Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager Rel. 7.1, Avaya Aura® Session Manager Rel. 7.1 and Avaya Session Border Controller for Enterprise Rel. 7.2 with CenturyLink SIP Trunking Service on Perimeta/BroadWorks Platform – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking Service on an enterprise solution consisting of Avaya Aura® Communication Manager Rel. 7.1.2, Avaya Aura® Session Manager Rel. 7.1.2 and Avaya Session Border Controller for Enterprise Rel. 7.2.1, to interoperate with the CenturyLink SIP Trunking service on Perimeta/BroadWorks Platform using UDP.

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 is conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

Table of Contents

1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking Service between the CenturyLink Perimeta/BroadWorks network and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Aura® Communication Manager Rel. 7.1.2 (Communication Manager), Avaya Aura® Session Manager Rel. 7.1.2 (Session Manager), Avaya Session Border Controller for Enterprise Rel. 7.2.1 (Avaya SBCE) and various Avaya endpoints, listed in **Section 4**.

Customers using this Avaya SIP-enabled enterprise solution with CenturyLink are able to place and receive PSTN calls via a broadband Internet connection. This converged network solution is an alternative to a traditional PSTN trunk such as analog and/or ISDN-PRI.

The terms "CenturyLink" and "Service Provider" will be used interchangeably throughout these Application Notes.

2. General Test Approach and Test Results

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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

CenturyLink is a member of the Avaya DevConnect Service Provider Program. The general test approach was to connect a simulated enterprise to CenturyLink's Perimeta/BroadWorks network via the Internet and exercise the features and functionalities listed in **Section 2.1**.

2.1. Interoperability Compliance Testing

To verify CenturyLink SIP Trunking service interoperability, the following features and functionalities were covered during the compliance testing:

- SIP Trunk Registration (Dynamic Authentication).
- Response to SIP OPTIONS.

- Inbound PSTN calls to various phone types including H.323, SIP, digital and analog telephone at the enterprise. All inbound calls from PSTN are routed to the enterprise across the SIP trunk from the service provider.
- Outbound PSTN calls from various phone types including H.323, SIP, digital and analog telephone at the enterprise. All outbound calls to the PSTN were routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls to/from Avaya one-X® Communicator (1XC) soft phone. Both the1XC Computer Mode (where 1XC is used for call control as well as audio path) and the 1XC Telecommuter Mode (where 1XC is used for call control and a separate telephone is used for audio path) are tested.
- Dialing plans including local, long distance, international, outbound toll-free, calls etc.
- Calling Party Name presentation and Calling Party Name restriction.
- Codec G.711MU.
- Media and Early Media transmissions.
- T.38 and G.711MU pass-through fax.
- DTMF tone transmissions as out-of-band RTP events as per RFC2833.
- Voicemail navigation for inbound and outbound calls.
- User features such as hold and resume, transfer, forward and conference.
- Off-net call forward with Diversion method.
- EC500 mobility (extension to cellular) with Diversion method.
- Routing inbound vector call to call center agent queues.
- Response to incomplete call attempts and trunk errors.
- Session Timers implementation.

Item that is supported but not tested includes the following:

- Inbound toll-free and emergency E911.

2.2. Test Results

Interoperability testing of CenturyLink SIP Trunking service on Perimeta/BroadWorks Platform with the Avaya SIP-enabled enterprise solution was completed with successful results with the exception of the observations and limitations described below:

- **OPTIONS –** CenturyLink does not send OPTIONS messages to the Avaya enterprise network but it does respond to OPTIONS messages received from the Avaya enterprise, this was sufficient to maintain the SIP trunk link up in service.
- **Informal Contact Header URI –** CenturyLink used long string of characters in the Contact header to hide the actual Contact URI. CenturyLink resolved this issue with configuration changes.
- **Outbound Calls with "+" –** CenturyLink does not accept "+" in front of the 10 digits number in the From, Contact, Diversion or P-Asserted-Identity headers. A signaling manipulation script (SigMa) was used in the Avaya SBCE to remove the "+" sign from the mentioned headers. Refer to **Section 7.3.3**.
- **Network Packets Limitation of 1500 bytes –** CenturyLink network packet size limitation is 1500 bytes. Therefore, it is necessary to reduce the packet size by removing unused headers from Avaya Aura SIP messages. The headers are Record-Route, User-

Agent, Accept-Language and Min-SE. This was accomplished by using a Signaling Manipulation script (SigMa) in the Avaya SBCE. Refer to **Section 7.3.3**.

- **URI in PAI Header should be set to the Pilot Number –** CenturyLink SIP trunking specification requires the URI in the PAI header to be the pilot number. This was accomplished by using a Signaling Manipulation script (SigMa) in the Avaya SBCE. Refer to **Section 7.3.3**.
- **Outbound call to PSTN being challenged twice with 401 –** On outbound calls from Avaya extensions to PSTN phones that are in busy state, CenturyLink would challenge the incoming INVITE two times. This caused the packet size to exceed the 1500 bytes CenturyLink's limitation. CenturyLink resolved this issue with configuration changes.
- **Miss-match Codecs –** CenturyLink did not respond with "488 Not Acceptable Here" error code when calls were made from the enterprise to the PSTN containing codecs not supported by CenturyLink. CenturyLink resolved this issue with configuration changes.
- **Anonymous Call –** Outbound Anonymous call from the enterprise to the PSTN with header Privacy=id was not working properly; the user would hear an announcement saying "Sorry your call can not be completed at this time…." CenturyLink resolved this issue by enabling the alien TN feature on their system to allow anonymous call to work.
- **Extra SIP Signaling –** After a call from the PSTN was effectively transferred off-net to another PSTN party using the SIP REFER method, signaling messages from either side for media shuffling (re-INVITE) or terminating pre-transfer calls between the PSTN and enterprise were observed. These non-recurring messages would receive 481 responses and had no negative impact on the transferred call.
- **Call Forward Off-Net –** On inbound calls from the PTSN to Avaya extension that were forwarded back to the PSTN, three different results were observed: Successful call forward with 2-way voice path, Failed call forward with "604 Does Not Exist" error code and successful call forward with 2-way voice path but with no ring back tone on the originating PSTN phone. CenturyLink resolved this issue with configuration changes.
- **T.38 Fax Support –** CenturyLink supports T.38 fax, but it will not perform fax tone detection. The Avaya system will perform the fax tone detection and will send a re-INVITE for T.38 fax negotiation. T.38 fax was successfully tested on inbound and outbound calls. The G.711 pass-through fax method was also successfully tested on inbound and outbound calls. It should be noted that due to the unpredictability of G.711 pass-through techniques, which only works well on networks with very few hops and with limited end-to-end delay, G.711 fax pass-through is delivered on a "best effort" basis; its success is not guaranteed, and it should be used at the customer's discretion.
- **Outbound T.38 Fax Issue with 405 –** Outbound T.38 fax calls from the enterprise to the PSTN failed with CenturyLink rejecting the call with "405 Method Not Allowed". CenturyLink resolved this issue with configuration changes.
- **Outbound T.38 Fax issue with G.729 –** CenturyLink supports codecs G.729 and G.711MU, for T.38 fax calls, the initial voice call was always set up with codec G.729 instead of G.711MU, this caused the Avaya system to timeout, the Avaya system was unable to detect fax tone from the far end, since the re-INVITE for T.38 fax negotiation was never sent by CenturyLink. The solution for this issue was to configure the Avaya system to only support codec G.711MU. The solution to this issue was approved by

CenturyLink, thus the testing was done only with codec G.711MU configured in the Avaya system. Refer to **Section 5.4**. This issue is being investigated by CenturyLink.

- **Avaya sends SIPS URI in Diversion header –** When TLS/SRTP is used within the enterprise; the SIP headers include the SIPS URI scheme for Secure SIP. The Avaya SBCE converts these header schemes from SIPS to SIP when it sends the SIP message toward CenturyLink. However, for call forward and EC500 calls, the Avaya SBCE was not changing the Diversion header scheme as expected. This anomaly is currently under investigation by the Avaya SBCE development team. The workaround is to include a signaling manipulation script (SigMa) for the CenturyLink Server Configuration profile on the Avaya SBCE to convert "sips" to "sip" in the Diversion header. Refer to **Section 7.3.3**.
- **SIP header optimization –** There are multiple SIP headers and parameters used by Communication Manager and Session Manager, some of them Avaya proprietary, that had no significance in the service provider's network. These headers were removed with the purpose of blocking enterprise information from being propagated outside of the enterprise boundaries, to reduce the size of the packets entering the service provider's network and to improve the solution interoperability in general. Refer to **Section 6.4**.

2.3. Support

For technical support on CenturyLink SIP Trunking service, contact CenturyLink at: http://www.centurylink.com/business/voice/sip-trunk.html

3. Reference Configuration

Figure 1 illustrates the sample Avaya SIP-enabled enterprise solution connected to the Service provider (Vendor Validation circuit) through a public Internet connection.

For security purposes, the real public IP addresses and PSTN routable phone numbers used in the compliance test are not shown in these Application Notes.

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

- Avaya Aura® Communication Manager running in VMware environment.
- Avaya Aura® System Manager running in VMware environment.
- Avaya Aura® Session Manager running in VMware environment.
- Avaya Aura® Messaging running in VMware environment.
- Avaya Aura® Media Server running in VMware environment.
- Avaya G450 Media Gateway.
- Avaya Session Border Controller for Enterprise.
- Avaya 96x1 Series IP Deskphones (H.323 and SIP).
- Avaya one-X® Communicator soft phones (H.323 and SIP).
- Avaya Equinox for Windows (SIP).
- Avaya digital and analog telephones.

Located at the edge of the enterprise network is the Avaya SBCE. It has a public side that connects to Service provider via Internet and a private side that connects to the enterprise network. All SIP and RTP traffic entering or leaving the enterprise network flows through the Avaya SBCE which can protect the enterprise against any outside SIP-based attacks. The Avaya SBCE provides network address translation at both the IP and SIP layers. The transport protocol between the Avaya SBCE and Service provider across the public network is UDP. The transport protocol between the Avaya SBCE, Session Manager and Communication Manager is TLS.

In the compliance test, the Avaya Customer-Premises Equipment (CPE) environment was configured with SIP domain "avayalab.com" for the enterprise. The Avaya SBCE was used to adapt the enterprise IP address to the Domain name known to CenturyLink. **Figure 1** shown below illustrates the network diagram used for the enterprise. All voice application elements are connected to the internal trusted LAN.

In the compliance test, the Avaya SIP trunk on the enterprise side is configured to periodically send OPTIONs messages to the CenturyLink network. SIP trunk registration messages were also sent by the Avaya SBCE to CenturyLink for SIP trunk authentication.

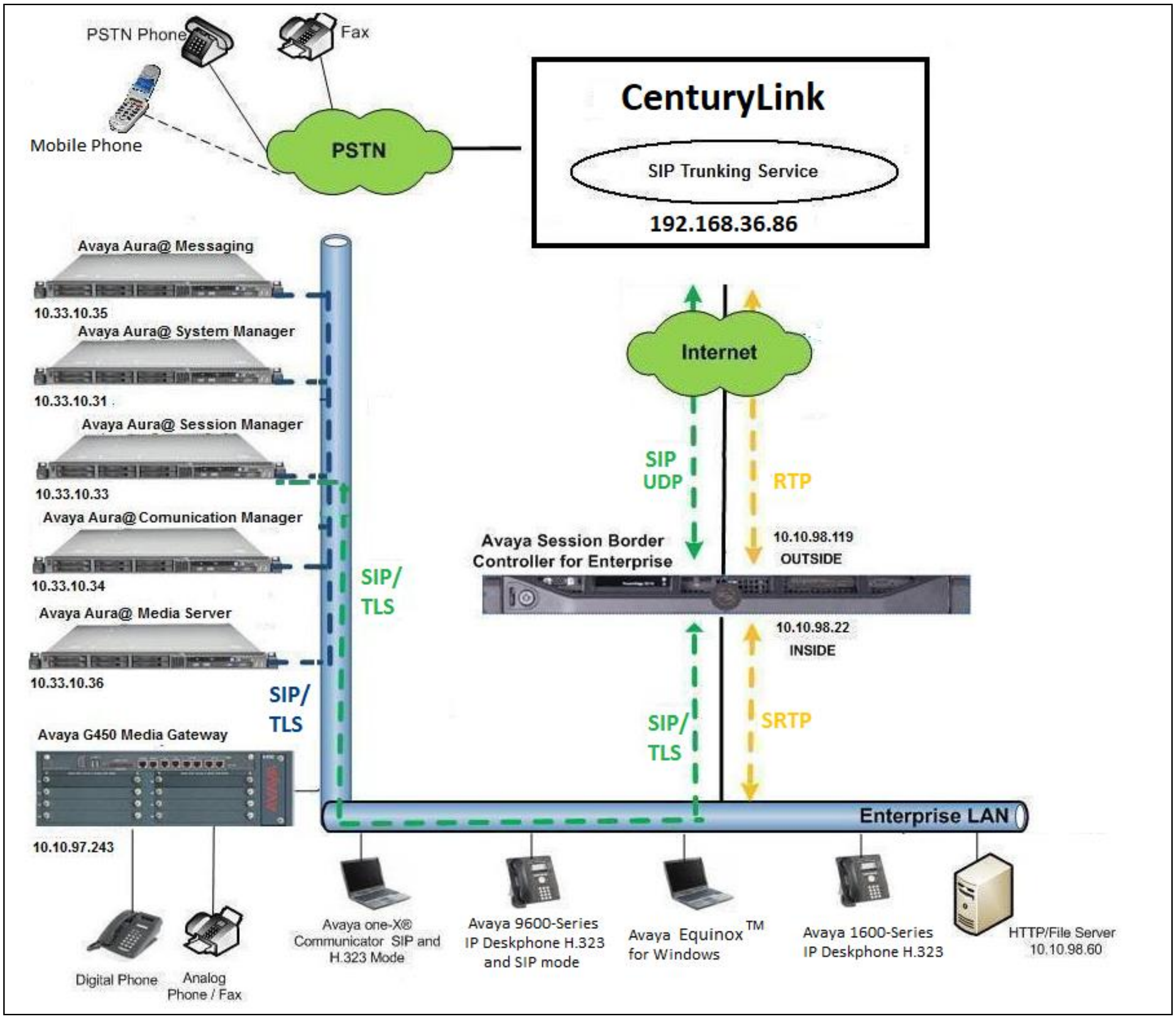


Figure 1: Avaya IP Telephony Network connecting to CenturyLink Networks

4. Equipment and Software Validated

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

Avaya IP Telephony Solution Components	
Component	Release
Avaya Aura® Communication Manager running on a Virtualized Server	7.1.2.0 (R017x.01.0.532.0 Patch 24184)
Avaya G450 Media Gateway	38.21.0
Avaya Aura® System Manager running on a Virtualized Server	7.1.2.0 Build No. - 7.1.0.0.1125193 Software Update Revision No: 7.1.2.0.057353 Feature Pack 2
Avaya Aura® Session Manager running on a Virtualized Server	7.1.2.0.712004
Avaya Aura® Messaging running in a Virtualized Server	7.0.441.017-1.262404
Avaya Aura® Media Server running on a Virtualized Server	7.8.0.333_2017.07.17
Avaya Session Border Controller for Enterprise	7.2.1-05-14222
Avaya 9621G IP Deskphone (H.323)	6.6.401
Avaya 9641G IP Deskphone (SIP)	7.1.1.0.9
Avaya one-X Communicator (H.323/SIP)	6.2.12.04-SP12
Avaya Equinox for Windows	3.2.2.2
Avaya 1608 IP Deskphone (H.323)	1.380B
Avaya 1408 Digital Telephone	1408D02A-003
Avaya Analog Telephone	n/a
CenturyLink SIP Trunking Service Components	
Component	Release
BroadSoft BroadWorks	R21.SP1
Metaswitch Perimeta SBC	V4.1.40_SU15_P01.02

Table 1: Equipment and Software Tested

Note: 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 the CenturyLink SIP Trunking service. It is assumed the general installation of Communication Manager, Avaya G450 Media Gateway and Media Server has been previously completed and is not discussed here.

The configuration of Communication Manager 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 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 sale representative to add the additional capacity or feature.

```
display system-parameters customer-options                      Page   2 of  12
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                             USED
                     Maximum Administered H.323 Trunks: 4000  0
           Maximum Concurrently Registered IP Stations: 2400  1
             Maximum Administered Remote Office Trunks: 4000  0
Maximum Concurrently Registered Remote Office Stations: 2400  0
              Maximum Concurrently Registered IP eCons: 68    0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 2400  0
                   Maximum Video Capable IP Softphones: 2400  3
                       Maximum Administered SIP Trunks: 4000  12
  Maximum Administered Ad-hoc Video Conferencing Ports: 4000  0
   Maximum Number of DS1 Boards with Echo Cancellation: 80    0


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

On page 5 verify **Media Encryption Over IP** is set to ***y***.

```
display system-parameters customer-options                      Page   5 of  12
                                OPTIONAL FEATURES

   Emergency Access to Attendant? y                                IP Stations? y
           Enable 'dadmin' Login? y
           Enhanced Conferencing? y                          ISDN Feature Plus? n
                  Enhanced EC500? y        ISDN/SIP Network Call Redirection? y
    Enterprise Survivable Server? n                            ISDN-BRI Trunks? y
       Enterprise Wide Licensing? n                                   ISDN-PRI? y
              ESS Administration? y                 Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                       Malicious Call Trace? y
     External Device Alarm Admin? y
  Five Port Networks Max Per MCC? n     Mode Code for Centralized Voice Mail? n
                Flexible Billing? n                 Media Encryption Over IP? y
   Forced Entry of Account Codes? y                 Multifrequency Signaling? y
      Global Call Classification? y          Multimedia Call Handling (Basic)? y
             Hospitality (Basic)? y       Multimedia Call Handling (Enhanced)? y
 Hospitality (G3V3 Enhancements)? y                Multimedia IP SIP Trunking? y
                       IP Trunks? y

           IP Attendant Consoles? y
        (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 an incoming call 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  19
                            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
```

On **Page 9**, verify that a text string has been defined to replace the Calling Party Number (CPN) for restricted or unavailable calls. The compliance test used the value of ***Restricted*** for restricted calls and ***Unavailable*** for unavailable calls.

```
change system-parameters features                              Page   9 of  19
                        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: 1
          International Access Code: 001

SCCAN PARAMETERS
   Enable Enbloc Dialing without ARS FAC? n

CALLER ID ON CALL WAITING PARAMETERS
     Caller ID on Call Waiting Delay Timer (msec): 200
```

5.3. IP Node Names

Use the **change node-names ip** command to verify that node names have been previously defined for the IP addresses of Communication Manager **(procr**), Session Manager (**SM**) and Media Server (**AMS**). These node names will be needed for defining the service provider signaling groups in **Section 5.6**.

```
change node-names ip                                              Page   1 of   2
                                  IP NODE NAMES
    Name                  IP Address
SM                      10.33.10.33
AMS                     10.33.10.36
default                 0.0.0.0
procr                   10.33.10.34
procr6                  ::
```

5.4. Codecs

Use the **change ip-codec-set** command to define a list of codecs to be used for calls between the enterprise and the service provider. The compliance test used ip-codec-set 1. CenturyLink supports codecs G.711MU and G.729, for the compliance test only codec G.711MU was configured, refer to **Section 2.2**. Enter ***G.711MU*** for **Audio Codec**. For **media encryption** within the Avaya system, ***1-srtp-aescm128-hmac80***, ***2-srtp-aescm128-hmac32*** *and* ***none*** were used, and ***best-effort*** in **Encrypted SRTCP** columns of the table.

The following screen shows the configuration for ip-codec-set 1.

```
change ip-codec-set 1                                          Page   1 of   2
                          IP CODEC SET
    Codec Set: 1

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

     Media Encryption                   Encrypted SRTCP: best-effort
 1: 1-srtp-aescm128-hmac80
 2: 2-srtp-aescm128-hmac32
 3: none
```

On **Page 2**, set the **Fax Mode** to ***t.38-standard***, CenturyLink supports T.38 fax.

```
change ip-codec-set 1                                       Page   2 of   2
                          IP CODEC SET

                              Allow Direct-IP Multimedia? n
                                                                    Packet
                          Mode                  Redundancy          Size(ms)
    FAX                   t.38-standard          1
    Modem                 off                    0
    TDD/TTY               US                     3
    H.323 Clear-channel   n                      0
    SIP 64K Data          n                      0                  20
```

5.5. IP Network Region

For the compliance test, ip-network-region 1 was created by the **change ip-network-region 1** command with the following parameters:

- Set the **Authoritative Domain** field to match the SIP domain of the enterprise. In the compliance test, the domain name is ***avayalab.com***. This domain name appears in the "From" header of SIP message originating from this IP region.
- Enter a descriptive name in the **Name** field.
- Enable **IP-IP Direct Audio** (shuffling) to allow audio traffic to be sent directly between IP endpoints without using media resources in the Media Gateway. By default, both **Intra-region** and **Inter-region IP-IP Direct Audio** are set to ***yes***. Shuffling can be further restricted at the trunk level under the Signaling Group form.
- Set the **Codec Set** field to the IP codec set defined in **Section 5.4**.
- Default values can be used for all other fields.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avayalab.com
    Name: ToSM
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
...
```

On **Page 4**, define the IP codec set to be used for traffic between region 1 and other regions. In the compliance test, Communication Manager, the Avaya G450 Media Gateway, IP/SIP phones and Session Manager were assigned to the same region 1.

```
change ip-network-region 1                                      Page   4 of  20

 Source Region: 1     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    1                                                               all
 2    1      y      NoLimit                                         n       t
 3                                                                  n       t
```

Non-IP telephones (e.g., analog, digital) derive network region from IP interface of the Avaya G450 Media Gateway to which the device is connected. IP telephones can be assigned a network region based on an IP address mapping.

To define network region 1 for IP interface **procr**, use **change ip-interface procr** command as shown in the following screen.

```
change ip-interface proc                                     Page   1 of   2
                                 IP INTERFACES


                  Type: PROCR
                                                   Target socket load: 4800

      Enable Interface? y                       Allow H.323 Endpoints? y
                                                 Allow H.248 Gateways? y
        Network Region: 1                         Gatekeeper Priority: 5
...
```

To define network region 1 for the Avaya G450 Media Gateway, use **change media-gateway** command as shown in the following screen.

```
change media-gateway 1                                     Page   1 of   2
                            MEDIA GATEWAY 1

                    Type: g450
                    Name: g450
               Serial No: 11N526797797
    Link Encryption Type: any-ptls/tls       Enable CF? n
          Network Region: 1                   Location: 1
                                             Site Data:
           Recovery Rule: none
...
```

If Avaya Aura® Media Server is used in place of Avaya Media Gateway G450 then network region 1 needs to be defined for the Avaya Aura® Media Server, use **change media-server** command as shown in the following screen.

```
change media-server 1                                      Page   1 of   1
                                 MEDIA SERVER

                    Media Server ID: 1

                    Signaling Group: 3
          Voip Channel License Limit: 30
     Dedicated Voip Channel Licenses: 30

                          Node Name: AMS
                     Network Region: 1
                           Location: 1
          Announcement Storage Area:
...
```

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 for this purpose and was configured using the parameters highlighted below.

- Set the **Group Type** field to ***sip***.
- Set the **IMS Enabled** field to ***n***.
- Set the **Transport Method** to ***tls***. The transport method specified here is used between Communication Manager and Session Manager.
- Set the **Near-end Listen Port** and **Far-end Listen Port** to ***5061***.
- Set the **Near-end Node Name** to ***procr***. This node name maps to the IP interface of **procr** defined in **Section 5.3**.
- Set the **Far-end Node Name** to ***SM***. This node name maps to the IP address of Avaya SBCE as defined in **Section 5.3**.
- Set the **Far-end Network Region** to the IP network region ***1*** defined for the service provider in **Section 5.5**.
- Set the **Far-end Domain** to ***avayalab.com***.
- Set the **DTMF over IP** to ***rtp-payload***. This setting enables Communication Manager to send or receive the DTMF transmissions using RFC2833.
- Set **Enable Layer 3 Test?** to ***y***. This setting allows Communication Manager to send OPTIONS heartbeat to Session Manager on the SIP trunk.
- 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 SIP trunk and the enterprise endpoint. If this value is set to ***n***, then the Avaya G450 Media Gateway will remain in the media path between the SIP trunk and the endpoint for the duration of the call. Depending on the number of media resources available in the Avaya G450 Media Gateway, these resources may be depleted during high call volume preventing additional calls from completing.
- Set the **Alternate Route Timer** to ***30***. This defines the number of seconds Communication Manager will wait for a response (other than 100 Trying) to an outbound INVITE before canceling the call.
- Default values may be used for all other fields.

