**Avaya Solution & Interoperability Test Lab**

# Application Notes for Configuring Avaya Aura® Communication Manager Rel. 7.0, Avaya Aura® Session Manager Rel. 7.0 and Avaya Session Border Controller for Enterprise Rel. 7.0 to support Clearcom SIP Trunking Services using TLS – Issue 1.0

## Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking service for an enterprise solution consisting of Avaya Aura® Communication Manager Rel. 7.0, Avaya Aura® Session Manager Rel. 7.0, and Avaya Session Border Controller for Enterprise Rel. 7.0 to support Clearcom SIP Trunking Services using TLS.

The test was performed to verify SIP trunk features including basic calls, call forward (all calls, busy, no answer), call transfer (blind and consult), conference, and voice mail. The calls were placed to and from the PSTN with various Avaya endpoints. For privacy, TLS for Signaling, SRTP for media encryption was used inside of the enterprise (private network side) and TLS for Signaling, RTP for media was used outside of the enterprise (public network side).

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

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

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# 1. Introduction

These Application Notes describe the steps required to configure Session Initiation Protocol (SIP) trunk service between the service provider Clearcom in Mexico and an Avaya SIP-enabled enterprise solution using Transport Layer Security (TLS).

In the sample configuration, the Avaya SIP-enabled enterprise solution consists of an Avaya Aura® Communication Manager Rel. 7.0 (hereafter referred to as Communication Manager), Avaya Aura® Session Manager Rel. 7.0 (hereafter referred to as Session Manager), Avaya Session Border Controller for Enterprise Rel. 7.0 (hereafter referred to as Avaya SBCE), and various Avaya endpoints. This solution does not extend to configurations without the Avaya Session Border Controller for Enterprise or Avaya Aura® Session Manager.

For privacy, TLS for Signaling, SRTP for media encryption was used inside of the enterprise (private network side) and TLS for Signaling, RTP for media was used outside of the enterprise (public network side) (refer to **Section 2.2**).

During interoperability testing, feature test cases were executed to ensure interoperability between Clearcom and Communication Manager.

Customers using an Avaya SIP-enabled enterprise solution with Clearcom SIP Trunking Service are able to place and receive PSTN calls via the SIP protocol. The converged network solution is an alternative to traditional analog trunks and/or PSTN trunks such as ISDN-PRI. This approach generally results in lower cost for the enterprise.

The terms "Service Provider" and "Clearcom" will be used interchangeable throughout these Application Notes.

# 2. General Test Approach and Test Results

The general test approach was to simulate an enterprise site in the Avaya Solution & Interoperability Test Lab by connecting Communication Manager, Session Manager and the Avaya SBCE to Clearcom SIP Trunking Service via the public internet, as depicted in **Figure 1**.

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 for full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

## 2.1. Interoperability Compliance Testing

To verify SIP trunk interoperability, the following areas were tested for compliance:

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

- Incoming calls from the PSTN were routed to DID numbers assigned by Clearcom. Incoming PSTN calls were terminated to the following endpoints: Avaya 96x1 Series IP Deskphones (H.323 and SIP), Avaya 2420 Digital Deskphones, Avaya one-X® Communicator soft phone (H.323 and SIP), Avaya Communicator for Windows (SIP) soft phone, analog Deskphones.
- Inbound and outbound PSTN calls to/from Remote Workers using Avaya 96x1 deskphones (SIP), Avaya one-X® Communicator (SIP) and Avaya Communicator for Windows (SIP).
- Outgoing calls to the PSTN were routed via Clearcom's network to various PSTN destinations.
- Proper disconnect when the caller abandons the call before the call is answered.
- Proper disconnect via normal call termination by the caller or the called parties.
- Proper disconnect by the network for calls that are not answered (with voicemail off).
- Proper response to busy endpoints.
- Proper response/error treatment when dialing invalid PSTN numbers.
- Proper Codec negotiation and two way speech-path. Testing was performed with codecs: G.729A, G.711A and G.711MU (Clearcom's preferred codec order).
- No matching codecs.
- Voicemail and DTMF tone support (leaving and retrieving voice mail messages, etc.).
- Outbound Toll-Free calls, interacting with IVR (Interactive Voice Response systems).
- Calling number blocking (Privacy).
- Call Hold/Resume (long and short duration).
- Call Forward (unconditional, busy, no answer).
- Blind Call Transfers.
- Consultative Call Transfers.
- Station Conference.
- EC500 (Extension to Cellular) calls.
- Simultaneous active calls.
- Long duration calls (over one hour).
- Proper response/error treatment to all trunks busy.
- Proper response/error treatment when disabling SIP connection.

**Note**: Remote Worker was tested as part of this solution. The configuration necessary to support remote workers is beyond the scope of these Application Notes and is not included in these Application Notes.

Items not supported or not tested included the following:

- Inbound toll-free calls, outbound Toll-Free calls, 911 calls (emergency), "0" calls (Operator) and 0+10 digits calls (Operator Assisted) were not tested.
- The SIP REFER method for call redirection was not tested for reasons noted in **Section 2.2**.
- T.38 fax was not tested for reasons noted in **Section 2.2**.

## 2.2. Test Results

Interoperability testing of Clearcom SIP Trunk service with an Avaya SIP-enabled enterprise solution was completed successfully with the following observations/limitations.

- **Secure Real-time Transport Protocol (SRTP)**: SRTP supports RTP media protection on a point to point basis providing confidentiality, message authentication, and replay protection. As SRTP is point to point, all individual links involved in the VoIP call, including key exchange/signaling, must be secure for the call to be secure from end to end. During the compliance test, it was observed that RTP, instead of SRTP, was always used outside of the enterprise (public network side). Calls would fail if the use of SRTP was enforced on the public network side. This behavior may be caused by the far-end not supporting SRTP. Thus **Best Effort** was used during the compliance test, allowing Avaya SBCE to use SRTP on the public network side if supported by the far-end, otherwise it defaults to RTP. SRTP for media encryption was used inside of the enterprise (private network side).
- **SIP REFER method**: PSTN calls that were transferred back to the network using the SIP REFER method did not work properly. Attended call transfers dropped. On blind transfers, the REFER message was accepted by Clearcom with a 202 message, but the trunks were not released after the call transfer was completed. For these reasons testing was done with REFER disabled in Communication Manager (**Network Call Redirection** set to "**n**" under the **trunk-group**, refer to **Section 5.7**). With REFER disabled, blind and attended call transfers to the PSTN completed successfully, with the caveat that Communication Manager trunk channels were not released from the call path after the call was transferred, two trunks channels remained busy/connected for the entire duration of the call.
- **Outbound Calling Party Number (CPN) Blocking**: To support user privacy on outbound calls (calling party number blocking), when enabled by the user, Communication Manager sends "anonymous" as the calling number in the SIP "From" header and includes "Privacy: id" in the INVITE message, while the actual number of the caller is sent in the "P-Asserted-Identity" header. On the called PSTN phone, the calling party number was not blocked, the first DID number assigned to the SIP trunk (5528810001) was displayed, instead of "anonymous".
- **Caller ID on incoming calls from U.S. based PSTN numbers**: Calls originating from PSTN telephones based in the U.S. to Communication Manager displayed "Unavailable". During the compliance test, Clearcom provided a local PSTN test number in Mexico, a SIP softphone was registered to this local PSTN number and was used to originate and terminate local PSTN calls to and from Communication Manager. The correct Caller ID was displayed at the Communication Manager extensions when calling from this local PSTN number. This behavior is not necessarily indicative of a limitation of the combined Avaya/Clearcom solution, this seems to be the expected behavior for international calls from the U.S., which is ultimately controlled by the PSTN providers, it is listed here simply as an observation.
- **Caller ID display on Outbound Calls, Call Forwards and Call transfers to the local PSTN in Mexico**: For outbound calls, calls from the local PSTN in Mexico to Communication Manager that were Forwarded or Transferred back out to the local PSTN in Mexico, the caller ID number displayed at the SIP softphone (local PSTN in Mexico)

was always of the first DID number assigned to the SIP Trunk (5528810001), regardless of the PSTN number being used to originate the call.

- **Caller ID display on EC500 extension to cellular**: For EC500 extension to cellular calls the Caller ID display at the Mobile/cellular station was always of the first DID number assigned to the SIP Trunk (5528810001), regardless of the PSTN number being used to originate the call.
- **Fax Support**: T.38 fax is the fax protocol officially supported by Communication Manager on SIP trunks. During the tests, Clearcom responded with "488 Not Acceptable Here" to the re-INVITE messages sent by Communication Manager to make the change from voice to T.38, causing the call to drop. Even though it was possible during the tests to complete G.711 fax pass-through calls using a local test number in Mexico, G.711 fax pass-through is available in Communication Manager on a "best effort" basis, and it's not guaranteed that it will work in every instance, thus G.711 fax pass-through is not recommended in Communication Manager.
- **From Header Manipulation**: Clearcom uses SIP trunk registration and digest authentication in order to accept calls from the enterprise into their network. Additionally, Clearcom requires the username associated with the SIP trunk credentials to be present in the "From" header of all outbound calls from the enterprise. Otherwise, the call is rejected with a "403 Username=From not allowed" message. A Signaling Script was created in the Avaya SBCE to include the SIP trunk credential's username in the "From" header of all outbound calls. (**Section 7.3.3**).
- **Request-URI Header Manipulation**: Clearcom sends the username associated with the SIP trunk credentials in the "Request-URI" header of all inbound calls, while the actual DID number of the party dialed is sent in the "To" header. Since the routing decision in Session Manager is based on Dial Patterns, by inspecting the number present in the "Request-URI" header of the incoming call, a Signaling Script was created in the Avaya SBCE to populate the "Request URI" header with the number present in the "To" header of inbound calls. (**Section 7.3.3**).
- **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. The following headers were removed from outbound messages using an Adaptation in Session Manager: AV-Global-Session-ID, AV-Correlation-ID, Alert-Info, Endpoint-View, P-AV-Message-id, P-Charging-Vector and P-Location (**Section 6.4**). Additionally, the parameters "gsid" and "epv" were removed from outbound Contact headers using a Signaling Script in the Avaya SBCE (**Section 7.3.3**).

## 2.3. Support

For support on Clearcom systems visit the corporate Web page at: http://www.clearcom.mx/

Avaya customers may obtain documentation and support for Avaya products by visiting http://support.avaya.com. Alternatively, in the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus.

# 3. Reference Configuration

**Figure 1** below illustrates the test configuration used. The test configuration simulates an enterprise site with an Avaya SIP-enabled enterprise solution connected to the Clearcom SIP Trunk service through the public Internet.

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

- Avaya Aura® Communication Manager running on VMware (ESXi 5.5) platform.
- Avaya Aura® Session Manager running on VMware (ESXi 5.5) platform.
- Avaya Aura® System Manager running on VMware (ESXi 5.5) platform.
- Avaya Session Border Controller for Enterprise running on a Dell R210 V2 Server.
- Avaya Aura® Messaging running on VMware (ESXi 5.5) platform.
- Avaya Aura® Media Server running on VMware (ESXi 5.5) platform.
- Avaya G450 Media Gateway.
- Avaya 96x1-Series IP Deskphones (H.323 and SIP).
- Avaya one-X® Communicator soft phones (H.323 and SIP).
- Avaya Communicator for Windows soft phone (SIP)
- Avaya 2420 Digital Deskphones.
- Analog Deskphones.
- Desktop PC running administration interfaces.

Located at the edge of the enterprise is the Avaya SBCE. It has a public side that connects to the public network and a private side that connects to the enterprise network. All SIP and RTP traffic entering or leaving the enterprise flow through the Avaya SBCE. This way, the Avaya SBCE can protect the enterprise against any 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 Clearcom, across the public Internet, was SIP over TLS. The transport protocol between the Avaya SBCE and Session Manager, across the enterprise network, was SIP over TLS. The transport protocol between Session Manager and Communication Manager, across the enterprise network, was SIP over TLS.

A separate SIP trunk group was created between Communication Manager and Session Manager to carry the traffic to and from the service provider (two-way trunk group). To separate the codec settings required by the service provider from the codec used by the telephones, two IP network regions were used, each with dedicated signaling groups.

For inbound calls, the calls flowed from the service provider to the Avaya SBCE, then to Session Manager. Session Manager used the configured dial patterns and routing policies to determine the recipient (in this case Communication Manager), and on which link to send the call. Once the call arrived at Communication Manager, further incoming call treatment, such as incoming digit translations and class of service restrictions are performed.

Outbound calls to the PSTN were first processed by Communication Manager for outbound feature treatment, such as Automatic Route Selection (ARS) and class of service restrictions. Once Communication Manager selected the proper SIP trunk, the call was routed to Session Manager. Session Manager once again used the configured dial patterns and routing policies to determine the route to the Avaya SBCE for egress to Clearcom's network.

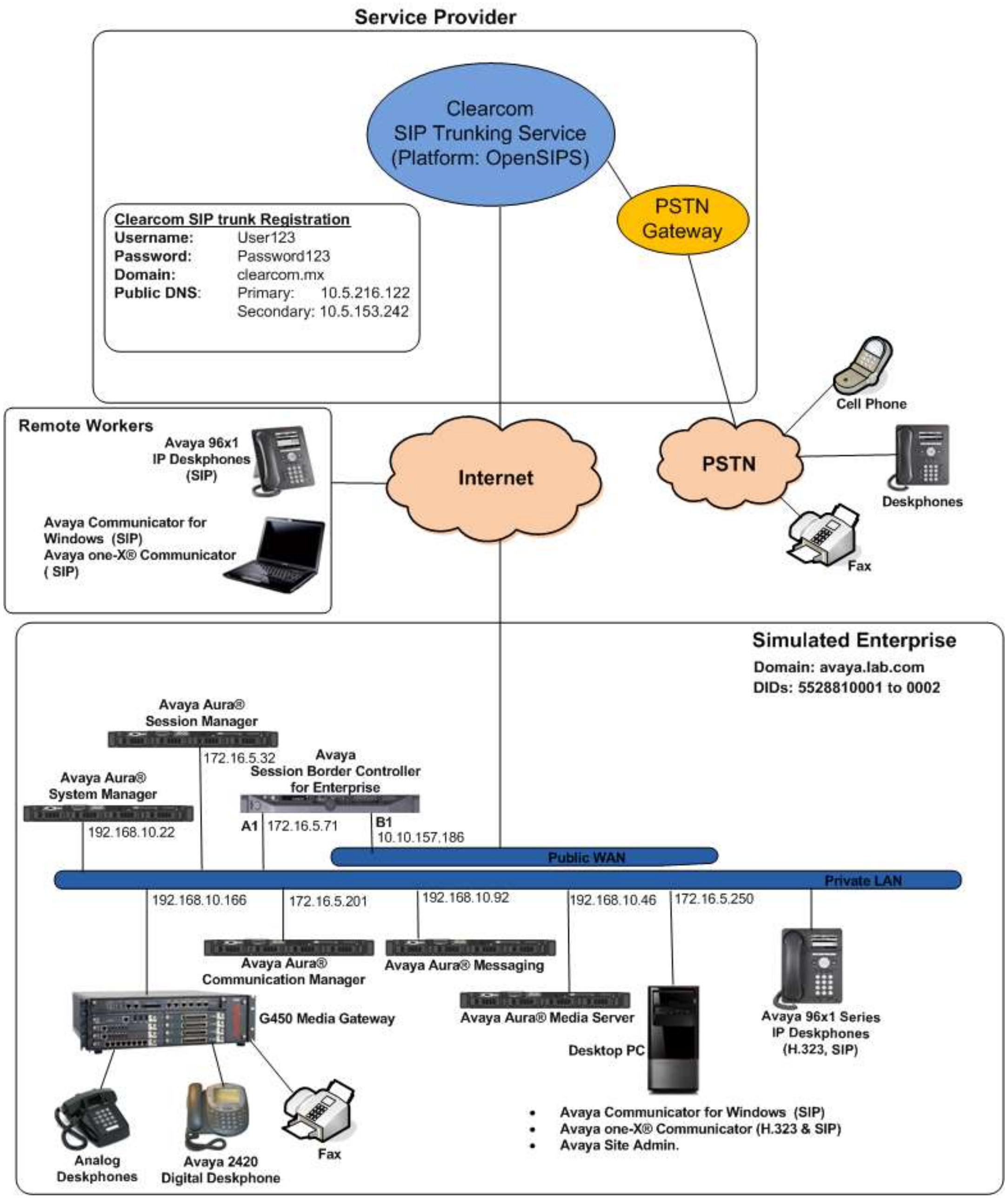



**Figure 1: Avaya SIP-enabled Enterprise Solution and Clearcom SIP Trunking Service**

# 4. Equipment and Software Validated

The following equipment and software were used for the compliance testing in the simulated enterprise:

| Equipment/Software | Release/Version |
|---|---|
| **Avaya** | |
| Avaya Aura® Communication Manager running on VMware ESXi 5.5 platform | 7.0.0.3.1 SP 3.1<br>(00.0.441.0-22903) |
| Avaya Aura® Session Manager running on VMware ESXi 5.5 platform | 7.0 SP2<br>(7.0.0.2.700201) |
| Avaya Aura® System Manager running on VMware ESXi 5.5 platform | 7.0.0.2<br>Build No. 7.0.0.0.16266-7.0.9.7002010<br>Software Update Rev. No. 7.0.0.2.4416 |
| G450 Gateway | 37.21.0 |
| Avaya Session Border Controller for Enterprise running on a DELL R210 V2 Server | 7.0.1-03-8739 |
| Avaya Aura® Media Server running on VMware ESXi 5.5 platform | 7.7.0.236 |
| Avaya Aura® Messaging running on VMware ESXi 5.5 platform | 6.3.3 Service Pack 3<br>(MSG-03.0.141.0-348_0304) |
| Avaya Aura® Integrated Management Site Administrator | 6.0.07 |
| Avaya one-X® Communicator (SIP & H.323) | 6.2.11.03-SP11 |
| Avaya Communicator for Windows (SIP) | 2.1.3.80 |
| Avaya 96x1 Series IP Deskphones (H.323) | Version 6.6029 |
| Avaya 96x1 Series IP Deskphones (SIP) | Version 7.0.0.39 |
| Avaya 2420 Series Digital Deskphone | -- |
| Lucent Analog Deskphone | -- |
| **Clearcom** | |
| OpenSIPS Softswitch | 1.9 |
| OpenSIPS Session Border Controller | 1.9 |

**Table 2 – Hardware and Software Components Tested**

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

# 5. Configure Avaya Aura® Communication Manager

This section describes the procedure for configuring Communication Manager. A SIP trunk is established between Communication Manager and Session Manager for use by signaling traffic to and from Clearcom. It is assumed that the general installation of Communication Manager, the Avaya G450 Media Gateway and the Avaya Aura® Media Server has been previously completed.

In configuring Communication Manager, various components such as ip-network-regions, signaling groups, trunk groups, etc. need to be selected or created for use with the SIP connection to the Service Provider. Unless specifically stated otherwise, any unused ip-network-region, signaling group, trunk group, etc. can be used for this purpose.

The Communication Manager configuration was performed using the Avaya Integrated Management Site Administrator. Some screens in this section have been abridged and highlighted for brevity and clarity in presentation. Note that the public IP addresses shown throughout these Application Notes have been edited so that the actual public IP addresses of the network elements are not revealed. Some screens captures will show the use of the **change** command instead of the **add** command, since the configuration used for the testing was previously added.

## 5.1. Licensing and Capacity

Use the **display system-parameter customer-options** to verify that **Media Encryption over IP** is set to ***y***.

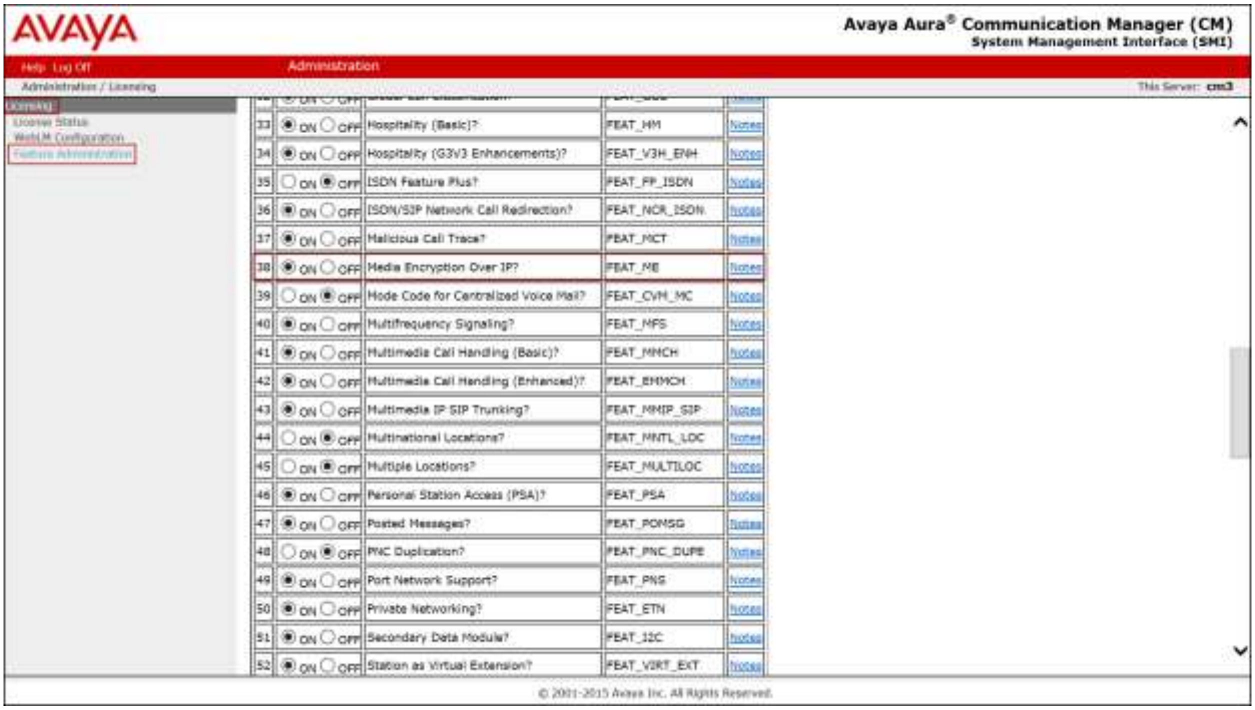
```
display system-parameters customer-options                      Page   5 of  12
                                OPTIONAL FEATURES

   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               Media Encryption Over IP? y
  Five Port Networks Max Per MCC? n     Mode Code for Centralized Voice Mail? n
                Flexible Billing? n
   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.)
```

If it's set to ***n***, obtain a license file for Communication Manager with the Media Encryption feature enabled.

After installing the license, in the SMI interface of Communication Manager (web interface), go to **Administration/Licensing → Feature Administration**. Go to Current Settings (not shown); look for **Media Encryption over IP?** and enable it (select ***ON***). Go to the bottom of the page and click on **Submit** (not shown).



In the Communication Manager SAT terminal, go back to display system customer options and that **Media Encryption over IP?** is set to ***y*** on page 5.

```
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                Media Encryption Over IP? y
  Five Port Networks Max Per MCC? n   Mode Code for Centralized Voice Mail? n
                Flexible Billing? n
   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.)
```

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 SIP trunks to the Service Provider. The example below shows one license with a capacity of ***24000*** trunks available and ***122*** in use. The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity.

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

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 12000 10
           Maximum Concurrently Registered IP Stations: 18000 1
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 41000 1
                   Maximum Video Capable IP Softphones: 18000 7
                       Maximum Administered SIP Trunks: 24000 122
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0








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

On **Page 4**, verify that **ARS** is set to ***y***.

```
display system-parameters customer-options                      Page   4 of  12
                                OPTIONAL FEATURES

    Abbreviated Dialing Enhanced List? y          Audible Message Waiting? y
        Access Security Gateway (ASG)? n              Authorization Codes? y
        Analog Trunk Incoming Call ID? y                       CAS Branch? n
 A/D Grp/Sys List Dialing Start at 01? y                         CAS Main? n
Answer Supervision by Call Classifier? y                Change COR by FAC? n
                                  ARS? y  Computer Telephony Adjunct Links? y
                 ARS/AAR Partitioning? y   Cvg Of Calls Redirected Off-net? y
          ARS/AAR Dialing without FAC? n                      DCS (Basic)? y
          ASAI Link Core Capabilities? n                DCS Call Coverage? y
          ASAI Link Plus Capabilities? n               DCS with Rerouting? y
       Async. Transfer Mode (ATM) PNC? n
  Async. Transfer Mode (ATM) Trunking? n    Digital Loss Plan Modification? y
              ATM WAN Spare Processor? n                          DS1 MSP? y
                                 ATMS? y            DS1 Echo Cancellation? y
                  Attendant Vectoring? y



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

## 5.2. System Features

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

```
display system-parameters features                              Page   1 of  19
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? n
                                   Trunk-to-Trunk Transfer: all
                Automatic Callback with Called Party Queuing? n
    Automatic Callback - No Answer Timeout Interval (rings): 3
                         Call Park Timeout Interval (minutes): 10
        Off-Premises Tone Detect Timeout Interval (seconds): 20
                                  AAR/ARS Dial Tone Required? y

               Music (or Silence) on Transferred Trunk Calls? all
               DID/Tie/ISDN/SIP Intercept Treatment: attendant
    Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                   Automatic Circuit Assurance (ACA) Enabled? n



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

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

```
display 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:
         International Access Code:

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 the Avaya server running Communication Manager (**procr**), and for Session Manager (**Lab-HG-SM**). These node names will be needed for defining the Service Provider signaling group in **Section 5.6**.

```
change node-names ip                                            Page   1 of   2
                                 IP NODE NAMES
    Name              IP Address
ASBCE A1            172.16.5.71
Lab-HG-SM           172.16.5.32
MA-CM               192.168.10.12
default             0.0.0.0
media_server        192.168.10.46
msgserver           172.16.5.12
procr               172.16.5.201
procr6              ::








( 8  of 8    administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

## 5.4. Codecs and Media Encryption

Use the **change ip-codec-set** command to define a list of codecs to use for calls between the enterprise and the Service Provider. For the compliance test, **ip-codec-set 2** was used for this purpose. Clearcom supports G.729A, G.711MU and G.711A. Thus, these codecs were included in this set. Enter ***G.729A, G.711A*** and ***G.711MU*** in the **Audio Codec** column of the table; this is Clearcom's preferred codec order. Set **Media Encryption** to ***1-srtp-aescm128-hmac80*** and ***2-srtp-aescm128-hmac32***, this value must match the Media Encryption value set under the Avaya SBCE **Media Rules**, **Section 7.4.2**. Set **Encrypted SRTCP** to ***enforce-unenc-srtcp***.

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

                          IP CODEC SET

    Codec Set: 2

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

     Media Encryption                        Encrypted SRTCP: enforce-unenc-srtcp
 1: 1-srtp-aescm128-hmac80
 2: 2-srtp-aescm128-hmac32
 3:
 4:
 5:
```

On **Page 2**, set the **Fax Mode** to ***off*** (T.38 fax is currently not supported by Clearcom, refer **Section 2.2**).

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

                          IP CODEC SET

                              Allow Direct-IP Multimedia? n


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

## 5.5. IP Network Region

Create a separate IP network region for the Service Provider trunk. This allows for separate codec or quality of service settings to be used (if necessary) for calls between the enterprise and the Service Provider versus calls within the enterprise or elsewhere. For the compliance test, **IP-network-region 2** was chosen for the Service Provider trunk. Use the **change ip-network-region *2*** command to configure region 2 with the following parameters:

- Set the **Authoritative Domain** field to match the SIP domain of the enterprise. In this configuration, the domain name is ***avaya.lab.com***. This name appears in the "From" header of SIP messages 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 Avaya Media Gateway. Set both **Intra-region** and **Inter-region IP-IP Direct Audio** to *yes*. This is the default setting. Shuffling can be further restricted at the trunk level on the Signaling Group form.
- Set the **Codec Set** field to the IP codec set defined in **Section 5.4**.
- Default values can be used for all other fields.

```
change ip-network-region 2                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 2
Location: 1         Authoritative Domain: avaya.lab.com
    Name: SP Region                Stub Network Region: n
MEDIA PARAMETERS                   Intra-region IP-IP Direct Audio: yes
      Codec Set: 2                 Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                         IP Audio Hairpinning? n
   UDP Port Max: 3349
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

On **Page 4**, define the IP codec set to be used for traffic between region 2 and region 1. Enter the desired IP codec set in the **codec set** column of the row with destination region (**dst rgn**) 1. Default values may be used for all other fields. The example below shows the settings used for the compliance test. It indicates that codec set ***2*** will be used for calls between region 2 (the Service Provider region) and region 1 (the rest of the enterprise).

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

 Source Region: 2     Inter Network Region Connection Management     I       M
                                                                     G  A    t
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    c
 rgn  set   WAN  Units     Total Norm  Prio Shr Regions          CAC  R  L    e
 1    2     y    NoLimit                                              n       t
 2    2                                                                  all
 3
 4
 5
 6
 7
 8
 9
 10
 11
 12
 13
 14
 15
```

## 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 SIP 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***. This specifies Communication Manager will serve as an Evolution Server for Session Manager.
- Set the **Transport Method** to the recommended default value of ***tls*** (Transport Layer Security).The transport method specified here is used between Communication Manager and Session Manager. The transport method used between Session Manager and the Avaya SBCE is specified as TLS in **Sections 6.6** and **7.3.4**. Lastly, the transport method between the Avaya SBCE and Clearcom is also TLS. This is defined in **Section 7.3.4**.
- Set the **Near-end Listen Port** and **Far-end Listen Port** to a valid unused port, instead of the default well-known port value. (For TLS, the well-known port value is 5061). This is necessary so Session Manager can distinguish this trunk from the trunk used for other enterprise SIP traffic. The compliance test was conducted with the **Near-end Listen Port** and **Far-end Listen Port** set to ***5071***.
- Set the **Peer Detection Enabled** field to ***y***. The **Peer-Server** field will initially be set to ***Others*** and cannot be changed via administration. Later, the **Peer-Server** field will

automatically change to ***SM*** once Communication Manager detects its peer as Session Manager.

- Set the **Near-end Node Name** to ***procr***. This node name maps to the IP address of the Avaya Server running Communication Manager as defined in **Section 5.3**.
- Set the **Far-end Node Name** to ***Lab-HG-SM***. This node name maps to the IP address of Session Manager as defined in **Section 5.3**.
- Set the **Far-end Network Region** to the IP network region defined for the Service Provider in **Section 5.5**.
- Set the **Far-end Domain** to the domain of the enterprise.
- Set **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 inside IP of the Avaya SBCE and the enterprise endpoint. If this value is set to ***n***, then the Avaya Media Gateway will remain in the media path of all calls between the SIP trunk and the endpoint.
- Set the **DTMF over IP** field to ***rtp-payload***. This value enables Communication Manager to send DTMF transmissions using RFC 2833.
- Default values may be used for all other fields.

