Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Broadvox SIP Trunking Service with Avaya Aura® Communication Manager 6.3, Avaya Aura® Session Manager 6.3, and Avaya Session Border Controller for Enterprise 6.2 – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between the Broadvox SIP Trunking service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Aura® Communication Manager 6.3, Avaya Aura® Session Manager 6.3 and Avaya Session Border Controller for Enterprise 6.2.

The Broadvox SIP Trunking service offered by Broadvox provides customers with PSTN access via a SIP trunk between the enterprise and the Broadvox network, as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

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

Table of Contents

1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between the Broadvox SIP Trunking service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Aura® Communication Manager 6.3, Avaya Aura® Session Manager 6.3, Avaya Session Border Controller for Enterprise (Avaya SBCE) 6.2 and various Avaya endpoints.

The Broadvox SIP Trunking service referenced within these Application Notes is designed for enterprise business customers. Customers using this service with the Avaya SIP-enabled enterprise solution are able to place and receive PSTN calls via a broadband WAN connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks such as analog and/or ISDN-PRI.

2. General Test Approach and Test Results

A simulated enterprise site containing all the equipment for the Avaya SIP-enabled solution was installed at the Avaya Solution and Interoperability Lab. The enterprise site was configured to connect to the Broadvox SIP Trunk service via a broadband connection to the public Internet.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

To verify SIP trunking interoperability, the following features and functionality were covered during the interoperability compliance test:

- SIP trunk registration with the service provider.
- Incoming PSTN calls to various phone types. Phone types included H.323, SIP, digital, and analog telephones at the enterprise. All inbound calls from the PSTN were routed to the enterprise across the SIP trunk from the service provider.
- Outgoing PSTN calls from various phone types. Phone types included H.323, SIP, digital, and analog telephones at the enterprise. All outbound calls to the PSTN were routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls to/from Avaya one-X® Communicator softphones in the "This Computer" and "Other Phone" modes. (H.323, SIP).
- Inbound and outbound PSTN calls to/from Avaya Flare® Experience for Windows softphones (SIP).
- Inbound and outbound PSTN calls to/from SIP remote workers using Avaya 96x1 deskphones, Avaya one-X® Communicator and Flare® Experience for Windows.
- Various call types, including: local, long distance and international.
- Codecs G.711U and G729A and proper codec negotiation.
- DTMF tone transmissions passed as out-of-band RTP events as per RFC 2833.
- Caller ID presentation and Caller ID restriction.
- Voicemail redirection and navigation.
- User features such as hold and resume, transfer and conference.
- Off-net call forwarding and mobility (extension to cellular).
- Routing inbound PSTN calls to call center agent queues.
- T.38 fax.

Items not supported or not tested included the following:

- Inbound toll-free and emergency (911) calls are supported but were not tested as part of the compliance test.
- Network Call Redirection methods using REFER or 302 Temporarily Unavailable messages are not supported by Broadvox and were not tested.
- Operator services such as dialing 0 or 0 + 10 digits are not supported.

2.2. Test Results

Interoperability testing of the Broadvox SIP Trunking service with the Avaya SIP-enabled enterprise solution was completed with successful results for all test cases with the exception of the observation described below:

- **Media Shuffling**: Direct IP-IP Audio Connections (media shuffling) must be disabled on the Communication Manager signaling group used by the SIP trunk connected to the Broadvox SIP Trunking service. The re-INVITEs used by Communication Manager to perform media shuffling are not supported by Broadvox.

2.3. Support

For technical support on the Broadvox SIP Trunking service, contact Broadvox by calling (888) 849-9608 options 2 or 3, or by sending an e-mail to techsupport@broadvox.com.
For all other inquiries visit http://www.broadvox.com/products/sip-trunking, or contact customer service by calling (888) 849-9608 opt. 1, or by email to customerservice@broadvox.com.

3. Reference Configuration

Figure 1 illustrates the sample Avaya SIP-enabled enterprise solution, connected to the Broadvox SIP Trunking service through a public Internet WAN connection.

For security purposes, references to any public IP addresses used during the compliance test have been replaced in these Application Notes with private addresses. Also, PSTN routable phone numbers used in the test have been changed to non-routable numbers.

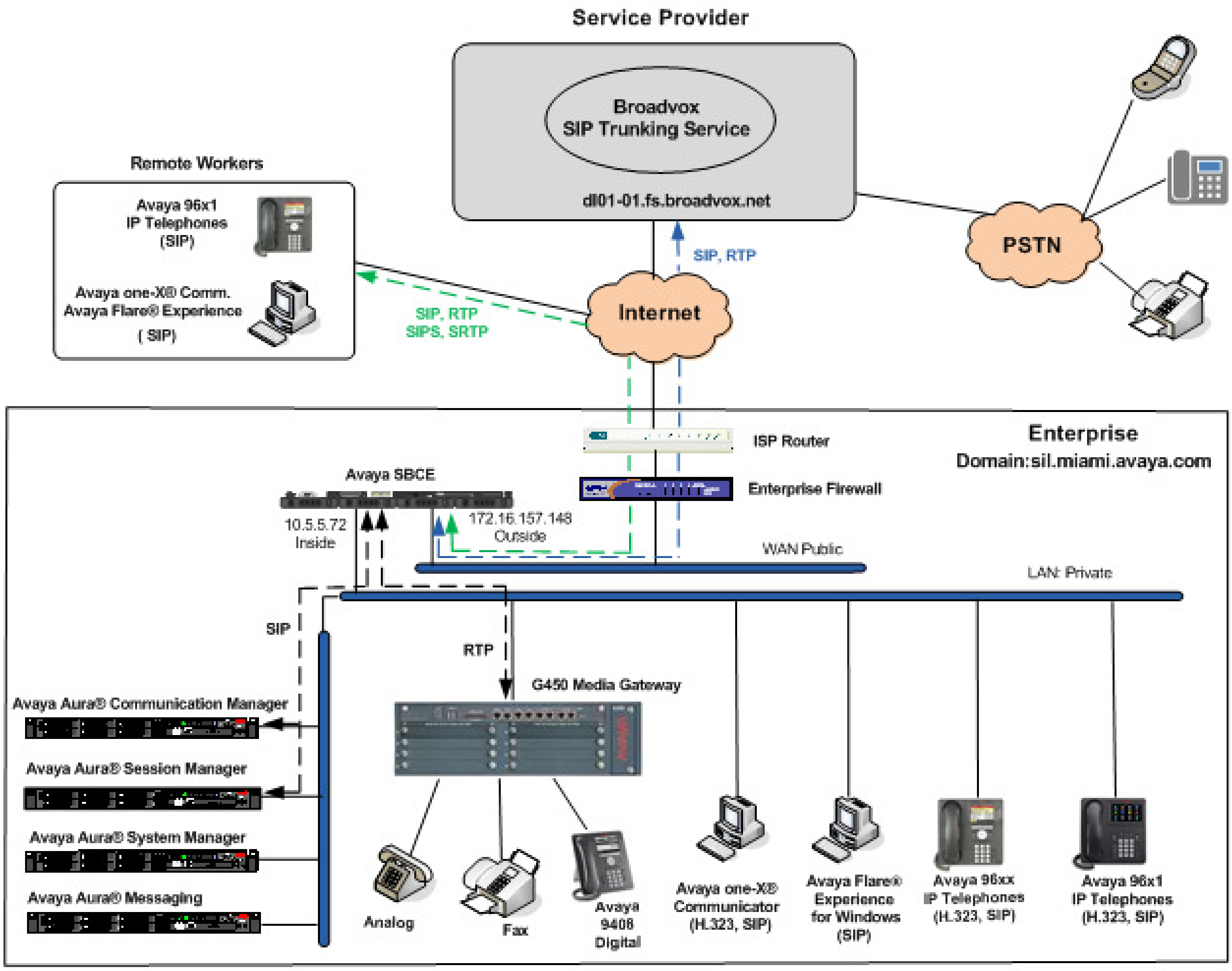


Figure 1: Avaya SIP Enterprise Solution connected to the Broadvox SIP Trunking service

The components used to create the simulated enterprise customer site included:

- Avaya Aura® Communication Manager.
- Avaya Aura® Session Manager.
- Avaya Aura® System Manager.
- Avaya Session Border Controller for Enterprise.
- Avaya Aura® Messaging.
- Avaya G450 Media Gateway.
- Avaya 96x0 and 96x1 Series IP Telephones (H.323 and SIP).
- Avaya one-X® Communicator soft phones (H.323 and SIP).
- Avaya Flare® Experience for Windows softphones.
- Avaya digital and analog telephones.

Additionally, the reference configuration included remote worker functionality, introduced with Avaya Aura® 6.2 Feature Pack 2. A remote worker is a SIP endpoint that resides in the untrusted network, registered to the Session Manager at the enterprise via the Avaya SBCE. Remote workers feature the same functionality as any other endpoint at the enterprise. This functionality was successfully tested during the compliance test, using the following endpoints and protocols:

- Avaya 96x1 SIP Deskphones (using TLS and SRTP).
- Avaya one-X® Communicator SIP (using TCP and RTP).
- Avaya Flare® Experience for Windows (using TCP and RTP).

The configuration tasks required to support remote workers are beyond the scope of these Application Notes; hence they are not discussed in this document. Consult [**7**] in the **References** section for more information on this topic.

The Avaya SBCE is located at the edge of the enterprise. It has a public side that connects to the external network and a private side that connects to the enterprise infrastructure. All signaling and media traffic entering or leaving the enterprise flows through the Avaya SBCE, which in this way can protect the enterprise against any SIP-based attacks. The Avaya SBCE provides the registration capability of the SIP trunk with the service provider, and also performs network address translation at both the IP and SIP layers.

The SIP Proxy server assigned by Broadvox during the compliance test used the fully qualified domain name (FQDN) "dl01-01.fs.broadvox.net". The Avaya SBCE was configured to use an external DNS server to resolve the IP addresses of the Broadvox SIP Proxy servers used to route outbound signaling traffic.

The transport protocol between the Avaya SBCE and Broadvox across the public IP network is UDP. The transport protocol between the Avaya SBCE and the enterprise Session Manager across the enterprise IP network is TCP.

For inbound calls, the calls flow from the service provider to the external firewall, to the Avaya SBCE, then to Session Manager. Session Manager uses the configured dial patterns (or regular expressions) and routing policies to determine the recipient (in this case the Communication Manager) and on which link to send the call. Once the call arrives at Communication Manager, further incoming call treatment, such as incoming digit translations and class of service restrictions may be performed.

Outbound calls to the PSTN were first processed by Communication Manager for outbound feature treatment such as automatic route selection and class of service restrictions. Once Communication Manager selected the proper SIP trunk, the call was routed to Session Manager. Session Manager once again used the configured dial patterns (or regular expressions) and routing policies to determine the route to the Avaya SBCE for egress to the Broadvox network.

A separate SIP trunk was created between Communication Manager and Session Manager to carry the service provider traffic. This was done so that any trunk or codec settings required by the service provider could be applied only to this trunk without affecting other enterprise SIP traffic. The trunk carried both inbound and outbound traffic.

Messaging was used during the compliance test to verify voice mail redirection and navigation, as well as the delivery of MWI (Message Waiting Indicator) messages to the enterprise telephones. Messaging was installed on a single standalone server located on the enterprise network, administered as a separate SIP entity in Session Manager. Since the configuration tasks for Messaging are not directly related to the interoperability tests with the Broadvox SIP Trunking service, they are not included in these Application Notes.

4. Equipment and Software Validated

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

Component	Version
Avaya	
Avaya Aura® Communication Manager on HP® Proliant DL360 G7 Server	6.3-03.0.124.0
Avaya Aura® Session Manager on HP® Proliant DL360 G7 Server	6.3.3.0633004
Avaya Aura® System Manager on HP® Proliant DL360 G7 Server	6.3.3 Software Update Rev. 6.3.3.5.1719
Avaya Session Border Controller for Enterprise on a Dell R210 V2 Server	6.2.0.Q48
Avaya Aura® Messaging on a Dell PowerEdge R610 server	6.2.SP3
Avaya G450 Media Gateway	33.13.0
Avaya 96xx Series IP Telephones (H.323)	Avaya one-X Deskphone Edition 3.2
Avaya 96x1 Series IP Telephones (SIP)	Avaya one-X Deskphone Edition SIP 6.2.2.17
Avaya 96x1 Series IP Telephones (H.323)	Avaya one-X Deskphone Edition H.323 6.2.2 (SP2)
Avaya one-X Communicator (H.323, SIP)	6.1.8.06-SP8-40314
Avaya Flare Experience for Windows	1.1.3.14
Broadvox SIP Trunking service	
Broadvox Fusion softswitch	Version 1.0

The specific configuration above was used for the compliance testing. Note that this solution will be compatible with other Avaya Server and Media Gateway platforms running similar versions of Communication Manager and Session Manager.

5. Configure Avaya Aura® Communication Manager

This section describes the procedure for configuring Communication Manager for the Broadvox SIP Trunking service. A SIP trunk is established between Communication Manager and Session Manager for use by signaling traffic to and from Broadvox. It is assumed that the general installation of Communication Manager, the Avaya G450 Media Gateway and Session Manager has been previously completed and is not discussed here.

The Communication Manager configuration was performed using the System Access Terminal (SAT). Some screens in this section have been abridged and highlighted for brevity and clarity in presentation.

5.1. Licensing and Capacity

Use the **display system-parameters customer-options** command to verify that the **Maximum Administered SIP Trunks** value on **Page 2** is sufficient to support the desired number of simultaneous SIP calls across all SIP trunks at the enterprise including any trunks to and from the service provider. The example shows that **24000** licenses are available and **391** are in use. 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   2 of   11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                                    USED
                     Maximum Administered H.323 Trunks: 12000 10
           Maximum Concurrently Registered IP Stations: 18000 3
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 41000 2
                   Maximum Video Capable IP Softphones: 18000 6
                       Maximum Administered SIP Trunks: 24000 391
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   0
                     Maximum Media Gateway VAL Sources: 250   1
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 100   0

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

5.2. System Features

Use the **change system-parameters features** command to set the **Trunk-to-Trunk Transfer** field to ***all*** to allow incoming calls from the PSTN to be transferred to another PSTN endpoint. If for security reasons, incoming calls should not be allowed to transfer back to the PSTN then leave the field set to ***none***.

```
change system-parameters features                              Page   1 of  20
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? y
                                    Trunk-to-Trunk Transfer: all
               Automatic Callback with Called Party Queuing? n
    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 (or Silence) on Transferred Trunk Calls? no
              DID/Tie/ISDN/SIP Intercept Treatment: attendant
    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
```

On **Page 9** verify that a text string has been defined to replace the Calling Party Number (CPN) for restricted or unavailable calls. This text string is entered in the two fields highlighted below. The compliance test used the value of ***restricted*** for restricted calls and ***unavailable*** for unavailable calls.

```
change system-parameters features                              Page   9 of  20
                        FEATURE-RELATED SYSTEM PARAMETERS

CPN/ANI/ICLID PARAMETERS
   CPN/ANI/ICLID Replacement for Restricted Calls: restricted
  CPN/ANI/ICLID Replacement for Unavailable Calls: unavailable

DISPLAY TEXT
                                       Identity When Bridging: principal
                                        User Guidance Display? n
 Extension only label for Team button on 96xx H.323 terminals? n

INTERNATIONAL CALL ROUTING PARAMETERS
                 Local Country Code: ___
           International Access Code: _____
```

5.3. IP Node Names

Use the **change node-names ip** command to verify that node names have been previously defined for the IP addresses of Communication Manager (**procr**) and the Session Manager Security Module (**asm**). These node names will be needed for defining the service provider signaling group in **Section 5.6**.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name                  IP Address
Acme_s1p0               192.168.10.52
HG_CM                   172.16.5.12
HG_SM                   172.16.5.32
asm                     192.168.10.32
default                 0.0.0.0
ip_office               192.168.10.60
msgserver               192.168.10.12
procr                   192.168.10.12
procr6                  ::
```

5.4. Codecs

Use the **change ip-codec-set** command to define a list of codecs to use for calls between the enterprise and the service provider. For the compliance test, ip-codec-set 4 was used for this purpose. The Broadvox SIP Trunking service used codecs G.711MU and G.729A, in this order of preference. Enter ***G.711MU*** and ***G.729A*** in the **Audio Codec** column of the table. Default values can be used for all other fields.

```
change ip-codec-set 4                                           Page   1 of   2

                          IP Codec Set

    Codec Set: 4

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

On **Page 2**, set the **Fax Mode** to ***t.38-standard***.

```
change ip-codec-set 4                                           Page   2 of   2

                          IP Codec Set

                              Allow Direct-IP Multimedia? n



                     Mode                   Redundancy
    FAX              t.38-standard           0              ECM: y
    Modem            off                     0
    TDD/TTY          US                      3
    Clear-channel    n                       0
```

5.5. IP Network Regions

Create a separate IP network region for the service provider trunk group. This allows for separate codec or quality of service settings to be used (if necessary) for calls between the enterprise and the service provider versus calls within the enterprise or elsewhere. For the compliance test, IP Network Region 4 was chosen for the service provider trunk. Use the **change ip-network-region 4** command to configure region 4 with the following parameters:

- Set the **Authoritative Domain** field to match the SIP domain of the enterprise. In this configuration, the domain name is ***sil.miami.avaya.com*** as assigned to the shared test environment in the Avaya test lab. This domain name appears in the "From" header of SIP messages originating from this IP region.
- Enter a descriptive name in the **Name** field.
- Leave both **Intra-region** and **Inter-region IP-IP Direct Audio** set to ***yes***, the default setting. This will enable **IP-IP Direct Audio** (shuffling), to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya Media Gateway. Shuffling can be further restricted at the trunk level on the Signaling Group form.
- Set the **Codec Set** field to the IP codec set defined in **Section 5.4**.
- Default values can be used for all other fields.

```
change ip-network-region 4                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 4
Location: 1         Authoritative Domain: sil.miami.avaya.com
    Name: Broadvox SIP Trunk       Stub Network Region: n
MEDIA PARAMETERS                   Intra-region IP-IP Direct Audio: yes
      Codec Set: 4                 Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                         IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        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
```

On **Page 4**, define the IP codec set to be used for traffic between region 4 and region 1 (the rest of the enterprise). Enter the desired IP codec set in the **codec set** column of the row with destination region (**dst rgn**) 1. Default values may be used for all other fields. The following example shows the settings used for the compliance test. It indicates that codec set **4** will be used for calls between region 4 (the service provider region) and region 1 (the rest of the enterprise).

```
change ip-network-region 4                                      Page   4 of  20

 Source Region: 4     Inter Network Region Connection Management     I       M
                                                                     G  A    t
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    c
 rgn  set   WAN  Units     Total Norm  Prio Shr Regions          CAC  R  L    e
 1    4     y    NoLimit                                             n  ___  t
 2    ____                                                              ___
 3    ____                                                              ___
 4    4                                                                 all
 5
```

5.6. Signaling Group

Use the **add signaling-group** command to create a signaling group between Communication Manager and Session Manager for use by the service provider trunk. This signaling group is used for inbound and outbound calls between the service provider and the enterprise. For the compliance test, signaling group 4 was used for this purpose and was configured using the parameters highlighted below:

- Set the **Group Type** field to ***sip***.
- Set the **IMS Enabled** field to ***n***. This specifies the Communication Manager will serve as an Evolution Server for the Session Manager.
- Set the **Transport Method** to the transport protocol to be used between Communication Manager and Session Manager. For the compliance test, ***tcp*** was used.
- Set the **Peer Detection Enabled** field to ***y***. The **Peer-Server** field will initially be set to ***Others*** and cannot be changed via administration. Later, the **Peer-Server** field will automatically change to ***SM*** once Communication Manager detects its peer is a Session Manager.

  Note: Once the **Peer-Server** field is updated to ***SM***, the system changes the default values of the following fields, setting them to display–only:
    - **Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers?** is changed to ***y***.
    - **Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers?** is changed to ***n***.
- Set the **Near-end Node Name** to ***procr***. This node name maps to the IP address of the Communication Manager as defined in **Section 5.3**.
- Set the **Far-end Node Name** to ***asm***. This node name maps to the IP address of Session Manager, as defined in **Section 5.3**.

```
change signaling-group 4                                        Page   1 of   2
                                SIGNALING GROUP

 Group Number: 4                  Group Type: sip
  IMS Enabled? n            Transport Method: tcp
        Q-SIP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n

   Near-end Node Name: procr                  Far-end Node Name: asm
 Near-end Listen Port: 5075                 Far-end Listen Port: 5075
                                         Far-end Network Region: 4
                                  Far-end Secondary Node Name:
Far-end Domain: sil.miami.avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? n
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y
                                                   Alternate Route Timer(sec): 6
```

- Set the **Near-end Listen Port** and **Far-end Listen Port** to a valid unused port instead of the default well-known port value. (For TCP, the well-known port value is 5060). This is necessary so the SM can distinguish this trunk from the trunk used for other enterprise SIP traffic. For the compliance test both the **Near-end Listen Port** and **Far-end Listen Port** were set to ***5075***.
- Set the **Far-end Network Region** to the IP network region defined for the service provider in **Section 5.5**.
- Set the **Far-end Domain** to the domain of the enterprise.
- Set the **DTMF over IP** field to ***rtp-payload***. This value enables Communication Manager to send DTMF transmissions using RFC 2833.
- Set **Direct IP-IP Audio Connections** to ***n***. This setting will effectively disable media shuffling on the SIP trunk. Broadvox does not support the re-INVITEs used by Communication Manager when performing media shuffling, as previously mentioned in **Section 2.2**.
- Default values may be used for all other fields.

5.7. Trunk Group

Use the **add trunk-group** command to create a trunk group for the signaling group created in **Section 5.6**. For the compliance test, trunk group 4 was configured using the parameters highlighted below.