Signaling Group 2:

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

 Group Number: 2             Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                 Far-end Node Name: SM
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1

Far-end Domain: avayalab.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                   Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n          Alternate Route Timer(sec): 30
```

Another signaling group is created between Communication Manager and Media Server to provide media resource for IP telephony in replacement/absent of media gateway G450. For the compliance test, signaling group ***3*** was used for this purpose and was configured in the capture shown below.

Signaling Group 3:

```
add signaling-group 3                                       Page   1 of   2
                                SIGNALING GROUP

 Group Number: 3             Group Type: sip
                        Transport Method: tls


  Peer Detection Enabled? n  Peer Server: AMS



   Near-end Node Name: procr                 Far-end Node Name: AMS
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1

Far-end Domain: 10.33.10.36
```

5.7. Trunk Group

Use the **add trunk-group** command to create a trunk group for the signaling group created in **Section 0**. 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 **Outgoing Display** to ***y*** to enable name display on the trunk.
- Set the **Service Type** field to ***public-ntwrk***.
- Set the **Signaling Group** to the signaling group **2** shown in **Section 0**.
- Set the **Number of Members** field to customer requirement. It is the number of trunk members in the SIP trunk group. This value determines how many simultaneous SIP calls can be supported by this trunk group.
- Default values are 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-Carrier                COR: 1        TN: 1        TAC: #02
   Direction: two-way       Outgoing Display? y
 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: 32
```

On **Page 2**, verify that the **Preferred Minimum Session Refresh Interval (sec)** is set to a value acceptable to the service provider. This value defines the interval a re-INVITEs must be sent to refresh the Session Timer. For the compliance testing, a 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

             XOIP Treatment: auto    Delay Call Setup When Accessed Via IGAR? N


Caller ID for Service Link Call to H.323 1xC: station-extension
```

On **Page 3**, set the **Numbering Format** field to ***public***. This field specifies the format of the CPN sent to the far-end. The public numbers are automatically preceded with a + sign when passed in the "From", "Contact" and "P-Asserted Identity" headers.

Set the **Replace Restricted Numbers** and **Replace Unavailable Numbers** fields to ***y***. This will allow the CPN displayed on the local endpoint to be replaced with the value set in **Section 5.2**, if the inbound call enabled CPN block. Default values are used for all other fields.

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

                     Numbering Format: public
                                                  UUI Treatment: service-provider

                                                   Replace Restricted Numbers? y
                                                  Replace Unavailable Numbers? Y
                                                    Hold/Unhold Notifications? y
                                Modify Tandem Calling Number: no

Show ANSWERED BY on Display? y
```

On **Page 4**, the settings are as follow:

- Set of **Network Call Redirection** flag to ***y*** to enable the use of SIP REFER message to transfer calls back to the PSTN. It can also be set to ***n*** for the use of re-INVITE.
- Set the **Send Diversion Header** field to ***y***. This is needed to support call forwarding of inbound call back to PSTN and Extension to Cellular (EC500) call scenarios.
- Set the **Support Request History** field to ***n***.
- Set the **Telephone Event Payload Type** to ***101***.

```
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? Y
          Build Refer-To URI of REFER From Contact For NCR? n
                                     Send Diversion Header? y
                                   Support Request History? n
                              Telephone Event Payload Type: 101


                        Convert 180 to 183 for Early Media? n
                  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
                                Request URI Contents: may-have-extra-digits
```

5.8. Calling Party Information

The calling party number is sent in the SIP “From”, “Contact” and “PAI” headers. Since public numbering is selected to define the format of this number (**Section 0**), use the **change public-unknown-numbering** command to create an entry for each extension which has a DID assigned. DID numbers are provided by the service provider. They are used to authenticate the caller.

The screen below shows a subset of the 10-digit DID numbers assigned for testing. These 4 numbers were mapped to the enterprise extensions 60396, 60397 and 60379. These same 10-digit numbers were used in the outbound calling party information on the service provider trunk when calls were originated from these extensions.

Note: When using 10-digit CPN the + will need to be removed from the SIP message by the Avaya SBCE.

```
change public-unknown-numbering 0                                 Page   1 of   2
                           NUMBERING - PUBLIC/UNKNOWN FORMAT

Ext Ext               Trk        CPN              Total
Len Code              Grp(s)     Prefix           Len
 5  60396             2         3036335744       10      Total Administered: 3
 5  60397             2         3036335747       10         Maximum Entries: 240
 5  60379             2         3036335748       10
```

5.9. Incoming Call Handling Treatment

In general, the incoming call handling treatment for a trunk group can be used to manipulate the digits received for an incoming call if necessary. DID number sent by the service provider can be mapped to an extension using the incoming call handling treatment of the receiving trunk-group. Use the **change inc-call-handling-trmt trunk-group** command to create an entry for each DID.

```
change inc-call-handling-trmt trunk-group 50                  Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 public-ntwrk    10 3036335744       10  60396
 public-ntwrk    10 3036335747       10  60397
 public-ntwrk    10 3036335748       10  60379
……
```

5.10. Outbound Routing

In these Application Notes, the **Automatic Route Selection** (ARS) feature is used to route an outbound call via the SIP trunk to the service provider via the Avaya SBCE. In the compliance test, a single digit 9 was used as the ARS access code. An enterprise caller will dial 9 to reach an outside line. To define feature access code (**fac**) ***9***, use the **change dialplan analysis** command as shown below.

```
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
   1           4   ext
   30          4   ext
   39          5   udp
   60          5   ext
   9           1   fac
   *           3   dac
   #           3   dac
```

Use the **change feature-access-codes** command to define ***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:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code: *05
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code:
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
```

Use the **change ars analysis** command to configure the routing of dialed digits following the first digit 9. The example patterns below show a subset of the dialed strings tested as part of the compliance test. All dialed strings are mapped to route pattern ***2*** for an outbound call which contains the SIP trunk to the service provider (as defined next).

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

          Dialed           Total      Route     Call   Node   ANI
          String          Min  Max   Pattern    Type   Num    Reqd
    0                     1    11     2         op            n
    011                   10   18     2         intl          n
    1                     11   11     2         pubu          n
    411                   3    3      2         svcl          n
    613                   10   10     2         pubu          n
    303                   10   10     2         pubu          n
    866                   10   10     2         pubu          n
    911                   3    3      2         svcl          n
```

As mentioned above, the route pattern defines which trunk group will be used for the outbound calls and performs necessary digit manipulation. Use the **change route-pattern** command to configure the parameters for route pattern ***2*** in the following manner.

- **Pattern Name**: Enter a descriptive name.
- **Grp No**: Enter the outbound trunk group for the SIP service provider. For the compliance testing, trunk group ***2*** was used.
- **FRL**: Set the Facility Restriction Level (**FRL**) field to a level that allows access to this trunk for all users that require it. The value of ***0*** is the least restrictive level.
- **Numbering Format**: ***pub-unk***, all calls using this route pattern will use the public numbering table. See setting of the **Numbering Format** in the trunk group form for full details in **Section 5.8**.

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

5.11. Saving Communication Manager Configuration Changes

The command "**save translation all**" can be used to save the configuration changes made on Communication Manager.

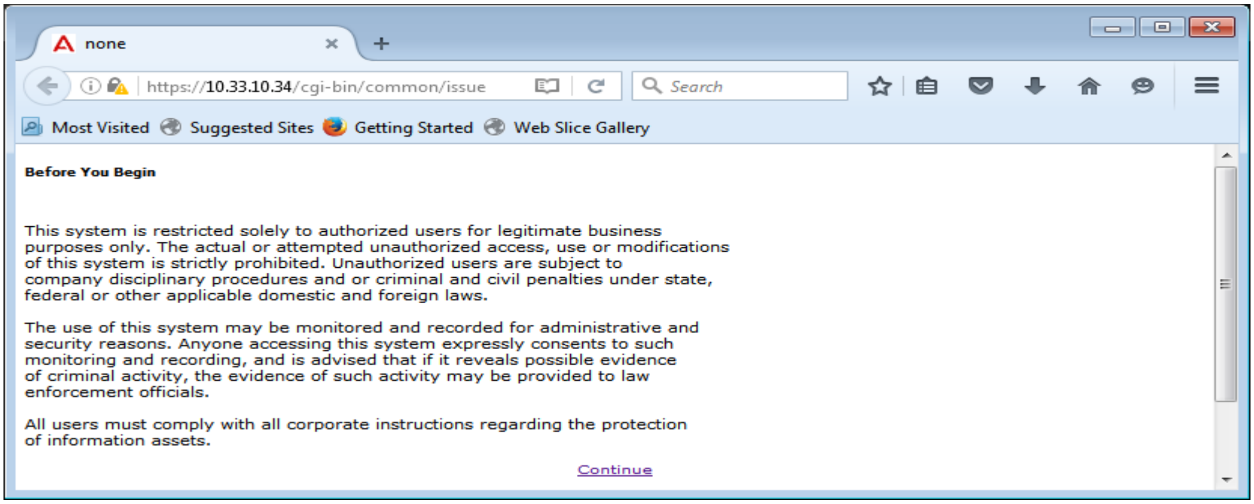
5.12. TLS Management on Communication Manager

It is (or may be) necessary to install System Manager Certificate Authority (CA) certificate on Communication Manager for the TLS signaling to work between Session Manager and Avaya Communication Manager if it is not previously installed.

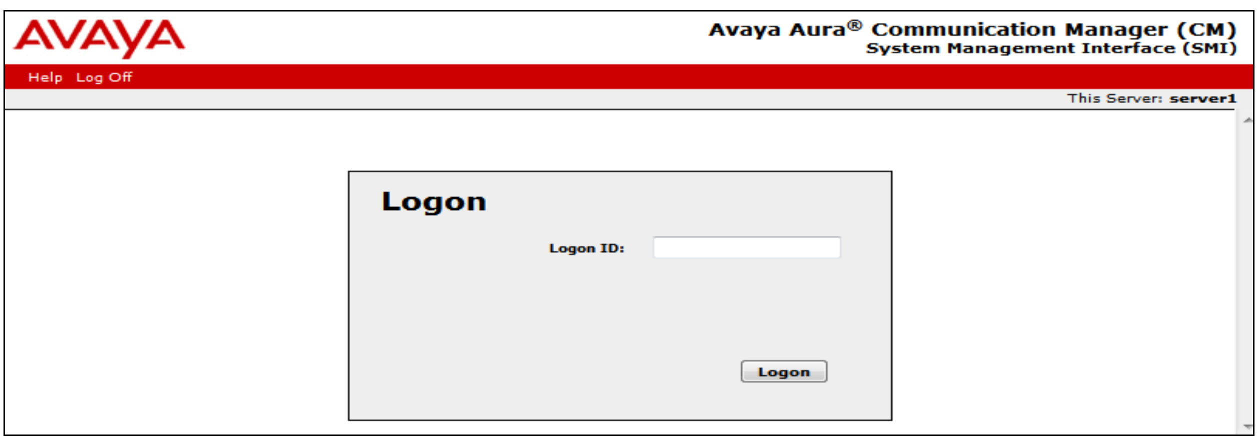
This section is to show how to install System Manager CA certificate on Communication Manager using web console.

System Manager CA certificate is obtained using procedure provided in **Section 6.9**.

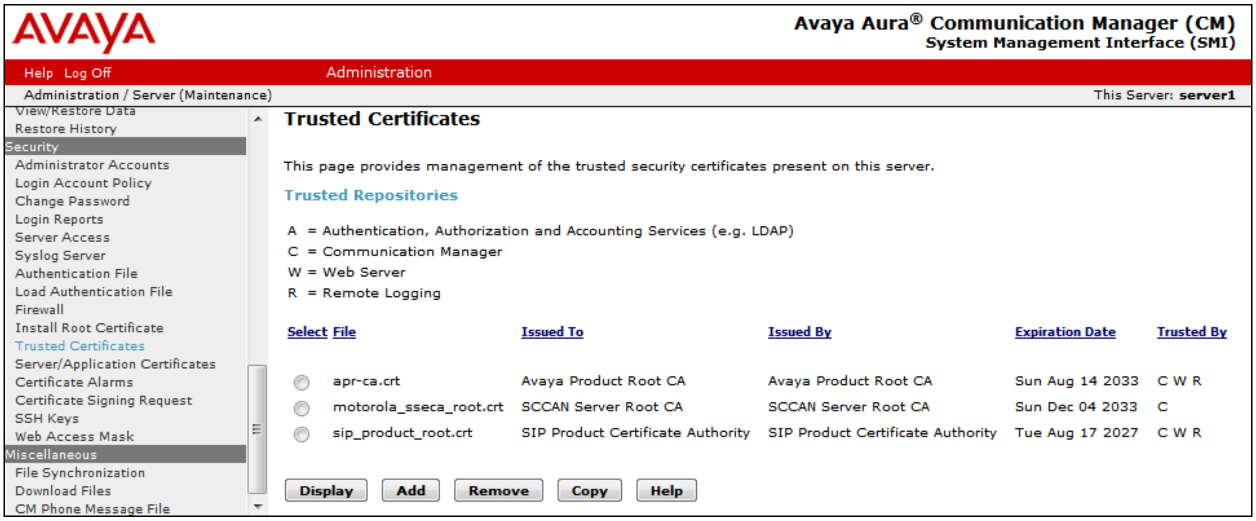
From a web browser, type in "https://<ip-address>", where "<ip-address>" is the IP address or FQDN of Communication Manager. Click on **Continue** and it will be redirect to login page.



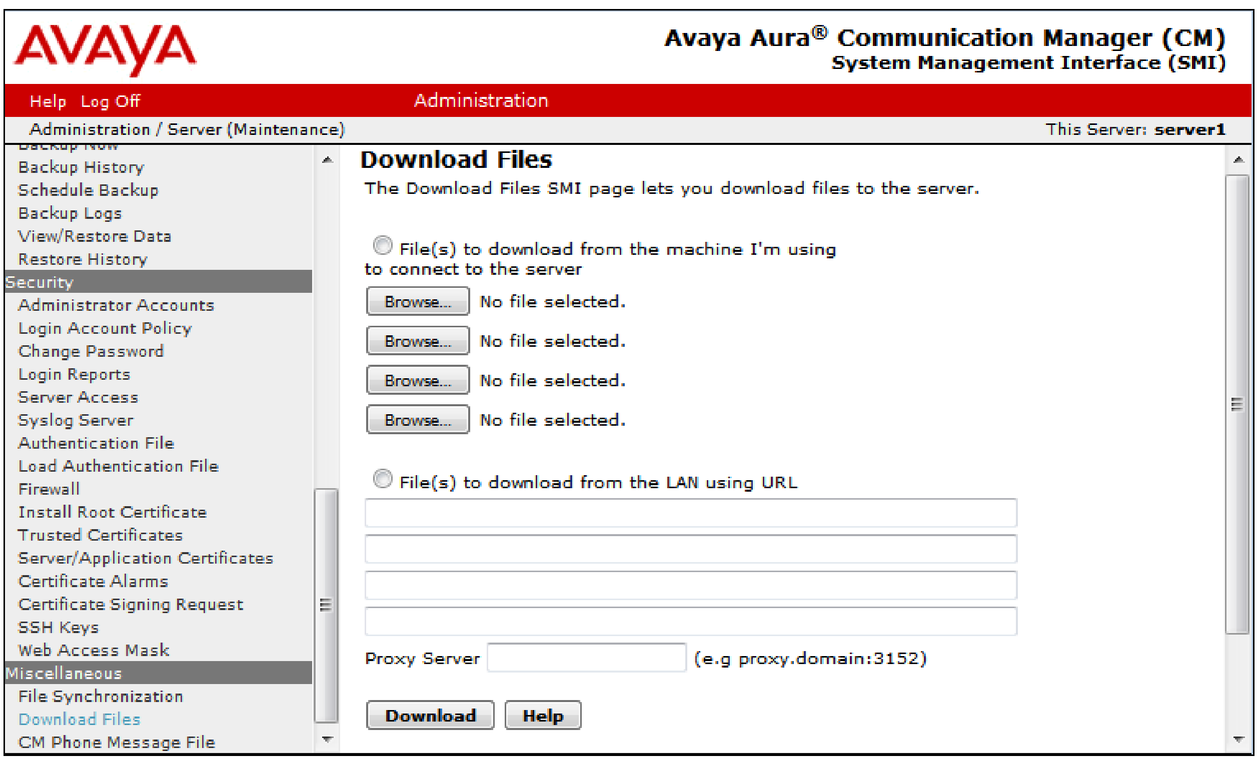
At login page, type in the login ID and its password credential.



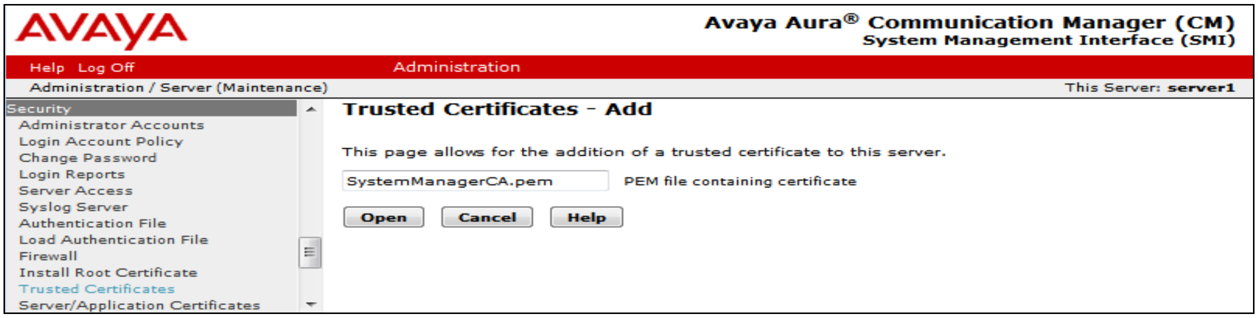
Click on **Continue** again (not shown), navigate to **Administration → Server (Maintenance) → Security → Trusted Certificates** to verify if the System Manager CA certificate is present or not. If it is not, then continue to the next step.



Navigate to **Miscellaneous → Download Files**, click on **File to download from the machine I'm using to connect to the server** and click on **Browse** to browse to where the System Manager CA certificate is located. Then click on **Download** button to load the System Manager CA certificate on Communication Manager server.

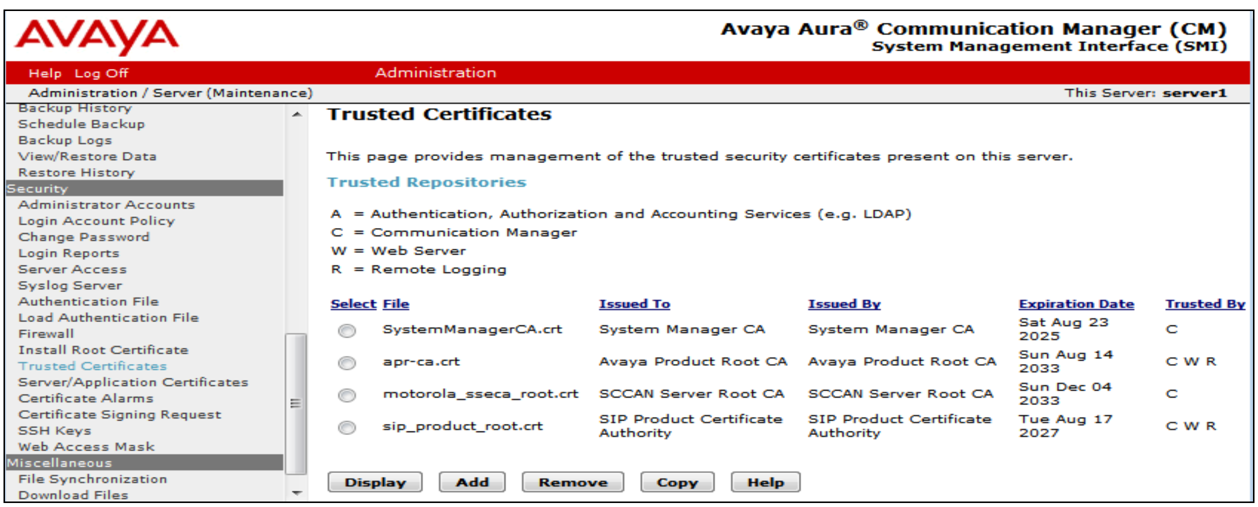


Navigate to **Security → Trusted Certificates**, click on **Add** button and enter the certificate name which has been downloaded from above step. Then click **Open**.



Enter the name of the System Manager CA certificate to store the certificate in Communication Manger. Check the Communication Manager check-box. Then click **Add**.

Navigate to **Security → Trusted Certificates** again. It now shows the System Manager CA certificate in the **Trusted Repositories**.



6. Configure Avaya Aura® Session Manager

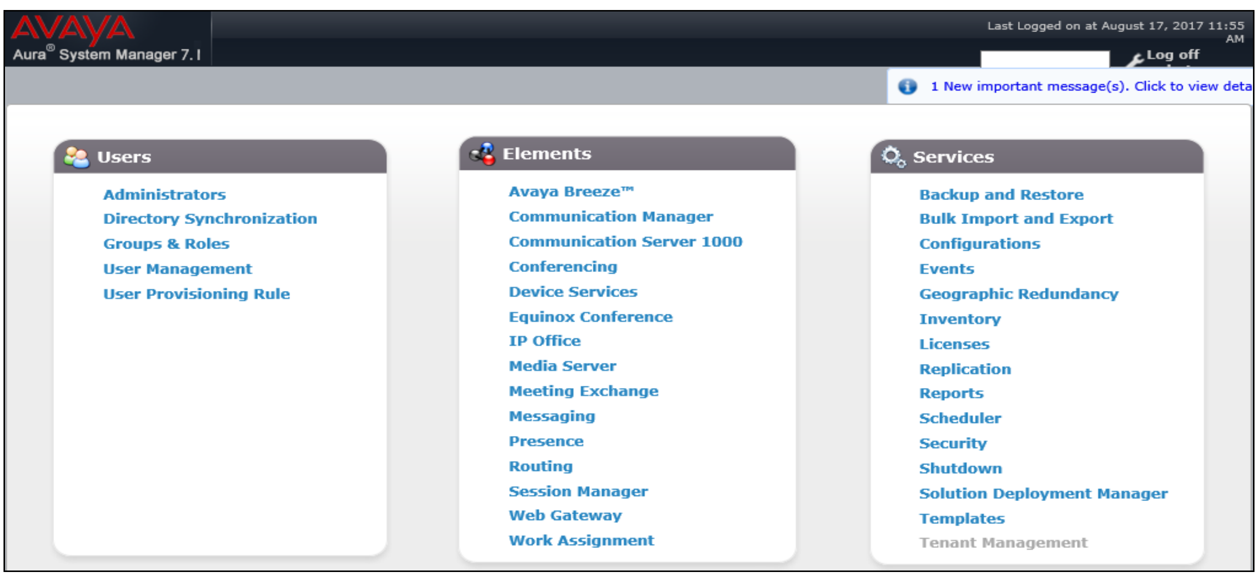
This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain
- Logical/physical Location that can be used by SIP Entities
- Adaptations
- 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
- TLS Certificate Management

It may not be necessary to configure all the items above when creating a connection to the service provider since some of these items would have already been defined as part of the initial Session Manager installation. This includes items such as certain SIP domains, locations, SIP entities, and Session Manager itself. However, each item should be reviewed to verify the configuration.

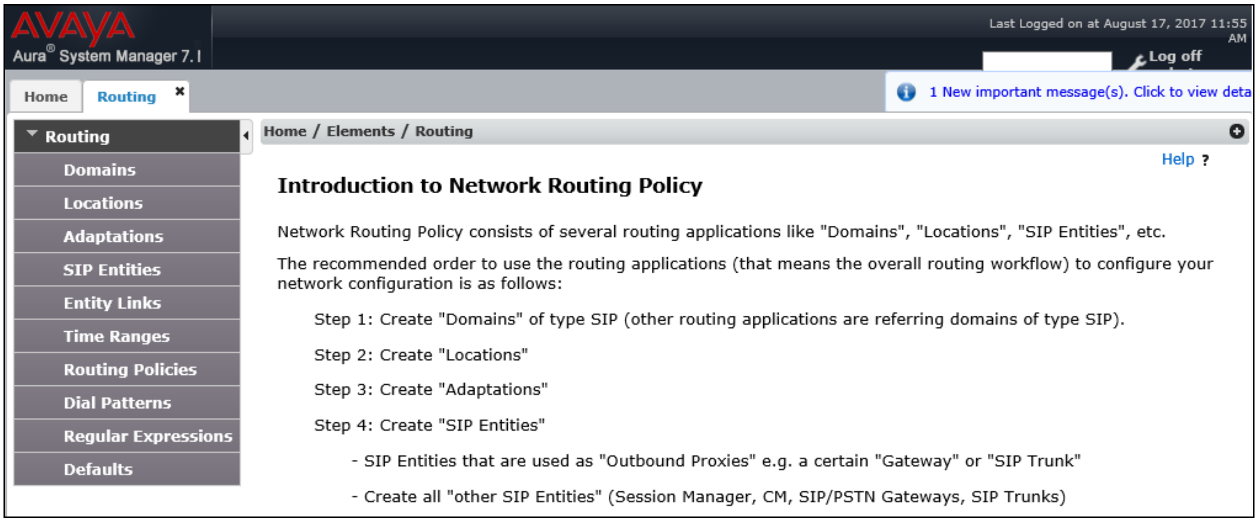
6.1. System Manager Login and Navigation

Session Manager configuration is accomplished by accessing the Web GUI of System Manager, using the URL "https://<ip-address>/SMGR", where "<ip-address>" is the IP address or FQDN of System Manager. At the **System Manager Log On** screen, provide the appropriate credentials and click on **Login** (not shown). The initial screen shown below is then displayed.



Most of the configuration items are performed in the Routing element. Click on **Routing** in the **Elements** column to bring up the **Introduction to Network Routing Policy** screen.

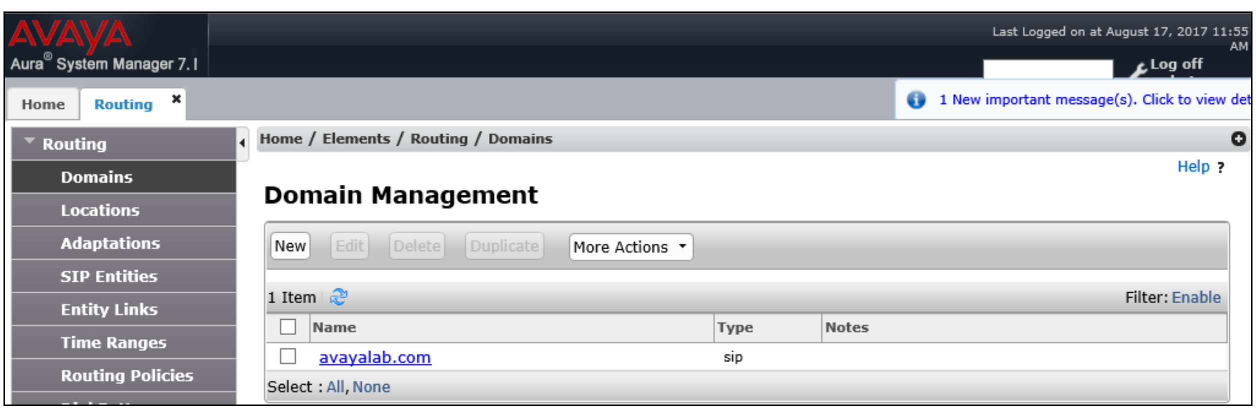
The navigation tree displayed in the left pane will be referenced in subsequent sections to navigate to items requiring configuration.



6.2. Specify SIP Domain

To view or to change SIP domains, select **Routing → Domains**. Click on the checkbox next to the name of the SIP domain and **Edit** to edit an existing domain, or the **New** button to add a domain. Click the **Commit** button (not shown) after changes are completed.

