except for G.711 a-law. **Note:** RFC2833 is required for shuffling SIP calls.
Define Dial Plan	10-Digit Dialing and International (011 Country Code) calls were supported by the test configuration.
Listed Directory Numbers provided by Global Crossing	Listed directory numbers should be assigned to the endpoints at the enterprise site. This allows calls to be delivered from the PSTN. In this configuration, listed directory numbers beginning with area code 602-555 were assigned to the SIP, H.323, digital, and analog endpoints in the enterprise network.
Global Crossing provides Proxy IP Address	The IP address of the Acme Packet Net-Net Session Director in the Global Crossing network was 20.1.1.54 and used to configure the host address maps in Avaya SIP Enablement Services.
Customer provides IP Address of Avaya SIP Enablement Services	The IP address of Avaya SIP Enablement Services in the enterprise network was 10.1.1.86. Global Crossing used this IP address for routing calls destined to the listed directory numbers assigned to the enterprise site. For a redundant system, the logical IP address is used.
SIP Transport Protocol and Port	SIP signaling was transported between Avaya SIP Enablement Services and Global Crossing using UDP and port 5060.

6. Interoperability Compliance Testing

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

6.1. General Test Approach

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

- Incoming calls to the Avaya SIP network from the PSTN. PSTN calls were routed through the Global Crossing VoIP network.
- Outgoing calls from the Avaya SIP network to the PSTN through the Global Crossing VoIP network.
- Calls to/from SIP, H.323, digital and analog endpoints in the Avaya enterprise network.
- Various call types including: local, long distance, international, toll free, operator and directory assistance calls.
- Calls using G.711 and G.729 codecs.
- DTMF transmission using RFC 2833.
- Voicemail coverage and retrieval for endpoints at the enterprise site.
- Direct IP-to-IP media (also known as "Shuffling" which allows SIP endpoints to send audio (RTP) packets directly to each other without using media resources on the Avaya Media Gateway).

6.2. Test Results

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

7. Verification Steps

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

1. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call can remain active for more than 35 seconds. This time period is included to verify that proper routing of the SIP messaging has satisfied SIP protocol timers.
2. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active for more than 35 seconds.
3. Verify that the user on the PSTN can terminate an active call by hanging up.
4. Verify that an endpoint at the enterprise site can terminate an active call by hanging up.

5. If Shuffling is enabled, verify that a call originated or terminated on an Avaya 4600 Series SIP Telephone is shuffled. To determine if the call is shuffled, identify the trunk member active on the call by running the **status trunk <*group*>** command on the SAT of Avaya Communication Manager. Next, run the **status trunk group/member** command and check the **Audio Connection** field. If the call is shuffled, the field should be set to *ip-direct*; otherwise, the field would be set to *ip-tdm*.

8. Support

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

9. Conclusion

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

10. References

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

[1] *Administrator Guide for Avaya Communication Manager*, June 2005, Issue 1, Document Number 03-300509.
[2] *Avaya Extension to Cellular and Off-PBX Station (OPS) Installation and Administration Guide Release 3.0*, June 2005, Issue 9, Document Number 210-100-500.
[3] *Converged Communications Server R3.0 Installation and Administration Guide (SIP Enablement Services R3.0)*, June 2005, Issue 5, Document Number 555-245-705.
[4] *IA 770 INTUITY AUDIX Release 3.0 Installation, Upgrades, and Troubleshooting*, April 2005, Issue 1, Document Number 11-300532.
[5] *SIP Support in Release 3.0 of Avaya Communication Manager Running on the Avaya S8300, S8500, S8500B, S8700, and S8710 Media Server*, June 2005, Issue 5, Document Number 555-245-206.

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

APPENDIX A: Sample SIP INVITE Messages

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

Sample SIP INVITE Message from Global Crossing VoIP Network:
```
Session Initiation Protocol
    Request-Line: INVITE sip:6025554069;npdi=yes@10.1.1.86:5060 SIP/2.0
        Method: INVITE
        Resent Packet: False
    Message Header
        Via: SIP/2.0/UDP 20.1.1.54:5060;branch=z9hG4bK2iij4m2048u1nbghj0g0.1sr
        From: <sip:7325551638@20.1.1.54:5060>;tag=SD18n4901-gK020b71b5
            SIP from address: sip:7325551638@20.1.1.54:5060
            SIP tag: SD18n4901-gK020b71b5
        To: <sip:6025554069@10.1.1.86:5060>
            SIP to address: sip:6025554069@10.1.1.86:5060
        Call-ID: SD18n4901-1078adf431bdef166f2bab28e134a6bb-v3000i1
        CSeq: 23091 INVITE
        Max-Forwards: 69
        Allow: OPTIONS,INVITE,ACK,CANCEL,BYE,REFER,INFO,SUBSCRIBE,NOTIFY,PRACK,UPDATE
        Accept: multipart/mixed, application/sdp, application/isup, application/dtmf,
application/dtmf-relay
        Contact: <sip:7325551638@20.1.1.54:5060;transport=udp>
        Content-Length: 288
        Content-Disposition: session; handling=required
        Content-Type: application/sdp
    Message body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): Sonus_UAC 13795 29028 IN IP4 20.1.1.54
                Owner Username: Sonus_UAC
                Session ID: 13795
                Session Version: 29028
                Owner Network Type: IN
                Owner Address Type: IP4
                Owner Address: 20.1.1.54
            Session Name (s): SIP Media Capabilities
            Connection Information (c): IN IP4 20.1.1.54
                Connection Network Type: IN
                Connection Address Type: IP4
                Connection Address: 20.1.1.54
            Time Description, active time (t): 0 0
                Session Start Time: 0
                Session Stop Time: 0
            Media Description, name and address (m): audio 10362 RTP/AVP 18 0 8 101
                Media Type: audio
                Media Port: 10362
                Media Proto: RTP/AVP
                Media Format: ITU-T G.729
                Media Format: ITU-T G.711 PCMU
                Media Format: ITU-T G.711 PCMA
                Media Format: 101
            Media Attribute (a): rtpmap:18 G729/8000
                Media Attribute Fieldname: rtpmap
                Media Attribute Value: 18 G729/8000
            Media Attribute (a): rtpmap:0 PCMU/8000
                Media Attribute Fieldname: rtpmap
```