- Set the **Group Type** field to ***sip***.
- Enter a descriptive name for the **Group Name**.
- Enter an available trunk access code (TAC) that is consistent with the existing dial plan in the **TAC** field.
- Set the **Service Type** field to ***public-ntwrk***.
- Set the **Signaling Group** to the signaling group shown in the previous step.
- Set the **Number of Members** field to the number of trunk members in the SIP trunk group. This value determines how many simultaneous SIP calls can be supported by this trunk.
- Default values were used for all other fields.

```
change trunk-group 4                                            Page   1 of  21
                                TRUNK GROUP

Group Number: 4                  Group Type: sip             CDR Reports: y
  Group Name: Broadvox                      COR: 1        TN: 1        TAC: 604
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                     Night Service:
Queue Length: 0
Service Type: public-ntwrk            Auth Code? n
                                           Member Assignment Method: auto
                                                    Signaling Group: 4
                                                  Number of Members: 6
```

On **Page 2**, verify that the **Preferred Minimum Session Refresh Interval** is set to a value acceptable to the service provider. This value defines the interval that re-INVITEs must be sent to keep the active session alive. For the compliance test, the default value of ***600*** seconds was used.

```
change trunk-group 4                                            Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 5000

            SCCAN? n                                Digital Loss Group: 18
                     Preferred Minimum Session Refresh Interval(sec): 600

 Disconnect Supervision - In? y  Out? y
```

On **Page 3**, set the **Numbering Format** field to ***public***. This field specifies the format of the calling party number (CPN) sent to the far-end. Beginning with Communication Manager 6.0, public numbers are automatically preceded with a + sign when passed in the SIP From, Contact and P-Asserted Identity headers. The addition of the + sign did not impact interoperability with Broadvox. Set the **Replace Restricted Numbers** and **Replace Unavailable Numbers** fields to ***y***. This will allow the CPN displayed on local endpoints to be replaced with the value set in **Section 5.2**, if the inbound call enabled CPN block.

```
change trunk-group 4                                            Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                           Maintenance Tests? y


                     Numbering Format: public
                                                  UUI Treatment: service-provider

                                                   Replace Restricted Numbers? y
                                                  Replace Unavailable Numbers? y
```

On **Page 4**, set the **Telephone Event Payload Type** to ***101***, and **Convert 180 to 183 for Early Media** to ***y***. Default values were used for all other fields.

```
change trunk-group 4                                            Page   4 of  21
                              PROTOCOL VARIATIONS

                                                 Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                                Send Transferring Party Information? n
                                           Network Call Redirection? n

                                              Send Diversion Header? n
                                            Support Request History? n
                                       Telephone Event Payload Type: 101


                                Convert 180 to 183 for Early Media? y
                          Always Use re-INVITE for Display Updates? n
                                Identity for Calling Party Display: P-Asserted-Identity
                    Block Sending Calling Party Location in INVITE? n
                         Accept Redirect to Blank User Destination? n
                                                      Enable Q-SIP? n
```

5.8. Calling Party Information

The calling party number is sent in the SIP "From", "Contact" and "PAI" headers. Since public numbering was selected to define the format of this number (**Section 5.7**), use the **change public-numbering** command to create an entry for each extension which has a DID assigned. DID numbers are provided by the SIP service provider. Each DID number is assigned in this table to one enterprise internal extension or Vector Directory Numbers (VDNs). In the sample configuration, three DID numbers were assigned for testing. These DID numbers, preceded by the country code "1" and the "+" sign which is automatically inserted by Communication Manager, were used as the outbound calling party information on the service provider trunk when calls were originated from these extensions.

```
change public-unknown-numbering 1                              Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                                     Total
Ext Ext              Trk       CPN                   CPN
Len Code             Grp(s)    Prefix                Len
                                                           Total Administered: 5
 4  2                                                 4       Maximum Entries: 9999
 4  3                                                 4
 4  3001             4         17325551234           11    Note: If an entry applies to
 4  3002             4         17325551235           11    a SIP connection to Avaya
 4  3003             4         17325551236           11    Aura(R) Session Manager,
                                                           the resulting number must
                                                           be a complete E.164 number.
```

5.9. Inbound Routing

In general, the "incoming call handling treatment" form for a trunk group can be used to manipulate the digits received for an incoming call if necessary. Since Session Manager is present, Session Manager can be used to perform digit conversion using an Adaptation, and digit manipulation via the Communication Manager incoming call handling table may not be necessary. If the 10 digit DID number sent by Broadvox is left unchanged by Session Manager, then the DID number can be mapped to an extension using the incoming call handling treatment of the receiving trunk group. Use the **change inc-call-handling-trmt** command to create an entry for each DID.

```
change inc-call-handling-trmt trunk-group 4                    Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/       Number   Number       Del  Insert
 Feature         Len      Digits
 public-ntwrk    10 7325551234        10   3001
 public-ntwrk    10 7325551235        10   3002
 public-ntwrk    10 7325551236        10   3003
 public-ntwrk
```

5.10. Outbound Routing

In these Application Notes, the Automatic Route Selection (ARS) feature is used to route outbound calls via the SIP trunk to the service provider. In the sample configuration, the single digit 9 is used as the ARS access code. Enterprise callers will dial 9 to reach an "outside line". This common configuration is illustrated below with little elaboration. Use the **change dialplan analysis** command to define a dialed string beginning with ***9*** of length ***1***, as a feature access code (***fac***).

```
change dialplan analysis                                        Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all          Percent Full: 2

    Dialed    Total  Call     Dialed    Total  Call     Dialed    Total  Call
    String    Length Type     String    Length Type     String    Length Type
   0          1    attd
   1          5    ext
   2          5    ext
   3          4    ext
   4          5    ext
   5          5    ext
   6          3    dac
   7          5    ext
   8          5    ext
   9          1    fac
   *          3    dac
   #          3    dac
```

Use the **change feature-access-codes** command to configure ***9*** as the **Auto Route Selection (ARS) – Access Code 1**.

```
change feature-access-codes                                     Page   1 of  10
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code: *10
         Abbreviated Dialing List2 Access Code: *12
         Abbreviated Dialing List3 Access Code: *13
Abbreviated Dial - Prgm Group List Access Code: *14
                      Announcement Access Code: *19
                       Answer Back Access Code:

      Auto Alternate Routing (AAR) Access Code: *00
    Auto Route Selection (ARS) - Access Code 1: 9      Access Code 2:
                 Automatic Callback Activation: *33     Deactivation: #33
Call Forwarding Activation Busy/DA: *30    All: *31     Deactivation: #30
   Call Forwarding Enhanced Status:        Act:         Deactivation:
```

Use the **change ars analysis** command to configure the routing of dialed digits following the first digit 9. The example below shows a subset of the dialed strings tested as part of the compliance test. See **Section 2.1** for the complete list of call types tested. All dialed strings are mapped to route pattern 4 which contains the SIP trunk group to the service provider.

```
change ars analysis 0                                           Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                  Location: all           Percent Full: 0

          Dialed           Total    Route     Call    Node   ANI
          String          Min  Max  Pattern   Type    Num    Reqd
    011                   10   14   4         intl           n
    1786                  11   11   4         fnpa           n
    1800                  11   11   4         fnpa           n
    732                   10   10   4         hnpa           n
    411                   3    3    4         svcl           n
                                                             n
```

The route pattern defines which trunk group will be used for the call and performs any necessary digit manipulation. Use the **change route-pattern** command to configure the parameters for the service provider trunk route pattern in the following manner. The example below shows the values used for route pattern 4 for the compliance test.

- **Pattern Name**: Enter a descriptive name.
- **Grp No**: Enter the outbound trunk group for the SIP service provider. For the compliance test, trunk group 4 was used.
- **FRL**: Set the Facility Restriction Level (**FRL**) field to a level that allows access to this trunk for all users that require it. The value of ***0*** is the least restrictive level.
- **Pfx Mrk**: Setting the prefix mark (**Pfx Mrk**) to ***1*** will prefix any FNPA 10-digit number with a "1" and leave numbers of any other length unchanged. This will ensure 1 + 10 digits are sent to the service provider for long distance calls.
- Default values were used for all other fields.

```
change route-pattern 4                                          Page   1 of   3
                    Pattern Number: 4      Pattern Name: Broadvox
                             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: 4   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 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                          none
```

6. Configure Avaya Aura® Session Manager

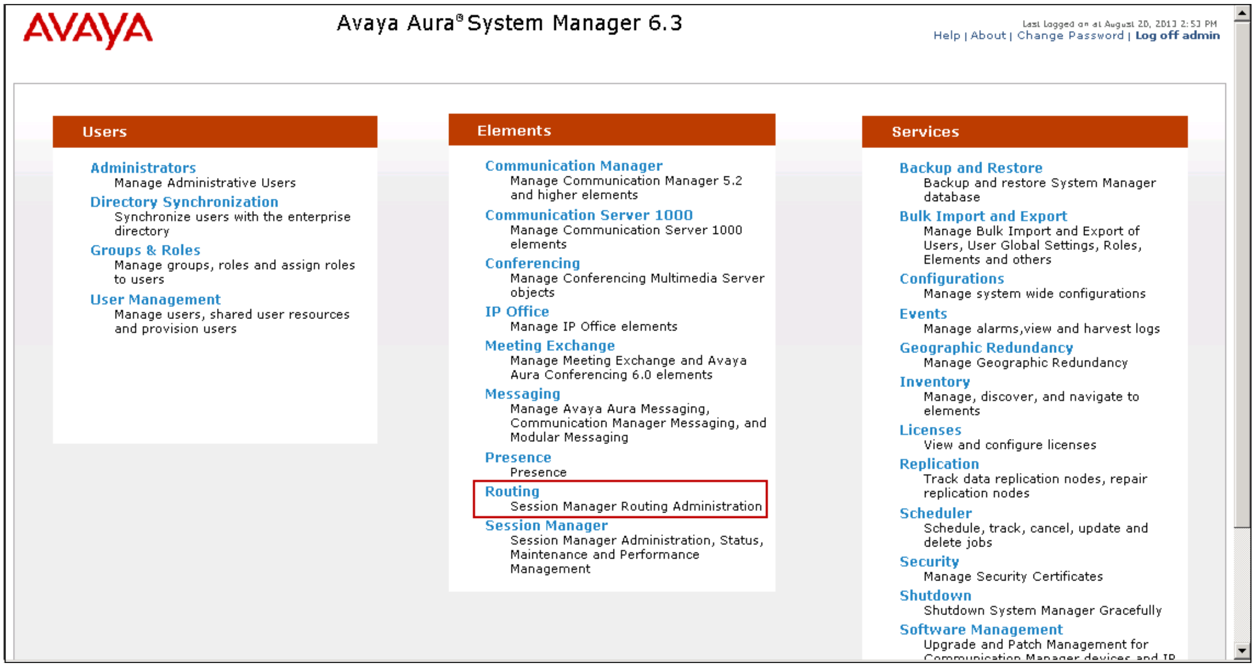
This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain.
- Logical/physical Locations that can be occupied by SIP Entities.
- SIP Entities corresponding to Communication Manager, Session Manager and the Avaya SBCE.
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities.
- Routing Policies, which control call routing between the SIP Entities.
- Dial Patterns, which govern to which SIP Entity a call is routed.

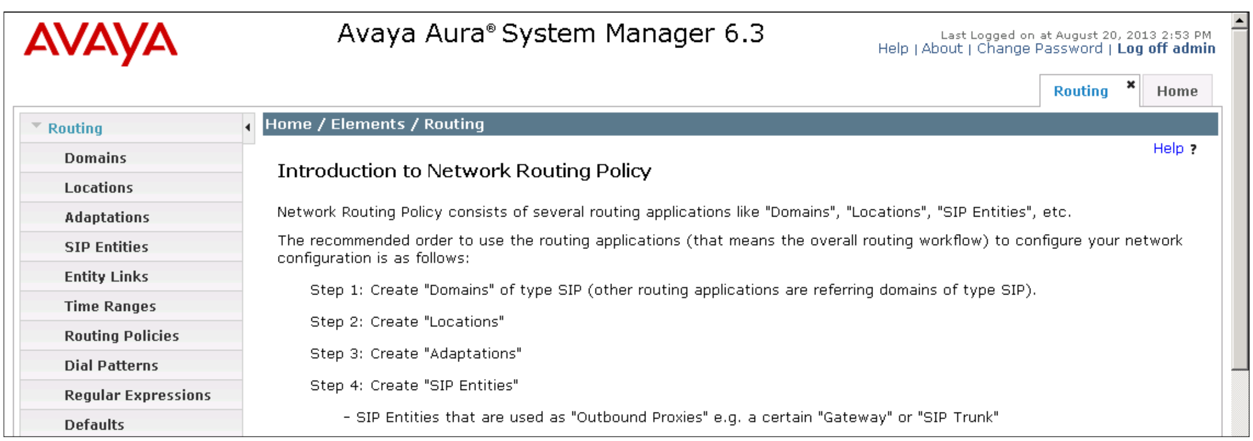
The following sections assume that the initial configuration of Session Manager and System Manager has already been completed, and that network connectivity exists between System Manager and Session Manager.

6.1. System Manager Login and Navigation

Session Manager configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL "https://<ip-address>/SMGR", where "<ip-address>" is the IP address of System Manager. Log in with the appropriate credentials and click on **Log On** (not shown). The screen shown below is then displayed; click on **Routing**.



The navigation tree displayed in the left pane below will be referenced in subsequent sections to navigate to items requiring configuration. Most items discussed in this section will be located under the **Routing** link shown below.

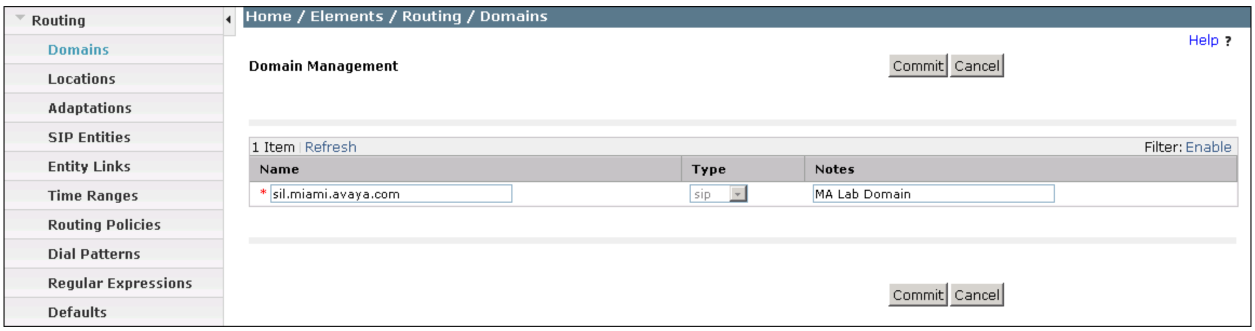


6.2. SIP Domain

Create a SIP domain for each domain for which Session Manager will need to be aware in order to route calls. For the compliance test, this will be the enterprise domain, **sil.miami.avaya.com**. Navigate to **Routing → Domains** in the left-hand navigation pane (**Section 6.1**) and click the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

- **Name:** Enter the domain name.
- **Type:** Select **sip** from the pull-down menu.
- **Notes:** Add a brief description (optional).

Click **Commit**. The screen below shows the entry for the enterprise domain.



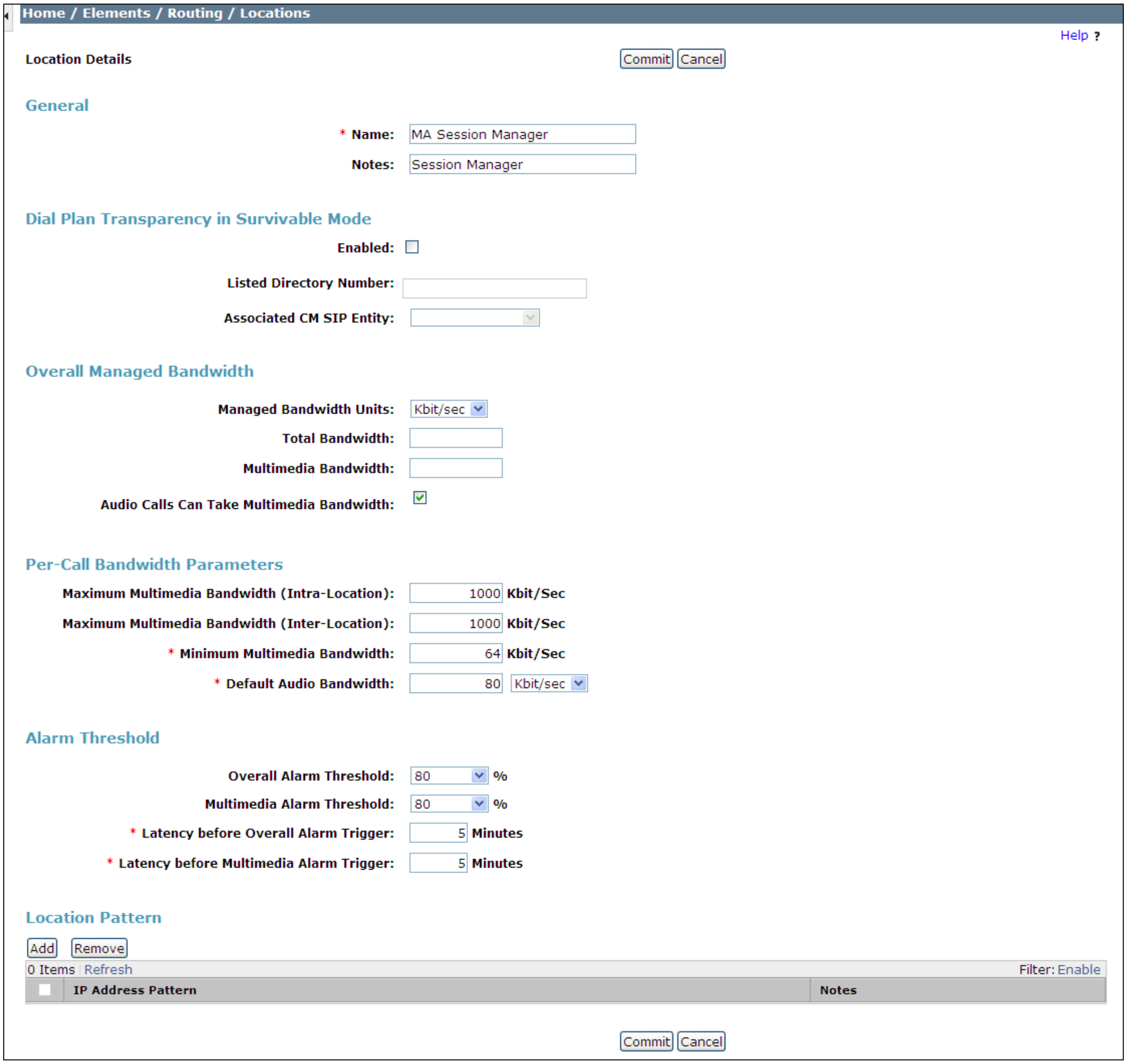
6.3. Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management, call admission control and location-based routing. To add a location, navigate to **Routing → Locations** in the left-hand navigation pane and click the **New** button in the right pane (not shown). In the **General** section, enter the following values.

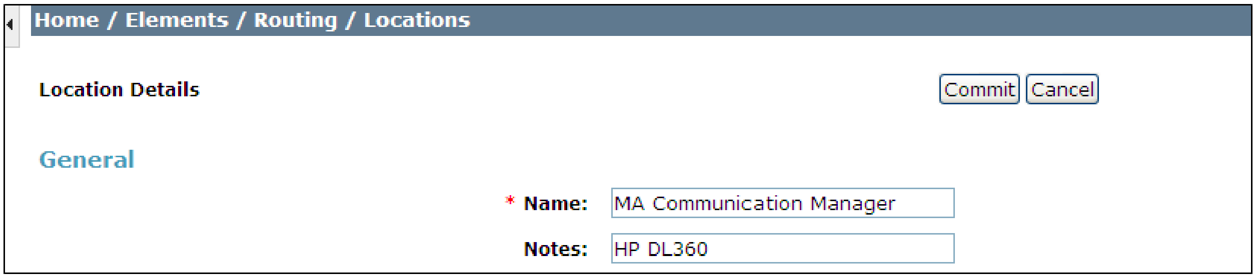
- **Name:** Enter a descriptive name for the location.
- **Notes:** Add a brief description (optional).

Defaults can be used for all other parameters.

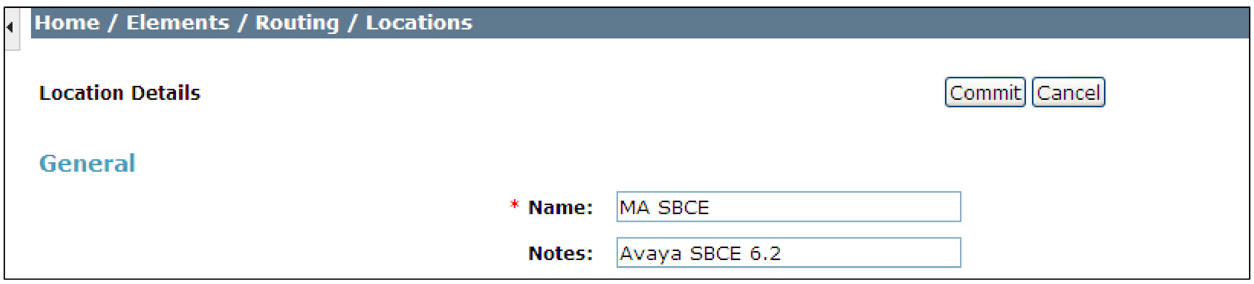
The following screen shows the location details for the location named "MA Session Manager". Later, this location will be assigned to the SIP Entity corresponding to Session Manager.



The following screen shows the location details for the location named “MA Communication Manager”. Later, this location will be assigned to the SIP Entity corresponding to Communication Manager. Other location parameters (not shown) retained the default values.



The following screen shows the location details for the location named “MA SBCE”. Later, this location will be assigned to the SIP Entity corresponding to the Avaya SBCE. Other location parameters (not shown) retained the default values.

