Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Communication Manager 6.3, Avaya Aura® Session Manager 6.3 and Avaya Session Border Controller for Enterprise 6.3 to support Clearcom SIP Trunk Services – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking on an enterprise solution consisting of Avaya Aura® Communication Manager 6.3, Avaya Aura® Session Manager 6.3 and Avaya Session Border Controller for Enterprise 6.3, to interoperate with Clearcom SIP Trunk Services.

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

Readers should pay attention to Section 2, in particular the scope of testing as outlined in Section 2.1 as well as the observations noted in Section 2.2, to ensure that their own use cases are adequately covered by this scope and results.

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.

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1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between Clearcom SIP Trunk Services 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) 6.3 and various Avaya endpoints, listed in **Section 4**.

The SIP trunking service provided by Clearcom and referenced within these Application Notes is designed for business customers in Mexico. Customers using this service with this Avaya 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 Clearcom SIP Trunk Services 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 via the service provider network.
- Inbound and outbound PSTN calls to/from Avaya one-X® Communicator softphones using "This Computer" and "Other Phone" modes. (H.323, SIP).
- Inbound and outbound PSTN calls to/from Avaya Communicator for Windows softphones (SIP).
- Various call types, including: local, long distance and international.
- Codecs G729A, G.711MU, G711A and proper codec negotiation.
- DTMF tones 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 transferring, call forwarding and mobility (extension to cellular).
- Routing inbound PSTN calls to call center agent queues.
- Fax.

The following items are not supported and were not tested:

- Operator (0), operator assisted (0+10) and Directory Assistance calls are not supported.
- Inbound toll-free and emergency calls are supported but were not tested as part of the compliance test.

2.2. Test Results

Interoperability testing of Clearcom SIP Trunk Services with the Avaya SIP-enabled enterprise solution was completed with successful results for all test cases with the observations/limitations noted below:

- **Caller ID on outbound calls**: On calls originating from enterprise extensions to PSTN telephones, the caller ID number shown on the PSTN endpoint was always the main number assigned by Clearcom to the SIP trunk, not the specific DID assigned to that extension. Additionally, on calls to EC500 mobile phones and calls that were forwarded or transferred back on the SIP trunk to the PSTN, the number displayed on the PSTN endpoint was the main number on the trunk, not the originator's caller's ID.
  This may be a requirement of the Clearcom service for all outbound calls coming from the enterprise site, it is listed here as an observation.
- **Caller ID on inbound calls**: On inbound calls made from the test lab in the U.S., the caller ID shown on the enterprise extensions occasionally showed "Unavailable", while in other cases showed numbers corresponding to local PSTN numbers in Mexico, not the number of the original caller. Calls made from test numbers in Mexico showed the correct caller ID.
- **Outbound Calling Party Number (CPN) Block**: When an enterprise user activated the CPN Block feature for privacy purposes, Communication Manager sent the "Privacy:id" header and "anonymous" in the "From" header of the outbound call, while the actual number of the caller is sent in the "P-Asserted-Identity" header. On the receiving end, the caller ID was still the main number assigned by Clearcom to the SIP trunk.
- **Call transfer to the PSTN using REFER**: PSTN calls that were transferred back to the network using REFER messages did not work properly. Attended call transfers dropped. On blind transfers, the REFER message was accepted by Clearcom with a 202 message, but the trunks were not released after the call transfer was completed. For these reasons, REFER was disabled for the compliance test, by setting Network Call Redirection in the Communication Manager trunk group form to "no". With REFER disabled, blind and attended call transfers to the PSTN completed successfully, with the caveat that Communication Manager was not released from the call path after the call was transferred, and two trunks remained busy for the complete duration of the call.
- **Fax support**: T.38 fax is the fax protocol officially supported by Communication Manager on SIP trunks. During the tests, Clearcom responded with "488 Not Acceptable Here" to the re-invites sent from Communication Manager to make the change from the voice setup to T.38 fax, and the calls dropped. Even though it was possible during the tests to complete G.711 fax calls using a local test number in Mexico, G.711 fax pass-through is available in Communication Manager on a "best effort" basis, and it's not guaranteed that it will work in every instance.
  For the reasons above, the use of fax is not recommended with this solution.
- **Conference in Avaya Communicator softclients**: The Communication Manager conference feature is not currently supported in Avaya Communicator release 2.1. An Avaya Aura® Conferencing server is required for ad-hoc conferences. This feature should be available in the upcoming release 3.0 of Avaya Communicator.

- **From Header Manipulation**: Clearcom uses SIP trunk registration and digest authentication in order to accept outbound calls from the enterprise into their network. Additionally, Clearcom requires the username associated with the SIP trunk credentials to be present in the "From" header of all outbound calls from the enterprise. Otherwise, the call is rejected with a "403 Username=From not allowed" message. A Signaling Script was created in the Avaya SBCE to include the SIP trunk credential's username in the "From" header of all outbound calls. (**Section 7.7**).
- **Request-URI Header Manipulation**: Clearcom sends the username associated with the SIP trunk credentials in the "Request URI" header of all inbound calls, while the actual DID number of the party dialed is sent in the "To" header. Since the routing decision in Session Manager is based on Dial Patterns, by inspecting the number present in the "Request URI" header of the incoming call, a Signaling Script was created in the Avaya SBCE to populate the "Request URI" header with the number present in the "To" header of inbound calls. (**Section 7.7**).
- **SIP header optimization**: There are multiple SIP headers used by Communication Manager, Session Manager and the Avaya SBCE that had no particular use in the service provider's network. These headers were removed in order to block private IP addresses and other enterprise information from being propagated outside of the enterprise boundaries, and also to reduce the size of the packets entering the Clearcom network. The parameters "gsid" and "epv" on outbound Contact headers were removed using a Signaling Script in the Avaya SBCE (**Section 7.7**). Additionally, the following outbound headers were blocked by the Avaya SBCE using Signaling Rules: AV-Global-Session-ID, Alert-Info, Endpoint-View, P-AV-Message-ID, P-Charging-Vector, P-Location and Remote-Address. (**Section 7.11**).

2.3. Support

For technical support on the Clearcom SIP Trunk Services offer, visit http://www.clearcom.mx/

3. Reference Configuration

Figure 1 illustrates the sample Avaya SIP-enabled enterprise solution, connected to Clearcom SIP Trunk 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.

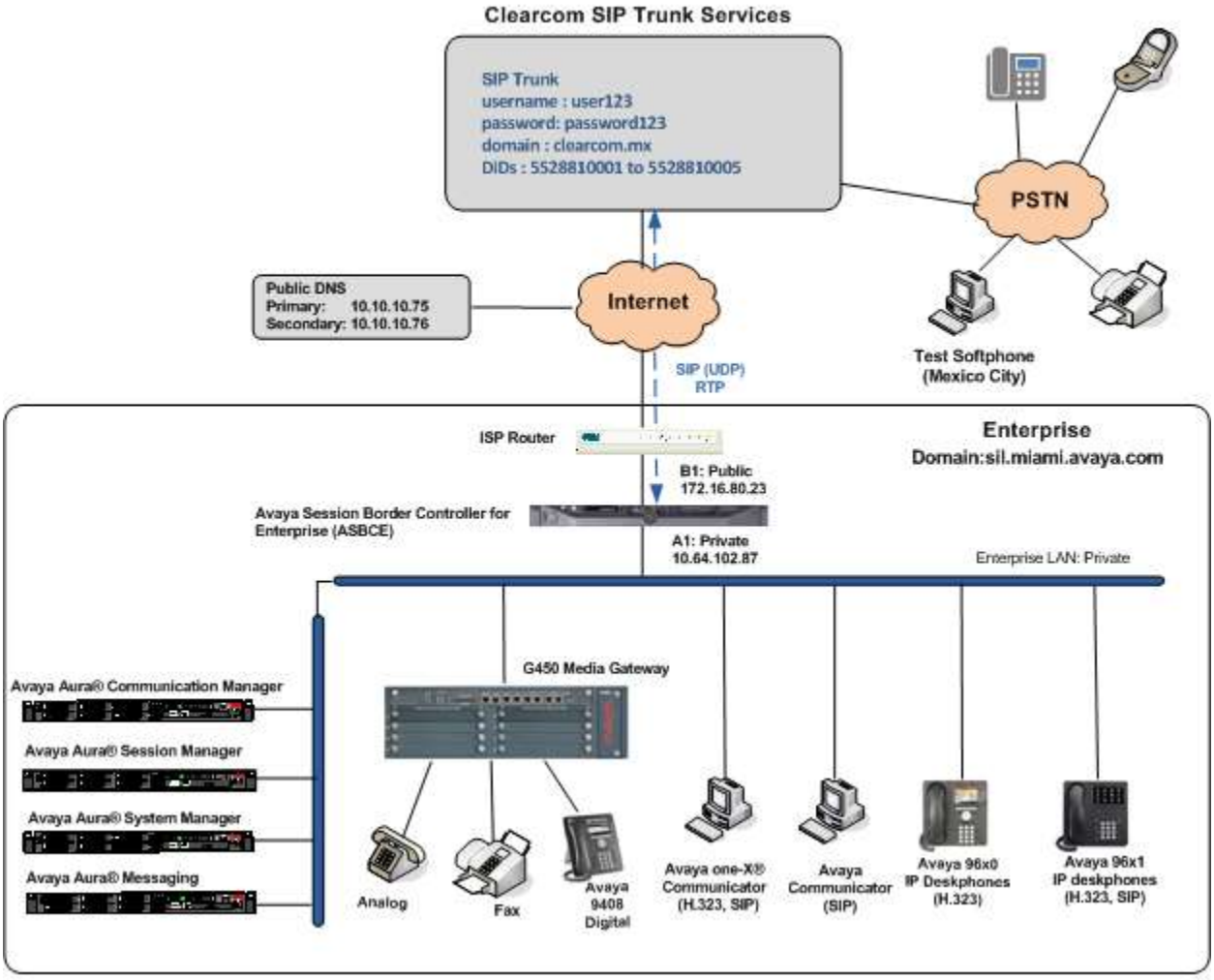


Figure 1: **Avaya SIP Enterprise Solution connected to Clearcom SIP Trunk Services**.

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 Deskphones (H.323 and SIP).
- Avaya one-X® Communicator softphones (H.323 and SIP).
- Avaya Communicator for Windows softphones.
- Avaya digital and analog telephones.

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 can protect the enterprise against any SIP-based attacks. The Avaya SBCE also performs network address translation at both the IP and SIP layers.

The transport protocol between the Avaya SBCE and Clearcom 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 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 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 selects the proper SIP trunk, the call is routed to Session Manager. Session Manager once again uses the configured dial patterns (or regular expressions) and routing policies to determine the route to the Avaya SBCE for egress to the Clearcom 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. This trunk carried both inbound and outbound traffic.

Avaya Aura® Messaging was used during the compliance test to verify voice mail redirection and navigation, as well as the delivery of Message Waiting Indicator (MWI) 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 Clearcom SIP Trunk Services, 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:

Equipment/Software	Release/Version
Avaya	
Avaya Aura® Communication Manager	6.3.10 (6.3-03.0.124.0-22147)
Avaya Aura® Session Manager	6.3.13 (6.3.13.0.631304)
Avaya Aura® System Manager	6.3.13 (Update Revision 6.3.13.10.3336)
Avaya Session Border Controller for Enterprise	6.3 Service Pack 2 (6.3.2-08-5478)
Avaya Aura® Messaging	6.3.2 SP2 Patch 3 (MSG-03.0.124.0-335_0217)
Avaya G450 Media Gateway	36.14.0
Avaya 96x0 Series IP Telephones (SIP)	Avaya one-X® Deskphone Edition SIP 2.6.13
Avaya 96x1 Series IP Telephones (SIP)	Avaya one-X® Deskphone Edition SIP 6.5.0.17
Avaya 96x1 Series IP Telephones (H.323)	Avaya one-X® Deskphone Edition H.323 6.6
Avaya one-X® Communicator (H.323, SIP)	6.2.6.03-FP6
Avaya Communicator for Windows	2.1.1.74
Avaya 9408 Digital Telephone	Rel 12.0
Avaya 6210 Analog Telephone	N/A
Clearcom	
OpenSIPS Softswitch	1.9
OpenSIPS Session Border Controller	1.9

The specific configuration above was used for the compliance testing. Note that this solution will be compatible with other Avaya Servers 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 to work with Clearcom SIP Trunk Services. A SIP trunk is established between Communication Manager and Session Manager for use by signaling traffic to and from the service provider. It is assumed that the general installation of Communication Manager and the Avaya G450 Media Gateway 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 **361** 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.

```
display system-parameters customer-options                    Page   2 of  11
                              OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 12000 0
          Maximum Concurrently Registered IP Stations: 18000 2
            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 0
                   Maximum Video Capable IP Softphones: 18000 3
                     Maximum Administered SIP Trunks: 24000 361
  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: 300   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 *(***proc**r) 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
ASBCE_A1                      10.64.102.87
asm                           10.64.102.82
default                       0.0.0.0
procr                         10.64.102.83
procr6                        ::
tftp                          192.168.10.150
```

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 2 was used for this purpose. Clearcom used codecs G.729A, G.711MU and G711A on the SIP trunk, in this order of preference. Enter the corresponding codecs in the **Audio Codec** column of the table. Default values can be used for all other fields.

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

                       IP CODEC SET

   Codec Set: 2

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

On **Page 2**, set the **Fax Mode** to ***off***. See the note regarding fax use in **Section 2.2**.

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

                       IP CODEC SET

                           Allow Direct-IP Multimedia? n

                                                                 Packet
                           Mode              Redundancy          Size(ms)
   FAX                     off                 0
   Modem                   off                 0
   TDD/TTY                 off                 3
   H.323 Clear-channel     n                   0
   SIP 64K Data            n                   0                 20
```

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 2 was chosen for the service provider trunk. Use the **change ip-network-region 2** command to configure region 2 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 if needed.
- Set the **Codec Set** field to the IP codec set defined in **Section 5.4**.
- Default values may be used for all other fields.

```
change ip-network-region 2                                    Page   1 of  20
                              IP NETWORK REGION
  Region: 2
Location: 1     Authoritative Domain: sil.miami.avaya.com
   Name: Service Provider     Stub Network Region: n
MEDIA PARAMETERS               Intra-region IP-IP Direct Audio: yes
    Codec Set: 2               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 2 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 **2** will be used for calls between region 2 (the service provider region) and region 1 (the rest of the enterprise).

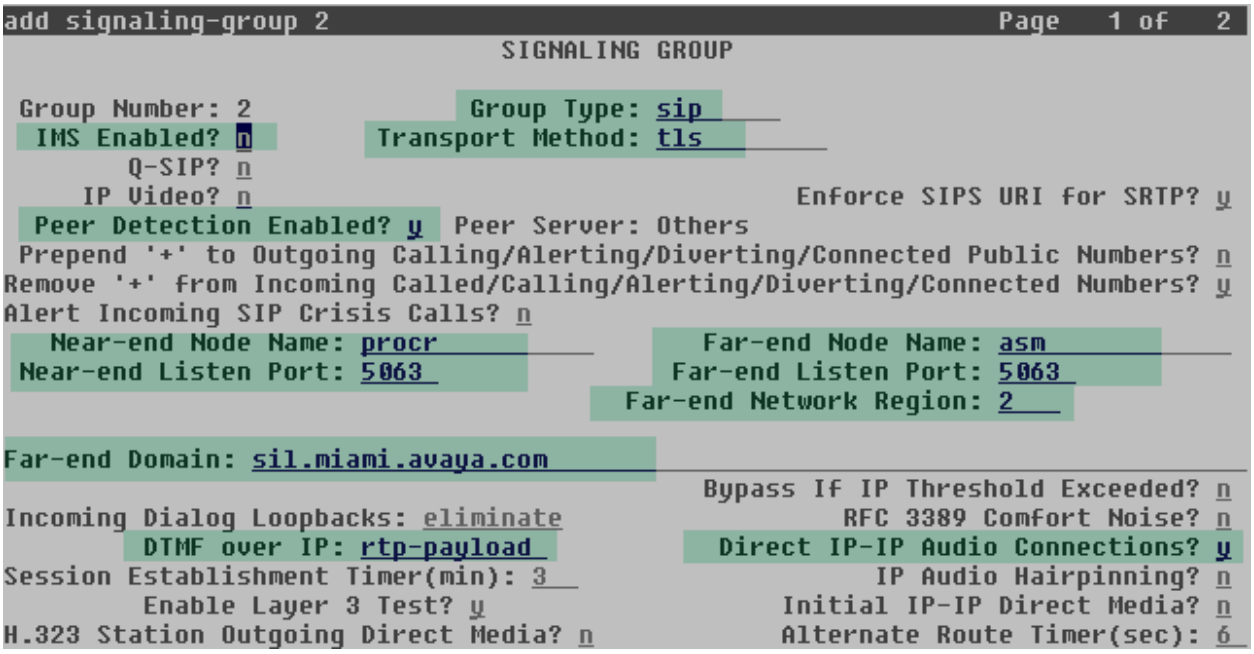
```
change ip-network-region 2                                    Page   4 of  20

 Source Region: 2     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    2___  y    NoLimit                                        n  ___    t
 2    2                                                          all
 3    ____                                                       ___
 4
```

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 2 was used and was configured using the parameters highlighted below, shown on the screen on the next page:

- 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, ***tls*** 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**.



- 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 TLS, the well-known port value is 5061). 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 ***5063***.
- 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 ***y***. This field will enable media shuffling on the SIP trunk allowing Communication Manager to redirect media traffic directly between the Avaya SBCE and the enterprise endpoint. If this value is set to **n**, then the Avaya Media Gateway will remain in the media path of all calls between the SIP trunk and the endpoint. Depending on the number of media resources available in the Avaya Media Gateway, these resources may be depleted during high call volume preventing additional calls from completing.
- 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 2 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 section.
- 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.

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

Group Number: 2                    Group Type: sip          CDR Reports: y
 Group Name: SIP Trunk to SP              COR: 1      TN: 1      TAC: 602
   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: 2
                                                  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. The default value of **600** seconds was used.

```
add trunk-group 2                                          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 ***private***. 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 has enabled CPN block.

```
add trunk-group 2                                          Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n           Measured: none
                                                      Maintenance Tests? y


                  Numbering Format: private
                                                 UUI Treatment: service-provider

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

On **Page 4**, set the **Network Call Redirection** field to ***n***. See note in **Section 2.2** regarding this setting. Set the **Send Diversion Header** field to ***n***. Set the **Support Request History** field to ***n***.

Set the **Telephone Event Payload Type** to **100**, the value preferred by Clearcom. Set **Convert 180 to 183 for Early Media** to ***y***. Default values were used for all other fields.

```
add trunk-group 2                                          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: 100


                          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

          Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
```

5.8. Calling Party Information

The calling party number is sent in the SIP "From", "Contact" and "PAI" headers. Since private numbering was selected to define the format of this number (**Section 5.7**), use the **change private-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 example below, five DID numbers are assigned by the service provider for testing. These DID numbers were used as the outbound calling party information on the service provider trunk when calls were originated from the mapped extensions.

```
change private-numbering 1                                    Page   1 of   2
                          NUMBERING - PRIVATE FORMAT

Ext  Ext             Trk        Private          Total
Len  Code            Grp(s)     Prefix           Len
 4   5002            2          5528810001       10    Total Administered: 10
 4   5004            2          5528810002       10       Maximum Entries: 540
 4   5005            2          5528810003       10
 4   5013            2          5528810004       10
 4   5033            2          5528810005       10
 __  ____________    _____      ____________     __
```

Note: During the compliance test, Clearcom did not inspect the calling party number sent in the origination headers from the enterprise to authenticate outbound calls; it used SIP trunk registration and Digest Authentication instead. This is shown on **Section 7.8.2** of the Avaya SBCE configuration, later in this document. Clearcom also inserted the main DID number assigned to the SIP trunk on all outbound calls sent to the PSTN, for caller ID purposes. Since the calling party information sent from the enterprise was for all practical purposes not used by Clearcom, the configuration shown on the screen above was not strictly required, and it is shown here simply for completeness.

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 DID number sent by Clearcom 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 2                    Page   1 of  30
                     INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 public-ntwrk    10 5528820001      10  5002
 public-ntwrk    10 5528810002      10  5004
 public-ntwrk    10 5528810003      10  5005
 public-ntwrk    10 5528810004      10  5013
 public-ntwrk    10 5528810005      10  5033
 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 2, 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: 1

       Dialed          Total      Route    Call   Node  ANI
       String          Min  Max   Pattern  Type   Num   Reqd
   00                  12   22    2        intl         n
   001                 13   13    2        intl         n
   01                  12   12    2        natl         n
   2                   8    8     2        hnpa         n
   55                  10   10    2        hnpa         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 2 in the compliance test.