```
change 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: Lab-HG-SM
 Near-end Listen Port: 5071                   Far-end Listen Port: 5071
                                           Far-end Network Region: 2

Far-end Domain: avaya.lab.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? n                       Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n               Alternate Route Timer(sec): 6
```

## 5.7. Trunk Group

Use the **add trunk-group** command to create a trunk group for the signaling group created in **Section 5.6**. For the compliance test, **trunk group 2** was configured using the parameters highlighted below.

- Set the **Group Type** field to ***sip***.
- Enter a descriptive name for the **Group Name**.
- Enter an available trunk access code (TAC) that is consistent with the existing dial plan in the **TAC** field.
- Set the **Service Type** field to ***public-ntwrk***.
- Set the **Signaling Group** to the signaling group shown in **Section 5.6**.
- Set the **Number of Members** field to the number of trunk members in the SIP trunk group. This value determines how many simultaneous SIP calls can be supported by this trunk.
- Default values were used for all other fields.

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

Group Number: 2                   Group Type: sip          CDR Reports: y
  Group Name: Service Provider           COR: 1        TN: 1        TAC: 602
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: public-ntwrk            Auth Code? n
                                            Member Assignment Method: auto
                                                     Signaling Group: 2
                                                   Number of Members: 10
```

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 that re-INVITEs must be sent to keep the active session alive. Note that the value assigned to the **Preferred Minimum Session Refresh Interval (sec)** field is doubled and assigned to the "Min-SE" Header Field in SIP INVITE messages for calls originating from Communication Manager. Using the default setting of ***600*** seconds as in the example, the "Min-SE" Header Field would be populated for 1200 seconds in SIP INVITE messages originating from Communication Manager.

```
change 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 ***private***. This field specifies the format of the calling party number (CPN) sent to the far-end. Beginning with Communication Manager 6.0, public numbers are automatically preceded with a + sign when passed in the SIP "From", "Contact", "P-Asserted Identity" and "Diversion" headers. The addition of the "+" sign impacted caller ID presentation on outbound calls sent to Clearcom. Thus, the **Numbering Format** was set to ***private*** and the **Numbering Format** in the route pattern was set to ***unk-unk*** (**Section 5.10**).

Set the **Replace Restricted Numbers** and **Replace Unavailable Numbers** fields to ***y***. This will allow the CPN displayed on local endpoints to be replaced with the value set in **Section 5.2**, if the inbound call enabled CPN block.

Default values were used for all other fields.

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



                     Numbering Format: private
                                              UUI Treatment: service-provider

                                               Replace Restricted Numbers? y
                                              Replace Unavailable Numbers? y

                                                Hold/Unhold Notifications? y
                               Modify Tandem Calling Number: no




 Show ANSWERED BY on Display? y
```

**Page 4** was configured using the parameters highlighted below.

- Set the **Network Call Redirection** field to ***n***. This setting directs Communication Manager **not** to use the SIP REFER method for transferring calls off-net to the PSTN, refer to **Section 2.2**.
- Set the **Send Diversion Header** field to ***n***.
- Set the **Support Request History** field to ***n***.
- Set the **Telephone Event Payload Type** to ***101***. The value preferred by Clearcom.
- Set the **Convert 180 to 183 for Early Media** to ***y***.
- Set the **Always Use re-INVITE for Display Updates** field to ***y***.
- Set the **Identity for Calling Party Display** to ***P-Asserted-Identity***.

```
change trunk-group 2                                            Page   4 of  21
                              PROTOCOL VARIATIONS

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

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


                        Convert 180 to 183 for Early Media? y
                  Always Use re-INVITE for Display Updates? y
                        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 private numbering was selected to define the format of this number (**Section 5.7**), use the **change private-numbering** command to create an entry for each extension which has a DID assigned. The DID numbers are assigned by the Service Provider. Each DID number is assigned to one enterprise internal extension or Vector Directory Numbers (VDNs).

The screen below shows DID numbers assigned for testing. The DID numbers were mapped to enterprise extensions 3041, 3042, 3044 and 3045.

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

Ext Ext              Trk          Private           Total
Len Code             Grp(s)       Prefix            Len
 4  3                                                4     Total Administered: 6
 4  5                                                4        Maximum Entries: 540
 4  3041             2            5528810001        10
 4  3042             2            5528810002        10
 4  3044             2            5528810003        10
 4  3045             2            5528810004        10
```

**Note**: During the compliance test, Clearcom did not inspect the calling party number sent in the origination headers from the enterprise to authenticate outbound calls; it used SIP trunk registration and Digest Authentication instead. This is shown in **Section 7.3.4** of the Avaya SBCE configuration, later in this document. Clearcom also inserted the main DID number assigned to the SIP trunk on all outbound calls sent to the PSTN, for caller ID purposes. Since the calling party information sent from the enterprise was for all practical purposes not used by Clearcom, the configuration shown on the screen above was not strictly required, and it is shown here simply for completeness.

## 5.9. Inbound Routing

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

```
change inc-call-handling-trmt trunk-group 2                     Page   1 of  30
                        INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number       Del  Insert
 Feature        Len      Digits
 public-ntwrk   10  5528810001        10   3041
 public-ntwrk   10  5528810002        10   3042
 public-ntwrk   10  5528810003        10   3044
 public-ntwrk   10  5528810004        10   3045
 public-ntwrk
 public-ntwrk
 public-ntwrk
 public-ntwrk
 public-ntwrk
 public-ntwrk
 public-ntwrk
 public-ntwrk
 public-ntwrk
 public-ntwrk
 public-ntwrk
 public-ntwrk
 public-ntwrk
 public-ntwrk
```

## 5.10. Outbound Routing

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

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

   Dialed     Total  Call     Dialed     Total  Call     Dialed     Total  Call
   String     Length Type     String     Length Type     String     Length Type
  0             13  udp
  1             4   dac
  2             4   ext
  3             4   ext
  4             4   udp
  5             4   ext
  6             3   dac
  7             4   ext
  8             1   fac
  9             1   fac
  *             3   dac
  #             2   dac
```

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

```
change feature-access-codes                                     Page   1 of  10
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code: ____
         Abbreviated Dialing List2 Access Code: ____
         Abbreviated Dialing List3 Access Code: ____
Abbreviated Dial - Prgm Group List Access Code: ____
                      Announcement Access Code: #7__
                       Answer Back Access Code: ____
                         Attendant Access Code: __
      Auto Alternate Routing (AAR) Access Code: 8___
    Auto Route Selection (ARS) - Access Code 1: 9___    Access Code 2: ____
                 Automatic Callback Activation: ____     Deactivation: ____
Call Forwarding Activation Busy/DA: ____   All: ____     Deactivation: ____
   Call Forwarding Enhanced Status: ____   Act: ____     Deactivation: ____
                         Call Park Access Code: ____
                       Call Pickup Access Code: *44_
CAS Remote Hold/Answer Hold-Unhold Access Code: ____
                   CDR Account Code Access Code: ____
                        Change COR Access Code: ____
                   Change Coverage Access Code: ____
            Conditional Call Extend Activation: ____     Deactivation: ____
                   Contact Closure   Open Code: ____       Close Code: ____
```

Use the **change ars analysis** command to configure the routing of dialed digits following the first digit 9. The example below shows a subset of the dialed strings tested as part of the compliance test. Refer to **Section 2.1** for the complete list of call types tested. All dialed strings are mapped to **route pattern 2** which contains the SIP trunk 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
    0                     13   13   1         hnpa           n
    00                    2    2    deny      op             n
    001                   13   13   2         intl           n
    01                    12   12   2         natl           n
    011                   10   18   2         intl           n
    040                   3    3    2         svcl           n
    045                   13   13   2         natl           n
    101xxxx0              8    8    deny      op             n
    101xxxx0              18   18   deny      op             n
    101xxxx01             16   24   deny      iop            n
    101xxxx011            17   25   deny      intl           n
    101xxxx1              18   18   deny      fnpa           n
    10xxx0                6    6    deny      op             n
    10xxx0                16   16   deny      op             n
```

The route pattern defines which trunk group will be used for the call and performs any necessary digit manipulation. Use the **change route-pattern** command to configure the parameters for the Service Provider trunk route pattern in the following manner. The example below shows the values used for route pattern 2 during the compliance test.

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

```
change route-pattern 2                                          Page   1 of   3
                    Pattern Number: 2      Pattern Name: Serv. Provider
    SCCAN? n     Secure SIP? n     Used for SIP stations? n

    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 2    0                                                             n   user
 2:                                                                    n   user
 3:                                                                    n   user
 4:                                                                    n   user
 5:                                                                    n   user
 6:                                                                    n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM Sub  Numbering LAR
     0 1 2 M 4 W    Request                                 Dgts Format
 1:  y y y y y n  n            rest                              unk-unk   none
 2:  y y y y y n  n            rest                                        none
 3:  y y y y y n  n            rest                                        none
 4:  y y y y y n  n            rest                                        none
 5:  y y y y y n  n            rest                                        none
 6:  y y y y y n  n            rest                                        none
```

**Note**: To save all Communication Manager provisioning changes, enter the command **save translations**.

# 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 occupied by SIP Entities
- Adaptation module to perform dial plan manipulation.
- SIP Entities corresponding to Communication Manager, the Avaya SBCE and Session Manager
- 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
- Session Manager, corresponding to the Session Manager server to be managed by System Manager.

It may not be necessary to create all the items above when configuring 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, Adaptations, SIP Entities, and Session Manager itself. However, each item should be reviewed to verify the configuration.

**Note**: Some of the default information in the screenshots that follow may have been cut out (not included) for brevity.

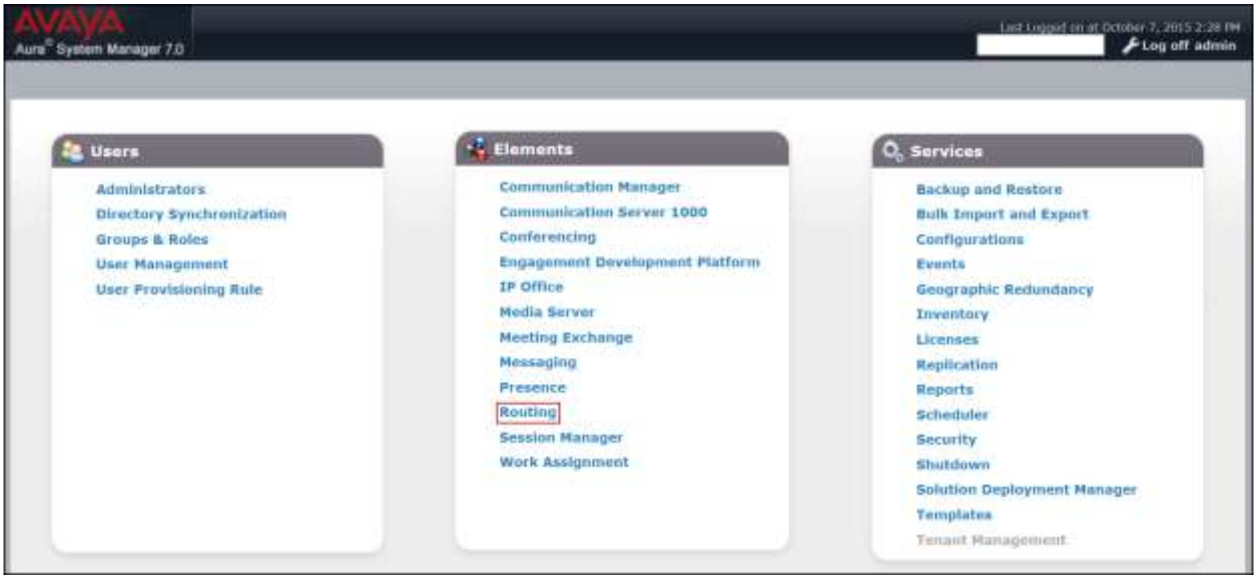
**Note**: Some Avaya products are shipped with a default identity TLS certificate signed by Avaya, to enable out-of-box support for TLS sessions. These are considered “demo” certificates which do not meet the current National Institute of Standards and Technology (NIST) security standards. For security reasons these default “demo” certificates should not be used in Production.

**Avaya recommends using 3rd Party Certificate Authority (CA) signed identity certificates for enhanced security.**

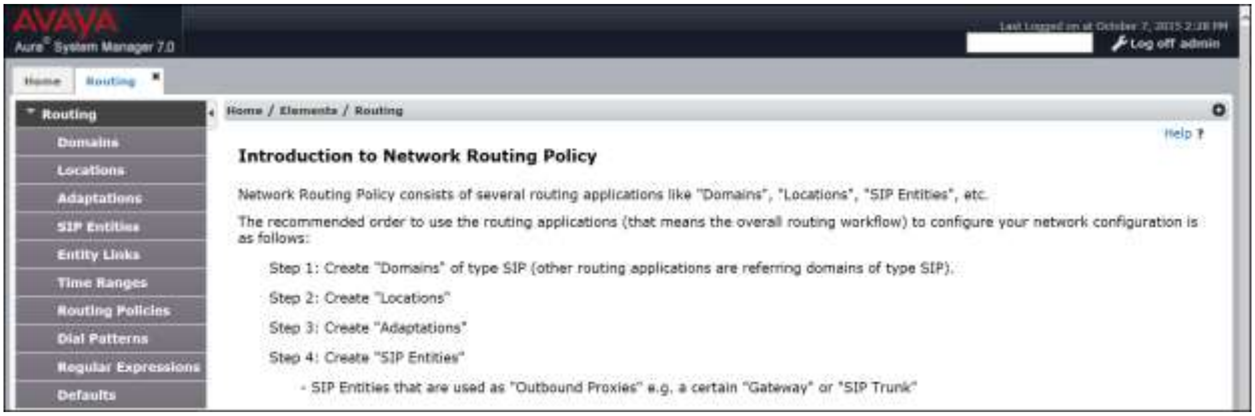
On the enterprise side (or private side), testing was done with the default “demo” TLS identity certificates. The procedure to obtain and install 3rd Party CA TLS certificates is outside the scope of these Application Notes. Refer to items [**3**], [**5**] and [**8**] in **Section 11**.

## 6.1. System Manager Login and Navigation

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



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



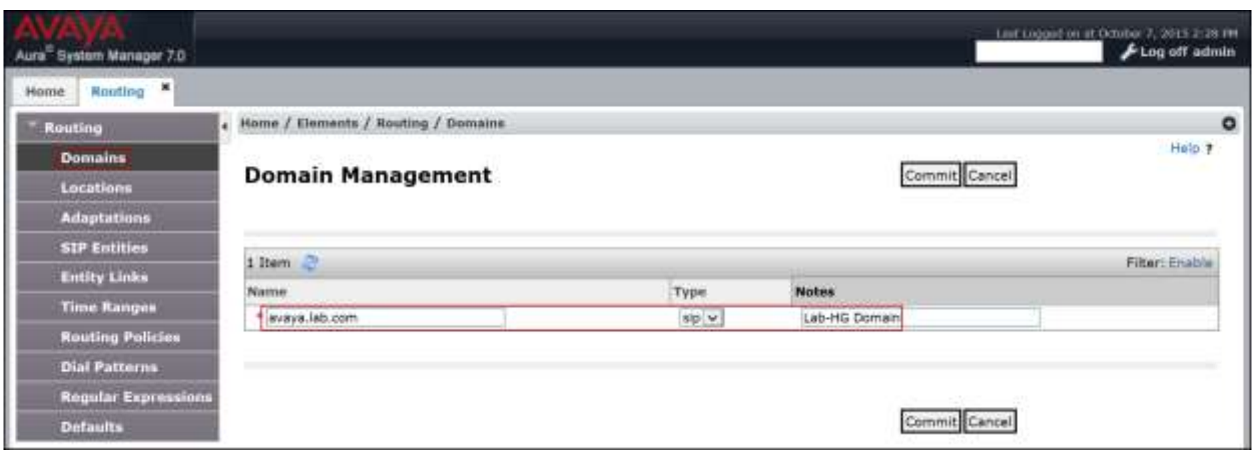

## 6.2. Specify SIP Domain

Create a SIP domain for each domain of which Session Manager will need to be aware in order to route calls. For the compliance test the enterprise domain ***avaya.lab.com*** was used.

To add a domain, navigate to **Routing → Domains** in the left-hand navigation pane and click the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

- **Name:** Enter the domain name.
- **Type:** Select ***sip*** from the pull-down menu.
- **Notes:** Add a brief description (optional).
- Click **Commit** to save (not show).

The screen below shows the entry for the enterprise domain **avaya.lab.com**.



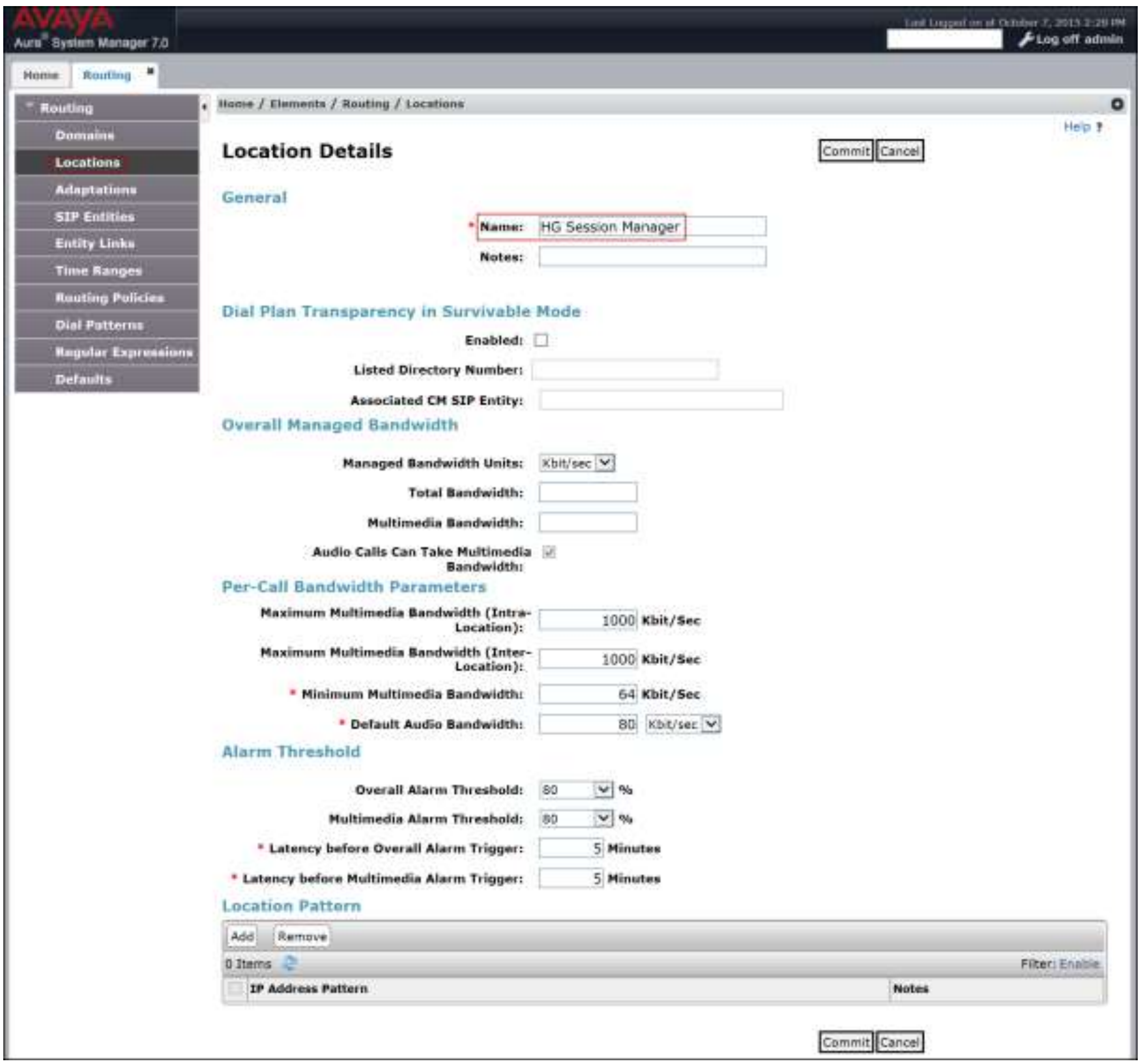
## 6.3. Add Location

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

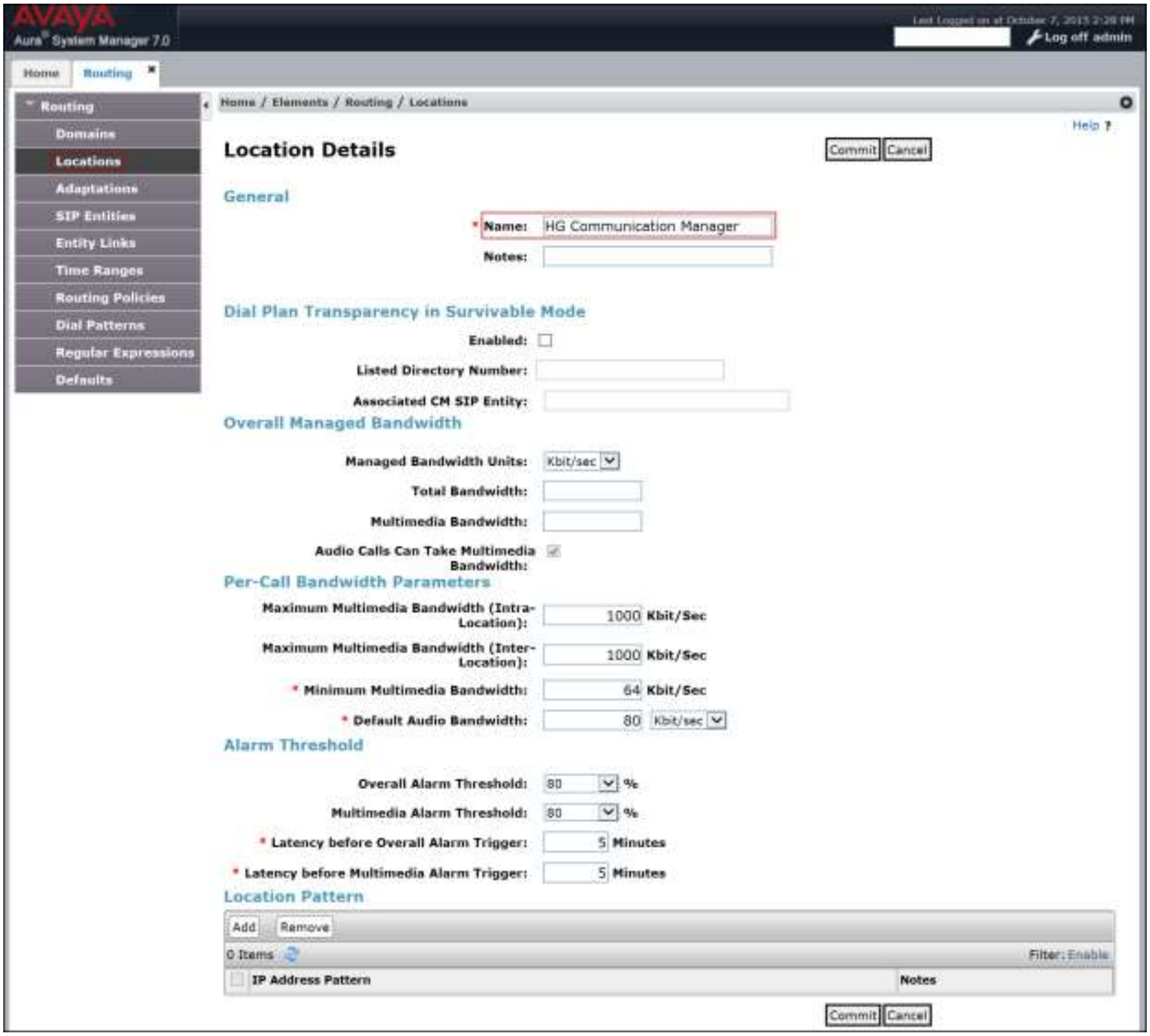
In the **General** section, enter the following values. Use default values for all remaining fields:

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

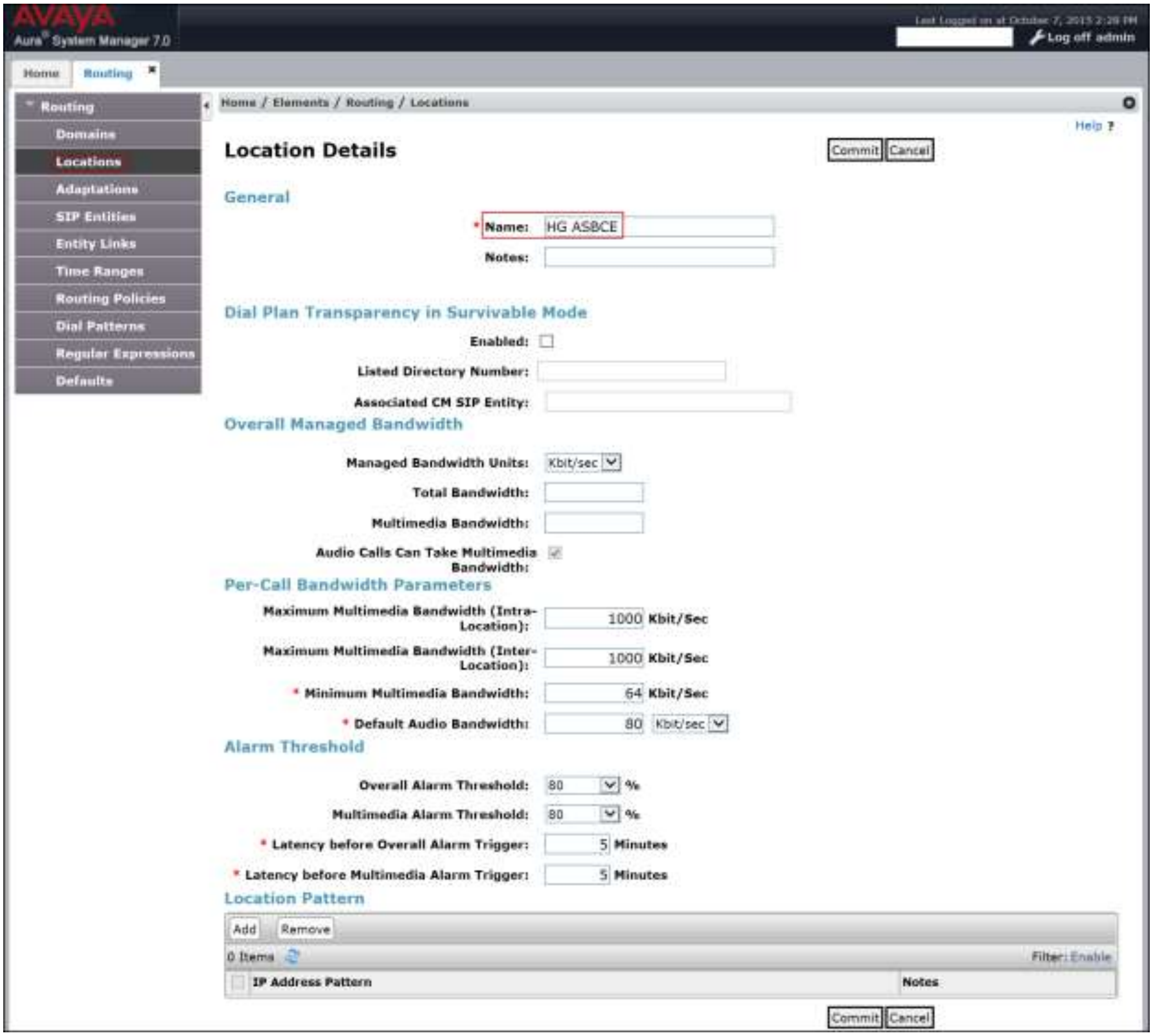
The screen below shows the **HG Session Manager** location. This location will be assigned later to the SIP Entity corresponding to Session Manager.



The following screen shows the **HG Communication Manager** location. This location will be assigned later to the SIP Entity corresponding to Communication Manager.



The following screen shows the **HG ASBCE** location. This location will be assigned later to the SIP Entity corresponding to the Avaya SBCE.



## 6.4. Adaptations

In order to improve interoperability with third party elements, Session Manager 7.0 incorporates the ability to use Adaptation modules to remove specific headers that are either Avaya proprietary or deemed excessive/unnecessary for non-Avaya elements.

For the compliance test, an Adaptation named "CM_Outbound_Header_Removal" was created to block the following headers from outbound messages, before they were forwarded to the Avaya SBCE: Alert-Info, P-Charging-Vector, AV-Global-Session-ID, AV-Correlation-ID, P-AV-Message-id, P-Location, and Endpoint-View. These headers contain private information from the enterprise, which should not be propagated outside of the enterprise boundaries. They also add unnecessary size to outbound messages, while they have no significance to the service provider.

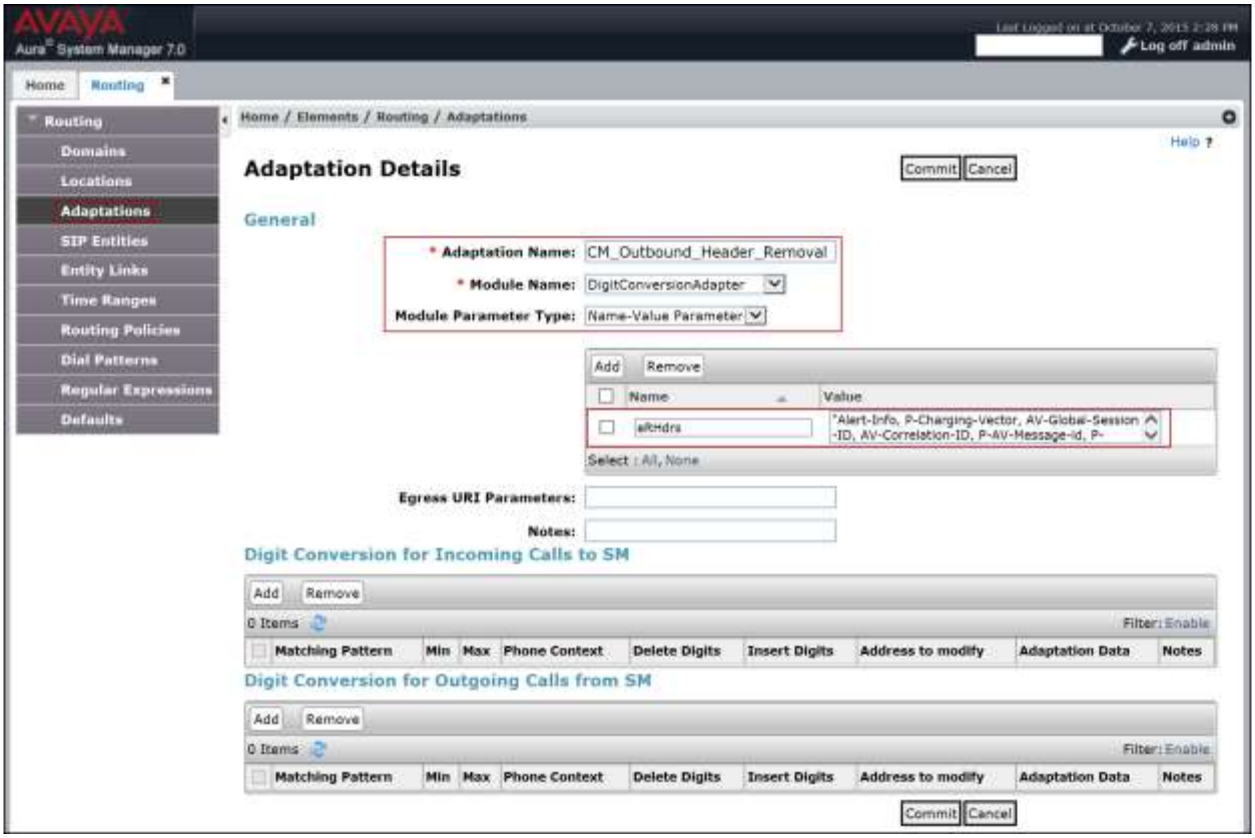
Navigate to **Routing → Adaptations** in the left-hand navigation pane and click the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

- **Adaptation Name**: Enter an appropriate name.
- **Module Name**: Select the ***DigitConversionAdapter*** option.
- **Module Parameter Type**: Select ***Name-Value Parameter***.

Click **Add** to add the name and value parameters.

- **Name**: Enter ***eRHdrs***. This parameter will remove the specified headers from messages in the egress direction.
- **Value**: Enter ***"Alert-Info, P-Charging-Vector, AV-Global-Session-ID, AV-Correlation-ID, P-AV-Message-id, P-Location, Endpoint-View"***

The screen below shows the adaptation created for the compliance test. This adaptation will later be applied to the SIP Entity corresponding to the Avaya SBCE. All other fields were left with their default values.



## 6.5. SIP Entities

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

In the **General** section, enter the following values. Use default values for all remaining fields:

- **Name:** Enter a descriptive name.
- **FQDN or IP Address:** Enter the FQDN or IP address of the SIP Entity interface that is used for SIP signaling.
- **Type:** Enter ***Session Manager*** for Session Manager, ***CM*** for Communication Manager and ***SIP Trunk*** (or ***Other***) for the Avaya SBCE.
- **Adaptation:** This field is only present if **Type** is not set to **Session Manager**. If applicable, select the **Adaptation Name.**
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Select the time zone for the location above.

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 **Session Manager** SIP entities.

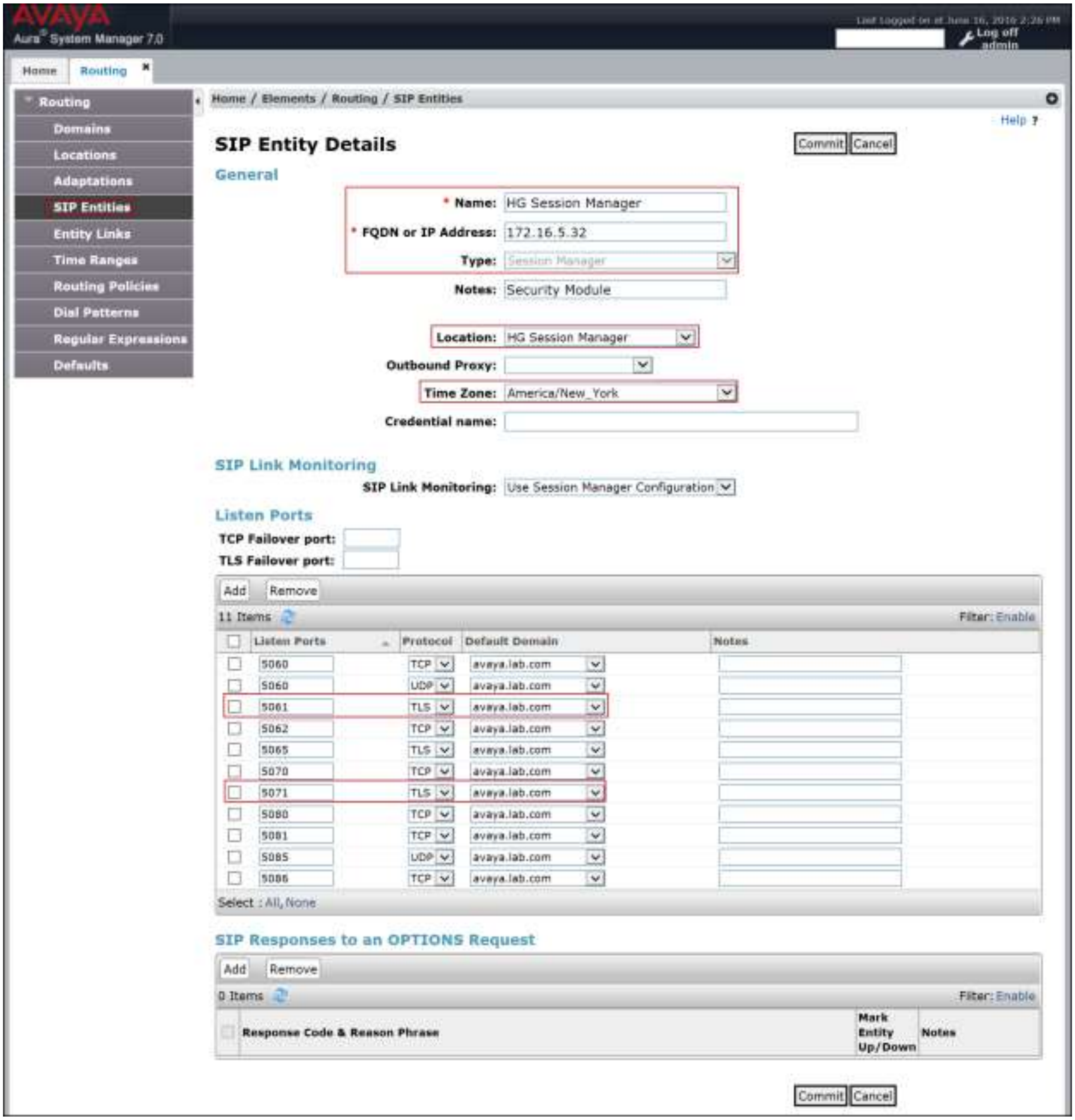
In the **Port** section, click **Add** and enter the following values. Use default values for all remaining fields:

- **Port:** Port number on which the Session Manager will listen for SIP requests.
- **Protocol:** Transport protocol to be used to send SIP requests.
- **Default Domain:** The domain used for the enterprise.
- Click **Commit** to save.