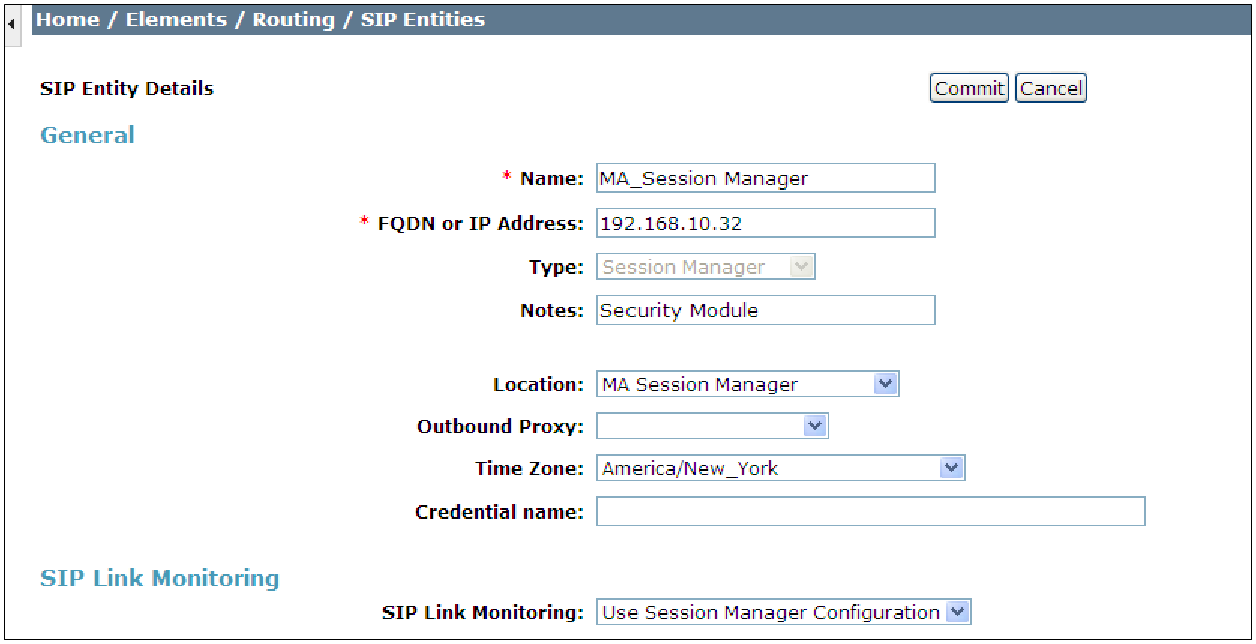


6.4. SIP Entities

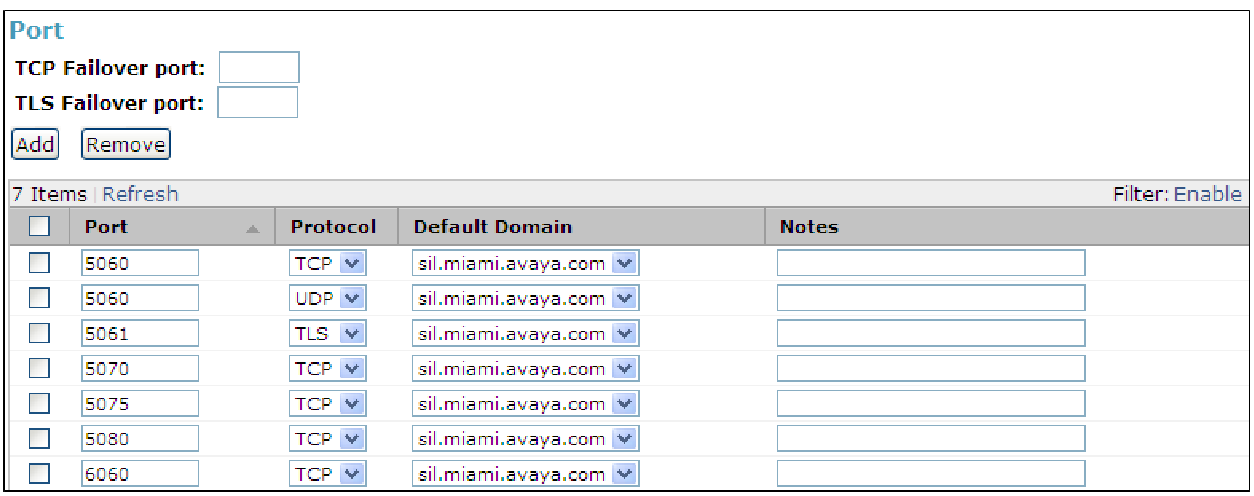
A SIP Entity must be added for Session Manager and for each SIP telephony system connected to it, which includes Communication Manager and the Avaya SBCE. Navigate to **Routing → SIP Entities** in the left navigation pane and click on the **New** button in the right pane (not shown). In the **General** section, enter the following values. Use default values for all remaining fields:

- **Name:** Enter a descriptive name.
- **FQDN or IP Address:** Enter the FQDN or IP address of the SIP Entity that is used for SIP signaling.
- **Type:** Select **Session Manager** for Session Manager, **CM** for Communication Manager and **SIP Trunk** for the Avaya SBCE.
- **Adaptation:** This field is only present if **Type** is not set to **Session Manager** If Adaptations were to be created, here is where they would be applied to the entity.
- **Location:** Select the location that applies to the SIP Entity being created, defined in **Section 6.3**.
- **Time Zone:** Select the time zone for the location above.

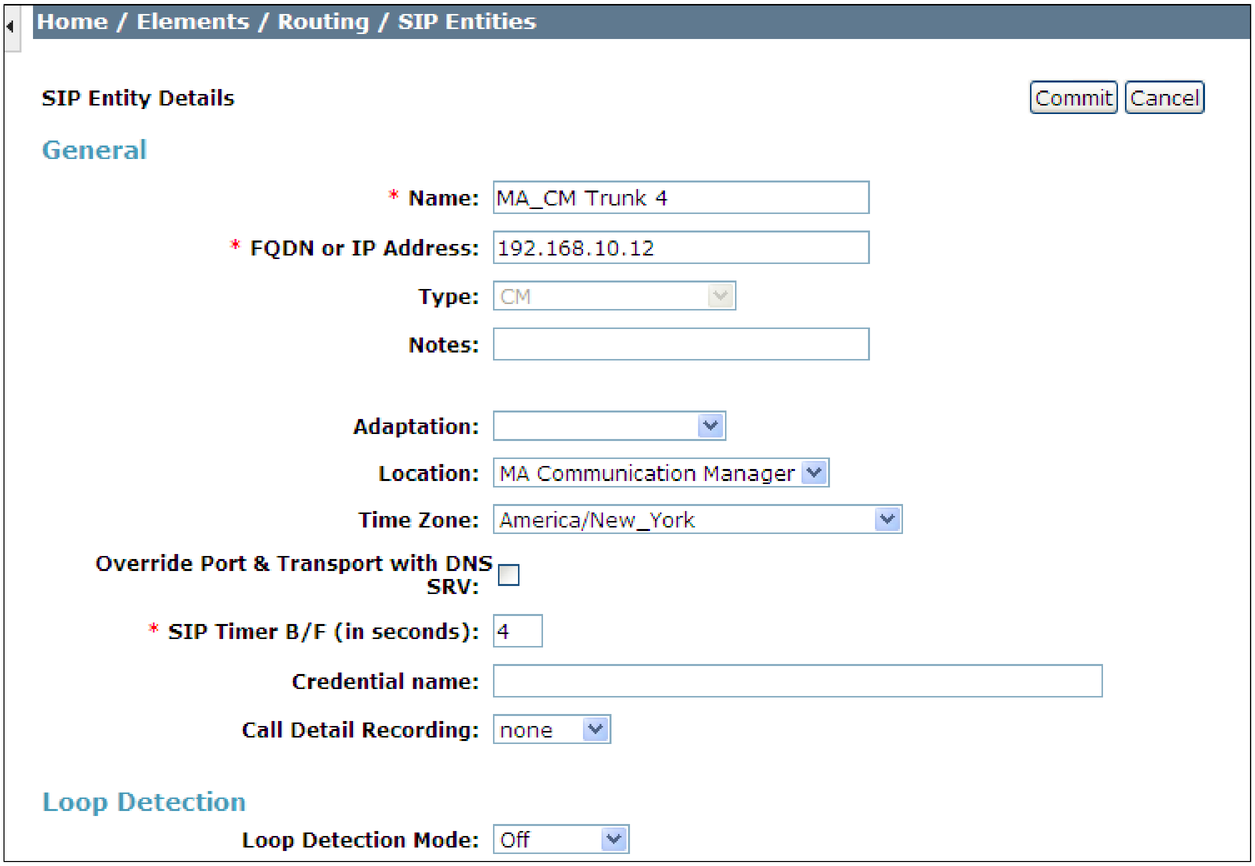
The following screen shows the addition of the Session Manager SIP Entity. The IP address of the Session Manager Security Module is entered for **FQDN or IP Address**.



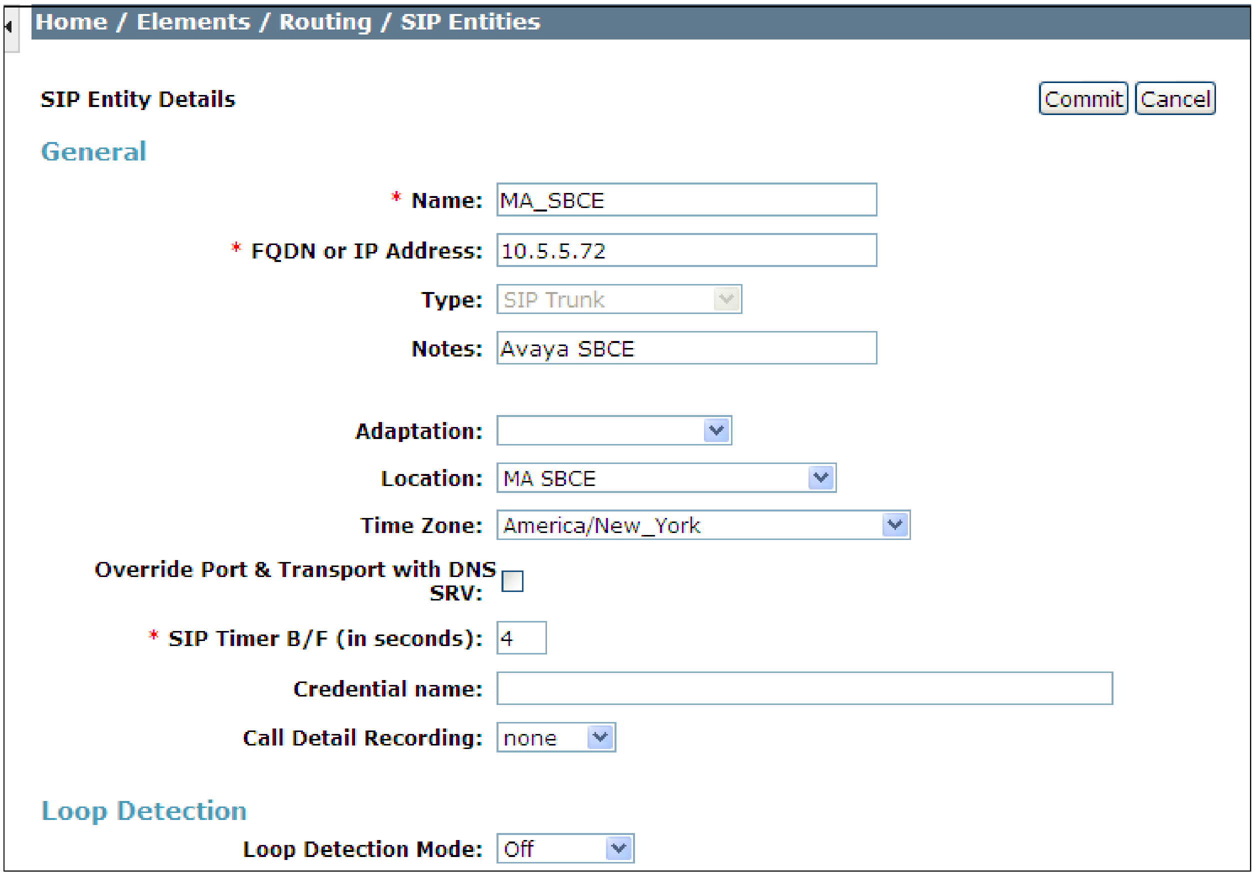
To define the ports that Session Manager will use to listen for SIP requests, scroll down to the **Port** section of the **SIP Entity Details** screen. This section is only present for **Session Manager** SIP entities. The screen below shows the ports used by Session Manager in the shared lab environment. TCP ports 5060 and 5075 are the ones directly relevant to the SIP trunk to Broadvox in the reference configuration.



The following screen shows the addition of the SIP Entity for Communication Manager. In order for Session Manager to send SIP service provider traffic on a separate entity link to Communication Manager, the creation of a separate SIP entity for Communication Manager is required. This SIP Entity should be different to the one created during the Session Manager installation, to be used by all other enterprise SIP traffic. The **FQDN or IP Address** field is set to the IP address of the "**procr**" interface in Communication Manager.



The following screen shows the addition of the Avaya SBCE Entity. The **FQDN or IP Address** field is set to the IP address of the SBC private network interface (see **Figure 1**).



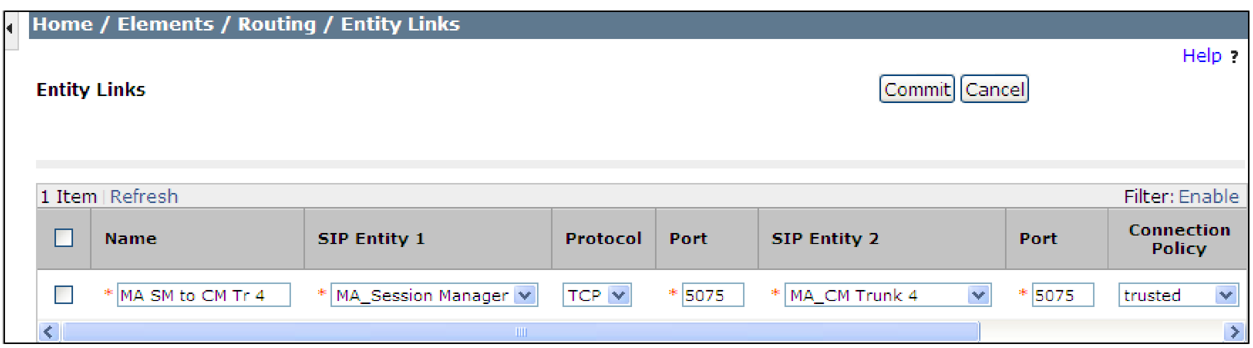
6.5. Entity Links

A SIP trunk between Session Manager and a telephony system is described by an Entity Link. Two Entity Links were created; one to the Communication Manager for use only by service provider traffic and one to the Avaya SBCE. To add an Entity Link, navigate to **Routing → Entity Links** in the left navigation pane and click on the **New** button in the right pane (not shown). Fill in the following fields in the new row that is displayed:

- **Name:** Enter a descriptive name.
- **SIP Entity 1:** Select the Session Manager from the drop-down menu.
- **Protocol:** Select the transport protocol used for this link.
- **Port:** Port number on which Session Manager will receive SIP requests from the far-end.
- **SIP Entity 2:** Select the name of the other system from the drop-down menu.
- **Port:** Port number on which the other system receives SIP requests from Session Manager.
- **Connection Policy:** Select **Trusted** to allow calls from the associated SIP Entity.

Click **Commit** to save.

The screen below shows the Entity Link to Communication Manager. The protocol and ports defined here must match the values used on the Communication Manager signaling group form in **Section 5.6**.



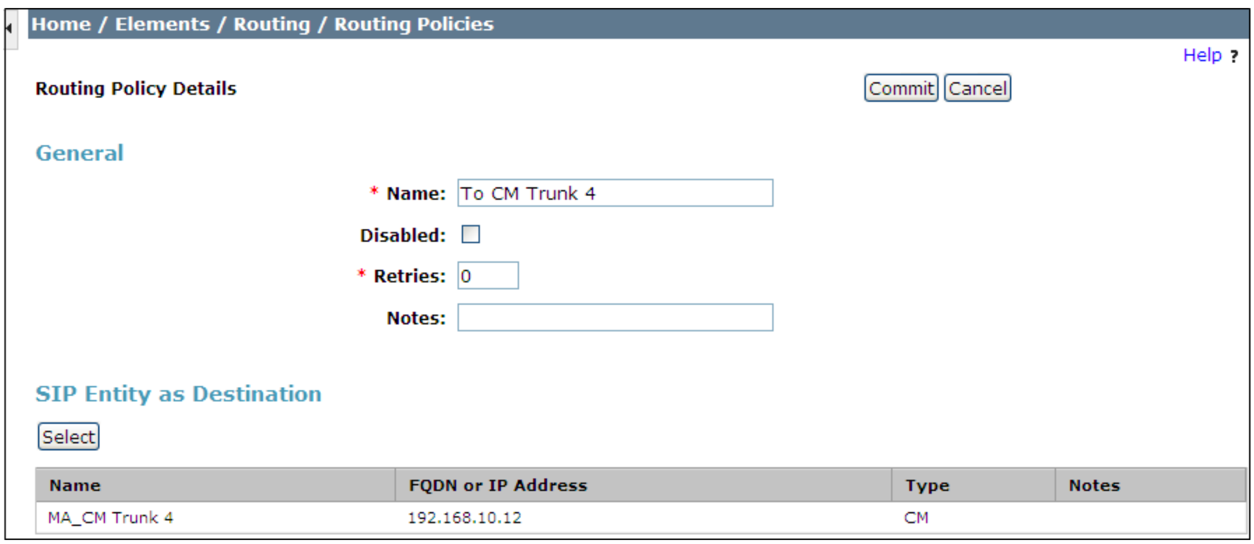
Entity Link to the Avaya SBCE:

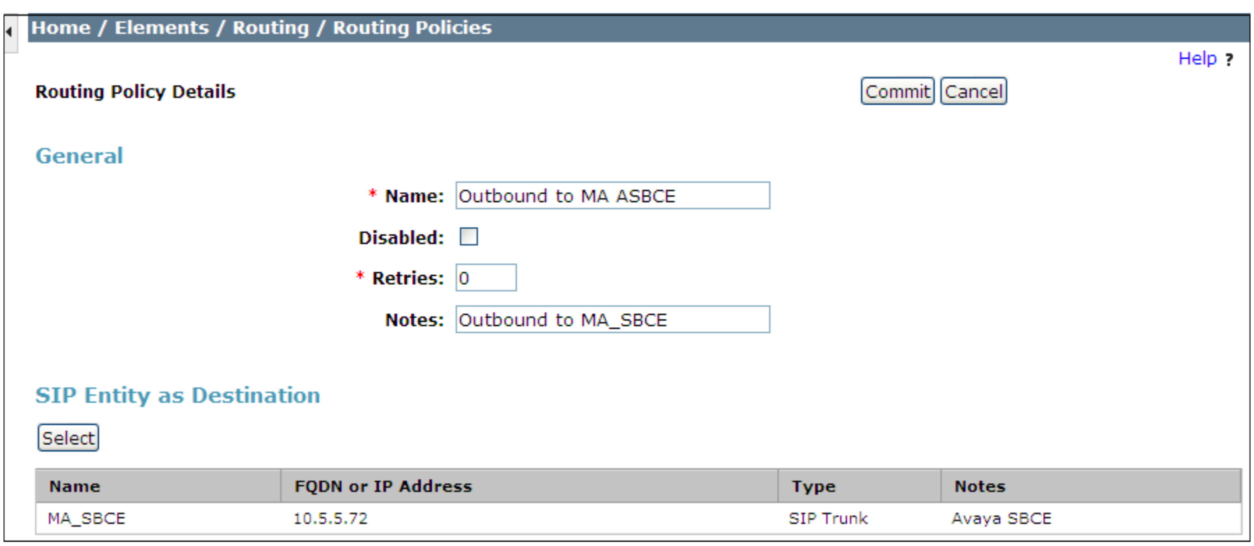
6.6. Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.4**. Two routing policies must be added; one for Communication Manager and one for the Avaya SBCE. To add a routing policy, navigate to **Routing → Routing Policies** in the left navigation pane and click on the **New** button in the right pane (not shown). The following screen is displayed. In the **General** section, enter a descriptive **Name** and add a brief description under **Notes** (optional).

In the **SIP Entity as Destination** section, click **Select**. The **SIP Entity List** page opens (not shown). Choose the appropriate SIP entity to which this routing policy applies and click **Select**. The selected SIP Entity displays on the **Routing Policy Details** page as shown below. Use default values for remaining fields. Click **Commit** to save.

The following screens show the Routing Policies for Communication Manager and the Avaya SBCE





6.7. Dial Patterns

Dial Patterns are needed to route specific calls through Session Manager. For the compliance test, dial patterns were needed to route calls from Communication Manager to Broadvox and vice versa. Dial Patterns define which route policy will be selected for a particular call based on the dialed digits, destination domain and originating location. To add a dial pattern, navigate to **Routing → Dial Patterns** in the left navigation pane and click on the **New** button in the right pane (not shown). Fill in the following, as shown in the screens below:

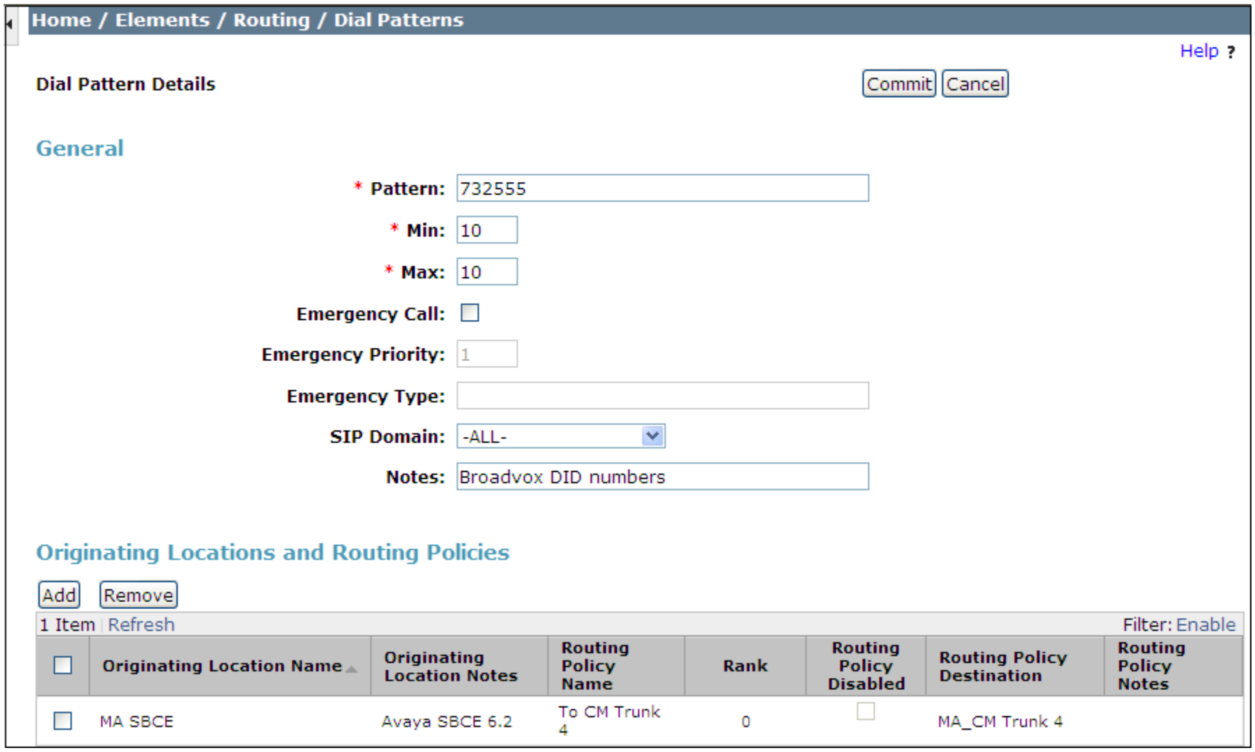
In the **General** section, enter the following values:

- **Pattern:** Enter a dial string that will be matched against the Request-URI of the call.
- **Min:** Enter a minimum length used in the match criteria.
- **Max:** Enter a maximum length used in the match criteria.
- **SIP Domain:** Enter the destination domain used in the match criteria, or select **"ALL**" to route incoming calls to all SIP domains.
- **Notes:** Add a brief description (optional).

In the **Originating Locations and Routing Policies** section, click **Add**. From the **Originating Locations and Routing Policy List** that appears (not shown), select the appropriate originating location for use in the match criteria. Lastly, select the routing policy from the list that will be used to route all calls that match the specified criteria. Click **Select**.

