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 Claro SIP Trunking Services – 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 Claro SIP Trunking Services.

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.

Claro SIP Trunking Service provides PSTN access via SIP trunks between the enterprise and Claro's network as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

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

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

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1. Introduction

These Application Notes describe the steps required to configure Session Initiation Protocol (SIP) trunk service between the service provider Claro and an Avaya SIP-enabled enterprise solution.

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.

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

During the interoperability testing, a VPN connection was used to connect the simulated Avaya enterprise network to Claro's network via the public Internet. The connection could also be done without the use of VPN, by directly connecting the Avaya SBCE to a public facing SBC located in Claro's network. This is accomplished by assigning public IP addresses, capable of being reached across the public Internet, to the Avaya SBCE (interface B1) and to the Claro SBC.

Customers using an Avaya SIP-enabled enterprise solution with Claro 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 "Claro" 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 Claro 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:

- Response to SIP OPTIONS queries.
- Incoming calls from the PSTN were routed to DID numbers assigned by Claro. Incoming PSTN calls were terminated to the following endpoints: Avaya 96x0 Series IP Deskphones (H.323), 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 Claro's network to the 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.711MU, G.711A and G.729A (Claro'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.
- T.38 fax.

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), 0+10 digits calls (Operator Assisted), and 411 calls (Local Directory Assistance) were not tested.
- The SIP REFER method for call redirection was not tested for reasons noted in **Section 2.2**.

2.2. Test Results

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

- **SIP REFER**: Calls from the PSTN to Communication Manager that are re-directed to another PSTN endpoint by the Communication Manager user, with REFER enabled in Communication Manager (**Network Call Redirection** set to “**y**” under the **trunk-group**), did not work properly. Testing was done with REFER disabled in Communication Manager (**Network Call Redirection** set to “**n**” under the **trunk-group)**, refer to **Section 5.7**.
- **Caller ID display on Call Forward to the PSTN**: For Calls from the PSTN to Communication Manager which were Forwarded back out to the PSTN, the caller ID number displayed at the PSTN was always of the first DID number assigned to the SIP Trunk, 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, regardless of the PSTN number being used to originate the call.
- **No matching codec on outbound calls**: If an unsupported audio codec is received by Claro on the SIP Trunk (e.g., 726A-32K), Claro will respond with “500 Server Internal Error” instead of “488 Not Acceptable Here”, the user will hear re-order tones. This issue does not have any user impact, it is listed here simply as an observation.
- **SIP UPDATE**: SIP UDATEs was causing problems with call transfers to the PSTN and with other call types to the PSTN. For this solution SIP UPDATEs needs to be disabled on the SIP Trunk. SIP UPDATE was disabled on the SIP Trunk, refer to **Section 5.7**.
- **Music on hold**: With Communication Manager configured to play music any time calls were placed on-hold by Communication Manager users, music was not played to PSTN users. The issue was related to the manner in which Claro currently handles the SIP messages Communication Manager includes in the SDP of re-INVITEs it sends when calls are placed on-hold. When a call from the PSTN is placed on-hold by a Communication Manager user, Communication Manager sends a re-INVITE with “sendonly” in the SDP, in response, Claro sends a 200 OK with “inactive” in the SDP. This response caused the audio path to be closed between Communication Manager and the PSTN user, thus resulting in the user not hearing music while on hold. The issue was solved at the Avaya SBCE by removing the “sendonly” message Communication Manager includes in the SDP of re-INVITEs, in response, Claro sends a 200 OK with “sendrecv” in the SDP, opening the audio path in between Communication Manager and the PSTN user, thus resulting in the PSTN user hearing music while he/she is on hold. Refer to **Section 7.2.3**.

- **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 7.2.3**).

2.3. Support

For support on Claro systems visit the corporate Web page at:
http://www.claro.com.do/wps/portal/do/sc/empresas

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 Claro 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 96x0-Series IP Deskphones (H.323).
- 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 a VPN Firewall, followed by the Avaya SBCE. The Avaya SBCE has two physical interfaces: interface **B1** is used to connect to the public network, interface **A1** is used to connect to the private network. Since a VPN connection was used with this solution to connect Claro's network to the enterprise, the **A1** interface was used for access to the private enterprise network and to route calls to Claro's network across the VPN tunnel. In this solution, the **B1** interface was used for remote workers access to the enterprise. The configuration required for the **B1** interface is not discussed in this document.

When a VPN connection is not used, the **B1** interface is normally used for remote workers access to the private network and to route calls to the Service Provider across the public Internet.

All SIP and RTP traffic entering or leaving the enterprise flows through the Avaya SBCE and through the VPN Firewall. The Avaya SBCE provides network address translation at both the IP and SIP layers. The transport protocol between the Avaya SBCE and Claro, through the VPN Tunnel, and across the public Internet, is SIP over UDP. The transport protocol between the Avaya SBCE and Session Manager across the enterprise IP network is SIP over TCP. The transport protocol between Session Manager and Communication Manager across the enterprise IP network is SIP over TLS. Note that for ease of troubleshooting during the testing, the compliance test was conducted with the Transport Method set to TCP between Session Manager and Communication Manager.