The following screen shows the addition of the Session Manager SIP entity. The name ***HG Session Manager***, the IP address of the Session Manager signaling interface, the Location ***HG Session Manager*** created in **Section 6.3** and the **Time Zone** were used.

For the compliance test, only two Ports were used:

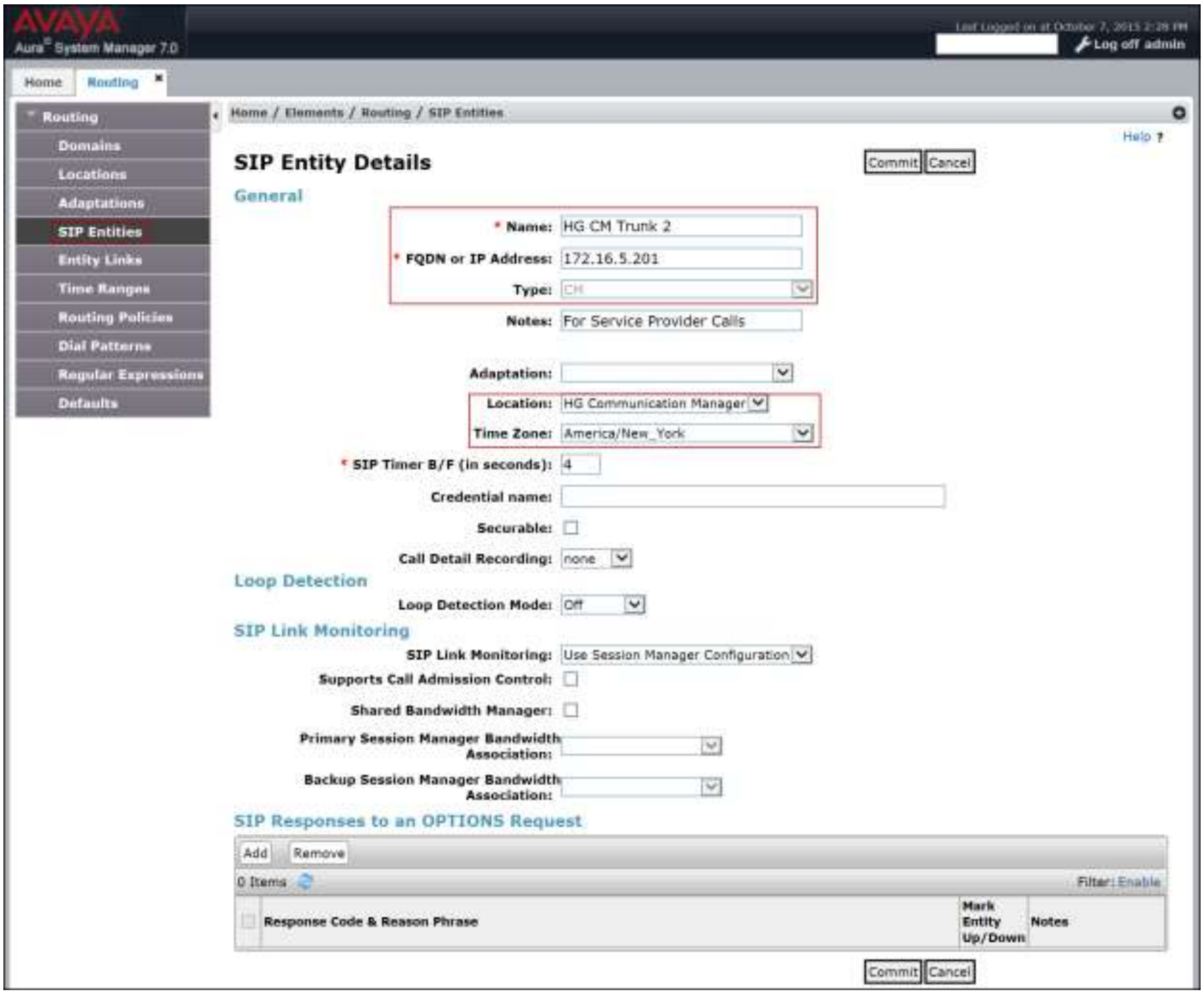
- ***5061*** with ***TLS*** for connecting to the Avaya SBCE.
- ***5071*** with ***TLS*** for connecting to Communication Manager.



The following screen shows the addition of the Communication Manager SIP Entity.

A separate SIP entity for Communication Manager is required in order to route traffic from Communication Manager to the Service Provider.

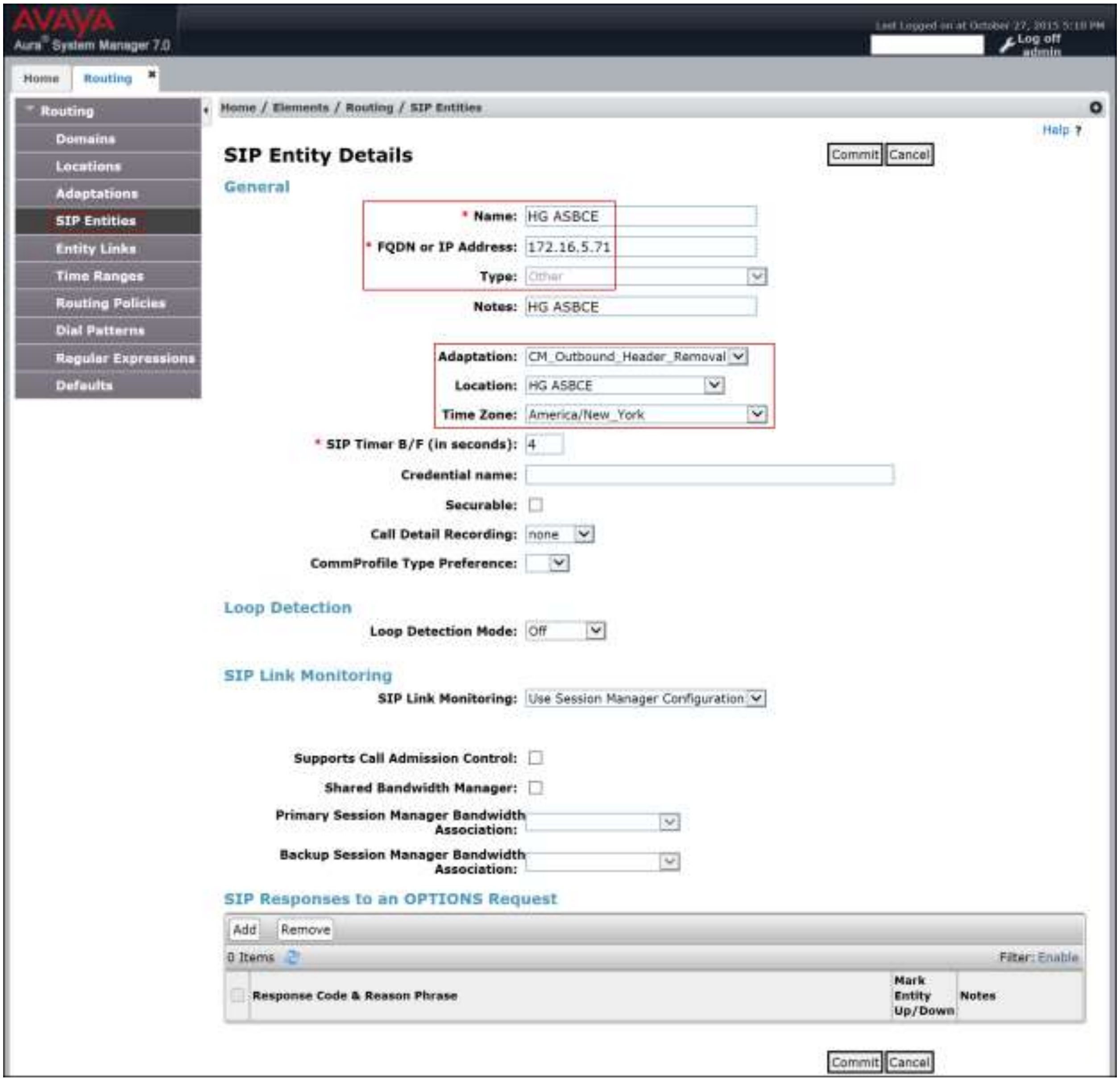
The name ***HG CM Trunk 2***, the IP of the Avaya Server running Communication Manager, the **Type** of ***CM*** for Communication Manager, the Location ***HG Communication Manager*** created in **Section 6.3** and the **Time Zone** were used.



The following screen shows the addition of the SIP entity for the Avaya SBCE.

The name ***HG ASBCE***, the inside IP address of the Avaya SBCE, the **Type** of ***Other***, the adaptation ***CM_Outbound_Header_Removal*** created in **Section 6.4**, the location ***HG ASBCE*** created in **Section 6.3** and the **Time Zone** were used.

Note: **Type**: ***Other*** was used during the testing; **SIP Trunk** could have been used instead.



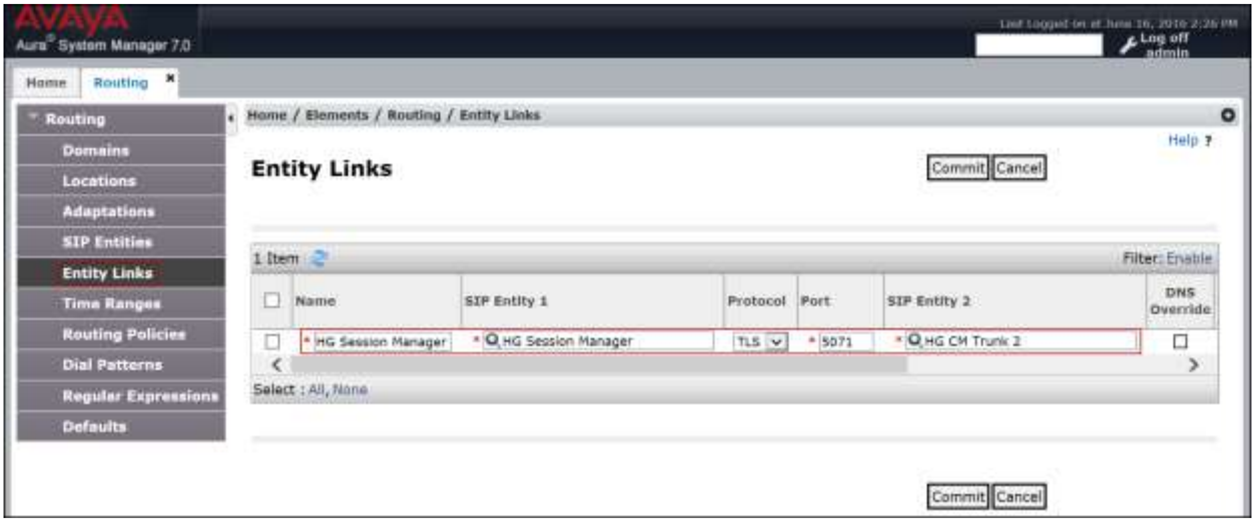
## 6.6. Entity Links

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

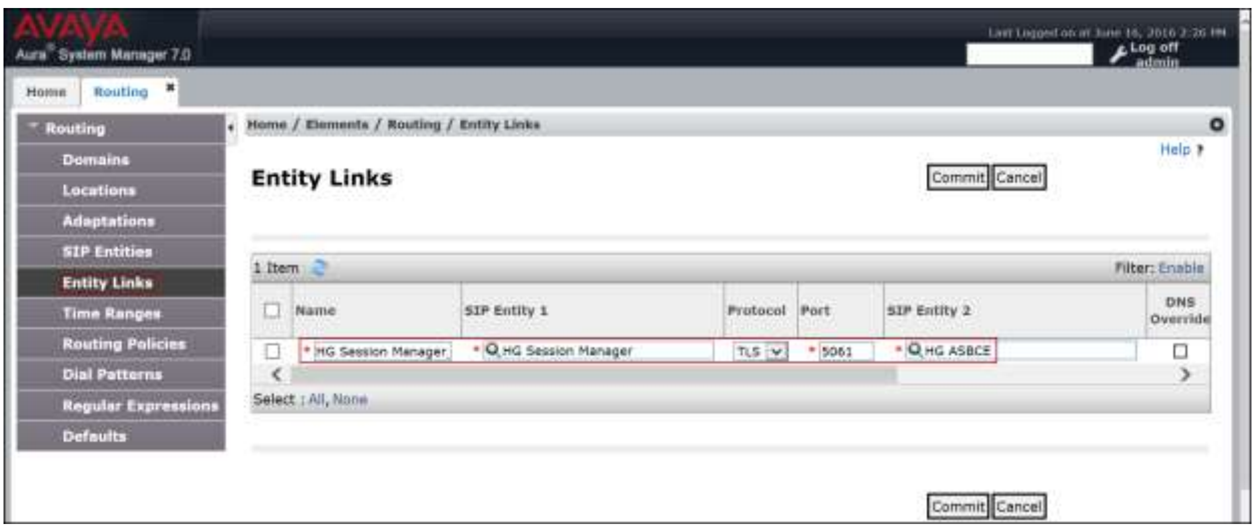
- **Name:** Enter a descriptive name.
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select the transport protocol used for this link. For Communication Manager this was matched to the **Transport Method** defined on the Communication Manager signaling group in **Section 5.6**. For the Avaya SBCE, this was matched to the **Transport** defined on the **Server Configuration** for Session Manager (Call Server) in **Section 7.3.4**.
- **Port:** Port number on which Session Manager will receive SIP requests from the far-end. For Communication Manager, this was matched to the **Far-end Listen Port** defined on the Communication Manager signaling group in **Section 5.6**. For the Avaya SBCE, this was matched to the **Port** defined on the **Server Configuration** for Session Manager (Call Server) in **Section 7.3.4**.
- **SIP Entity 2:** Select the name of the other system. For Communication Manager or the Avaya SBCE select the respective SIP Entity defined in **Section 6.5**.
- **Port:** Port number on which the other system will receive SIP requests from Session Manager. For Communication Manager, this was matched to the **Near-end Listen Port** defined on the Communication Manager signaling group in **Section 5.6**. For the Avaya SBCE, this was matched to the **TLS Port** defined for the private **Signaling Interface** on the Avaya SBCE in **Section 7.5.3**.
- **Connection Policy:** Select ***Trusted***.
- Click **Commit** to save.

The following screens illustrate the entity links to Communication Manager and to the Avaya SBCE.

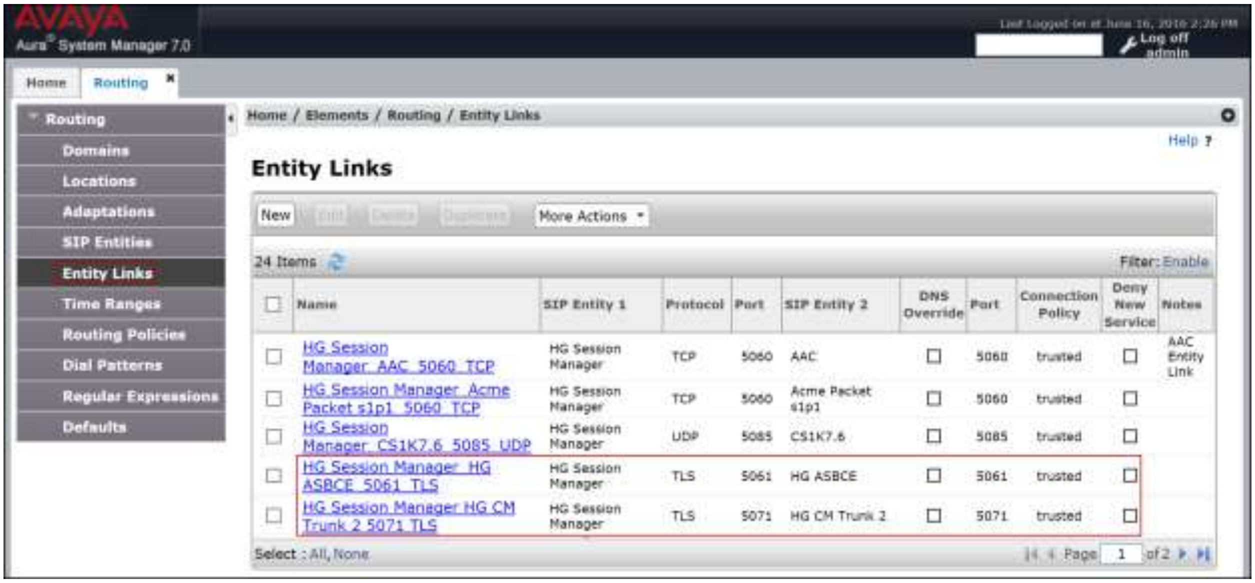
The following screen shows the entity link to Communication Manager:




The following screen shows the entity link to the Avaya SBCE:



The following screen shows the list of the newly added entity links. Note that only the highlighted entity links were created for the compliance test, and are the ones relevant to these Application Notes.



## 6.7. Routing Policies

Routing Policies describe the conditions under which calls are routed to the SIP entities specified in **Section 6.5**. Two routing policies must be added: one for Communication Manager and one for the Avaya SBCE. To add a routing policy, navigate to **Routing → Routing Policies** in the left-hand navigation pane and click on the **New** button in the right pane (not shown). The following screen is displayed. Fill in the following:

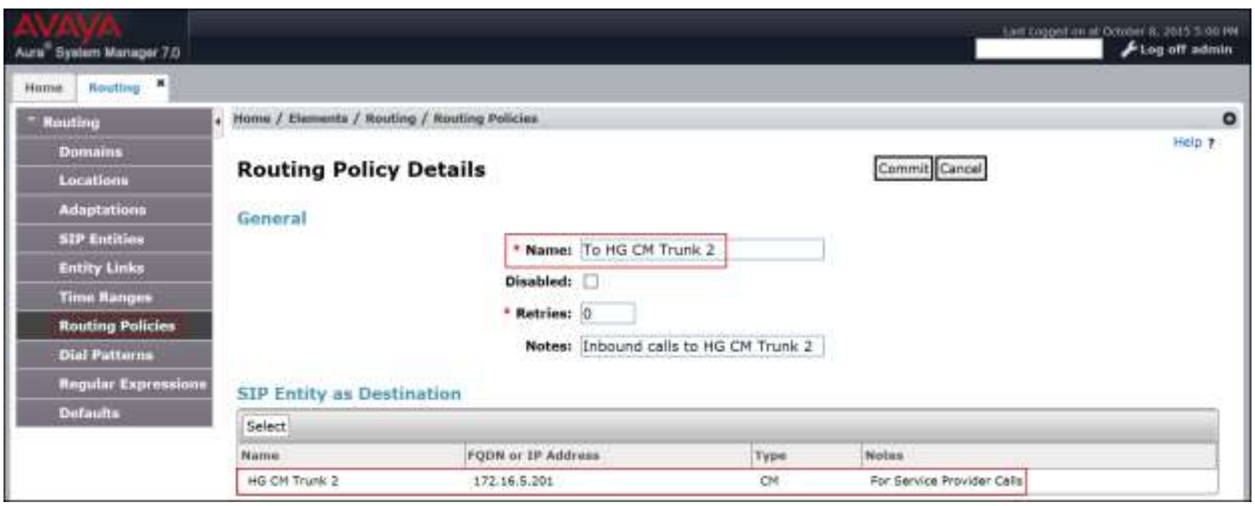
In the **General** section, enter the following values. Use default values for all remaining fields:

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

In the **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 displays on the **Routing Policy Details** page as shown below. Use default values for 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:


## 6.8. Dial Patterns

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

In the **General** section, enter the following values. Use default values for all remaining fields:

- **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).

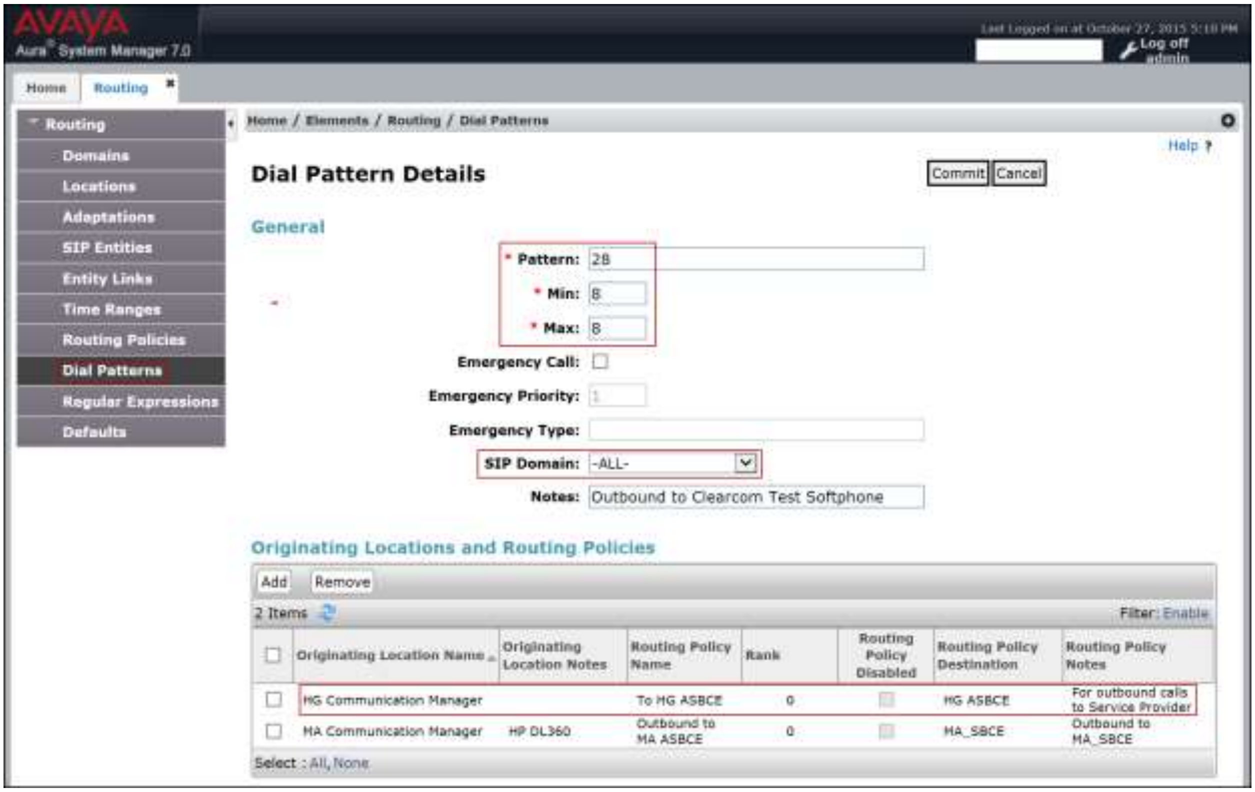
In the **Originating Locations and Routing Policies** section, click **Add**. From the **Originating Locations and Routing Policy List** that appears (not shown), select the appropriate originating location for use in the match criteria. Lastly, select the routing policy from the list that will be used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

- Click **Commit** to save.

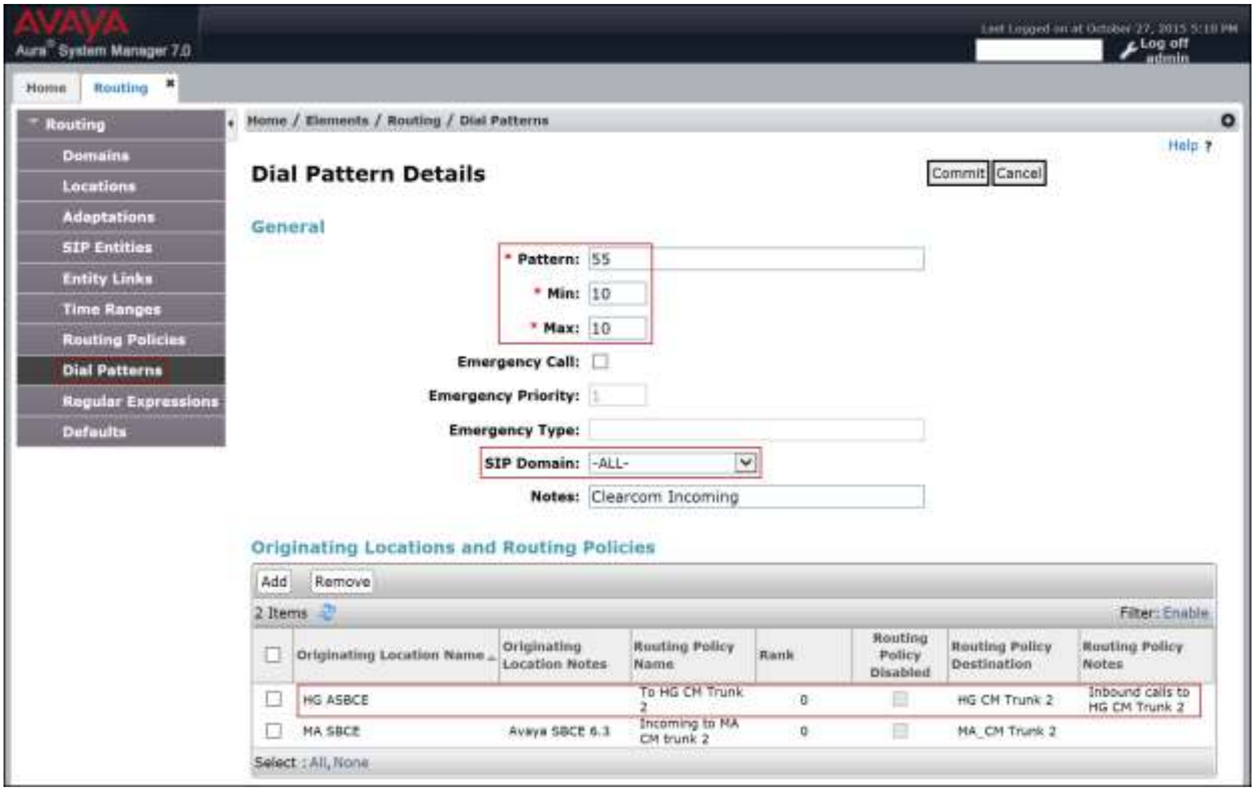
Examples of dial patterns used for the compliance testing are shown below.

The first example shows dial pattern ***28***, with destination SIP Domain of ***-ALL-***, Originating Location Name ***HG Communication Manager*** and Routing Policy name ***To HG ASBCE***. This dial pattern was used for local outbound calls in Mexico.

**Note**: The SIP Domain was set to -ALL- since dial pattern 28 is shared among multiple SIP Domains in the Avaya lab, SIP Domain ***Avaya.lab.com*** could have been used instead.



The following dial pattern used for the compliance testing was for inbound calls to the enterprise. It uses dial pattern ***55*** matching the first two digits of the DID numbers assigned to Communication Manager. This dial pattern was configured with the destination SIP Domain of -***ALL***-, Originating Location Name ***HG ASBCE***, and Routing Policy name ***To HG CM Trunk 2***.




**Note**: The SIP Domain was set to -ALL- since dial pattern 55 is shared among multiple SIP Domains in the Avaya lab, SIP Domain ***Avaya.lab.com*** could have been used instead.

**Note:** The same procedure should be followed to add other required dial patterns.

## 6.9. Add/View Avaya Aura® Session Manager

The creation of a Session Manager element provides the linkage between System Manager and Session Manager. This was most likely done as part of the initial Session Manager installation. To add Session Manager, navigate to **Elements → Session Manager → Session Manager Administration** in the left-hand navigation pane and click on the **New** button in the right pane (not shown). If Session Manager already exists, click **View** (not shown) to view the configuration. Enter/verify the data as described below and shown in the following screen:

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

- **SIP Entity Name:** Select the SIP Entity created for Session Manager.
- **Description**: Add a brief description (optional).
- **Management Access Point Host Name/IP:** Enter the IP address of the Session Manager management interface.

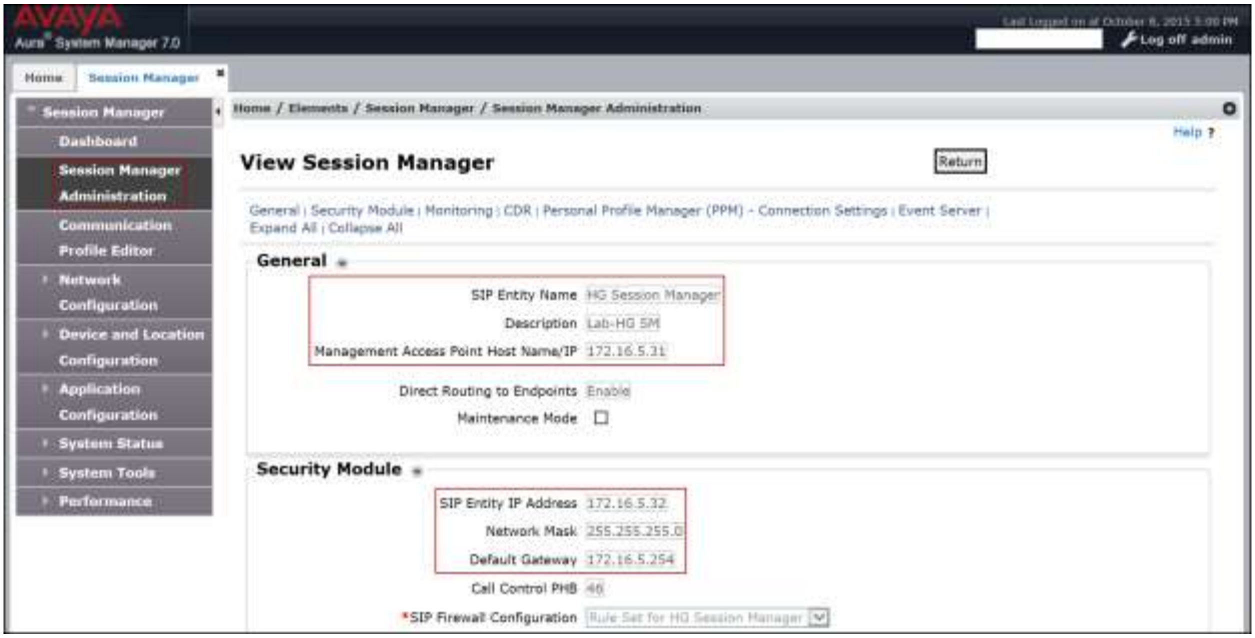
In the **Security Module** section, enter the following values:

- **SIP Entity IP Address:** Should be filled in automatically based on the SIP Entity Name. Otherwise, enter IP address of the Session Manager signaling interface.
- **Network Mask:** Enter the network mask corresponding to the IP address of the Session Manager signaling interface.
- **Default Gateway**: Enter the IP address of the default gateway for Session Manager.

Use default values for the remaining fields.

- Click **Save** (not shown).

The screen below shows the Session Manager values used for the compliance test.



# 7. Configure Avaya Session Border Controller for Enterprise

This section describes the required configuration of the Avaya SBCE to connect to Clearcom's SIP Trunking service.