- **Pattern Name**: Enter a descriptive name.
- **Grp No**: Enter the outbound trunk group for the SIP service provider.
- **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.
- **Numbering Format**: Set to ***unk-unk***. All calls using this route pattern will use the private numbering table. See setting of the **Numbering Format** in the trunk group form for full details in **Section 5.7**.

```
change route-pattern 2                                       Page   1 of   3
                    Pattern Number: 2     Pattern Name: Route to SP
                                  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:  2   0                                                           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                            unk-unk   none
```

Enter the **save translation** command (not shown) to save all the changes made to the Communication Manager configuration.

6. Configure Avaya Aura® Session Manager

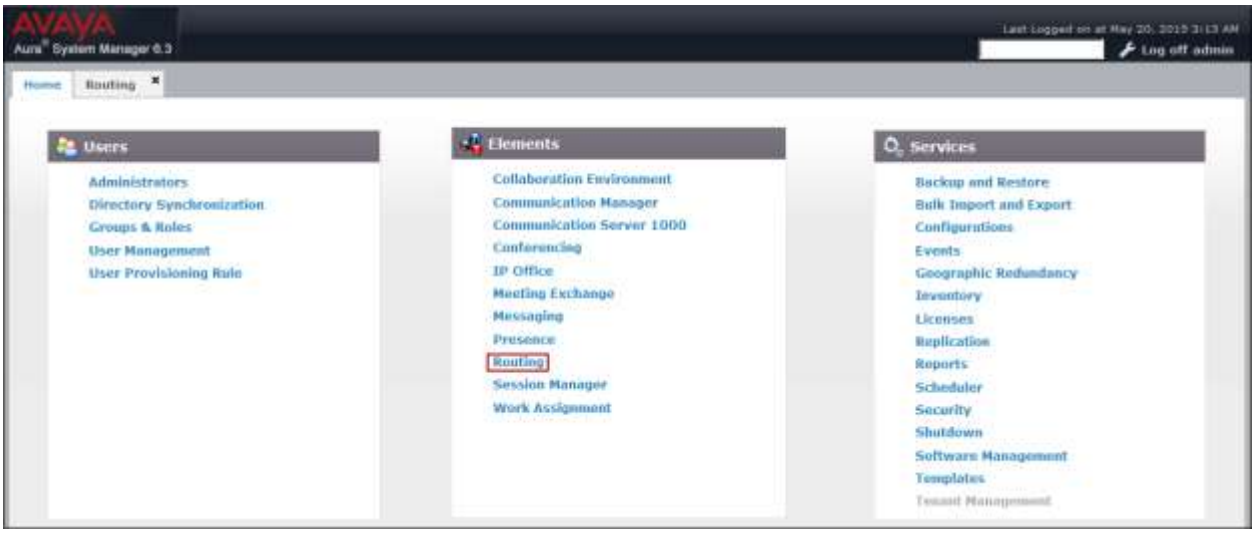
This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- 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.

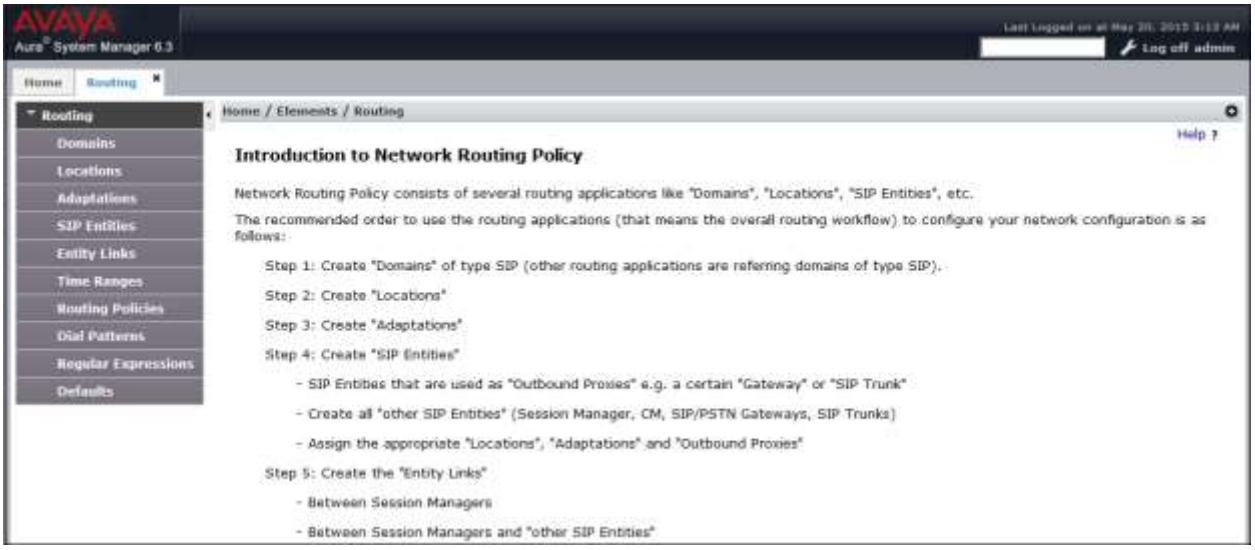
The following sections assume that the initial configuration of Session Manager and System Manager has already been completed, and that network connectivity exists between System Manager and Session Manager.

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 an entry for each SIP domain for which Session Manager will need to be aware in order to route calls. For the compliance test, this was the enterprise domain, ***sil.miami.avaya.com***. Navigate to **Routing → Domains** in the left-hand navigation pane 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.

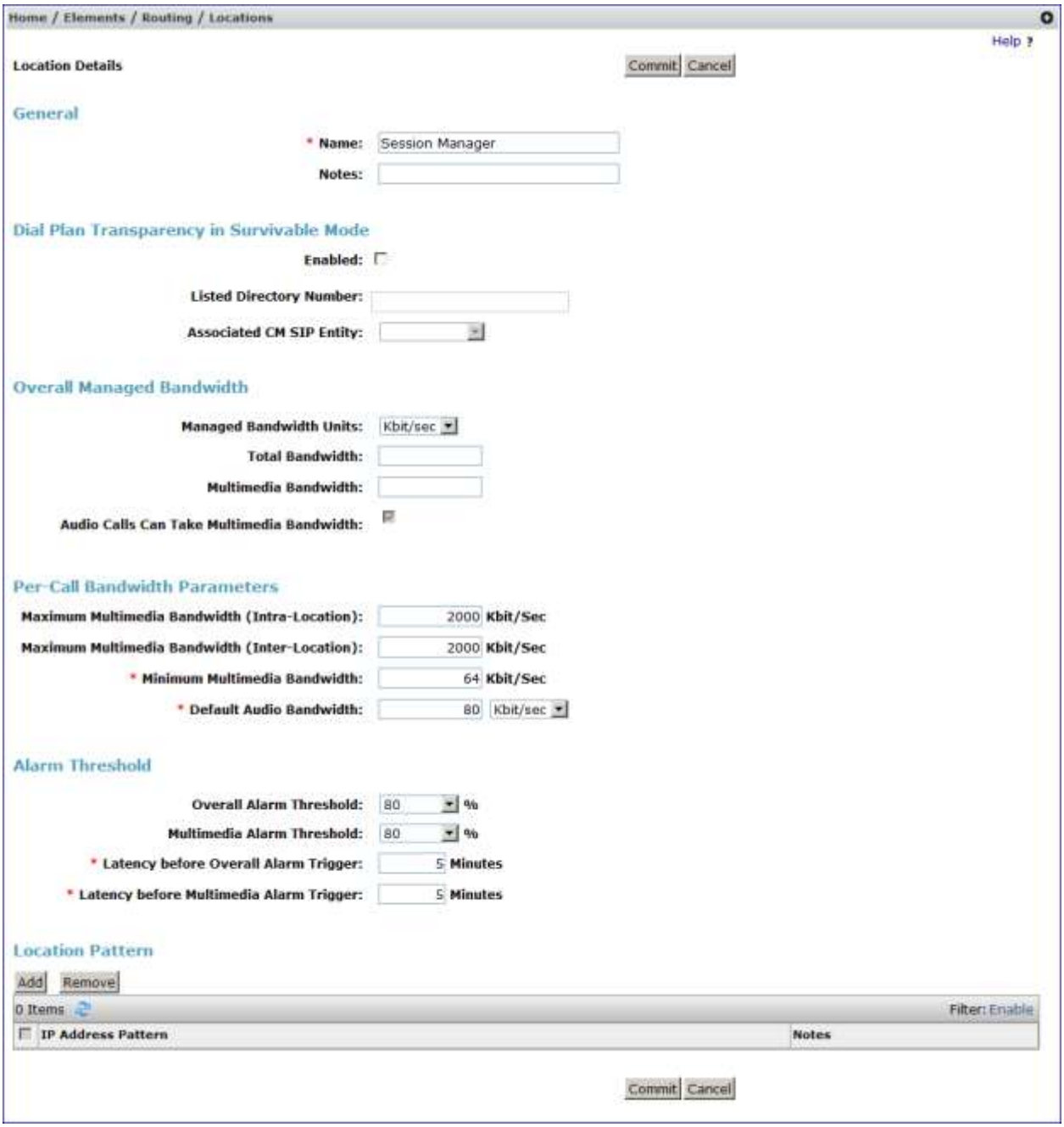
6.3. Locations

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.

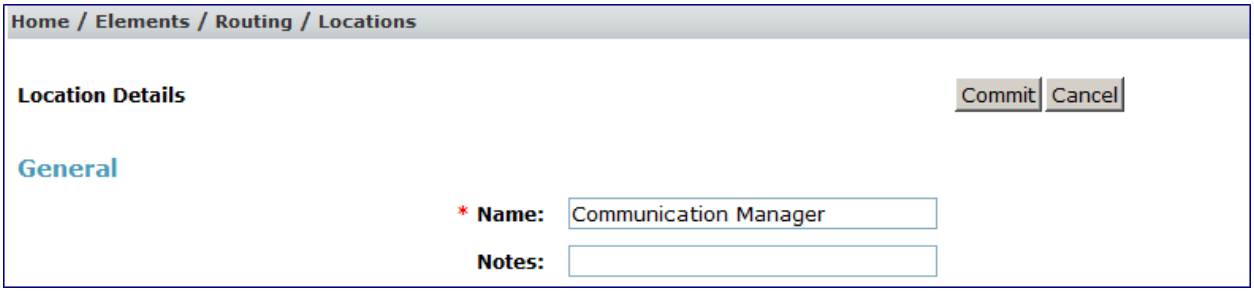
- **Name:** Enter a descriptive name for the location.
- **Notes:** Add a brief description (optional).

Defaults can be used for all other parameters.

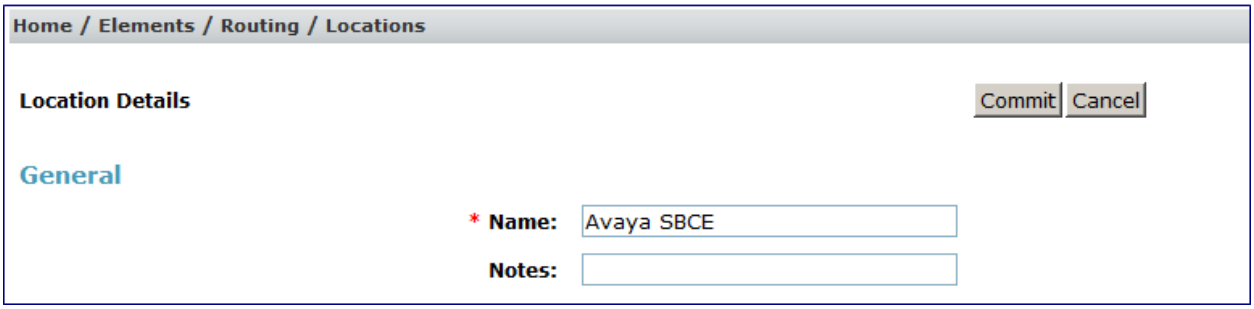
The following screen shows the location details for the location named ***Session Manager***. Later, this location will be assigned to the SIP Entity corresponding to Session Manager. Default values were used for all parameters.



The following screen shows the location details for the location named ***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 ***Avaya 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.

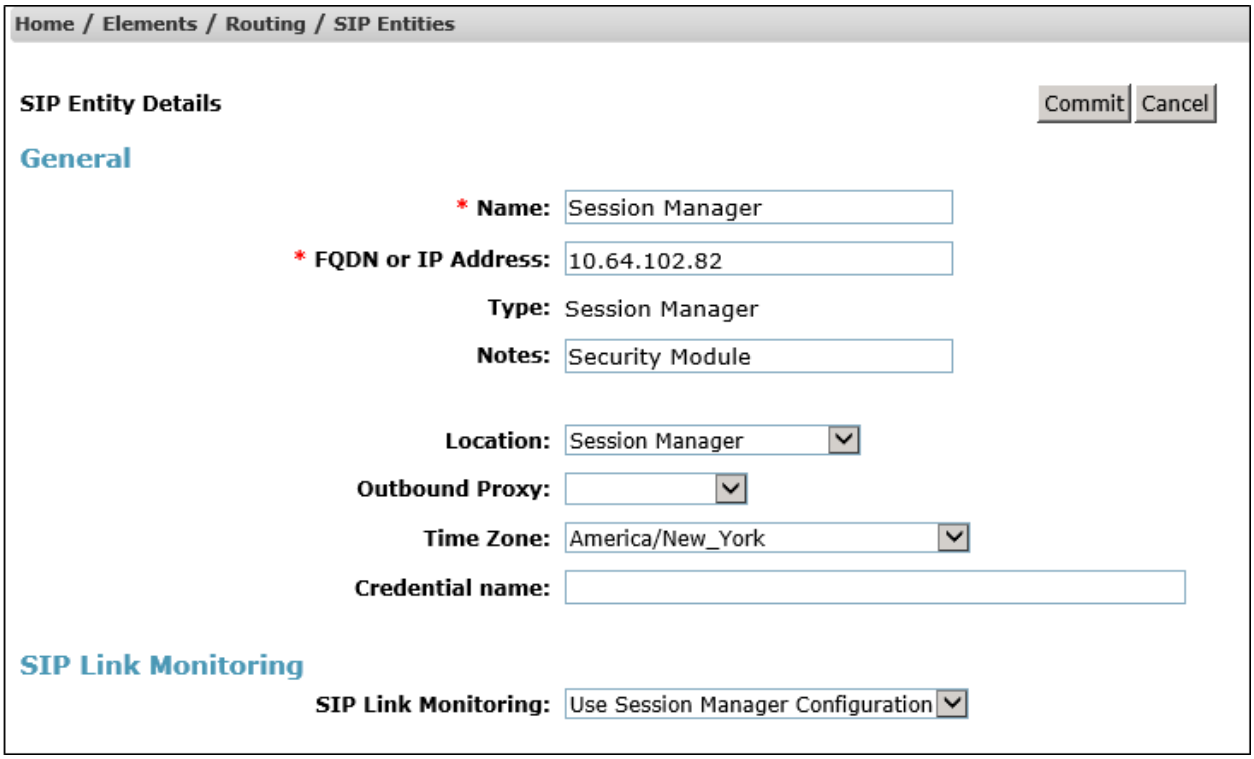


6.4. SIP Entities

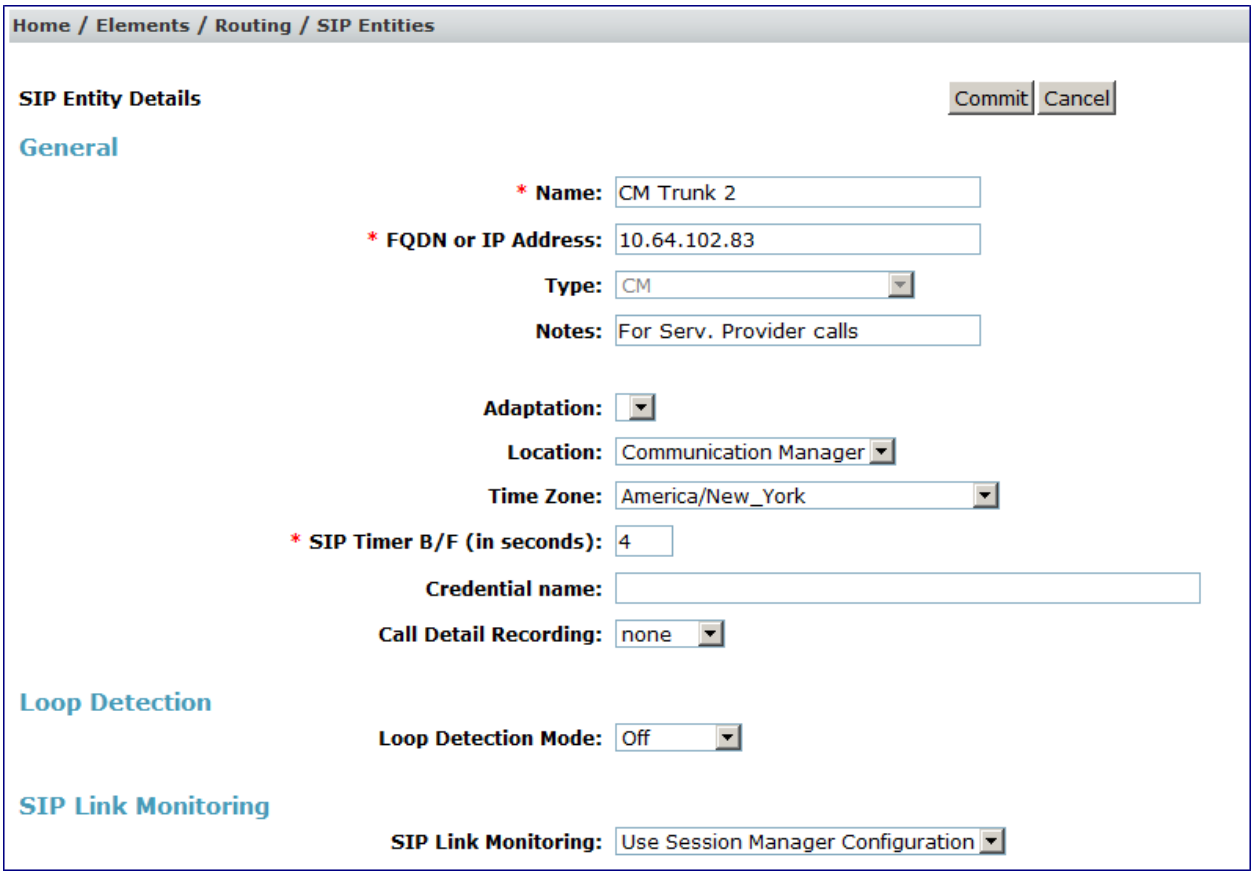
A SIP Entity must be added for Session Manager and for each SIP telephony system connected to it, which includes Communication Manager and the Avaya SBCE. Navigate to **Routing** → **SIP Entities** in the left navigation pane and click on the **New** button in the right pane (not shown). In the **General**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

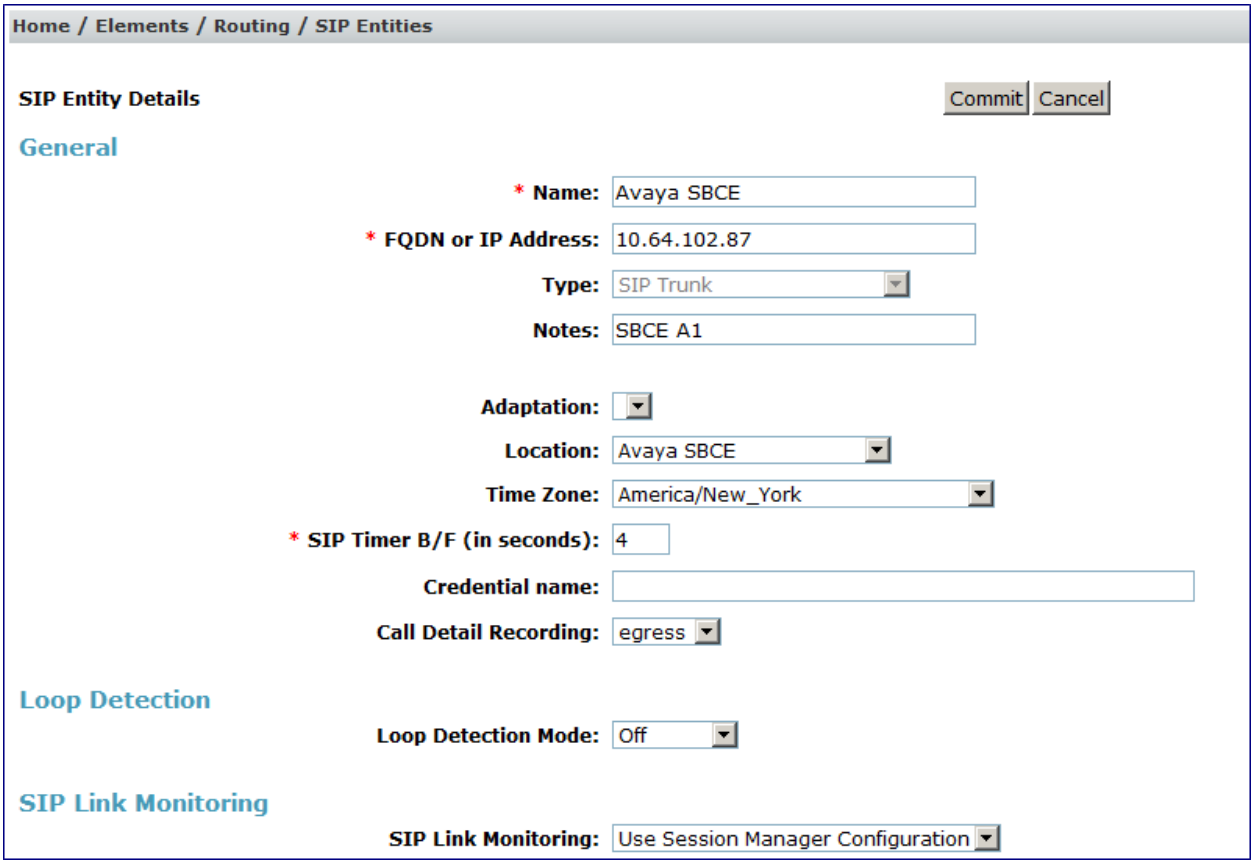
The following screen shows the addition of the Session Manager SIP Entity. The IP address of the Session Manager Security Module is entered in the **FQDN or IP Address** field.