For security reasons, any actual public IP addresses and routable DID numbers used in the reference configuration have been masked.

One 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 a dedicated signaling group.

For inbound calls, the calls flowed from Claro to the Avaya SBCE through the VPN Tunnel, 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 Claro's network through the VPN Tunnel.

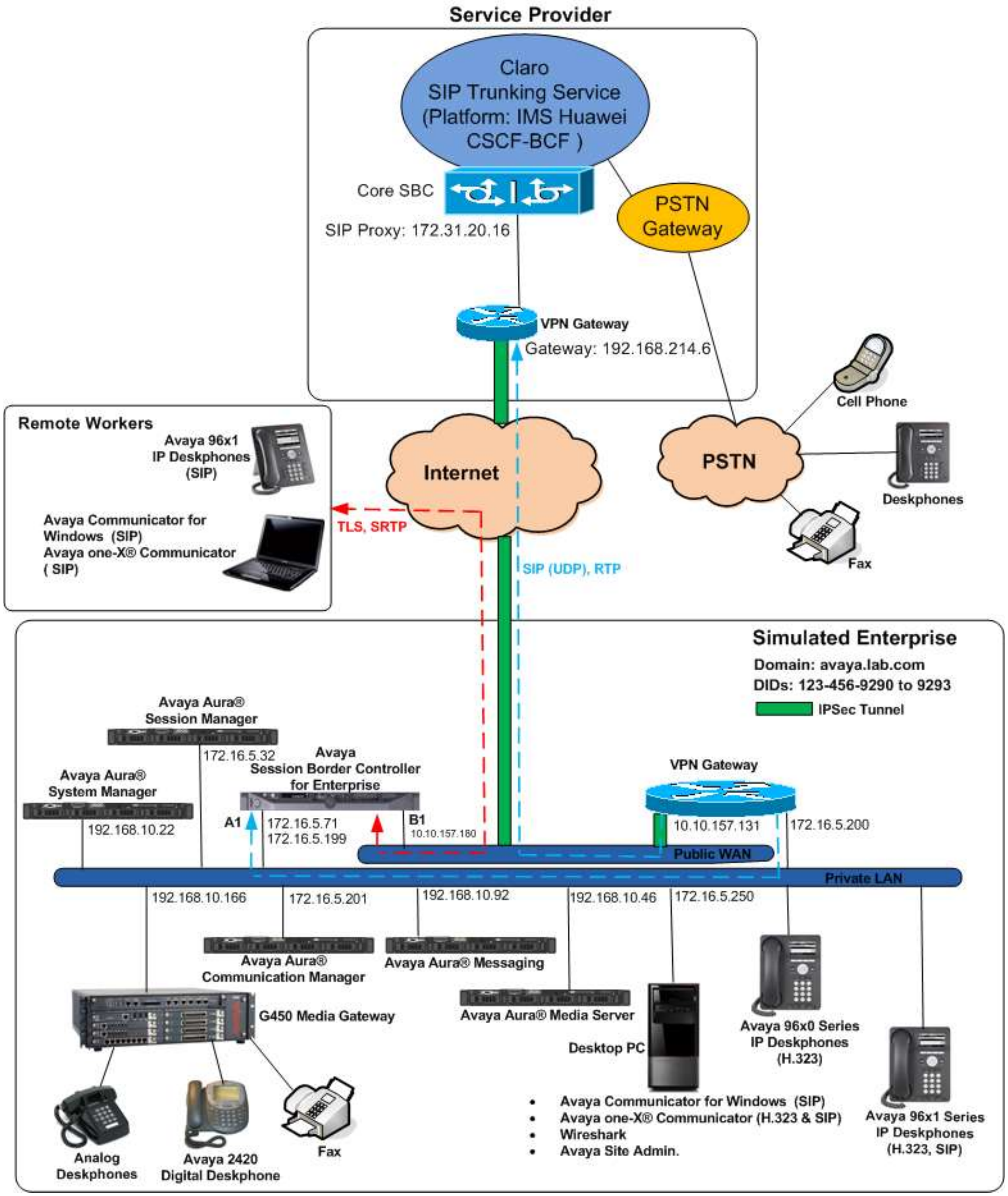


Figure 1: Avaya SIP-enabled Enterprise Solution and Claro SIP Trunking Service

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	7.0.0.1.0 (00.0.441.0-22477)
Avaya Aura® Session Manager running on VMware ESXi 5.5 platform	7.0.0.0 (7.0.0.0.700007)
Avaya Aura® System Manager running on VMware ESXi 5.5 platform	7.0.0.0 Build No. 7.0.0.0.16266-7.0.9.912 Software Update Rev. No. 7.0.0.0.3929
G450 Gateway	37.19.0
Avaya Session Border Controller for Enterprise running on a DELL R210 V2 Server	7.0.0-21-6602
Avaya Aura® Media Server running on VMware ESXi 5.5 platform	7.7.0.226
Avaya Aura® Messaging running on VMware ESXi 5.5 platform	6.3.3 Service Pack 3 (MSG-03.0.141.0-348_0304)
Avaya Aura® Integrated Management Site Administrator	6.0.07
Avaya one-X® Communicator (SIP & H.323)	6.2.7.03-SP7
Avaya Communicator for Windows (SIP)	2.1.2.75
Avaya 96x0 Series IP Deskphones (H.323)	Avaya one-X® Desk phone Edition Version S3.250A
Avaya 96x1 Series IP Deskphones (H.323)	Avaya one-X® Deskphone H.323 Version 6.6029
Avaya 96x1 Series IP Deskphones (SIP)	Avaya one-X® Deskphone SIP Version 7.0.0.39
Avaya 2420 Series Digital Deskphone	--
Lucent Analog Deskphone	--
Claro	
IMS Huawei CSCF-BCF	V100R010C00SPC100
SBC Huawei SessionEngine2600	V200R009ENGC30SPC100

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 Claro. 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-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 are available and ***122*** are in use. The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity.