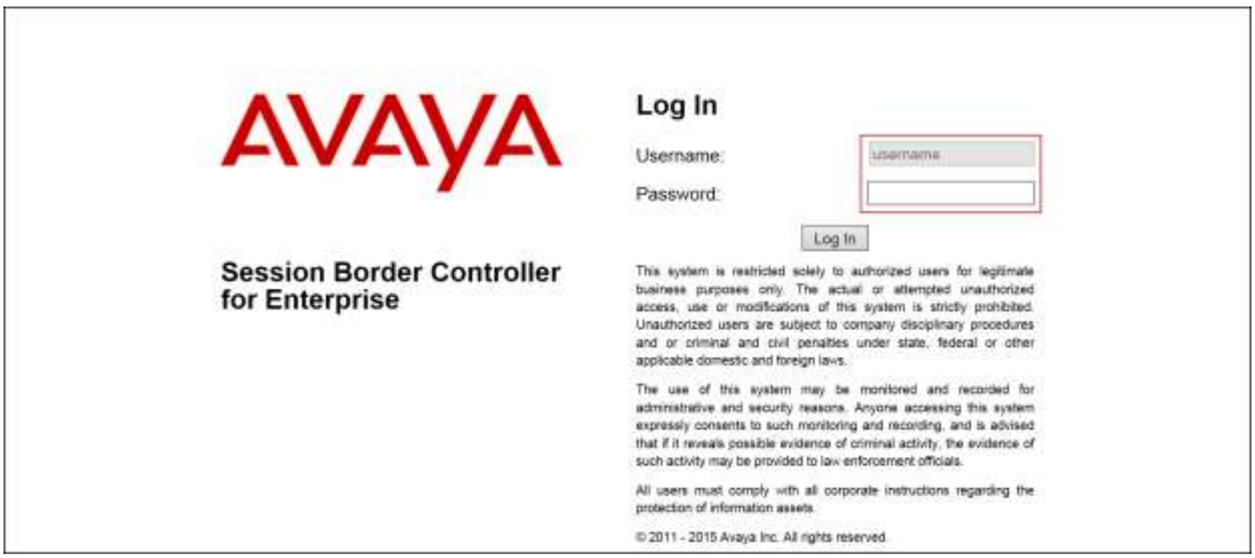
It is assumed that the Avaya SBCE was provisioned and is ready to be used; the configuration shown here is accomplished using the Avaya SBCE web interface.

**Note:** In the following pages, and for brevity in these Application Notes, not every provisioning step will have a screenshot associated with it. Some of the default information in the screenshots that follow may have been cut out (not included) for brevity.

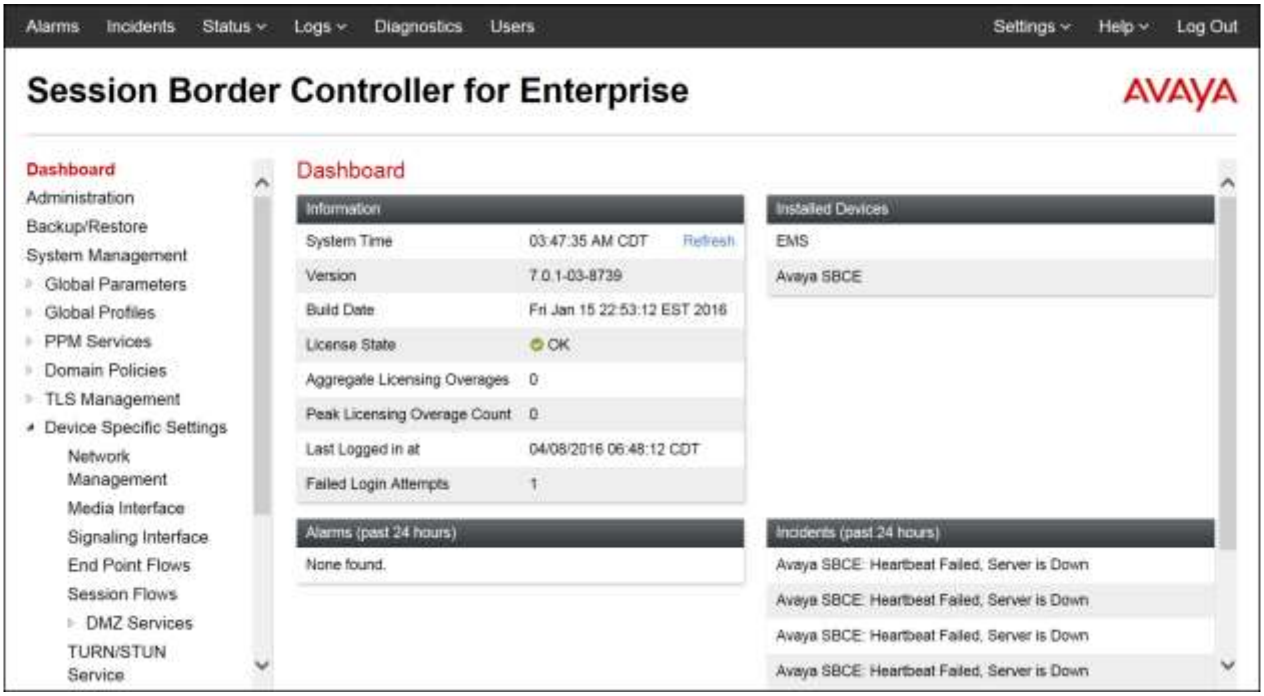
## 7.1. Log in Avaya SBCE

Use a web browser to access the Avaya SBCE web interface, enter https://<ip-addr>/sbc in the address field of the web browser, where <ip-addr> is the management IP address of the Avaya SBCE.

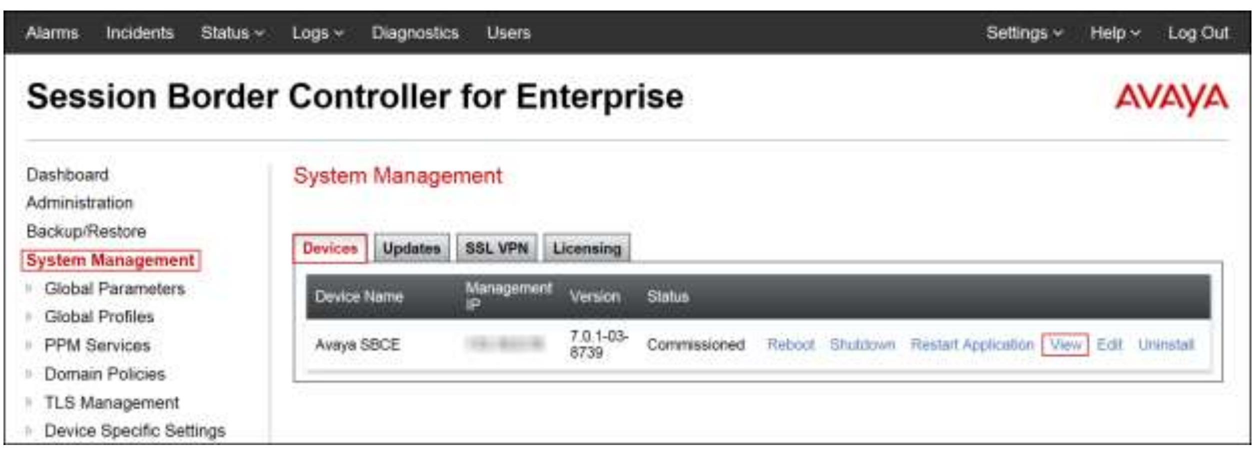
Enter the appropriate credentials and then click **Log In**.



The **Dashboard** main page will appear as shown below.

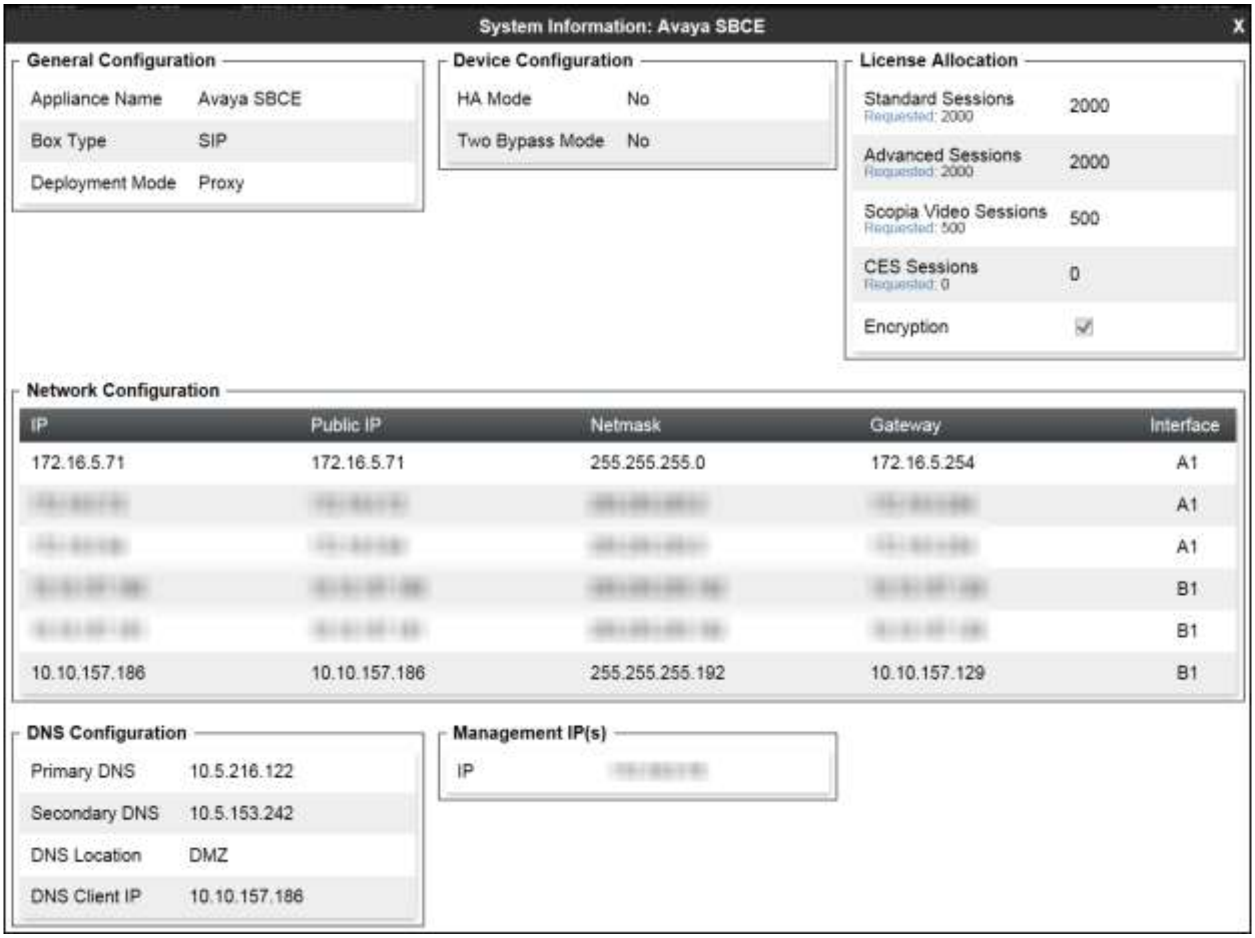



To view the system information that has been configured during installation, navigate to **System Management**. A list of installed devices is shown in the right pane. For the compliance test, a single Device Name **Avaya SBCE** was already added.



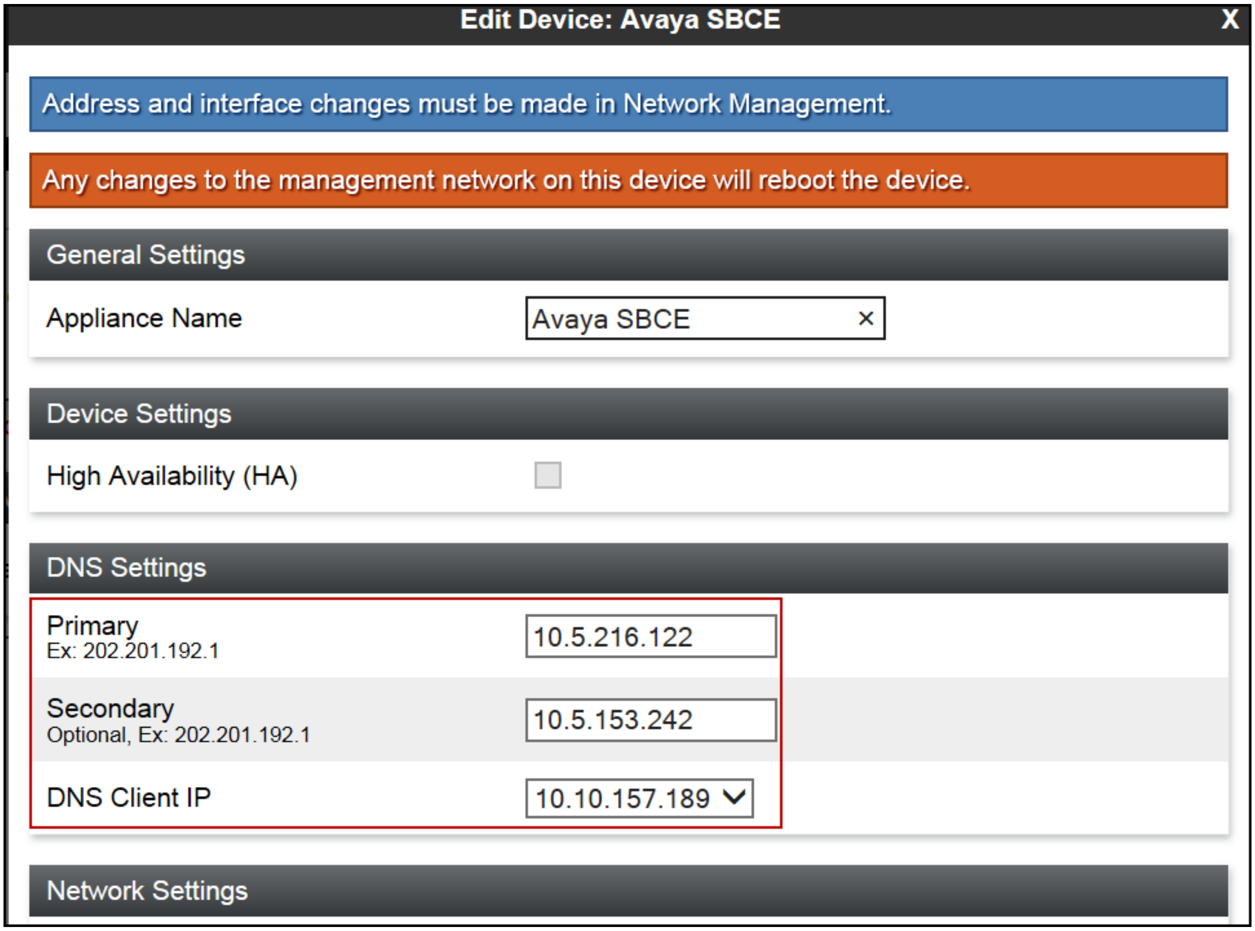
To view the network configuration assigned to the Avaya SBCE, click **View** as shown on the screen above. The **System Information** window is displayed as shown below.

The **System Information** screen shows **Network Configuration**, **DNS Configuration** and **Management IP** information provided during installation and corresponds to **Figure 1**.




On the previous screen, note that the **A1** interface corresponds to the inside interface (Private Network side) and **B1** interface corresponds to the outside interface (Public Network side) of the Avaya SBCE. On the **License Allocation** area of the **System Information**, verify that the number of **Standard Sessions** is sufficient to support the desired number of simultaneous SIP calls across all SIP trunks at the enterprise. The number of sessions and encryption features are primarily controlled by the license file installed. Refer to **Figure 1** for the IP addresses for both the A1 and B1 interfaces on the Avaya SBCE.

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




**IMPORTANT! – During the Avaya SBCE installation, the Management interface, (labeled "M1"), of the Avaya SBCE <u>must</u> be provisioned on a different subnet than either of the Avaya SBCE private and public network interfaces (e.g., A1 and B1).**

## 7.2. TLS Management

### 7.2.1. TLS Certificates

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. UC-Sec security products utilize TLS primarily to facilitate secure communications with remote servers.

This section describes the TLS profiles that were created for the Avaya SBCE, including the following:

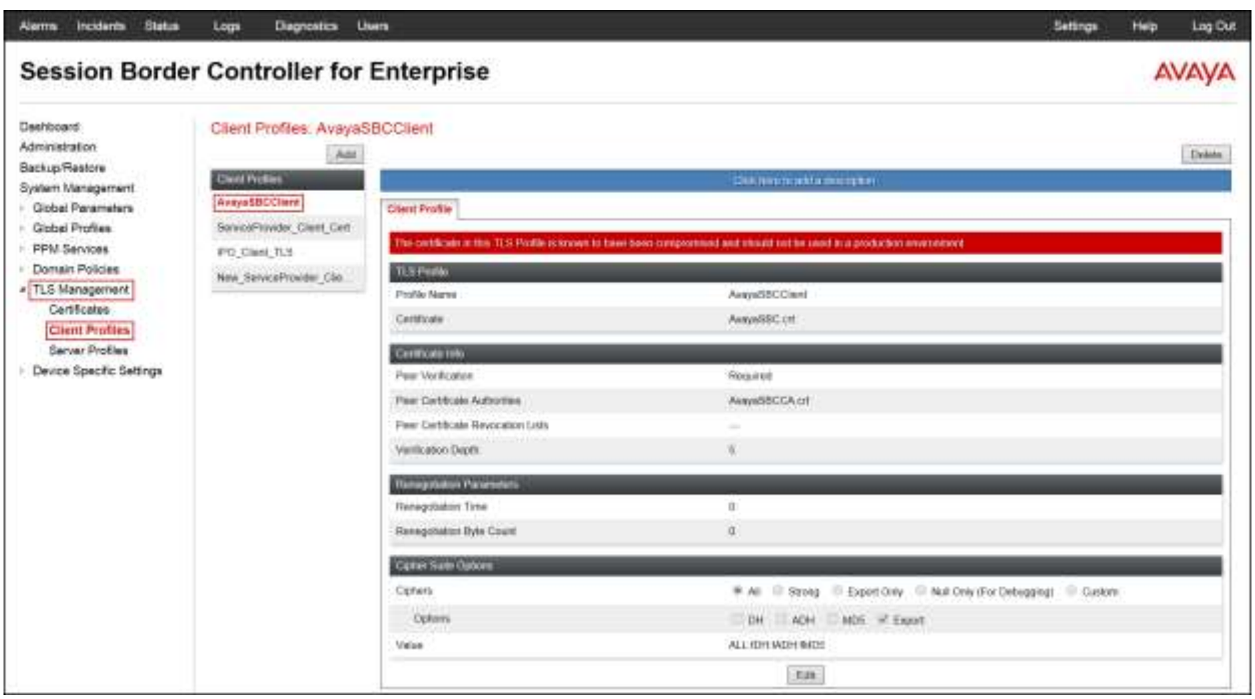
- Create TLS client and server profiles to identify which certificates will be used in various TLS connections on the Avaya SBCE.

It is assumed that generation and installation of certificates on the Avaya SBCE, and the exchange of TLS CA certificates with the Service Provider, have been previously completed, and is not discussed in this document. Refer to items [**8**] in **Section 11**.

### 7.2.2. TLS Client Profile – Avaya Session Manager

For the TLS client profile toward Session Manager, the pre-existing (pre-installed) demo TLS client profile by the name ***AvayaSBCClient*** was used.

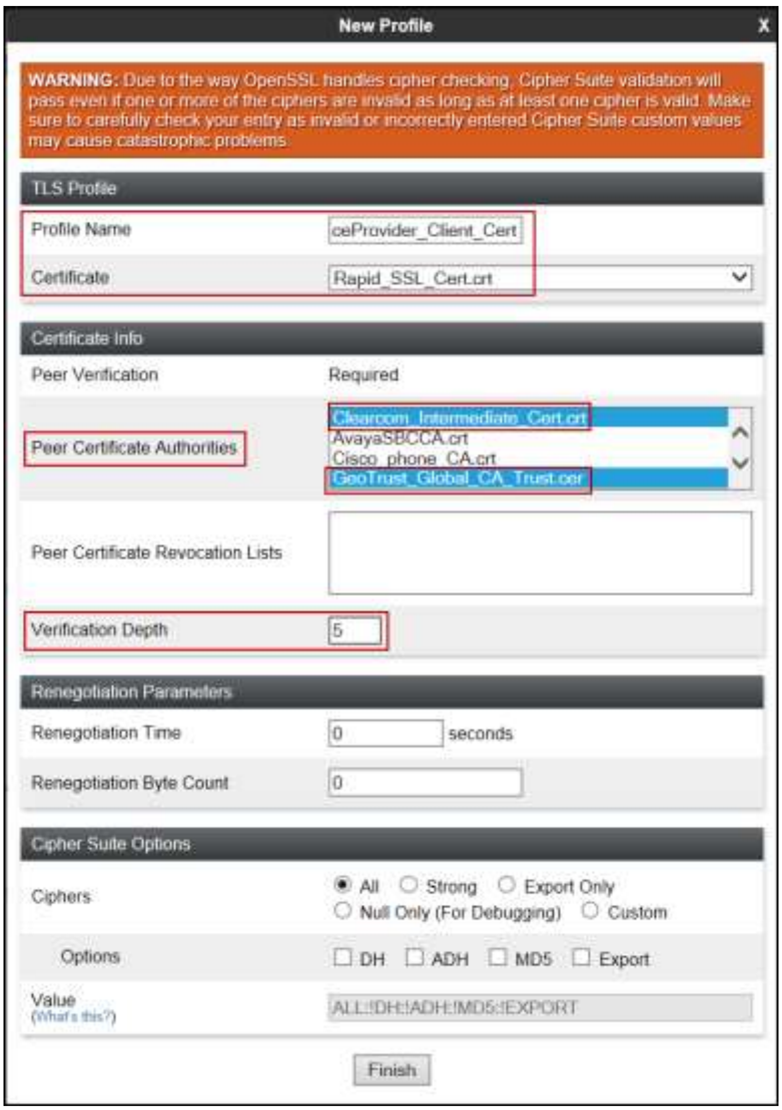
The following screen capture shows the pre-existing TLS client Profile **AvayaSBCClient**.



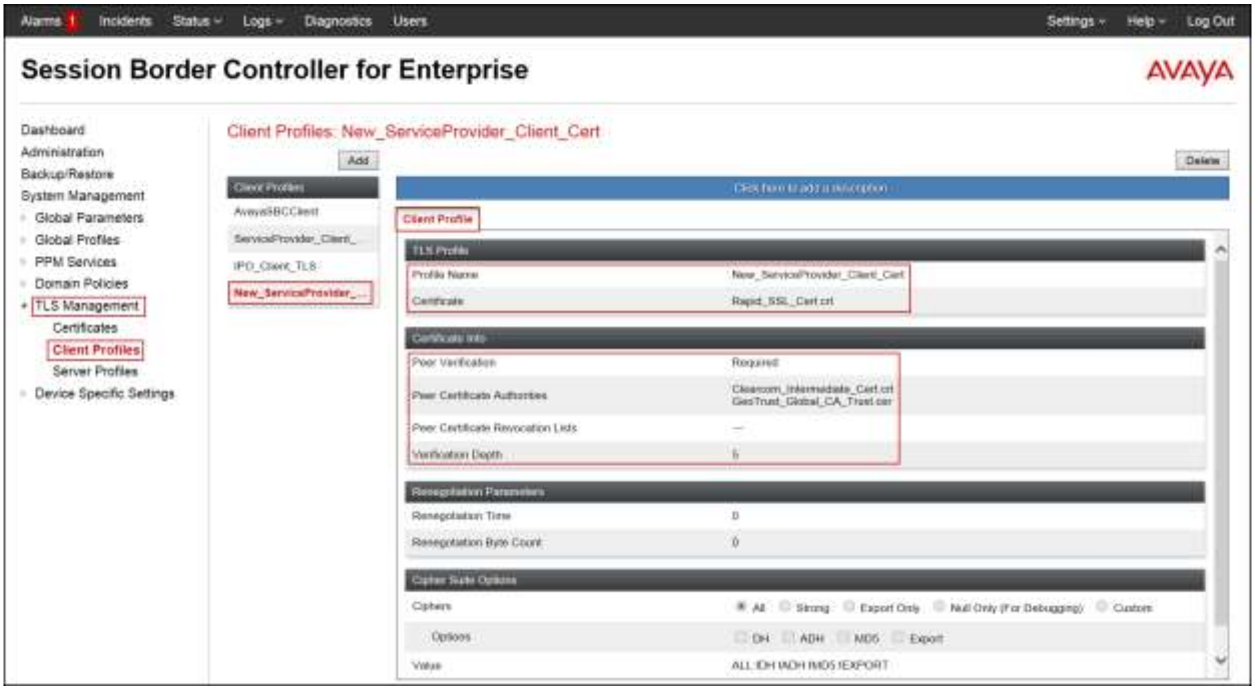
### 7.2.3. TLS Client Profile – Service Provider

To create a TLS client profile toward the Service Provider, navigate to **TLS Management → Client Profiles** and click **Add**. Configure the following parameters:

- Under TLS Profile enter the **profile name**; the name of ***New_ServiceProvider_Client_Cert*** was used in this example.
- Under TLS Profile select the **Certificate** to be used from the pull down menu; ***Rapid_SSL_Cert.crt*** was selected in the sample configuration.
- Under Certificate Info, by using Crtl+Click, select the CA certificates to be used for the **Peer Certificate Authorities** field, ***Clearcom_Intermediate_Cert.crt*** and ***GeoTrust_Global_CA_Trust.cer*** were selected in the sample configuration.
- Set the **Verification Depth** to ***5***.
- Default values can be used for the remaining fields.
- Click **Finish**.



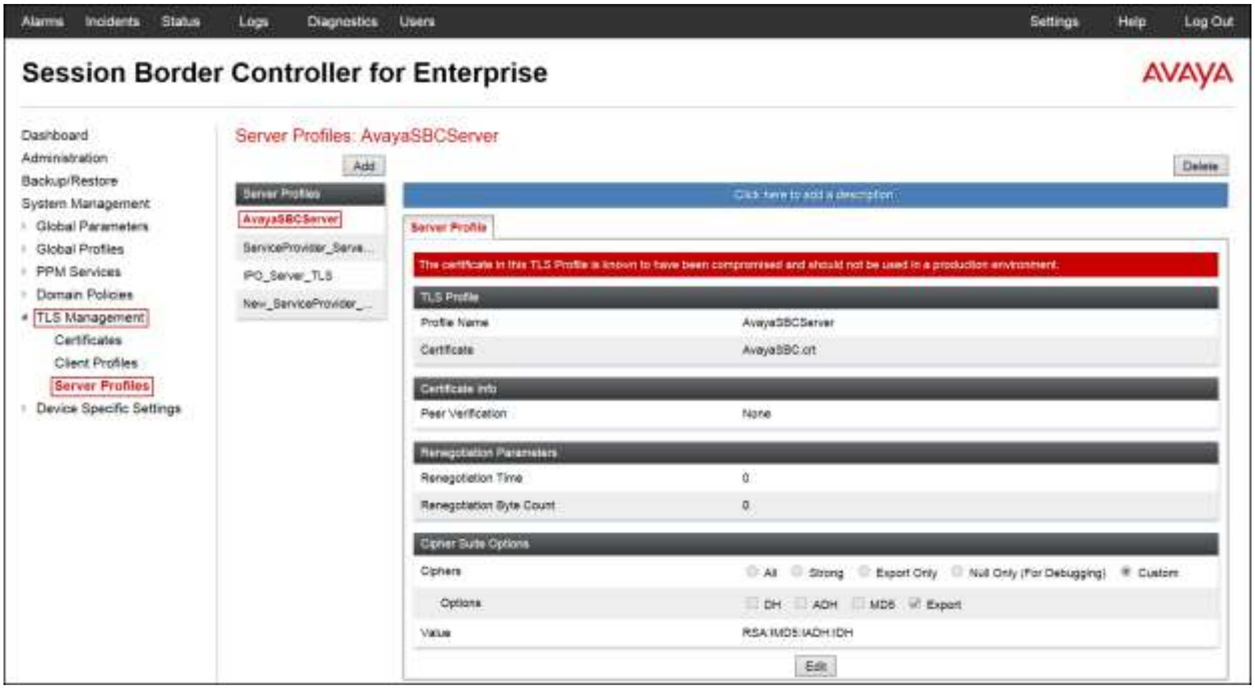
The following screen capture shows the newly created **New_ServiceProvider_Client_Cert** client Profile.



### 7.2.4. TLS Server Profile – Avaya Session Manager

For the TLS server profile toward Session Manager, the pre-existing (pre-installed) demo TLS server profile by the name ***AvayaSBCServer*** was used.

The following screen capture shows the pre-existing TLS server Profile **AvayaSBCServer**.

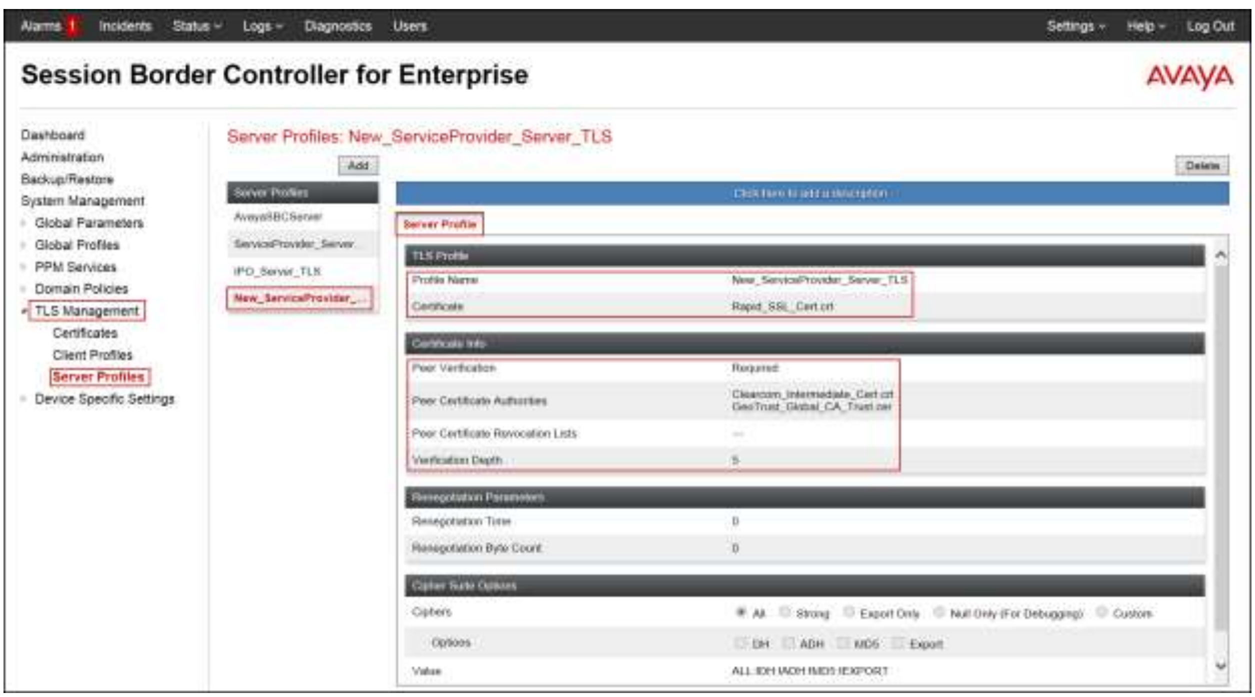


### 7.2.5. TLS Server Profile – Service Provider

To create a TLS server profile toward the Service Provider, navigate to **TLS Management → Server Profiles** and click **Add**. Configure the following parameters:

- Under TLS Profile enter the **profile name**; the name of ***New_ServiceProvider_Server_TLS*** was used in this example.
- Under TLS Profile select the **Certificate** to be used from the pull down menu; ***Rapid_SSL_Cert.crt*** was selected in the sample configuration.
- Under Certificate Info, **Peer Verification**, select **Required** from the pull down menu.
- Under Certificate Info, by using Crtl+Click, select the CA certificates to be used for the **Peer Certificate Authorities** field, ***Clearcom_intermediate_Cert.crt*** and ***GeoTrust_Global_CA_Trust.cer*** were selected in the sample configuration.
- Set the **Verification Depth** to ***5***.
- Default values can be used for the remaining fields.
- Click **Finish**.

The following screen capture shows the newly created **New_ServiceProvider_Server_TLS** server Profile.



## 7.3. Global Profiles

The Global Profiles Menu, on the left navigation pane, allows the configuration of parameters across all Avaya SBCE appliances.