The following screen shows the list of configured SIP domains. The domain "***avayalab.com***" was already created for communication between Session Manager and Communication Manager. The domain "***avayalab.com***" is not known to the service provider. It will be adapted by the Avaya SBCE to IP address based URI-Host to meet the SIP specification of the service provider.



6.3. Add Location

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

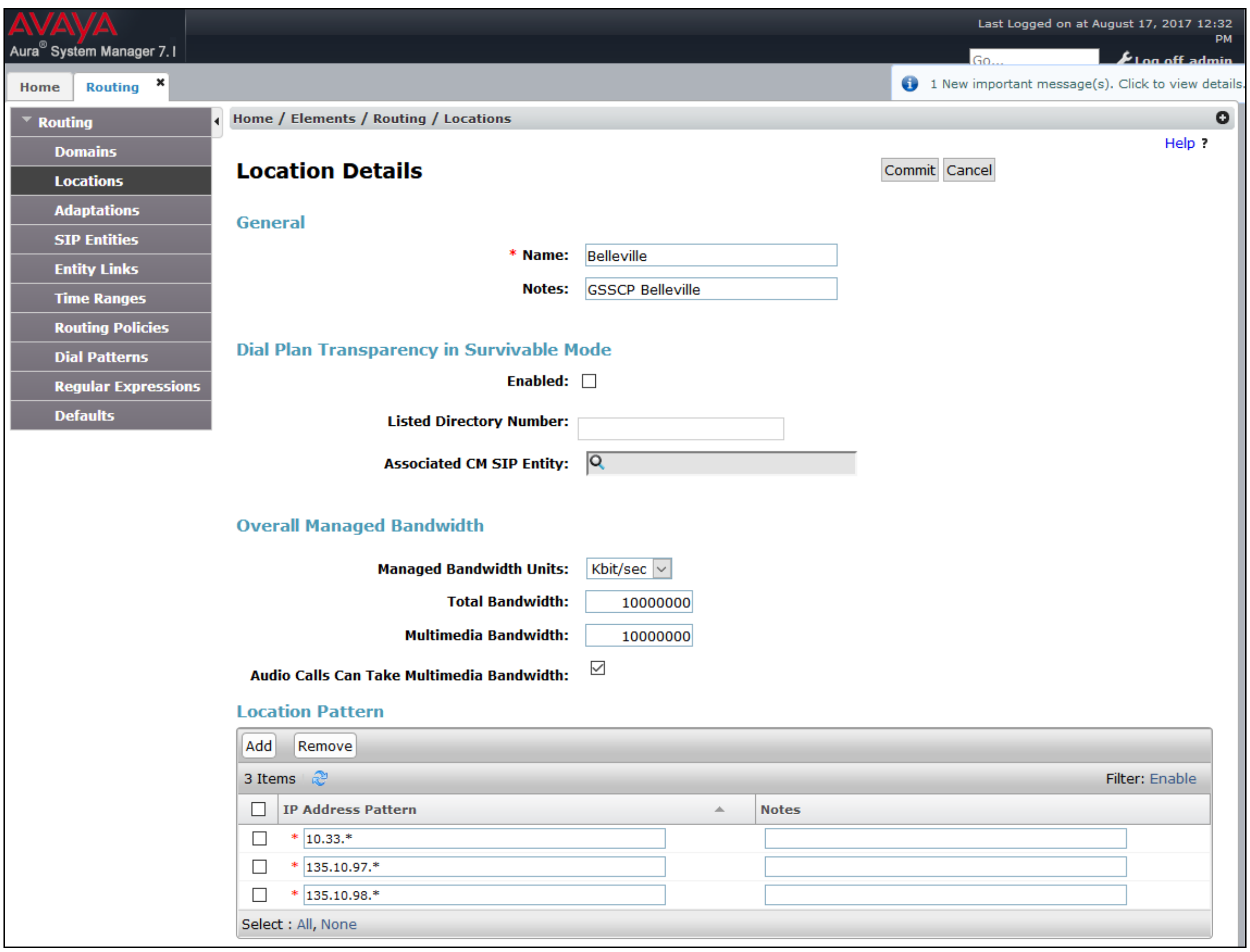
In **General** section, enter the following values:

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

In the **Location Pattern** section (see the screen below), click **Add** and enter following values:

- **IP Address Pattern:** An IP address pattern used to identify the location.
- **Notes:** Add a brief description (optional).

Displayed below are the screenshots for location ***Belleville***, which includes all equipment on the ***10.33.x, 10.10.98.x*** and ***10.10.97.x*** subnet including Communication Manager, Session Manager and Avaya SBCE. Click **Commit** to save.

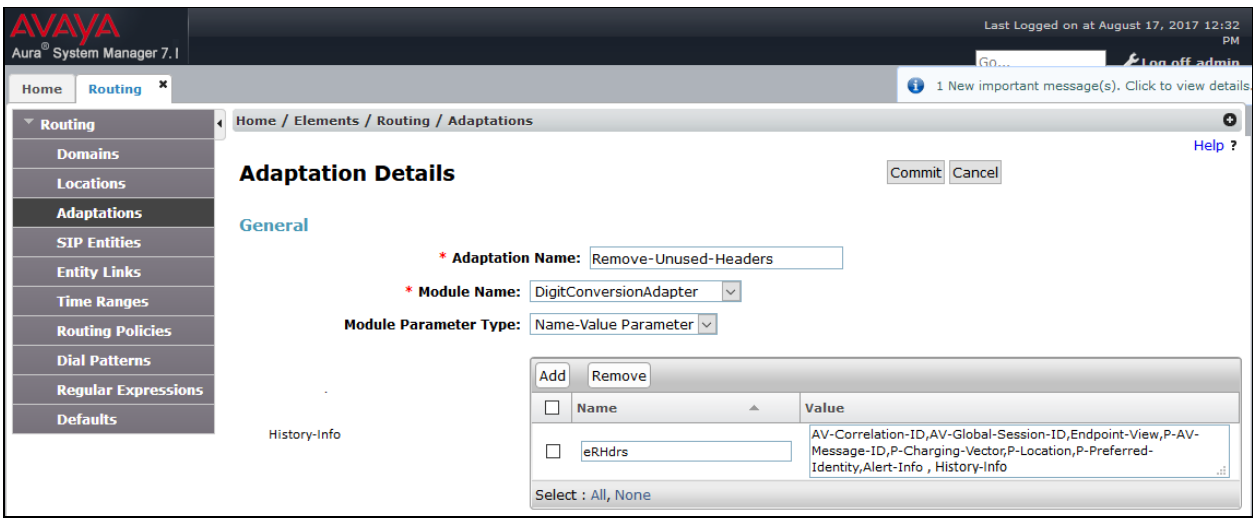


6.4. Add Adaptations

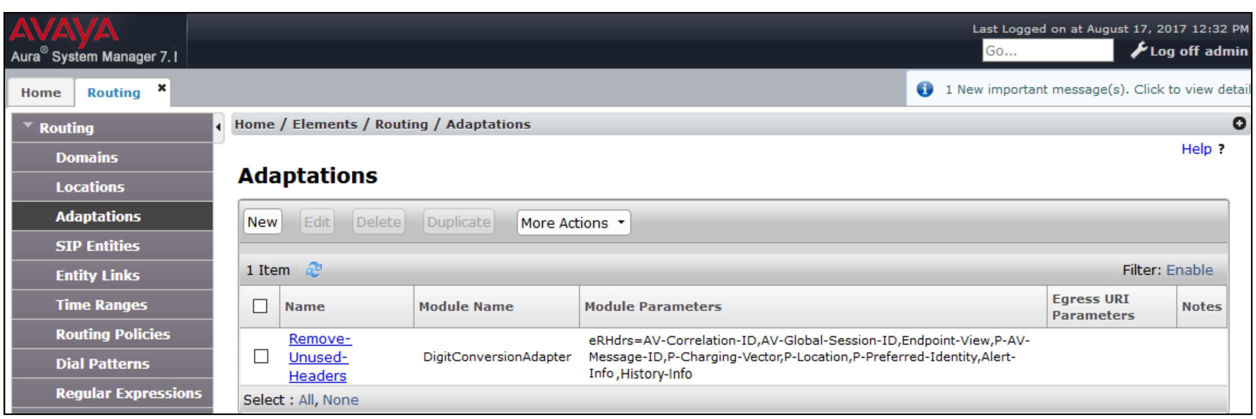
An adaptation is required by the service provider in order to remove un-wanted or proprietary headers that are not used or understood by the service provider.

To add a new adaptation, navigating to **Routing → Adaptations** in the left navigation pane and click **New** button in the right pane (not shown).

- **Adaptation Name**: Enter a descriptive name.
- **Module Name**: Select ***DigitConversionAdapter*** from pull down list.
- **Module Parameter Type:** Select ***Name-Value Parameter*** from pull down list.
- Click the **Add** button to enter a **Name** as shown in capture.
- **Value**: Enter the following information as shown in capture and click **Commit** button.



The newly created Adaptation is shown below.



6.5. Add SIP Entities

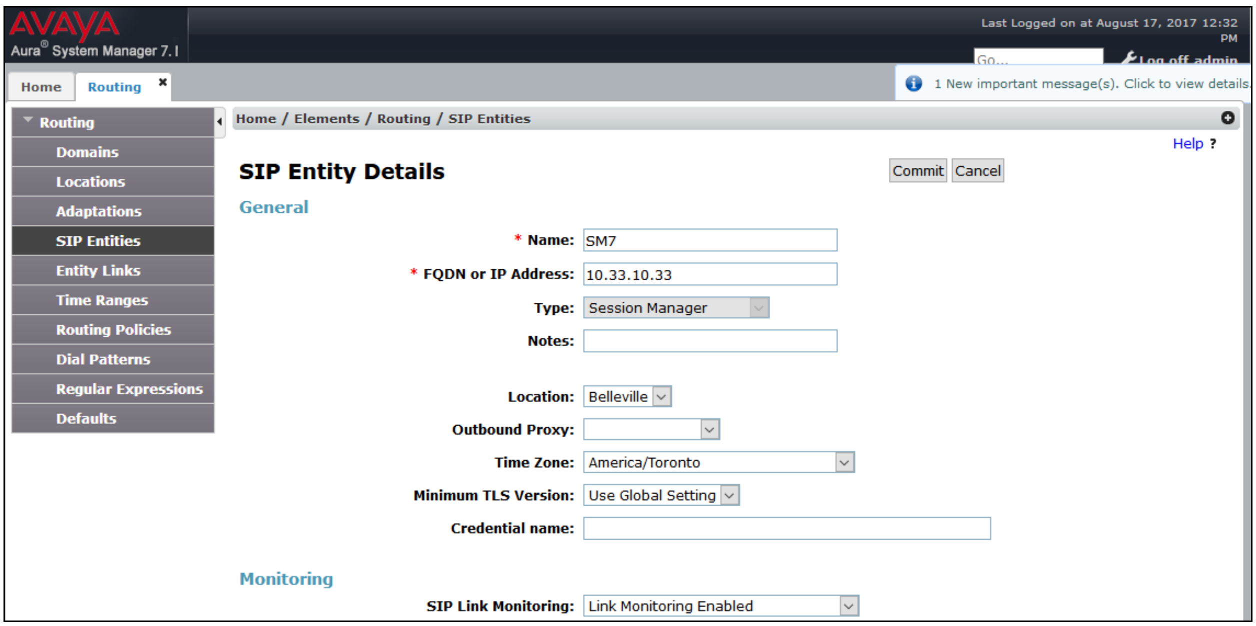
A SIP Entity must be added for Session Manager and for each SIP telephony system connected to it which includes Communication Manager and Avaya SBCE.

To add a new SIP Entity, navigate to **Routing → SIP Entities** in the left navigation pane and click **New** button in the right pane (not shown).

In **General** section, enter 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.
- **Location:** Select one of the locations defined in **Section** Error! Reference source not found..
- **Time Zone**: Select the time zone for the location above.

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



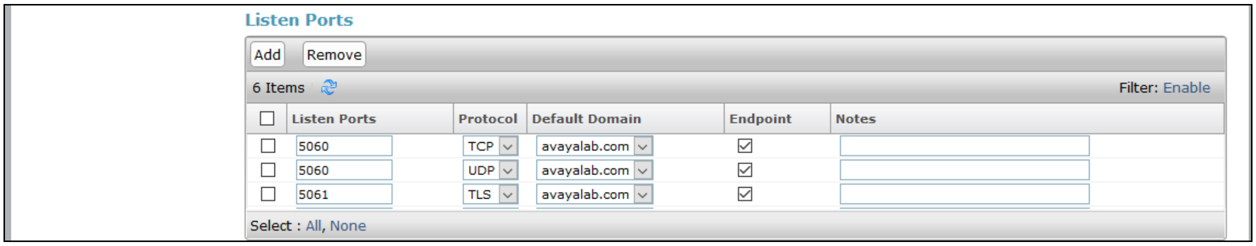
To define the ports used by Session Manager, scroll down to the **Port** section of the **SIP Entity Details** screen. This section is only present for the **Session Manager** SIP Entity.

In the **Port** section, click **Add** and enter following values. Use default values for all remaining fields:

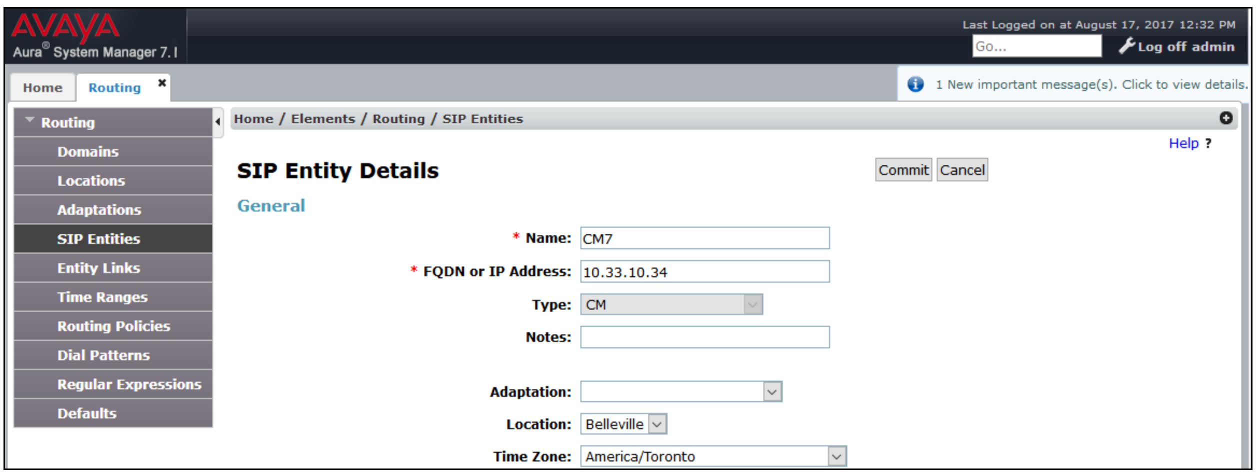
- **Listen Ports:** Port number on which the Session Manager can listen for SIP requests.
- **Protocol:** Transport protocol to be used to receive SIP requests.
- **Default Domain:** The domain used for the enterprise.

Defaults can be used for the remaining fields. Click **Commit** to save (not shown).

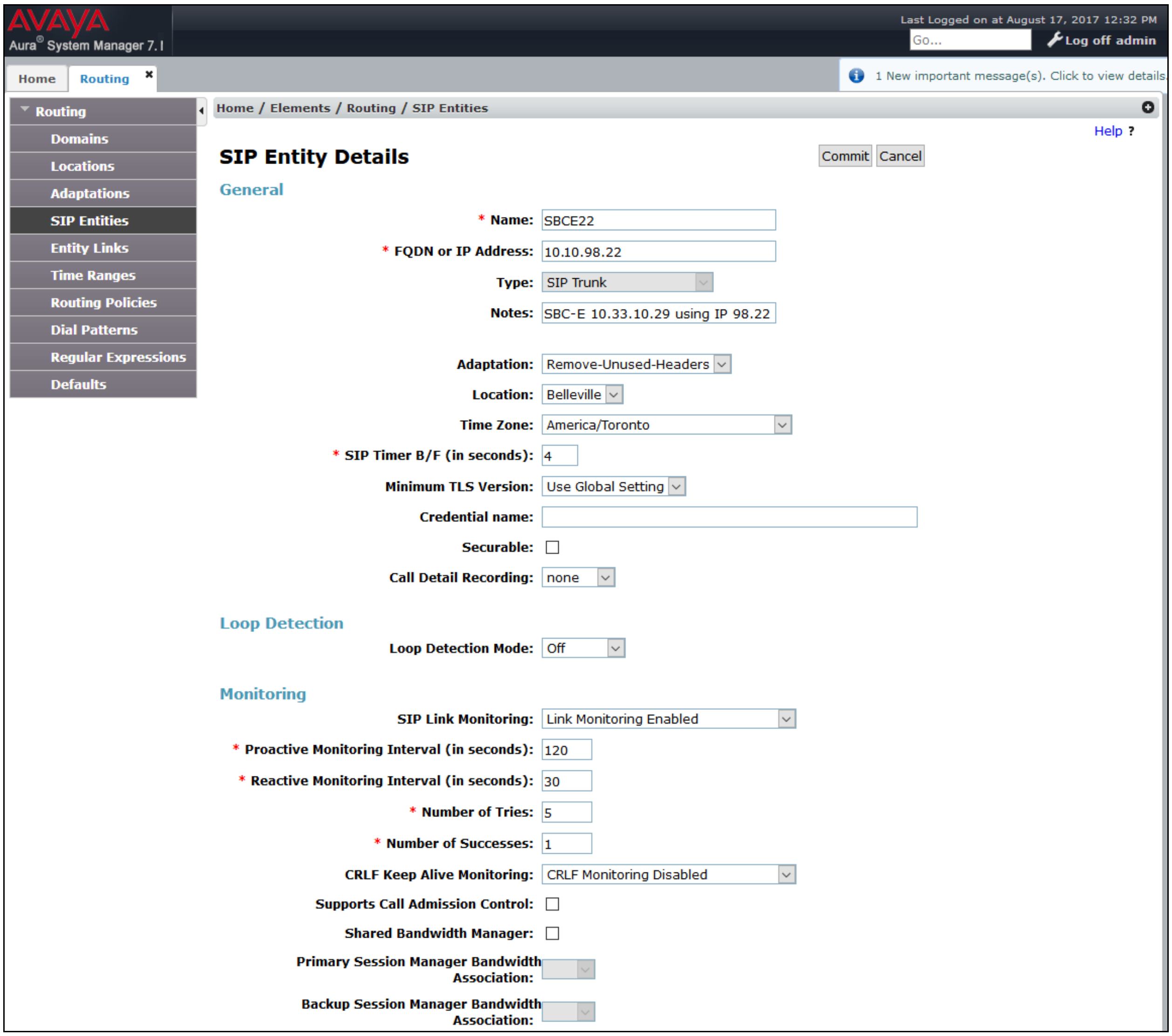
The compliance test used **Listen Ports** entry ***5061*** with ***TLS*** for connecting to Communication Manager and for connecting to the Avaya SBCE.



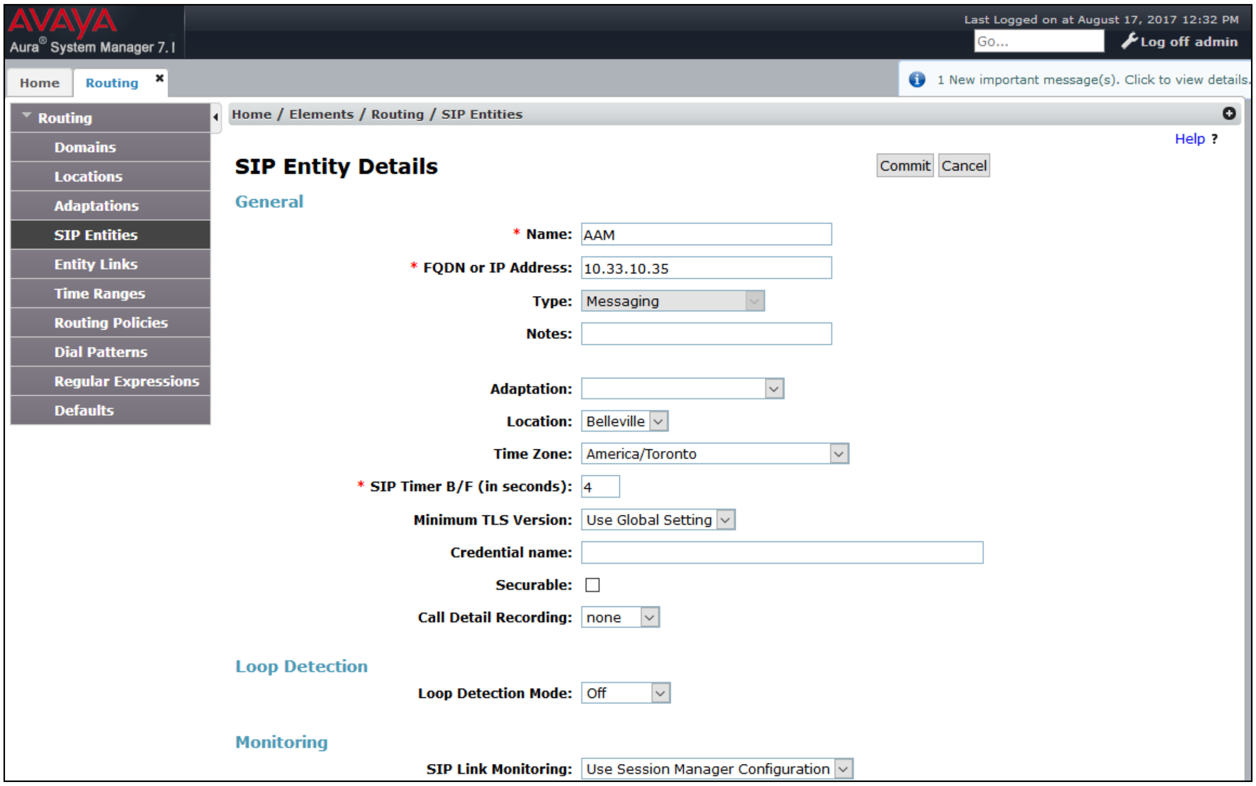
The following screen shows the addition of the Communication Manager SIP Entity. In order for Session Manager to send SIP traffic on an entity link to Communication Manager, it is necessary to create a SIP Entity for Communication Manager. The **FQDN or IP Address** field is set to IP address of Communication Manager and **Type** to ***CM***. The **Location** and **Time Zone** parameters are set as shown in screen below.



The following screen shows the addition of the SIP Entity for the Avaya SBCE. The **FQDN or IP Address** field is set to the IP address of its private network interface (see **Figure 1**). Select **Type** as ***SIP Trunk***. Select the created **Adaptation** in **Section 6.4** from pull the pull-down menu list. Select **SIP Link Monitoring** as **Link Monitoring Enabled** with the interval of ***120*** seconds. This setting allows Session Manager to send outbound OPTIONS heartbeat every ***120*** seconds to the service provider (which is forwarded by the Avaya SBCE) to query the status of the SIP trunk connecting to the service provider.



Similarly, a SIP Entity is added for Avaya Aura® Messaging server as shown in the capture below.



6.6. Add Entity Links

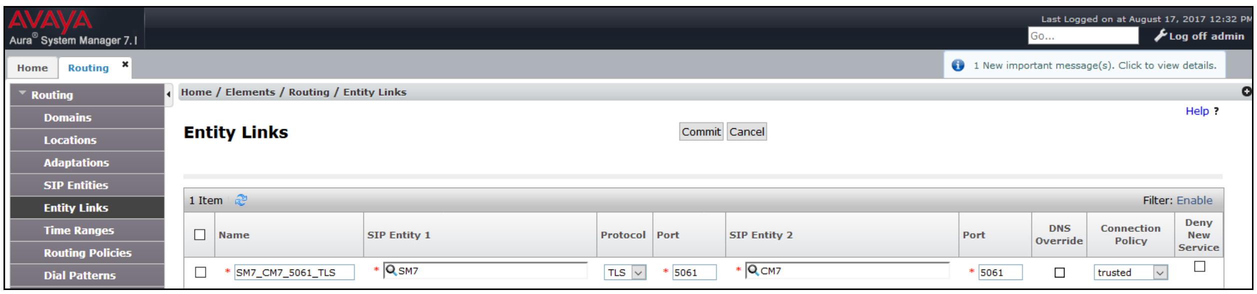
A SIP trunk between Session Manager and a telephony entity is described by an Entity Link. During compliance testing, three Entity Links were created, one for Communication Manager, Avaya Aura® Messaging and other for Avaya SBCE. To add an Entity Link, navigate to **Routing → Entity Links** in the left navigation pane and click **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 entity defined in **Section 6.5**.
- **Protocol:** Select the transport protocol used for this link, ***TLS*** for the Entity Link to Communication Manager and Avaya Aura® Messaging and ***TLS*** for the Entity Link to the Avaya SBCE.
- **Port:** Port number on which Session Manager will receive SIP requests from the far-end. For Communication Manager, this must match the **Far-end Listen Port** defined on the Communication Manager in **Section 5.6**.
- **SIP Entity 2:** Select the name of the other systems. For Communication Manager, select the Communication Manager SIP Entity defined in **Section** Error! Reference source not found.**5**. For Avaya SBCE, select Avaya SBCE SIP Entity defined in **Section** Error! Reference source not found.**5**.

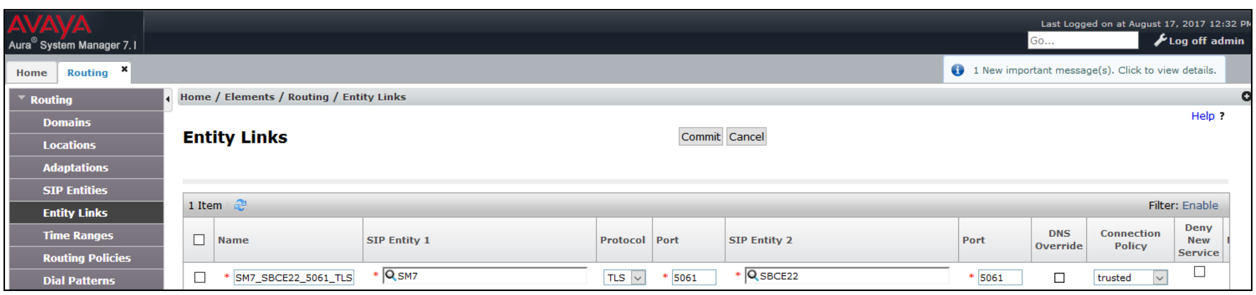
- **Port:** Port number on which the other system receives SIP requests from Session Manager. For Communication Manager, this must match the **Near-end Listen Port** defined on the Communication Manager in **Section 5.6**.
- **Connection Policy:** Select **Trusted**. **Note**: If this is not selected, calls from the associated SIP Entity specified in **Section 6.5** will be denied.
- Click **Commit** to save.