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 than the one created during the Session Manager installation, 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, as seen in **Section 5.3**.



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**).



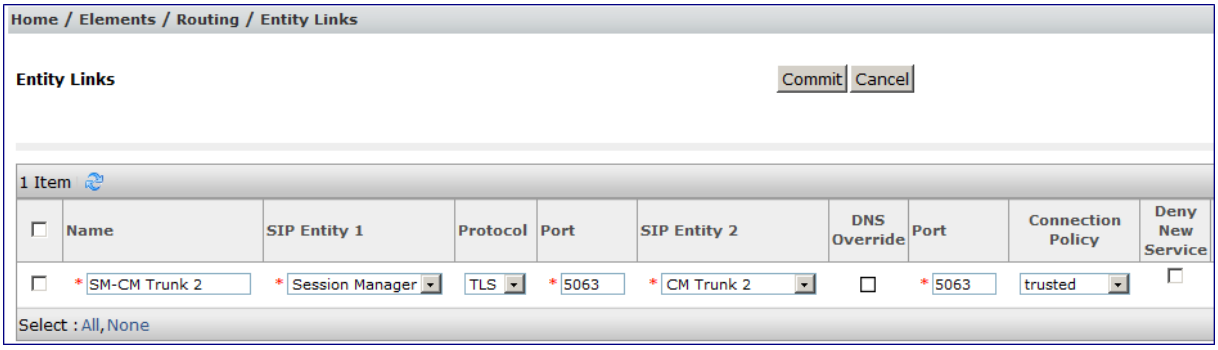
6.5. Entity Links

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:

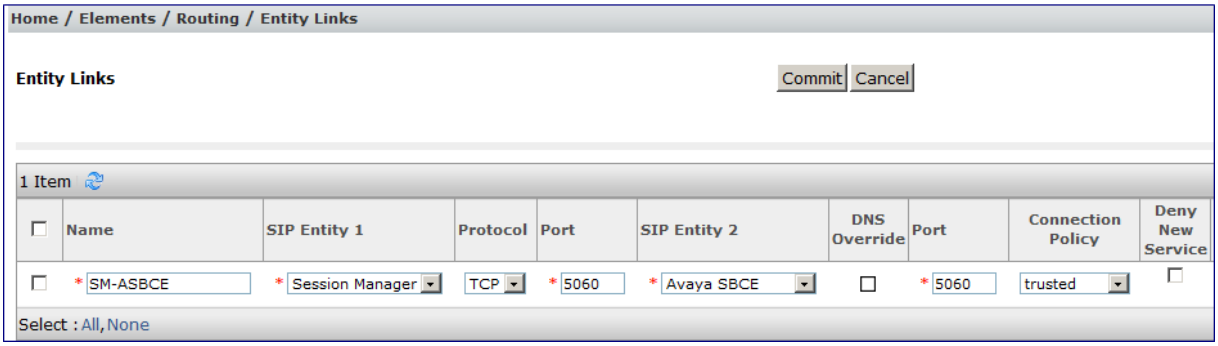
- **Name:** Enter a descriptive name.
- **SIP Entity 1:** Select the Session Manager from the drop-down menu.
- **Protocol:** Select the transport protocol used for this link.
- **Port:** Port number on which Session Manager will receive SIP requests from the far-end.
- **SIP Entity 2:** Select the name 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**.



The Entity Link to the Avaya SBCE is show below. ***TCP*** and port ***5060*** were used.

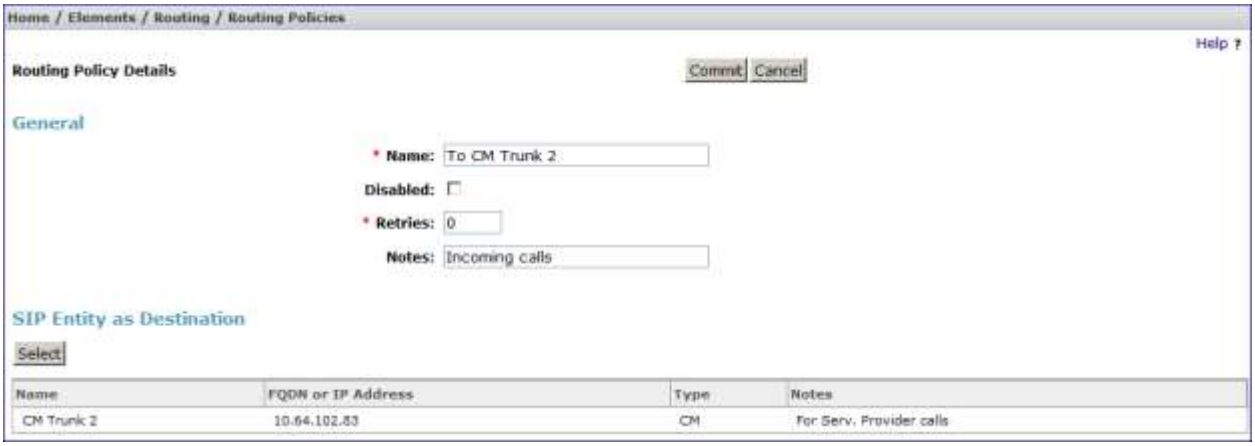


6.6. Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.4**. Two routing policies were added: an incoming policy with Communication Manager as the destination, and an outbound policy to 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 the service provider 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:

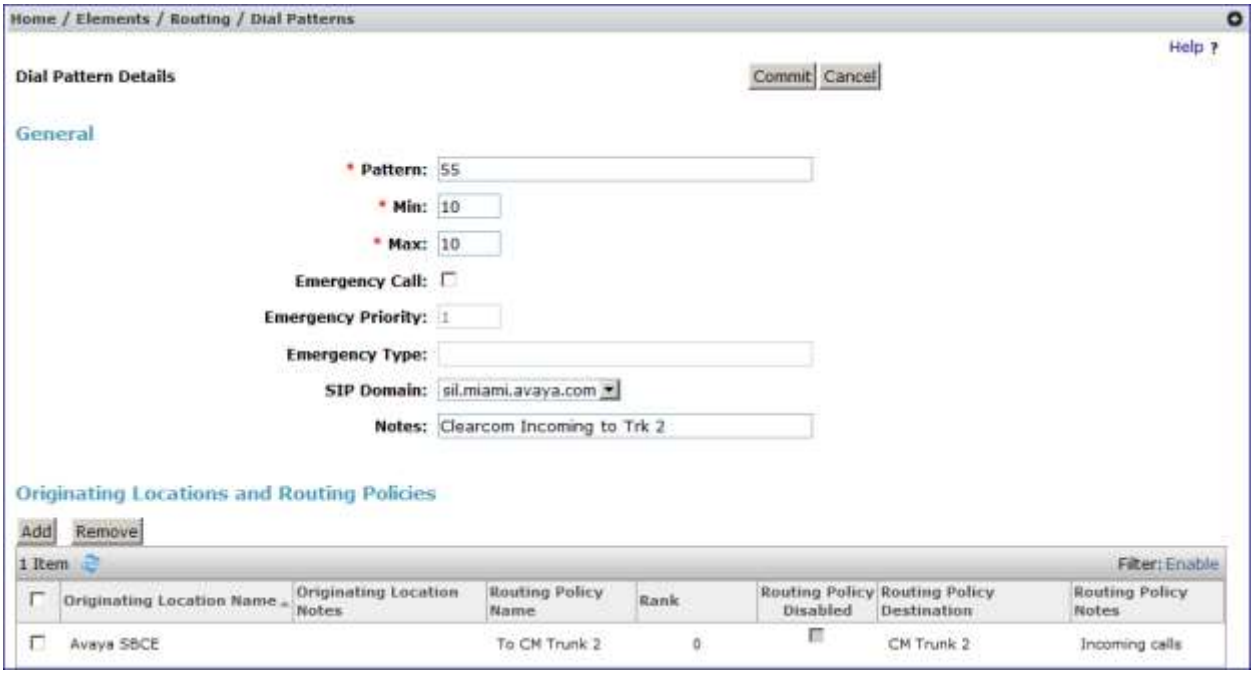
In the **General** section, enter the following values:

- **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**.

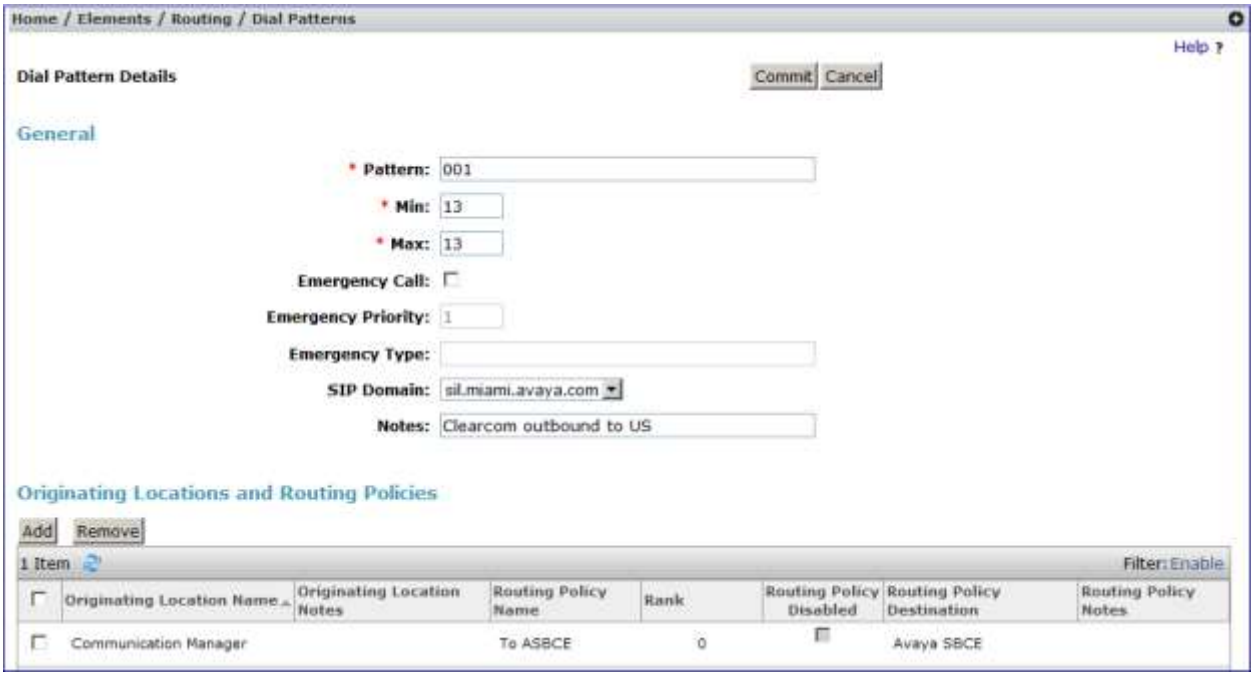
Default values can be used for the remaining fields. Click **Commit** to save.

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 ***55***, which was in the range of the DID numbers assigned by the service provider to the SIP trunk, arriving from location ***Avaya SBCE***, used route policy ***To CM Trunk 2*** to Communication Manager.



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

The following screen shows an example dial pattern used during the compliance test to verify outbound long distance PSTN calls to the United States. The screen shows that 13 digit dialed numbers, beginning with ***001***, arriving from the ***Communication Manager*** location**,** will use route policy ***To ASBCE***, which sends the call out to the PSTN via Avaya SBCE and the Clearcom SIP Trunk.



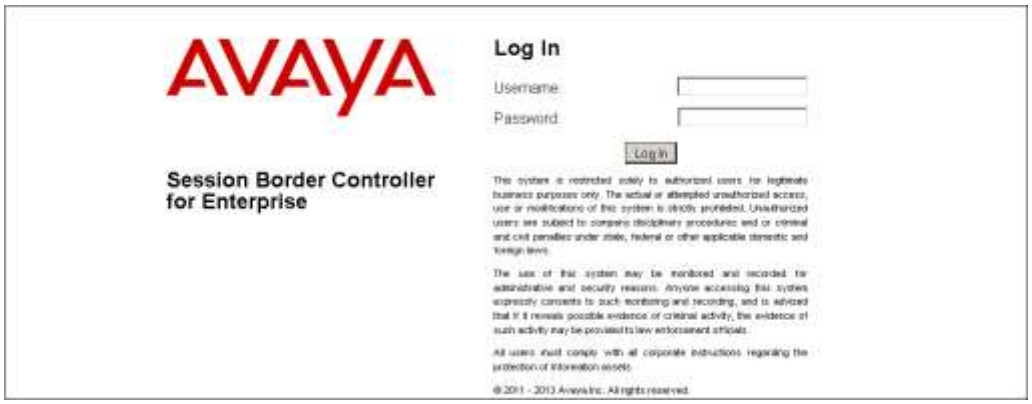
Repeat this procedure as needed, to define additional dial patterns for PSTN numbers to be routed to the service provider's network via the Avaya SBCE.

7. Configure Avaya Session Border Controller for Enterprise

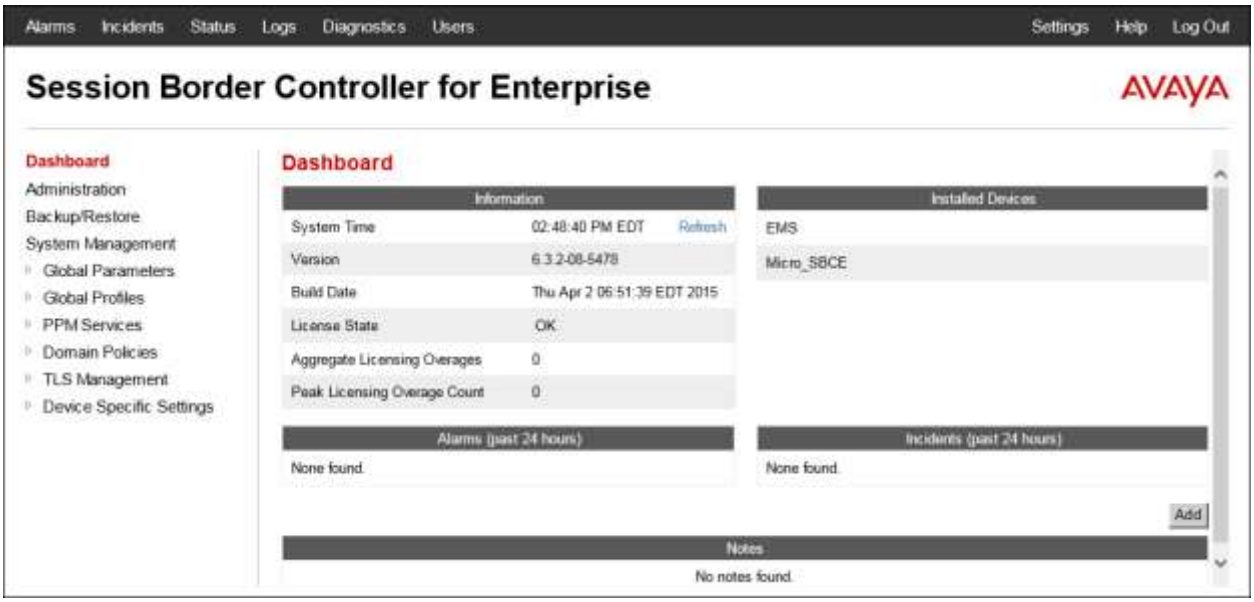
This section describes the configuration of the Avaya SBCE. It is assumed that the initial installation of the Avaya SBCE, the assignment of the management interface IP Address and license installation have already been completed; hence these tasks are not covered in these Application Notes. For more information on the installation and initial provisioning of the Avaya SBCE consult the Avaya SBCE documentation in the **Additional 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.



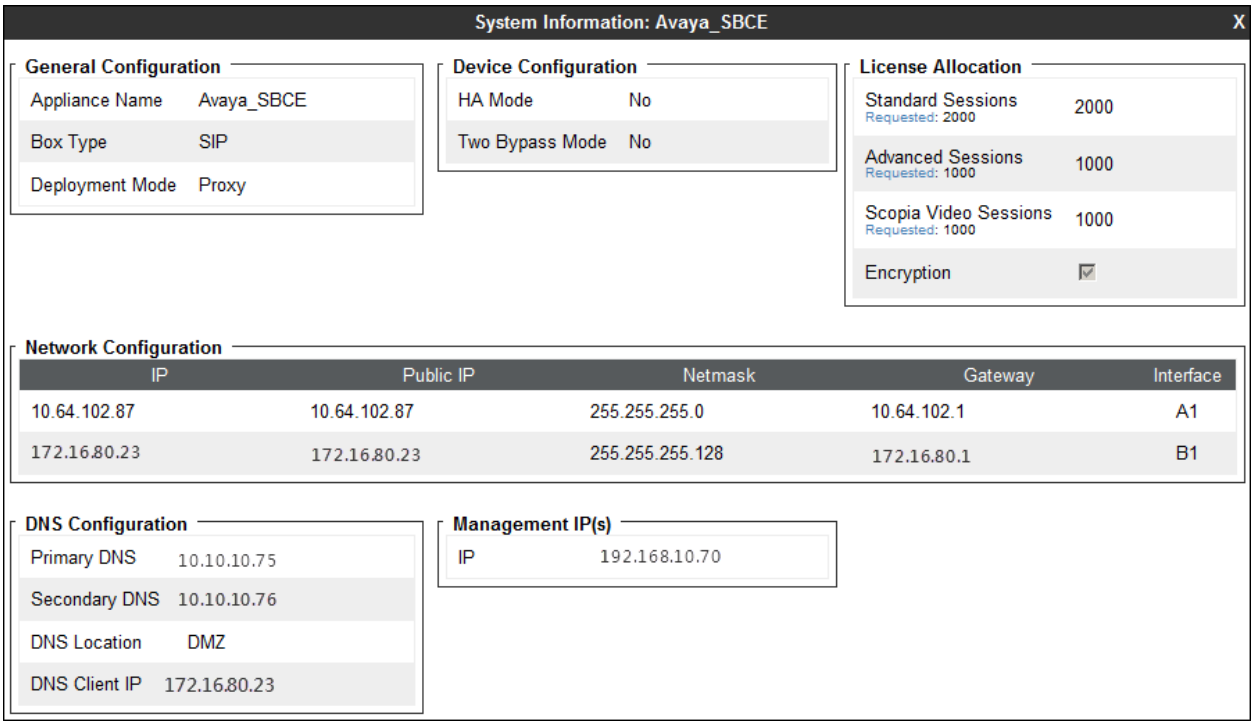
Once logged in, the Dashboard screen is presented. The left navigation pane contains the different available menu items used for the configuration of the Avaya SBCE. Verify that the status of the **License State** field is **OK**, indicating that a valid license is present. Contact an authorized Avaya sales representative if a license is needed.