### 7.3.1. Server Interworking Avaya-SM

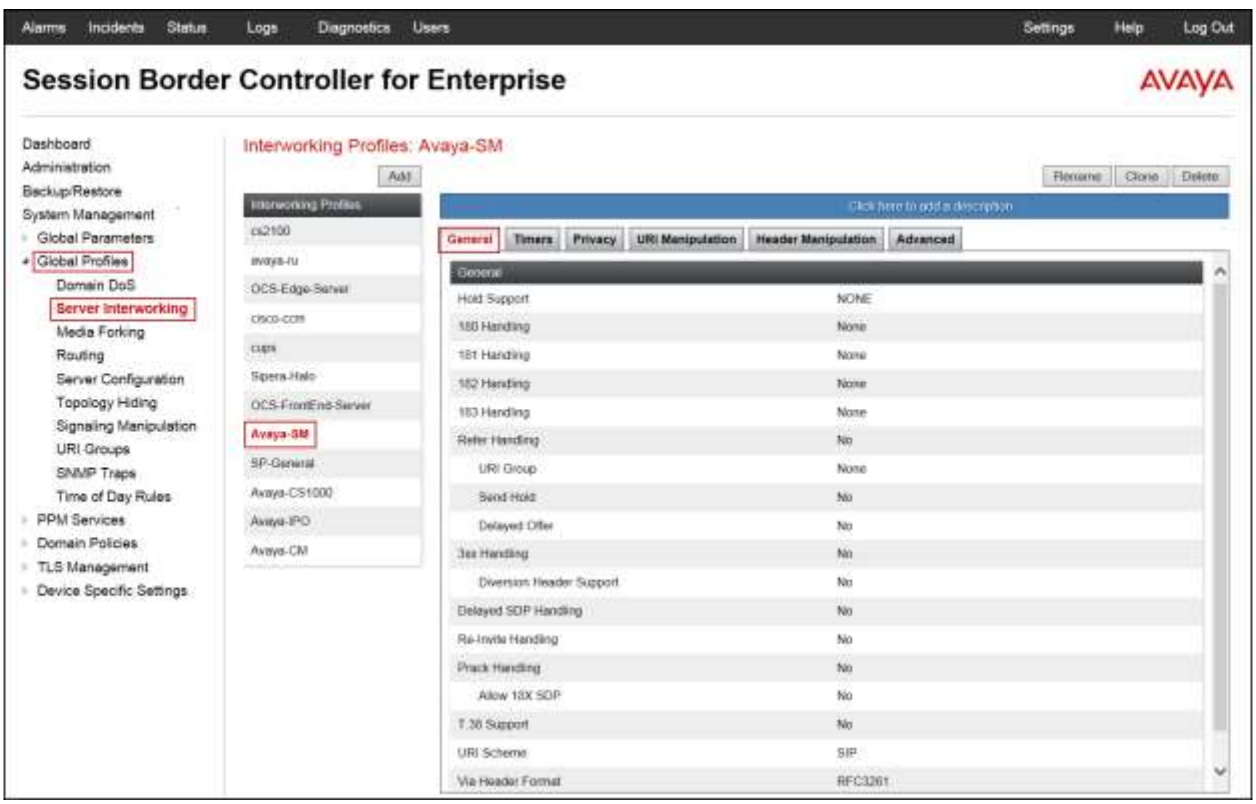
Interworking Profile features are configured to facilitate interoperability of implementations between enterprise SIP-enabled solutions and different SIP trunk Service Providers.

Several profiles have been already pre-defined and they populate the list under **Interworking Profiles** on the screen below. If a different profile is needed, a new Interworking Profile can be created, or an existing default profile can be modified or “cloned”. Since directly modifying a default profile is generally not recommended, for the test configuration the default **avaya-ru** profile was duplicated, or “cloned”, and then modified to meet specific requirements for the enterprise SIP-enabled solution.

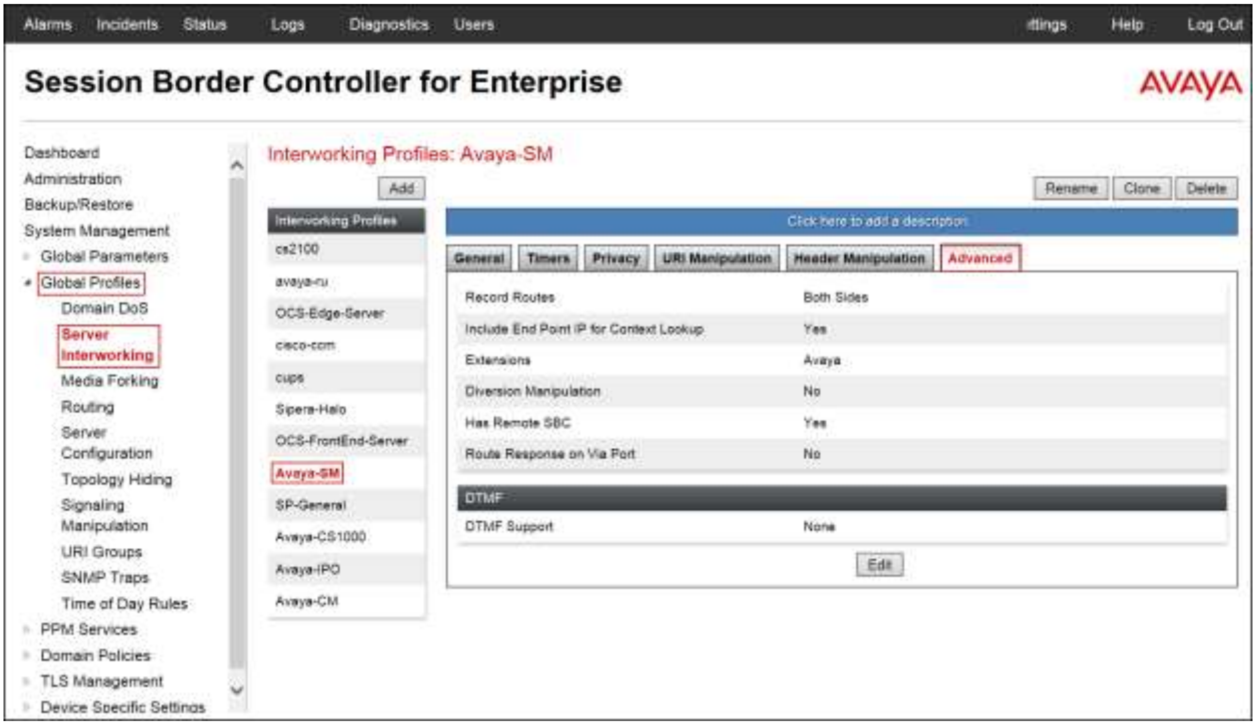
On the left navigation pane, select **Global Profiles → Server Interworking**. From the **Interworking Profiles** list, select **avaya-ru**. Click **Clone** on top right of the screen.

Enter the new profile name in the **Clone Name** field; the name of ***Avaya-SM*** was chosen in this example. Click **Finish** (not shown).

The following screen capture shows the **General** tab of the newly created **Avaya-SM** Server Interworking Profile.



The following screen capture shows the **Advanced** tab of the newly created **Avaya-SM** Server Interworking Profile.



### 7.3.2. Server Interworking SP-General

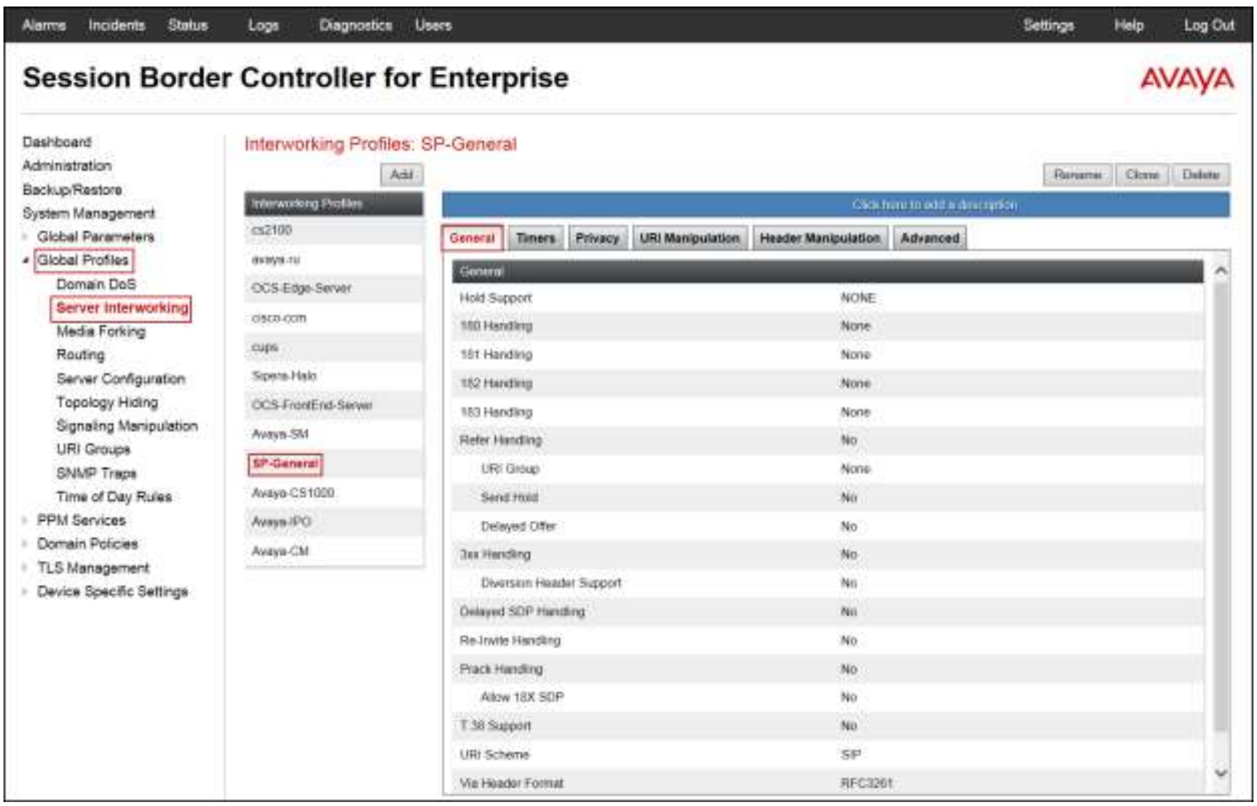
A second Server Interworking profile named **SP-General** was created for the Service Provider.

On the left navigation pane, select **Global Profiles → Server Interworking**. From the **Interworking Profiles** list, select **Add**.

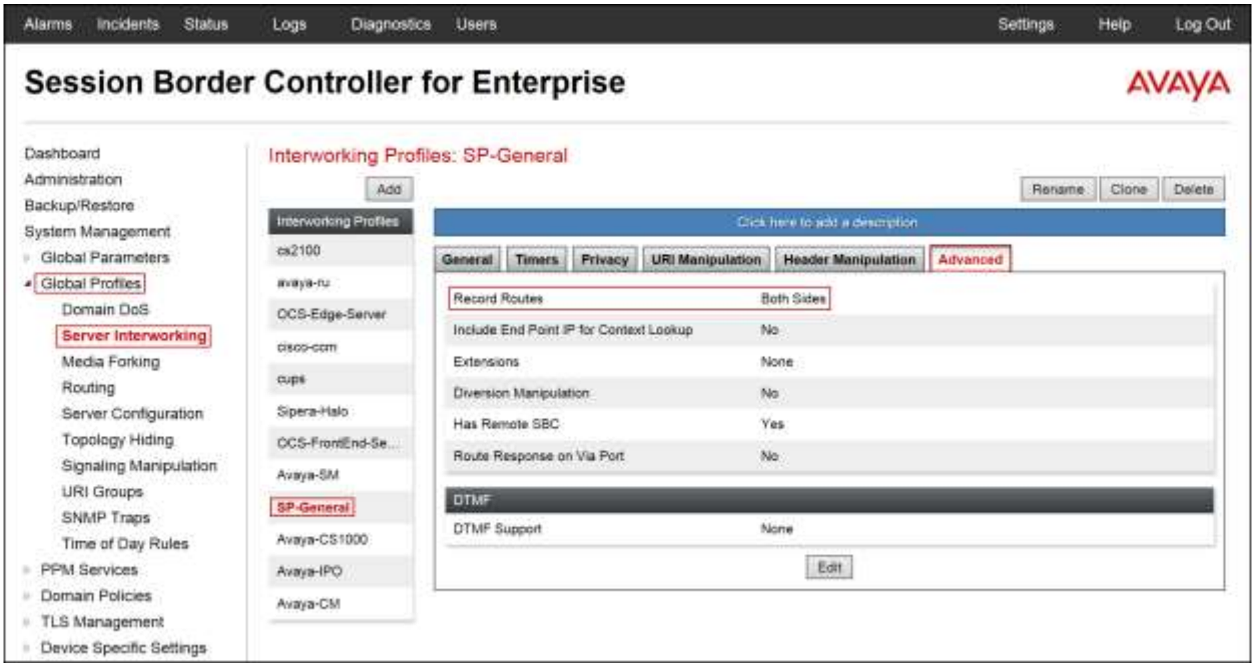
Enter the new profile name (not shown); the name of ***SP-General*** was chosen in this example. Click **Next**:

- Leaving other fields with their default values, click **Next** until the Advanced tab is reached, check ***Both Sides*** then click **Finish** on the Advanced tab.

The following screen capture shows the **General** tab of the newly created **SP-General** Server Interworking Profile.



The following screen capture shows the **Advanced** tab of the newly created **SP-General** Server Interworking Profile.



### 7.3.3. Signaling Manipulation

The Signaling Manipulation feature of the Avaya SBCE allows an administrator to perform granular header manipulations on the headers of the SIP messages, which sometimes is not possible by direct configuration on the web interface. This ability to configure header manipulation in such a highly flexible manner is achieved by the use of a proprietary scripting language called SigMa.

The script can be created externally as a regular text file and imported in the Signaling Manipulation screen, or they can be written directly in the page using the embedded Sigma Editor. In the reference configuration, the Editor was used. A detailed description of the structure of the SigMa scripting language and details on its use is beyond the scope of these Application Notes. Consult **[8]** in the **References** section for more information on this topic.

Sigma scripts were created during the compliance test to correct the following interoperability issues (refer to **Section 2.2**):

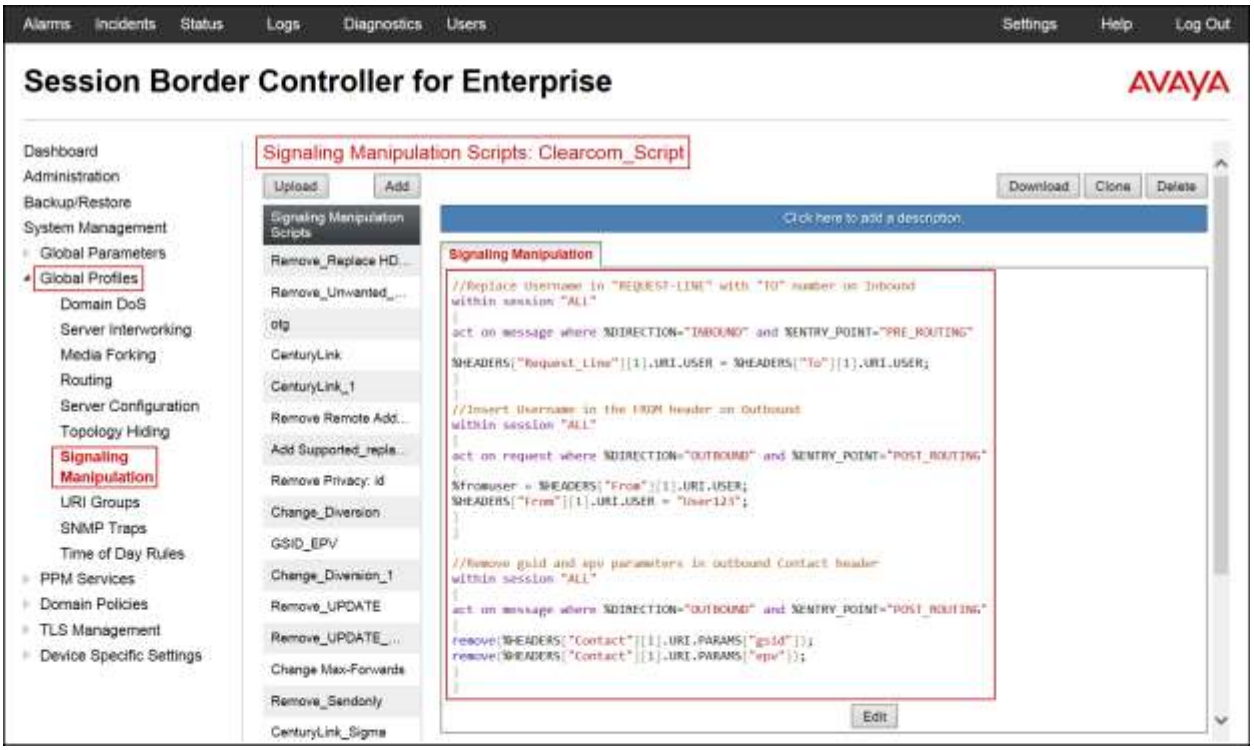
- Include the SIP trunk credential's username in the "From" header of all outbound calls.
- Copy the destination DID number present in the "To" header of incoming calls to the "Request-URI" header.
- Remove the "gsid" and "epv" parameters from outbound "Contact" headers.

The script will later be applied to the Server Configuration profile corresponding to the service provider in **Section 7.3.4**.

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 ***Clearcom_Script*** was chosen in this example.
- Copy the complete script from **Appendix A**.
- Click **Save**.

The following screen capture shows the **Clearcom_Script** script after it was added.



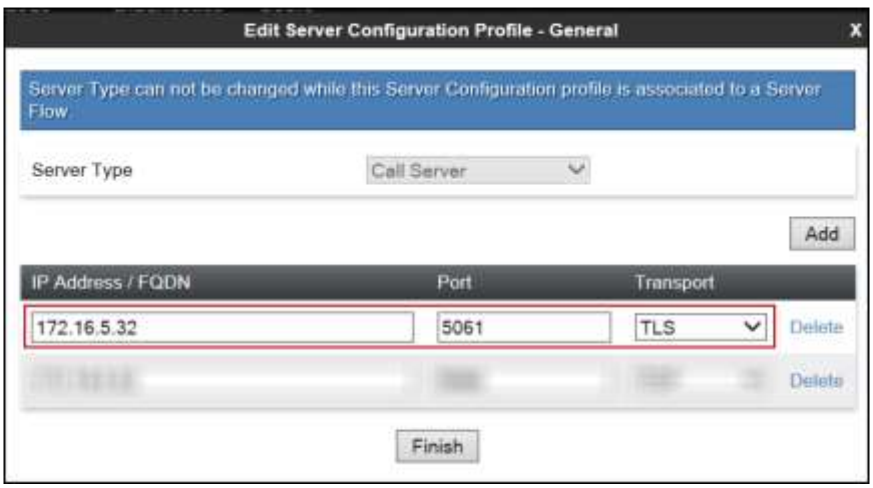
### 7.3.4. Server Configuration

Server Profiles should be created for the Avaya SBCE's two peers, the Call Server (Session Manager) and the Trunk Server which is the SIP Proxy at the Service Provider's network.

To add the profile for the Call Server, from the **Global Profiles** menu on the left-hand navigation pane, select **Server Configuration**. Click **Add** in the **Server Profiles** section and enter the profile name: ***Session Manager***.

On the **Edit Server Configuration Profile – General** window:

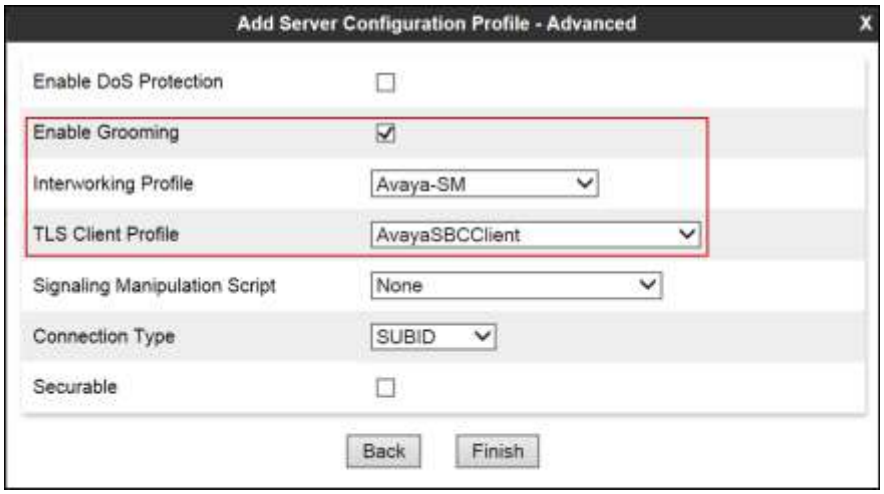
- **Server Type**: select ***Call Server***.
- **IP Address / FQDN:** ***172.16.5.32*** (IP Address of the Session Manager SIP entity).
- **Port:** ***5061*** (This port must match the port number defined in **Section 6.6**).
- **Transports:** Select ***TLS***.
- Click **Next**.



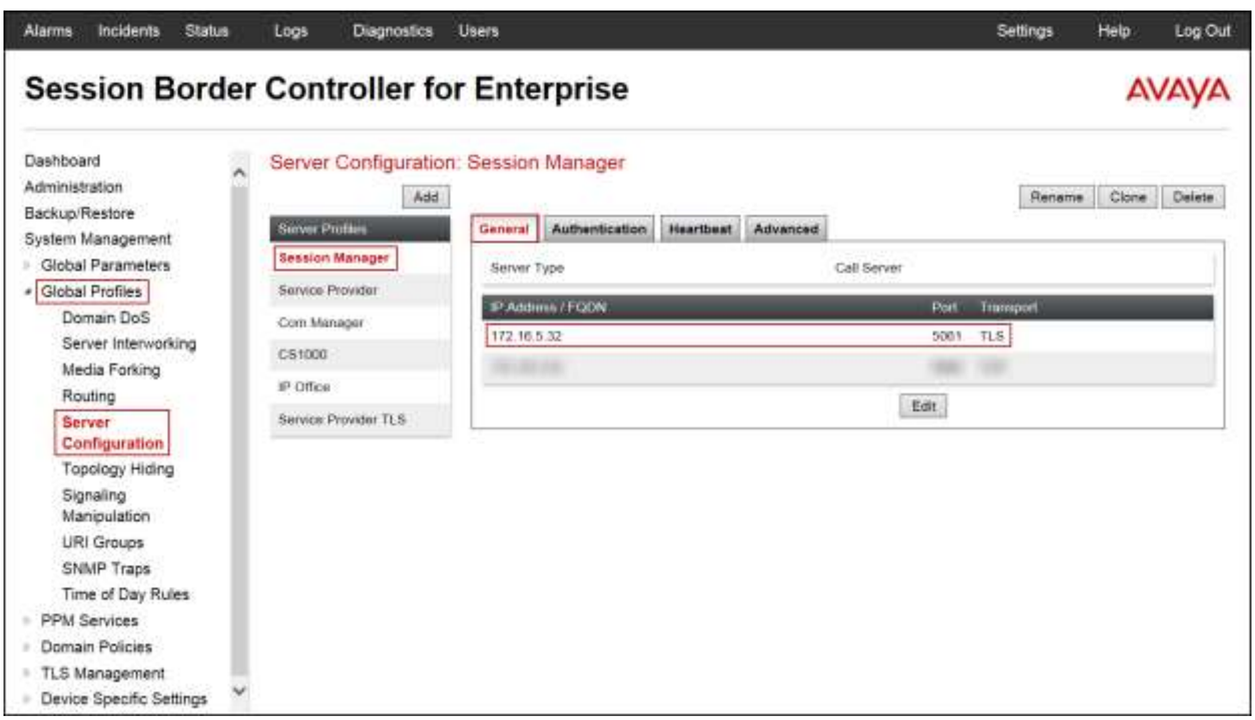

- Click **Next** in the **Add Server Configuration Profile - Authentication** window (not shown).
- Click **Next** in the **Add Server Configuration Profile - Heartbeat** window (not shown).

On the **Add Server Configuration Profile - Advanced** window:

- Check ***Enable Grooming***.
- Select ***Avaya-SM*** from the **Interworking Profile** drop down menu.
- Select ***AvayaSBCClient*** from the **TLS Client Profile** drop down menu.
- Leave the **Signaling Manipulation Script** at the default ***None***.
- Click **Finish**.




The following screen capture shows the **General** tab of the newly created **Session Manager** Server Profile.

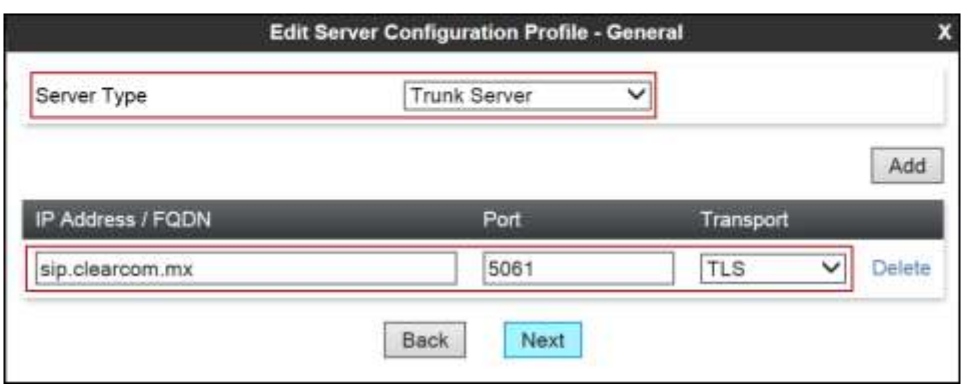


The following screen capture shows the **Advanced** tab of the newly created **Session Manager** Server Profile.


To add the profile for the Trunk Server, from the **Server Configuration** screen, click **Add** in the **Server Profiles** section and enter the profile name: ***Service Provider TLS***.

On the **Edit Server Configuration Profile – General** window

- **Server Type:** select ***Trunk Server***.
- **IP Address/FQDN:** ***sip.clearcom.mx*** (the Fully Qualified Domain Name of the service provider SIP proxy server. This information was provided by Clearcom.).
- **Port:** ***5061***.
- **Transports**: Select ***TLS***.
- Click **Next**.



On the **Add Server Configuration Profile - Authentication** window:

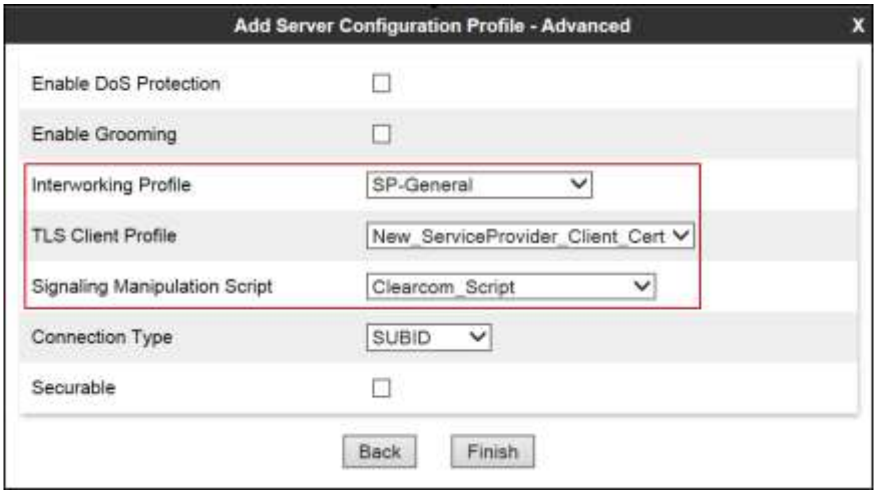
- Check the ***Enable Authentication*** box.
- Enter the **User Name** credential provided by the service provider for SIP trunk registration.
- Enter the **Realm** credential provided by the service provider for SIP trunk registration. Note that the Service Provider's Domain Name was used (Must be entered, currently cannot be detected automatically from the challenge).
- Enter **Password** credential provided by the service provider for SIP trunk registration.
- Click **Next**.

On the **Add Server Configuration Profile - Heartbeat** window:

- Check the **Enable Heartbeat** box.
- Under **Method**, select ***REGISTER*** from the drop down menu.
- **Frequency:** Enter the amount of time (in seconds) between REGISTER messages that will be sent from the enterprise to the Service Provider Proxy Server to refresh the registration binding of the SIP trunk. This value should be chosen in consultation with the service provider, ***120*** seconds was the value used during the compliance test.
- The **From URI** and **To URI** entries for the REGISTER messages are built using the following:
  - **From URI**: Use the **User Name** entered above in the **Authentication** screen (***User123***) and the Service Provider's domain name (***clearcom.mx***), as shown on the screen below.
  - **To URI**: Use the **User Name** entered above in the **Authentication** screen (***User123***) and the Service Provider's domain name (***clearcom.mx***), as shown on the screen below.
- Click **Next**.

On the **Add Server Configuration Profile - Advanced** window:

- Select ***SP-General*** from the **Interworking Profile** drop down menu.
- Select ***New_ServiceProvider_Client_Cert*** from the **TLS Client Profile** drop down menu
- Select the ***Clearcom_Script*** from the **Signaling Manipulation Script** drop down menu (**Section 7.4.3**.).
- Click **Finish**.

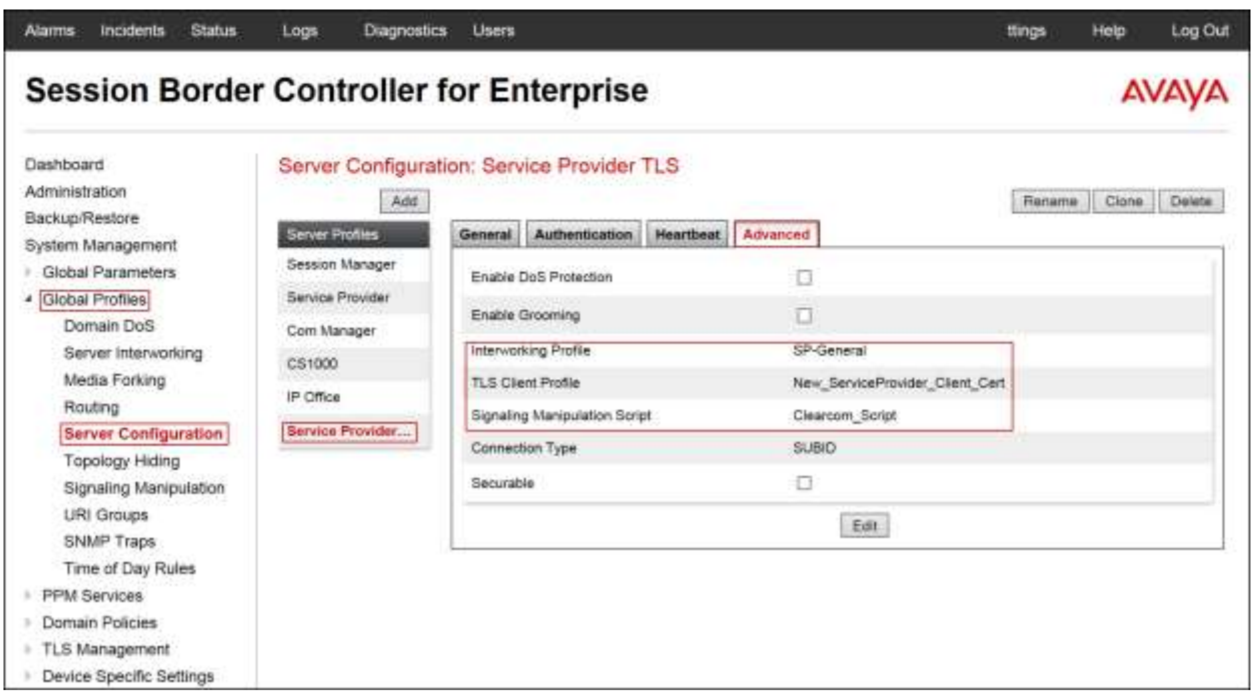



The following screen capture shows the **General** tab of the newly created **Service Provider TLS** Server Configuration Profile.

The following screen capture shows the **Authentication** tab of the newly created **Service Provider TLS** Server Configuration Profile.

The following screen capture shows the **Heartbeat** tab of the newly created **Service Provider TLS** Server Configuration Profile.


The following screen capture shows the **Advanced** tab of the newly created **Service Provider TLS** Server Configuration Profile.



## 7.3.5. Routing Profiles

Routing profiles define a specific set of routing criteria that are used, in conjunction with other types of domain policies, to determine the route that SIP packets should follow to arrive at their intended destination.

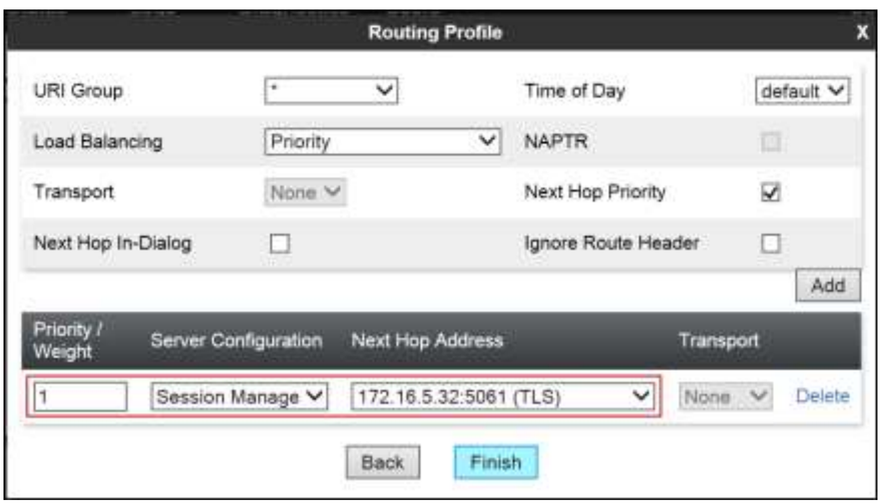
Two Routing profiles were created; one for inbound calls, with Session Manager as the destination, and the second one for outbound calls, which are sent to the service provider.