The following screens illustrate the Entity Links to Communication Manager and to the Avaya SBCE.

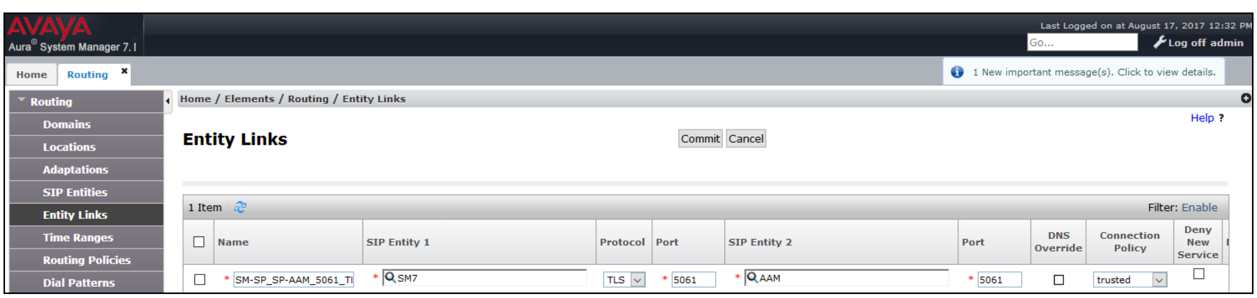
Entity Link to Communication Manager



Entity Link to Avaya SBCE



Entity Link to Avaya Aura® Messaging



6.7. Add Routing Policies

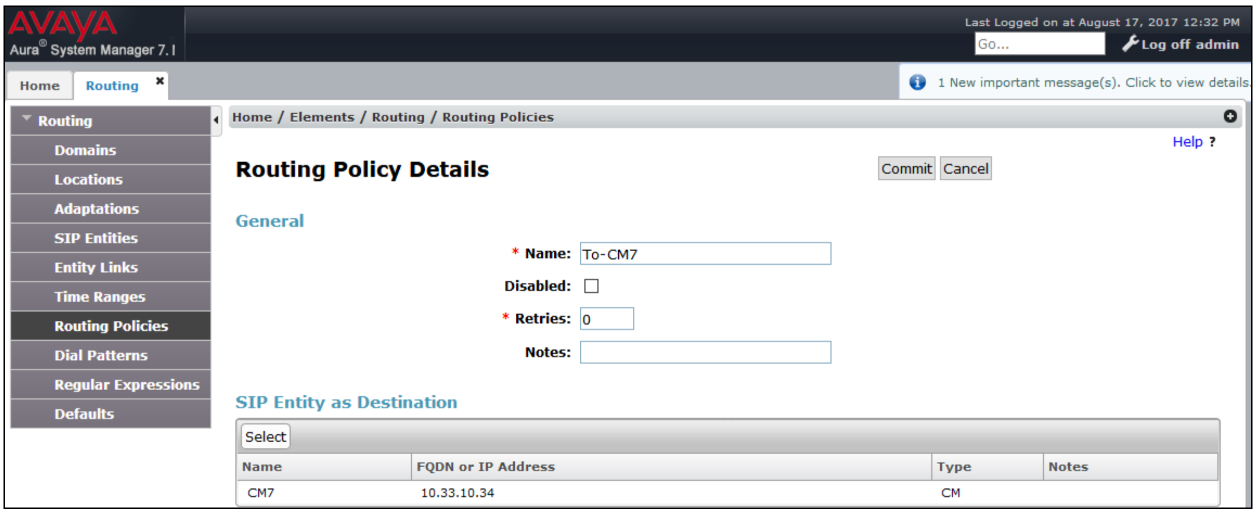
Routing Policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section** Error! Reference source not found.**5**. Three routing policies were added,

Communication Manager, Avaya Aura® Messaging and Avaya SBCE. To add a routing policy, navigate to **Routing → Routing Policies** in the left navigation pane and click **New** button in the right pane (not shown). The following screen is displayed.
In the **General** section, configure the following fields:

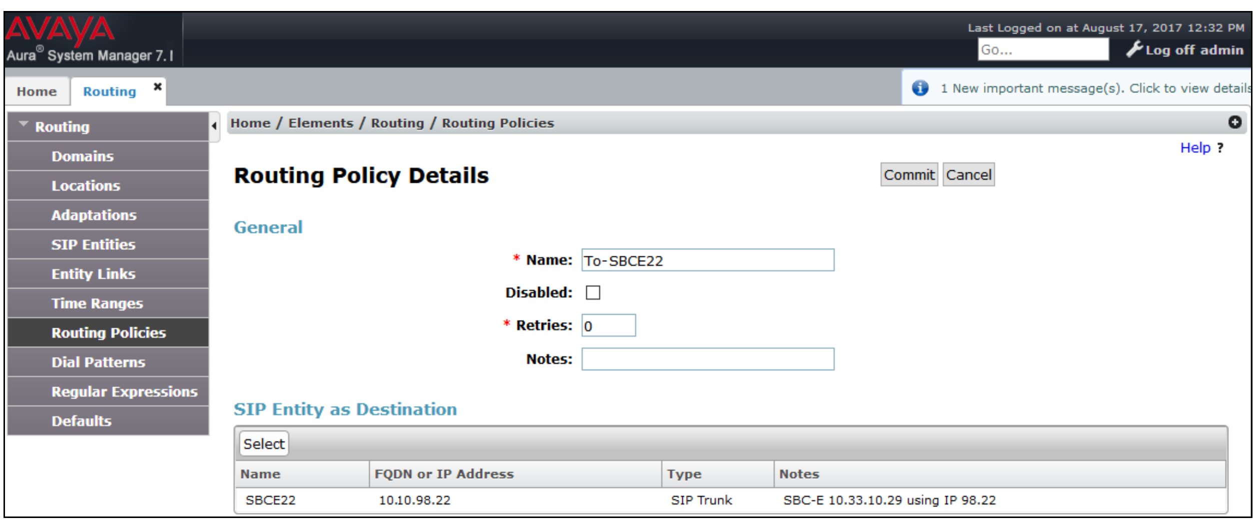
- **Name:** Enter a descriptive name.
- **Notes:** Add a brief description (optional).

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

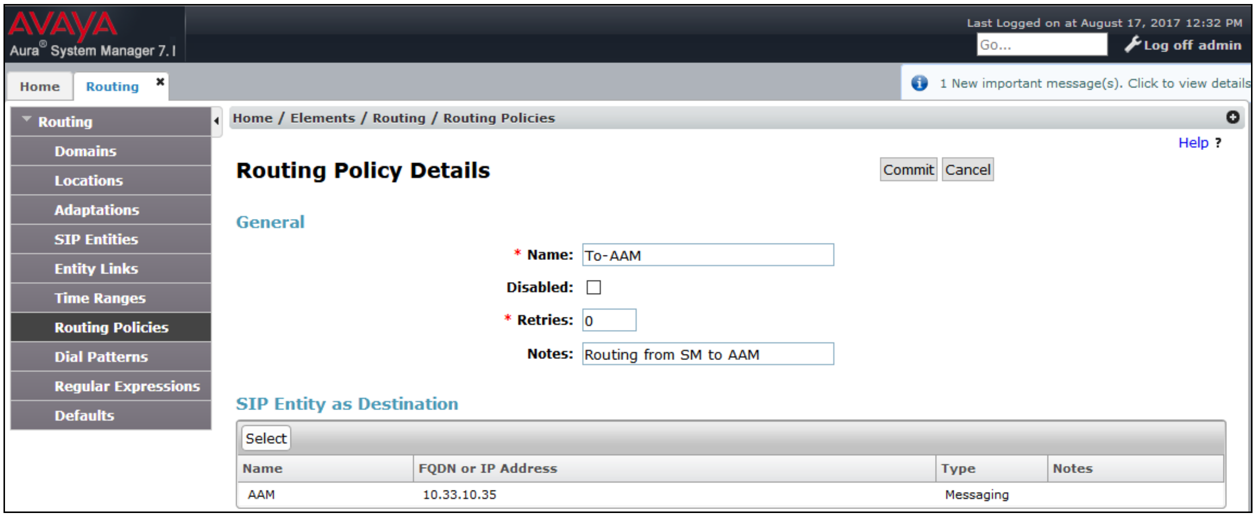
The following screen shows the Routing Policy for Communication Manager.



The following screen shows the Routing Policy for the Avaya SBCE.



The following screen shows the Routing Policy for the Avaya Aura® Messaging.



6.8. Add Dial Patterns

Dial Patterns are needed to route specific calls through Session Manager. For the compliance testing, dial patterns were needed to route calls from Communication Manager to Avaya Aura® Messaging and from Communication Manager to the service provider and vice versa. Dial Patterns define which routing 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 **New** button in the right pane (not shown).

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

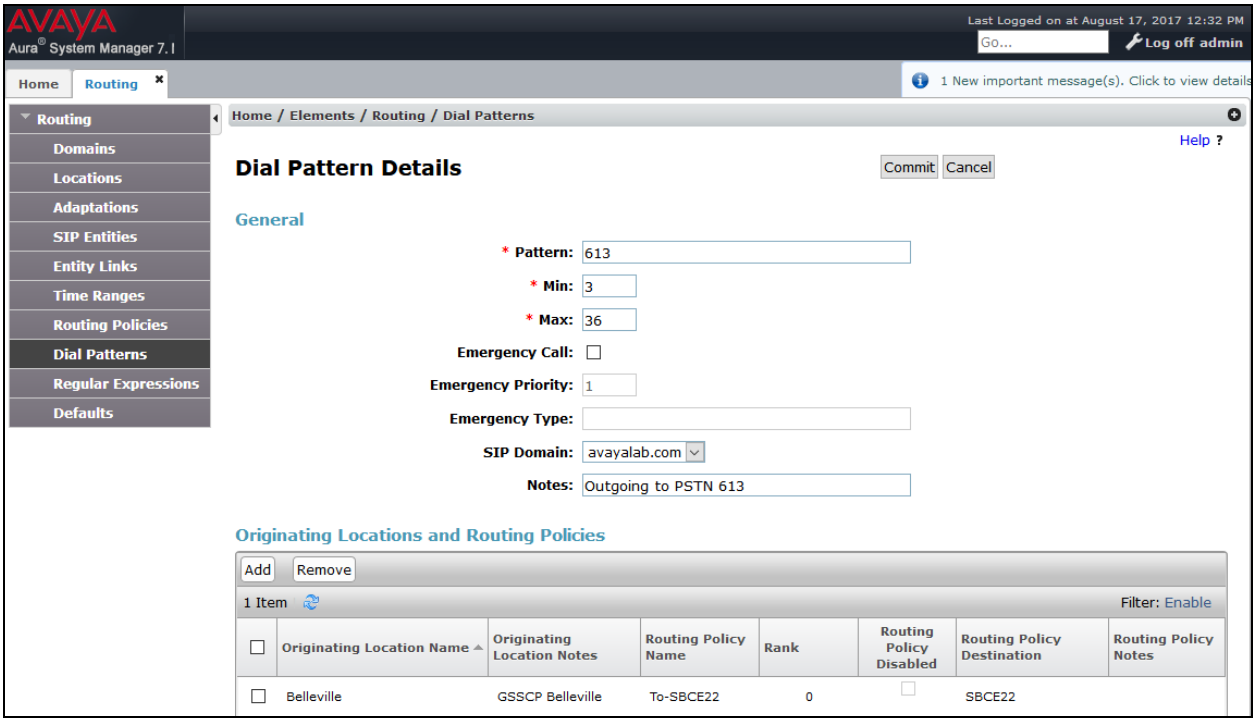
- **Pattern:** Enter a dial string that will be matched against the "Request-URI" of the call.
- **Min:** Enter a minimum length used in the match criteria.
- **Max:** Enter a maximum length used in the match criteria.
- **SIP Domain:** Enter the destination domain used in the match criteria.
- **Notes:** 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**.

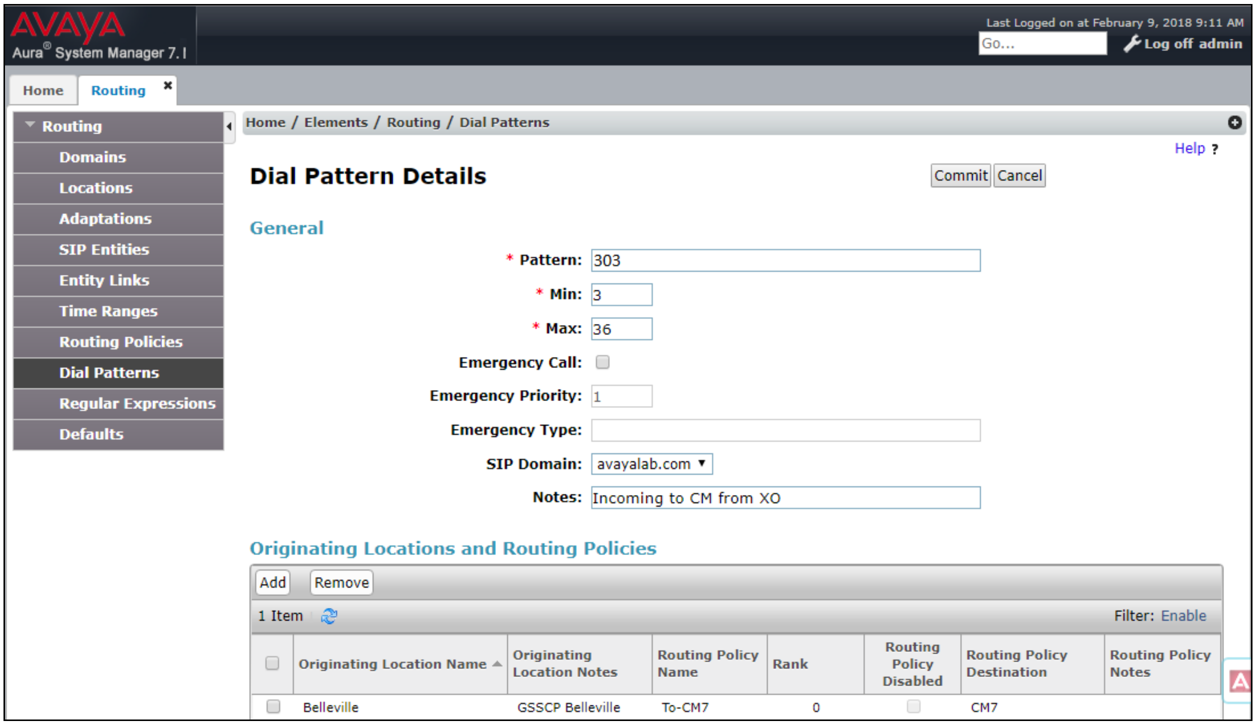
Default values can be used for the remaining fields. Click **Commit** to save.

Two examples of the dial patterns used for the compliance testing are shown below, one for outbound calls from the enterprise to the PSTN and one for inbound calls from the PSTN to the enterprise.

The first example shows that 10-digit dialed numbers that have a destination domain of "avayalab.com" uses route policy to Avaya SBCE as defined in **Section** Error! Reference source not found.**7**.



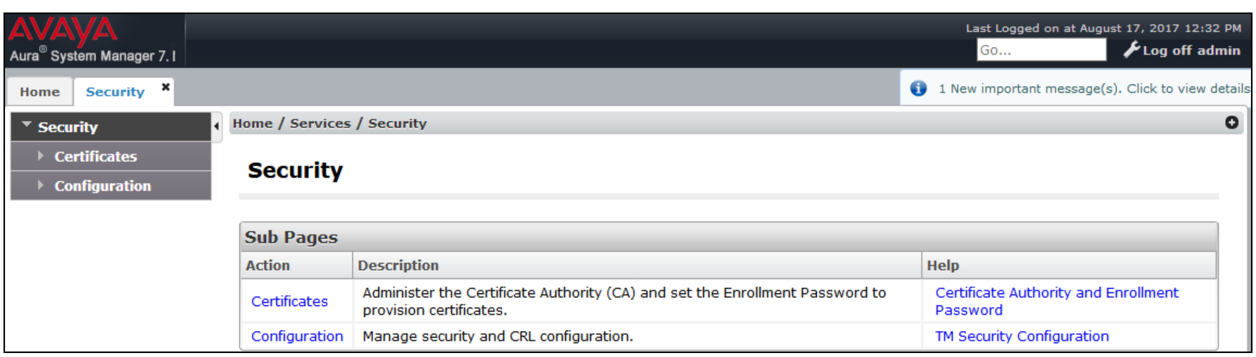
The second example shows that inbound 10-digit numbers assigned by the Service provider with domain “avayalab.com” to use route policy to Communication Manager as defined in **Section** Error! Reference source not found.**7**.



6.9. TLS Certificate Management on System Manager

This section is to provide a procedure how to download System Manger CA certificate which is being installed on Avaya Communication Manager and Avaya SBCE for the communication between Avaya system components using TLS connectivity.

From System Manager Menu in **Section 6.1**, navigate to **Services → Security**. Click on arrow tab to show navigation tree as shown.



Navigate to **Certificates → Authority → CA Functions → CA Structure & CRLs**. Then click on **Download PEM file** to download the System Manager CA certificate and save it as ***SystemManagerCA.pem*** to a directory on local management PC.

7. Configure Avaya Session Border Controller for Enterprise

In the sample configuration, an Avaya SBCE is used as the edge device between the Avaya CPE and the Service provider SIP Trunking Service.

These Application Notes assume that the installation of the Avaya SBCE and the assignment of a management IP Address have already been completed.

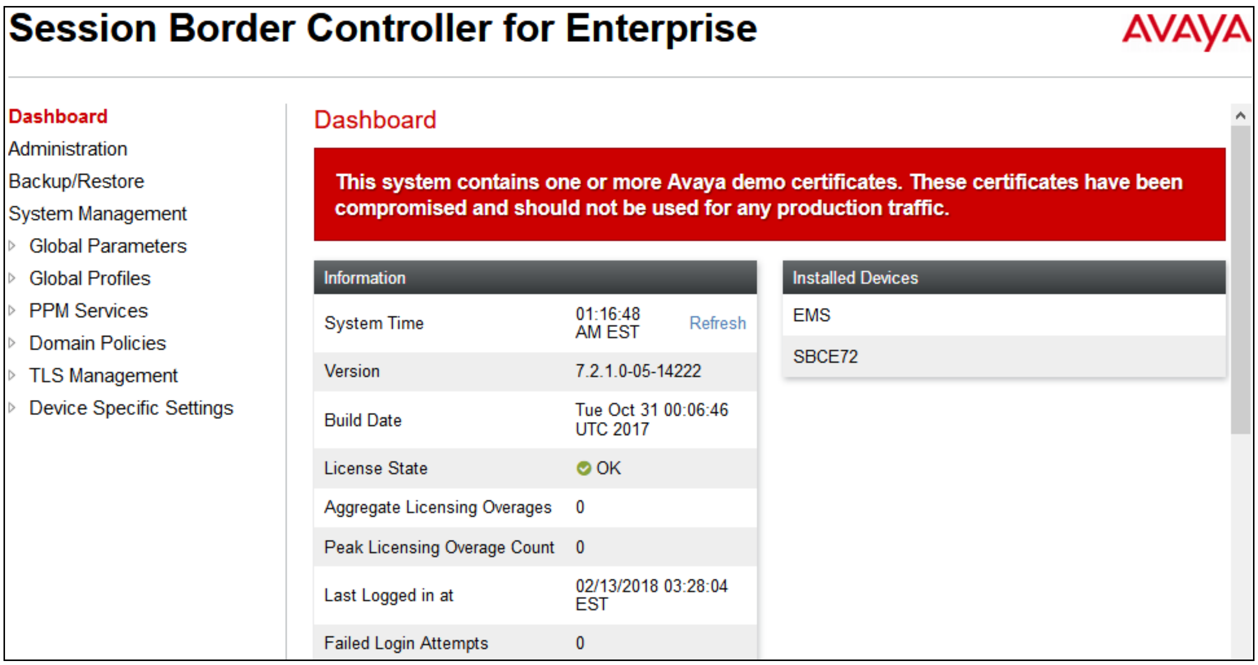
In this session, the naming convention used for the Service Provider is **SP**, which is connected to the external interface of the Avaya SBCE. The Avaya Enterprise side is **EN**, which is connected to the internal interface of the Avaya SBCE.

7.1. Avaya Session Border Controller for Enterprise Login

Use a Web browser to access the Avaya SBCE web interface, enter “https://<ip-addr>/sbc” in the address field of the web browser (not shown), where “<ip-addr>” is the management LAN IP address of Avaya SBCE.

Enter appropriate credentials and click ***Continue***. Then enter password to login.

The main page of the Avaya SBCE will appear as shown below.



7.2. TLS Management

Transport Layer Security (TLS) is a standard protocol that is used extensively to provide a secure channel by encrypting communications over IP networks. It enables clients to authenticate servers or, optionally, servers to authenticate clients.

Avaya SBCE supports the configuration of third-party certificates and TLS settings. For optimum security, Avaya recommends using third-party CA certificates for enhanced security

Testing was done with System Manager signed identity certificates. The procedure to obtain and install 3rd party CA certificates is outside the scope of these application notes.

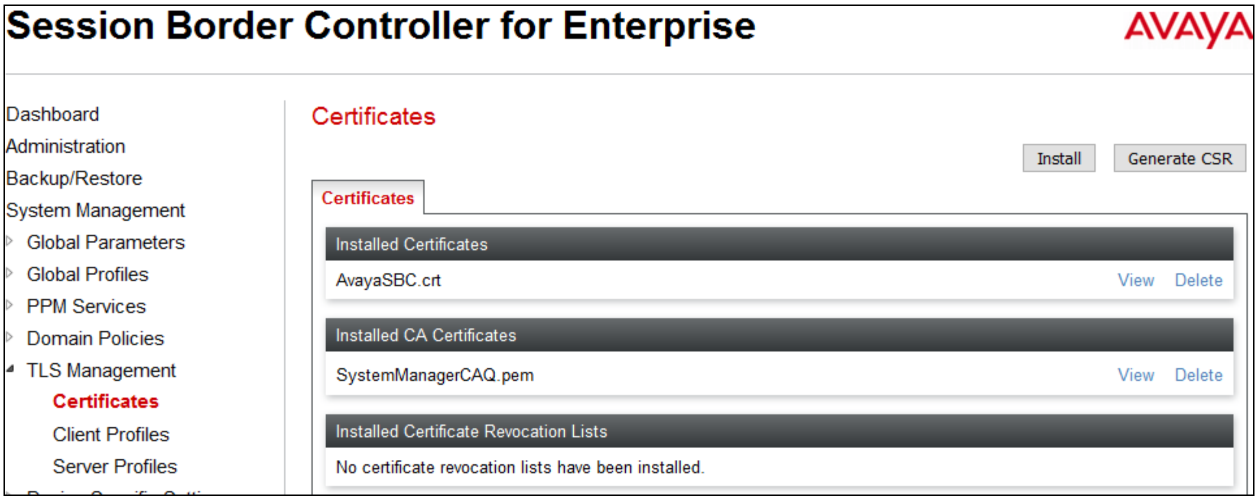
In this compliance testing, TLS transport is used for the communication between Session Manager and Avaya SBCE. The following procedures show how to create the client and server profiles.

7.2.1. Certificates

You can use the certificate management functionality that is built into the Avaya SBCE to control all certificates used in TLS handshakes. You can access the Certificates screen from **TLS Management → Certificates**.

Ensure the preinstalled certificates are presented in the system as shown below.

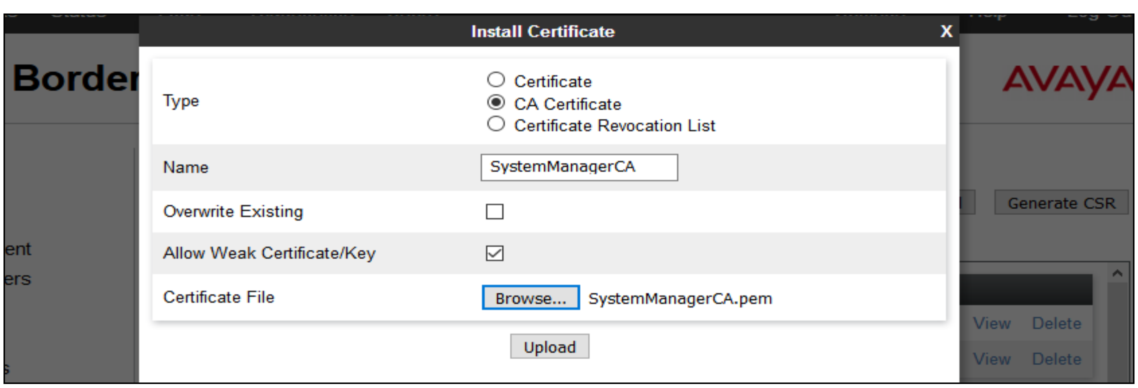
- ***AvayaSBC.crt*** is Avaya SBCE identify certificate.
- ***SystemManagerCAQ.pem*** is System Manager CA certificate.



If System Manager CA certificate (SystemManagerCAQ.pem) is not present, the following procedure shows how to install it on the Avaya SBCE.

System Manager CA certificate is obtained using the procedure provided in **Section 6.9**. Then on the Avaya SBCE, navigate to **TLS Management → Certificates**. Click on **Install** button.

- Select **CA Certificate**.
- Provide a descriptive **Name**.
- **Browse** to the directory where the System Manager CA certificate previously saved and select it.
- Click **Upload**.