7.2. System Management

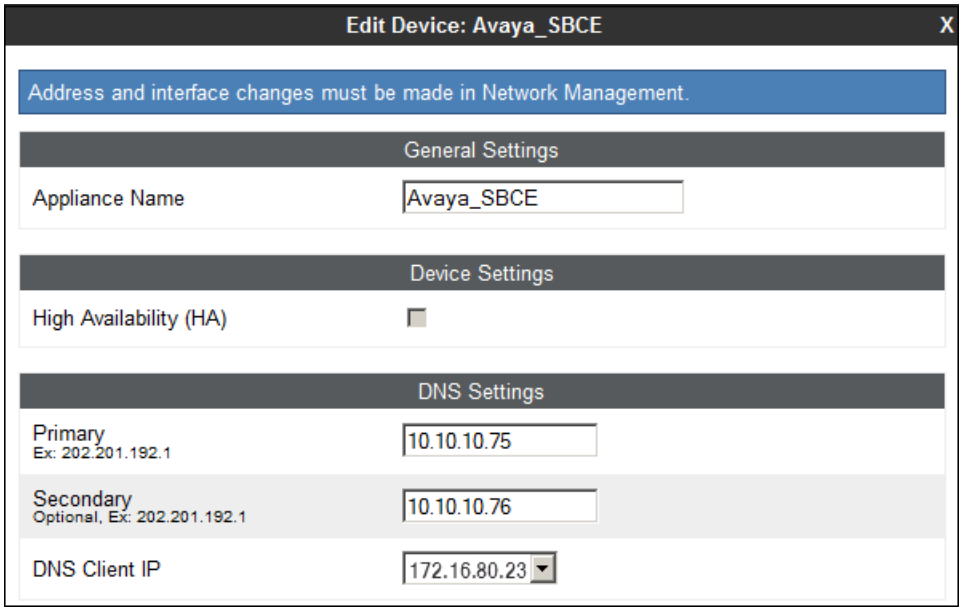
To view current system information, select **System Management** on the left navigation pane. A list of installed devices is shown in the **Devices** tab on the right pane. In the reference configuration, a single device named **Avaya_SBCE** is shown. The management IP address that was configured during 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 all 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 configuration assigned to the Avaya SBCE, click **View** on the screen above. The **System Information** window is displayed, containing the current device configuration and network settings. Note that for security purposes, any public IP addresses used during the compliance test have been masked in this document.



Note that the **A1** and **B1** interfaces correspond to the private and public interfaces of the Avaya SBCE. On the **License Allocation** area of the **System Information**, verify that the number of **Standard Sessions** is sufficient to support the desired number of simultaneous SIP calls across all SIP trunks at the enterprise. The number of sessions and encryption features are primarily controlled by the license file installed.

DNS server configuration can be entered or modified as needed, by clicking **Edit** on the **System Management**/**Devices** tab shown on the previous page. Under **DNS Settings**, enter the IP addresses of the **Primary** and **Secondary** DNS servers. During the compliance test, public DNS servers were used, and the IP address corresponding to the public interface of the Avaya SBCE was selected from the **DNS Client IP** scroll down menu, as shown on the screen below. Click **Finish** (not shown) when done.

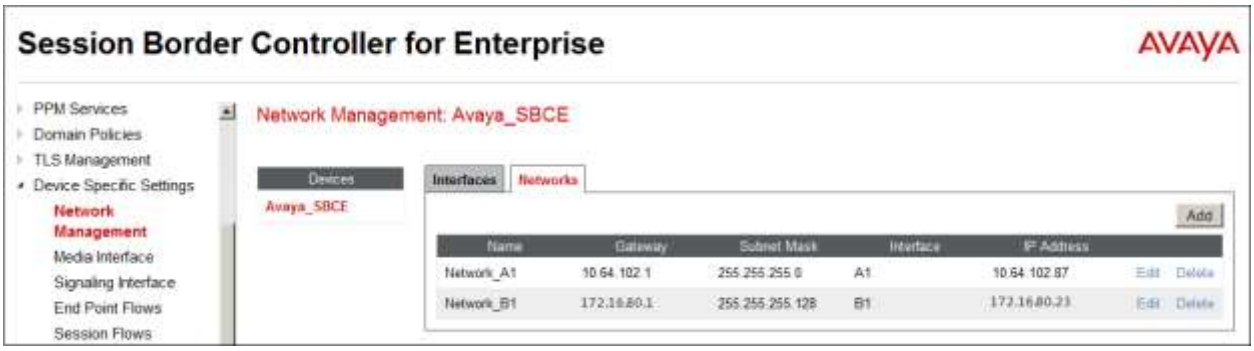


7.3. Network Management

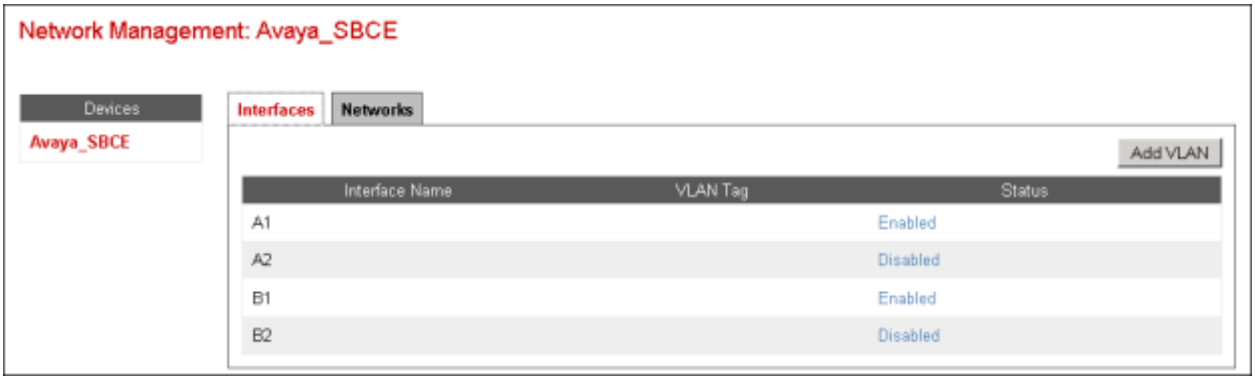
The network configuration parameters should have been previously specified during installation of the Avaya SBCE. In the event that changes need to be made to the network configuration, they can be entered here.

Select **Network Management** from **Device Specific Settings** on the left-side menu. Under **Devices** in the center pane, select the device being managed, **Avaya_SBCE** in the sample configuration. On the **Networks** tab, verify or enter the network information as needed. Note that the **A1** and **B1** interfaces correspond to the private and public interfaces for the Avaya SBCE.

In the configuration used during the compliance test, IP address ***10.64.102.87*** was assigned to interface **A1**, and IP address ***172.16.80.23*** was assigned to interface **B1**.



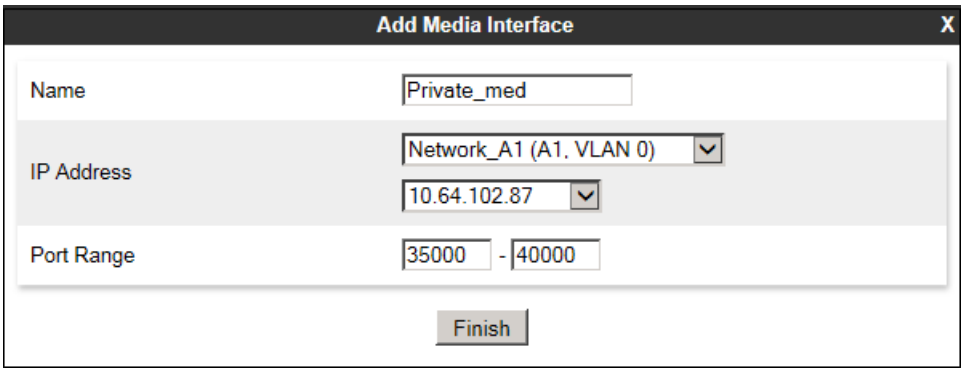
On the **Interfaces** tab, verify the **Administrative Status** is **Enabled** for both the **A1** and **B1** interfaces. Click the buttons under the **Status** column if necessary to enable the interfaces.



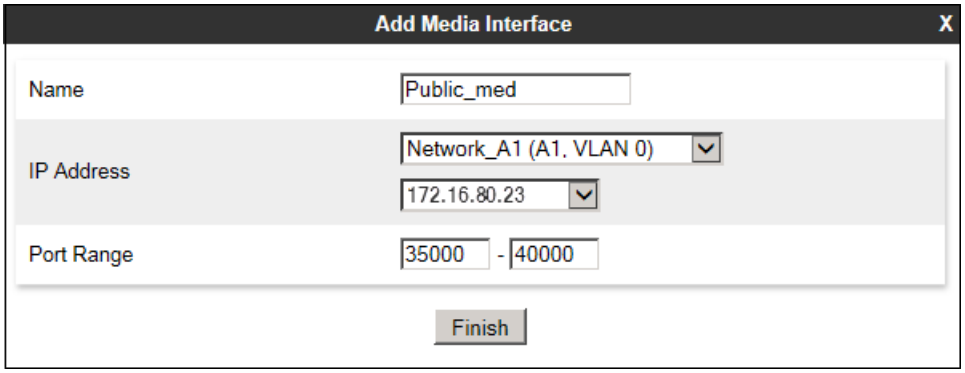
7.4. Media Interfaces

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 the 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. Under **IP Address**, select from the drop-down menus the network and IP address associated with the private interface of the SBCE (A1). The **Port Range** was left at the default values of ***35000-40000***. Click **Finish**.



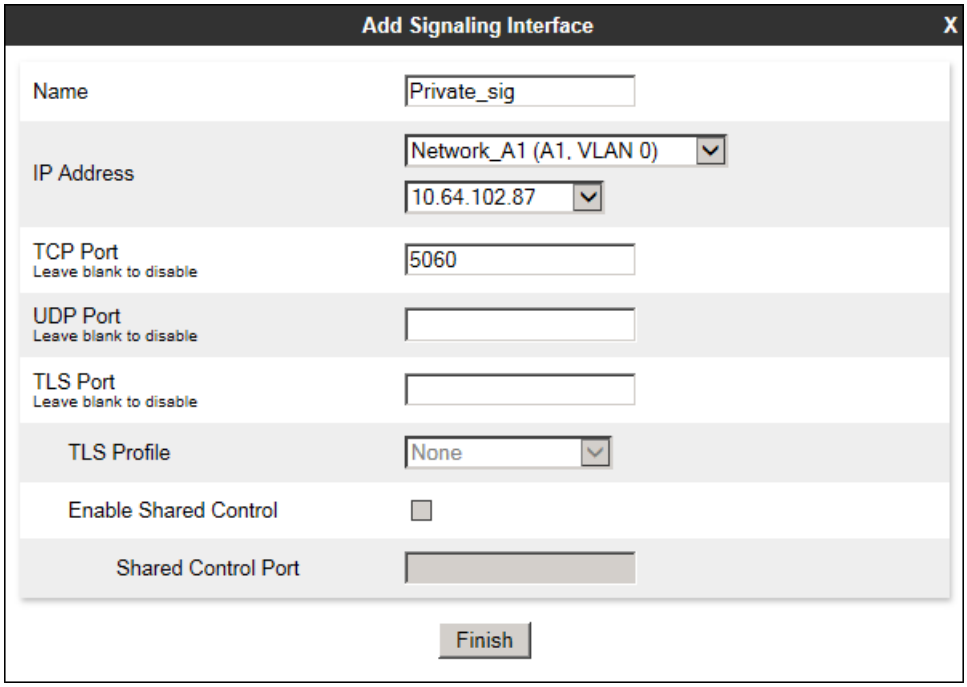
A Media Interface facing the public network side was similarly created with the name **Public_med**, as shown below. Under **IP Address**, the network and IP address associated with the public interface of the SBCE (B1) were selected from the drop-down menus. The **Port Range** was left at the default values. Click **Finish**.



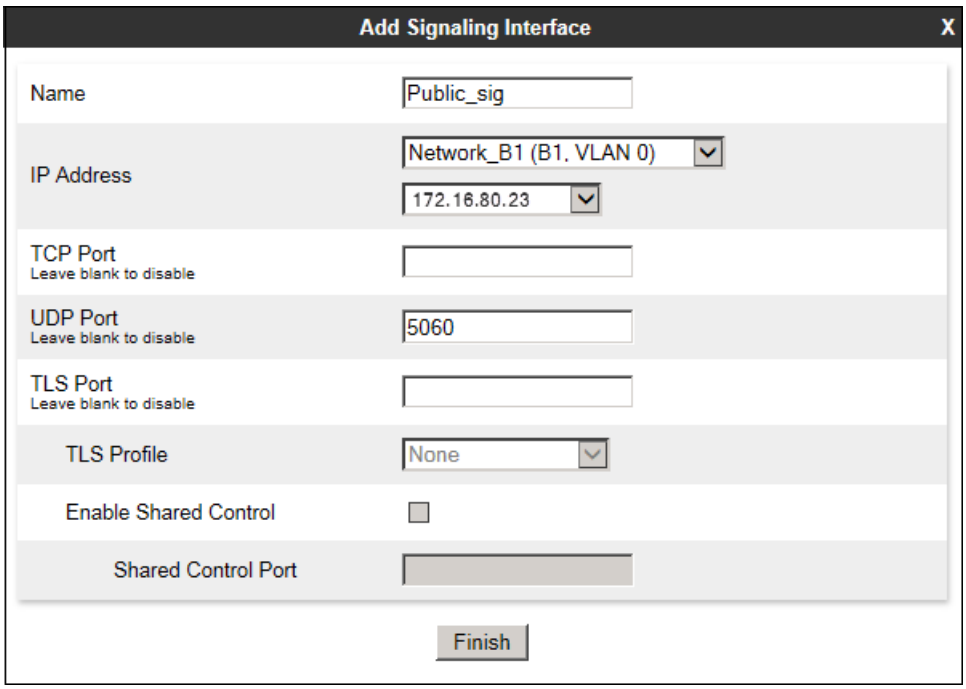
7.5. Signaling Interfaces

Signaling Interfaces are created to specify the IP addresses and ports in which the Avaya SBCE will listen for signaling traffic in the connected 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. Under **IP Address**, select from the drop-down menus the network and IP address associated with the private interface of the SBCE (A1). Enter ***5060*** for **TCP Port**, since TCP port 5060 is used to listen for signaling traffic from Session Manager in the sample configuration, as defined in **Section 6.5**. Click **Finish**.



A second Signaling Interface with the name **Public_sig** was similarly created in the service provider's direction. Under **IP Address**, the network and IP address associated with the public interface of the SBCE (B1) were selected from the drop-down menus. Enter ***5060*** for **UDP Port**, since this is the protocol and port used by the Avaya SBCE to listen to the service provider's SIP traffic. Click **Finish**.

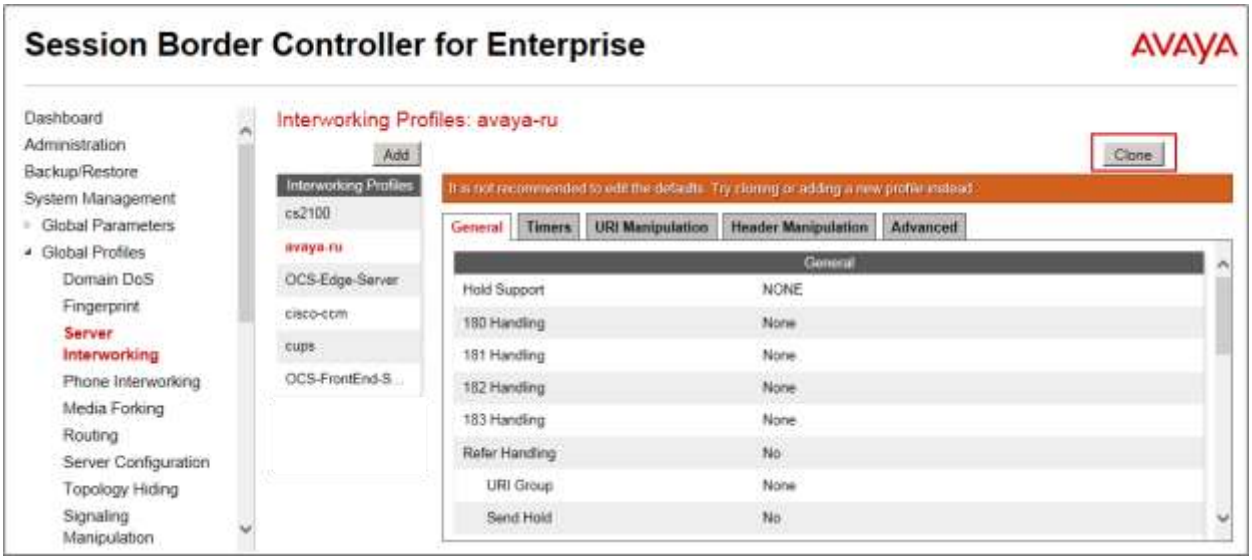


7.6. Server Interworking

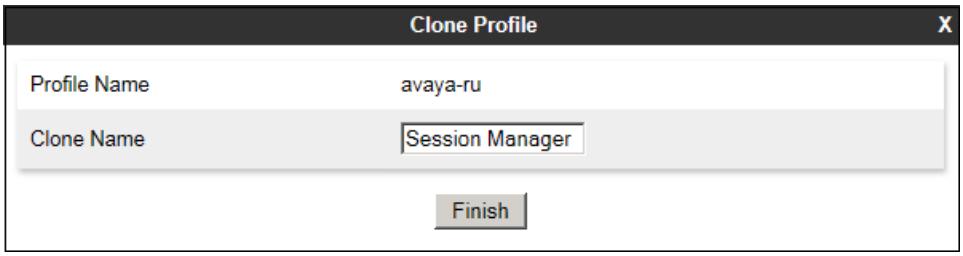
Interworking Profile features are configured to facilitate the interoperability between the enterprise SIP-enabled solution (Call Server) and the SIP trunk service provider (Trunk Server). In the reference configuration, Session Manager functions as the Call Server and the Clearcom SIP Proxy as the Trunk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

7.6.1. Server Interworking Profile – Enterprise

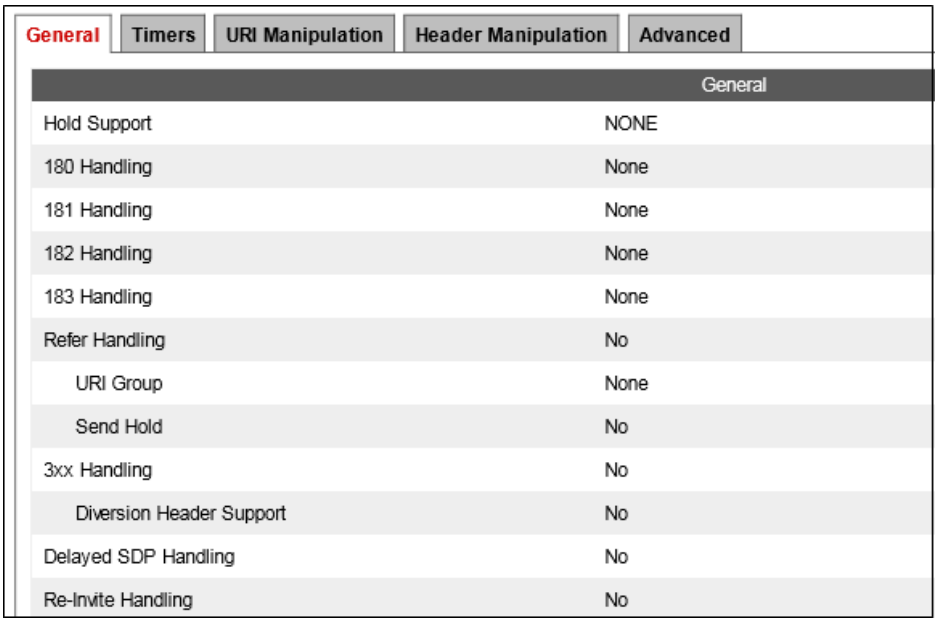
Interworking profiles can be created by cloning one of the pre-defined default profiles, or by adding a new profile. To configure the interworking profile in the enterprise direction, select **Global Profiles → Server Interworking** on the left navigation pane. Under **Interworking Profiles**, select ***avaya-ru*** from the list of pre-defined profiles. Click **Clone**.