To create the inbound route, from the **Global Profiles** menu on the left-hand side:

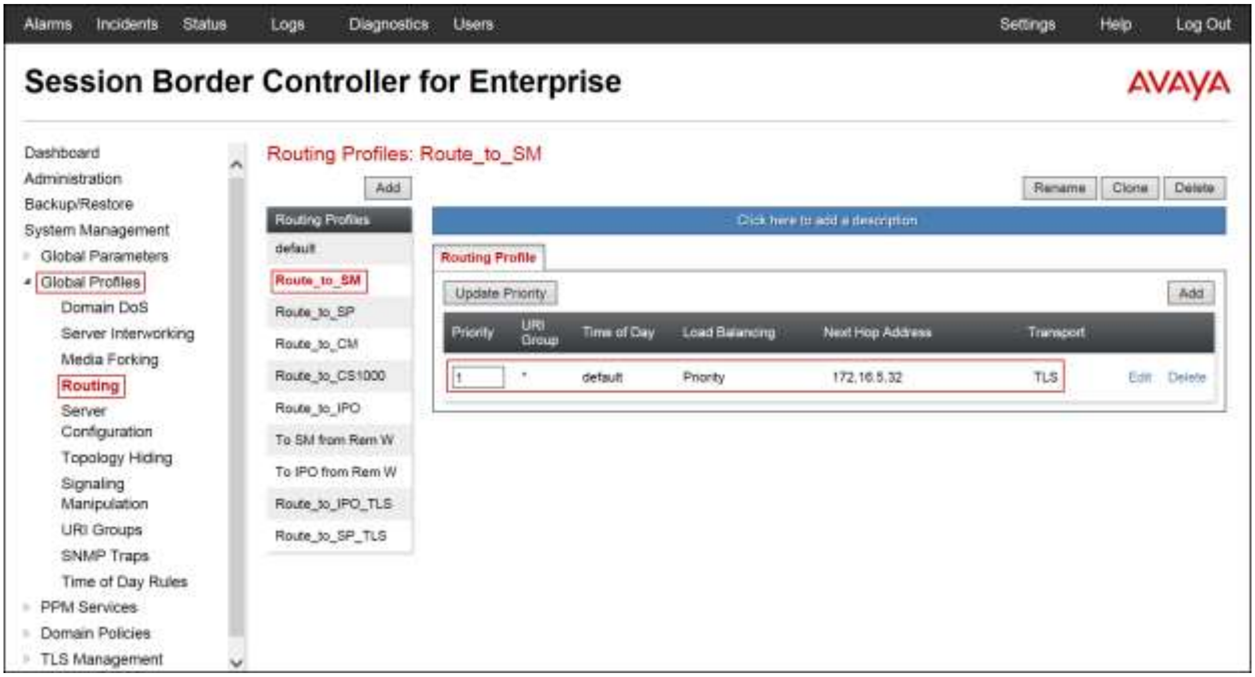
- Select **Routing**.
- Click **Add** in the **Routing Profiles** section.
- Enter Profile Name: ***Route_to_SM***.
- Click **Next**.

On the **Routing Profile** screen complete the following:

- Click on the **Add** button to add a **Next-Hop Address**.
- **Priority / Weight**: ***1***
- **Server Configuration**: Select ***Session Manager***.
- The **Next Hop Address** is populated automatically with ***172.16.5.32:5061 (TLS)*** (Session Manager IP address, Port and Transport).
- Click **Finish**.



The following screen capture shows the newly created **Route_to_SM** Routing Profile.

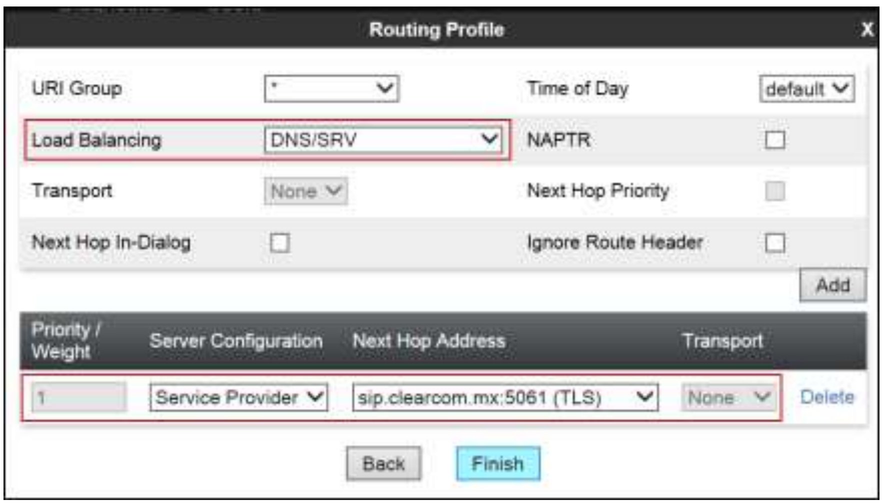


Similarly, for the outbound route:

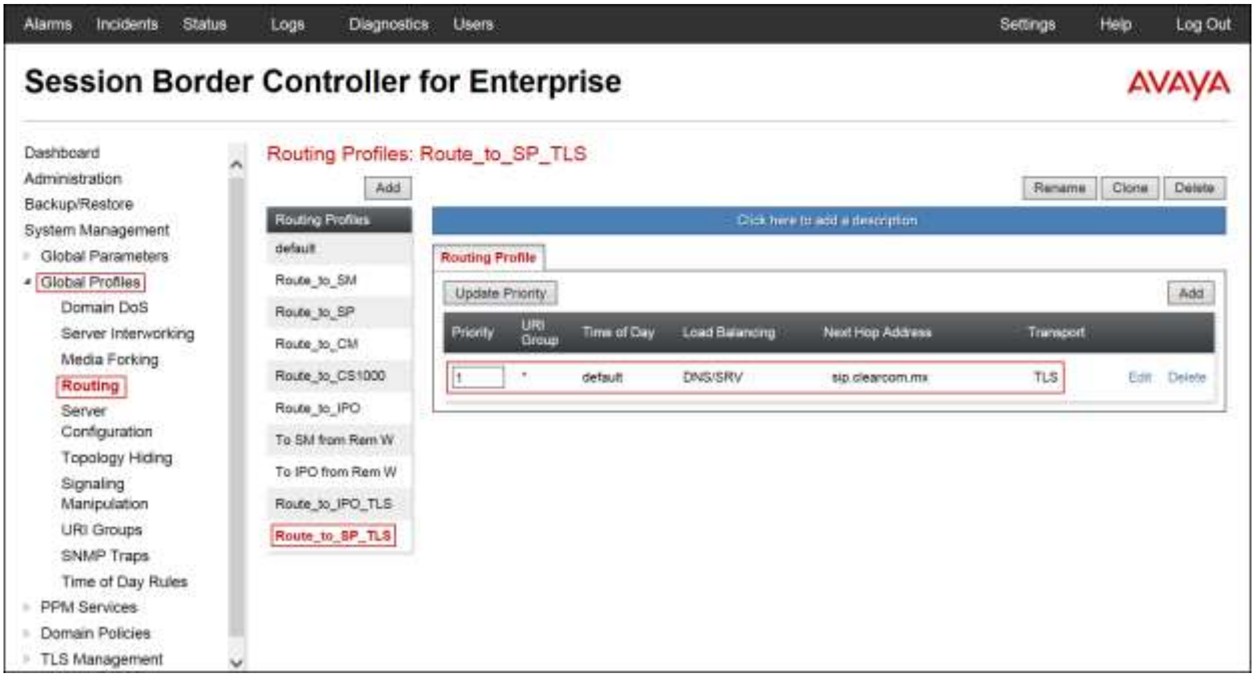
- Select **Routing**.
- Click **Add** in the **Routing Profiles** section.
- Enter Profile Name: ***Route_to_SP_TLS***.
- Click **Next**.

On the **Routing Profile** screen complete the following:

- **Load Balancing**: Select ***DNS/SRV***
- **Priority / Weight**: ***1***
- Click on the **Add** button to add a **Next-Hop Address**.
- **Server Configuration**: Select ***Service Provider***.
- The **Next Hop Address** is populated automatically with ***sip.clearcom.mx:5061 (TLS)*** (Service Provider FQDN, Port and Transport).
- Click **Finish**.



The following screen capture shows the newly created **Route_to_SP_TLS** Routing Profile.



### 7.3.6. Topology Hiding

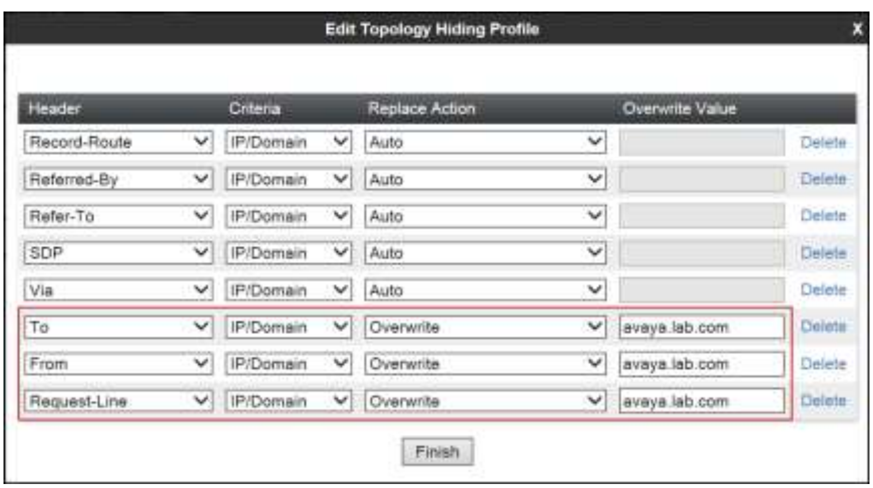
Topology Hiding is a security feature which allows changing several parameters of the SIP packets, preventing private enterprise network information from being propagated to the un-trusted public network.

Topology Hiding can also be used as an interoperability tool to adapt the host portion in SIP headers like To, From, Request-URI, Via, Record-Route and SDP to the IP addresses or domains expected by Session Manager and the SIP trunk Service Provider, allowing the call to be accepted in each case.

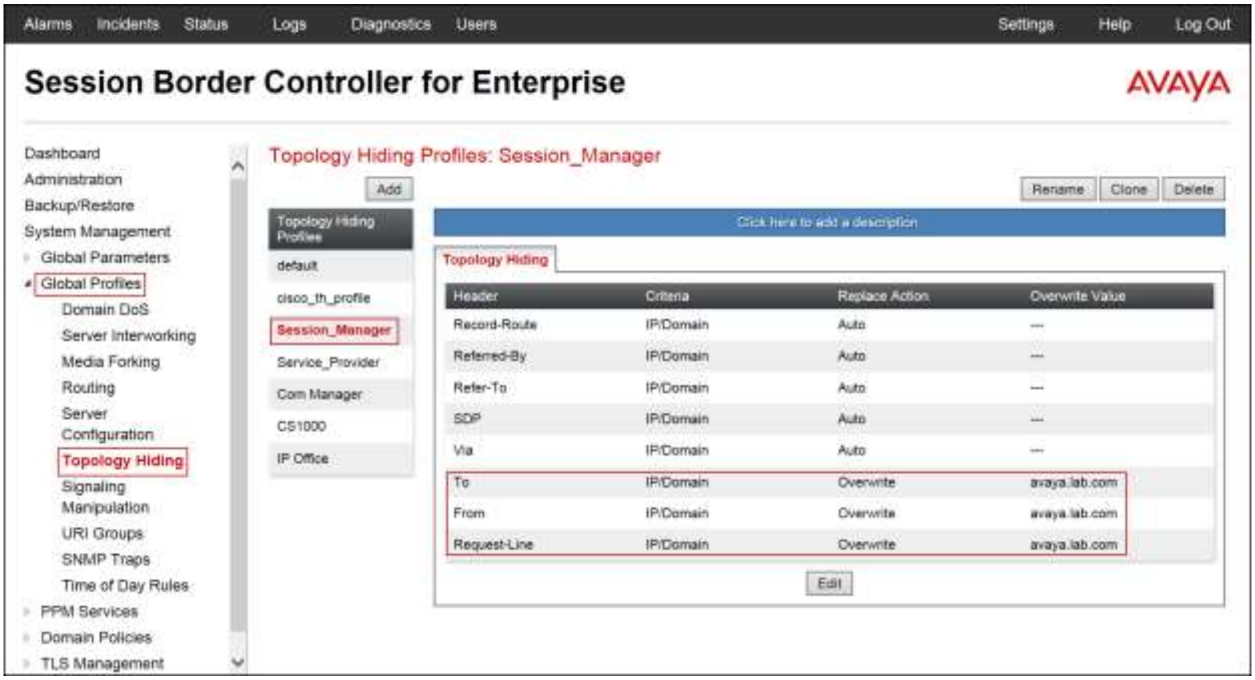
For the compliance test, only the minimum configuration required to achieve interoperability on the SIP trunk was performed. Additional steps can be taken in this section to further mask the information that is sent from the enterprise to the public network.

To add the Topology Hiding profile in the enterprise direction, select **Topology Hiding** from the **Global Profiles** menu on the left-hand side:

- Select the **default** profile in the **Topology Hiding Profiles** list, then click **Clone** on top right of the screen.
- Enter the **Profile Name**: ***Session_Manager***.
- Click **Finish**.
- Click **Edit** on the newly added **Session_Manager** Topology Hiding profile.
- For **To** under **Header**, choose ***Overwrite*** from the pull-down menu under **Replace Action**, enter the domain name for the Enterprise **(*avaya.lab.com*)** under **Overwrite Value**.
- For **From** under **Header**, choose ***Overwrite*** from the pull-down menu under **Replace Action**, enter the domain name for the enterprise **(*avaya.lab.com*)** under **Overwrite Value**.
- For **Request-Line** under **Header**, choose ***Overwrite*** from the pull-down menu under **Replace Action**; enter the domain name for the Enterprise **(*avaya.lab.com*)** under **Overwrite Value**.



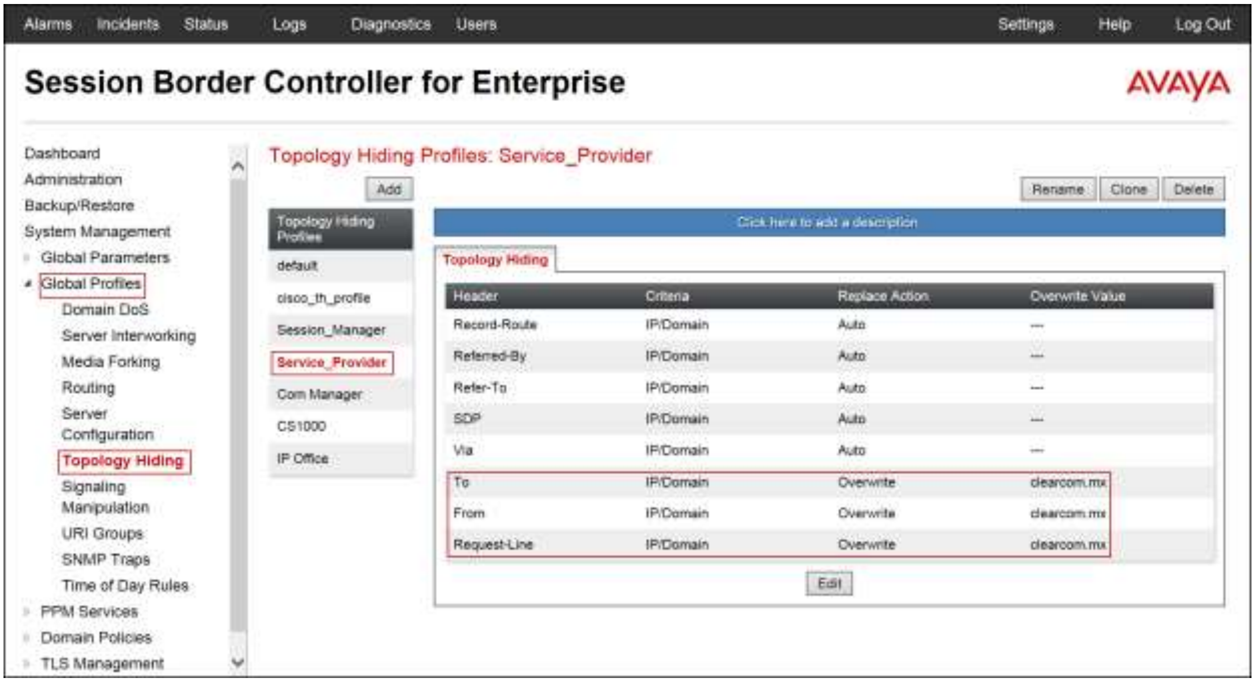
The following screen capture shows the newly created **Session_Manager** Topology Hiding Profile.



To add the Topology Hiding profile in the service provider direction, select **Topology Hiding** from the **Global Profiles** menu on the left-hand side:

- Select the **default** profile in the **Topology Hiding Profiles** list, then click **Clone** on top right of the screen.
- Enter the **Profile Name**: ***Service_Provider***.
- Click **Finish**.
- Click **Edit** on the newly added **Service_Provider** Topology Hiding profile.
- For **To** under **Header**, choose ***Overwrite*** from the pull-down menu under **Replace Action,** enter the domain name for the service provider **(*Clearcom.mx*)** under **Overwrite Value**.
- For **From** under **Header**, choose ***Overwrite*** from the pull-down menu under **Replace Action,** enter the domain name for the service provider **(*Clearcom.mx*)** under **Overwrite Value**.
- For **Request-Line** under **Header**, choose ***Overwrite*** from the pull-down menu under **Replace Action,** enter the domain name for the service provider **(*Clearcom.mx*)** under **Overwrite Value**.

The following screen capture shows the newly created **Service_Provider** Topology Hiding Profile.



## 7.4. Domain Policies

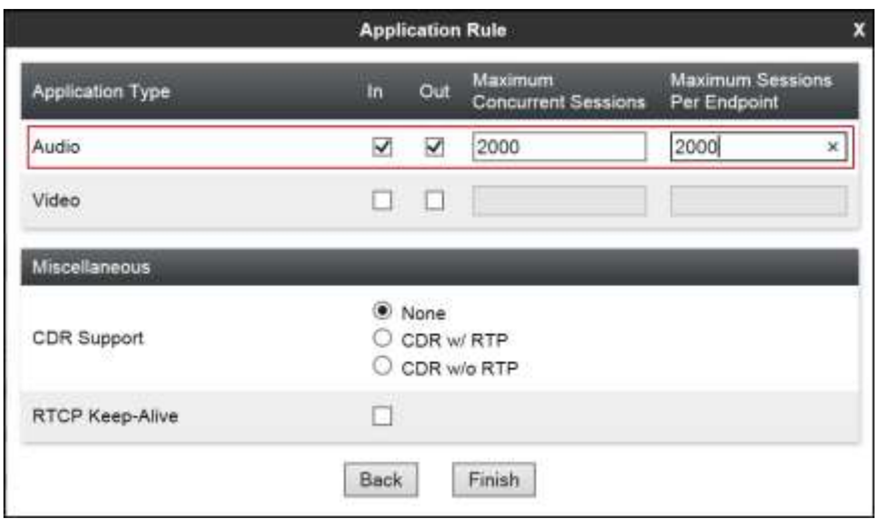
Domain Policies allow the configuration of sets of rules designed to control and normalize the behavior of call flows, based upon various criteria of communication sessions originating from or terminating in the enterprise. Domain Policies include rules for Application, Media, Signaling, Security, etc.

**Note**: The **default-trunk** Application Rule could have been used instead of creating a new one, but a new Application Rule was created to allow changes in the future.

### 7.4.1. Application Rules

Application Rules define which types of SIP-based Unified Communications (UC) applications the UC-Sec security device will protect: voice, video, and/or Instant Messaging (IM). In addition, Application Rules define the maximum number of concurrent voice and video sessions the network will process in order to prevent resource exhaustion. From the menu on the left-hand side, select **Domain Policies → Application Rules**.

- Click on the **Add** button to add a new rule.
- **Rule Name:** enter the name of the profile, e.g., ***2000 Sessions***.
- Under **Audio** check ***In*** and ***Out*** and set the **Maximum Concurrent Sessions** and **Maximum Sessions Per Endpoint** to recommended values; the value of ***2000*** was used in the sample configuration.
- Click **Finish**.



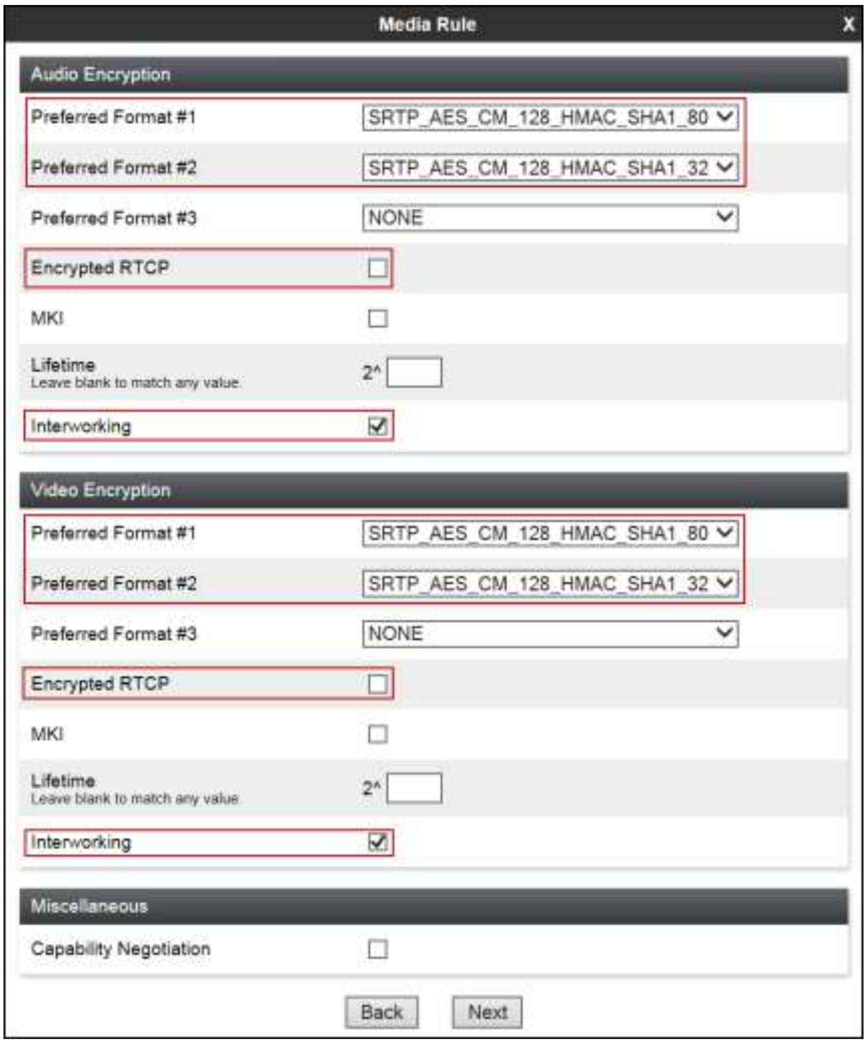
The following screen capture shows the newly created **2000 Sessions** Application Rule.


## 7.4.2. Media Rules

Media Rules allow one 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 to determine how media packets matching these criteria will be handled by the Avaya SBCE security product. For the compliance test, two media rules (shown below) were used; one toward Session Manager and one toward the Service Provider.

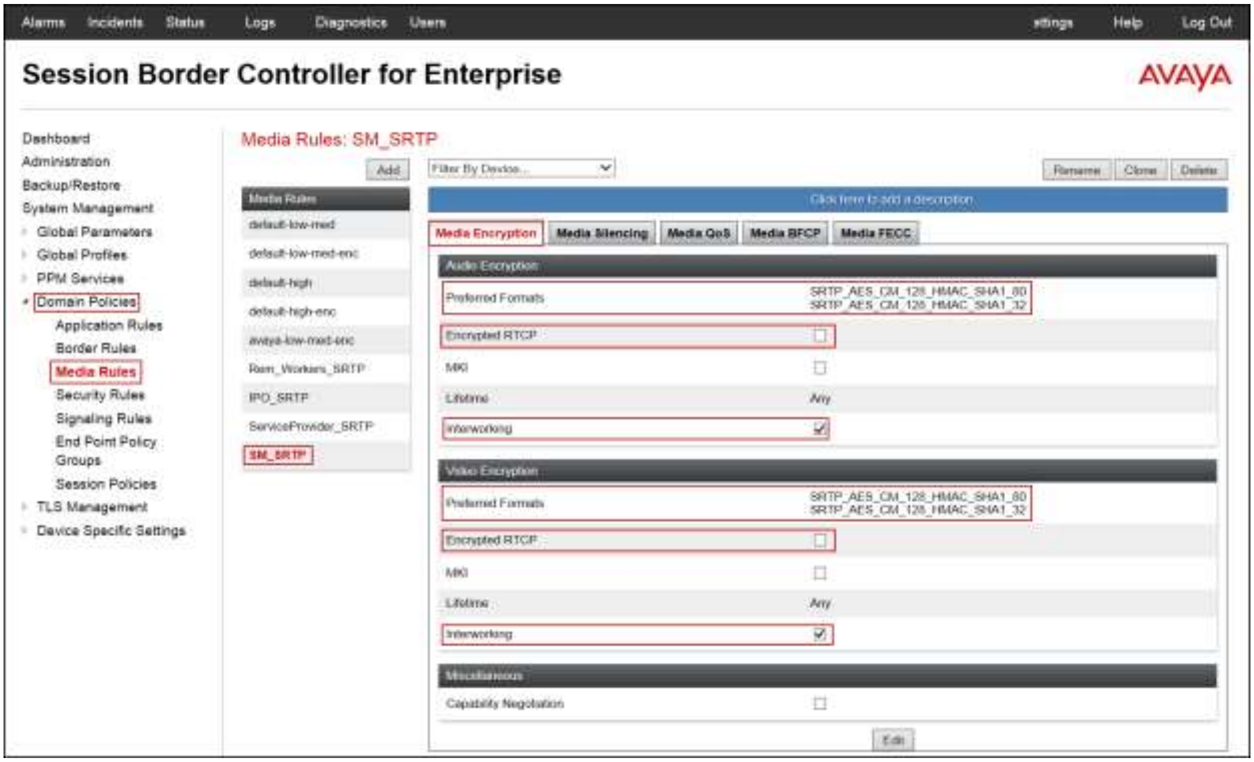
To add a media rule in the Session Manager direction, from the menu on the left-hand side, select **Domain Policies → Media Rules**.

- Click on the **Add** button to add a new media rule (not shown).
- Under **Rule Name** enter ***SM_SRTP***.
- Click Next (not shown).
- Under Audio Encryption, **Preferred Format #1**, select ***SRTP_AES_CM_128_HMAC_SHA1_80***.
- Under Audio Encryption, **Preferred Format #2**, select ***SRTP_AES_CM_128_HMAC_SHA1_32***.
- Under Audio Encryption, uncheck ***Encrypted RTCP***.
- Under Audio Encryption, check ***Interworking***.
- Repeat the above steps under Video Encryption.
- Under Miscellaneous verify that ***Capability Negotiation*** is unchecked.
- Click **Next**.



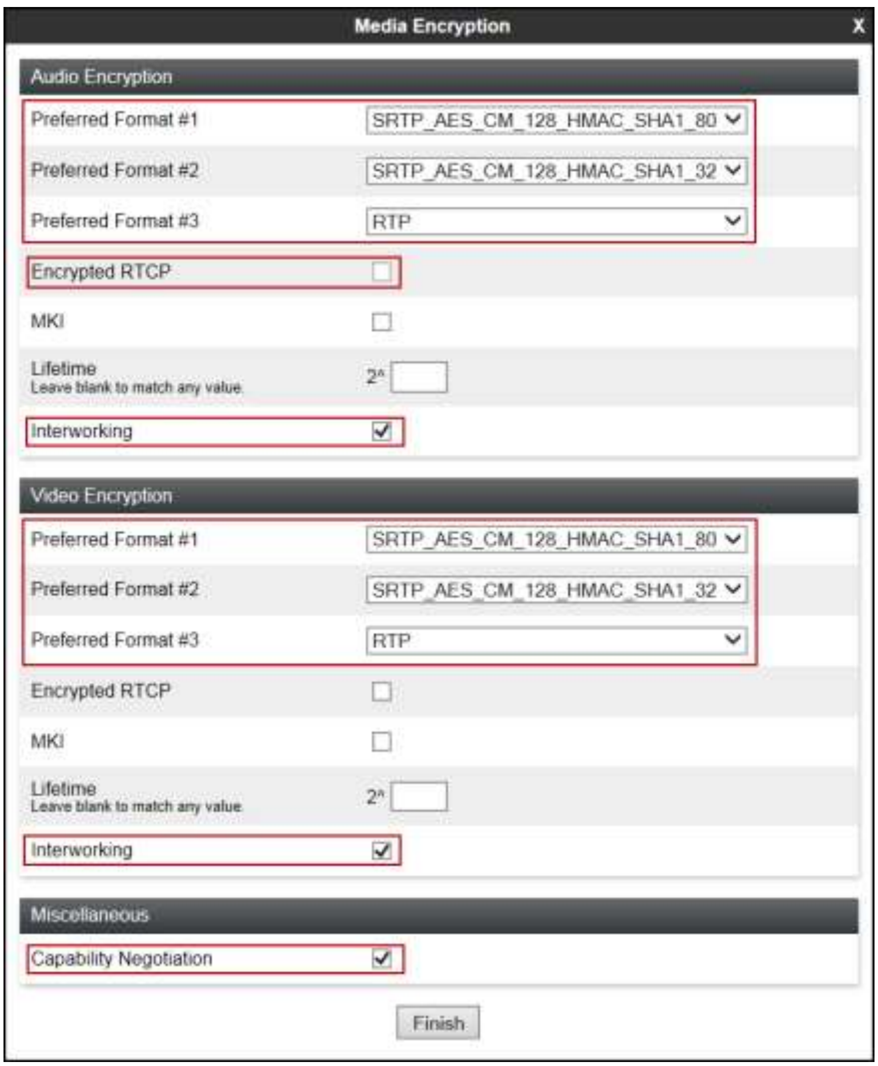
Accept default values in the remaining sections by clicking **Next** (not shown), and then click **Finish** (not shown).

The following screen capture shows the newly created **SM_SRTP** Media Rule



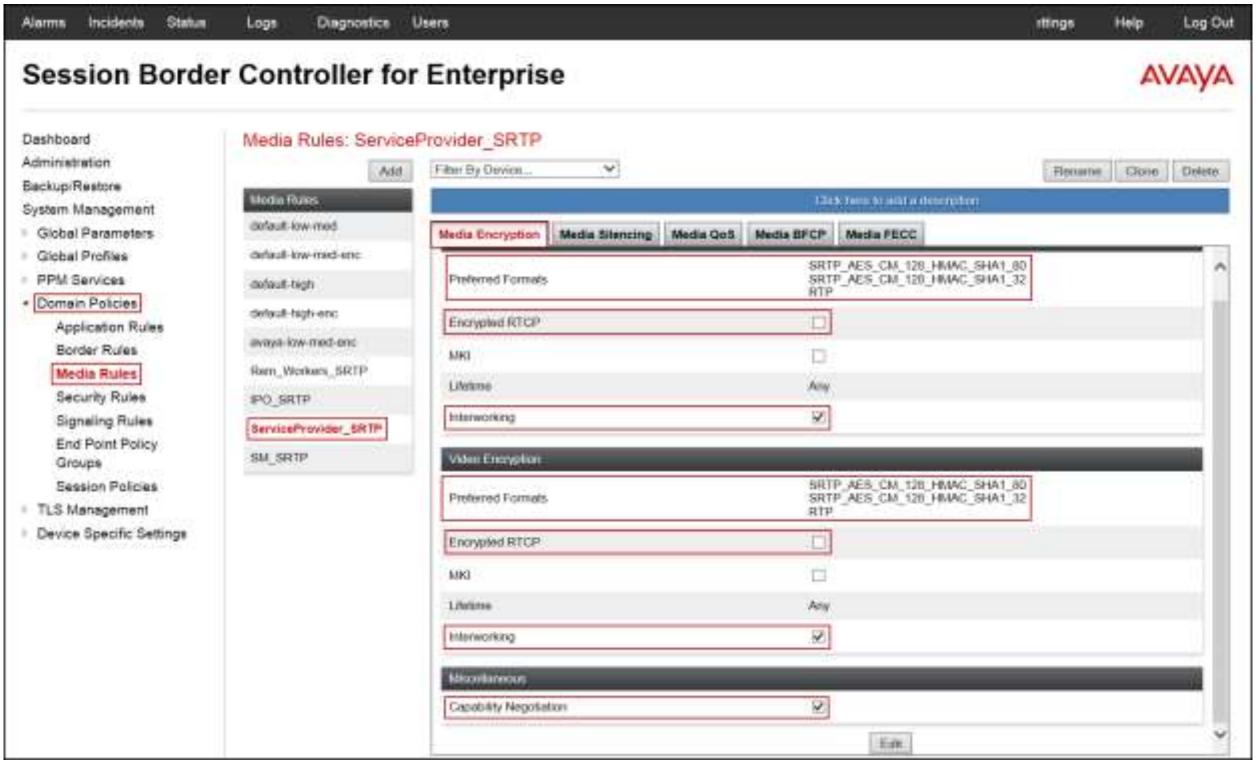
To add a media rule in the Service Provider direction, from the menu on the left-hand side, select **Domain Policies → Media Rules**.