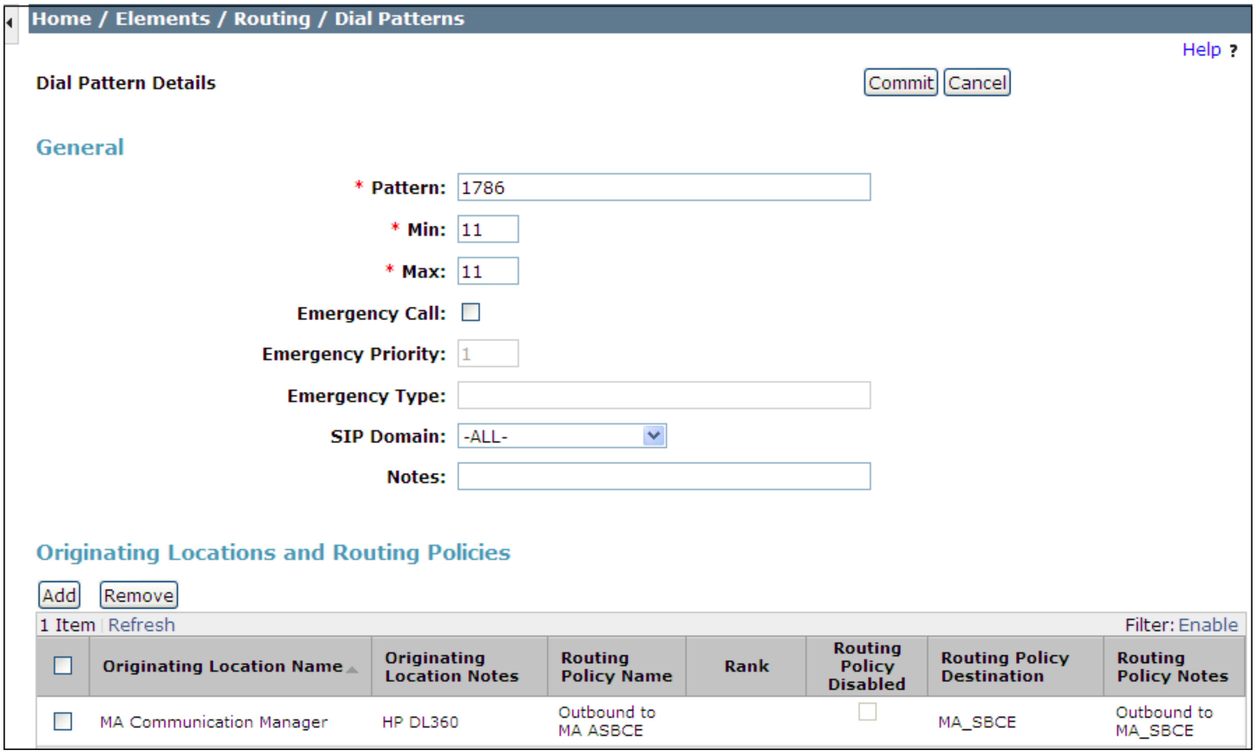
Default values can be used for the remaining fields. Click **Commit** to save.

The following screen illustrates an example dial pattern used to verify inbound PSTN calls to the enterprise. In the example, calls to 10 digit numbers starting with ***732555***, which is the DID range assigned by Broadvox to the SIP trunk, arriving from location **MA SBCE**, under **Originating Location Name**, will use route policy **To CM Trunk 4** to Communication Manager.



Repeat this procedure as needed to define additional dial patterns for other numbers assigned by Broadvox to the enterprise, to be routed to Communication Manager.

The example in this screen shows that 11 digit dialed numbers for outbound calls, beginning with a number such as ***1786*** used for test purposes during the compliance test, arriving from the **MA Communication Manager** location, under **Originating Location Name**, will use route policy **Outbound to MA ASBCE**, which sends the call out to the PSTN via the Avaya SBCE to the Broadvox SIP Trunk.



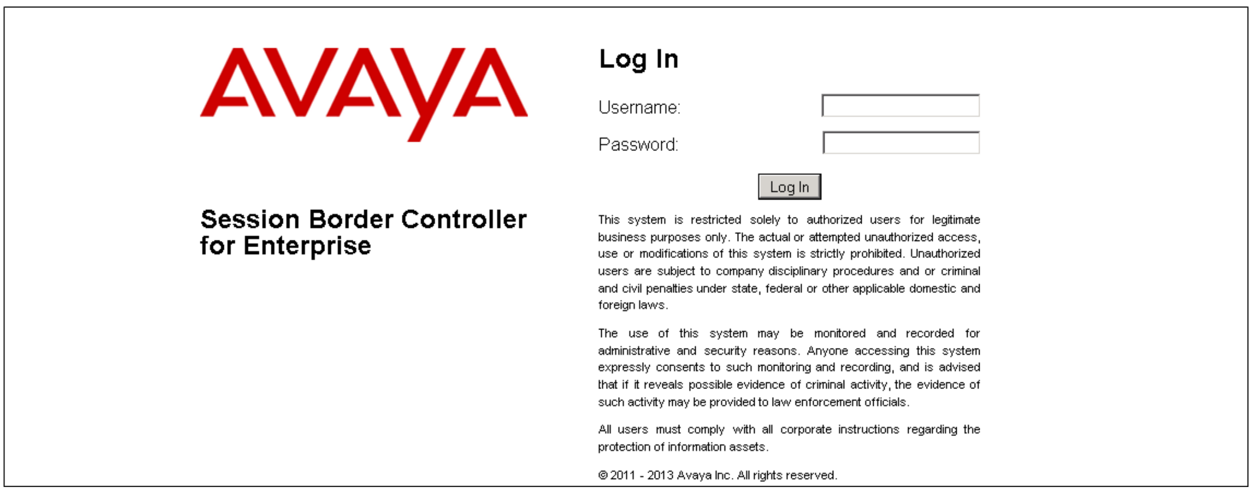
Repeat this procedure as needed, to define additional dial patterns for PSTN numbers to be routed to the Broadvox network via the Avaya SBCE.

7. Configure Avaya Session Border Controller for Enterprise

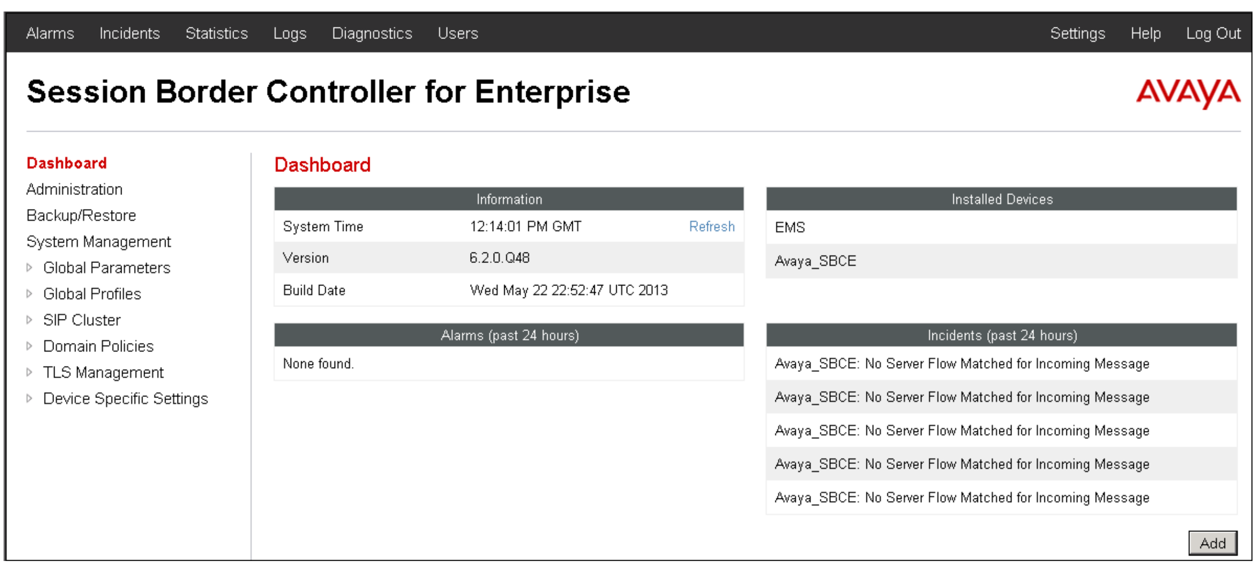
In the sample configuration, the Avaya SBCE is used as the edge device between the Avaya CPE and the Broadvox SIP Trunking service. It is assumed that the initial installation of the Avaya SBCE and the assignment of the management interface IP Address have already been completed; hence these tasks are not covered in these Application Notes. For more information on the SBC installation and initial provisioning, consult the Avaya SBCE documentation listed in the **References** section.

7.1. System Access

Access the Session Border Controller web management interface by using a web browser and entering the URL **https://<ip-address>**, where **<ip-address>** is the management IP address configured at installation. Log in using the appropriate credentials.

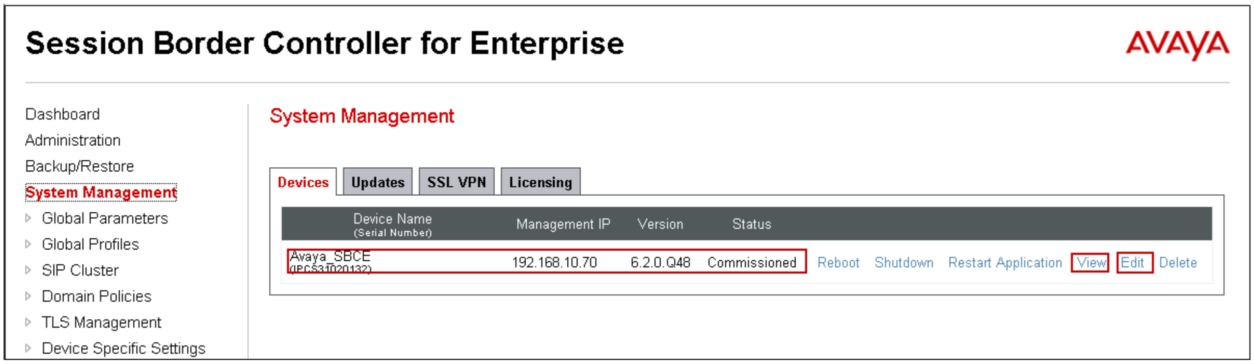


Once logged in, the Dashboard screen is presented. The left navigation pane contains the different available menu items used for the configuration of the Avaya SBCE.

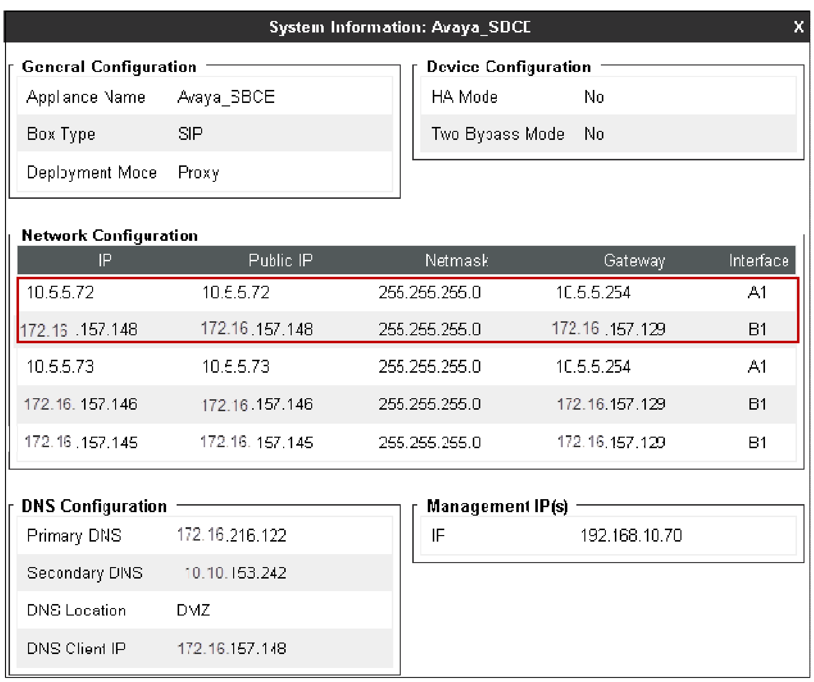


7.2. System Management

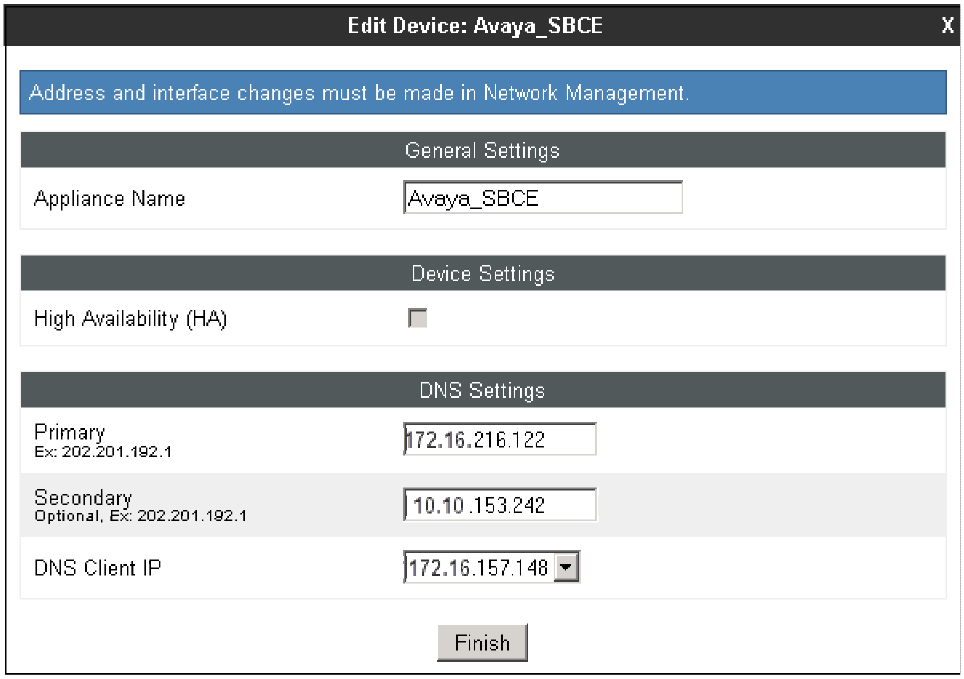
To view current system information, select **System Management** on the left navigation pane. A list of installed devices is shown in the **Devices** tab on the right pane. In the reference configuration, a single device named **Avaya_SBCE** is shown. The management IP address that was configured during installation and the current software version are shown here. Note that the management IP address needs to be on a subnet separate from the ones used in the other interfaces of the Avaya SBCE, segmented from all VoIP traffic. Verify that the **Status** is **Commissioned**, indicating that the initial installation process of the device has been previously completed, as shown on the screen below.



To view the network information assigned to the Avaya SBCE, click **View** on the screen above. The **System Information** window is displayed, containing the current device and the network settings. Note that the **A1** and **B1** interfaces correspond to the inside and outside interfaces for the Avaya SBCE. The highlighted **A1** and **B1** IP addresses are the ones relevant to the configuration of the SIP trunk to Broadvox. Other IP addresses assigned to these interfaces on the screen below are used to support remote workers and they are not discussed in this document.



DNS server information can be entered or modified if needed, by clicking **Edit** on the **System Management/Devices** tab shown on the previous page. On the screen below, note that even though public DNS servers were used in the reference configuration, for security reasons the public IP addresses of the **Primary** and **Secondary** DNS servers have been masked, and private IP addresses are shown. The **DNS client** IP, also masked, corresponds to the **B1** interface used for SIP trunking, Click **Finish** when done.



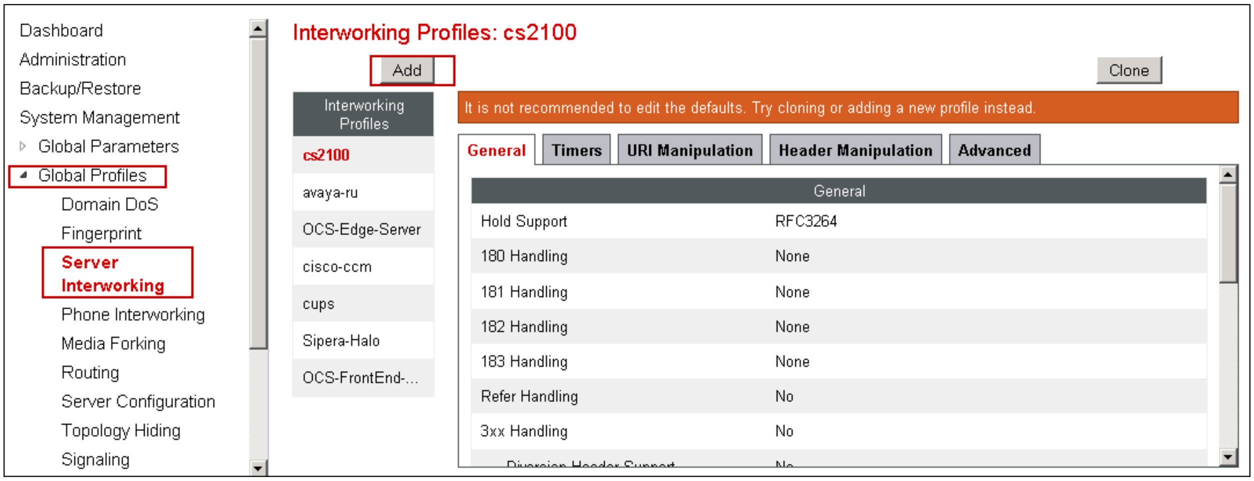
7.3. Global Profiles

The Global Profiles Menu, on the left navigation pane, allows the configuration of parameters across all UC-Sec appliances.

7.3.1. Server Interworking

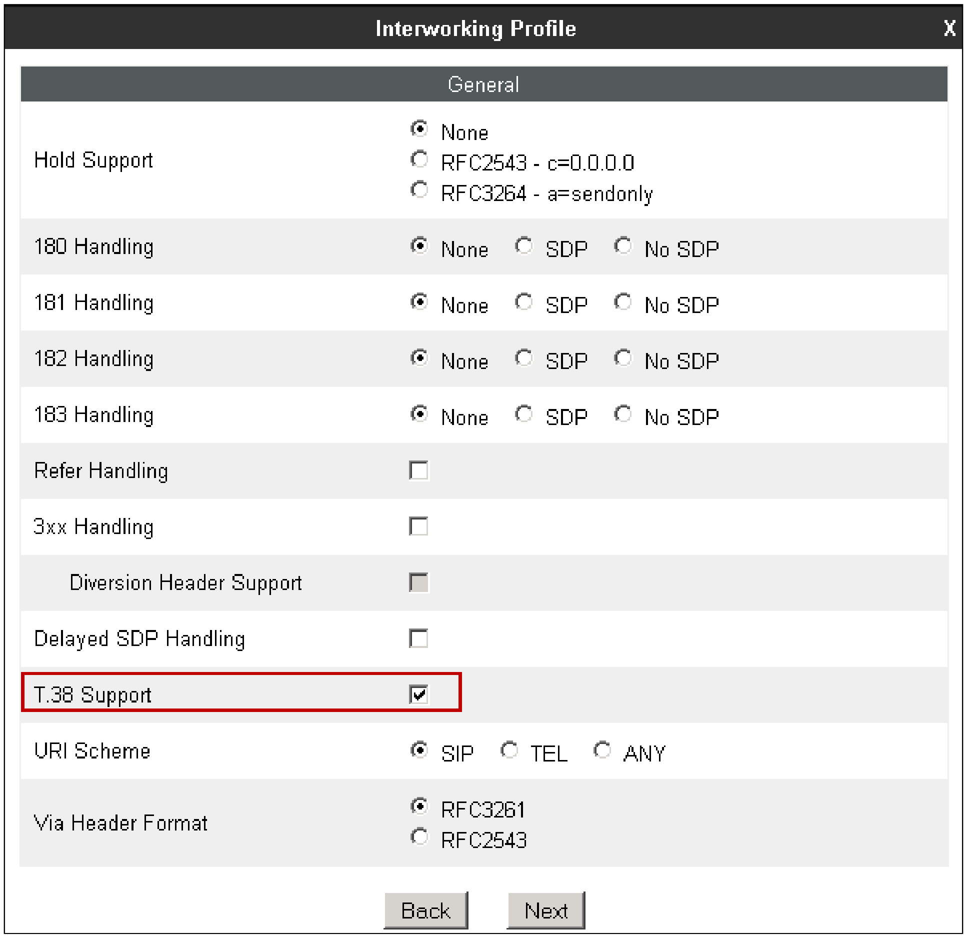
Interworking Profile features are configured to facilitate the interoperability between the enterprise SIP-enabled solution (Call Server) and the SIP trunk service provider (Trunk Server). In the reference configuration, Session Manager functions as the Call Server and the Broadvox SIP Proxy as the Trunk Server.

To configure the interworking profile in the enterprise direction, select **Global Profiles → Server Interworking** on the left navigation pane. Click **Add**.

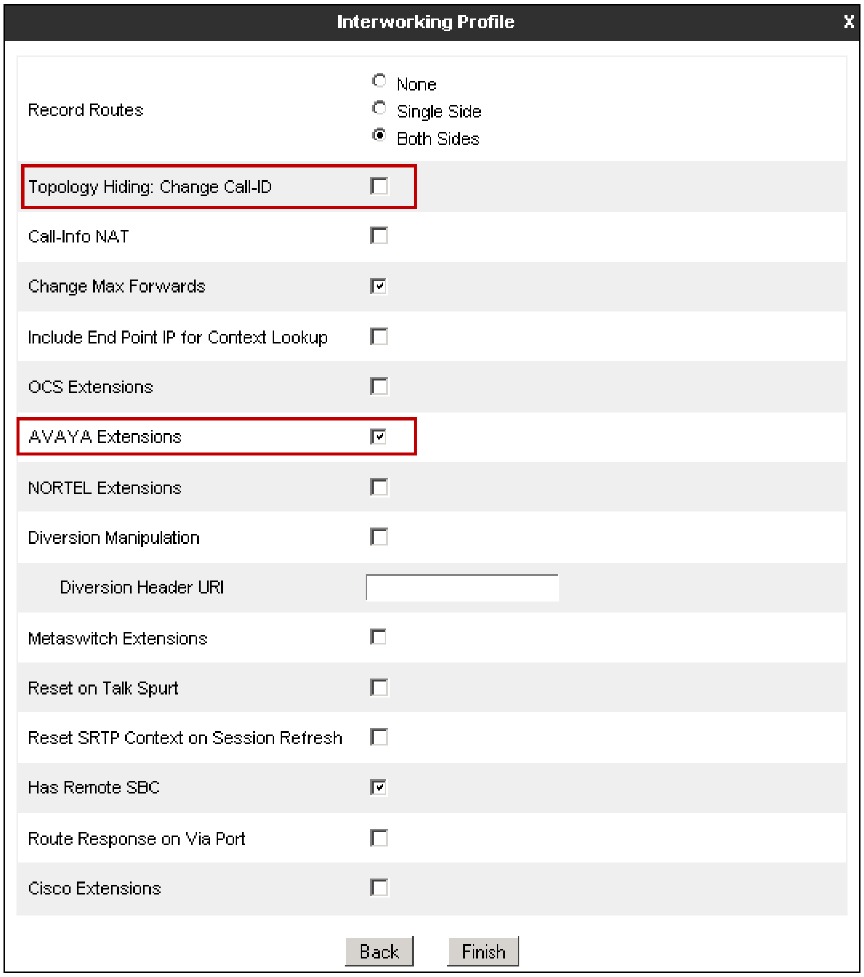


Enter a descriptive name for the new profile. Click **Next**.