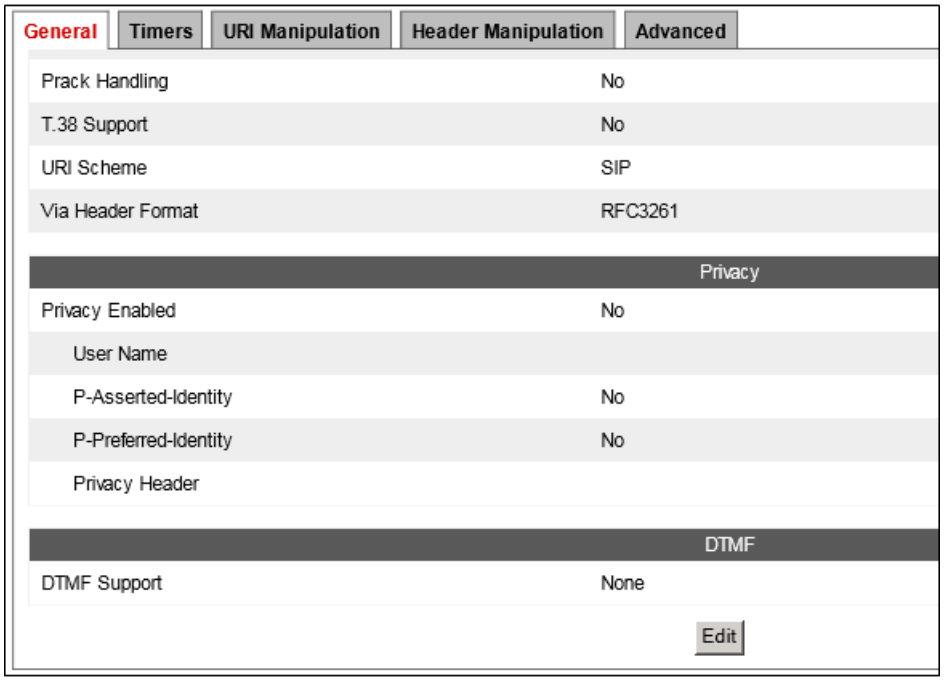
Enter a descriptive name for the cloned profile. Click **Finish**.



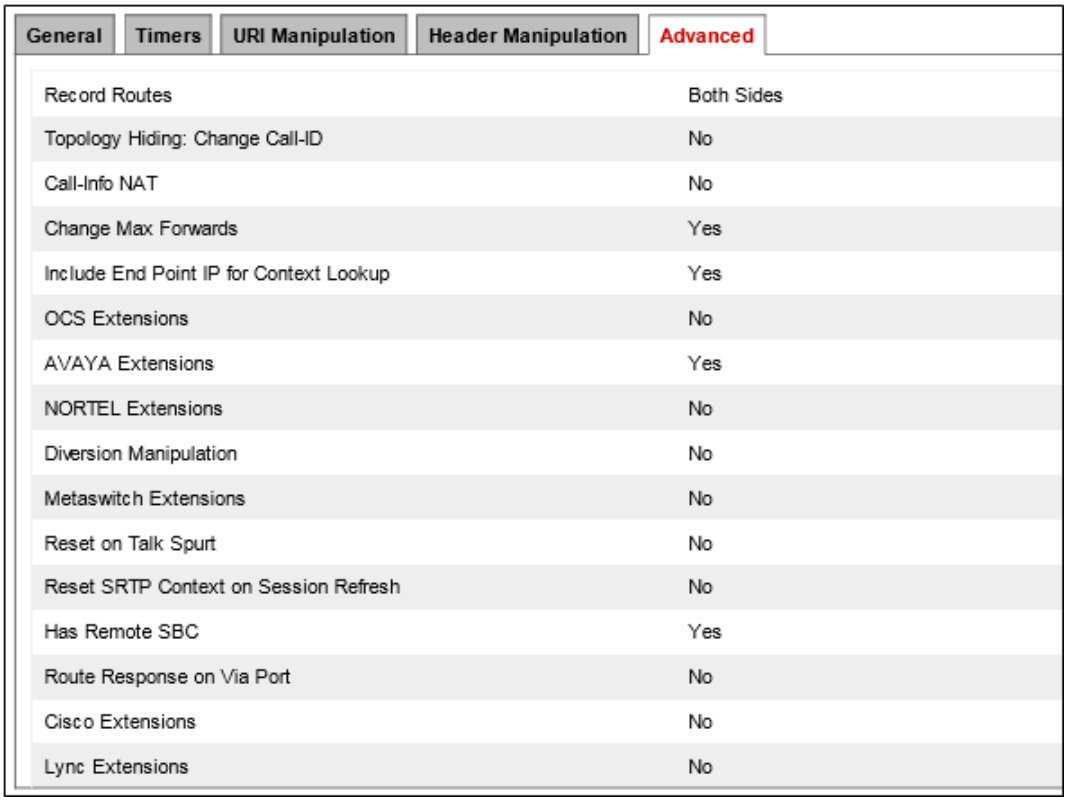
On the newly cloned ***Session Manager*** interworking profile, verify the settings on the **General** tab:



Scroll down to the bottom of the tab to see the rest of the settings. Click **Edit** if changes to any of the parameters are needed.



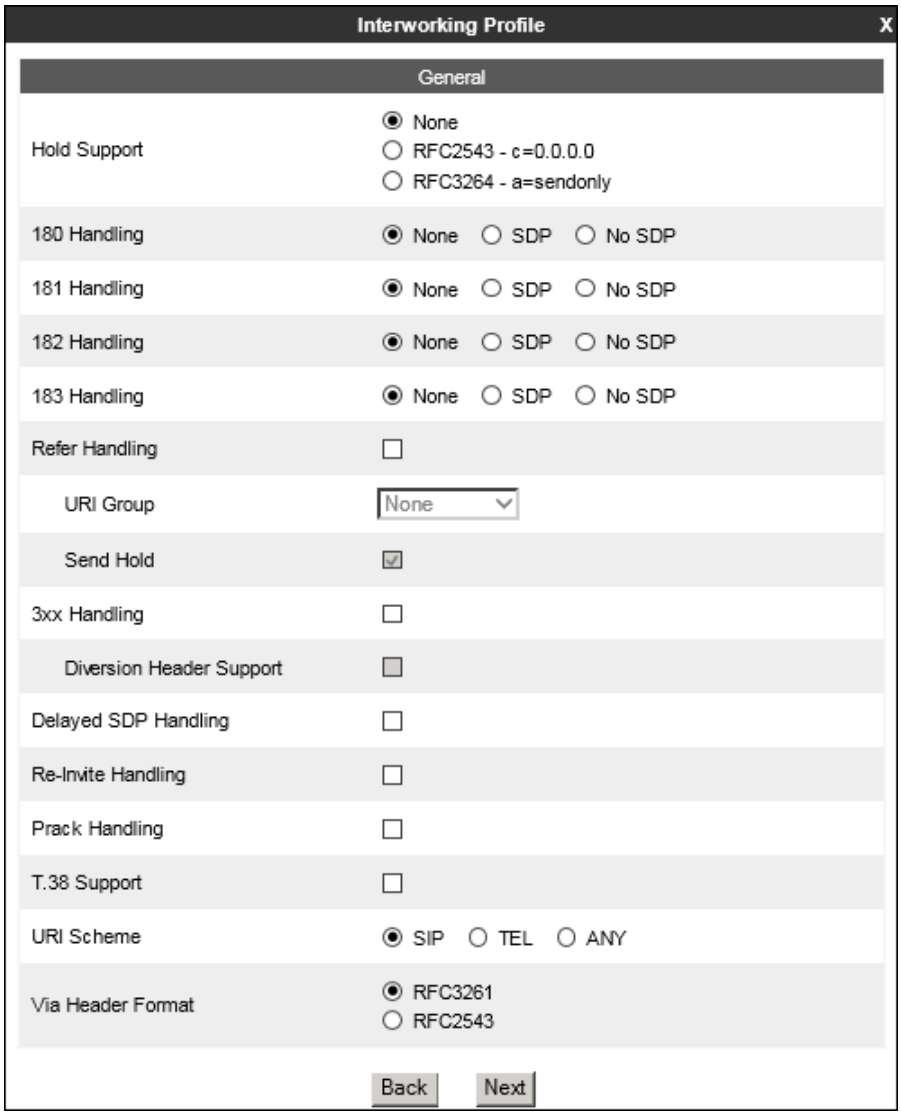
The **Timers**, **URI Manipulation** and **Header Manipulation** tabs contain no entries.
The **Advaced** tab settings are shown on the screen below:



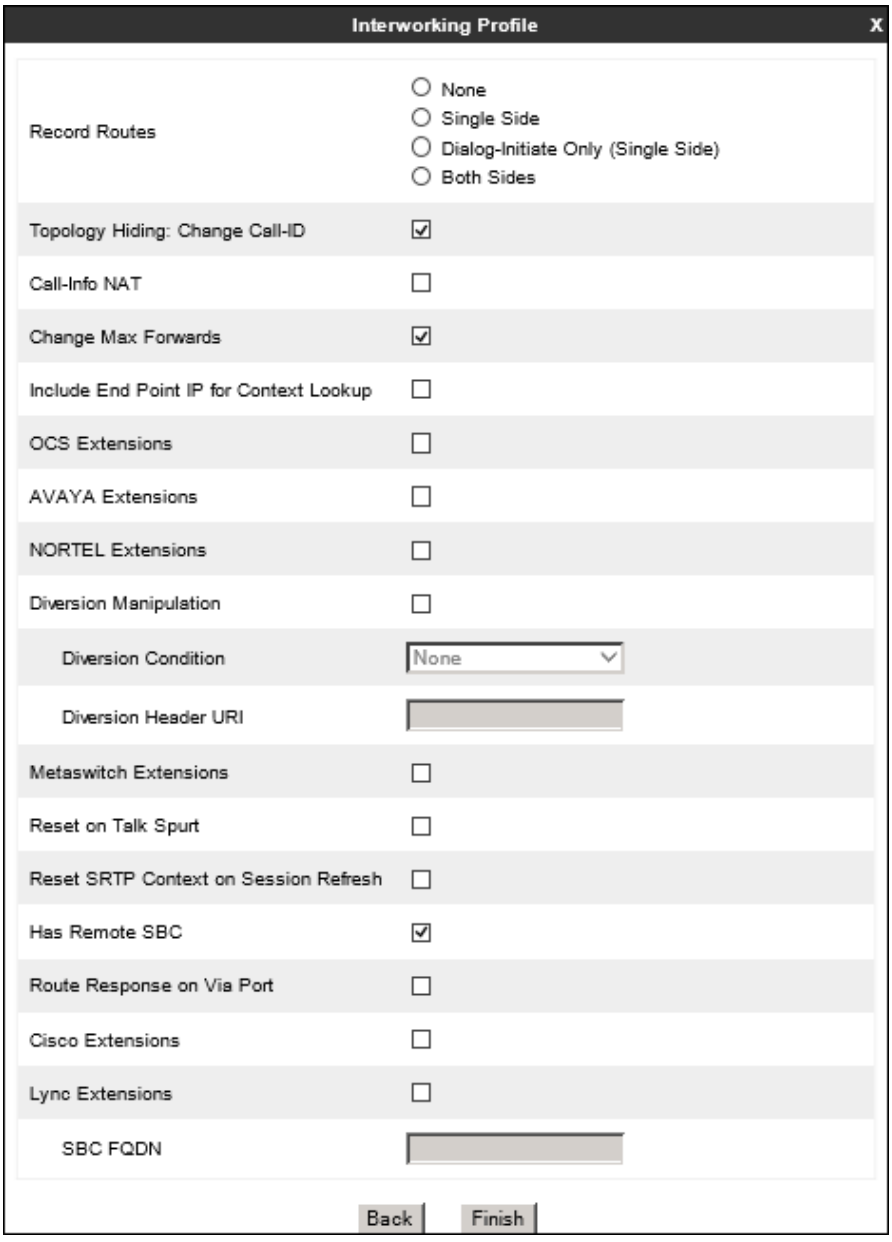
7.6.2. Server Interworking Profile – Service Provider

A second interworking profile in the direction of the SIP trunk was created, by adding a new profile in this case. Select **Global Profiles → Server Interworking** on the left navigation pane and click **Add** (not shown). Enter a descriptive name for the new profile. Click **Next**.

On the **General** screen, all parameters retain their default values. Click **Next**.



Click **Next** on the **Privacy/DTMF** and **SIP Timers/Transport Timers** tabs (not shown). Accept all defaults in the **Advanced Settings** tab. Click **Finish**.



7.7. Signaling Manipulation

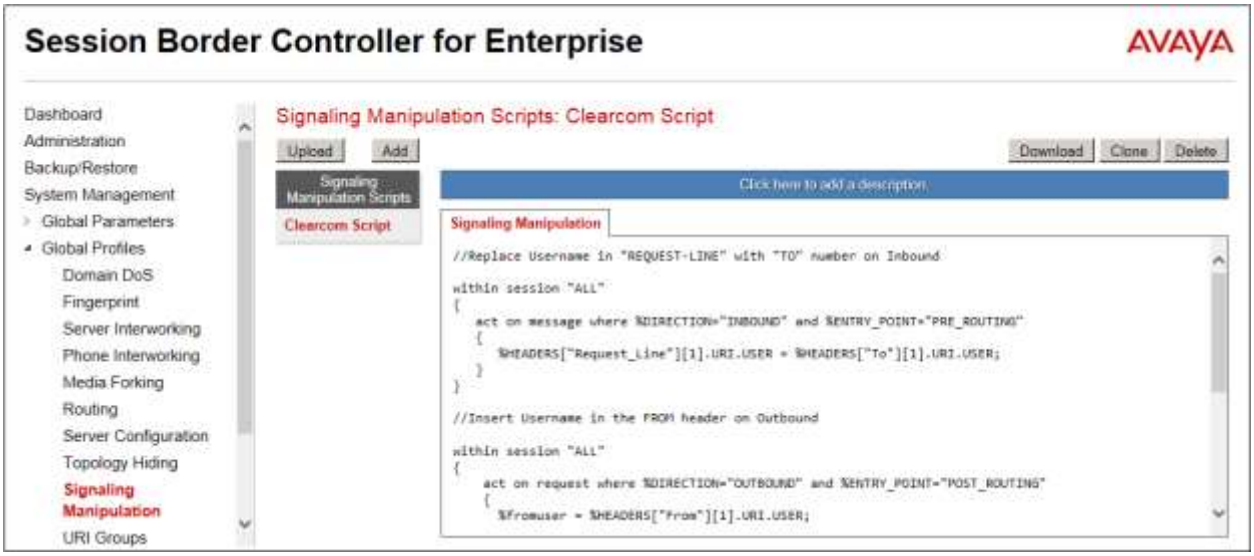
The screen below shows the finished Signaling Manipulation script named ***Clearcom Script*** created during the compliance test. This script was used to:

- Include the SIP trunk credential's username in the "From" header of all outbound calls.
- Copy the destination DID number present in the "To" header of incoming calls to the "Request-URI" header.
- Remove the "gsid" and "epv" parameters from outbound "Contact" headers.

See **Section 2.2** for more details on why these header manipulations were needed.

The script will later be applied to the Server Configuration profile corresponding to the service provider, later in **Section 7.8.2**.

To add a Signaling Manipulation script, from the **Global Profiles** menu on the left panel, select **Signaling Manipulation**. Click **Add** to open the SigMa Editor screen, where the text of the script can be entered or copied.



The details of the script used can be found in **Appendix A** in this document.

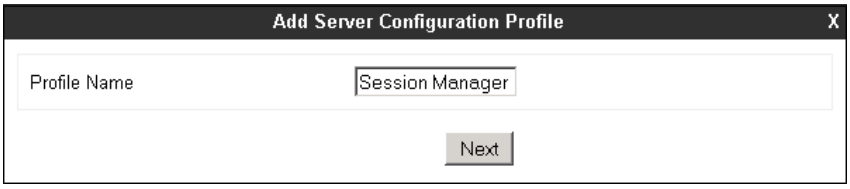
Note: Additional Avaya SBCE header manipulation was performed to remove unnecessary headers from outbound messages, by implementing Signaling Rules later in **Section 7.11**.

7.8. Server Configuration

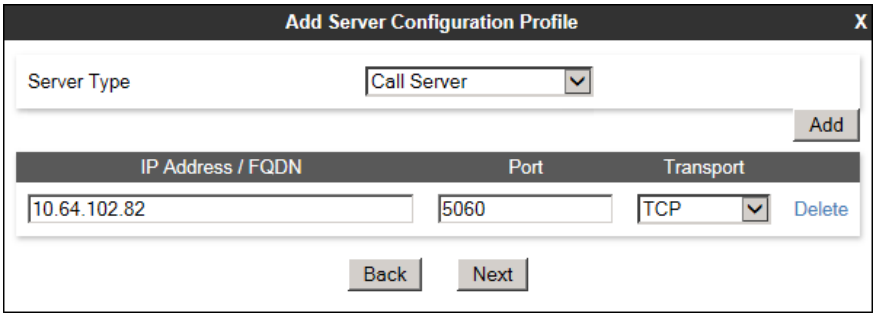
Server Profiles are created to define the parameters for the Avaya SBCE peers; Session Manager (Call Server) at the enterprise and the SIP Proxy at the service provider network (Trunk Server).

7.8.1. Server Configuration Profile – Enterprise

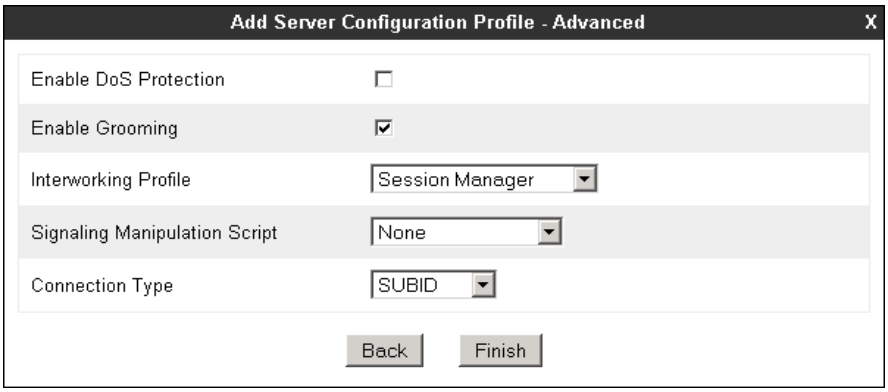
From the **Global Profiles** menu on the left-hand navigation pane, select **Server Configuration** and click the **Add** button (not shown) to add a new profile for the Call Server. Enter an appropriate **Profile Name** similar to the screen below. Click **Next**.



On the **Add Server Configuration Profile** Tab select ***Call Server*** from the drop down menu under the **Server Type**. On the **IP Addresses / FQDN** field**,** enter the IP address of the Session Manager Security Module (**Section 6.4)**. Enter ***5060*** under **Port** and select ***TCP*** for **Transport**. The transport protocol and port selected here must match the values defined for the Entity Link to the Session Manager previously in **Section 6.5**. Click **Next**.

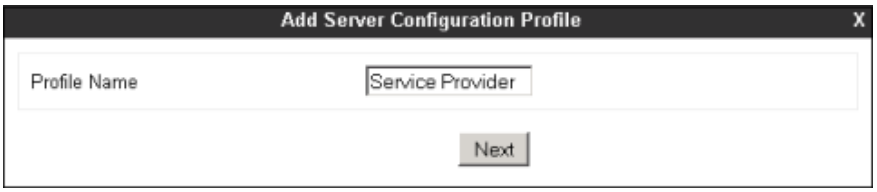


Click **Next** on the **Authentication** and **Heartbeat** tabs (not shown). On the **Advanced** tab, since TCP is used, check the **Enable Grooming** box. Select ***Session Manager*** from the **Interworking Profile** drop down menu. Click **Finish**.