- Click on the **Add** button to add a new media rule (not shown).
- Under **Rule Name** enter ***ServiceProvider_SRTP***.
- Click Next.
- Under Audio Encryption, **Preferred Format #1**, select ***SRTP_AES_CM_128_HMAC_SHA1_80***.
- Under Audio Encryption, **Preferred Format #2**, select ***SRTP_AES_CM_128_HMAC_SHA1_32***.
- Under Audio Encryption, **Preferred Format #3**, select ***RTP***.
- Under Audio Encryption, uncheck ***Encrypted RTCP***.
- Under Audio Encryption, check ***Interworking***.
- Repeat the above steps under Video Encryption.
- Under Miscellaneous check ***Capability Negotiation***.
- Click **Finish**.



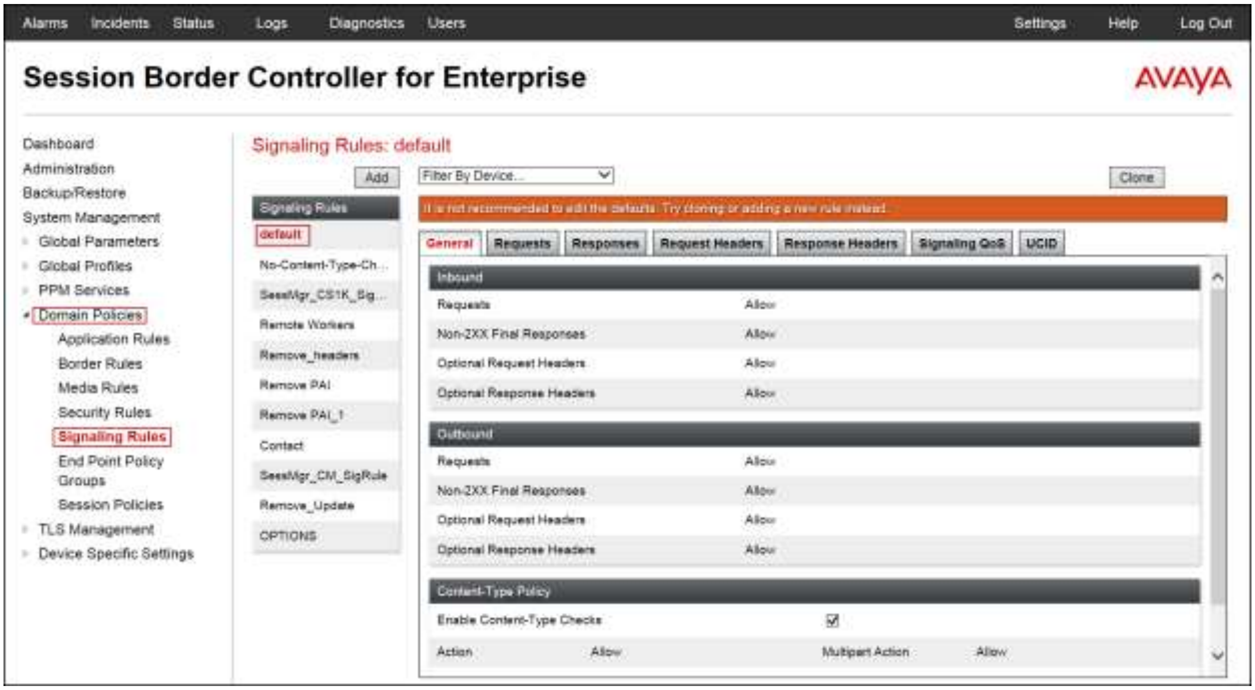
Accept default values in the remaining sections by clicking **Next** (not shown), and then click **Finish** (not shown).

The following screen capture shows the newly created **ServiceProvider_SRTP** Media Rule.



### 7.4.3. Signaling Rules

For the compliance test, the **default** Signaling Rule was used.

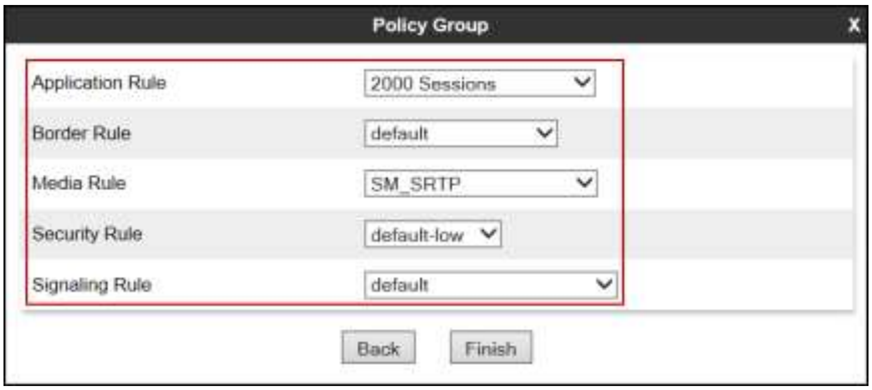


### 7.4.4. End Point Policy Groups

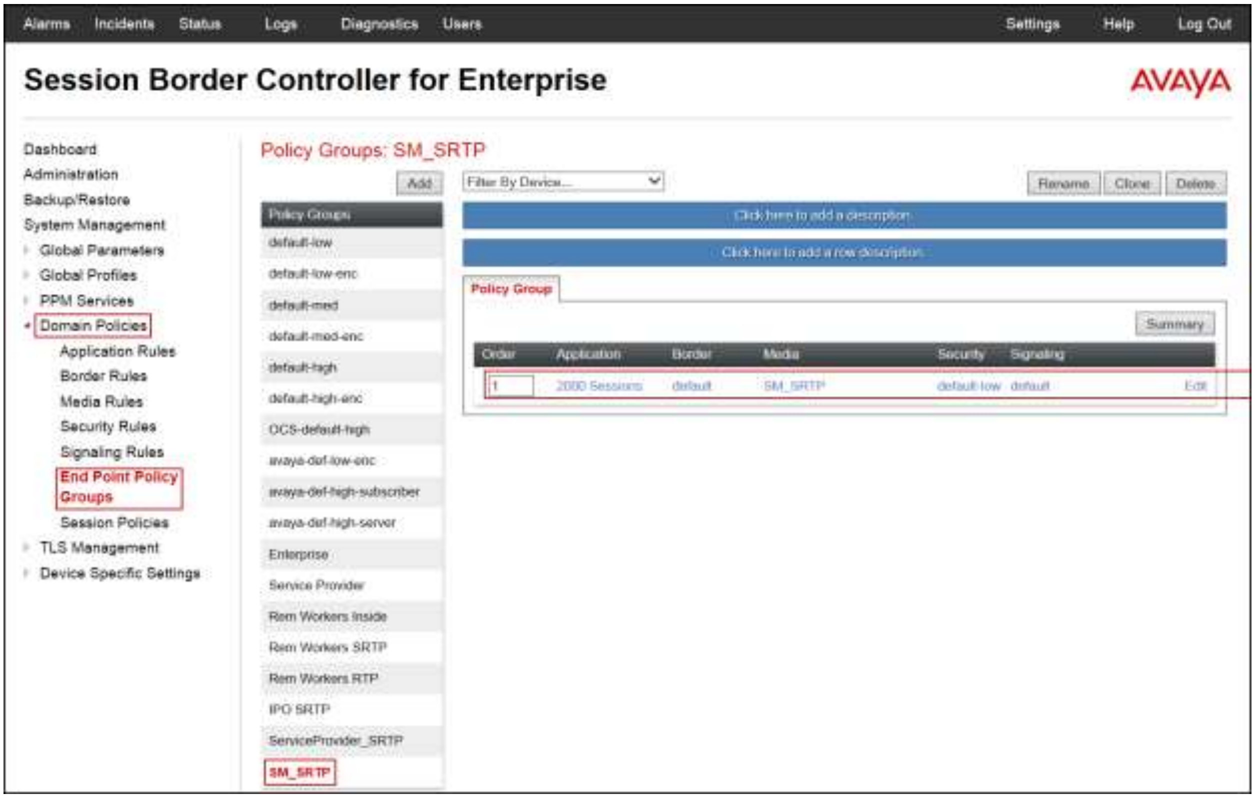
End Point Policy Groups are associations of different sets of rules (Media, Signaling, Security, etc.) to be applied to specific SIP messages traversing through the Avaya SBCE.

To create an End Point Policy Group for the Enterprise, from the **Domain Policies** menu, select **End Point Policy Groups**. Select **Add Group**, under **Group Name** enter ***SM_SRTP***.

- **Application Rule: *2000 Sessions***.
- **Border Rule: *default***.
- **Media Rule: *SM_SRTP***.
- **Security Rule: *default-low***.
- **Signaling Rule: *default***
- Click **Finish**.

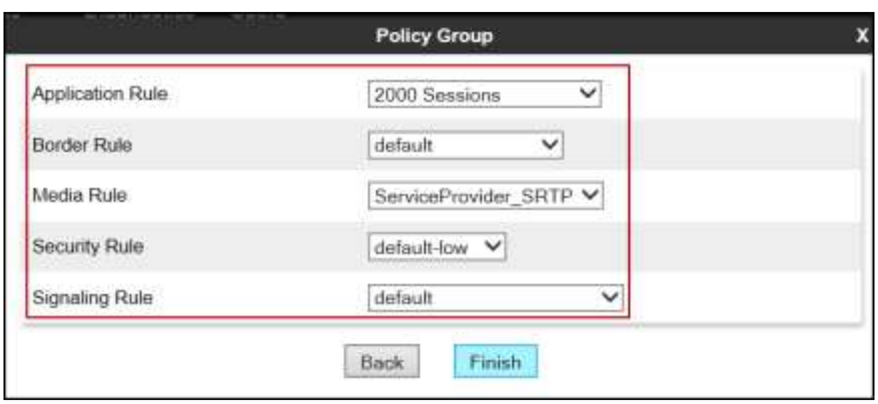


The following screen capture shows the newly created **SM_SRTP** End Point Policy Group.

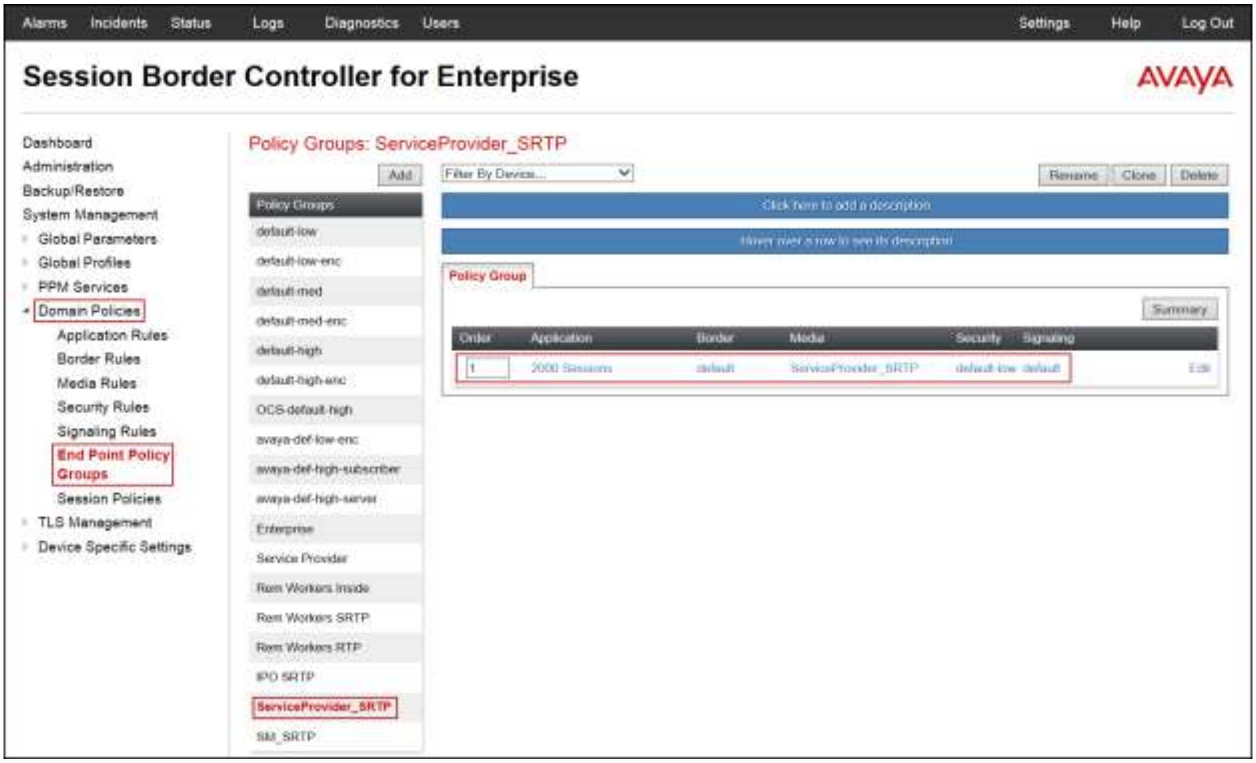



Similarly, to create an End Point Policy Group for the Service Provider SIP Trunk, select **Add Group**, under **Group Name** enter ***ServiceProvider_SRTP***.

- **Application Rule:** ***2000 Sessions***.
- **Border Rule:** ***default***.
- **Media Rule:** ***ServiceProvider_SRTP***.
- **Security Rule:** ***default-low***.
- **Signaling Rule:** ***default***.
- Click **Finish**.



The following screen capture shows the newly created **ServiceProvider_SRTP** End Point Policy Group.



## 7.5. Device Specific Settings

The **Device Specific Settings** allow the management of various device-specific parameters, which determine how a particular device will function when deployed in the network. Specific server parameters, like network and interface settings, as well as call flows, etc. are defined here.

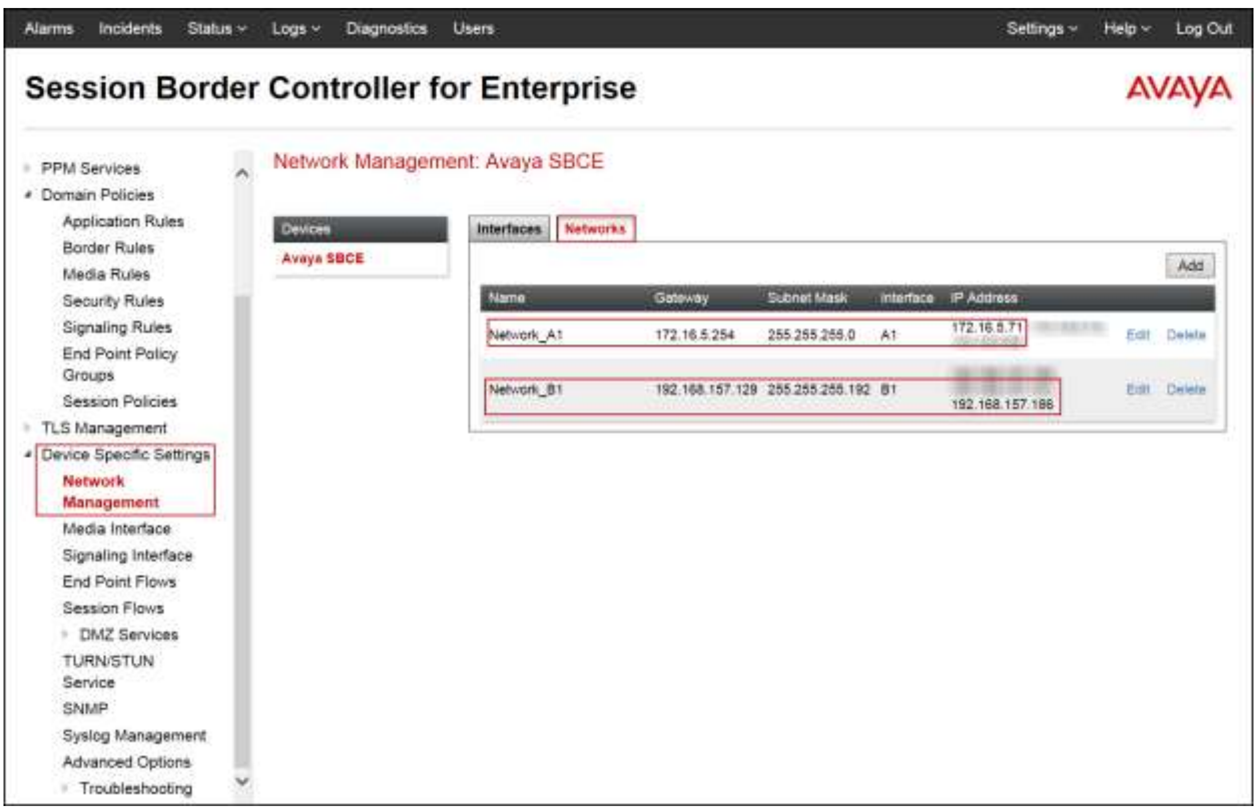
### 7.5.1. Network Management

The network information should have been previously completed. To verify the network configuration, from the **Device Specific Settings** on the left hand side, select **Network Management**. Select the **Network Configuration** tab.

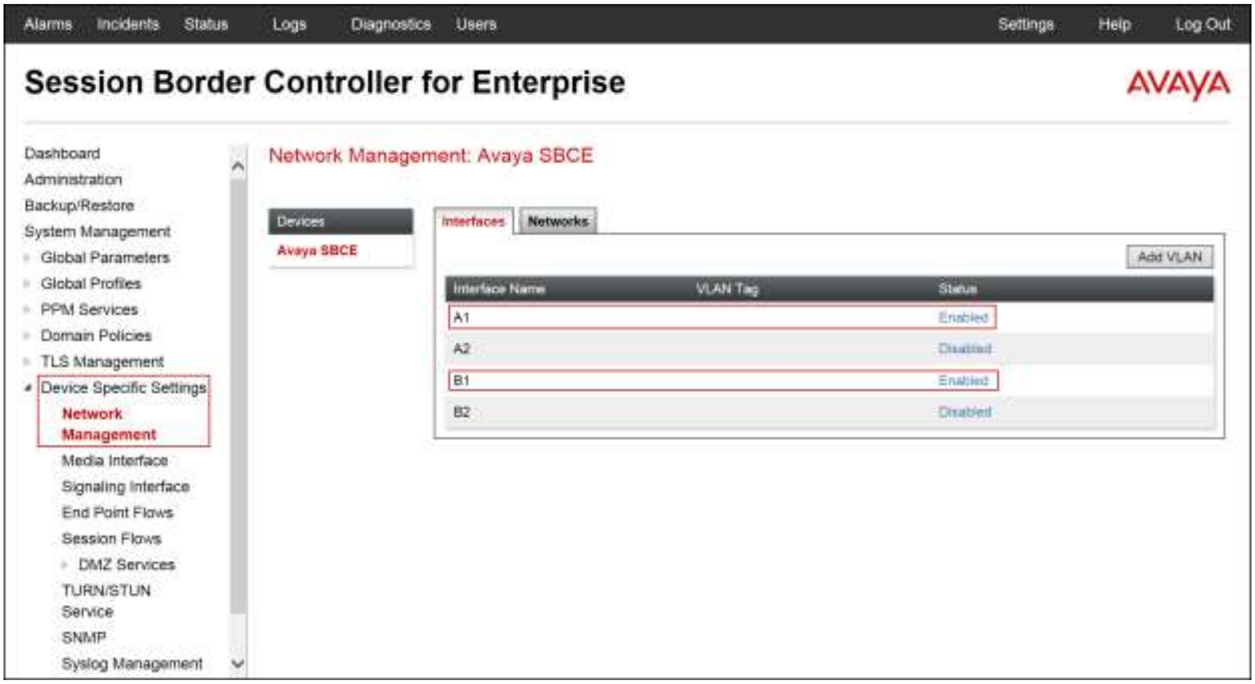
In the event that changes need to be made to the network configuration information, they can be entered here.

Use **Figure 1** as reference for IP address assignments.

| |
|---|
| **Note**: Only the highlighted entity items were created for the compliance test, and are the ones relevant to these Application Notes. Blurred out items are part of the Remote Worker configuration, which is not discussed in these Application Notes. |



On the Interface Configuration tab, click the **Status** for interfaces **A1** and **B1** to change the status to ***Enabled***. It should be noted that the default state for all interfaces is **disabled**, so it is important to perform this step or the Avaya SBCE will not be able to communicate on any of its interfaces.

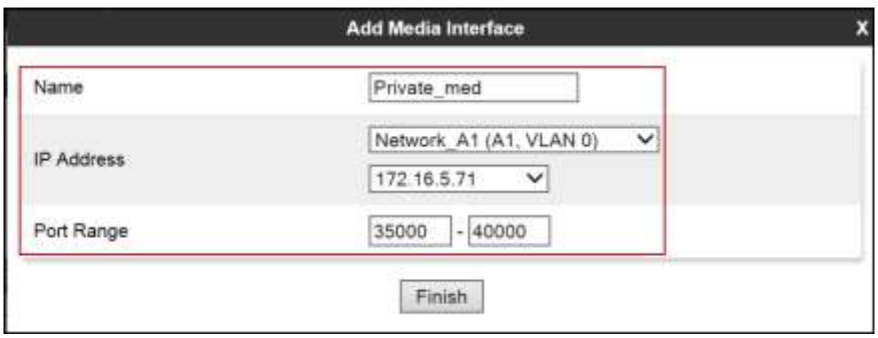


### 7.5.2. Media Interface

Media Interfaces were created to adjust the port range assigned to media streams leaving the interfaces of the Avaya SBCE. On the Private and Public interfaces of the Avaya SBCE, port range 35000 to 40000 was used.

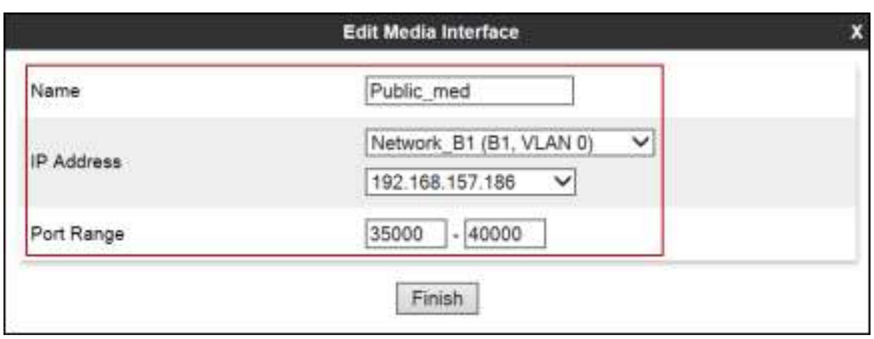
From the **Device Specific Settings** menu on the left-hand side, select **Media Interface**. Below is the configuration of the inside, private Media Interface of the Avaya SBCE.

- Select **Add** in the **Media Interface** area.
- **Name**: ***Private_med***.
- Under **IP Address** select: ***Network_A1 (A1, VLAN 0)***
  Select **IP Address**: ***172.16.5.71*** (Private or A1 IP Address of the Avaya SBCE, toward Session Manager).
- Enter **Port Range**: ***35000-40000***.
- Click **Finish**.

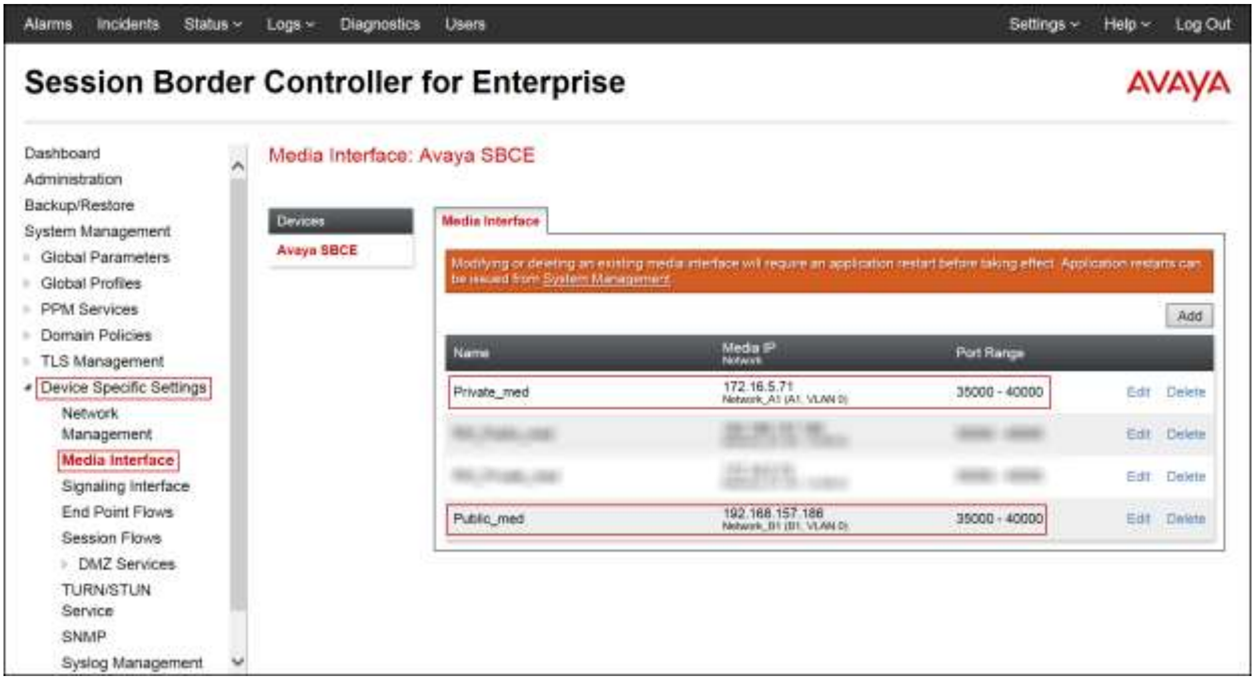



Below is the configuration of the outside, public Media Interface of the Avaya SBCE.

- Select **Add** in the **Media Interface** area.
- **Name**: ***Public_med***.
- Under **IP Address** select: ***Network_B1 (B1, VLAN 0)***
  Select **IP Address**: ***10.10.157.186*** (Public or B1 IP Address of the Avaya SBCE toward the Service Provider).
- **Port Range**: ***35000-40000***.
- Click **Finish**.



The following screen capture shows the newly created Media Interfaces.

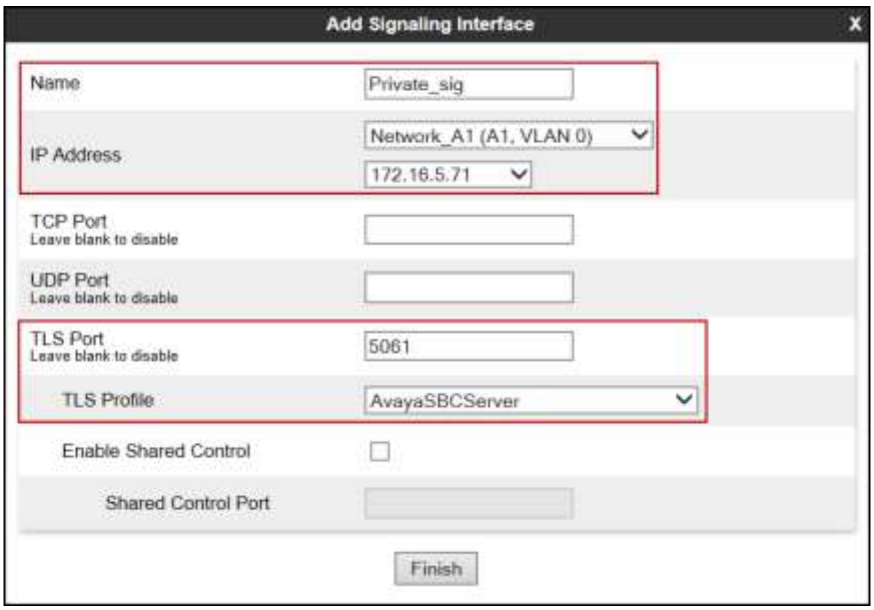


### 7.5.3. Signaling Interface

To create the Signaling Interface toward Session Manager, from the **Device Specific** menu on the left hand side, select **Signaling Interface**.

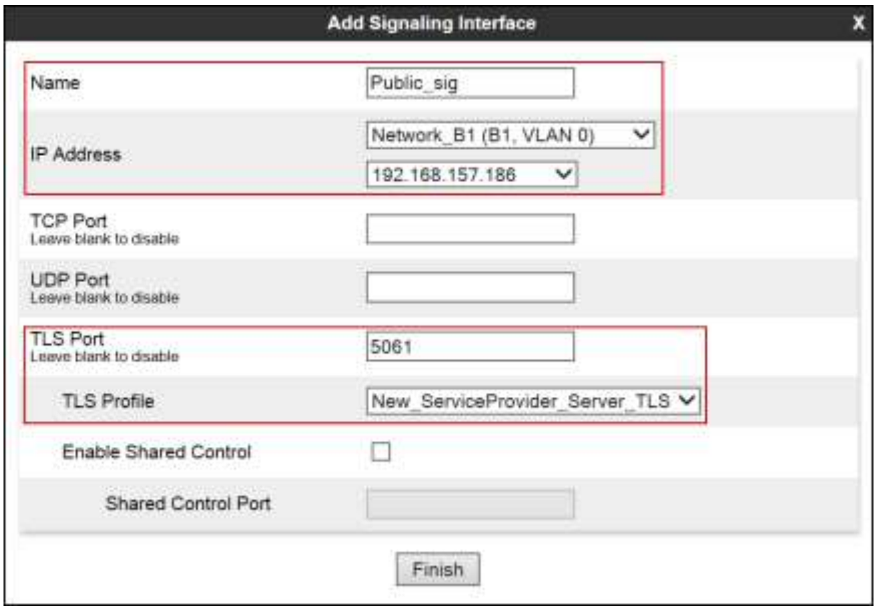
Below is the configuration of the inside private Signaling Interface of the Avaya SBCE.

- Select **Add** in the **Signaling Interface** area.
- **Name**: ***Private_sig***.
- Under **IP Address** select: ***Network_A1 (A1, VLAN 0)***
  Select **IP Address**: ***172.16.5.71*** (Inside or A1 IP Address of the Avaya SBCE, toward Session Manager).
- **TLS Port**: ***5061***
- Under **TLS Profile** select: **AvayaSBCServer**
- Click **Finish**.

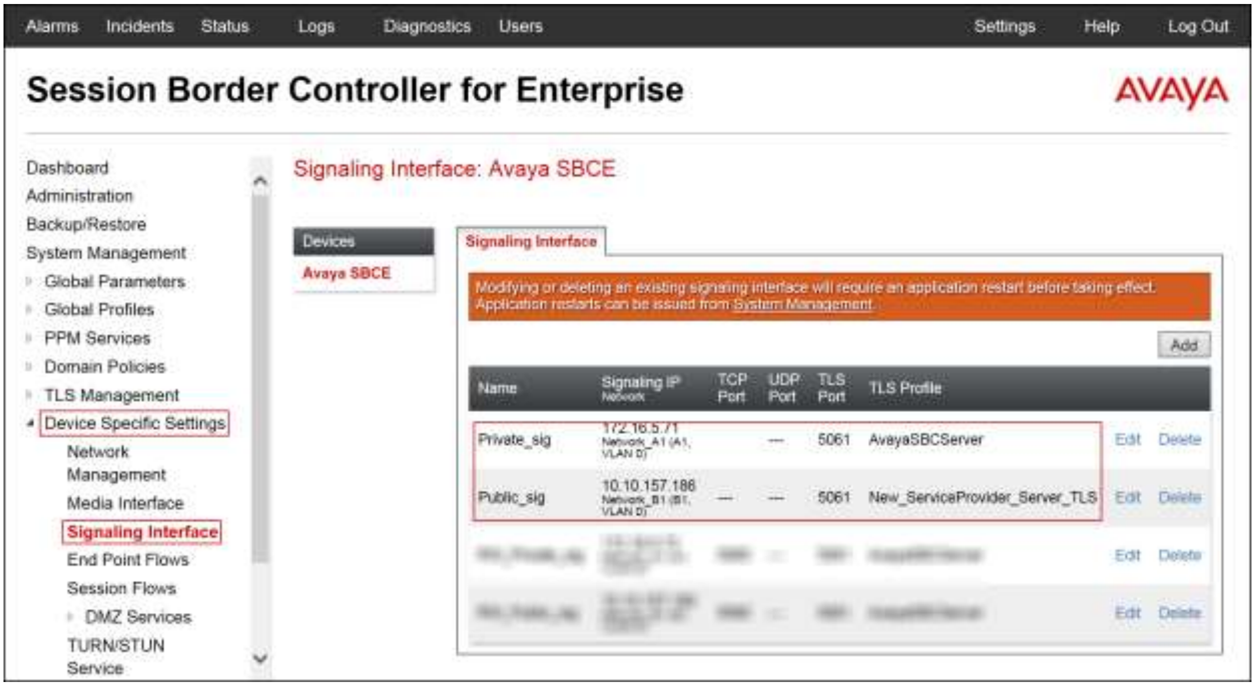


Below is the configuration of the outside, public Signaling Interface of the Avaya SBCE.