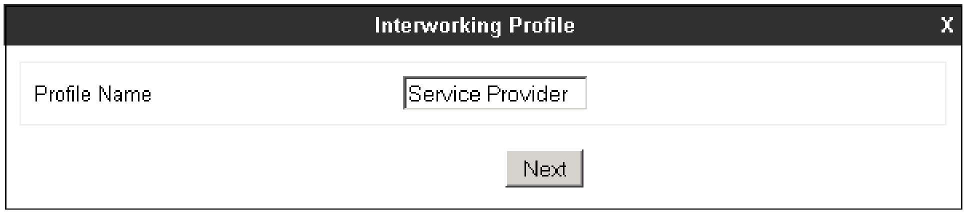
On the **General** screen, check the **T.38 Support box**. All other parameters retain their default values. Click **Next**.



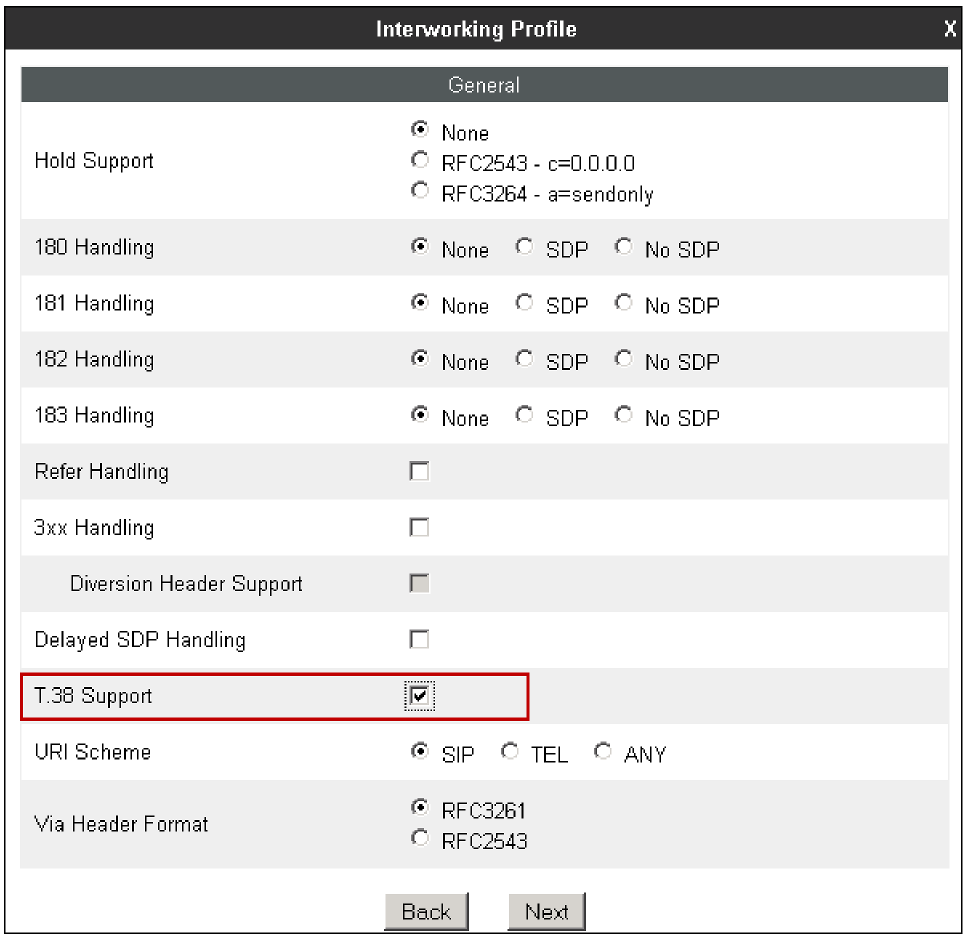
Click **Next** on the **Privacy/DTMF** and **SIP Timers/Transport Timers** tabs (not shown). On the **Advanced Settings** tab, uncheck the **Topology Hiding: Change Call-ID** box and check the **AVAYA Extensions** box. Click **Finish** to save and exit.



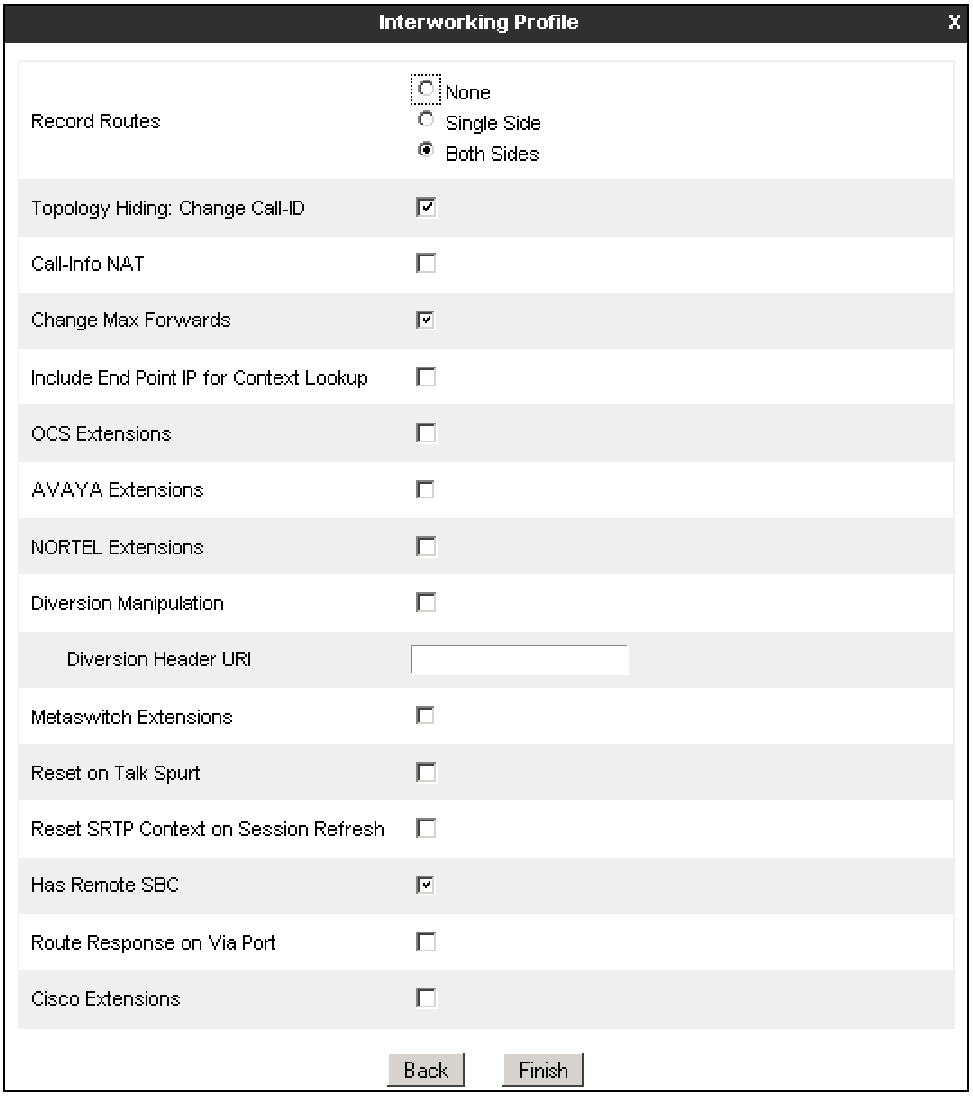
A second interworking profile named ***Service Provider*** in the direction of the SIP trunk to Broadvox was similarly created. For this profile default values were used for all parameters except for **T.38** Support, which was enabled.



General tab:

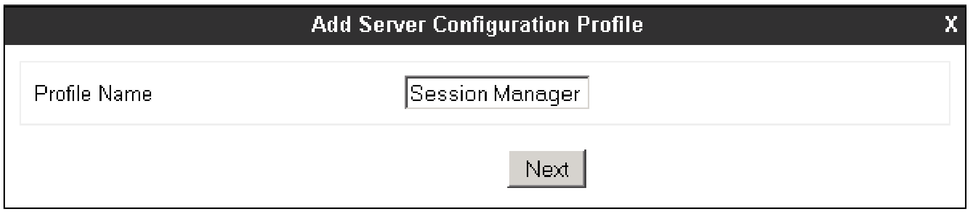


Advanced Settings tab:

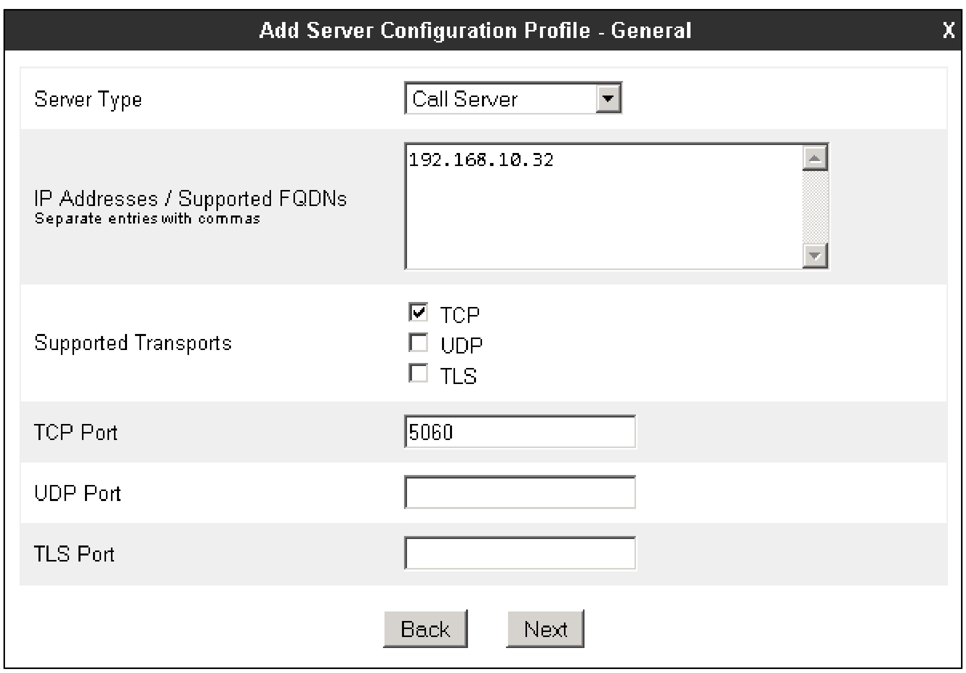


7.3.2. Server Configuration

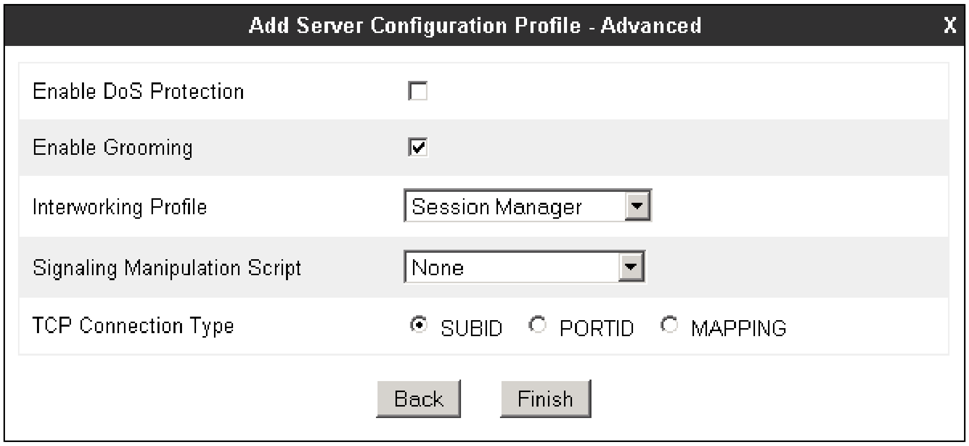
Server Profiles are created to define the parameters for the Avaya SBCE two peers, i.e., Session Manager (Call Server) and the SIP Proxy at the service provider's network (Trunk Server). From the **Global Profiles** menu on the left-hand navigation pane, select **Server Configuration** and click the **Add** button (not shown) to add a new profile for the Call Server. Enter an appropriate **Profile Name** similar to the screen below. Click **Next**.



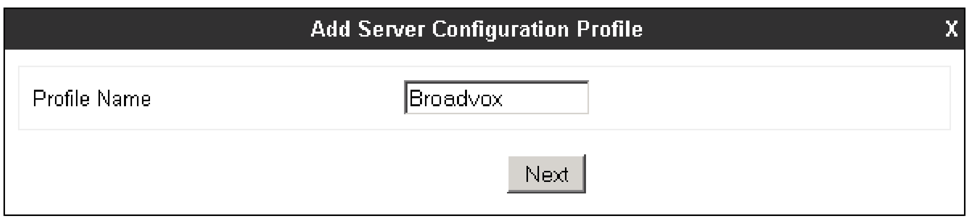
On the **Add Server Configuration Profile - General** Tab select ***Call Server*** from the drop down menu for the **Server Type**. On the **IP Addresses / Supported FQDNs** field, enter the IP address of the Session Manager Security Module. Select **TCP** for **Supported Transports**, and enter ***5060*** under **TCP Port**. The transport protocol and port selected here must match the values defined for the Session Manager SIP entity in **Section 6.4**. Click **Next**.



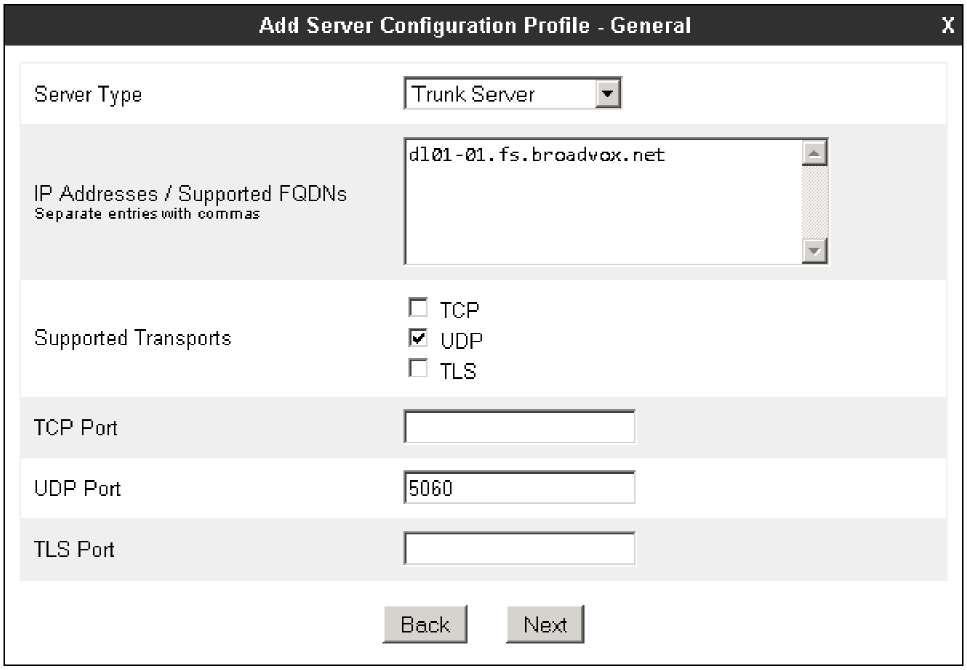
Click **Next** on the **Authentication** and **Heartbeat** tabs (not shown). On the **Advanced** tab, since TCP is used, check the **Enable Grooming** box. Select ***Session Manager*** from the **Interworking Profile** drop down menu. Click **Finish**.



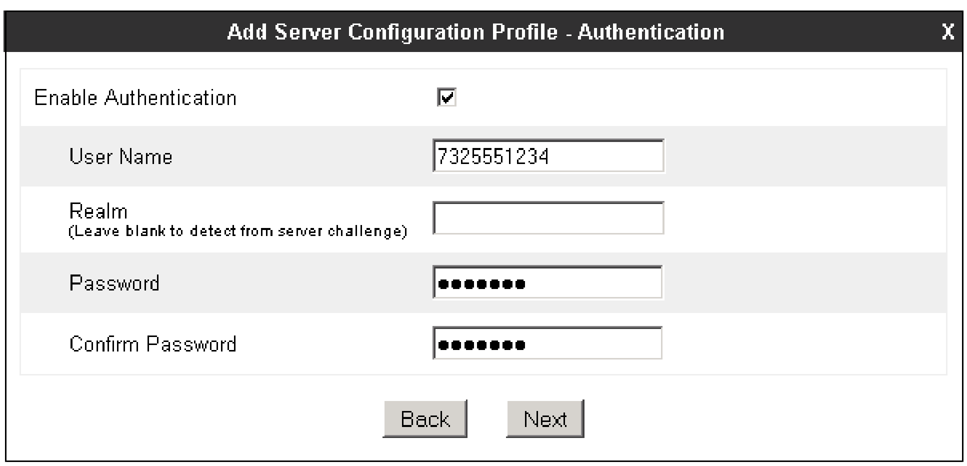
Similarly, to add the profile for the Trunk Server, click the **Add** button on the **Server Configuration** screen (not shown). Enter an appropriate **Profile Name** similar to the screen below. Click **Next**.



On the **Add Server Configuration Profile-General** Tab select ***Trunk Server*** from the drop down menu for the **Server Type**. On the **IP Addresses / Supported FQDNs** field, enter ***dl01-01.fs.broadvox.net***, the fully qualified domain name of the Broadvox SIP proxy server. Select **UDP** for **Supported Transports**, and enter ***5060*** under **UDP Port**, as specified by Broadvox.

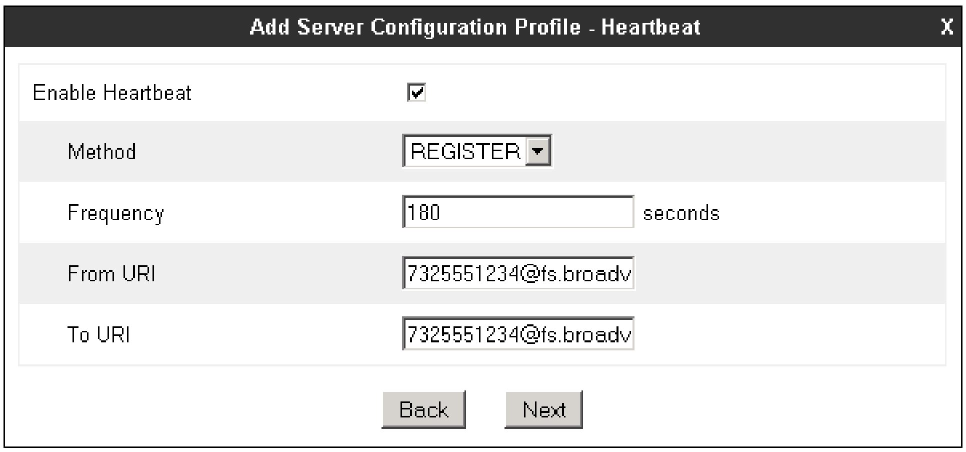


On the **Authentication** tab, check the **Enable Authentication** box. Enter the **User Name**, and **Password** credential information supplied by Broadvox for the authentication of the SIP trunk. Leave the **Realm** field blank. Click **Next**.

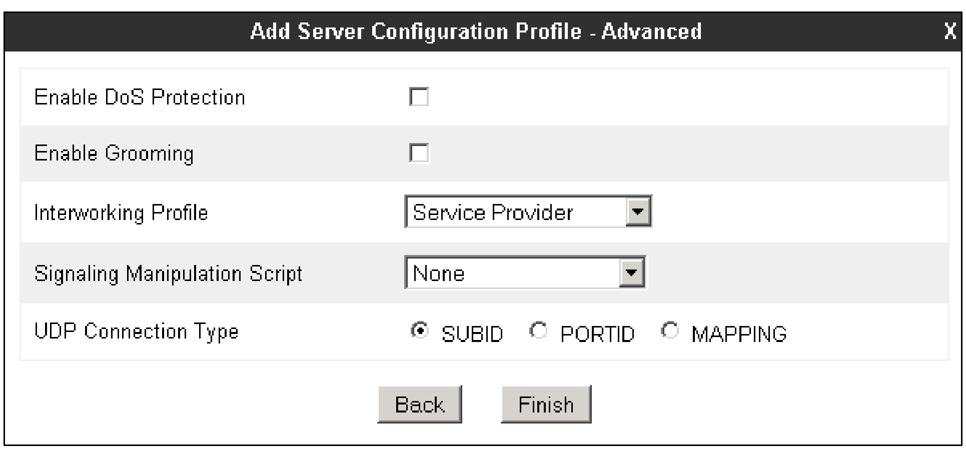


On the **Heartbeat** tab:

- Check the **Enable Heartbeat** box.
- Under **Method**, select **REGISTER** from the drop down menu.
- **Frequency:** Enter the amount of time (in seconds) between REGISTER messages that will be sent from the enterprise to the Broadvox proxy server in order to refresh the registration binding of the SIP trunk. This value should be chosen in consultation with the service provider. ***180*** seconds was the value used during the compliance test.
- The **From URI** and **To URI** entries for the REGISTER messages are built using the **User Name** entered in the **Authentication** screen, and the service's provider domain ***fs.broadvox.net***, like shown on the example below.
- Click **Next**.