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

Sample SIP INVITE Message from Avaya SIP Enablement Services to Global Crossing:
```
Session Initiation Protocol
    Request-Line: INVITE sip:7325551638@20.1.1.54:5060;transport=udp SIP/2.0
        Method: INVITE
        Resent Packet: False
    Message Header
        Call-ID: 0a993e9d18da12a3e43176d700
        CSeq: 1 INVITE
        From: "70000" <sip:70000@10.1.1.86>;tag=0a993e9d18da1293e43176d700
            SIP Display info: "70000"
            SIP from address: sip:70000@10.1.1.86
            SIP tag: 0a993e9d18da1293e43176d700
        Record-Route:
<sip:10.1.1.86:5060;transport=udp;lr>,<sip:10.1.1.84:5061;lr;transport=tls>
        To: sip:7325551638@devcon.com
            SIP to address: sip:7325551638@devcon.com
        Via: SIP/2.0/UDP
10.1.1.86:5060;branch=z9hG4bK03030316161693939348dd.0,SIP/2.0/TLS
10.1.1.84;psrrposn=2;branch=z9hG4bK0a993e9d18da12b3e43176d700,SIP/2.0/UDP
10.1.1.72:5060;psrrposn=1;branch=z9hG4bK1f0b47acb
        Content-Length: 160
        Content-Type: application/sdp
        Contact: "70000" <sip:7327770000@10.1.1.84:5061;transport=tls>
        Max-Forwards: 67
        User-Agent: Avaya CM/R013x.00.0.340.3
        Allow: INVITE,CANCEL,BYE,ACK,PRACK,SUBSCRIBE,NOTIFY,REFER,OPTIONS
        Session-Expires: 240;Refresher=uac
        Min-SE: 240
        Supported: 100rel,timer,replaces,join
        P-Asserted-Identity:"SIP70000"<sip:7327770000@devcon.com:5061>
    Message body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): - 1 1 IN IP4 10.1.1.84
                Owner Username: -
                Session ID: 1
                Session Version: 1
                Owner Network Type: IN
                Owner Address Type: IP4
                Owner Address: 10.1.1.84
            Session Name (s): -
            Connection Information (c): IN IP4 10.1.1.85
                Connection Network Type: IN
                Connection Address Type: IP4
                Connection Address: 10.1.1.85
            Time Description, active time (t): 0 0
                Session Start Time: 0
                Session Stop Time: 0
            Media Description, name and address (m): audio 3026 RTP/AVP 0 127
                Media Type: audio
                Media Port: 3026
                Media Proto: RTP/AVP
                Media Format: ITU-T G.711 PCMU
                Media Format: 127
            Media Attribute (a): rtpmap:0 PCMU/8000
                Media Attribute Fieldname: rtpmap
                Media Attribute Value: 0 PCMU/8000
            Media Attribute (a): rtpmap:127 telephone-event/8000
                Media Attribute Fieldname: rtpmap
                Media Attribute Value: 127 telephone-event/8000
            Data (2 bytes)
```

APPENDIX B: Specifying Pattern Strings in Address Maps

The syntax for the pattern is a Linux regular expression that will match the extension numbers to be mapped. 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). For example, **[0-9]** represents any single digit and ***** represents any number of digits or characters. For example, **^sip:538[0-9]*** would match any SIP INVITE message (**^** matches the beginning of a line) for any extension 3 or more digits in length, beginning with the digits **538**, and ending with any other sequence of digits. Square brackets contain a selection of characters to be matched, with a hyphen indicating a range. In the example, **[0-9]** matches any digit, or as another example, **[13579]** matches odd-numbered digits. Curly brackets containing a whole number match that number of instances of the preceding item. For example, **[0-9]{4}** matches any four digits. Another helpful *metacharacter* is dot (period), which matches any single character. For example, the regular expression **.*** matches any quantity (including zero) of any character(s).