- Select **Add** in the **Signaling Interface** area.
- **Name**: ***Public_sig***.
- Under **IP Address** select: ***Network_B1 (B1, VLAN 0)***
- Select **IP Address**: ***10.10.157.186*** (Public or B1 IP Address of the Avaya SBCE toward the Service Provider).
- **UDP Port**: ***5061***.
- Under **TLS Profile** select: ***New_ServiceProvider_Server_TLS***.
- Click **Finish**.

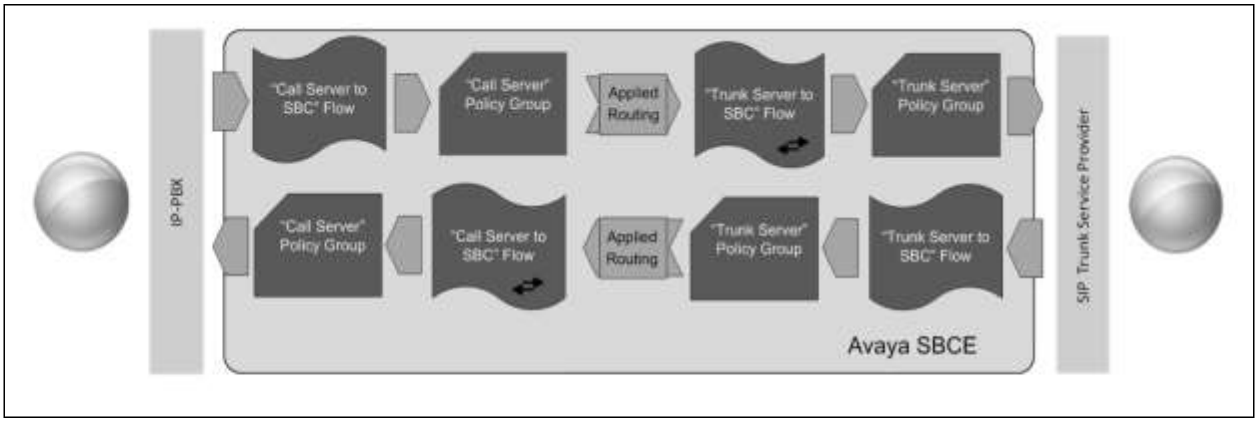


The following screen capture shows the newly created Signaling Interfaces.



### 7.5.4. End Point Flows

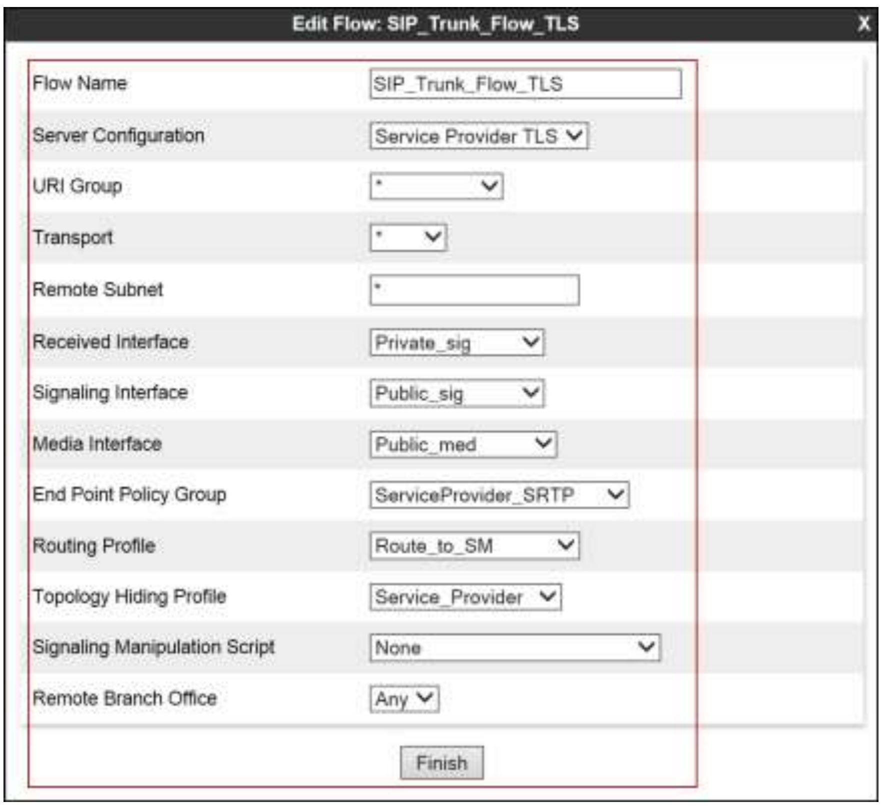
When a packet is received by Avaya SBCE, the content of the packet (IP addresses, URIs, etc.) is used to determine which flow it matches. Once the flow is determined, the flow points to a policy group which contains several rules concerning processing, privileges, authentication, routing, etc. Once routing 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 screen illustrates the flow through the Avaya SBCE to secure a SIP Trunk call.




The **End-Point Flows** defines certain parameters that pertain to the signaling and media portions of a call, whether it originates from within the enterprise or outside of the enterprise.

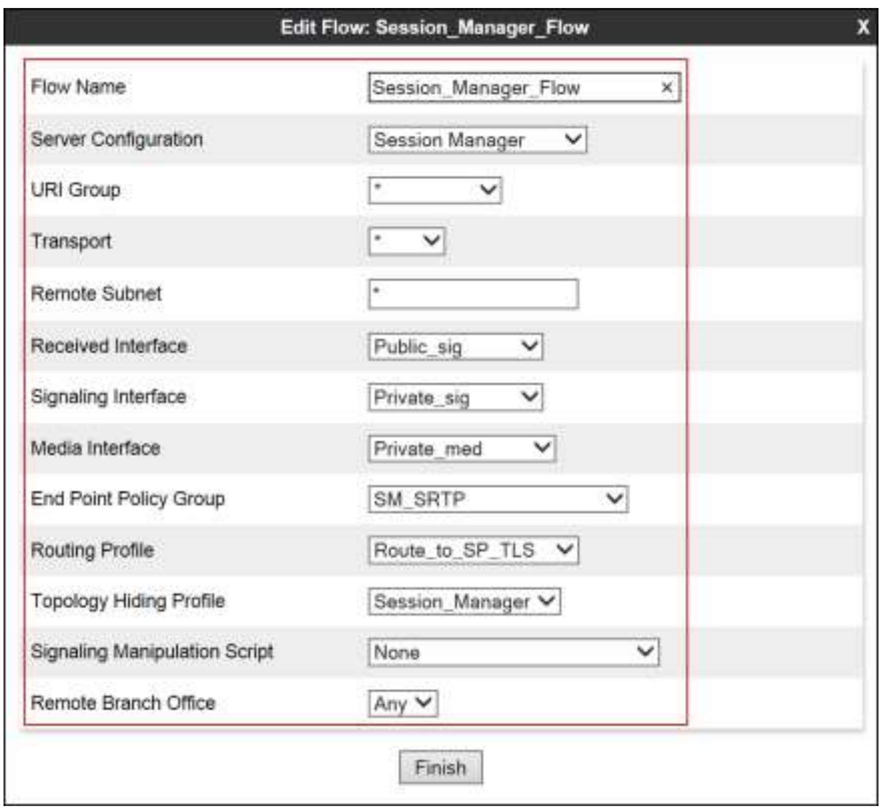
To create the call flow toward the Service Provider SIP trunk, from the **Device Specific Settings** menu, select **End Point Flows**, and then the **Server Flows** tab. Click **Add** (not shown).

- **Flow Name:** ***SIP_Trunk_Flow_TLS***.
- **Server Configuration**: ***Service Provider TLS***.
- **URI Group: ***
- **Transport: ***
- **Remote Subnet: ***
- **Received Interface**: ***Private_sig***.
- **Signaling Interface:** ***Public_sig***.
- **Media Interface**: ***Public_med***.
- **End Point Policy Group:** ***ServiceProvider_SRTP***.
- **Routing Profile*:*** ***Route_to_SM*** (Note that this is the reverse route of the flow).
- **Topology Hiding Profile:** ***Service_Provider***.
- **Signaling Manipulation Script:** ***None***.
- **Remote Brach Office**: ***Any***.
- Click **Finish**.

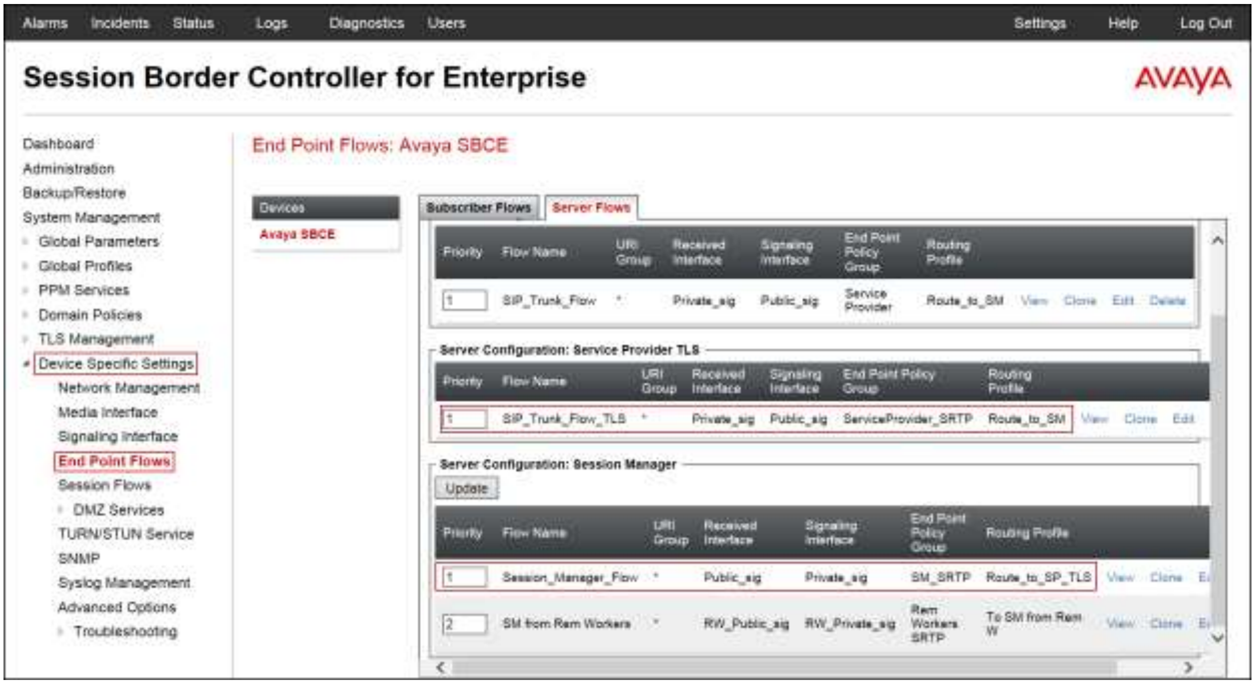


To create the call flow toward the Session Manager, click **Add**.

- **Flow Name:** ***Session_Manager_Flow***.
- **Server Configuration**: ***Session Manager***.
- **URI Group: ***
- **Transport: ***
- **Remote Subnet: ***
- **Received Interface**: ***Public_sig***.
- **Signaling Interface:** ***Private_sig***.
- **Media Interface**: ***Private_med***.
- **End Point Policy Group:** ***SM_SRTP***
- **Routing Profile:** ***Route_to_SP_TLS*** (Note that this is the reverse route of the flow).
- **Topology Hiding Profile:** ***Session_Manager***.
- **Signaling Manipulation Script:** ***None***.
- **Remote Brach Office**: ***Any***.
- Click **Finish**.



The following screen capture shows the newly created **End Point Flows**.



# 8. Clearcom SIP Trunking Service Configuration

To use Clearcom's SIP Trunking Service, a customer must request the service from Clearcom using the established sales processes. The process can be started by contacting Clearcom via the corporate web site at: http://www.clearcom.mx/ and requesting information.

During the signup process, Clearcom and the customer will discuss details about the preferred method to be used to connect the customer's enterprise network to Clearcom's network.

Clearcom is responsible for the configuration of Clearcom SIP Services. The customer will need to provide a public IP address to be used to reach the Avaya SBCE at the enterprise. In the case of the compliance test, this is the outside or public IP address of the Avaya SBCE (B1 interface). The customer will also need the IP addresses for the primary and the secondary public DNS servers, these addresses can be obtained from the local ISP in Mexico.

Clearcom will provide the following information:

- SIP Trunk registration credentials (user name, password, SIP domain).
- Fully Qualified Domain Name of the Clearcom SIP proxy server.
- DID numbers.
- Supported codecs and order of preference.
- Any IP addresses and port numbers used for signaling or media that will need access to the enterprise network through any security devices.

# 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 that can be used to troubleshoot the solution.

Verification Steps:

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

## 9.1. Troubleshooting

### 9.1.1. Communication Manager

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

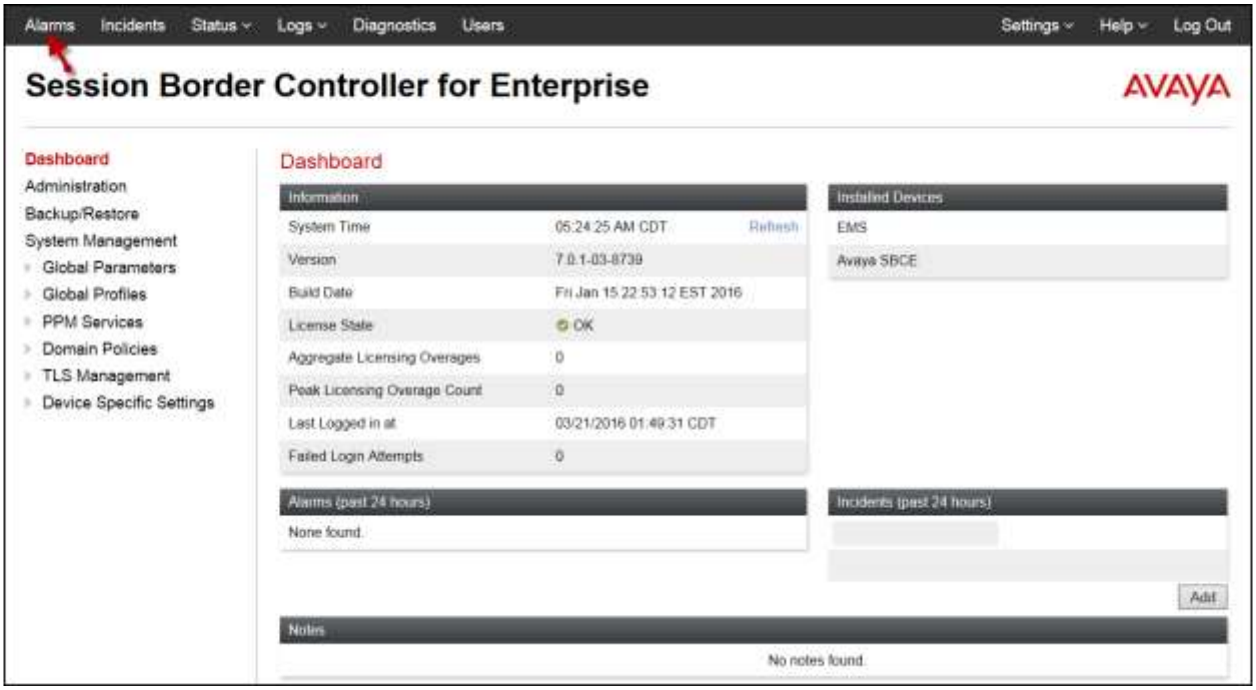
### 9.1.2. Session Manager

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

### 9.1.3. Avaya Session Border Controller for Enterprise (Avaya SBCE)

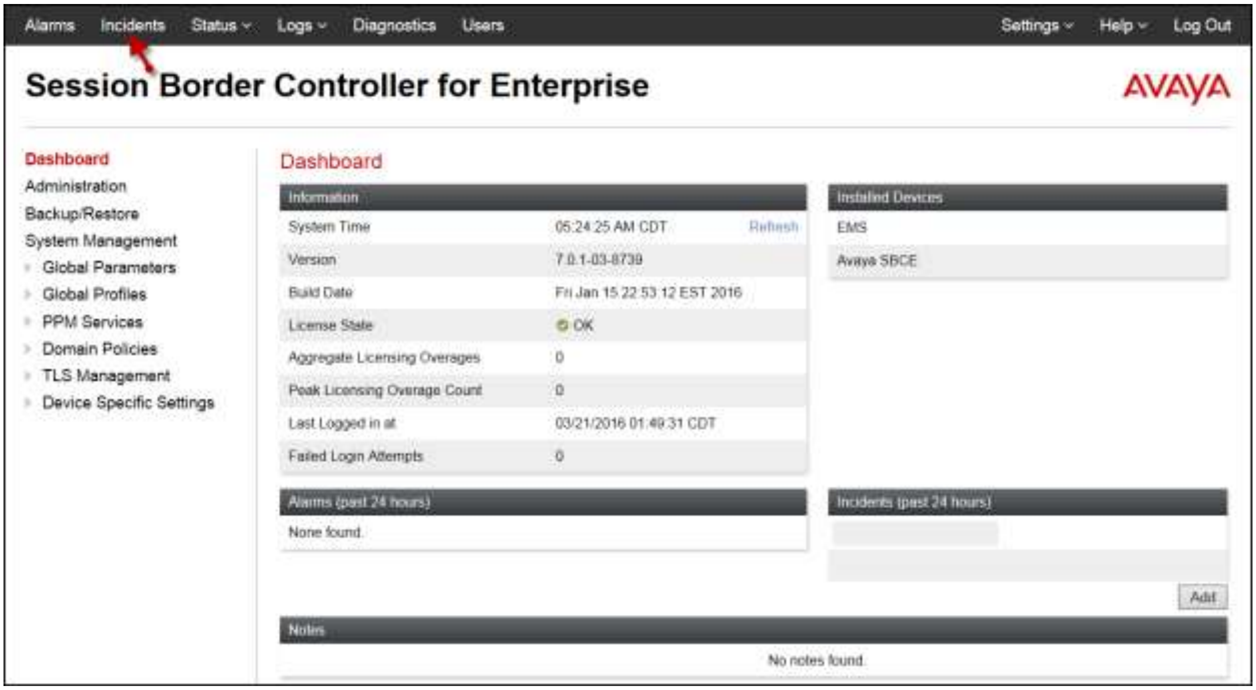
There are several links and menus located on the taskbar at the top of the screen of the web interface that can be used for diagnostic and troubleshooting.

**Alarms**: Provides information about the health of the Avaya SBCE.

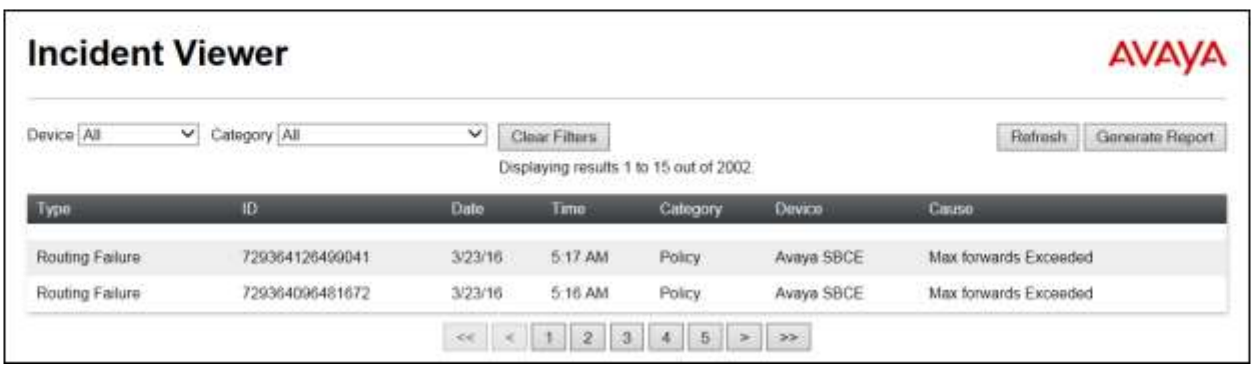



The following screen shows the **Alarm Viewer** page.

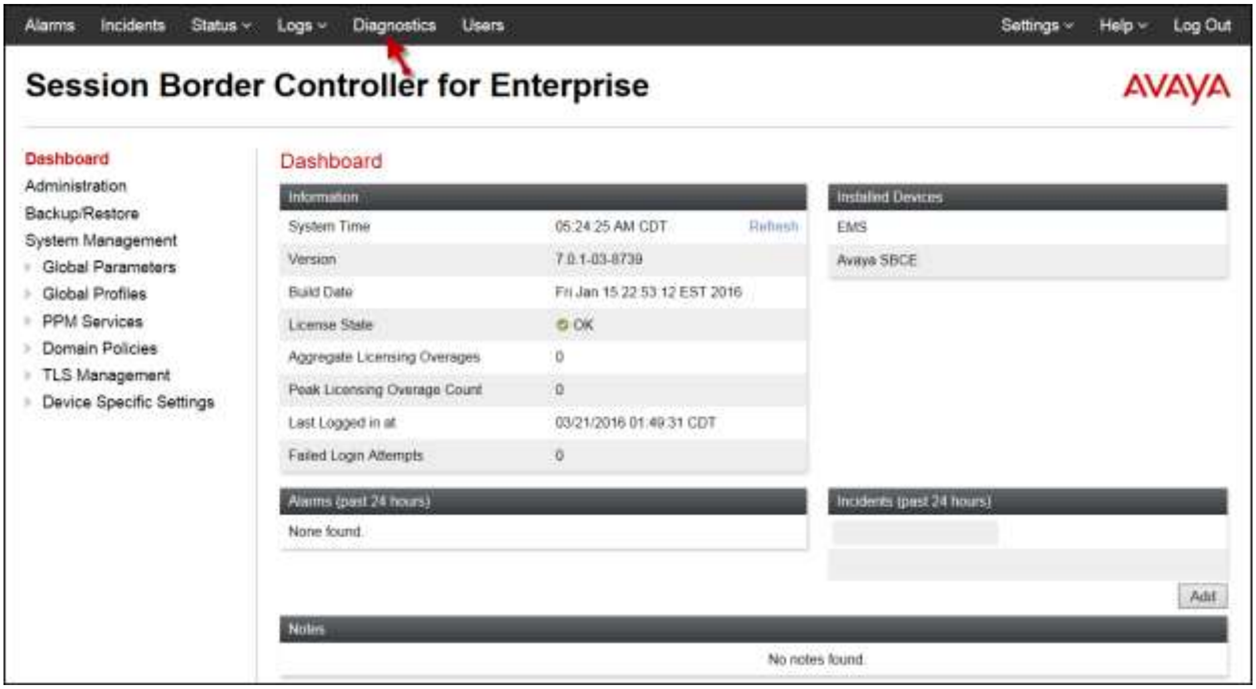
**Incidents**: Provides detailed reports of anomalies, errors, policies violations, etc.



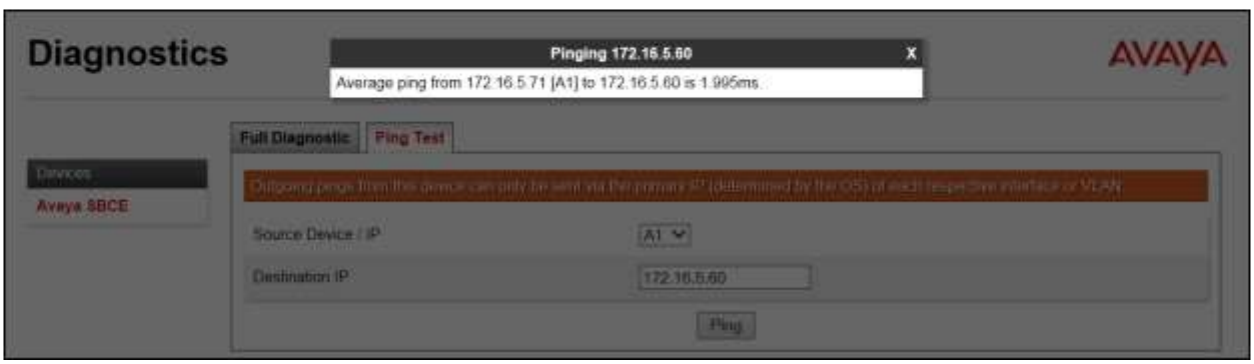

The following screen shows the Incident Viewer page.



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

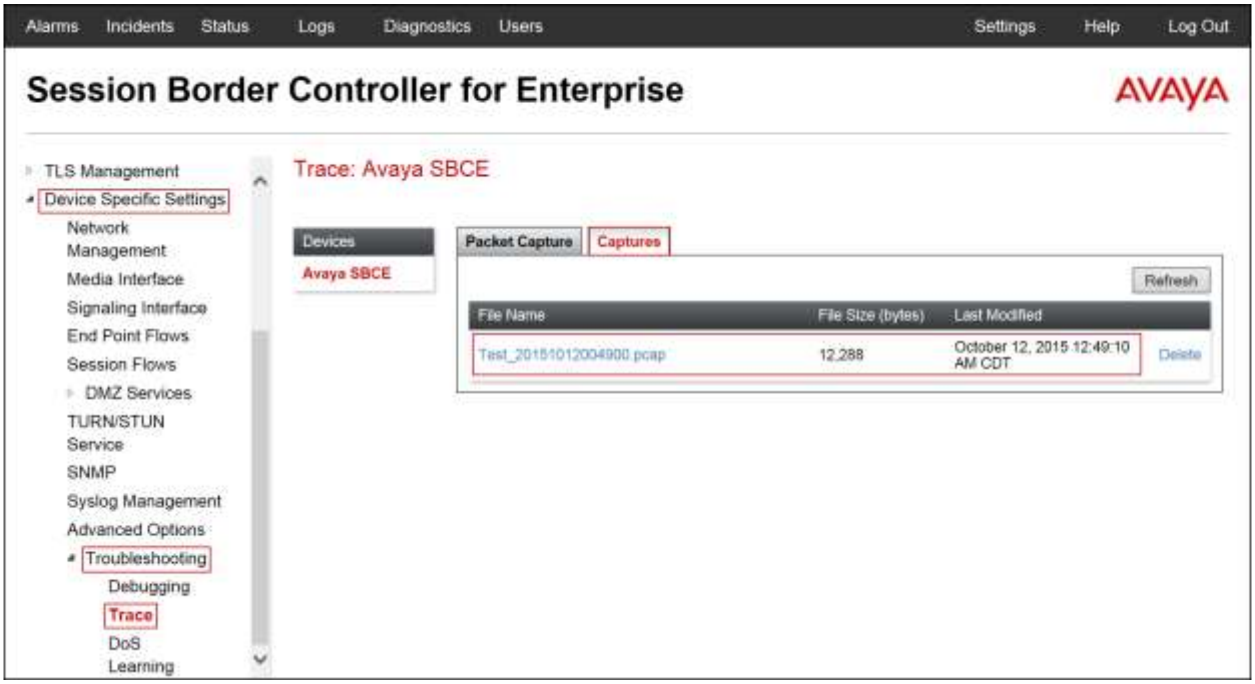



The following screen shows the Diagnostics page with the results of a ping test.



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

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




## 9.2. TraceSBC Tool

traceSBC is a perl script that parses Avaya SBCE log files and displays SIP and PPM messages in a ladder diagram. Because the logs contain the decrypted messages, the tool can easily be used in case of TLS and HTTPS. traceSBC can parse the log files downloaded from Avaya SBCE. traceSBC can also process log files in real time on Avaya SBCE, so that IP and PPM traffic can be checked during live calls. Refer to items [**8**] in **Section 11**

**Operation modes:**

- **Non real-time mode**:
  The tool starts with at least one file in the command line parameters. The tool automatically detects the type of files, processes the files, and finally displays messages from the different files in one diagram ordered by the timestamp. If filters are set, only the messages that match the filters are displayed in the diagram. In this mode, enabling live capture is not an option.
  Example: # traceSBC tracesbc_sip_1408635251
- **Real-time mode**
  In this mode, traceSBC must be on active Avaya SBCE. traceSBC is started without specifying a file in the command line parameters. The tool automatically starts processing the log files. The live capture can be started and stopped anytime without affecting service.
  Example: # traceSBC

**Log Files:**
Avaya SBCE can log SIP messages as processed by different subsystems and also log PPM messages. The traceSBC utility can process the log files real-time by opening the latest log files in the given directories. TraceSBC also checks regularly if a new file is generated, in which case the old one is closed and processing continues with the new one. A new log file is generated every time the relevant processes restart, or when the size reaches the limit of ~10 Megabytes.

**Log Locations:**
SIP messages are found at /archive/log/tracesbc/tracesbc_sip/ and PPM messages can be found at /archive/log/tracesbc/tracesbc_ppm/.

**Active files are of the following format**:
-rw-rw---- 1 root root 112445 Aug 21 10:12 tracesbc_sip_1408631651

**Inactive or closed files are of the following format**:
-rw-rw---- 1 root root 175236 Aug 21 06:33 tracesbc_sip_1408617250_1408620820_1

or

-rw-rw---- 1 root root 31706 Jul 10 13:34 tracesbc_sip_1436549674_1436553270_1.gz

# 10. Conclusion

These Application Notes describe the procedures necessary for configuring Session Initiation Protocol (SIP) Trunk service for an enterprise solution consisting of Avaya Aura® Communication Manager Release 7.0, Avaya Aura® Session Manager Release 7.0, and Avaya Session Border Controller for Enterprise Release 7.0 to support Clearcom SIP Trunking Service using TLS, as shown in **Figure 1**.

Interoperability testing was completed successfully with the observations/limitations outlined in the scope of testing in **Section 2.1** as well as under test results in **Section 2.2**.

# 11. References

This section references the documentation relevant to these Application Notes.

Product documentation for Avaya Aura® Communication Manager, including the following, is available at: http://support.avaya.com/


[1] *Administering Avaya Aura® Communication Manager*, Release 7.0, August 2015, Document Number 03-300509.

[2] *Avaya Aura® Communication Manager Feature Description and Implementation,* Release 7.0, August 2015, Document Number 555-245-205.

[3] *Avaya Aura® Communication Manager Security Design, Release 6.3,* Issue 6, June 2015, Document Number 03-601973.


Product documentation for Avaya Aura® System Manager, including the following, is available at: http://support.avaya.com/


[4] *Administering Avaya Aura® System Manager for Release 7.0*, Release 7.0, Issue 1, August 2015.

[5] *Avaya Aura® System Manager Release 7.0 Security Guide,* Release 7.0, Issue 1, August 2015


Product documentation for Avaya Aura® Session Manager, including the following, is available at: http://support.avaya.com/


[6] *Administering Avaya Aura® Session Manager*, Release 7.0, August 2015.


Product documentation for the Avaya Session Border Controller for Enterprise, including the following, is available at: http://support.avaya.com/


[7] *Deploying Avaya Session Border Controller for Enterprise*, Release 7.0, August 2015.

[8] *Administering Avaya Session Border Controller for Enterprise*, Release 7.0, August 2015.


Product documentation for Avaya Aura® Media Server, is available at: http://support.avaya.com/


[9] *Implementing and Administering Avaya Aura® Media Server*. Release 7.7. August 2015.

[10] *Quick Start Guide to Using the Avaya Aura® Media Server with Avaya Aura® Communication Manager*. *White Paper*. August 2015.


Other resources:


[11] *RFC 3261 SIP: Session Initiation Protocol,* http://www.ietf.org/.

[12] *RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, http://www.ietf.org/

# 12. Appendix A: SigMa Script

Following is the Signaling Manipulation script that was used in the configuration of the Avaya SBCE, **Section 7.3.3**. When adding this script as instructed in **Section 7.3.4** enter a name for the script in the Title (e.g., **Clearcom_Script**) and copy/paste the entire script. Note that the user name, shown below as “User123”, will need to be changed with the correct user name provided by Clearcom for registration purpose.

**Title: Clearcom_Script**

```
//Replace Username in "REQUEST-LINE" with "TO" number on Inbound
within session "ALL"
{
act on message where %DIRECTION="INBOUND" and %ENTRY_POINT="PRE_ROUTING"
{
%HEADERS["Request_Line"][1].URI.USER = %HEADERS["To"][1].URI.USER;
}
}
//Insert Username in the FROM header on Outbound
within session "ALL"
{
act on request where %DIRECTION="OUTBOUND" and
%ENTRY_POINT="POST_ROUTING"
{
%fromuser = %HEADERS["From"][1].URI.USER;
%HEADERS["From"][1].URI.USER = "User123";
}
}

//Remove gsid and epv parameters in outbound Contact header
within session "ALL"
{
act on message where %DIRECTION="OUTBOUND" and
%ENTRY_POINT="POST_ROUTING"
{
remove(%HEADERS["Contact"][1].URI.PARAMS["gsid"]);
remove(%HEADERS["Contact"][1].URI.PARAMS["epv"]);
}
}
```

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