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

IP PORT CAPACITIES                                              USED
                    Maximum Administered H.323 Trunks: 12000 10
          Maximum Concurrently Registered IP Stations: 18000 2
            Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
             Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                       Maximum Video Capable Stations: 41000 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***.

```
change 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.

```
change system-parameters features                               Page   9 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS

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

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

INTERNATIONAL CALL ROUTING PARAMETERS
                 Local Country Code:
          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

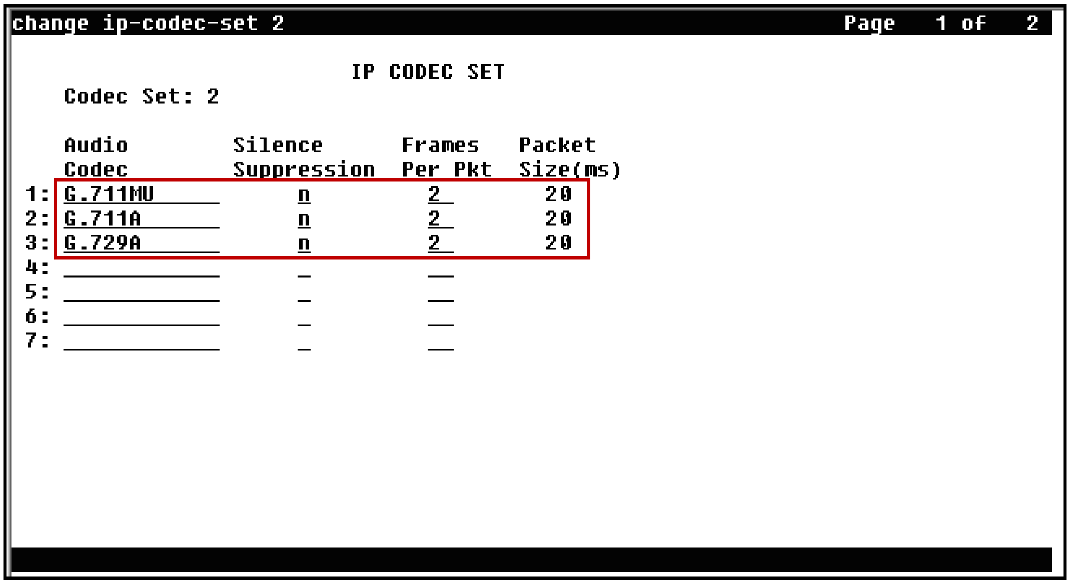
Use the **change node-names ip** command to verify that node names have been previously defined for the IP addresses of the Avaya server running Communication Manager **(procr**), and for Session Manager (**Lab-HG-SM**). These node names will be needed for defining the Service Provider signaling group 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

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. Claro SIP Trunking supports G.711MU, G.711A and G.729A. Thus, these codecs were included in this set. Enter ***G.711MU, G.711A*** and ***G.729A*** in the **Audio Codec** column of the table; this is Claro's preferred codec order. Default values can be used for all other fields.