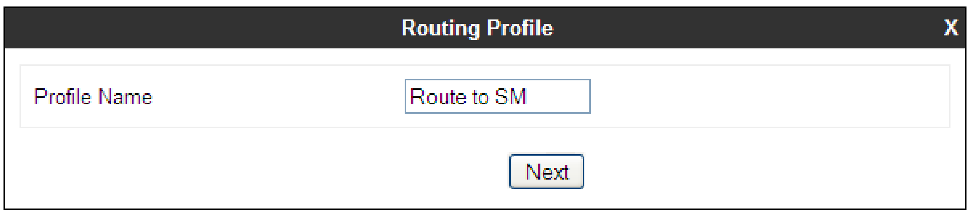
On the **Advanced** tab, select ***Service Provider*** from the **Interworking Profile** drop down menu. Click **Finish**



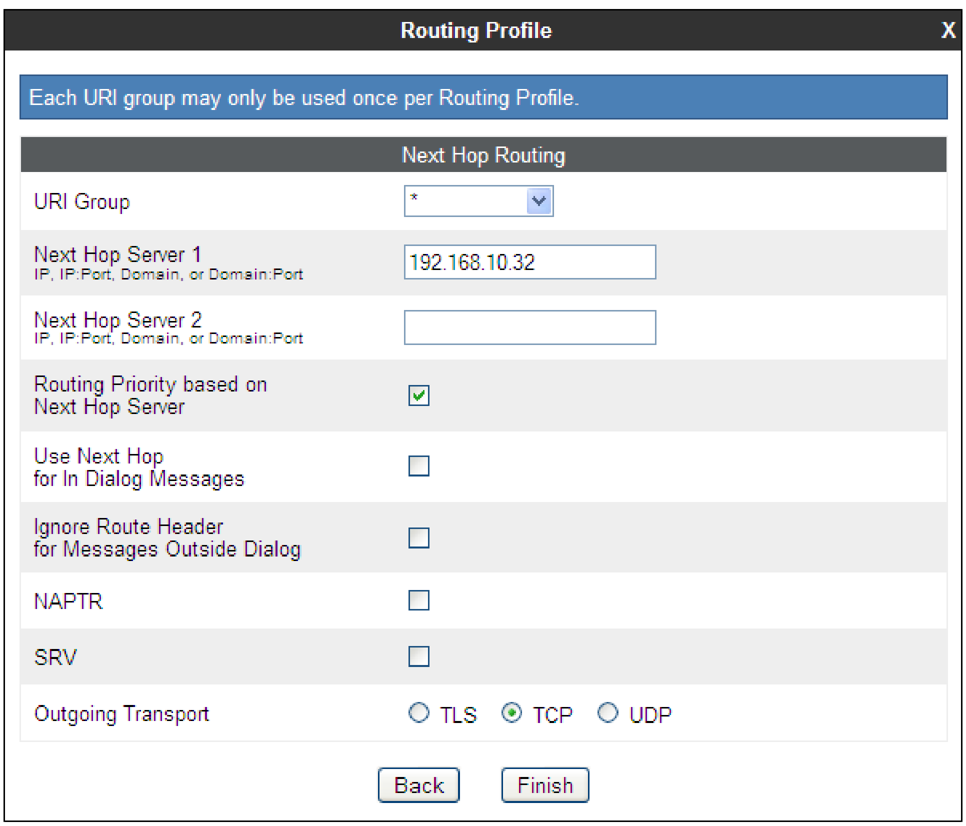
7.3.3. Routing Profiles

Routing profiles define a specific set of routing criteria that is used, in addition to other types of domain policies, to determine the path that the SIP traffic will follow as it flows through the Avaya SBCE interfaces.

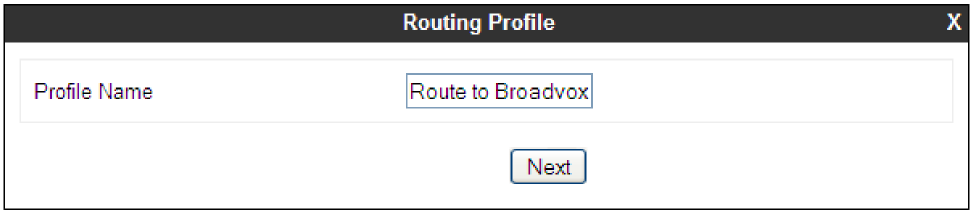
Two Routing Profiles were created in the test configuration, one for inbound calls, with Session Manager as the destination, and the second one for outbound calls, which are routed to the Broadvox SIP trunk. To create the inbound route, select the **Routing** tab from the **Global Profiles** menu on the left-hand side and select **Add** (not shown). Enter an appropriate **Profile Name** similar to the example below. Click **Next**.



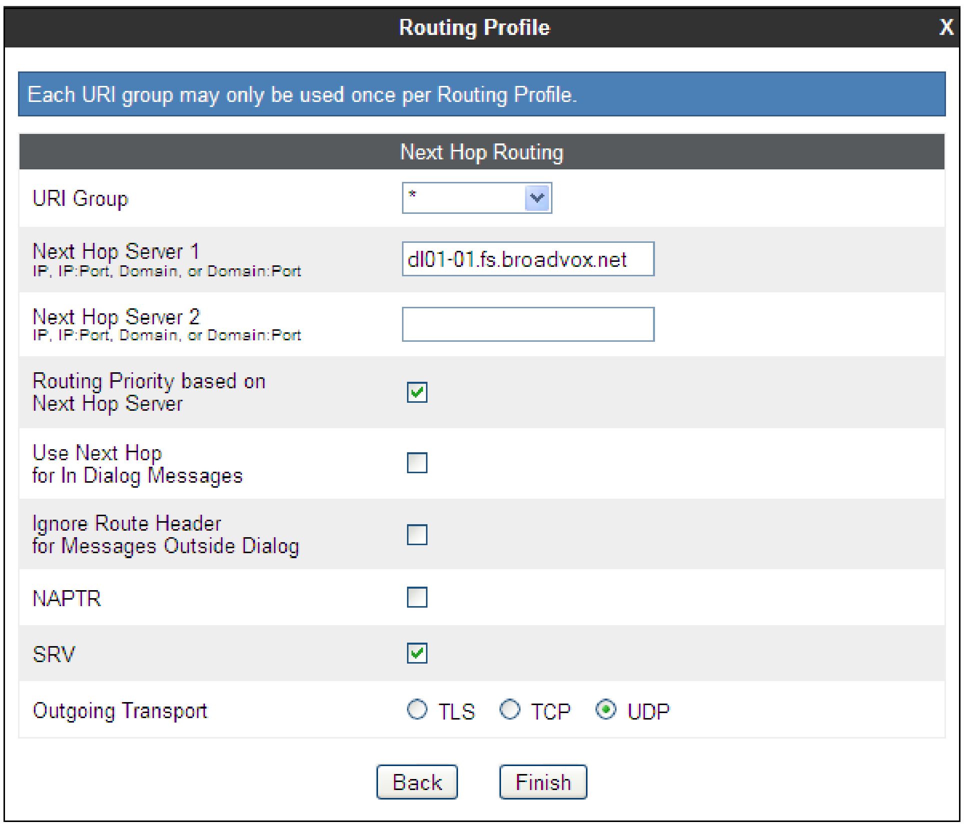
On the **Next Hop Routing** tab, enter the IP Address of Session Manager as **Next Hop Server 1**. Since the default well-known port value of 5060 for TCP was used, it is not necessary to enter the port number here. Check **Routing Priority based on Next Hop Server**. Choose **TCP** for **Outgoing Transport**. Click **Finish**.



Back at the **Routing** tab, select **Add** (not shown) to repeat the process in order to create the outbound route. Enter an appropriate **Profile Name**. Click **Next**.



On the Next Hop Routing tab, enter the FQDN of the service provider SIP proxy server as **Next Hop Server 1**. Since the default well-known port value of 5060 for UDP was used, it is not necessary to enter the port number here. Check the **Routing Priority based on Next Hop Server**. Check the **SRV** box. Choose **UDP** for **Outgoing Transport**. Click **Finish**.



7.3.4. Topology Hiding

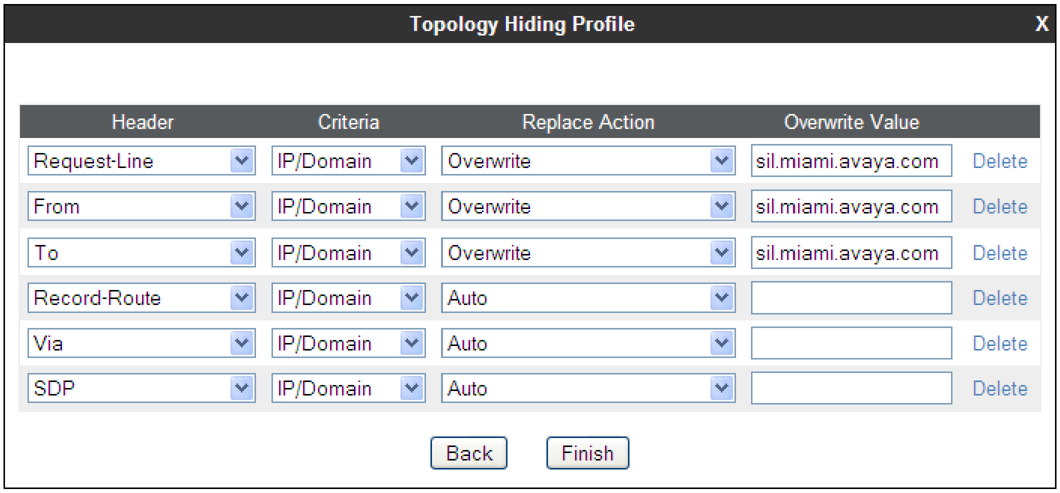
Topology Hiding is a security feature that allows the modification of several SIP headers, preventing private enterprise network information from being propagated to the untrusted public network.

Topology Hiding can also be used as an interoperability tool to adapt the host portion in the SIP headers to the IP addresses or domains expected on the service provider and the enterprise networks. For the compliance test, only the minimum configuration required to achieve interoperability on the SIP trunk was performed. Additional steps can be taken in this section to further mask the information that is sent from the enterprise to the public network.

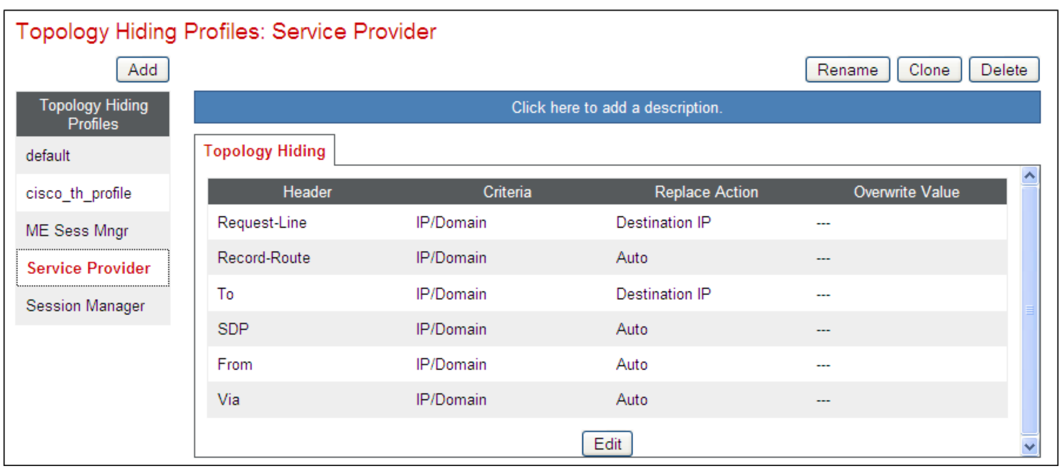
To add the **Topology Hiding Profile** in the enterprise direction, select **Topology Hiding** from the **Global Profiles** menu on the left-hand side and click the **Add** button (not shown). Enter a **Profile Name** such as the one shown below. Click **Next**.

On the **Topology Hiding Profile** screen, click the **Add Header** button repeatedly to show the rest of the headers in the profile.

For the **Request-Line**, **From** and **To** headers, select ***Overwrite*** in the **Replace Action** column and enter the enterprise SIP domain know by the Session Manager, ***sil.miami.avaya.com***, in the **Overwrite Value** column of these headers, as shown below. Default values were used for all other fields. Click **Finish**.



A Topology Hiding profile named **Service Provider** was similarly configured in the direction of the SIP trunk to Broadvox. During the compliance tests, IP addresses were used in the domain part of headers in messages between the Broadvox SIP Proxy and the Avaya SBCE. For the **Request-Line** and **To** headers, ***Destination IP*** was selected under the **Replace Action** column.



7.4. Domain Policies

Domain Policies allow the configuration of sets of rules designed to control and normalize the behavior of call flows, based upon various criteria of communication sessions originating from or terminating in the enterprise. Domain Policies include rules for Application, Media, Signaling, Security, etc.

In the reference configuration, only a new Signaling Rule was defined. All other rules under Domain Policies, linked together on End Point Policy Groups later in this section, used one of the default sets already pre-defined in the configuration. Please note that changes should not be made to any of the defaults. If changes are needed, it is recommended to create a new rule by cloning one the defaults and then make the necessary changes to the new rule.

7.4.1. Signaling Rules

A Signaling Rule was created in the sample configuration to remove (block) the following headers:

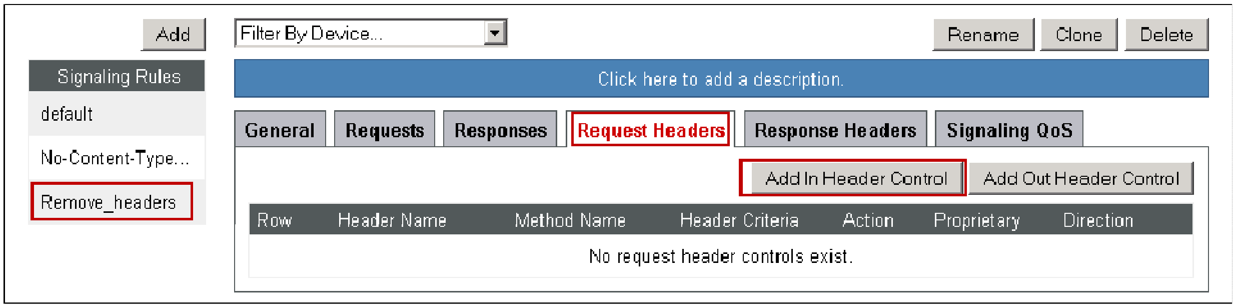
- AV-Global-Session-ID
- Alert-Info
- Endpoint-View
- P-AV-Message-ID
- P-Location
- P-Charging-Vector

These headers are sent in SIP messages from the Session Manager to the Avaya SBCE. They contain private IP addresses and SIP Domains from the enterprise, which should not be propagated outside of the enterprise boundaries.

In the **Domain Policies** menu on the left-hand side, select **Signaling Rules**, then **Add Rule** (not shown). Enter an appropriate name like in the example below. Click **Next**.

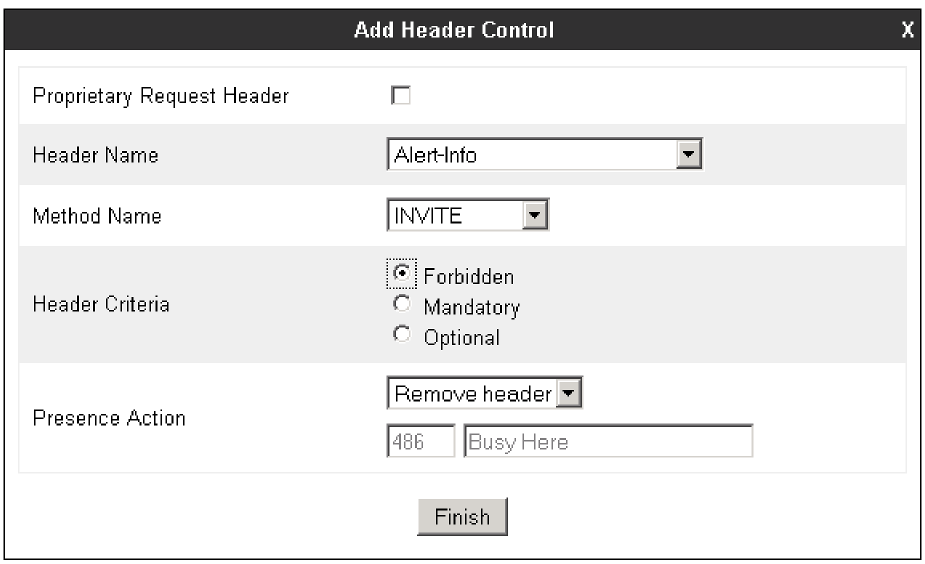
On the next three pages (not shown), leave sections **Inbound**, **Outbound** and **Content-Type Policies** with their default values. Click **Next**. On the **Signaling QoS** tab**,** default values were used. Click **Finish**.

On the newly created **Remove_headers** Signaling Rule, select the **Request Headers** tab to create the manipulations performed on request messages. Select **Add In Header Control**.

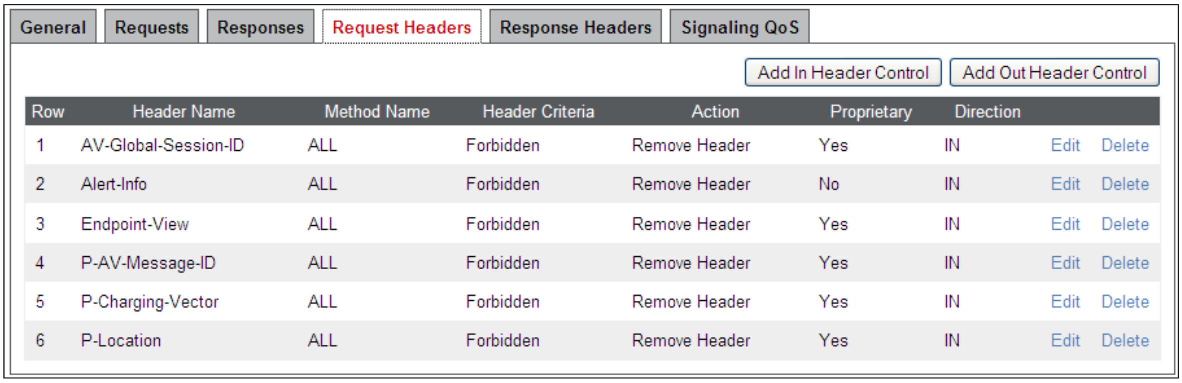


In the **Add Header Control** screen select the following:

- **Header Name: Alert-Info**
- **Method Name: INVITE**
- **Header Criteria: Forbidden**
- **Presence Action: Remove Header**
- Click **Finish**



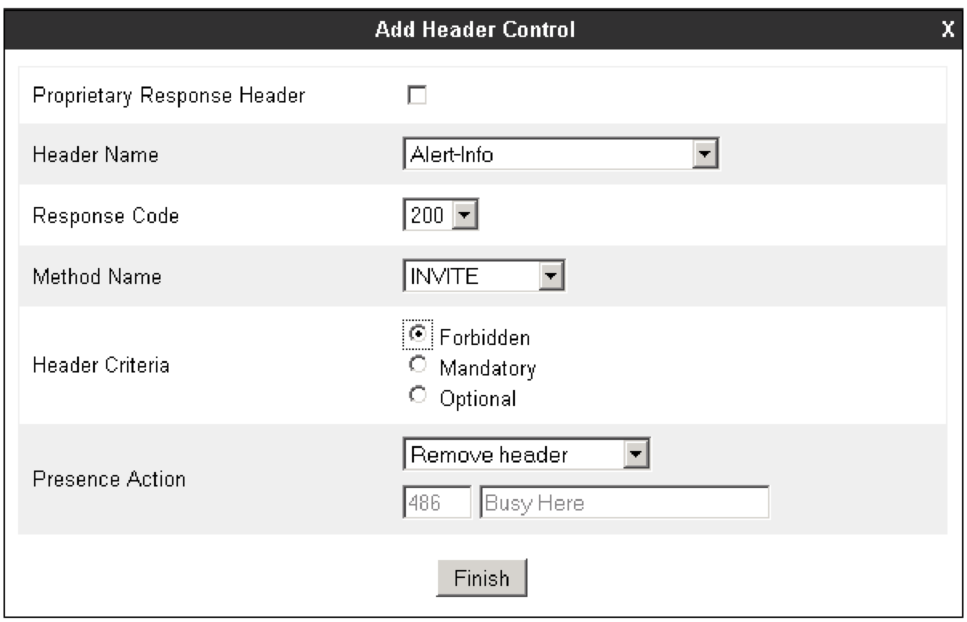
Select **Add In Header Control** as needed to configure the remaining header control rules. For these headers, make sure to check the **Proprietary Request Header** box in the **Add Header Control** tab. This will allow typing the name of the specific header on the **Header Name** box. Once completed, the **Request Headers** tab should look like the following screen.



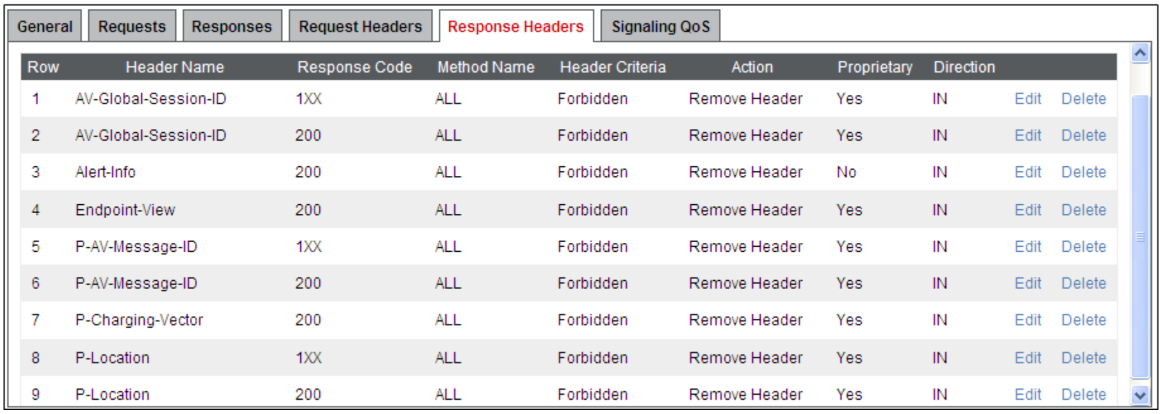
Row	Header Name	Method Name	Header Criteria	Action	Proprietary	Direction		
1	AV-Global-Session-ID	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
2	Alert-Info	ALL	Forbidden	Remove Header	No	IN	Edit	Delete
3	Endpoint-View	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
4	P-AV-Message-ID	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
5	P-Charging-Vector	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
6	P-Location	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete

Select the **Response Headers** tab to similarly create the manipulations performed on response messages. Select **Add In Header Control** (not shown).

The screen below shows the settings for the Alert-Info header on response messages.



Select **Add In Header Control** as needed to configure the remaining header control rules. For these headers, make sure to check the **Proprietary Request Header** box in the **Add Header Control** tab. This will allow typing the name of the specific header on the **Header Name** box. Once completed, the **Response Headers** tab should look like the following screen.