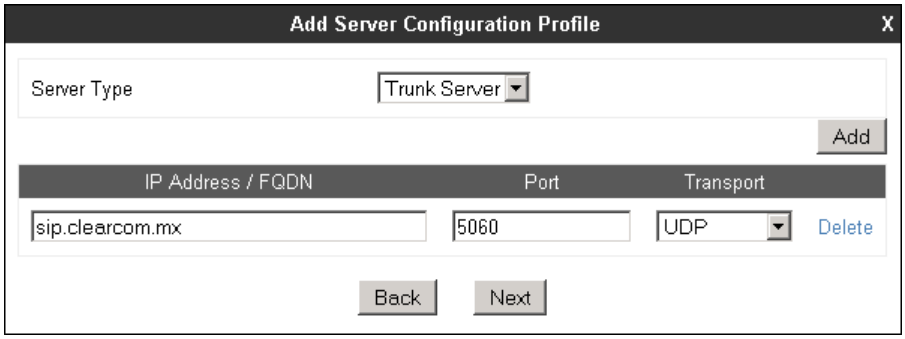


7.8.2. Server Configuration Profile – Service Provider

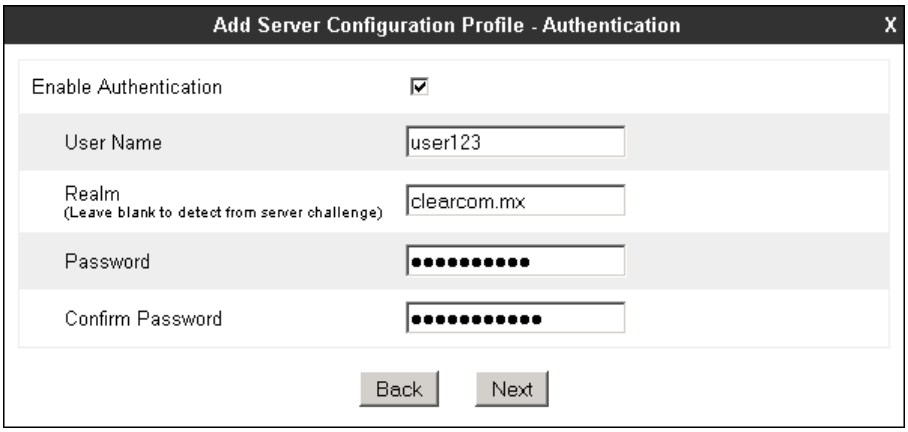
Similarly, to add the profile for the Trunk Server, click the **Add** button on the **Server Configuration** screen (not shown). Enter an appropriate **Profile Name** similar to the screen below. Click **Next**.



On the **Add Server Configuration Profile** Tab select ***Trunk Server*** from the drop down menu for the **Server Type**. On the **IP Addresses / FQDN** field, enter ***sip.clearcom.mx***, the Fully Qualified Domain Name of the service provider SIP proxy server. This information was provided by Clearcom. Enter ***5060*** under **Port**, and select **UDP** for **Transport**. Click **Next**.

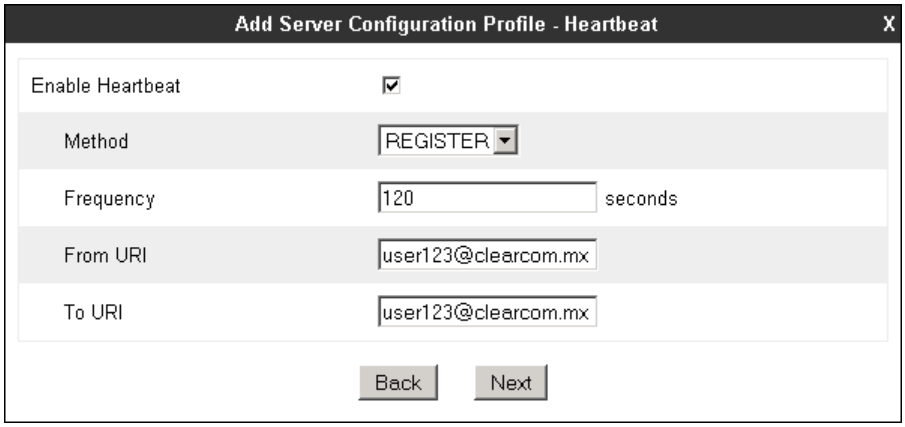


On the **Authentication** tab, check the **Enable Authentication** box. Enter the **User Name**, **Realm** and **Password** credential information supplied by the service provider for the authentication of the SIP trunk. Click **Next**.

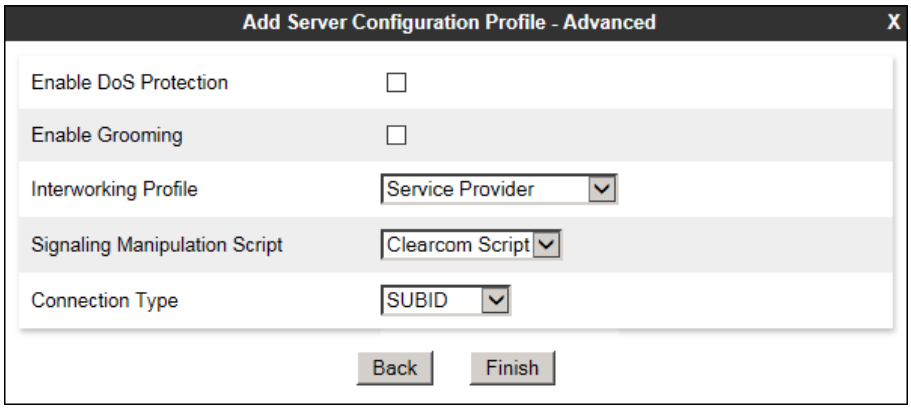


On the **Heartbeat** tab:

- 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 Clearcom proxy server in order to refresh the registration binding of the SIP trunk. **120** 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 ***clearcom.mx*** like shown on the example below.
- Click **Next**.



On the **Advanced** tab, select ***Service Provider*** from the **Interworking Profile** drop down menu. Under **Signaling Manipulation Script**, select the ***Clearcom Script*** created in **Section 7.7**. Click **Finish**.

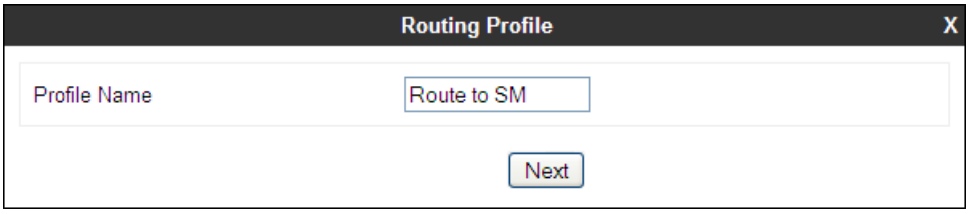


7.9. Routing

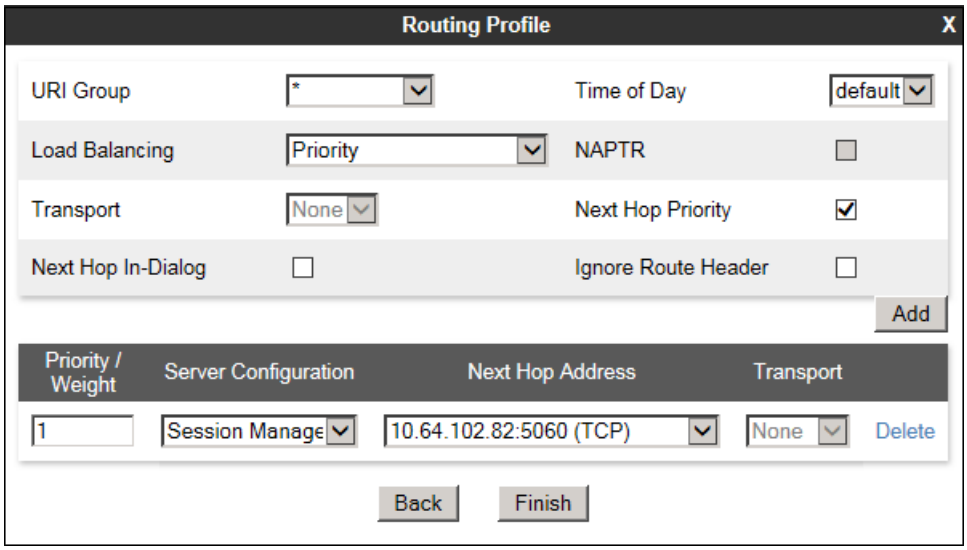
Routing profiles define a specific set of routing criteria that is used, in addition to other types of domain policies, to determine the path that the SIP traffic will follow as it flows through the Avaya SBCE interfaces. Two Routing Profiles were created in the test configuration, one for inbound calls, with Session Manager as the destination, and the second one for outbound calls, which are routed to the Clearcom SIP trunk.

7.9.1. Routing Profile – Enterprise

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**.

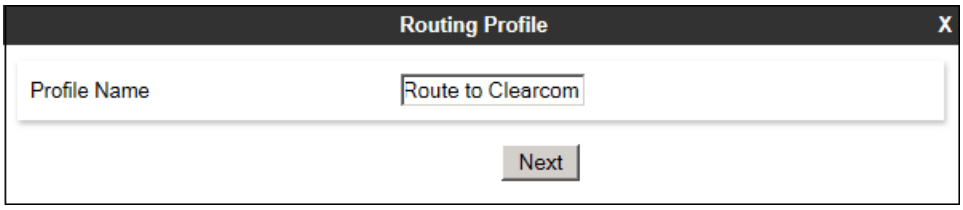


On the **Routing Profile** tab, click the **Add** button to enter the next-hop address. Enter ***1*** under **Priority/Weight**. Under **Server Configuration**, select ***Session Manager***. The **Next Hop Address** field will be populated with the IP address, port and protocol defined for the Session Manager Server Configuration Profile in **Section 7.8.1**. Defaults were used for all other parameters. Click **Finish**.

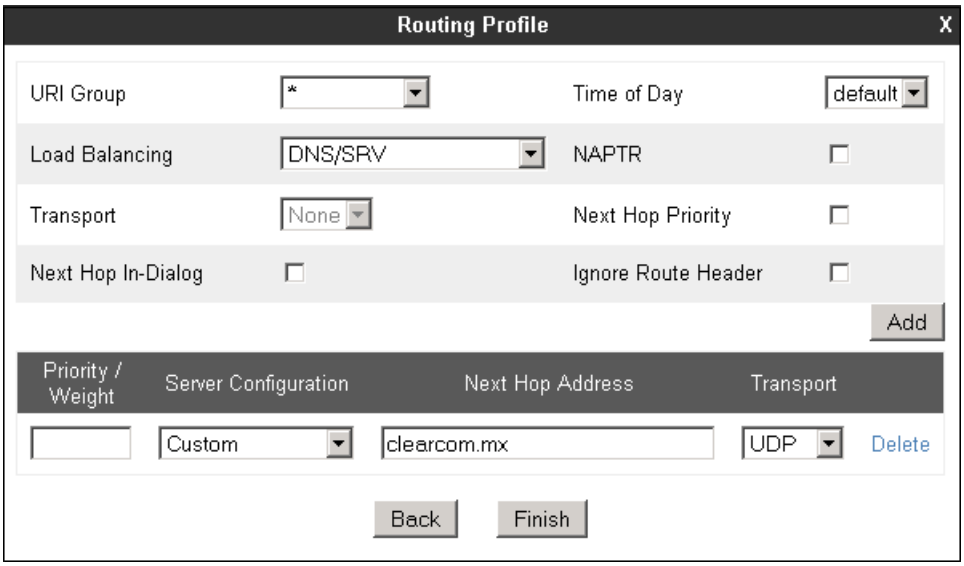


7.9.2. Routing Profile – Service Provider

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** similar to the example below. Click **Next**.



On the **Routing Profile** tab, select ***DNS/SRV*** under **Load Balancing**. Uncheck the **Next Hop Priority** box. Click the **Add** button. Under **Server Configuration**, select ***Custom***. On the **Next Hop Address** field, enter the Service Provider domain ***clearcom.mx***. Under **Transport** select ***UDP***. Defaults were used for all other parameters. Click **Finish**.



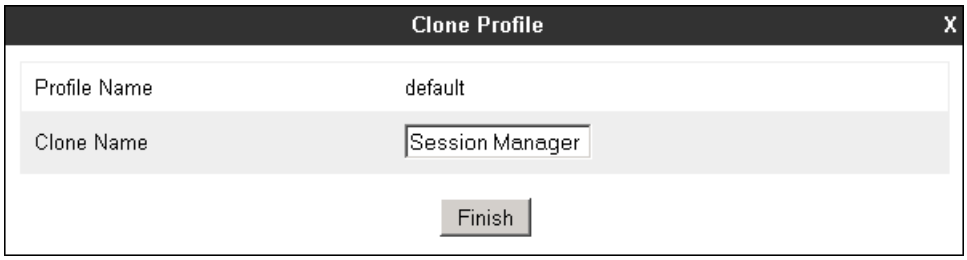
7.10. Topology Hiding

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, the default Topology Hiding Profile was cloned and modified accordingly. 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.

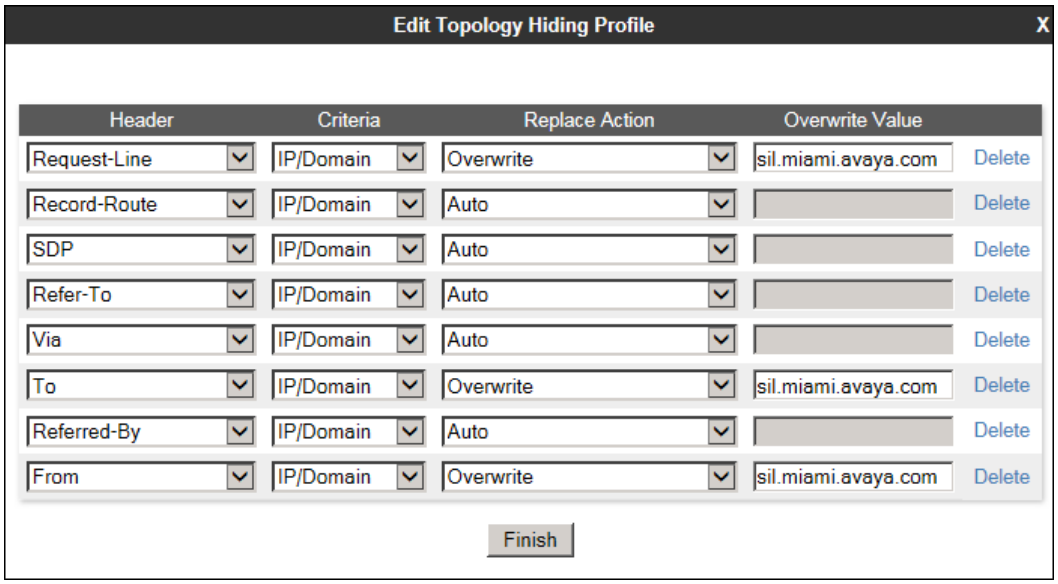
7.10.1. Topology Hiding Profile – Enterprise

To add the Topology Hiding Profile in the enterprise direction, select **Topology Hiding** from the **Global Profiles** menu on the left-hand side, select ***default*** from the list of pre-defined profiles and click the **Clone** button (not shown). Enter a **Clone Name** such as the one shown below. Click **Finish**.



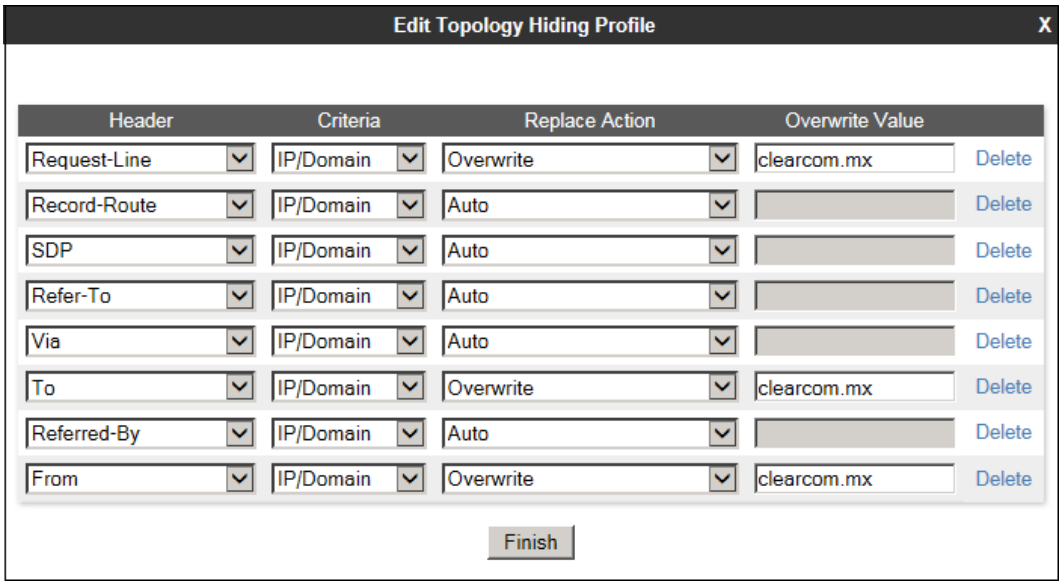
On the newly cloned **Session Manager** profile screen, click the **Edit** button (not shown).

For the **Request-Line**, **To** and **From** headers, select ***Overwrite*** in the **Replace Action** column and enter the enterprise SIP domain ***sil.miami.avaya.com***, in the **Overwrite Value** column of these headers, as shown below. This is the domain known by Session Manager, defined in **Section 6.2**. Default values were used for all other fields. Click **Finish**.



7.10.2. Topology Hiding Profile – Service Provider

A Topology Hiding profile named ***Service Provider*** was similarly configured in the direction of the SIP trunk to the service provider. In this case, for the **Request-Line**, **To** and **From** headers, select **Overwrite** in the **Replace Action** column. In the **Overwrite Value** column, enter the SIP domain ***clearcom.mx***, used and expected by the service provider on these headers. Click **Finish**.



7.11. Signaling Rules

In the reference configuration, two Signaling Rules were created in order to block unnecessary headers from being propagated outside of the enterprise boundaries.

7.11.1. Signaling Rule – Enterprise

A signaling rule was used to remove (block) the following headers, sent in SIP messages from the Session Manager to the Avaya SBCE:

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

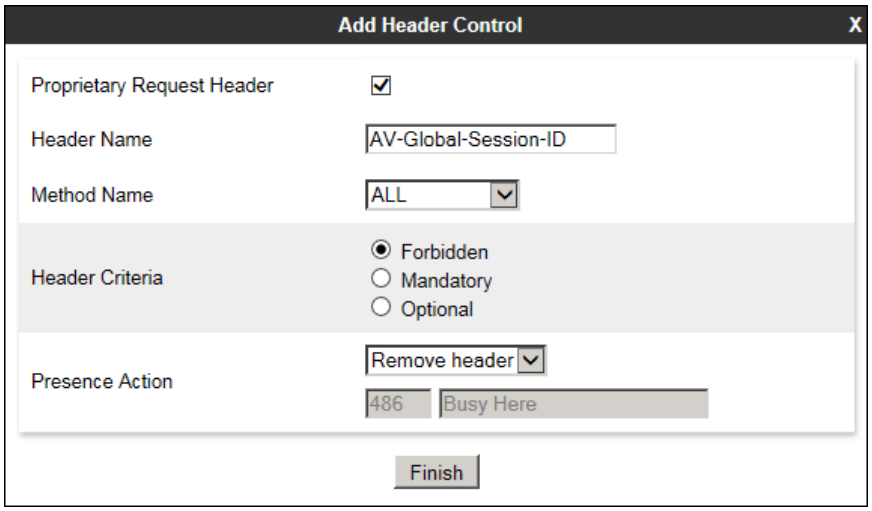
These headers contain private IP addresses and SIP Domains from the enterprise, which should not be propagated outside of the enterprise boundaries. They also add unnecessary size to outbound messages, while they have no significance to the service provider.

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**.