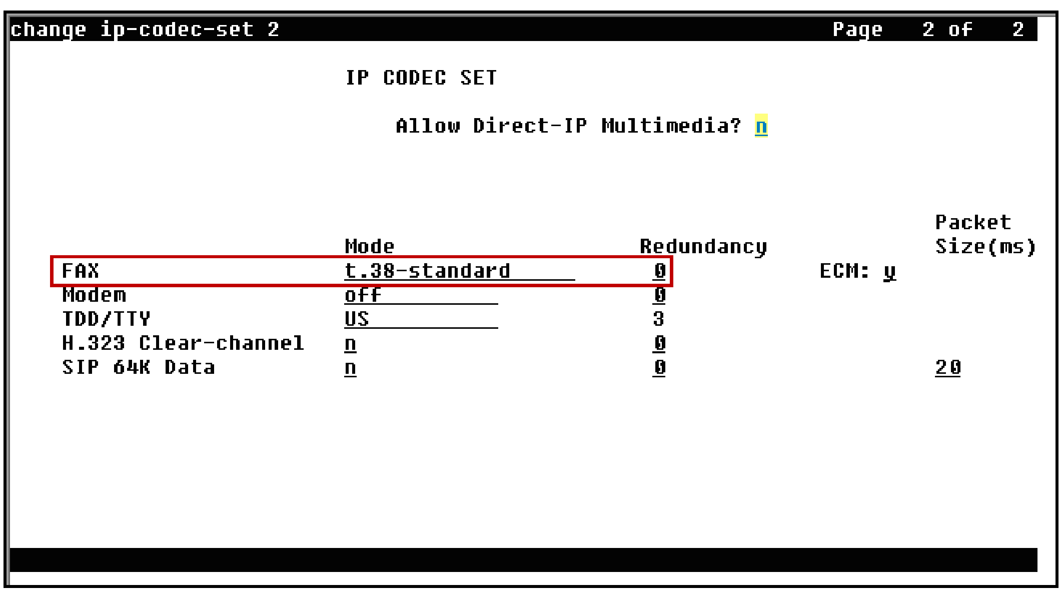


```
change ip-codec-set 2                                         Page   1 of   2

                          IP CODEC SET
    Codec Set: 2

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

On **Page 2**, set the **Fax Mode** to ***t.38-standard*** (T.38 fax is supported by Claro).



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

                          IP CODEC SET

                              Allow Direct-IP Multimedia? n



                                                                    Packet
                      Mode                   Redundancy             Size(ms)
    FAX               t.38-standard              0          ECM: y
    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 or the Avaya Aura® Media Server. 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). Note that for ease of troubleshooting during testing, the compliance test was conducted with the **Transport Method** set to ***tcp***. 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 TCP in **Sections 6.6** and **7.2.4**. Lastly, the transport method between the Avaya SBCE and Claro is UDP. This is defined in **Section 7.2.4**.
- 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 ***5070***. (For TCP, the well-known port value for SIP is 5060).

- 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: tcp
       Q-SIP? n
    IP Video? n                                     Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                      Far-end Node Name: Lab-HG-SM
 Near-end Listen Port: 5070                     Far-end Listen Port: 5070
                                             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 ***public***. This field specifies the format of the calling party number (CPN) sent to the far-end. Public numbers are automatically preceded with a + sign when passed in the SIP “From”, “Contact”, “P-Asserted Identity” and “Diversion” headers.

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: public
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? y
                                                Replace Unavailable Numbers? y

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




 Show ANSWERED BY on Display? y
```

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 message for transferring calls off-net to the PSTN, refer to **Section 2.2**.
- Set the **Send Diversion Header** field to ***y***. When enabled, the Diversion Header (in the outbound INVITE message) provides additional information to the network if the call has been re-directed. This is needed to support call forwarding of inbound calls back to the PSTN and some Extension to Cellular (EC500) call scenarios.
- Set the **Support Request History** field to ***n***.
- Set the **Telephone Event Payload Type** to ***101***. The value preferred by Claro.
- Set the **Convert 180 to 183 for Early Media** to ***y***.
- Set the **Always Use re-INVITE for Display Updates** to ***y***. When enabled, UDATE messages are disabled and are replaced with re-INVITE messages. UPDATE messages were causing problems and need to be disabled in Communication Manager (refer to **Section 2.2**).
- Set the **Identity for Calling Party Display** to ***From***.

```
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? y
                                   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: From
            Block Sending Calling Party Location in INVITE? n
                 Accept Redirect to Blank User Destination? n
                                              Enable Q-SIP? n

          Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
                                      Request URI Contents: may-have-extra-digits
```

5.8. Calling Party Information

The calling party number is sent in the SIP "From", "Contact" and "PAI" headers. Since public numbering was selected to define the format of this number (**Section 5.7**), use the **change public-unknown-numbering** command to create an entry for each extension which has a DID assigned. The DID numbers are assigned by the Service Provider. They are used to authenticate the caller. 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, 3044, 3045 and 3048. These 11-digit numbers were used for the outbound calling party information on the Service Provider trunk when calls were originated from these extensions.

```
change public-unknown-numbering 1                               Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                                Total
Ext Ext             Trk       CPN               CPN
Len Code            Grp(s)    Prefix            Len
                                                       Total Administered: 6
 4  3                                            4        Maximum Entries: 9999
 4  5                                            4
 4  3041            2         11234569290       11     Note: If an entry applies to
 4  3044            2         11234569226       11     a SIP connection to Avaya
 4  3045            2         11234569254       11     Aura(R) Session Manager,
 4  3048            2         11234569243       11     the resulting number must
                                                       be a complete E.164 number.

                                                       Communication Manager
                                                       automatically inserts
                                                       a '+' digit in this case.
```

In a real customer environment, normally DID numbers are comprised of the local extension plus a prefix. If this is true, then a single public numbering entry can be applied for all extensions. In the example below, all stations with a 4-digit extension beginning with 9 will send the calling party number as the **CPN Prefix** plus the extension number. The example shown in the screenshot below is assuming that the local extensions in the DID numbers begin with a 9 (e.g., 1123456**9**xxx).

```
change public-unknown-numbering 1                               Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                          Total
Ext Ext                 Trk       CPN            CPN
Len Code                Grp(s)    Prefix         Len
                                                       Total Administered: 6
 4  3                                             4       Maximum Entries: 9999
 4  5                                             4
 4  9                   2         1123456        11    Note: If an entry applies to
                                                       a SIP connection to Avaya
                                                       Aura(R) Session Manager,
                                                       the resulting number must
                                                       be a complete E.164 number.

                                                       Communication Manager
                                                       automatically inserts
                                                       a '+' digit in this case.
```