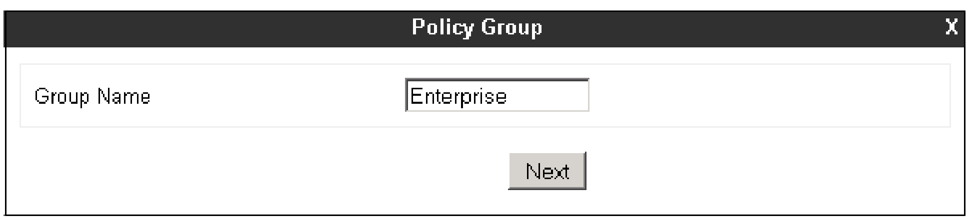
Row	Header Name	Response Code	Method Name	Header Criteria	Action	Proprietary	Direction		
1	AV-Global-Session-ID	1XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
2	AV-Global-Session-ID	200	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
3	Alert-Info	200	ALL	Forbidden	Remove Header	No	IN	Edit	Delete
4	Endpoint-View	200	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
5	P-AV-Message-ID	1XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
6	P-AV-Message-ID	200	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
7	P-Charging-Vector	200	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
8	P-Location	1XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
9	P-Location	200	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete

7.4.2. End Point Policy Groups

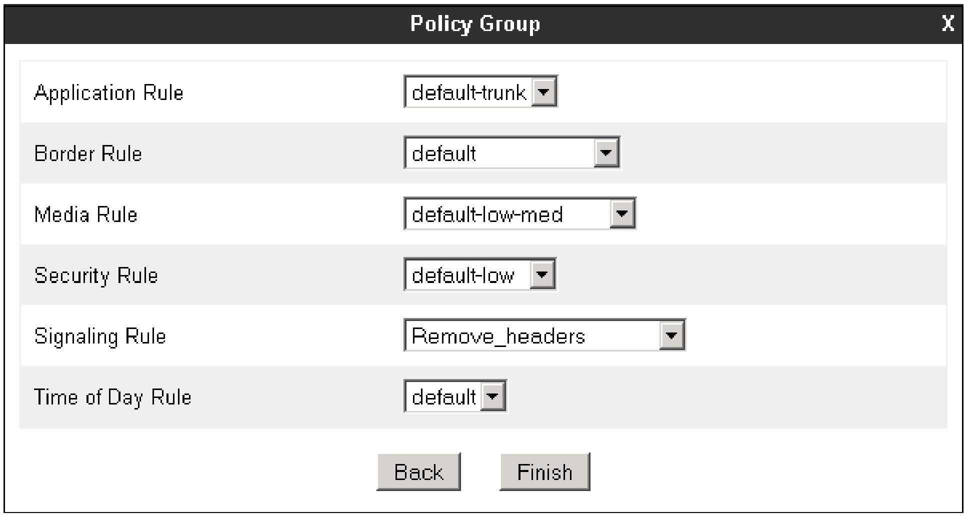
End Point Policy Groups associate the different sets of rules (Media, Signaling, Security, etc) to be applied to specific SIP messages traversing through the Avaya SBCE.

To create an End Point Policy Group for the enterprise, select **End Point Policy Groups** under the **Domain Policies** menu. Select **Add** (not shown).

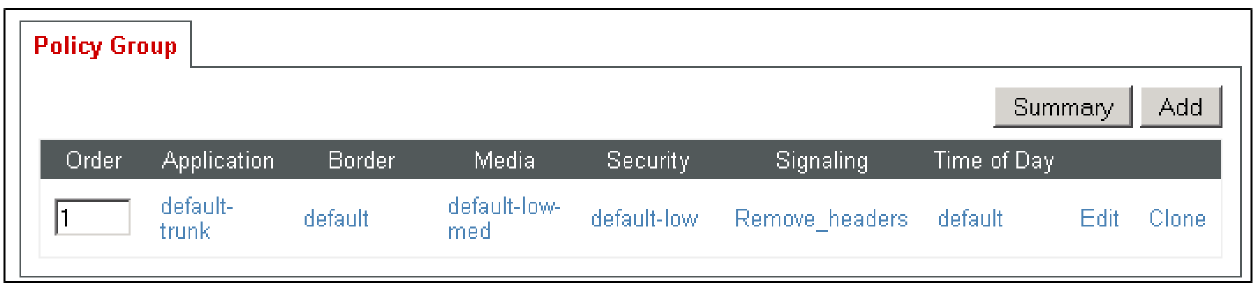
Enter an appropriate name in the **Group Name** field. ***Enterprise*** was used. Click **Next**.



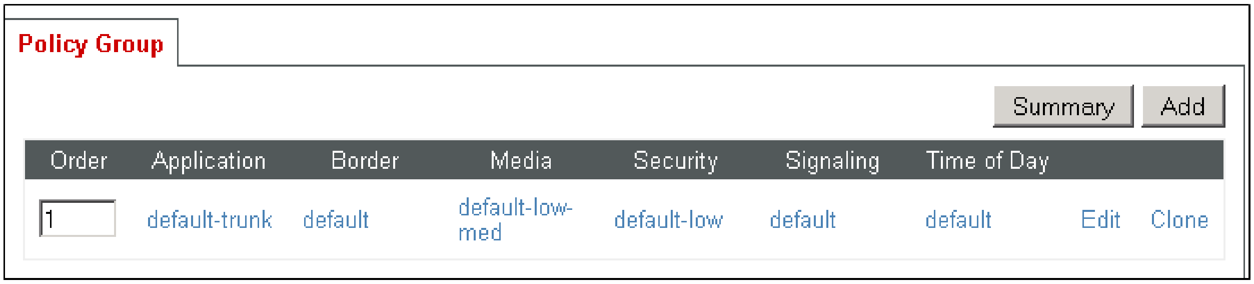
In the Policy Group tab, all fields used one of the default sets already pre-defined in the configuration, with the exception of the **Signaling Rule**, where the ***Remove_headers*** rule created in **Section 7.4.1** was selected. Click **Finish**.



The screen below shows the **Enterprise** End Point Policy Group after the configuration was completed.



A second End Point Policy Group was created for the service provider, repeating the steps described above. Defaults were used in this case for all fields. The screen below shows the **Service Provider** End Point Policy Group after the configuration was completed.



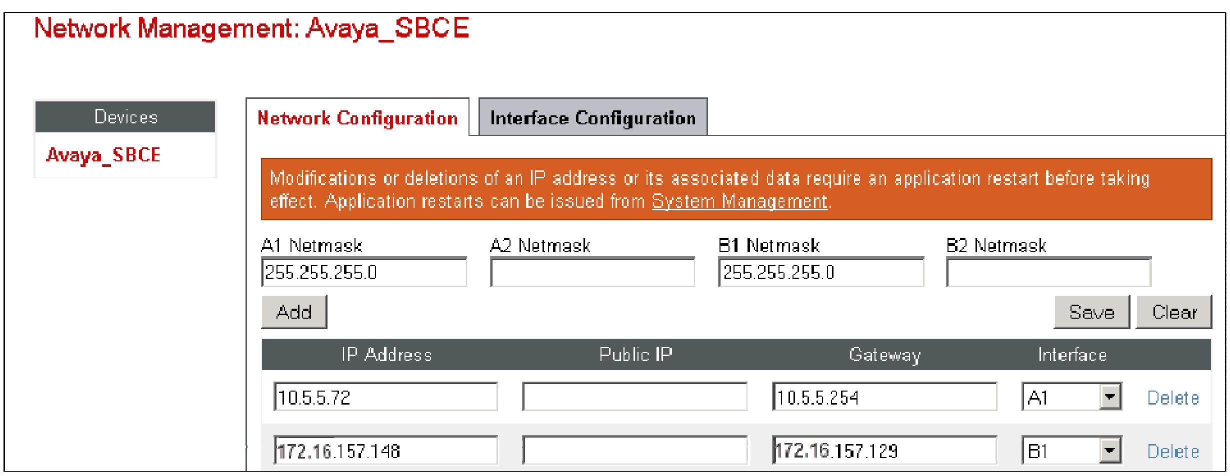
7.5. Device Specific Settings

The **Device Specific Settings** determine server specific parameters that determine how the device will work when deployed on the network. Among the parameters defined here are IP addresses, media and signaling interfaces, call flows, etc.

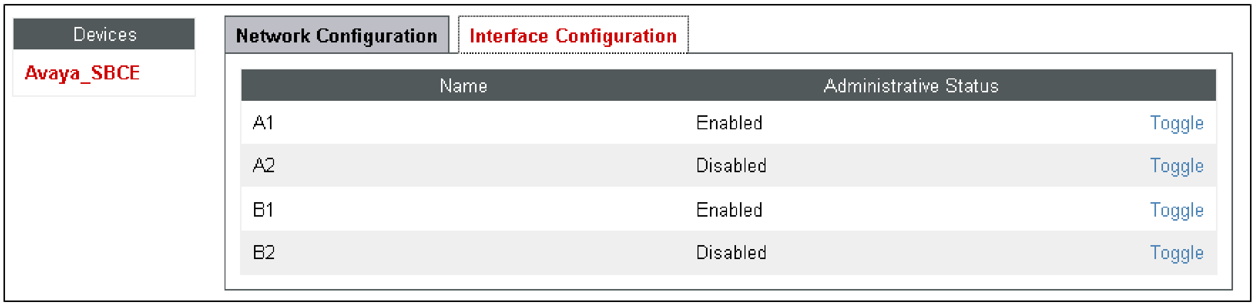
7.5.1. Network Management

The network configuration parameters should have been previously specified during installation of the Avaya SBCE. In the event that changes need to be made to the network configuration, they can be made here.

Select **Network Management** from **Device Specific Settings** on the left-side menu (not shown). Under **Devices** in the center pane, select the device being managed, **Avaya_SBCE** in the sample configuration. On the **Network Configuration** tab, verify or enter the network information as needed. Note that the **A1** interface is used for the internal side and **B1** is used for the external side of the Avaya SBCE.



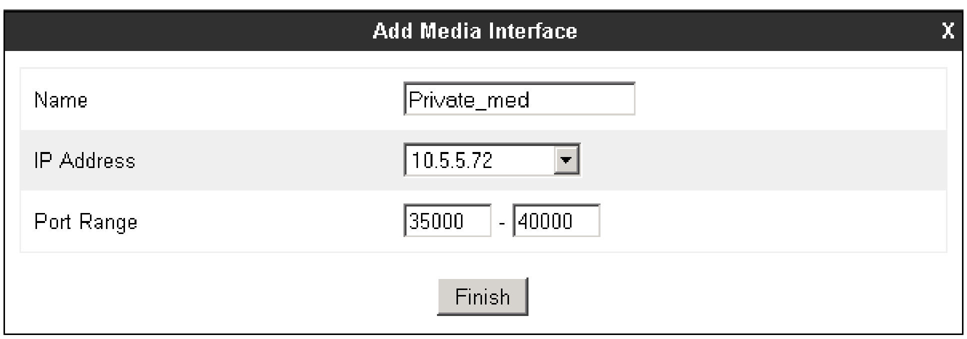
On the **Interface Configuration** tab, verify the **Administrative Status** is **Enabled** for both the **A1** and **B1** interfaces. Click the **Toggle** buttons if necessary to enable the interfaces.



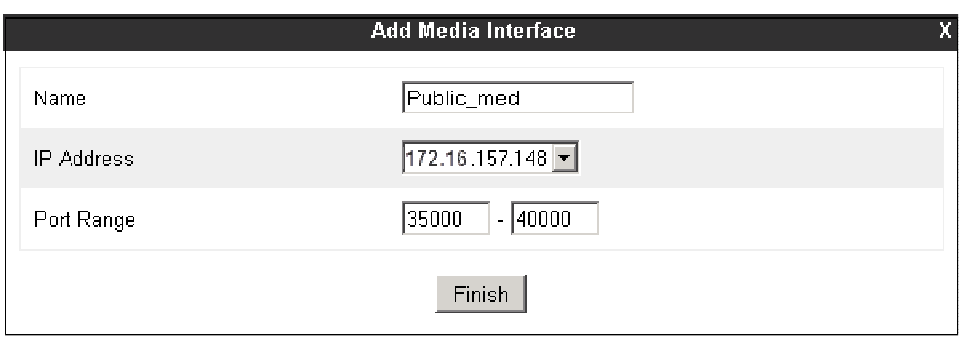
7.5.2. Media Interface

Media Interfaces were created to specify the IP address and port range in which the Avaya SBCE will accept media streams on each interface. Packets leaving the interfaces of the Avaya SBCE will advertise this IP address and one of the ports in this range as the listening IP address and port in which it will accept media from the Call or Trunk Server.

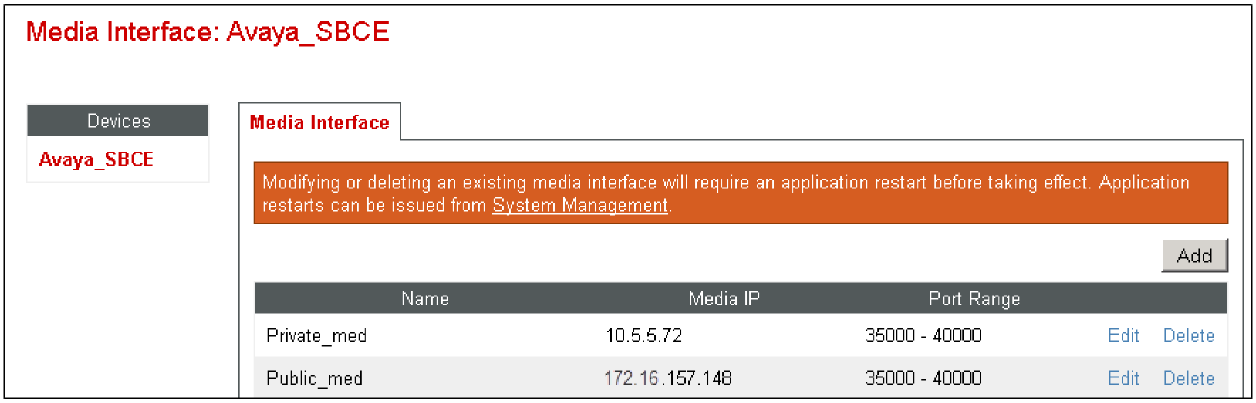
To add the Media Interface in the enterprise direction, select **Media Interface** from the **Device Specific Settings** menu on the left-hand side, select the **Avaya_SBCE** device and click the **Add** button (not shown). On the **Add Media Interface** screen, enter an appropriate **Name** for the Media Interface. Select the private IP Address for the Avaya SBCE from the **IP Address** drop-down menu. The **Port Range** was left at the default values of ***35000-40000***. Click **Finish**.



A second Media Interface facing the public network side was similarly created with the name **Public_med**, as shown below. The outside IP Address of the Avaya SBCE was selected from the drop-down menu. The **Port Range** was left at the default values.



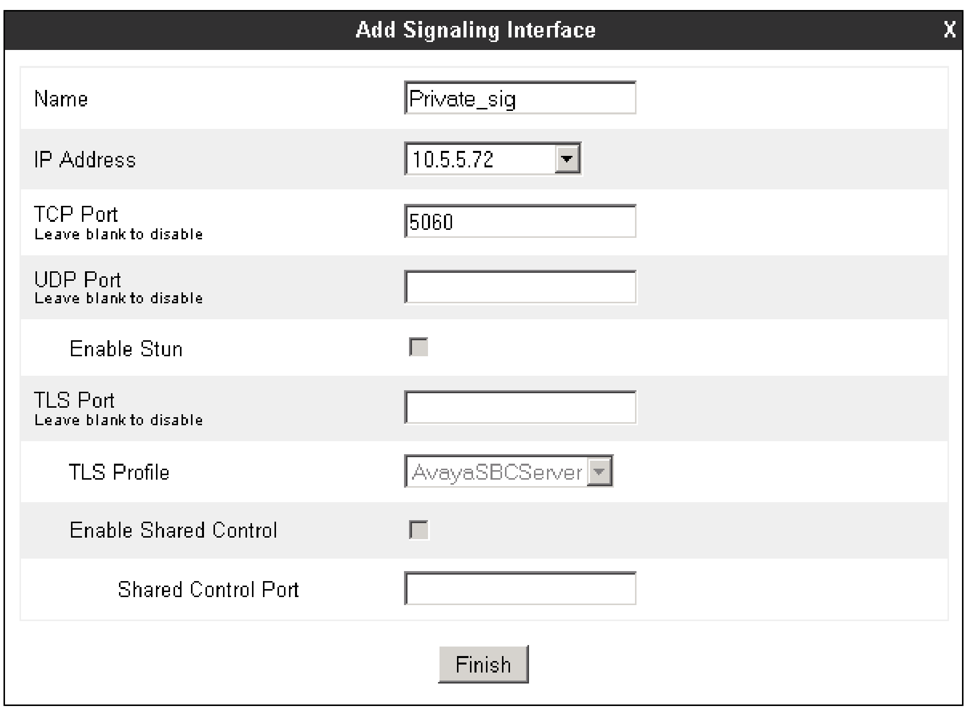
Once the configuration is complete, the **Media Interface** screen will appear as follows.



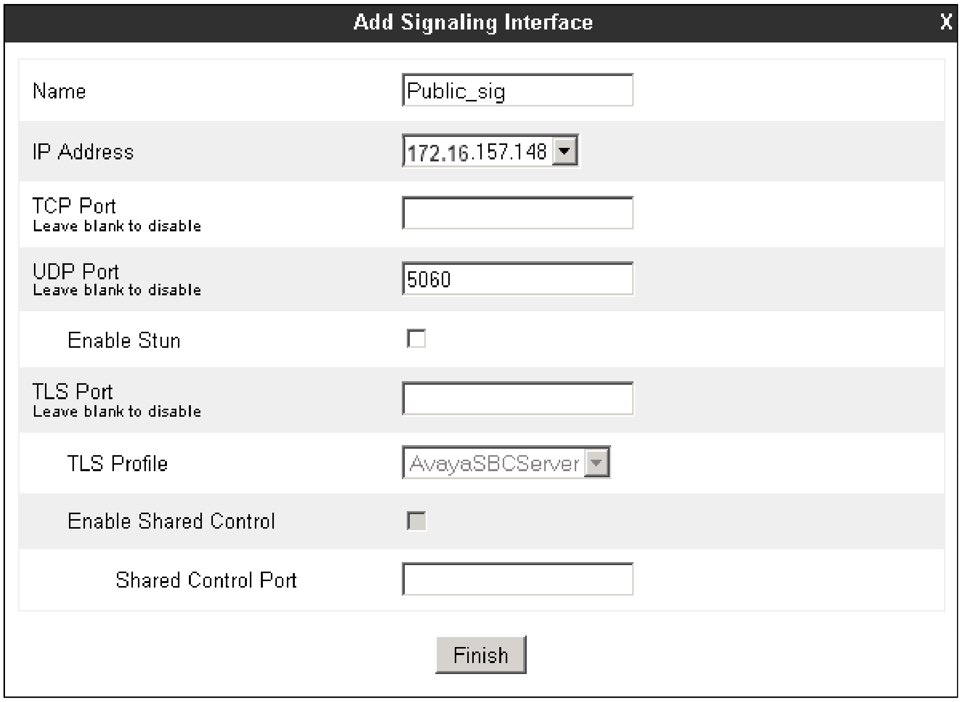
7.5.3. Signaling Interface

Signaling Interfaces are created to specify the IP addresses and ports in which the Avaya SBCE will listen for signaling traffic in both the inside and outside networks.

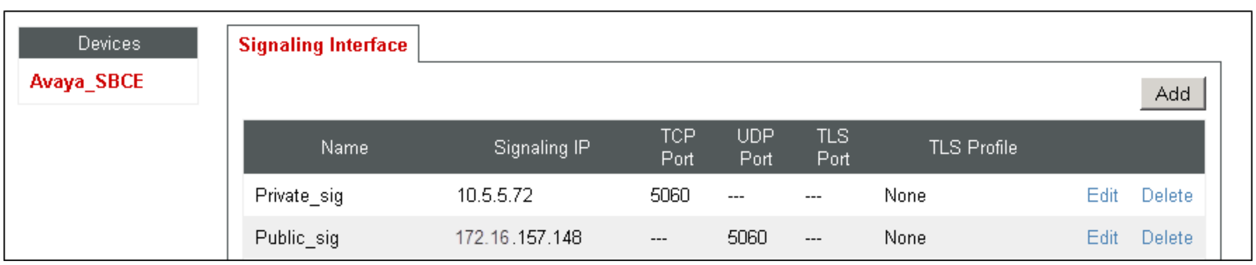
To add the Signaling Interface in the enterprise direction, select **Signaling Interface** from the **Device Specific Settings** menu on the left-hand side, select the **Avaya_SBCE** device and click the **Add** button (not shown). On the **Add Signaling Interface** screen, enter an appropriate **Name** for the interface. Select the private IP Address for the Avaya SBCE from the **IP Address** drop-down menu. Enter ***5060*** for **TCP Port**, since TCP port 5060 is used to listen to signaling traffic from Session Manager in the sample configuration. Click **Finish**.



A second Signaling Interface with the name ***Public_sig*** was similarly created in the network direction. The outside IP Address of the Avaya SBCE was selected from the drop-down menu. **UDP Port *5060*** was selected since this is the protocol and port used by the Avaya SBCE to listen to the service provider's SIP traffic.



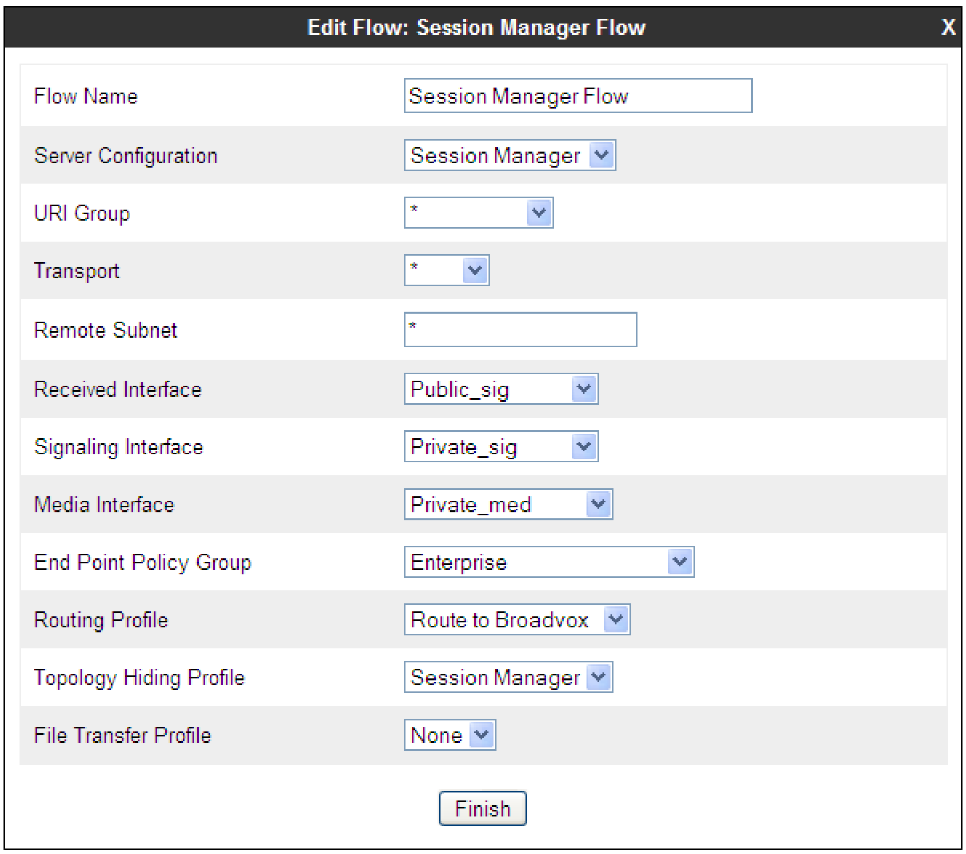
Once the configuration is complete, the **Signaling Interface** screen will appear as follows:



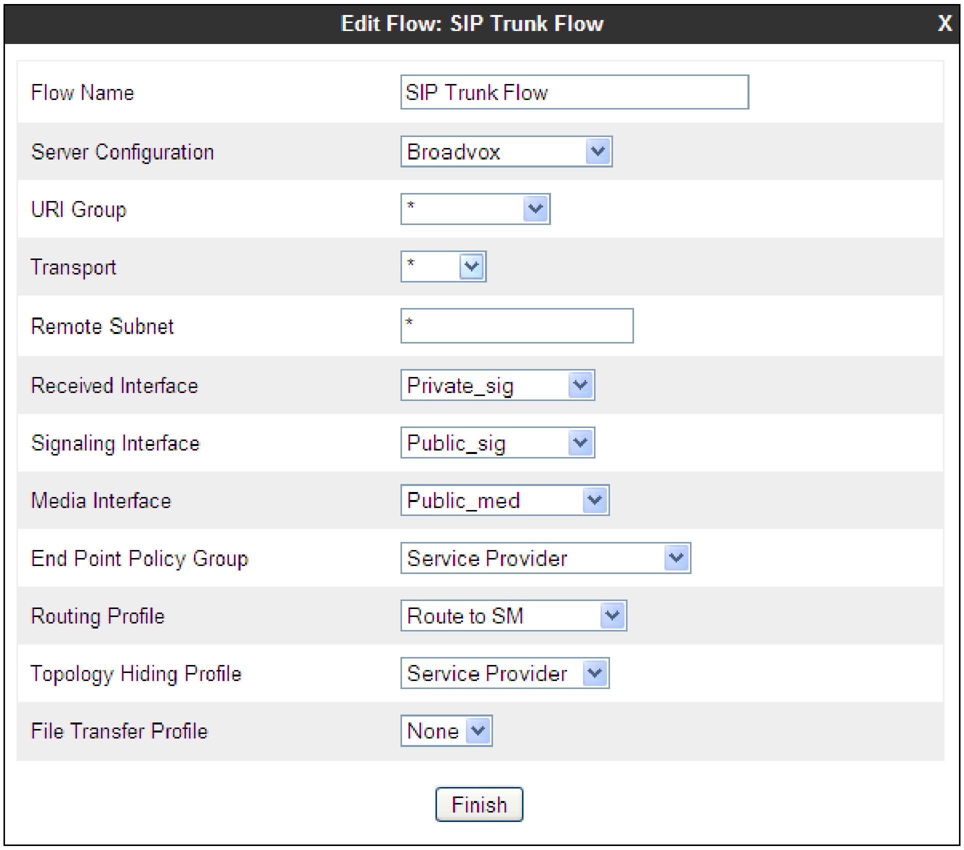
7.5.4. End Point Flows

End Point Flows determine the path to be followed by the packets traversing through the Avaya SBCE. They also combine the different sets of rules and profiles previously configured, to be applied to the SIP traffic traveling in each direction.