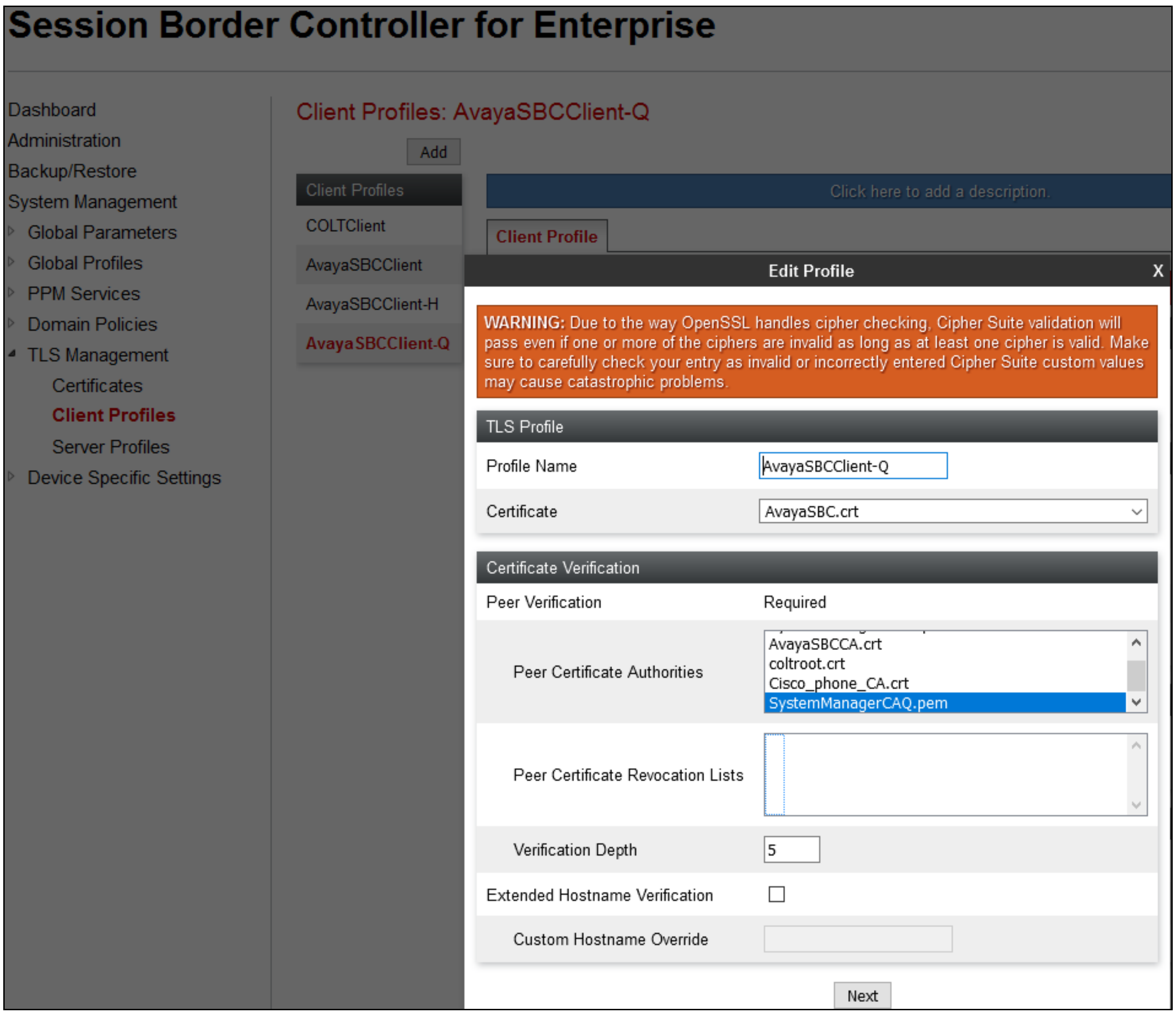


7.2.2. Client Profiles

This section describes the procedure to create client profile for Avaya SBCE to communicate with Session Manager via TLS signaling. This profile will be used in **Section 7.3.4**.

To create Client profile, navigate to **TLS Management → Client Profiles**, click on **Add**.

- Enter descriptive name in **Profile Name**.
- Select ***AvayaSBC.crt*** from pull down menu of **Certificate**.
- Select ***SystemManagerCAQ.pem*** from pull down of **Peer Certificate Authorities**.
- Enter ***5*** as **Verification Depth**.
- Click **Next** and **Finish** (not shown).

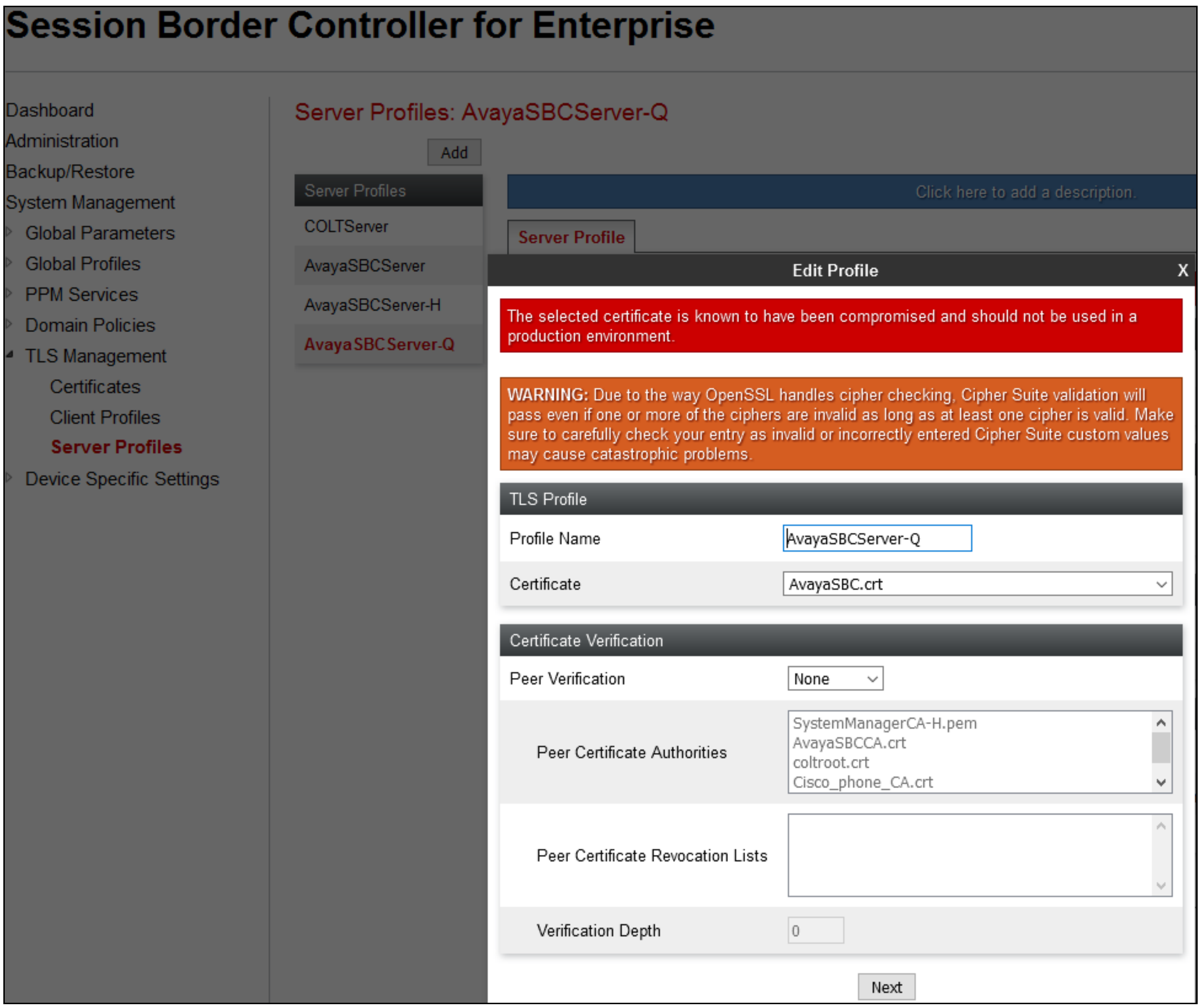


7.2.3. Server Profiles

This section describes the procedure to create server profile for Avaya SBCE to communicate with Session Manager via TLS signaling. This will be used in **Section 7.5.3**.

To create Server profile, navigate to **TLS Management** → **Server Profiles**, click on **Add**.

- Enter descriptive name in **Profile Name**.
- Select ***AvayaSBC.crt*** from pull down menu of **Certificate**.
- Select ***None*** from pull down menu of **Peer Verification**.
- Others are left at default.
- Click **Next** and **Finish** (not shown).



7.3. Global Profiles

Global Profiles allows for configuration of parameters across all Avaya SBCE appliances.

7.3.1. Uniform Resource Identifier (URI) Groups

URI Group feature allows a user to create any number of logical URI Groups that are comprised of individual SIP subscribers located in that particular domain or group. These groups are used by the various domain policies to determine which actions (Allow, Block, or Apply Policy) should be used for a given call flow.

For this configuration testing, "*" is used for all incoming and outgoing traffic.

7.3.2. Server Interworking Profile

Interworking Profile features are configured differently for Call Server and Trunk Server.

To create a Server Interworking profile, select **Global Profiles → Server Interworking**. Click on the **Add** button.

In the compliance testing, two Server Interworking profiles were created for SP and EN respectively.

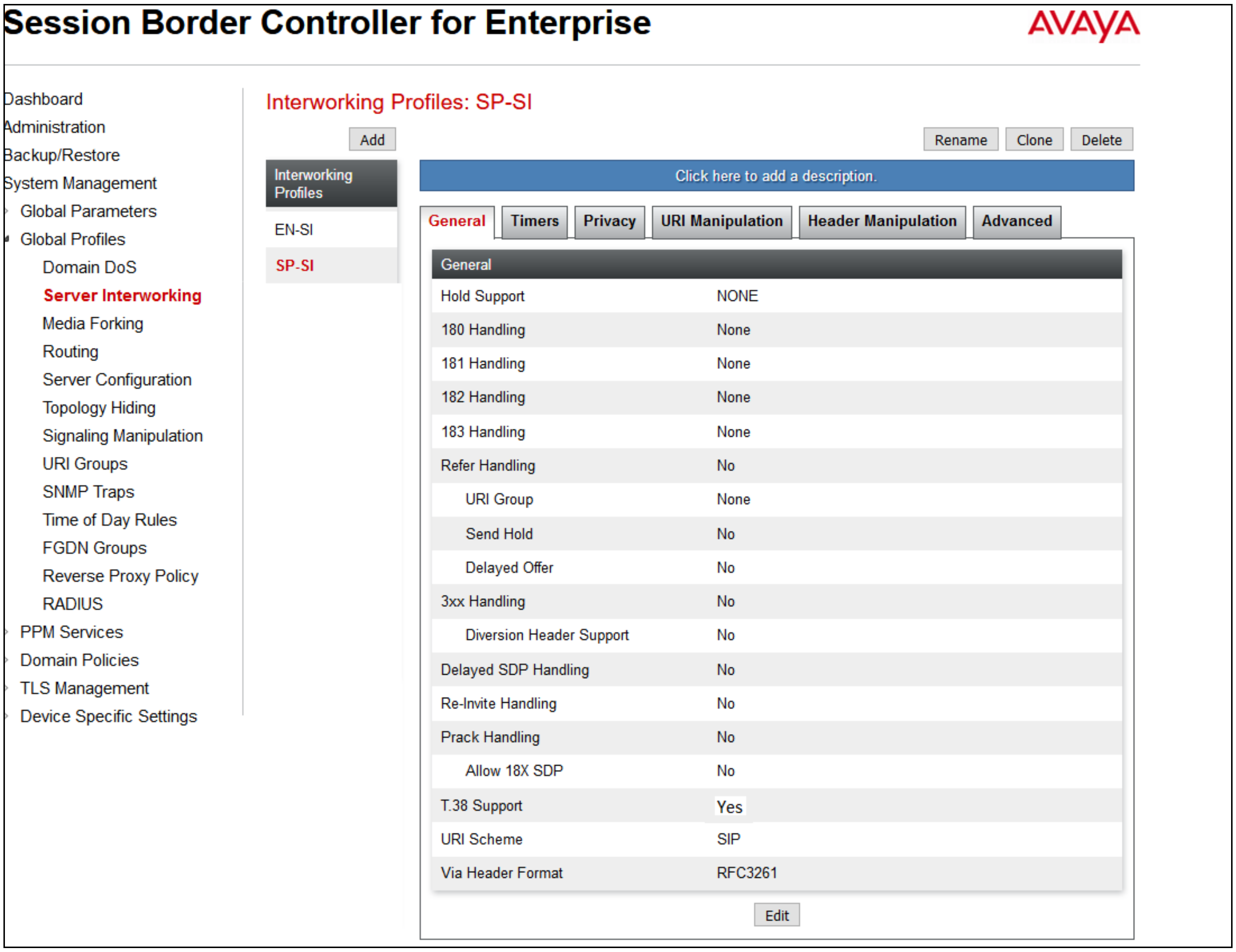
Server Interworking profile for SP

Profile **SP-SI** was defined to match the specification of SP. The **General** and **Advanced** tabs are configured with the following parameters while the other tabs for **Timers, Privacy, URI Manipulation** and **Header Manipulation** are kept as default.

General tab:

- **Hold Support** = ***NONE***. The Avaya SBCE will not modify the hold/ resume signaling from EN to SP.
- **18X Handling** = ***None***. The Avaya SBCE will not handle 18X, it will keep the 18X messages unchanged from EN to SP.
- **Refer Handling** = ***No***. The Avaya SBCE will not handle REFER. It will keep the REFER message unchanged from EN to SP.
- **T.38 Support** = ***Yes***. SP supports T.38 fax in the compliance testing.
- Others are left as default values.

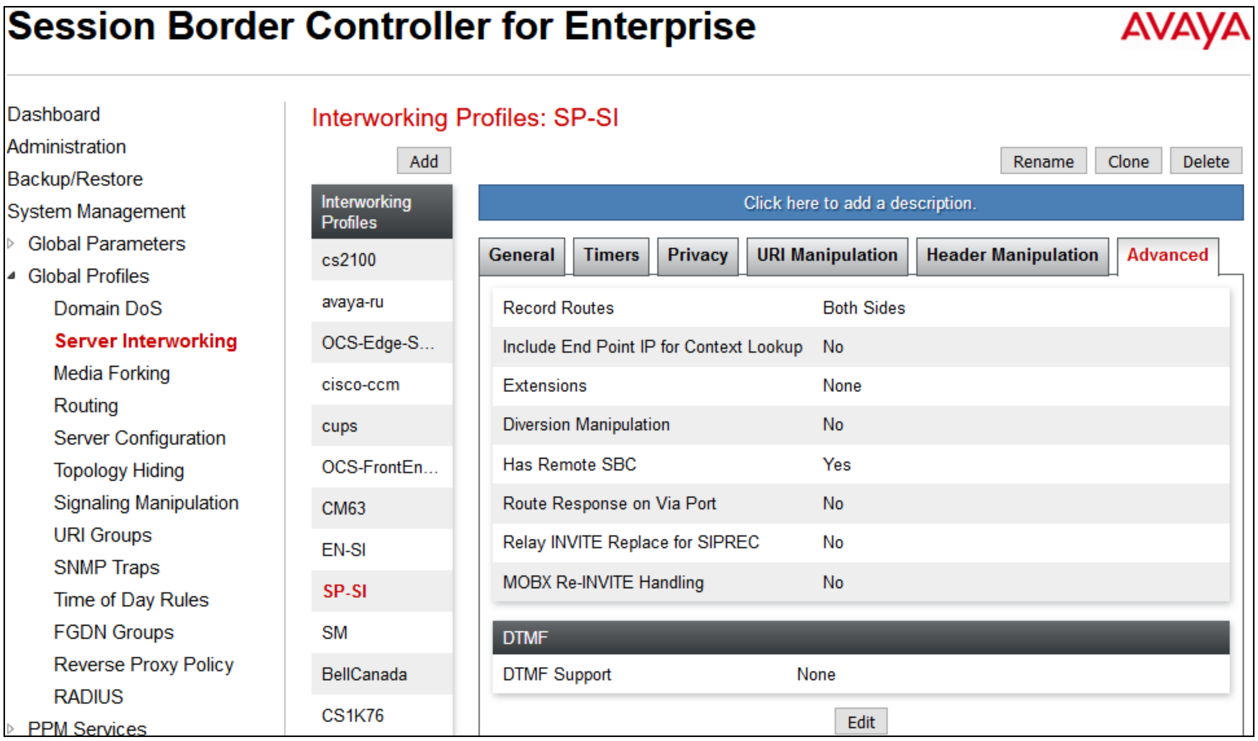
The screenshots below illustrate the Server Interworking profile **SP-SI, General**.



Advanced tab:

- **Record Routes**: ***Both Sides***.
- **Include End Point IP for Context Lookup**: ***No***.
- **Extensions**: ***None***.
- **Has Remote SBC**: ***Yes***. SP has an SBC which interfaces its Central Office (CO) to the enterprise SIP trunk. This setting allows the Avaya SBCE to always use the SDP received from SP for the media.
- **DTMF Support**: ***None***. The Avaya SBCE will send original DTMF method from EN to SP.
- Others are left as default values.

The screenshots below illustrate the Server Interworking profile **SP-SI**, **Advanced**.



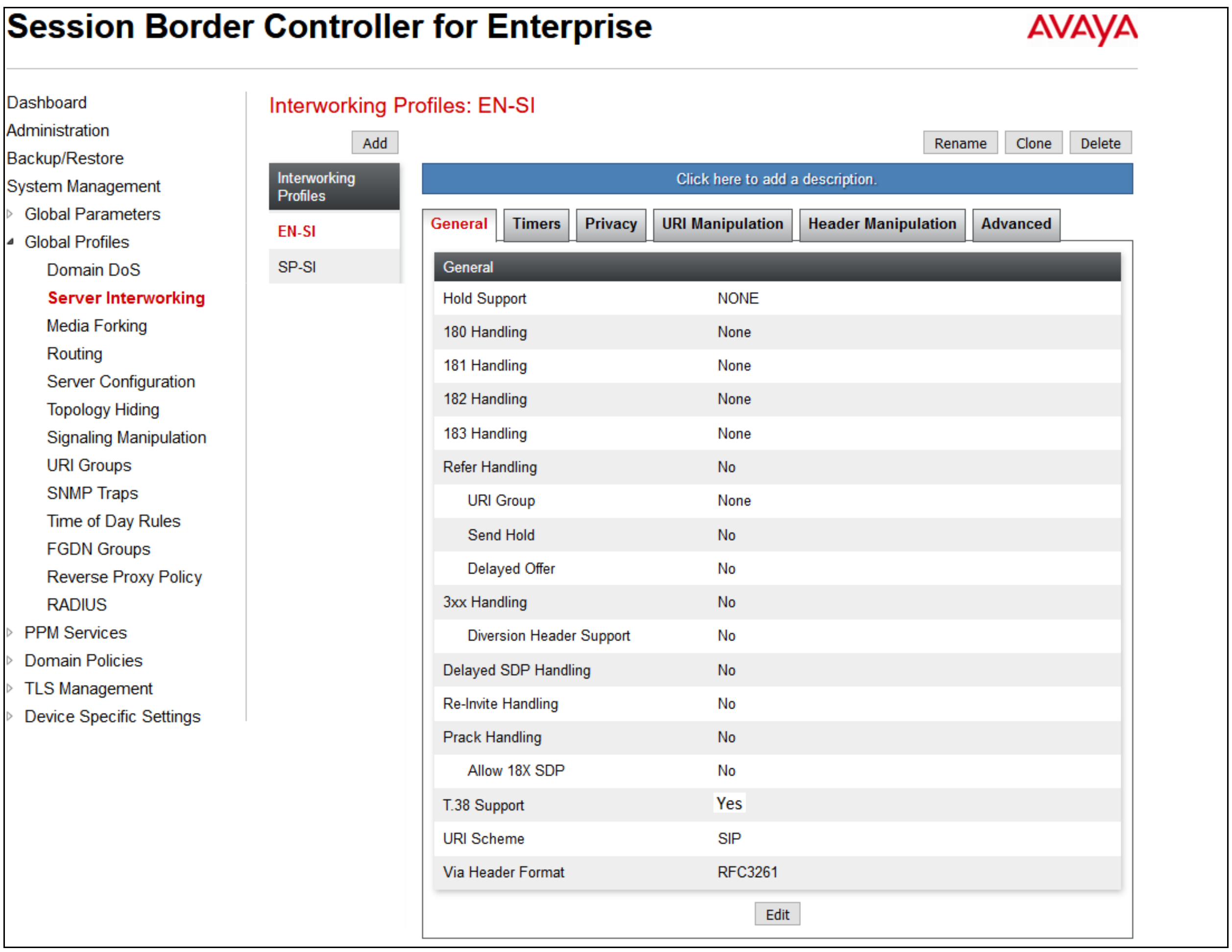
Server Interworking profile for EN

Profile **EN-SI** was defined to match the specification of EN. The **General** and **Advanced** tabs are configured with the following parameters while the other settings for **Timers**, **Privacy, URI Manipulation** and **Header Manipulation** are kept as default.

General tab:

- **Hold Support**: *NONE*.
- **18X Handling**: *None*. The Avaya SBCE will not handle 18X, it will keep the 18X messages unchanged from SP to EN.
- **Refer Handling**: *No*. The Avaya SBCE will not handle REFER, it will keep the REFER messages unchanged from SP to EN.
- **T.38 Support**: *Yes*.
- Others are left as default values.

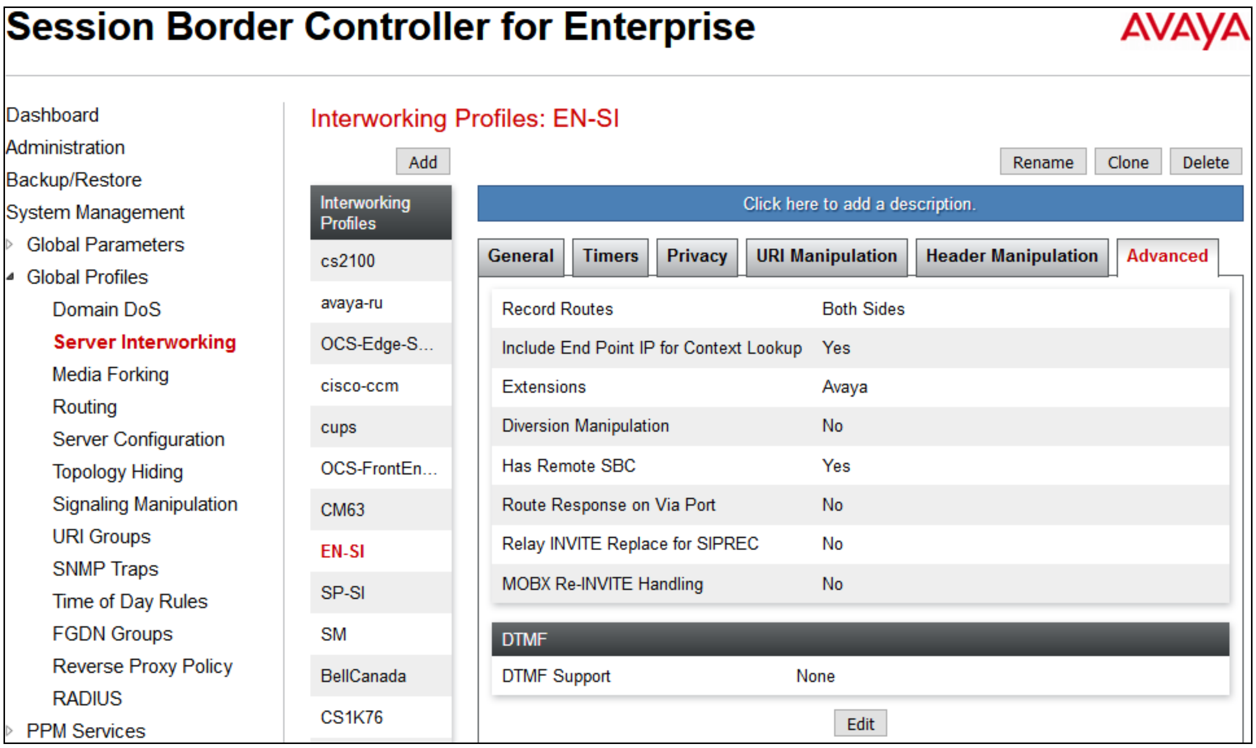
The screenshots below illustrate the Server Interworking profile **EN-SI**, **General**.



Advanced tab:

- **Record Routes**: ***Both Sides***. The Avaya SBCE will send Record-Route header to both call and trunk servers.
- **Include End Point IP for Context Lookup** = ***Yes***.
- **Extensions**: ***Avaya***.
- **Has Remote SBC**: ***Yes***. This setting allows the Avaya SBCE to always use the SDP received from EN for the media.
- **DTMF Support**: ***None***. The Avaya SBCE will send original DTMF method from SP to EN.
- Others are left as default values.

The screenshots below illustrate the Server Interworking profile **EN-SI**, **Advanced**.