5.9. Inbound Routing

DID numbers received from Claro were mapped to extensions using the incoming call handling treatment of the receiving trunk group. Use the **change inc-call-handling-trmt** command to create an entry for each DID number. Note that since Claro includes the “+” in the **Request-URI** of INVITE messages, the “+” was added to the **Incoming Number Digits**.

```
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     12  +11234569290      12   3041
 public-ntwrk     12  +11234569226      12   3044
 public-ntwrk     12  +11234569254      12   3045
 public-ntwrk     12  +11234569243      12   3048
 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
```

In a real customer environment, where DID numbers are usually comprised of a local extension plus a prefix, a single entry can be applied for all extensions, like in the example shown below.

```
change inc-call-handling-trmt trunk-group 2                  Page   1 of   3
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 public-ntwrk   12 +1123456         8
 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
 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. See **Section 2.1** for the complete list of call types tested. All dialed strings are mapped to **route pattern 2** which contains the SIP trunk to the Service Provider (as defined next).

```
change ars analysis 178                                         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
    178                    11   11   deny       fnpa           n
    1786                   11   11   2          fnpa           n
    179                    11   11   deny       fnpa           n
    180                    11   11   deny       fnpa           n
    1800                   11   11   2          fnpa           n
    1800555                11   11   deny       fnpa           n
    1809                   11   11   2          hnpa           n
    181                    11   11   deny       fnpa           n
    182                    11   11   deny       fnpa           n
    183                    11   11   deny       fnpa           n
    184                    11   11   deny       fnpa           n
    185                    11   11   deny       fnpa           n
    186                    11   11   deny       fnpa           n
    187                    11   11   deny       fnpa           n
    188                    11   11   deny       fnpa           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.