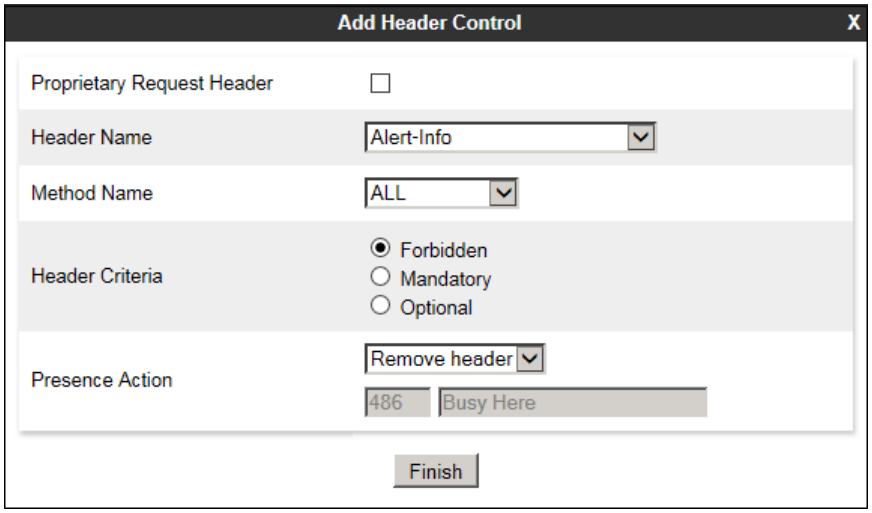
Click **Next** on the next four tabs (not shown), leaving all fields in sections **Inbound Outbound**, **Content-Type Policy**, **QoS** and **UCDI** with their default values, then click **Finish**. On the newly created Signaling Rule, select the **Request Headers** tab to create the manipulations performed on request messages. Select **Add In Header Control**.

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

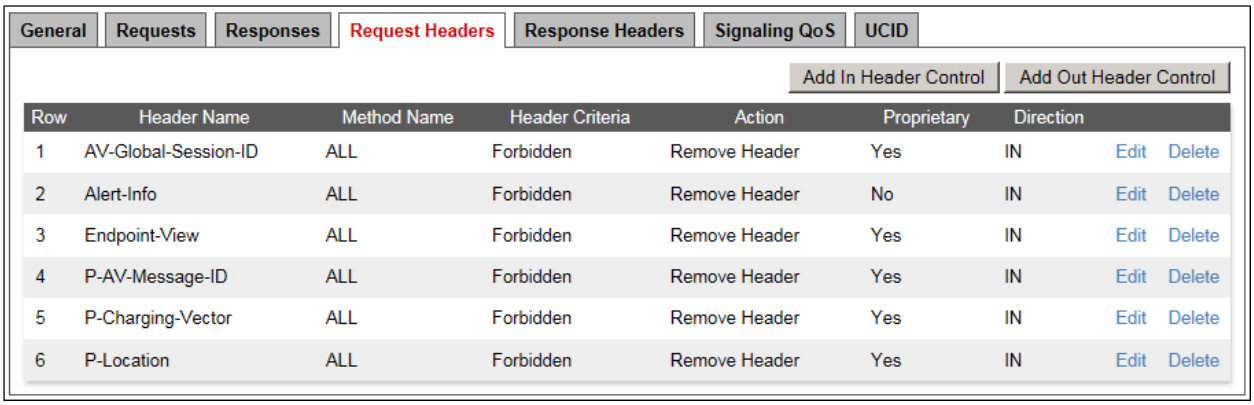
- Check the **Proprietary Request Header** box.
- **Header Name:** Type in the header name, ***AV-Global-Session-ID*** in this example.
- **Method Name:** Select ***ALL***.
- **Header Criteria:** Check **Forbidden**.
- **Presence Action:** Select ***Remove Header***.
- Click **Finish**.



The remaining header control rules are similarly configured, by selecting **Add In Header Control** as needed. The exception is the Alert-Info header, where the **Proprietary Request Header** box is not checked, and the **Header name** is selected directly from the scroll down menu, as shown on the screen below.

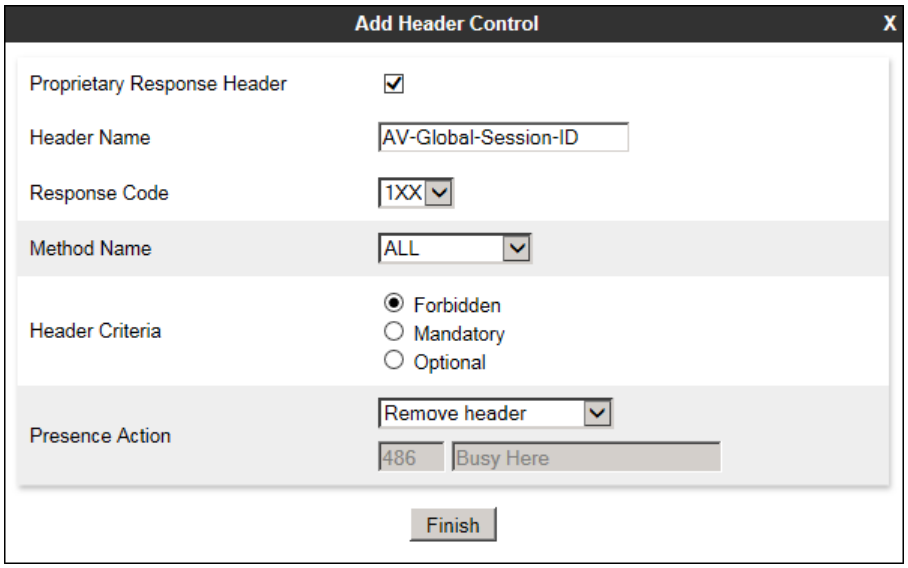


Once the configuration is completed, the **Request Headers** tab should look like the following screen.

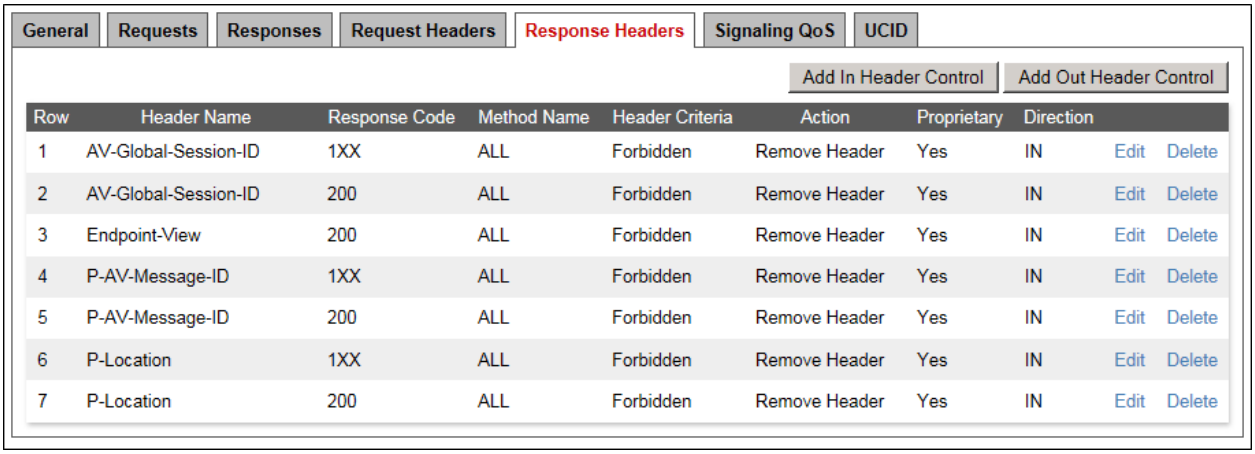


Select the **Response Headers** tab to similarly create the manipulations performed on response messages. Select **Add In Header Control** (not shown).

The screen below shows the settings of one of the entries for the AV-Global-Session-ID header on response messages.



Select **Add In Header Control** as needed to configure the remaining header control rules. Once the configuration is 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	Endpoint-View	200	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
4	P-AV-Message-ID	1XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
5	P-AV-Message-ID	200	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
6	P-Location	1XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
7	P-Location	200	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete

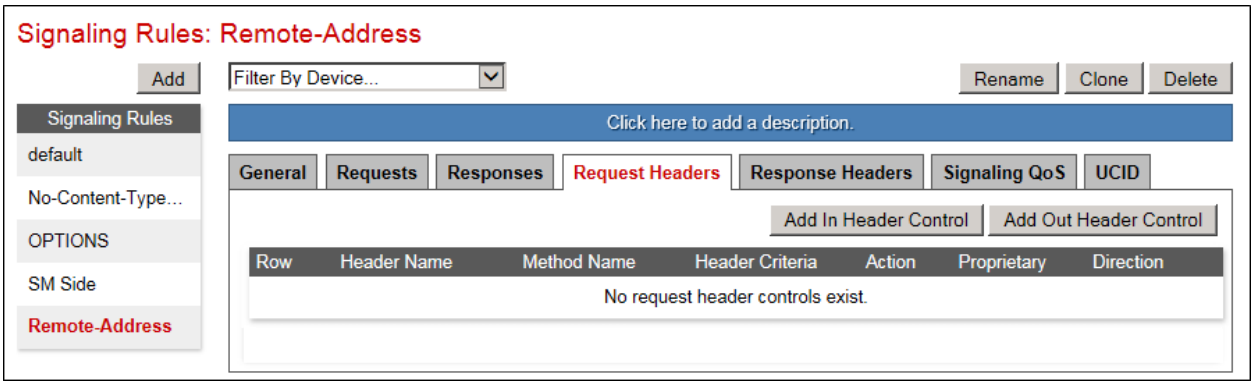
This Signaling Rule will be applied to the End Point Policy Group corresponding to the enterprise, later in **Section 7.12.1**.

7.11.2. Signaling Rule – Service Provider

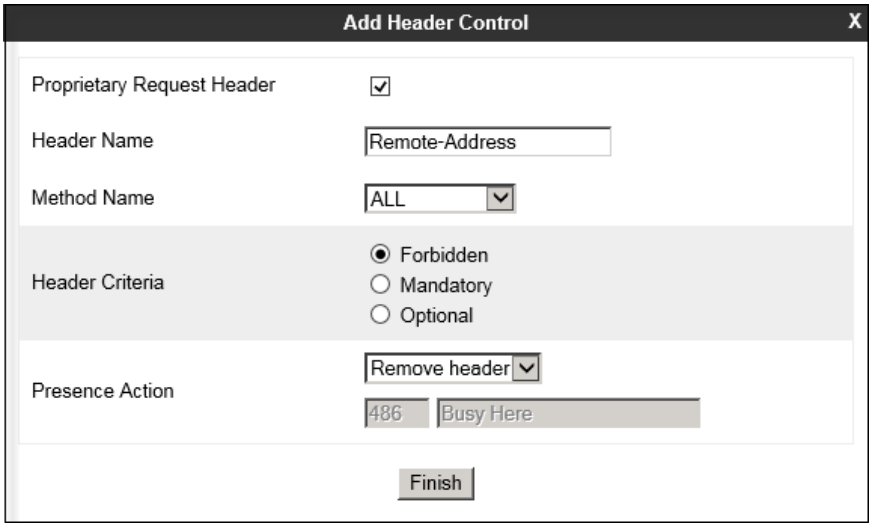
Another Signaling Rule was used in the Avaya SBCE to remove the "Remote-Address" header, generated by the Avaya SBCE, from outbound messages to the service provider. This header has local significance only and should not be propagated on the SIP trunk.

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**.

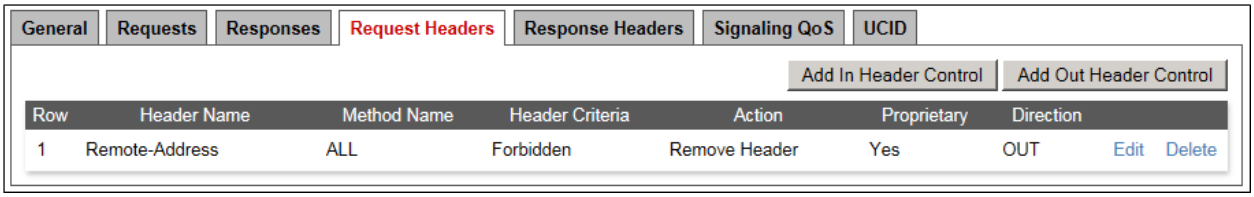
On the newly created **Remote-Address** Signaling Rule, select the **Request Headers** tab to create the manipulations performed on request messages. Select **Add Out Header Control**.



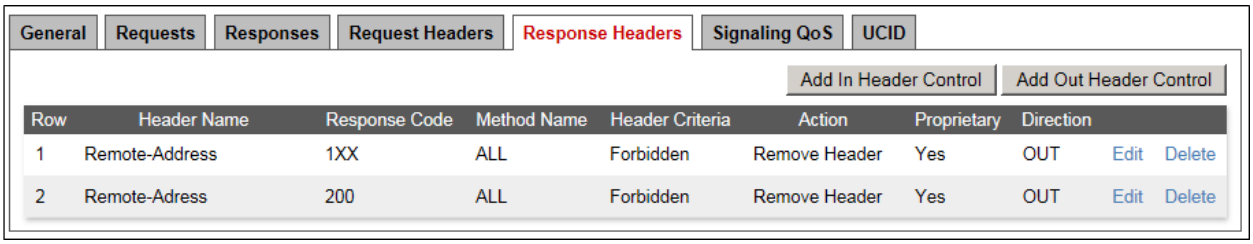
Enter the settings on the **Add Header Control** screen as show below. Click **Finish**.



Once the configuration is completed, the **Request Headers** tab should look like the following screen.



Select the **Response Headers** tab to similarly create the manipulations performed on response messages. Once the configuration is completed, the **Response Headers** tab should look like the following screen.



The Signaling Rule will be applied to the End Point Policy Group corresponding to the service provider, later in **Section 7.12.2**.

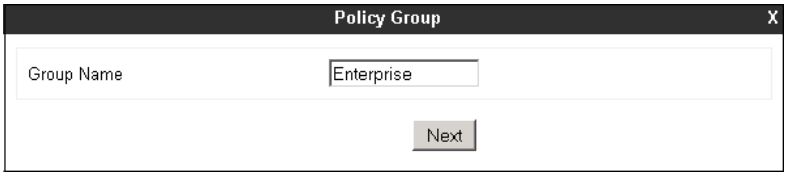
7.12. End Point Policy Groups

End Point Policy Groups associate the different sets of rules under Domain Policies (Media, Signaling, Security, etc.) to be applied to specific SIP messages traversing through the Avaya SBCE. In the reference configuration, the End Point Policy Groups used default sets of rules already pre-defined in the configuration, with the exception of the new Signaling Rules defined in **Section 7.11**. Please note that changes should not be made to any of the default rules used in these End Point Policy Groups. 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.

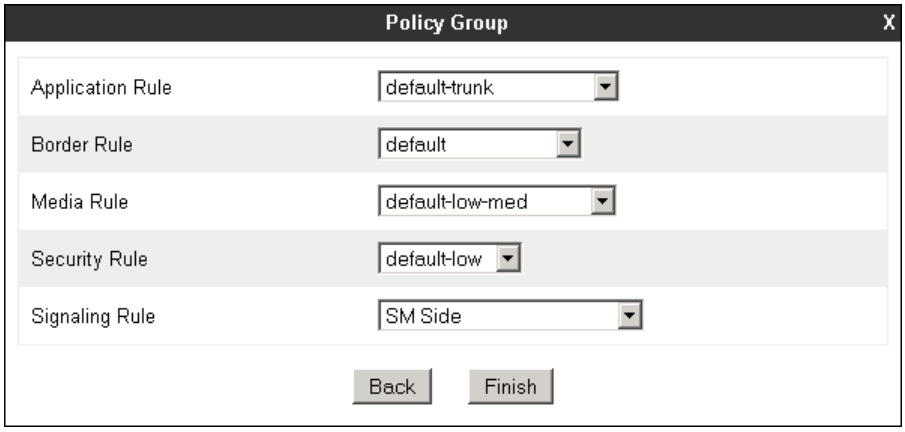
7.12.1. End Point Policy Group – Enterprise

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

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



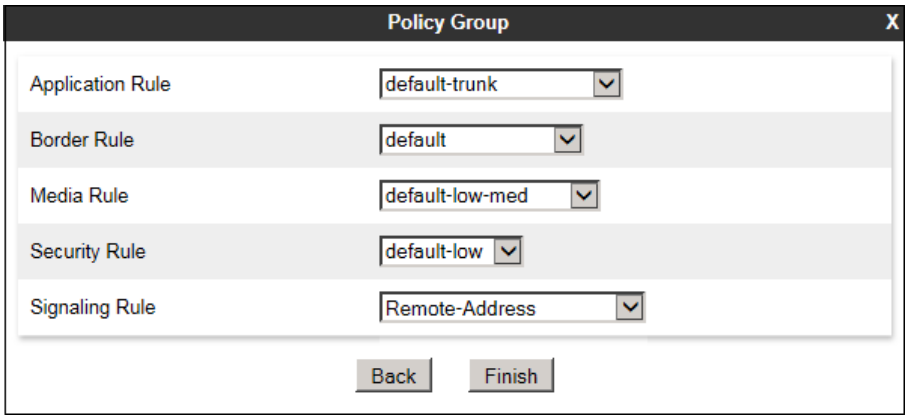
In the Policy Group tab, all fields used one of the default sets already pre-defined in the configuration, with the exception of the **Signaling Rule**, where the ***SM Side*** rule created in **Section 7.11.1** was selected. Click **Finish**.



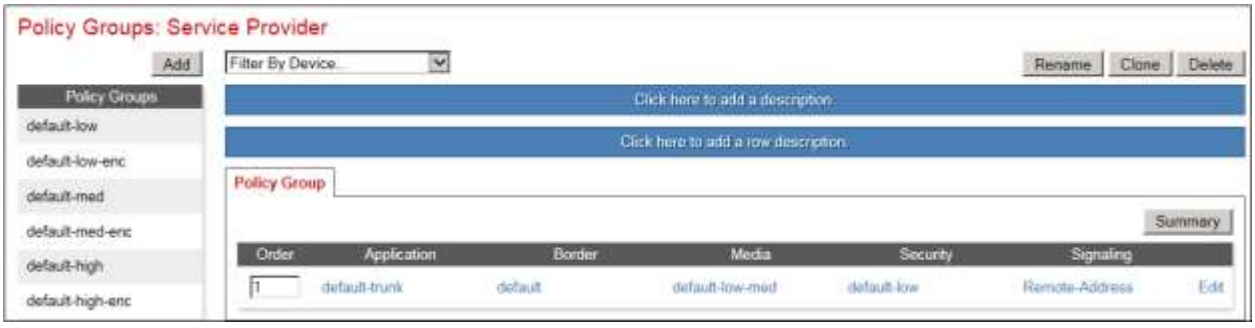
The screen below shows the ***Enterprise*** End Point Policy Group after the configuration was completed.

7.12.2. End Point Policy Group – Service Provider

A second End Point Policy Group was created for the service provider, repeating the steps previously described. In the Policy Group tab, all fields used one of the default sets already pre-defined in the configuration, except on the **Signaling Rule**, where the ***Remote-Address*** rule created in **Section 7.11.2** was selected. Click **Finish**.



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

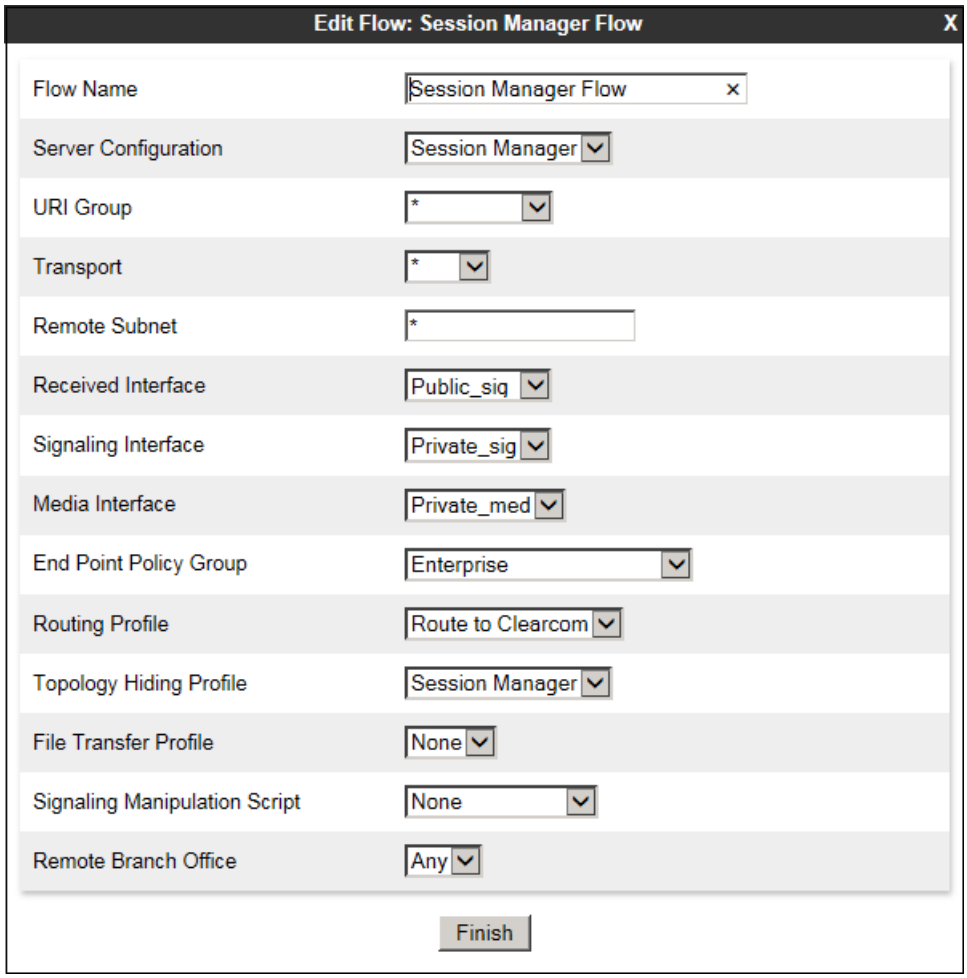


7.13. End Point Flows

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.