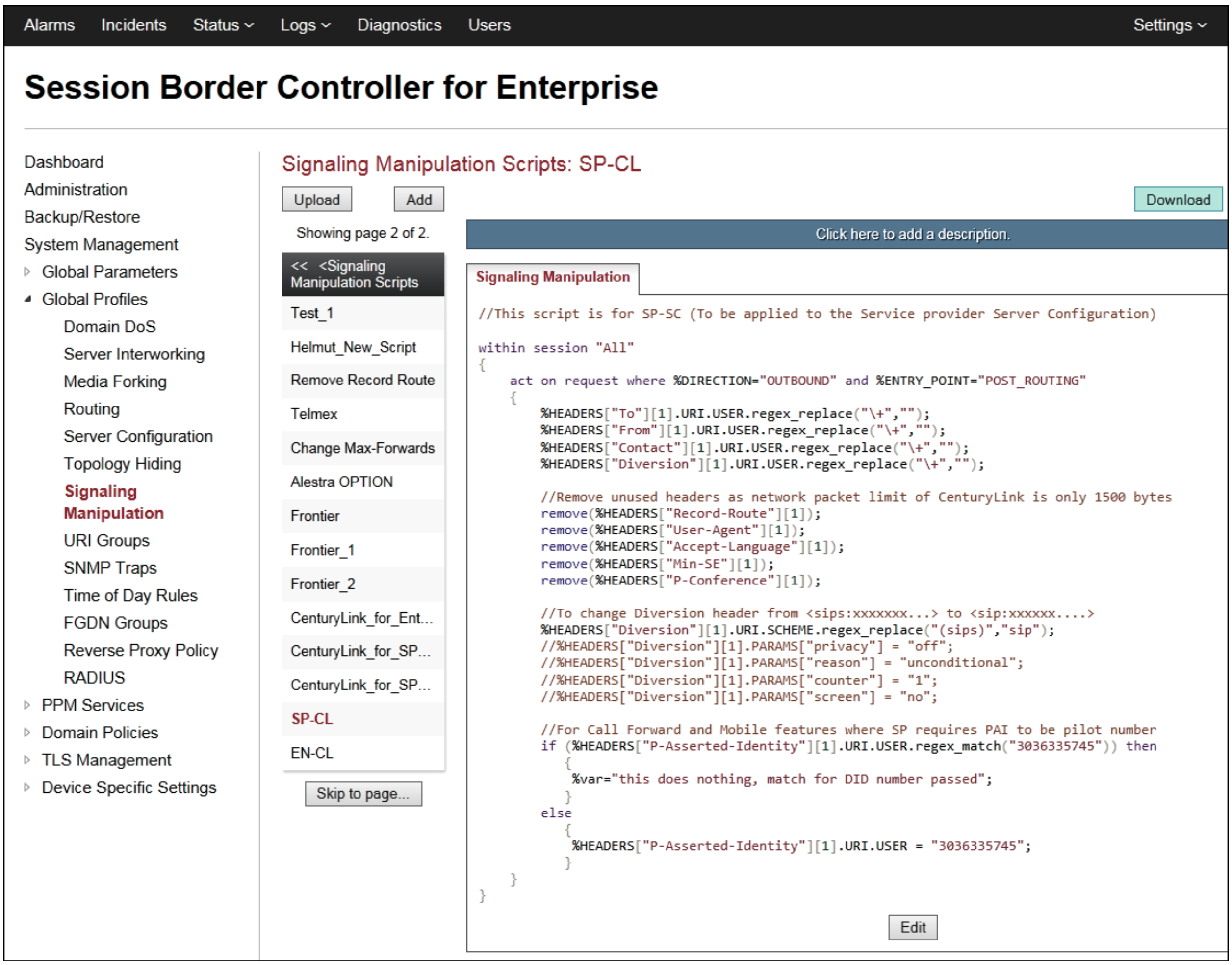
7.3.3. Signaling Manipulation

Signaling Manipulation feature allows the ability to add, change and delete any of the headers in a SIP message. This feature adds the ability to configure such manipulation in a highly flexible manner using a proprietary scripting language called **SigMa**.

To create a Signaling Manipulation script to be used toward the Service Provider direction, on the left navigation pane, select **Global Profiles → Signaling Manipulation**. From the **Signaling Manipulation Scripts** list, select **Add**.

- For **Title** enter a name, the name ***SP-CL*** was chosen in this example.
- Copy the complete script from **Appendix A**.
- Click **Save**.

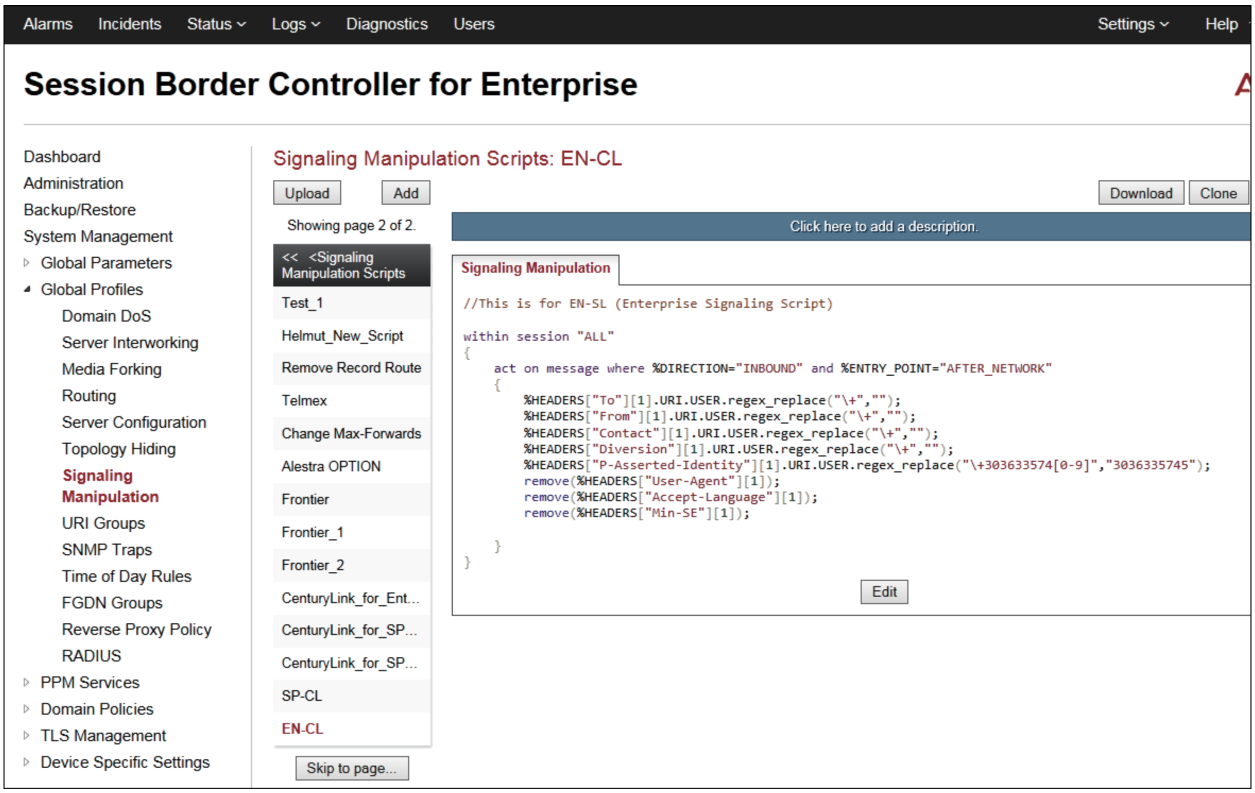
In the compliance testing, the SigMa **SP-CL** script was created for the Service Provider direction, details are captured below.



To create a Signaling Manipulation script to be used toward the enterprise direction, on the left navigation pane, select **Global Profiles → Signaling Manipulation**. From the **Signaling Manipulation Scripts** list, select **Add**.

- For **Title** enter a name, the name ***EN-CL*** was chosen in this example.
- Copy the complete script from **Appendix A**.
- Click **Save**.

In the compliance testing, the SigMa **EN-CL** script was created for the enterprise direction, details are captured below.



7.3.4. Server Configuration

The Server Configuration screen contains tabs: **General**, **Authentication**, **Heartbeat, Ping** and **Advanced**. These tabs are used to configure and manage various SIP Call Server specific parameters such as TCP and UDP port assignments, heartbeat signaling parameters, DoS security statistics and trusted domains. No configuration of **Authentication and Ping** is required.

To create a Server Configuration entry, select **Global Profiles →Server Configuration**. Click on the **Add** button.

In the compliance testing, two separate Server Configurations were created, server entry **SP-SC** for SP and server entry **EN-SC** for EN.

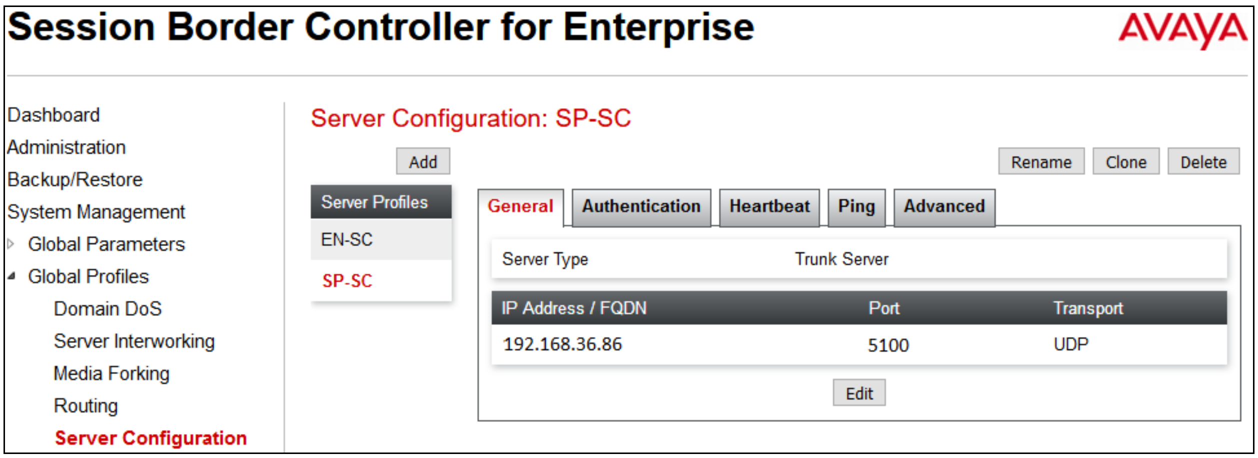
Server Configuration for SP

Server Configuration named **SP-SC** was created for SP.

General tab:

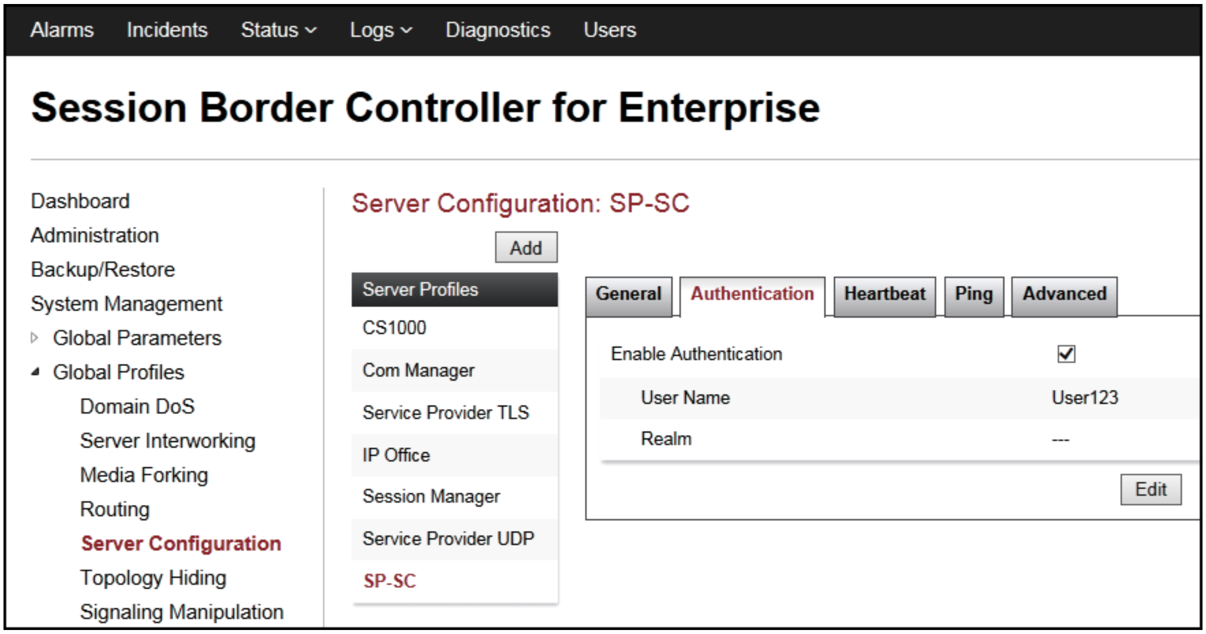
- Enter **Profile Name *SP-SC*** and click **Next** button.
- Set **Server Type** for SP as ***Trunk Server***.
- Enter the **IP Addresss/FQDN** of the SP SIP Proxy server (provided by the SP).
- In the compliance testing, SP supported ***UDP*** and listened on port ***5100***.
- Others are kept at default.

The Server Configuration profile is shown below.



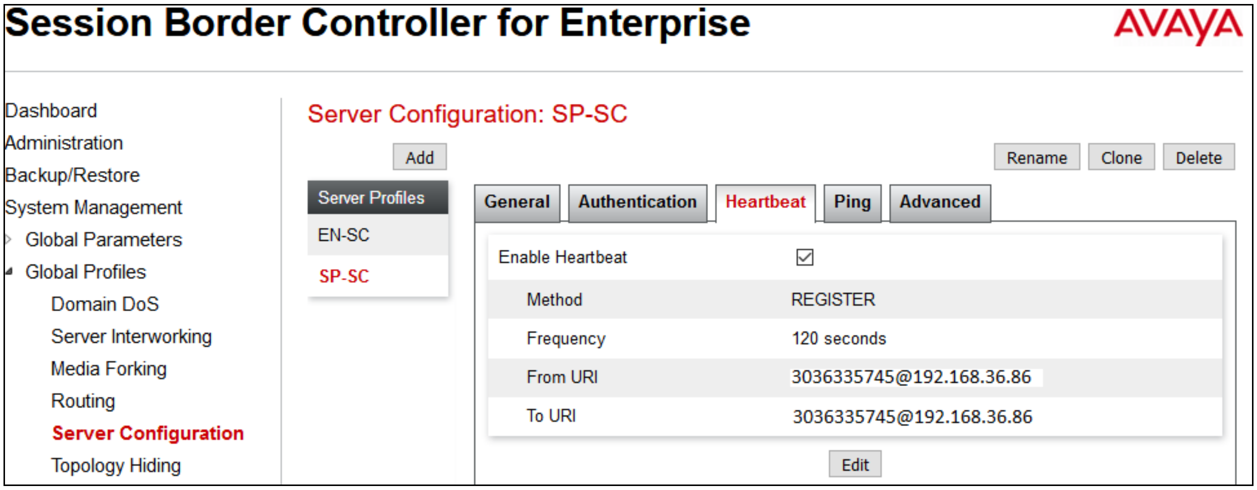
Authentication tab:

- Check **Enable Authentication** check box.
- Enter ***User Name/Trunk Group SIP ID*** provided by the SP for SIP trunk registration purpose.
- Leave ***Realm*** blank.
- Enter the ***Password*** and ***Confirm Password*** provided by the SP for SIP trunk registration purpose.



Heartbeat tab:

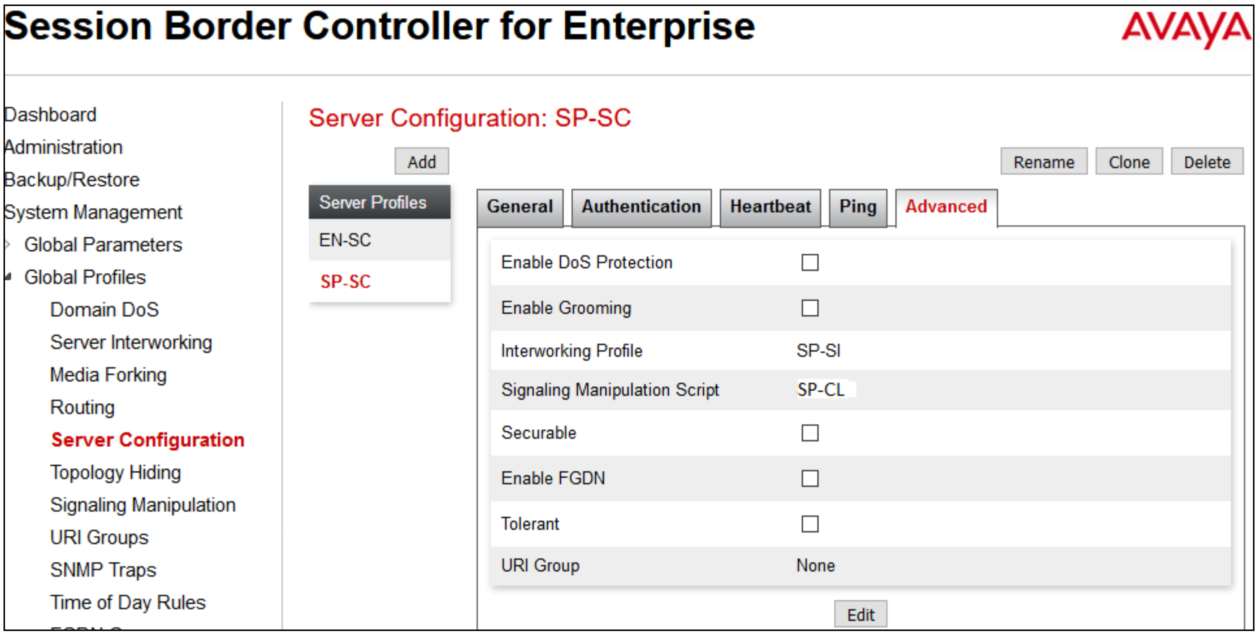
- Check **Enable Heartbeat** check box.
- Select ***REGISTER*** for **Method**.
- Enter ***120 seconds*** for **Frequency**.
- Enter ***3036335745@192.168.36.86*** for **From URI and To URI** fields, consisting of the pilot TN and the SP SIP Proxy IP address, this information should be provided by the SP.



Advanced tab:

Click on the **Edit** button and enter following information.

- **Interworking Profile** drop down list, select ***SP-SI*** as defined in **Section 7.3.2**.
- **Signaling Manipulation Script** drop down list, select ***SP-CL*** as defined in **Section 7.3.3.**
- The other settings are kept as default.

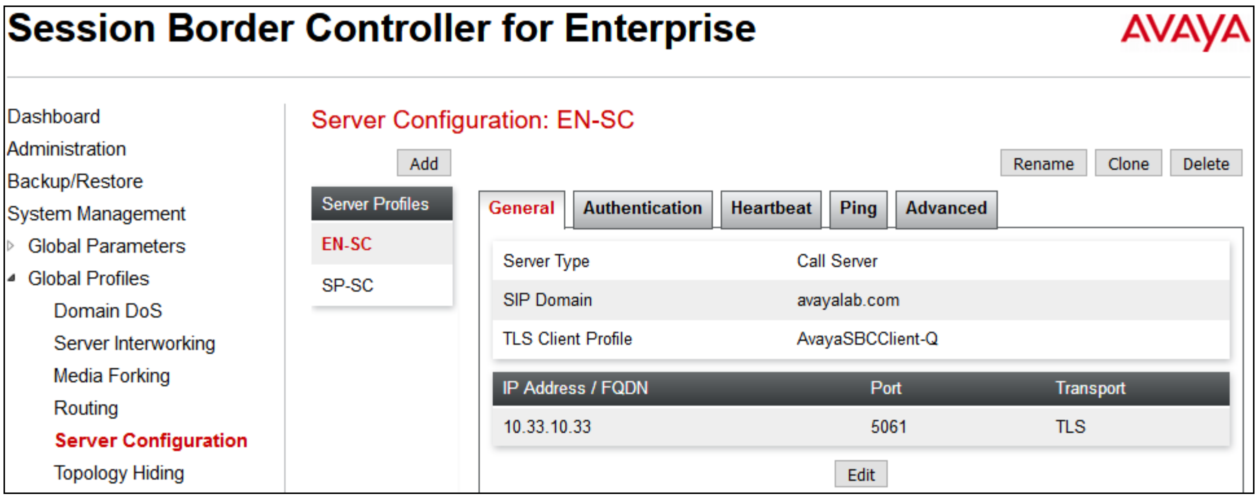


Server Configuration for EN

Server Configuration named **EN-SC** created for EN is discussed in detail below. **General** and **Advanced** tabs are provisioned but no configuration is done for **Authentication and Ping** tab. The **Heartbeat** tab is kept as ***disabled*** as default to allow the Avaya SBCE to forward the OPTIONS heartbeat from the SP to the EN to query the status of the SIP trunk.

General tab:

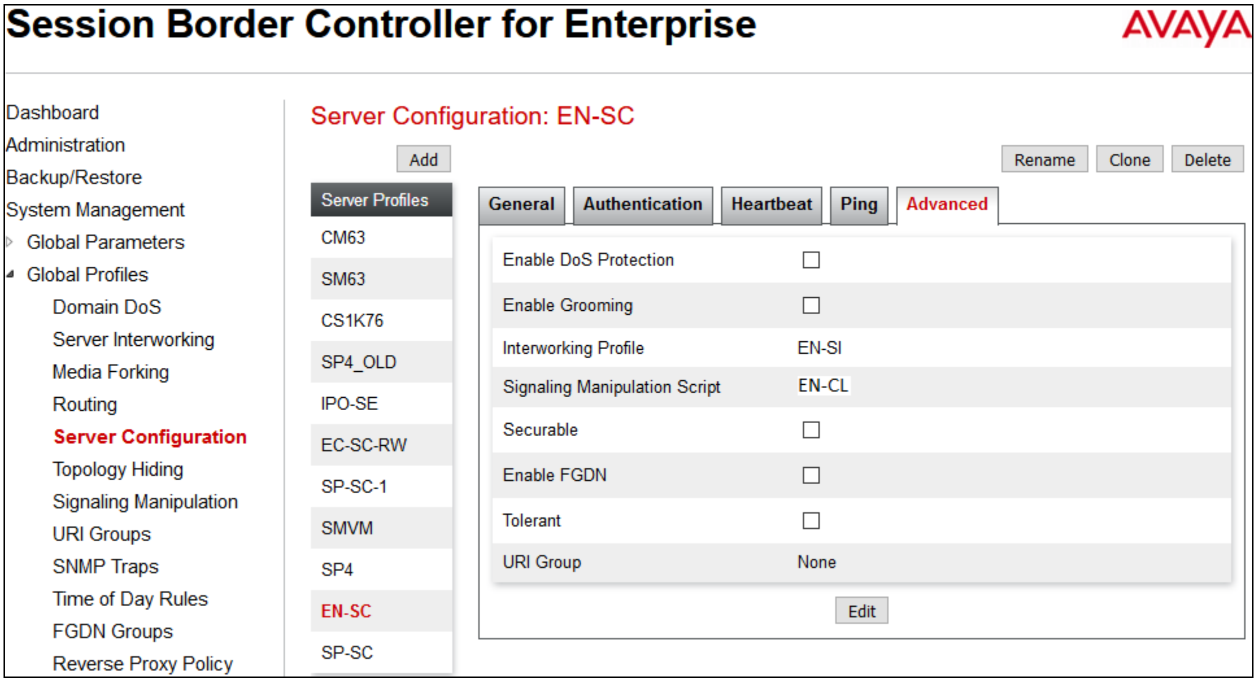
- Enter **Profile Name** as ***EN-SC*** and click **Next** button.
- Select the **Server Type** for EN as ***Call Server***.
- Select ***AvayaSBCClient-Q*** for **TLS Client Profile.**
- The **IP Address/FQDN** is the Session Manager IP address.
- **Transport**, the link between the Avaya SBCE and EN was ***TLS***.
- Listened on **Port *5061***.
- Other fields are kept at defaults.



Advanced tab:

Click on the **Edit** button to enter the following information.

- **Interworking Profile** drop down list select ***EN-SI*** as defined in **Section** Error! Reference source not found.**.**
- **Signaling Manipulation Script** drop down list select ***EN-CL***.
- The other settings are kept as default.



7.3.5. Routing Profiles

Routing Profiles define a specific set of packet routing criteria that are used in conjunction with other types of domain policies to identify a particular call flow and thereby ascertain which security features will be applied to those packets. Parameters defined by Routing Profiles include packet transport settings, name server addresses and resolution methods, next hop routing information and packet transport types.

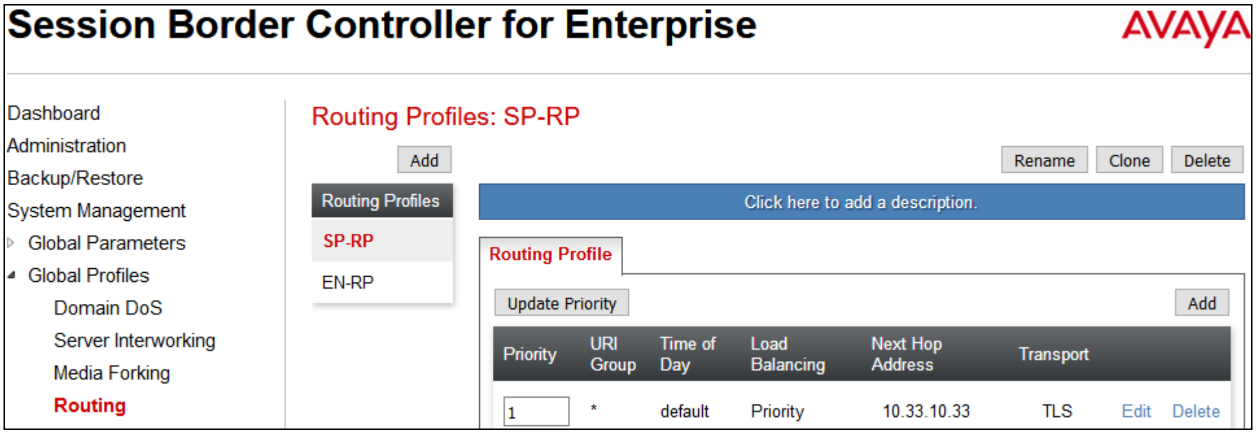
To create a Routing Profile, select **Global Profiles → Routing**. Click on the **Add** button.

In the compliance testing, a Routing Profile **EN-RP** was created to use in conjunction with the server flow defined for EN. This entry is to route the outbound call from the enterprise to the service provider.

In the opposite direction, a Routing Profile named **SP-RP** was created to be used in conjunction with the server flow defined for SP. This entry is to route the inbound call from the service provider to the enterprise.