- **Pfx Mrk**: ***1*** The prefix mark (**Pfx Mrk**) of one will prefix any FNPA 10-digit number with a 1 and leave numbers of any other length unchanged. This will ensure 1 + 10 digits are sent to the Service Provider for long distance North American Numbering Plan (NANP) numbers. All HNPA 10 digit numbers are left unchanged.

```
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          1                                                    n   user
 2:                                                                      n   user
 3:                                                                      n   user
 4:                                                                      n   user
 5:                                                                      n   user
 6:                                                                      n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM Sub  Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
 1: y y y y y n  n            rest                                          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:

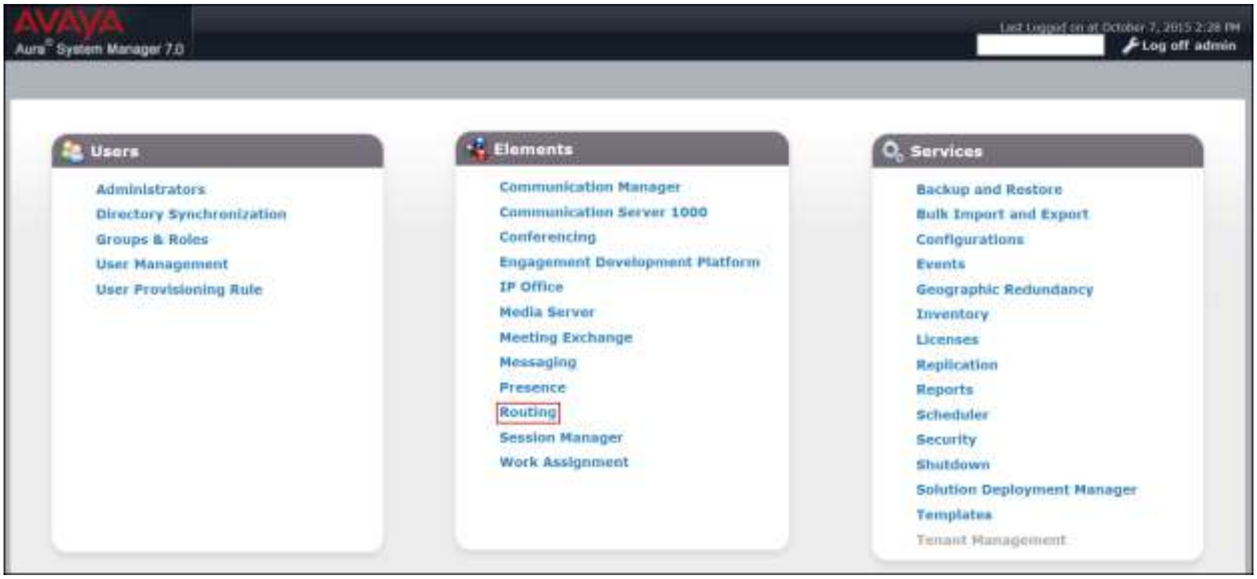
- SIP domain