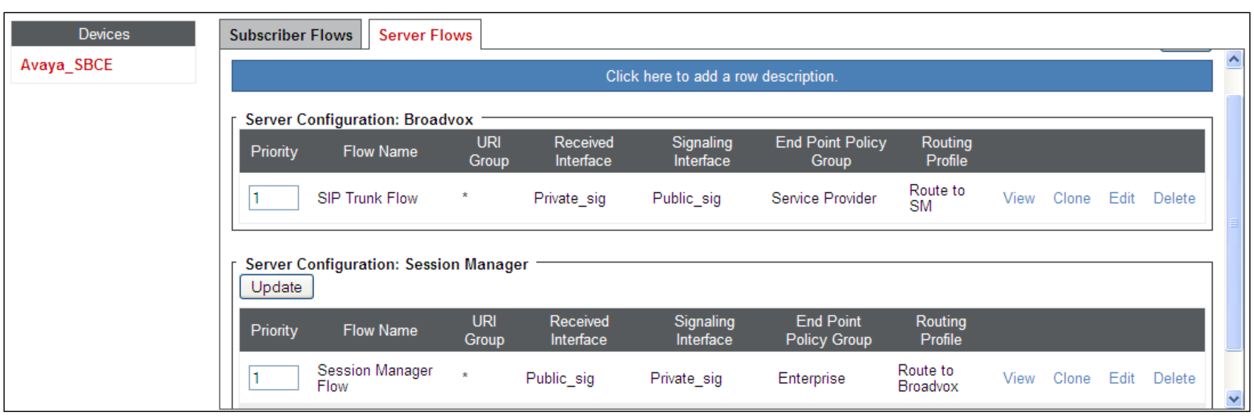
To create the call flow toward the enterprise, from the **Device Specific** menu, select **End Point Flows**, then select the **Server Flows** tab. Click **Add** (not shown). The screen below shows the flow named ***Session Manager Flow*** created in the sample configuration. The flow uses the interfaces, policies, and profiles defined in previous sections. Note the **Routing Profile** selection, which is the reverse route of the flow. Click **Finish**.



A second Server Flow with the name ***SIP Trunk Flow*** was similarly created in the network direction. The flow uses the interfaces, policies, and profiles defined in previous sections. Note the **Routing Profile** selection, which is the reverse route of the flow. Click **Finish**.



The two Server Flows created in the sample configuration are summarized on the screen below:



8. Broadvox SIP Trunking Service Configuration

Broadvox is responsible for the configuration of the Broadvox SIP Trunking service on its network. The customer will need to provide the IP address used to reach the Avaya SBCE at the enterprise. Broadvox will provide the customer the necessary information to configure the SIP connection from the enterprise site to the Broadvox network, including:

- Credentials for SIP trunk registration (username and password).
- Fully qualified domain name of the Broadvox SIP Proxy server.
- Supported codecs and order of preference.
- DID numbers.
- All IP addresses and port numbers used for signaling or media that will need access to the enterprise network through any security devices.

This information is used to complete the configuration of Communication Manager, Session Manager and the Avaya SBCE discussed in the previous sections.

9. Verification and Troubleshooting

This section provides verification steps that may be performed in the field to verify that the solution is configured properly. This section also provides a list of commands that can be used to troubleshoot the solution.

9.1. General Verification Steps

- 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.
- 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.
- Verify that the user on the PSTN can end an active call by hanging up.
- Verify that an endpoint at the enterprise site can end an active call by hanging up.

9.2. Communication Manager Verification

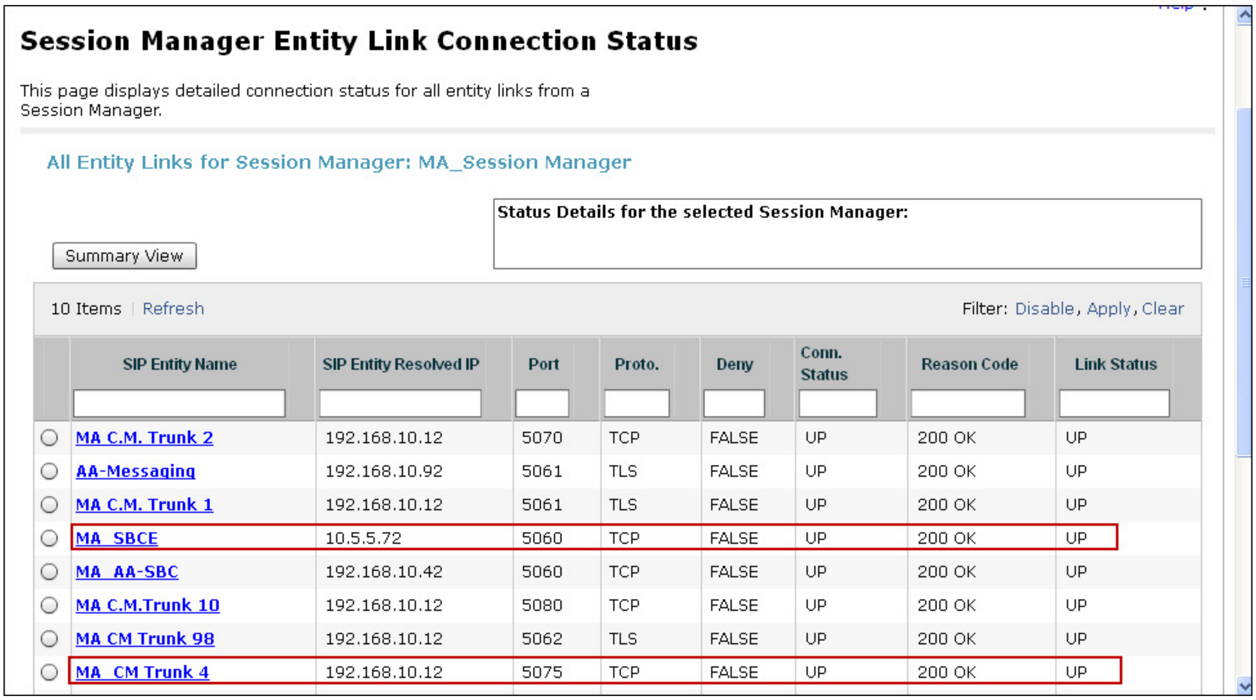
The following commands can be entered in the Communication Manager SAT terminal to verify the SIP trunk functionality:

- **list trace station** <extension number>
  Traces calls to and from a specific station.
- **list trace tac** <trunk access code number>
  Trace calls over a specific trunk group.
- **status signaling-group** <signaling group number>
  Displays signaling group service state.
- **status trunk** <trunk group number>
  Displays trunk group service state.

- **status station** <extension number>
  Displays signaling and media information for an active call on a specific station.

9.3. Session Manager Verification

Log in to System Manager. Under the **Elements** section, navigate to **Session Manager → System Status → SIP Entity Monitoring.** Verify that the state of the Session Manager links under the **Conn. Status** and **Link Status** columns to Communication Manager and the Avaya SBCE is **UP**, like shown on the screen below.



Session Manager Entity Link Connection Status

This page displays detailed connection status for all entity links from a Session Manager.

All Entity Links for Session Manager: MA_Session Manager

Summary View

Status Details for the selected Session Manager:

10 Items | Refresh

Filter: Disable, Apply, Clear

	SIP Entity Name	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
○	MA C.M. Trunk 2	192.168.10.12	5070	TCP	FALSE	UP	200 OK	UP
○	AA-Messaging	192.168.10.92	5061	TLS	FALSE	UP	200 OK	UP
○	MA C.M. Trunk 1	192.168.10.12	5061	TLS	FALSE	UP	200 OK	UP
○	MA_SBCE	10.5.5.72	5060	TCP	FALSE	UP	200 OK	UP
○	MA_AA-SBC	192.168.10.42	5060	TCP	FALSE	UP	200 OK	UP
○	MA C.M.Trunk 10	192.168.10.12	5080	TCP	FALSE	UP	200 OK	UP
○	MA CM Trunk 98	192.168.10.12	5062	TLS	FALSE	UP	200 OK	UP
○	MA_CM Trunk 4	192.168.10.12	5075	TCP	FALSE	UP	200 OK	UP

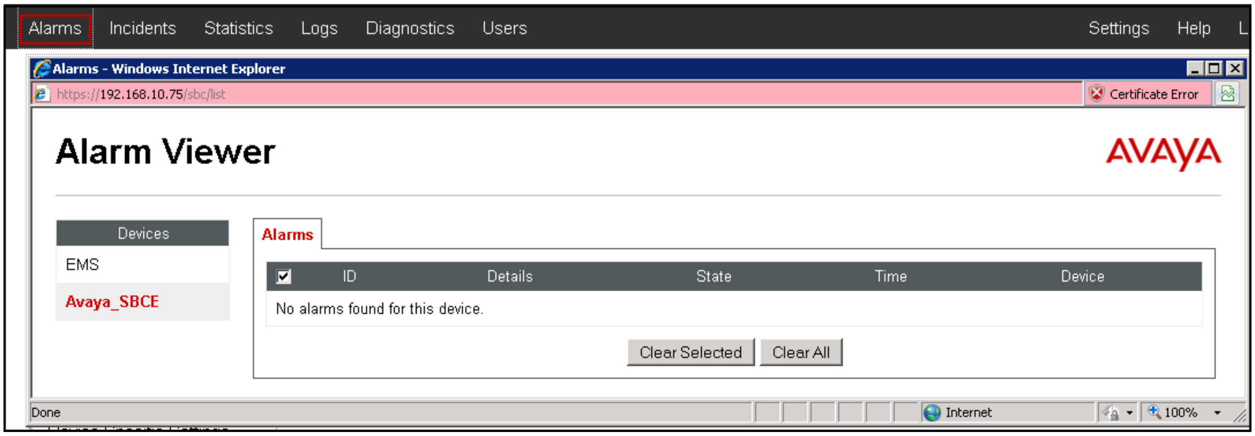
Other Session Manager useful verification and troubleshooting tools include:

- **traceSM** – Session Manager command line tool for traffic analysis. Login to the Session Manager command line management interface to run this command.
- **Call Routing Test** – The Call Routing Test verifies the routing for a particular source and destination. To run the routing test, from the System Manager Home screen navigate to **Elements → Session Manager →System Tools → Call Routing Test**. Enter the requested data to run the test

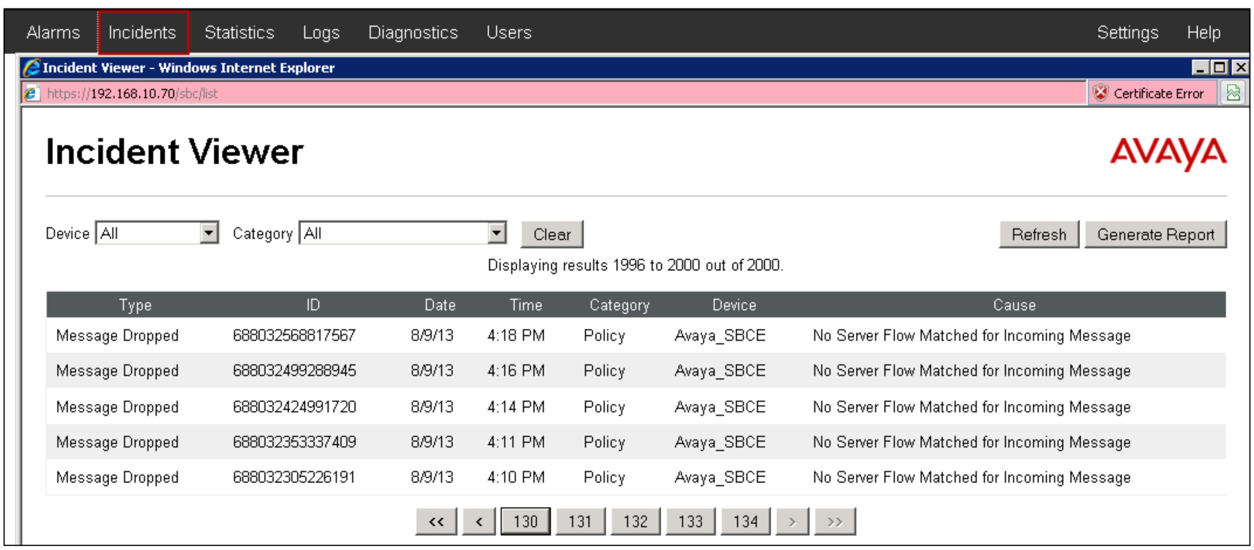
9.4. Avaya SBCE Verification

There are several links and menus located on the taskbar at the top of the screen of the web interface that can provide useful diagnostic or troubleshooting information.

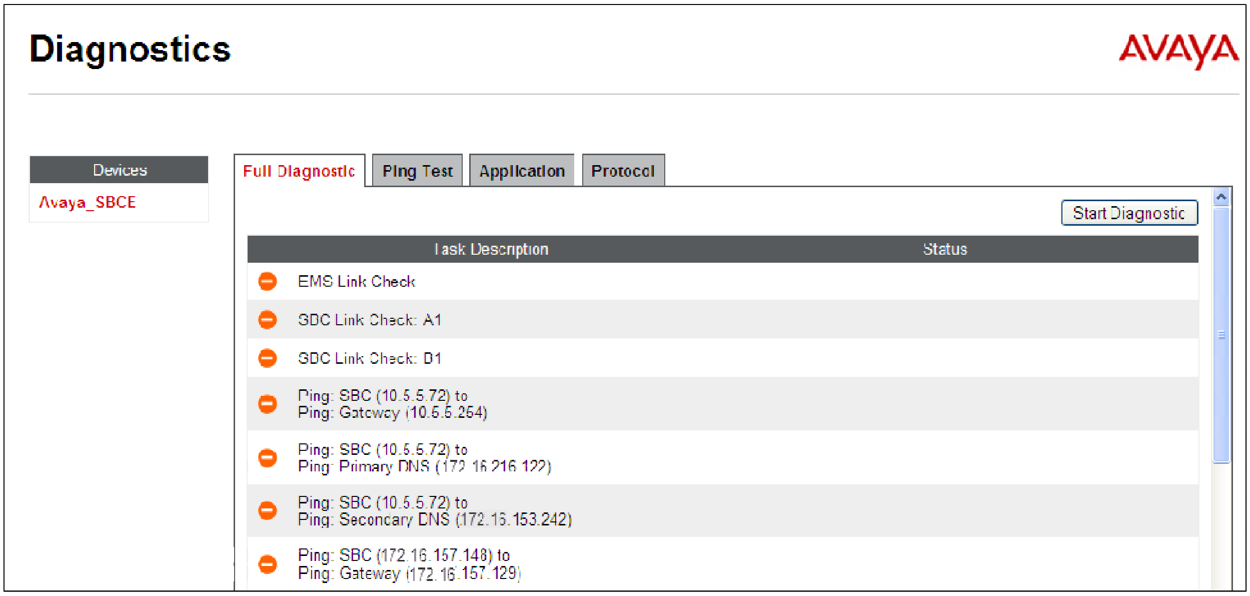
Alarms: Provides information about the health of the SBC.



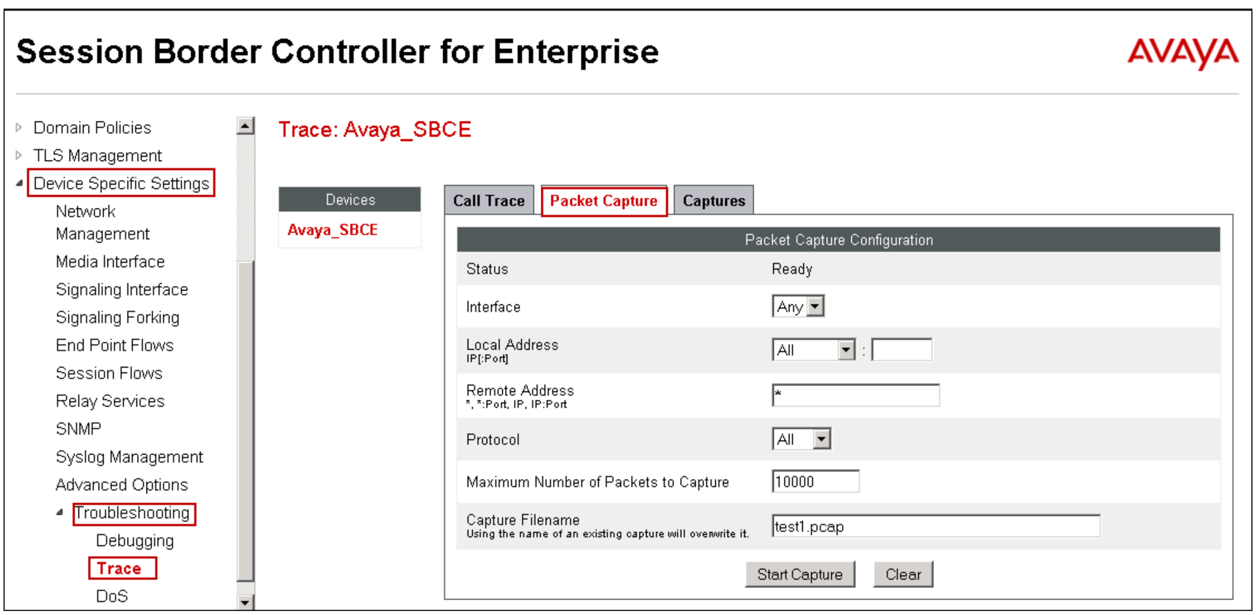
Incidents: Provides detailed reports of anomalies, errors, policies violations, etc.



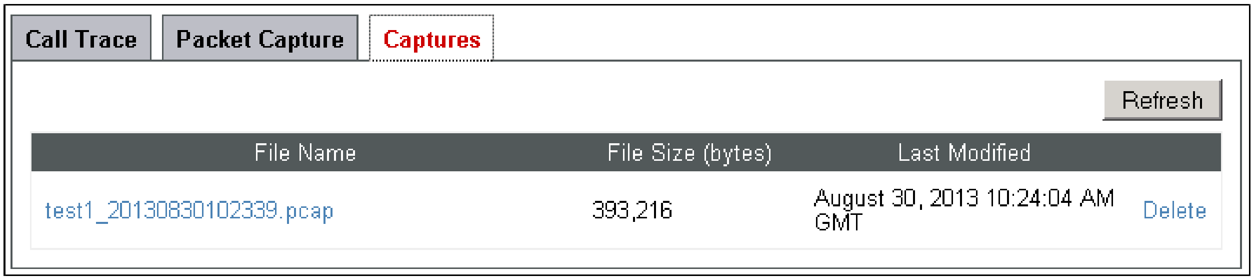
Diagnostics: This screen provides a variety of tools to test and troubleshoot the SBC network connectivity.



Additionally, the Avaya SBCE contains an internal packet capture tool that allows the capture of packets on any of its interfaces, saving them as *pcap* files. Navigate to **Device Specific Settings → Troubleshooting → Trace**. Select the **Packet Capture** tab, set the desired configuration for the trace and click **Start Capture**.



Once the capture is stopped, click the **Captures** tab and select the proper *pcap* file. Note that the date and time is appended to the filename specified previously. The file can now be saved to the local PC, where it can be opened with an application such as Wireshark.



10. Conclusion

These Application Notes describe the procedures required to configure an Avaya Aura® Communication Manager 6.3, Avaya Aura® Session Manager 6.3 and Avaya Session Border Controller for Enterprise 6.2, to connect to the Broadvox SIP Trunking service, as shown in **Figure 1**.

Interoperability testing of the sample configuration was completed with successful results for all test cases with the exception of the observations/limitations described in **Section 2.2.**

11. References

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

[1] *Installing and Configuring Avaya Aura® System Platform*, Release 6.3, May 2013.
[2] *Administering Avaya Aura® System Platform*, Release 6.3, May 2013.
[3] *Administering Avaya Aura® Communication Manager*, Release 6.3, May 2013, Document Number 03-300509.
[4] *Avaya Aura® Communication Manager Feature Description and Implementation,* Release 6.3, May 2013, Document Number 555-245-205.
[5] *Administering Avaya Aura® Session Manager*, Release 6.3, June 2013.
[6] *Installing Avaya Session Border Controller for Enterprise*, Release 6.2, June 2013
[7] *Administering Avaya Session Border Controller for Enterprise*, Release 6.2, May 2013
[8] *Avaya Session Border Controller for Enterprise Release Notes.* Release 6.2, June 2013
[9] *Administering Avaya one-X® Communicator*, December 2012.
[10] *Using Avaya one-X® Communicator, Release 6.1,* October 2011.
[11] *Implementing Avaya Flare® Experience for Windows.* Release 1.1 February 2013.
[12] *RFC 3261 SIP: Session Initiation Protocol,* http://www.ietf.org/.
[13] *RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, http://www.ietf.org/

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.