7.13.1. End Point Flow – Enterprise

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 that the **Routing Profile** selection is the profile created for the Service Provider in **Section 7.9.2**, which is the reverse route of the flow. Click **Finish**.



7.13.2. End Point Flow – Service Provider

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 that the **Routing Profile** selection is the profile created for Session Manager in **Section 7.9.1**, which is the reverse route of the flow. Also note that there is no selection under the **Signaling Manipulation Script** field. Since the script created in **Section 7.7** was previously applied to the service provider's Server Configuration Profile in **Section 7.8.2**, it is not necessary to make a selection here. Click **Finish**.

8. Clearcom SIP Trunking Configuration

Clearcom is responsible for the configuration of the SIP Trunking service in its network. The customer will need to provide the IP address and port used to reach the Avaya IP Office at the enterprise. Clearcom will provide the customer the necessary information to configure the SIP trunk connection from the enterprise site to the network, including:

- SIP Trunk registration credentials (user name, password, SIP domain).
- Fully Qualified Domain Name of the Clearcom SIP proxy server.
- DID numbers.
- Supported codecs and order of preference.
- Any 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.

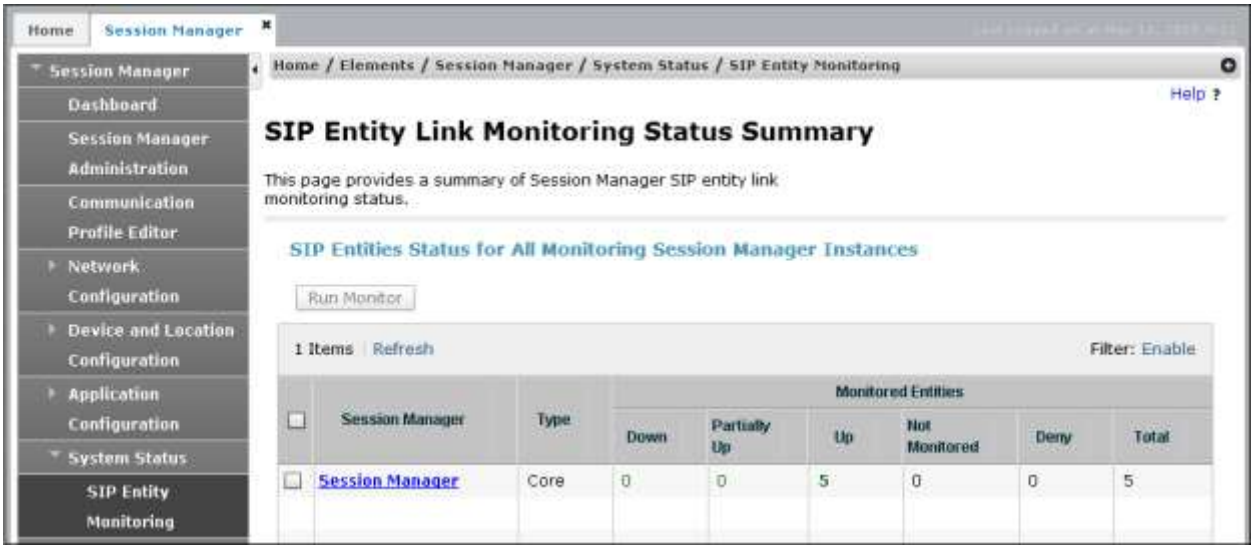
9.2. Communication Manager Verification

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

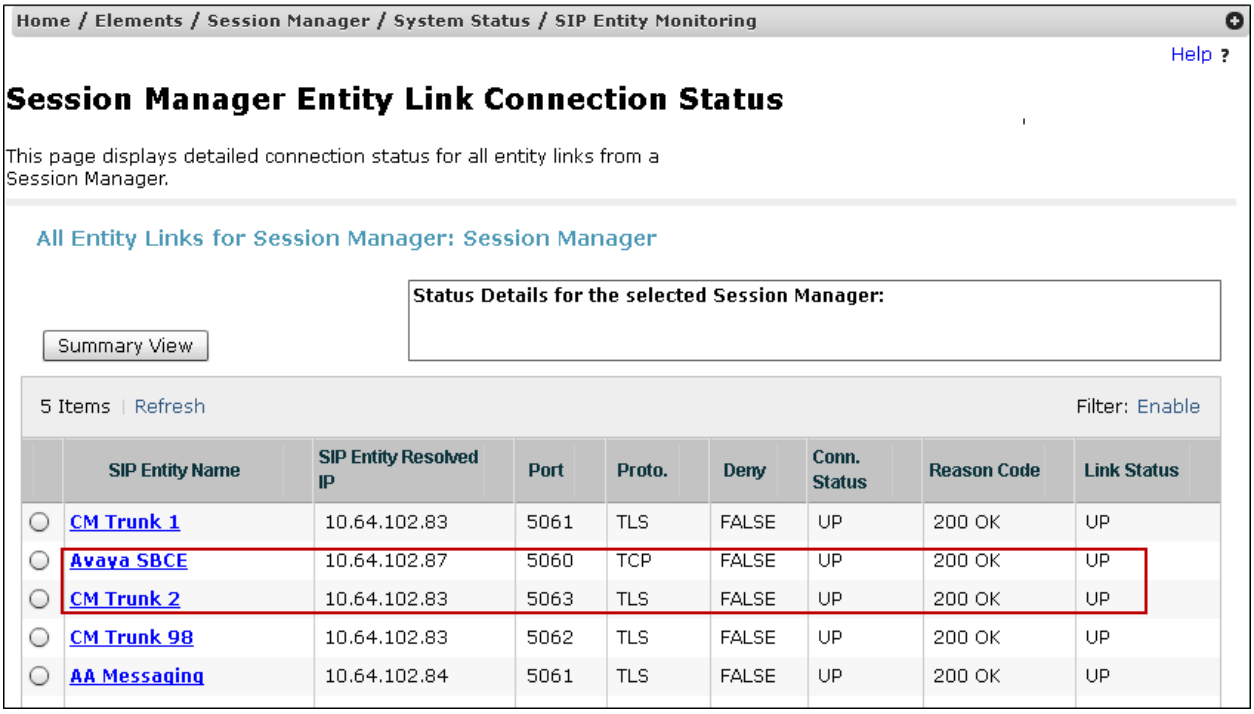
- **list trace station** <extension number>
  Traces calls to and from a specific station.
- **list trace tac** <trunk access code number>
  Trace calls over a specific trunk group.
- **status signaling-group** <signaling group number>
  Displays signaling group service state.
- **status trunk** <trunk 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**. Click the Session Manager instance (***Session Manager*** in the example below).



Verify that the state of the Session Manager links to Communication Manager and the Avaya SBCE under the **Conn. Status** and **Link Status** columns is ***UP***, like shown on the screen below.



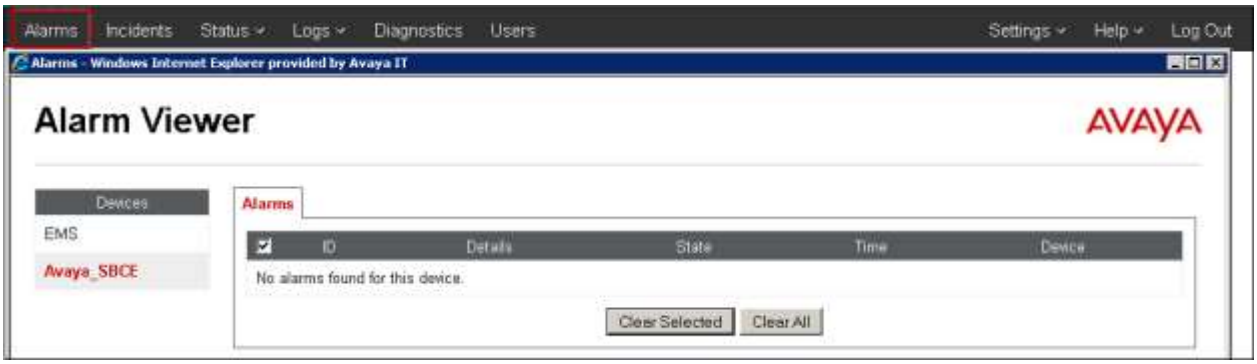
Other Session Manager useful verification and troubleshooting tools include:

- **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.

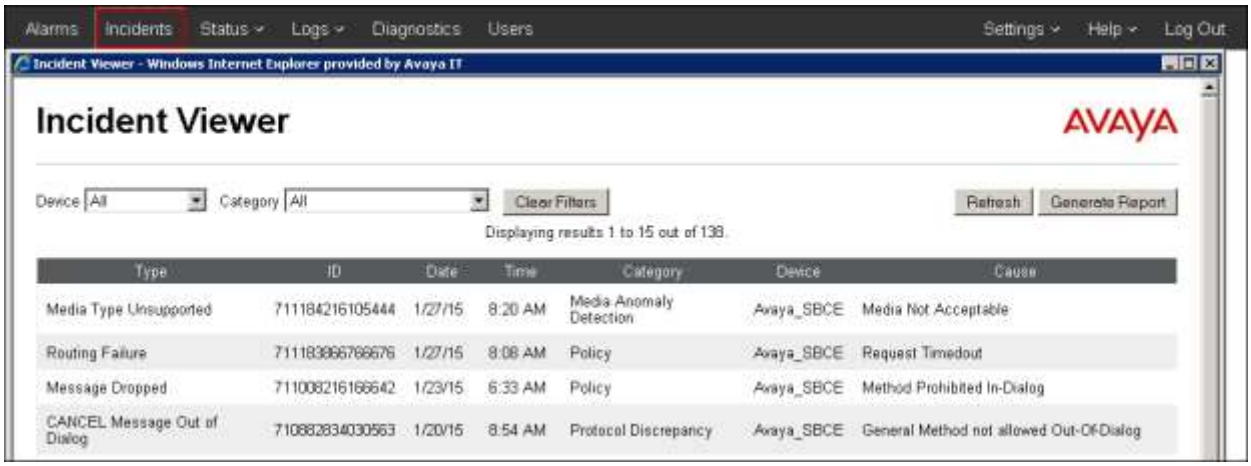
9.4. Avaya SBCE Verification

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.

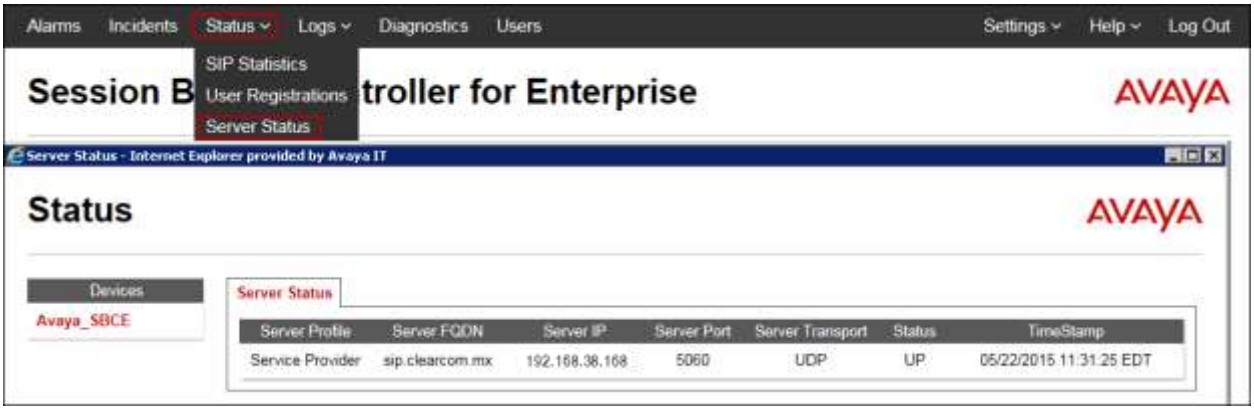
Alarms: This screen provides information about the health of the SBC.



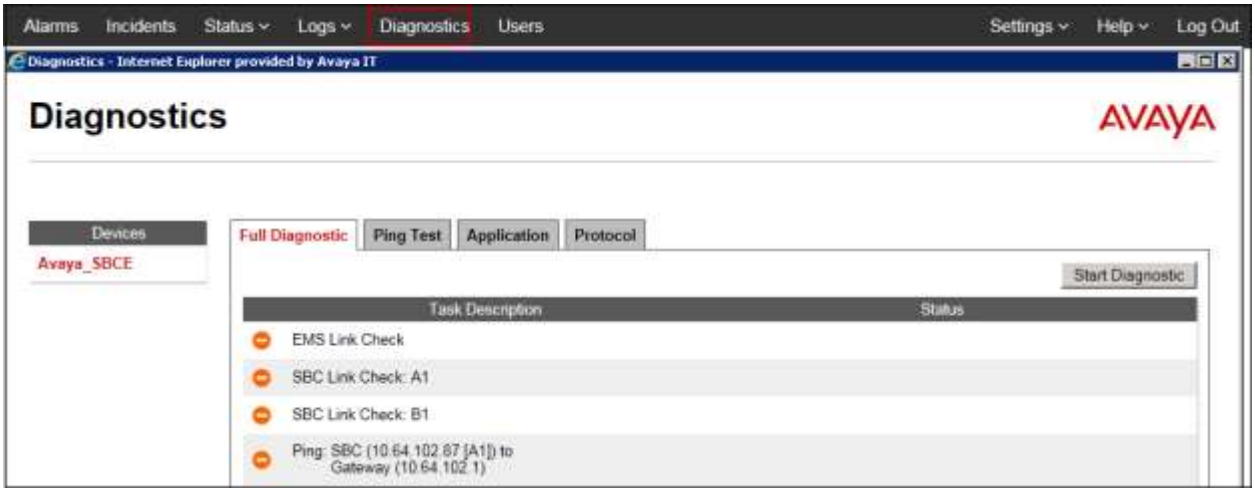
Incidents : This screen provides detailed reports of anomalies, errors, policies violations, etc.



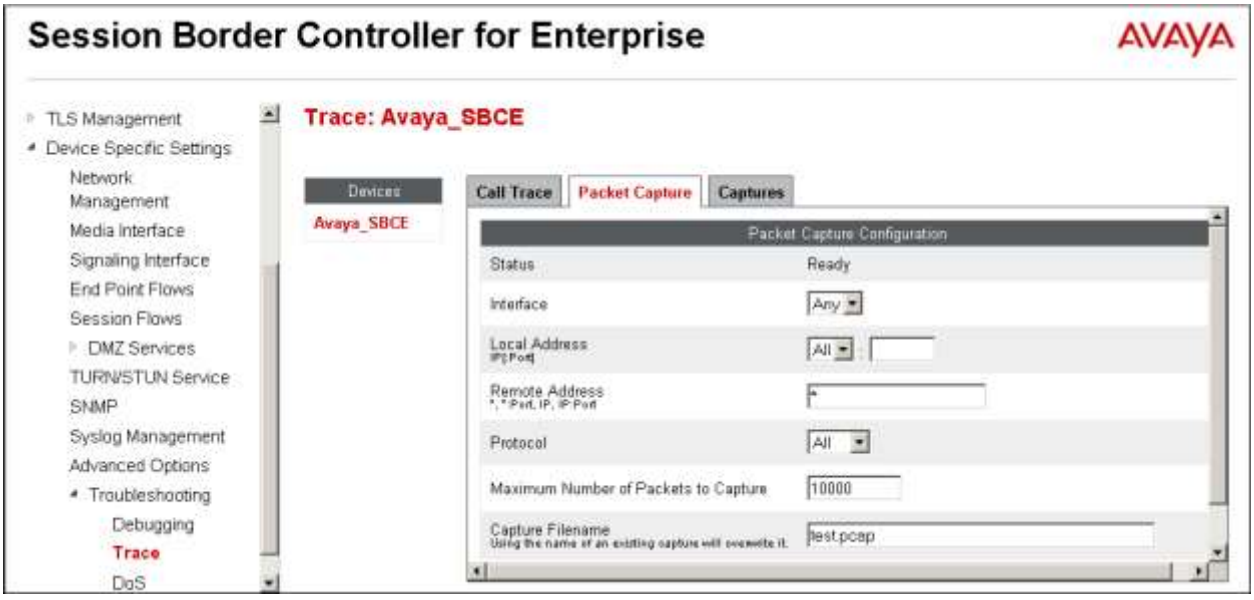
Status: Statistical and current status information. The **Server Status** screen below provides information about the condition of the connection to the Service Provider. This requires Heartbeat to be enabled on the Server Configuration profile, as configured in **Section 7.8.2**.



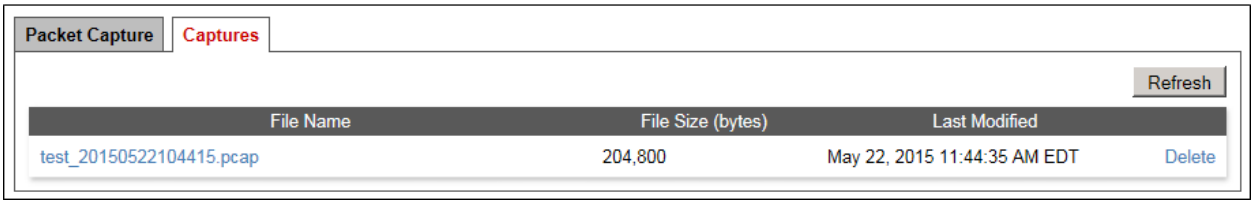
Diagnostics: This screen provides a variety of tools to test and troubleshoot the Avaya SBCE network connectivity.



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 Avaya Aura® Communication Manager 6.3, Avaya Aura® Session Manager 6.3 and Avaya Session Border Controller for Enterprise 6.3, to connect to Clearcom SIP Trunk Services, as shown in **Figure 1**.

Interoperability testing of the sample configuration was completed with successful results for all test cases with 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] *Administering Avaya Aura® Communication Manager*, Release 6.3, June 2014, Document Number 03-300509.
[2] *Avaya Aura® Communication Manager Feature Description and Implementation*, Release 6.3, December 2014, Document Number 555-245-205.
[3] *Administering Avaya Aura® Session Manager*, Release 6.3, September 2014.
[4] *Deploying Avaya Session Border Controller for Enterprise*, Release 6.3, October 2014.
[5] *Administering Avaya Session Border Controller for Enterprise*, Release 6.3, October 2014.
[6] *RFC 3261 SIP: Session Initiation Protocol*, http://www.ietf.org/
[7] *RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, http://www.ietf.org/

12. Appendix A: SigMa Script

The following is the Signaling Manipulation script used in the configuration of the Avaya SBCE, on **Section 7.7**.

```
//Replace Username in "REQUEST-LINE" with "TO" number on Inbound

within session "ALL"
{
   act on message where %DIRECTION="INBOUND" and %ENTRY_POINT="PRE_ROUTING"
   {
      %HEADERS["Request_Line"][1].URI.USER = %HEADERS["To"][1].URI.USER;
   }
}

//Insert Username in the FROM header on Outbound

within session "ALL"
{
    act on request where %DIRECTION="OUTBOUND" and ENTRY_POINT="POST_ROUTING"
    {
      %fromuser = %HEADERS["From"][1].URI.USER;

      %HEADERS["From"][1].URI.USER = "user123";
    }
}

//Remove gsid and epv parameters in outbound Contact header

within session "ALL"
{
   act on message where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"
   {
      remove(%HEADERS["Contact"][1].URI.PARAMS["gsid"]);

      remove(%HEADERS["Contact"][1].URI.PARAMS["epv"]);
   }
}
```

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.