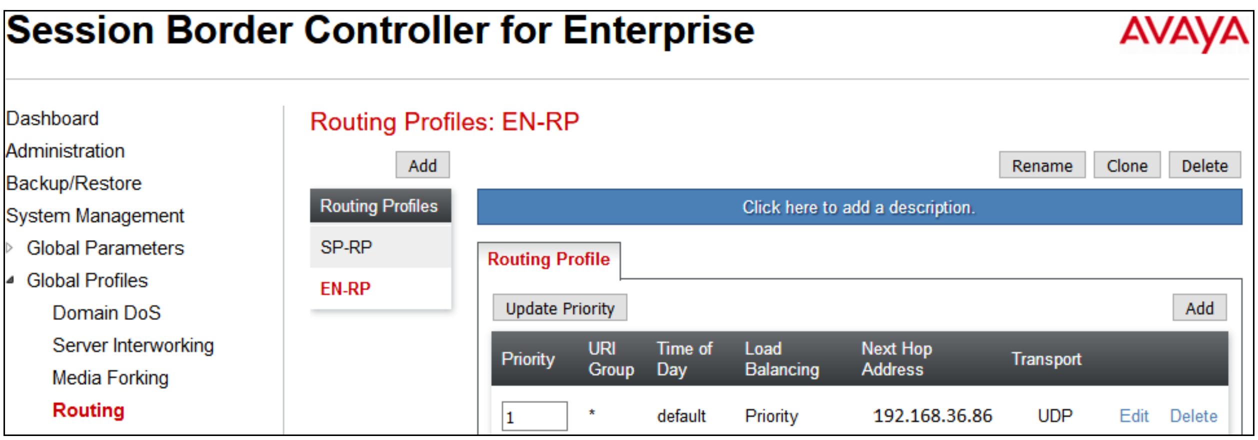
Routing Profile for SP

The screenshot below illustrates the routing profile from SP to Avaya network, **Global Profiles → Routing**: **SP-RP**. If there is a match in the "To" or "Request URI" headers with the URI Group "*" as described in **Section** Error! Reference source not found., the call will be routed to the **Next Hop Address** which is the IP address of Session Manager. As shown in **Figure 1**, the SIP trunk between EN and the Avaya SBCE is connected with transport protocol ***TLS***.



Routing Profile for EN

The Routing Profile for EN to SP, **EN-RP**, was defined to route call where the "To" header matches the URI Group **SP** defined in **Section** Error! Reference source not found. to **Next Hop Address** which is the IP address of SP SIP proxy as a destination. As shown in **Figure 1**, the SP SIP trunk is connected with transport protocol ***UDP***.



7.3.6. Topology Hiding

Topology Hiding is an Avaya SBCE security feature which allows changing certain key SIP message parameters to 'hide' or 'mask' how the enterprise network may appear to an unauthorized or malicious user.

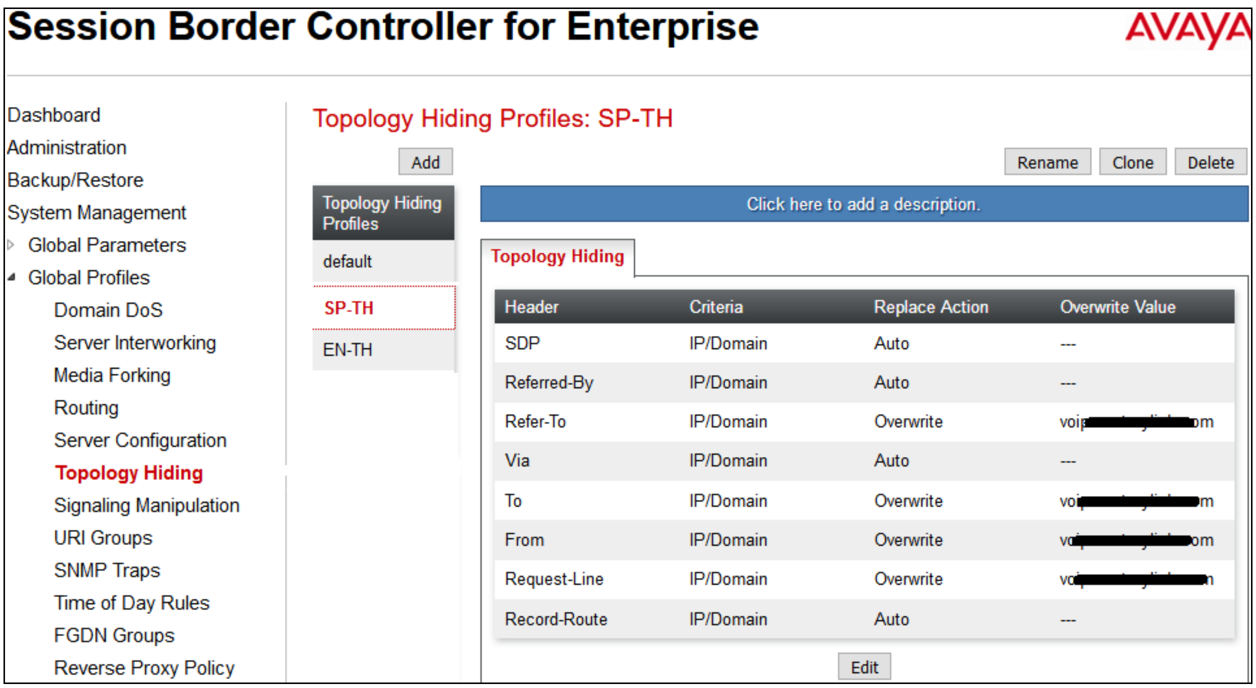
To create a Topology Hiding profile, select **Global Profiles → Topology Hiding**. Click on the **Add** button.

In the compliance testing, two Topology Hiding profiles **EN-TH** and **SP-TH** were created.

Topology Hiding Profile for SP

Profile **SP-TH** was defined to mask the enterprise in the "Refer-To", "Request-Line", "From" and "To" headers with the SP provided full qualified domain name (FQDN). This is done to secure the enterprise network topology and to meet the SIP requirement of the service provider. The **Criteria** should be selected as **IP/Domain** and the **Replace Action** as **Overwrite**.

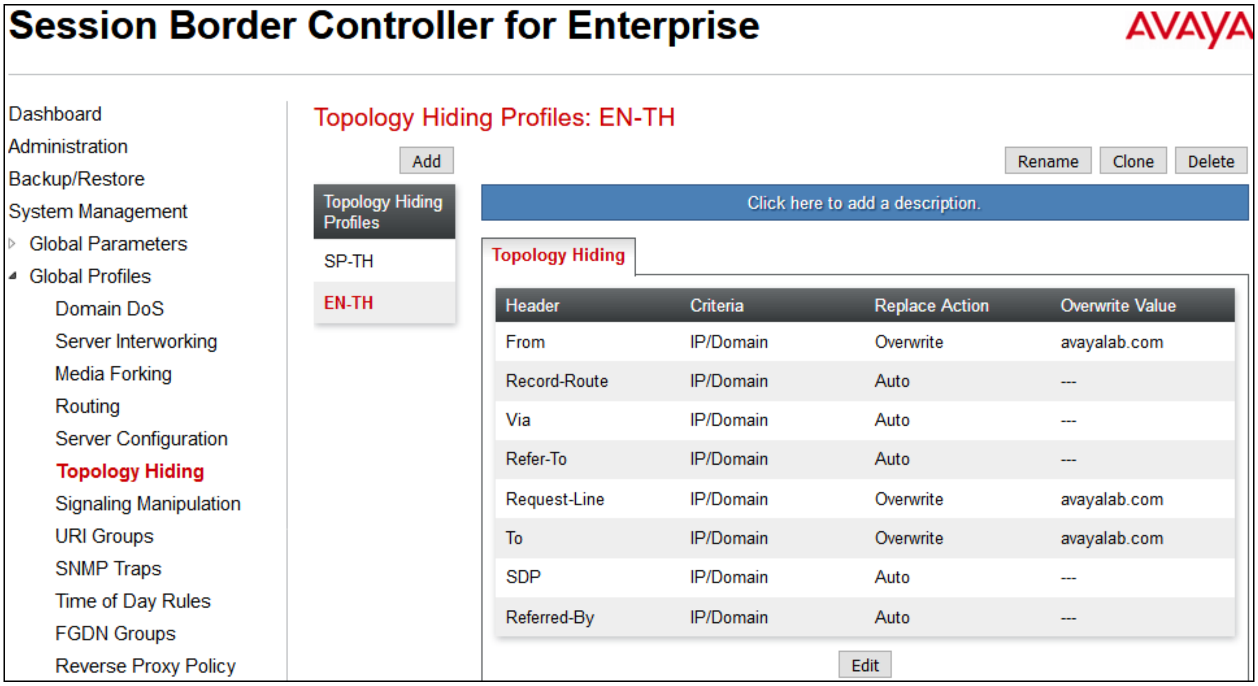
The screenshots below illustrate the Topology Hiding profile **SP-TH**.



Topology Hiding Profile for EN

Profile **EN-TH** was also created to mask SP URI-Host in "Request-Line", "From" and "To", headers with the enterprise domain ***avayalab.com***. The **Criteria** should be selected as **IP/Domain** and the **Replace Action** as **Overwrite**.

The screenshots below illustrate the Topology Hiding profile **EN-TH**.



7.4. Domain Policies

Domain Policies configures various rule sets (policies) to control unified communications based upon criteria of communication sessions originating from or terminating at the enterprise. These criteria can be used to trigger policies which, in turn, activate various security features of the Avaya SBCE security device to aggregate, monitor, control and normalize call flow. There are default policies available for use, or a custom domain policy can be created.

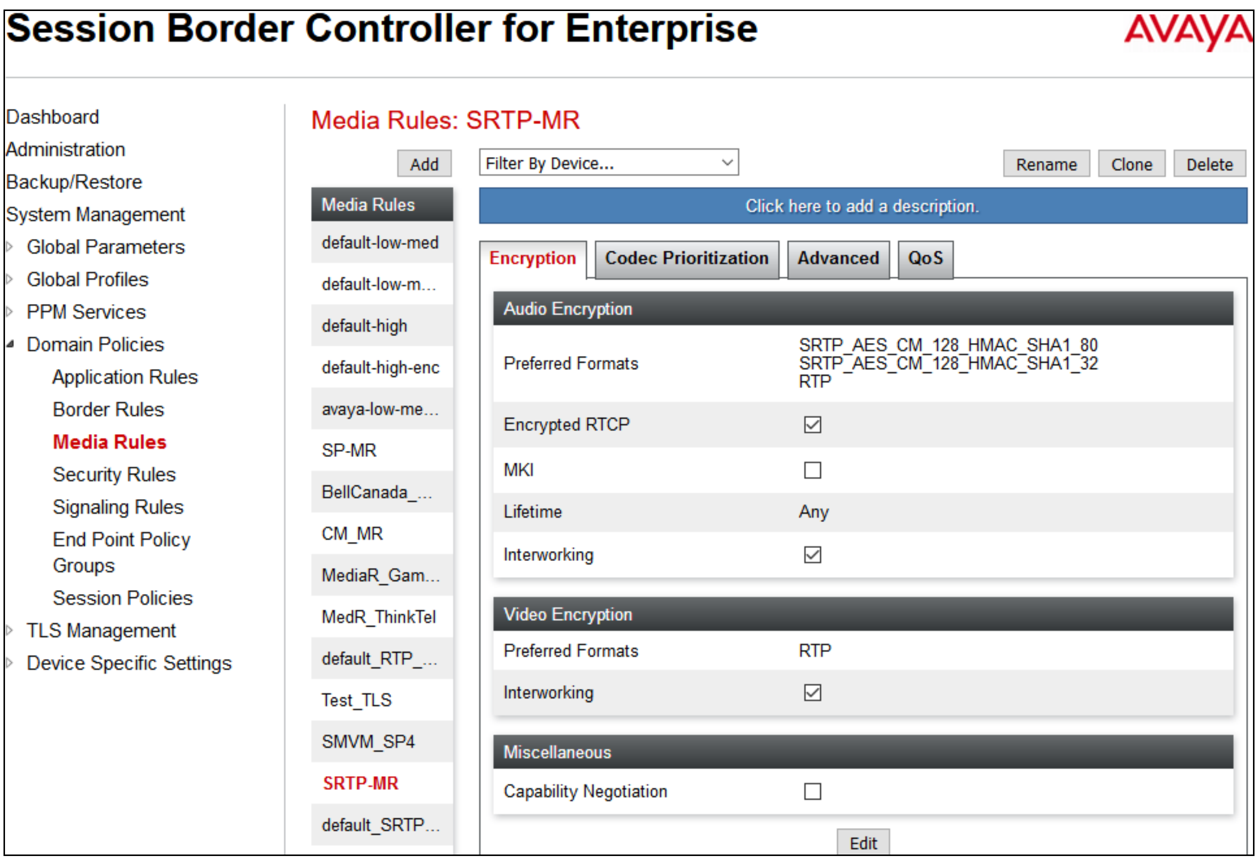
7.4.1. Media Rules

Media rules can be used to define RTP media packet parameters, such as prioritizing encryption techniques and packet encryption techniques. Together these media-related parameters define a strict profile that is associated with other SIP-specific policies. You can also define how Avaya SBCE must handle media packets that adhere to the set parameters.

To clone a Media Rule, navigate to **Domain Policies → Media Rules**. With ***default-low-med*** rule chosen, click on the **Clone** button.

Media Rules for EN

In this compliance testing, Secure Real-Time Transport Protocol (SRTP, media encryption) is used within enterprise network only. Therefore, it is necessary to create a media rule to apply to the internal interface of Avaya SBCE and EN. The created **SRTP-MR** rule is shown below.



Media Rules for SP
In the compliance test, the media rule used for the service provider was ***default-low-med***.

7.4.2. Signaling Rules

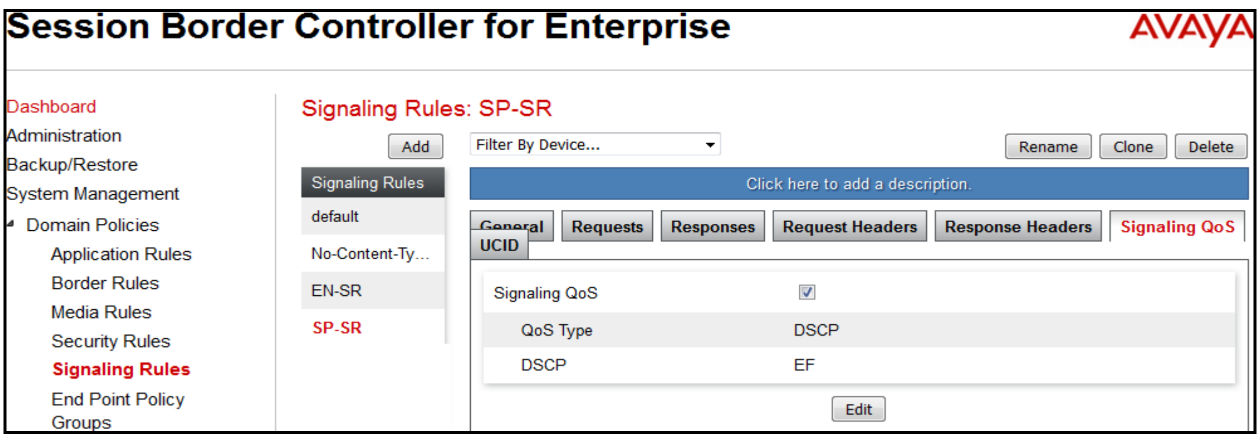
Signaling Rules define the action to be taken (Allow, Block, Block with Response, etc.) for each type of SIP-specific signaling request and response message. When SIP signaling packets are received by the Avaya SBCE, they are parsed and "pattern-matched" against the particular signaling criteria defined by these rules. Packets matching the criteria defined by the Signaling Rules are tagged for further policy matching.

To clone a Signaling Rule, navigate to **Domain Policies → Signaling Rules**. With the **default** rule chosen, click on the **Clone** button.

Signaling Rules for SP

In the compliance testing, created signaling rule **SP-SR** is discussed below. All the tabs are kept as default values except the **Signaling QoS** tab.

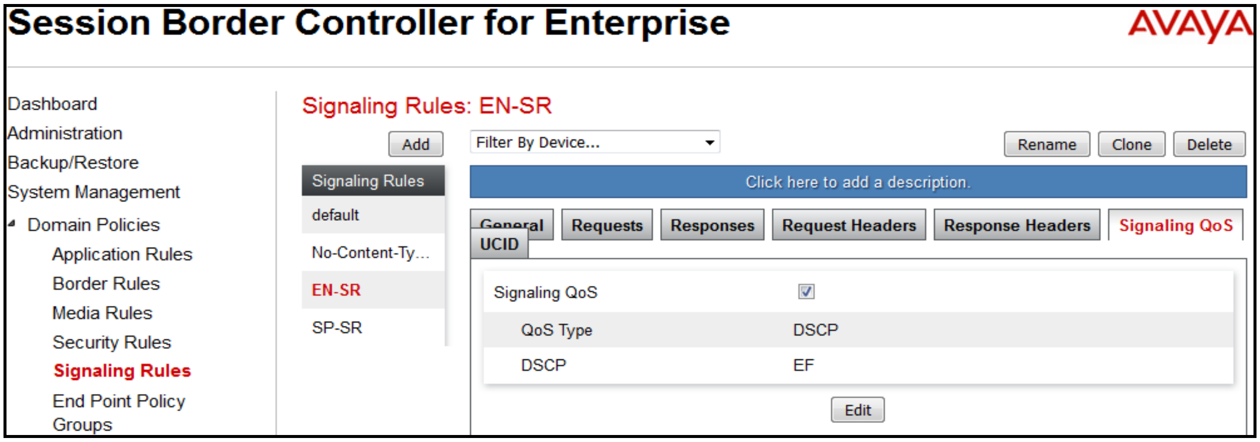
In the **Signaling QoS** tab, click on **Edit** button then check on checkbox. Then select ***EF*** value for **DSCP** option.



Signaling Rules for EN

In the compliance testing, created signaling rule **EN-SR** is discussed below. All the tabs are kept as default values except **Signaling QoS** tab.

In **Signaling QoS** tab, click on **Edit** button then check on checkbox. Then select ***EF*** value for **DSCP** option.



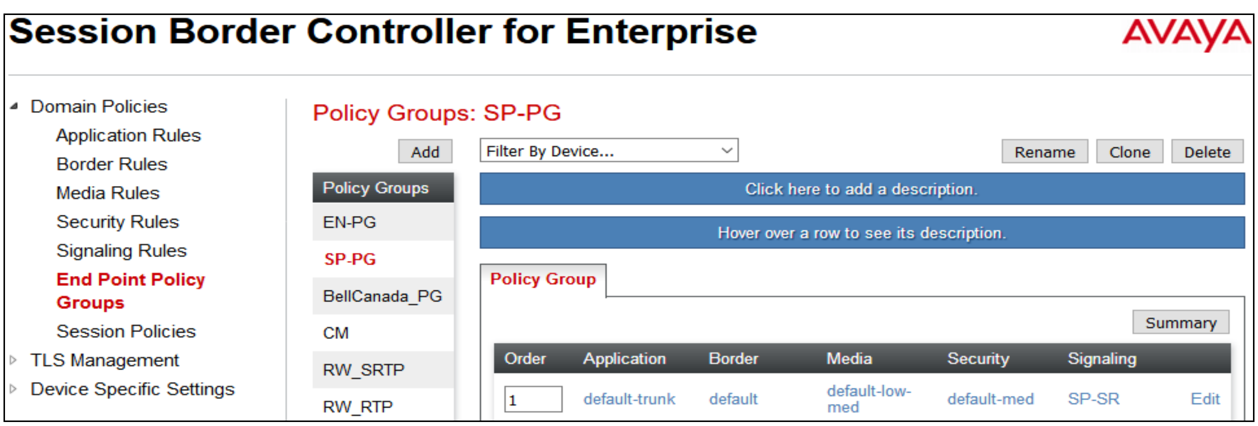
7.4.3. Endpoint Policy Groups

The rules created within the **Domain Policies** section are assigned to an **Endpoint Policy Group**. The **Endpoint Policy Group** is then applied to a **Server Flow** defined in the next section. Endpoint Policy Groups were created for SP and EN. To create a new policy group, navigate to **Domain Policies → Endpoint Policy Groups** and click on **Add**.

Endpoint Policy Group for SP

The following screen shows **SP-PG** created for SP:

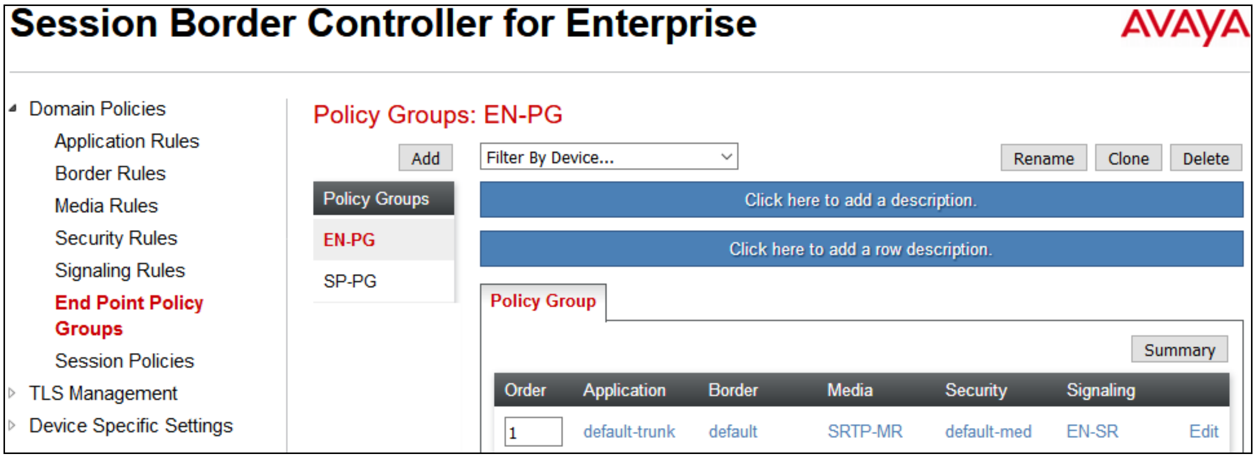
- Set Application Rule to ***default-trunk***.
- Set Border Rule to ***default***.
- Set Media Rule to ***default-low-med*** as created in **Section 7.4.1**.
- Set Security Rule to ***default-med***
- Set Signaling Rule to ***SP-SR*** as created in **Section 7.4.2**.



Endpoint Policy Group for EN

The following screen shows **EN-PG** created for EN:

- Set Application Rule to ***default-trunk***.
- Set Border Rule to ***default***.
- Set Media Rule to ***SRTP-MR*** as created in **Section 7.4.1**.
- Set Security Rule to ***default-med.***
- Set Signaling Rule to ***EN-SR*** as created in **Section 7.4.2**.



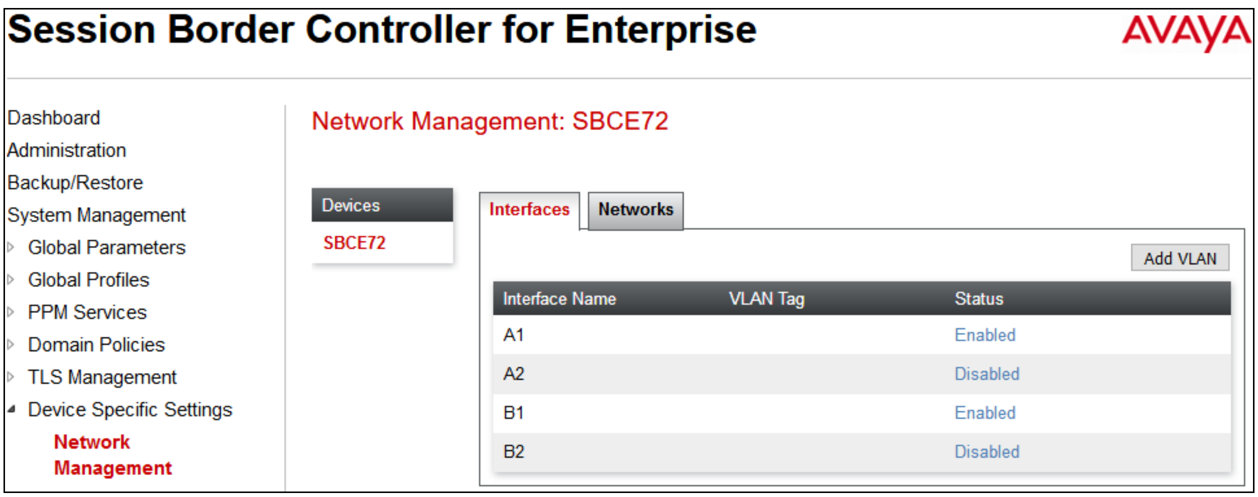
7.5. Device Specific Settings

Device Specific Settings allows aggregate system information to be viewed and various device-specific parameters to be managed to determine how a particular device will function when deployed in the network. Specifically, it gives the ability to define and administer various device-specific protection features such as Message Sequence Analysis (MSA) functionality and protocol scrubber rules, end-point and session call flows, as well as the ability to manage system logs and control security features.

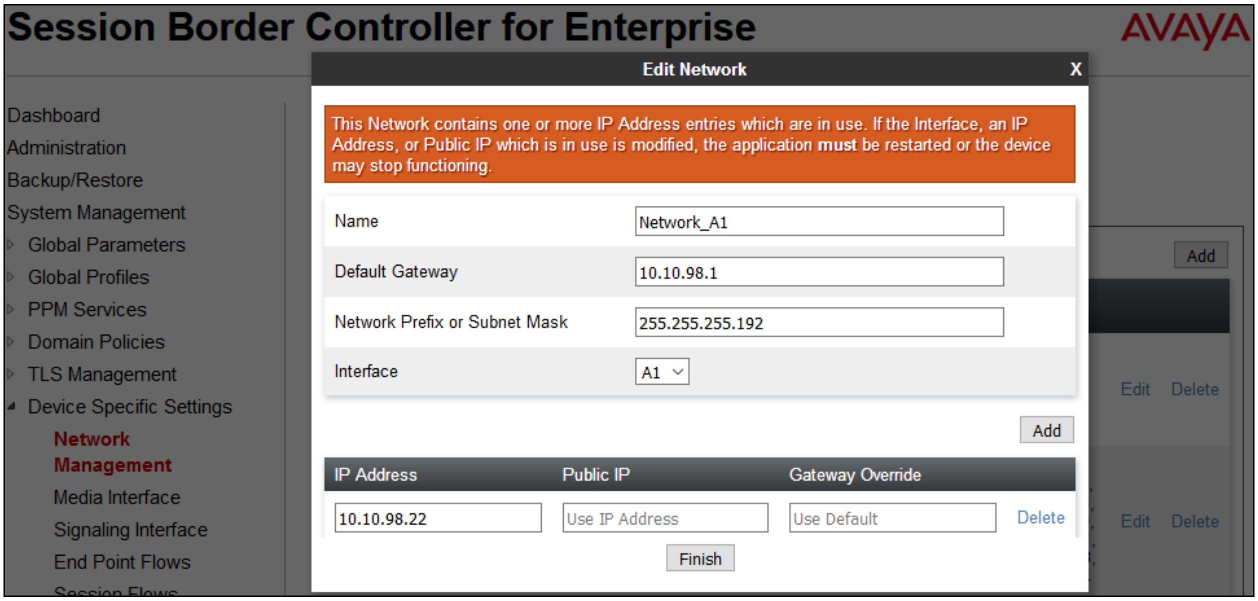
7.5.1. Network Management

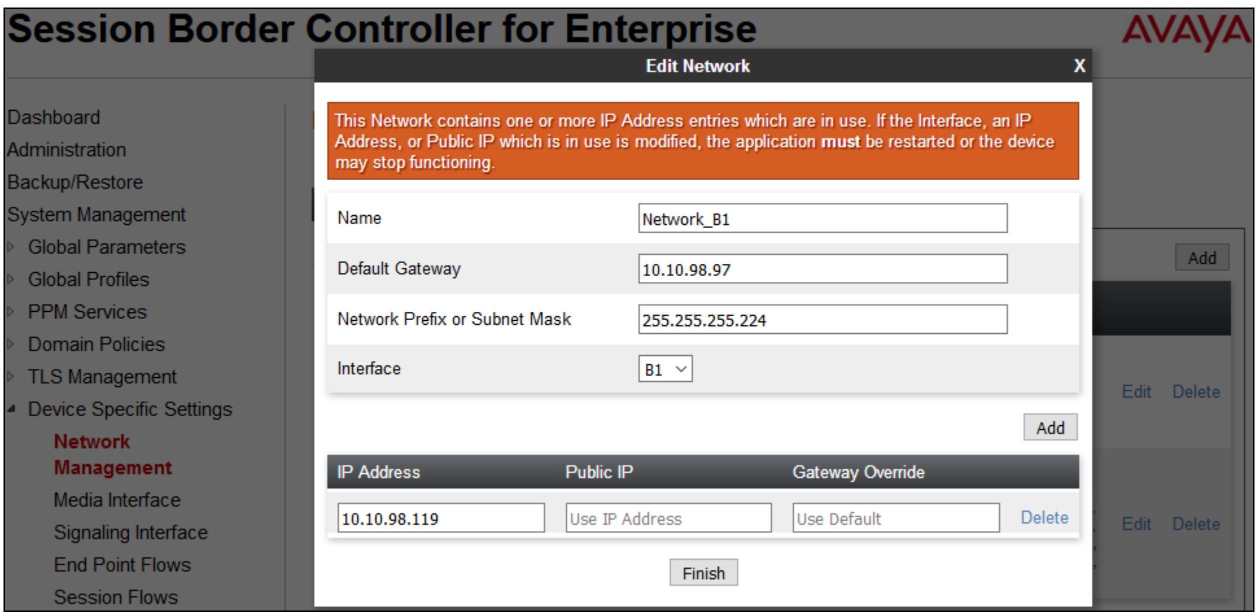
The Network Management screen is where the network interface settings are configured and enabled. During the installation process of the Avaya SBCE, certain network-specific information was defined such as; device IP address(es), public IP address(es), netmask, gateway, etc., to interface the device to the network. This information populates the **Network Management** tab, which can be edited as needed to optimize device performance and network efficiency.

Enable the interfaces used to connect to the inside and outside networks on the **Interface** tab. The following screen shows **Interface Names**, **A1** and **B1** are **Enabled**. To enable an interface, click on its **Status** corresponding to the interface names.



Navigate to **Device Specific Settings → Network** and under the **Network Configuration** tab verify the IP addresses assigned to the interfaces. The following screens show the private interface is assigned to **A1** and the public interface is assigned to **B1** respectively.





7.5.2. Media Interface

The Media Interface screen is where the media ports are defined. The Avaya SBCE will open a connection for RTP on the defined ports.

To create a new Media Interface, navigate to **Device Specific Settings → Media Interface** and click **Add**.

Separate Media Interfaces were created for both inside and outside interfaces. The following screen shows the Media Interfaces created in the compliance testing.

Note: After the media interfaces are created, an application restart is necessary before the changes will take effect.

7.5.3. Signaling Interface

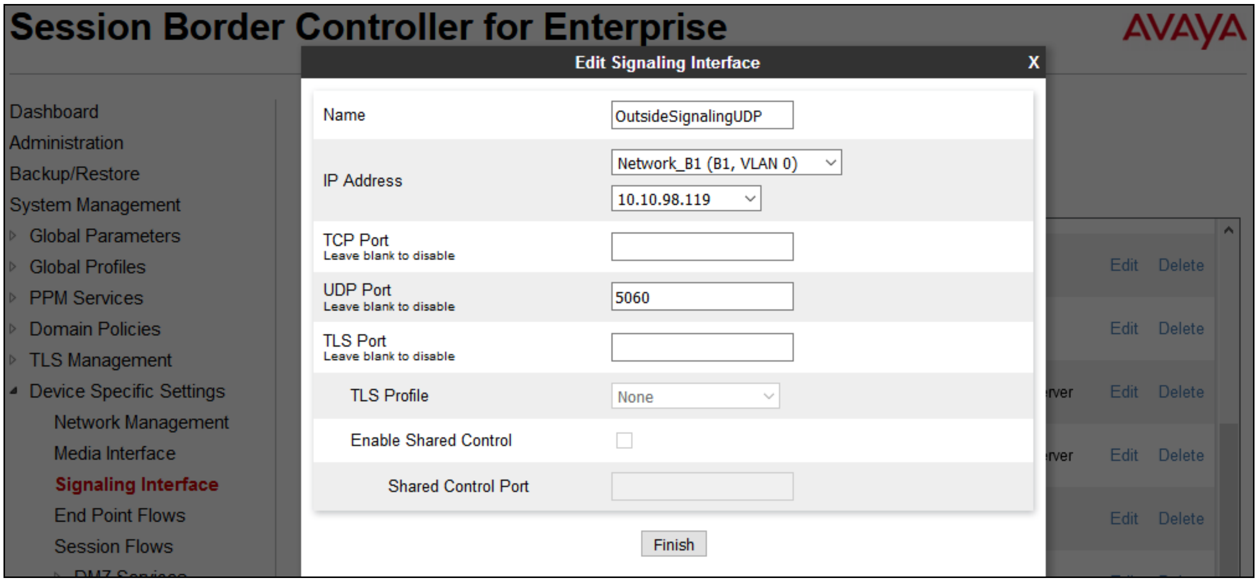
The Signaling Interface screen is where the SIP signaling port is defined. The Avaya SBCE will listen for SIP requests on the defined port.

To create a new Signaling Interface, navigate to **Device Specific → Settings → Signaling Interface** and click **Add**.

Separate Signaling Interfaces were created for both inside and outside interfaces.

Signaling Interface for SP

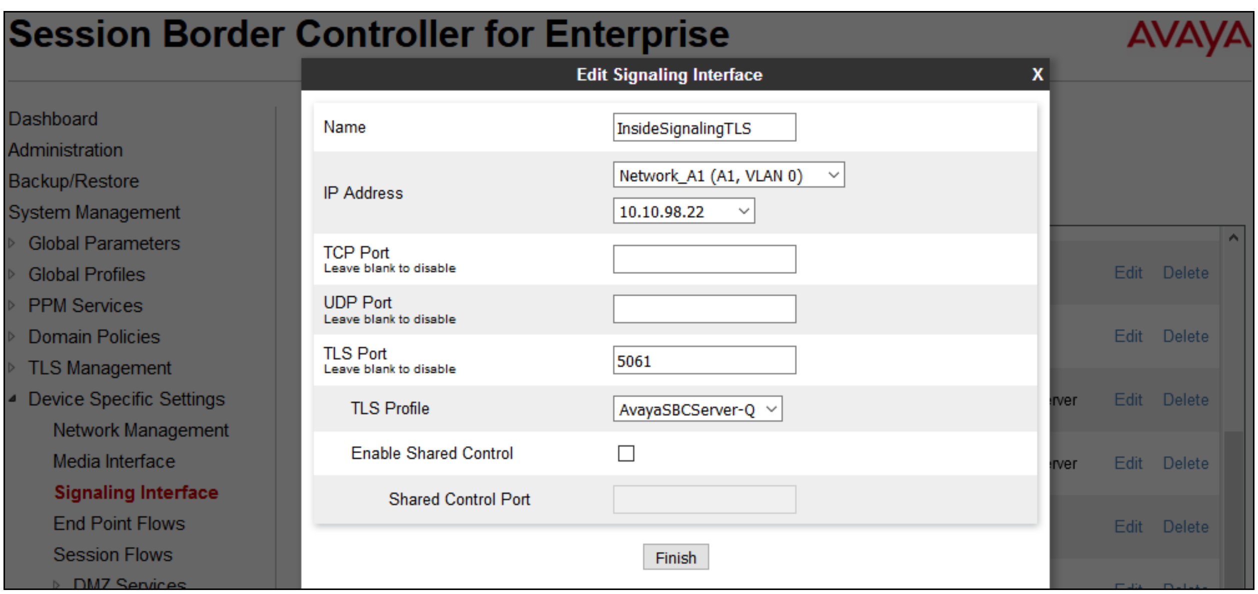
The outside interface to the service provider is created with UDP port 5060 as shown below.



Signaling Interface for EN

The inside interface to the service provider interface is created with TLS port 5061 as shown below.

- Enter descriptive name for **Name** field.
- Select **IP Address** from pull down menu defined as internal network interface **Section 7.5.1**.
- Specified ***5061*** for **TLS Port**. Then select **TLS profile** from pull down menu as defined in **Section 7.2.3**.
- Click **Finish**.



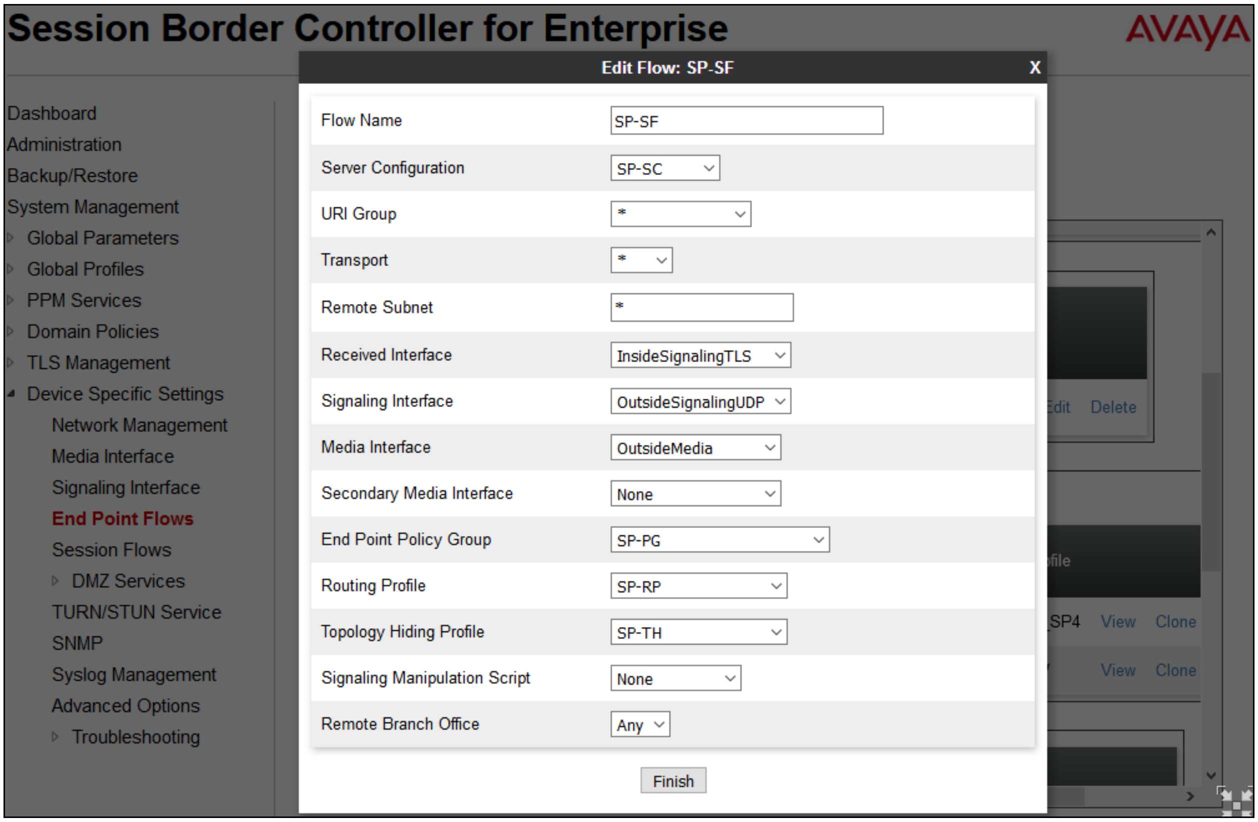
7.5.4. End Point Flows - Server Flow

When a packet is received by the Avaya SBCE, the content of the packet (IP addresses, URIs, etc.) is used to determine which flow it matches. Once the flow is determined, the flow points to a policy which contains several rules concerning processing, privileges, authentication, routing, etc. Once routing is applied and the destination endpoint is determined, the policies for this destination endpoint are applied. The context is maintained, so as to be applied to future packets in the same flow. The following screens illustrate the flow through the Avaya SBCE to secure a SIP Trunk call.

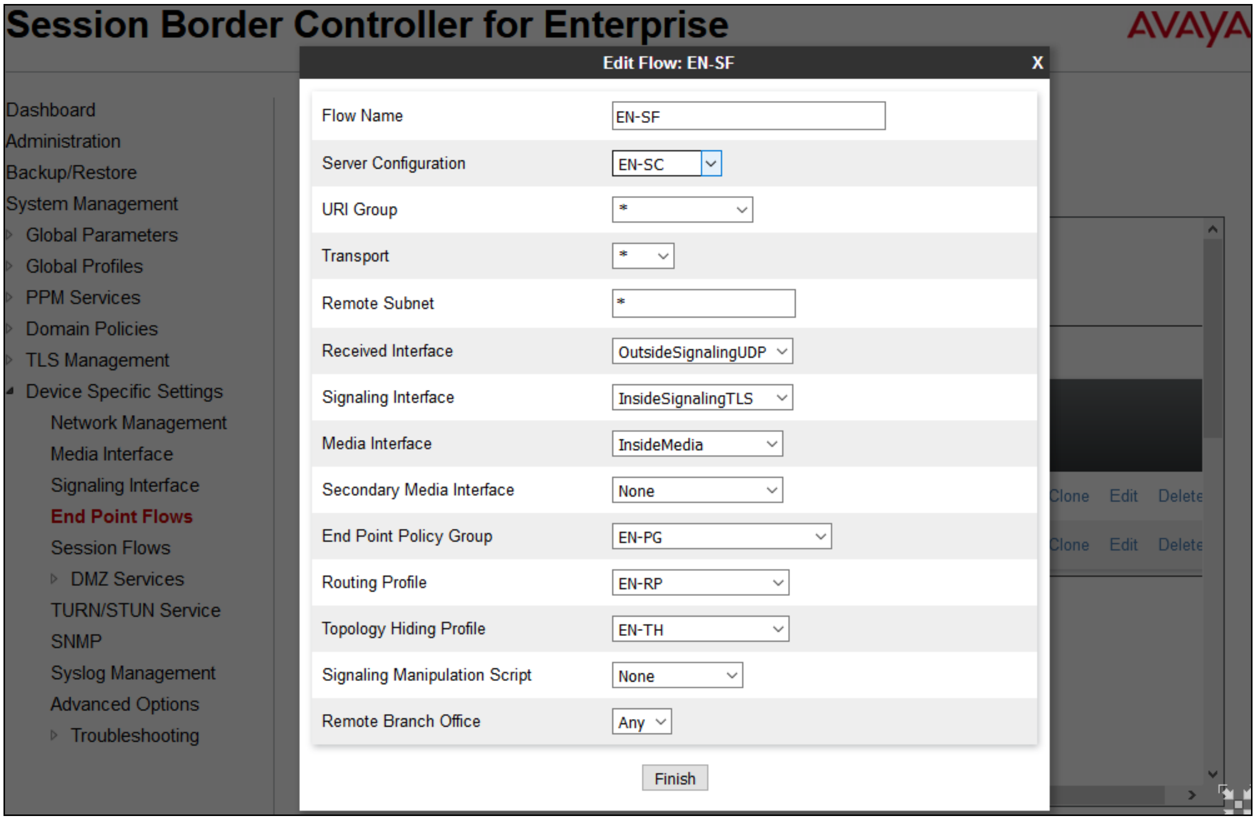
In the compliance testing, separate Server Flows were created for SP and EN. To create a Server Flow, navigate to **Device Specific Settings → End Point Flows**. Select the **Server Flows** tab and click **Add** (not shown). In the new window that appears, enter the following values. The other fields are kept default.

- **Flow Name**: Enter a descriptive name.
- **Server Configuration**: Select a Server Configuration created in **Section 7.3.4** to assign to the Flow.
- **URI Group**: Select the URI Group created in **Section 7.3.1** to assign to the Flow. **Note**: URI Group can be set to "*" to match all calls.
- **Received Interface**: Select the Signaling Interface created in **Section** Error! Reference source not found. that the Server Configuration is allowed to receive SIP messages from.
- **Signaling Interface**: Select the Signaling Interface created in **Section** Error! Reference source not found. used to communicate with the Server Configuration.
- **Media Interface**: Select the Media Interface created in **Section** Error! Reference source not found. used to communicate with the Server Configuration.
- **End Point Policy Group**: Select the End Point Policy Group created in **Section 7.4.3** to assign to the Server Configuration.
- **Routing Profile**: Select the Routing Profile created in **Section** Error! Reference source not found. that the Server Configuration will use to route SIP messages to.
- **Topology Hiding Profile**: Select the Topology-Hiding profile created in **Section** Error! Reference source not found. to apply to the Server Configuration.
- Click **Finish**.

The following screen shows the Server Flow **SP-SF** configured for SP.



Similarly, the following screen shows the Server Flow **EN-SF** configured for EN.



8. Service Provider Configuration

CenturyLink is responsible for the configuration of its SIP Trunking Service. The customer will need to provide the IP address used to reach the Avaya SBCE at the enterprise. CenturyLink will provide the customer with the necessary information to configure the SIP connection from the enterprise to CenturyLink. The information provided by CenturyLink includes:

- CenturyLink's SIP proxy IP address and UDP port number. For the compliance test UDP port number 5100 was used.
- CenturyLink's full qualified domain name (FQDN).
- Supported codecs.
- TNs and Pilot number.
- SIP Trunk registration credentials (user name/trunk group SIP ID and password).

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 useful troubleshooting commands.

9.1. Verification Steps

- Verify that endpoints at the enterprise site can place calls to PSTN and that the call remains active for more than 35 seconds. This time period is included to satisfy SIP protocol timers.
- Verify that endpoints at the enterprise site can receive calls from PSTN and that the call can remain active for more than 35 seconds. This time period is included to satisfy SIP protocol timers.
- Verify that the user on 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. Protocol Traces

The following SIP headers are inspected using Wireshark trace analysis:

- Request-URI: verify the called party number and SIP domain.
- From: verify the calling party name and number.
- To: verify the called party name and number.
- P-Asserted-Identity: verify the calling party name and number.
- Privacy: verify the value "user" and/or "id" presents the private call scenario.

The following attributes in SIP message body are inspected using Wireshark trace analysis:

- Connection Information (c line): verify IP address of near end and far end endpoints.
- Time Description (t line): verify session timeout value of near end and far end endpoints.
- Media Description (m line): verify audio port, codec, DTMF event description.
- Media Attribute (a line): verify specific audio port, codec, ptime, send/ receive ability, DTMF event and fax attributes.

9.3. Troubleshooting:

9.3.1. The Avaya SBCE

Use Avaya SBCE trace tool, traceSBC to monitor the SIP signaling messages between the Service provider and the Avaya SBCE.

9.3.2. Communication Manager

- **list trace station** <extension number>. Traces call to and from a specific station.
- **list trace tac** <trunk access code number>. Trace call over a specific trunk group.
- **status station** <extension number>. Displays signaling and media information for an active call on a specific station.
- **status trunk** <trunk group number>. Displays trunk group information.
- **status trunk** <trunk group number/channel number>. Displays signaling and media information for an active trunk channel.

10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Communication Manager 7.1.2, Avaya Aura® Session Manager 7.1.2 and the Avaya Session Border Controller for Enterprise 7.2.1 to CenturyLink SIP Trunking service on Perimeta/BroadWorks platform using UDP. The CenturyLink SIP Trunking service on Perimeta/BroadWorks platform is a SIP-based Voice over IP solution for customers ranging from small to large the enterprises. CenturyLink SIP Trunking service on Perimeta/BroadWorks Platform provides a flexible, cost-saving alternative to traditional analog and ISDN-PRI trunks.

All of the test cases were executed. Despite the number of observations seen during testing as noted in **Section 2.2**, the test results met the objectives outlined in **Section 2.1**. The CenturyLink SIP Trunking service on Perimeta/BroadWorks platform is considered **compliant** with Avaya Aura® Communication Manager 7.1.2 Avaya Aura® Session Manager 7.1.2 and Avaya Session Border Controller for Enterprise 7.2.1.

11. References

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

[1] *What's New in Avaya Aura Release 7.1.1,* Release 7.1.1, Issue 2, August 2017.
[2] *Upgrading Avaya Aura® System Manager to Release 7.1.1*, Issue 2, August 2017.
[3] *Administering Avaya Aura® System Manager for Release 7.1.1*, Issue 5, August 2017.
[4] *Administering Avaya Aura® Session Manager for* Release 7.1.1, Issue 2, August 2017.
[5] *Deploying Avaya Aura Communication Manager in Virtualized Environment,* Release 7.1.1, Issue 2, August 2017.
[6] *Avaya Session Border Controller for Enterprise Overview and Specification*, Release 7.2, Issue 2, June 2017.
[7] *Deploying Avaya Session Border Controller for Enterprise,* Release 7.2, Issue 3, September 2017.
[8] *Deploying Avaya Session Border Controller in Virtualized Environment,* Release 7.2, Issue 1, June 2017.
[9] *Administering Avaya Session Border Controller for Enterprise*, Release 7.2, January 2017.
[10] *Deploying and Updating Avaya Aura Media Server Appliance*, Release 7.8, Issue 3, August 2017.
[11] *9600 Series IP Deskphones Overview and Specification,* Release 7.1, June 2017.
[12] *Installing and Maintaining Avaya 9601/9608/9611G/9621G/9641G/9641GS IP Deskphones SIP*, Release 7.1, June 2017.
[13] *Administering Avaya 9601/9608/9611G/9621G/9641G/9641GS IP Deskphones SIP,* Release 7.1, June 2017.
[14] *Avaya Equinox™ Overview and Specification for Android, iOS, Mac, and Window, Release 3.0, January 2017.*
[15] *Administering Avaya one-X® Communicator,* Release 6.2, April 2015.
[16] *RFC 3261 SIP: Session Initiation Protocol,* http://www.ietf.org/
[17] *RFC 3515, The Session Initiation Protocol (SIP) Refer Method,* http://www.ietf.org/
[18] *RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals,* http://www.ietf.org/

Product documentation for CenturyLink SIP Trunking service on Perimeta/BroadWorks Platform is available from CenturyLink.

12. Appendix A: SigMa Scripts

Following are the Signaling Manipulation scripts (SigMa) that were used in the configuration of the Avaya SBCE. When adding this script as instructed in **Section 7.3.3** enter a name for the script in the Title (e.g., **SP-CL** or **EN-CL**) and copy/paste each individual script shown below. Note that some fields will need to be changed with the correct information, such as the pilot number “3036335745” this information should be provided by CenturyLink.

Title: SP-CL

```
//This script is for SP-SC (To be applied to the Service provider Server Configuration)

within session "All"
{
   act on request where %DIRECTION="OUTBOUND" and
%ENTRY_POINT="POST_ROUTING"
   {
      %HEADERS["To"][1].URI.USER.regex_replace("\+","");
      %HEADERS["From"][1].URI.USER.regex_replace("\+","");
      %HEADERS["Contact"][1].URI.USER.regex_replace("\+","");
      %HEADERS["Diversion"][1].URI.USER.regex_replace("\+","");

      //Remove unused headers as network packet limit of CenturyLink is only 1500 bytes
      remove(%HEADERS["Record-Route"][1]);
      remove(%HEADERS["User-Agent"][1]);
      remove(%HEADERS["Accept-Language"][1]);
      remove(%HEADERS["Min-SE"][1]);
      remove(%HEADERS["P-Conference"][1]);

      //To change Diversion header from <sips:xxxxxxx...> to <sip:xxxxxx....>
      %HEADERS["Diversion"][1].URI.SCHEME.regex_replace("(sips)","sip");
      //%HEADERS["Diversion"][1].PARAMS["privacy"] = "off";
      //%HEADERS["Diversion"][1].PARAMS["reason"] = "unconditional";
      //%HEADERS["Diversion"][1].PARAMS["counter"] = "1";
      //%HEADERS["Diversion"][1].PARAMS["screen"] = "no";

      //For Call Forward and Mobile features where SP requires PAI to be pilot number
      if (%HEADERS["P-Asserted-Identity"][1].URI.USER.regex_match("3036335745")) then
        {
         %var="this does nothing, match for DID number passed";
        }
      else
        {
         %HEADERS["P-Asserted-Identity"][1].URI.USER = "3036335745";
        }
   }
}
```

Title: EN-CL
//This is for EN-SC (To be applied to the Enterprise Server Configuration)

```
within session "ALL"
{
   act on message where %DIRECTION="INBOUND" and
%ENTRY_POINT="AFTER_NETWORK"
   {
      %HEADERS["To"][1].URI.USER.regex_replace("\+","");
      %HEADERS["From"][1].URI.USER.regex_replace("\+","");
      %HEADERS["Contact"][1].URI.USER.regex_replace("\+","");
      %HEADERS["Diversion"][1].URI.USER.regex_replace("\+","");
      %HEADERS["P-Asserted-Identity"][1].URI.USER.regex_replace("\+303633574[0-
9]","3036335745");
      remove(%HEADERS["User-Agent"][1]);
      remove(%HEADERS["Accept-Language"][1]);
      remove(%HEADERS["Min-SE"][1]);

   }
}
```

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.