- Logical/physical Location that can be occupied by SIP Entities
- Adaptation module to perform dial plan manipulation.
- SIP Entities corresponding to Communication Manager, the Avaya SBCE and Session Manager
- Entity Links, which define the SIP trunk parameters used by Session Manager 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.

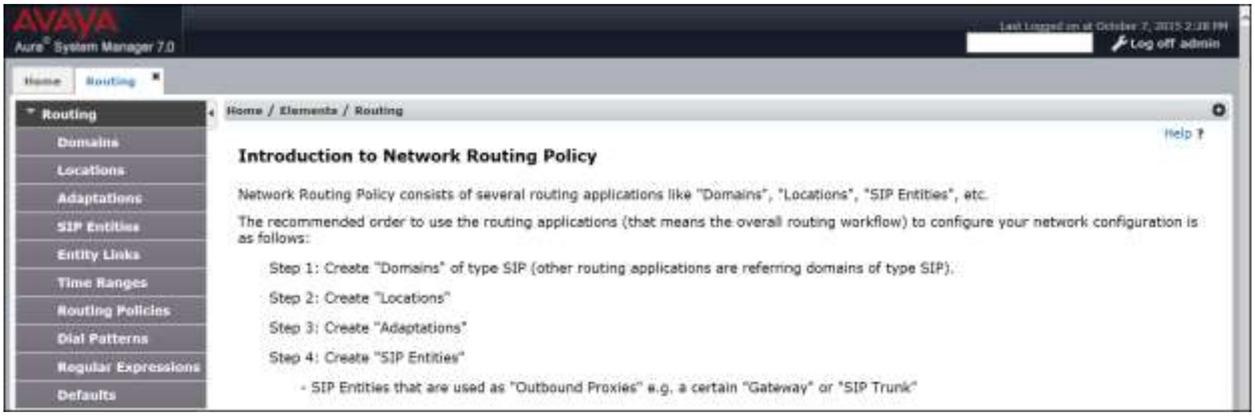
Note: Some of the default information in the screenshots that follow may have been cut out (not included) for brevity

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.



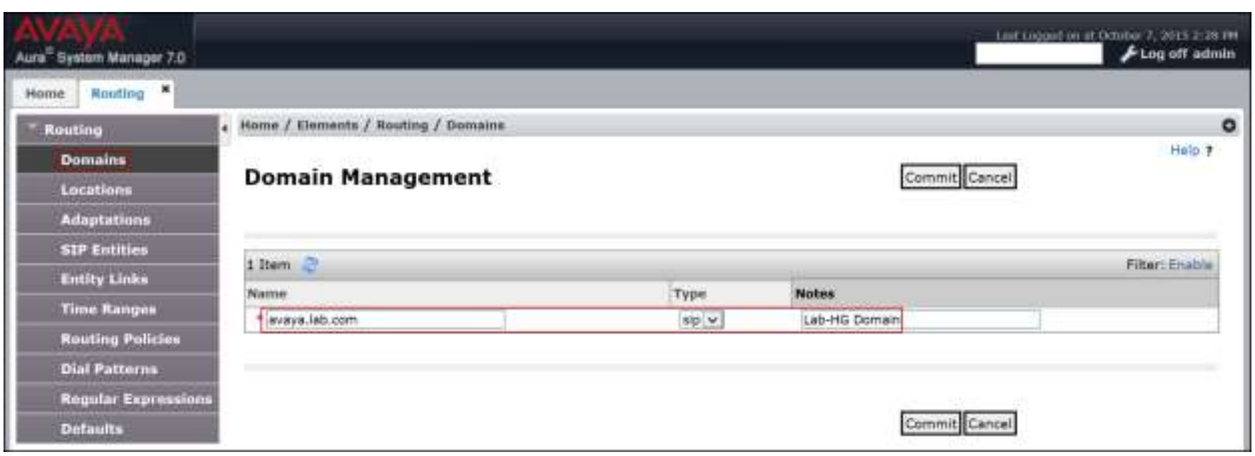
6.2. Specify SIP Domain

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**.



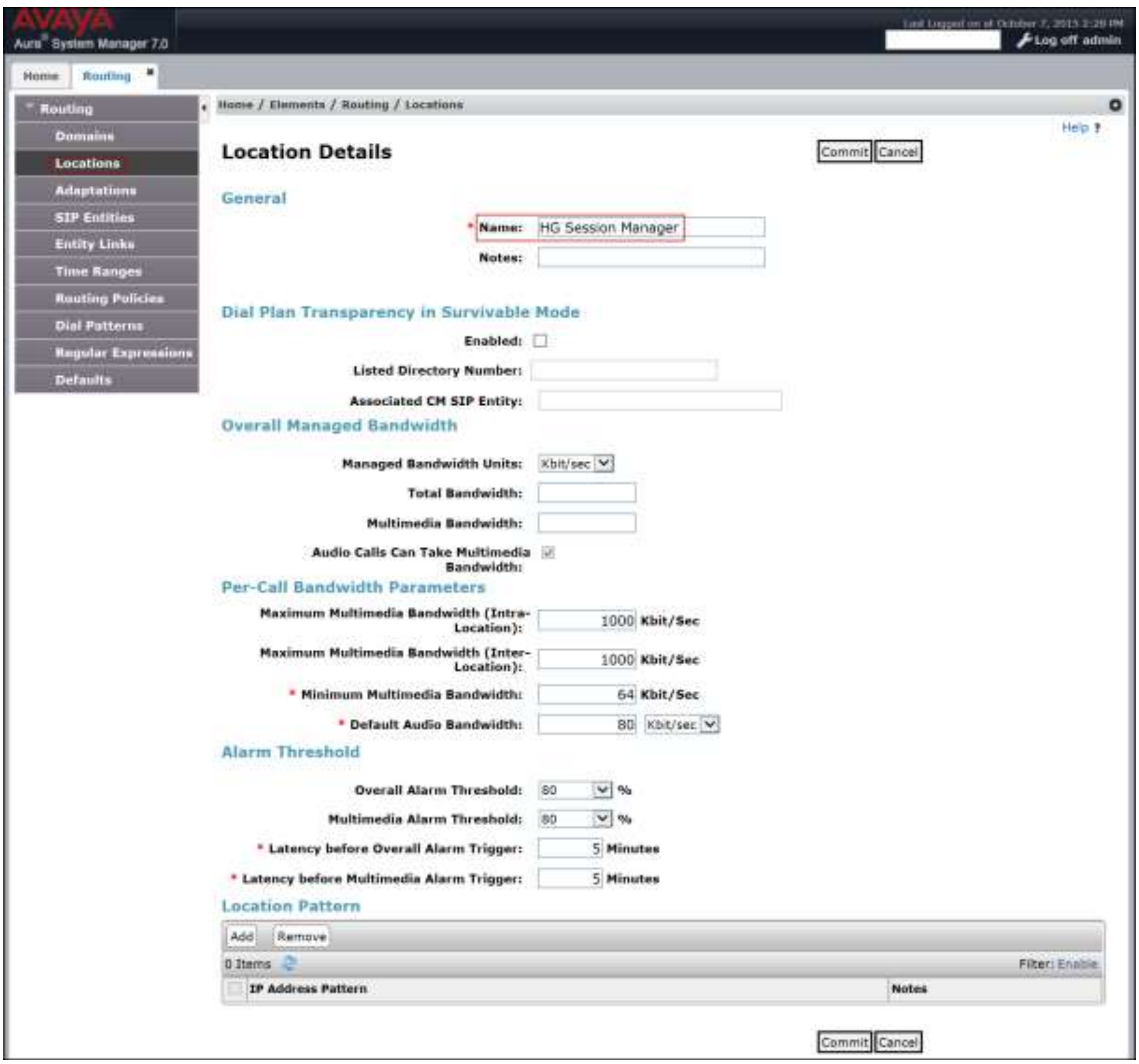
6.3. Add Location

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).

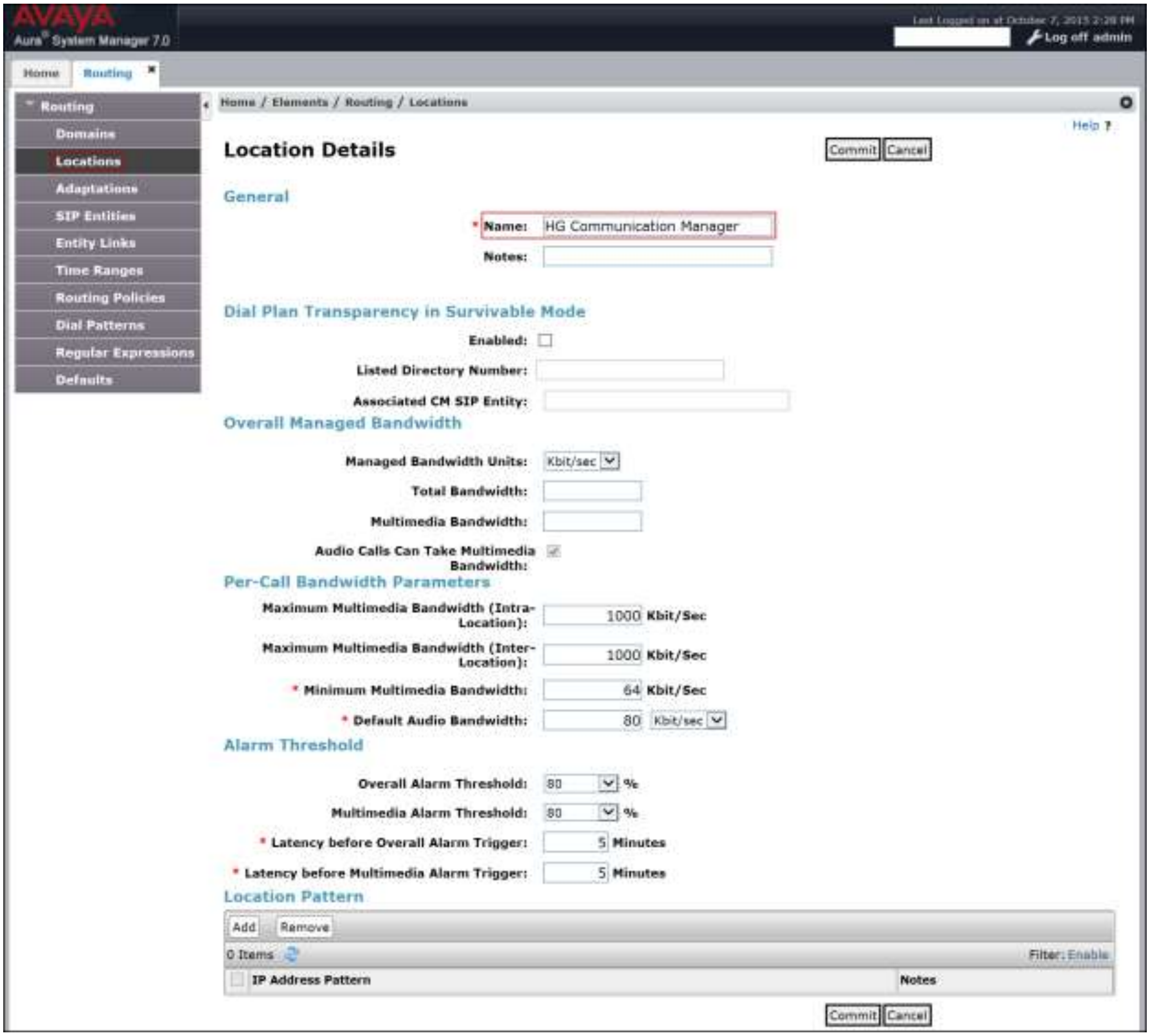
In the **General** section, enter the following values. Use default values for all remaining fields:

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

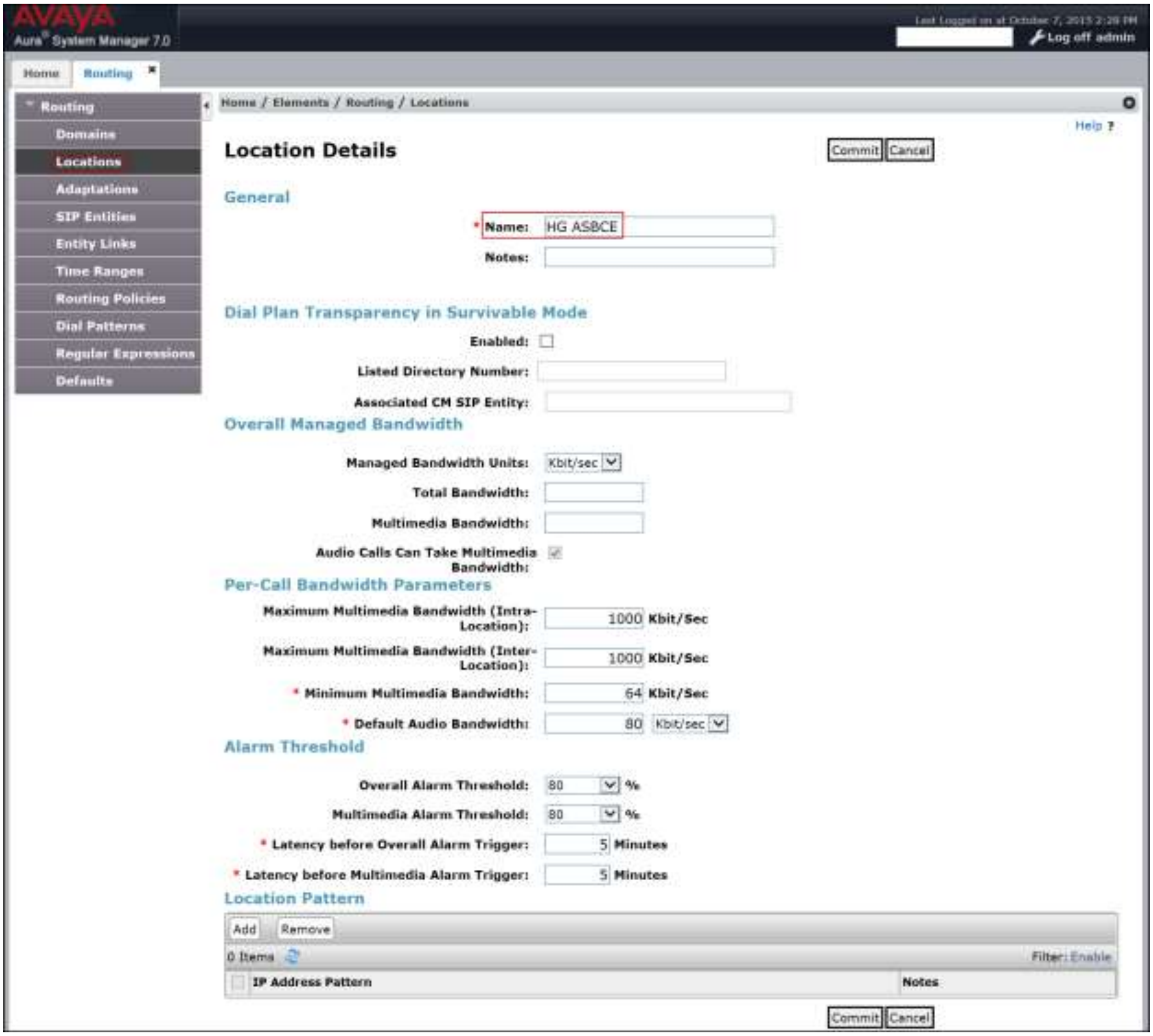
The screen below shows the **HG Session Manager** location. This location will be assigned later to the SIP Entity corresponding to Session Manager.



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.

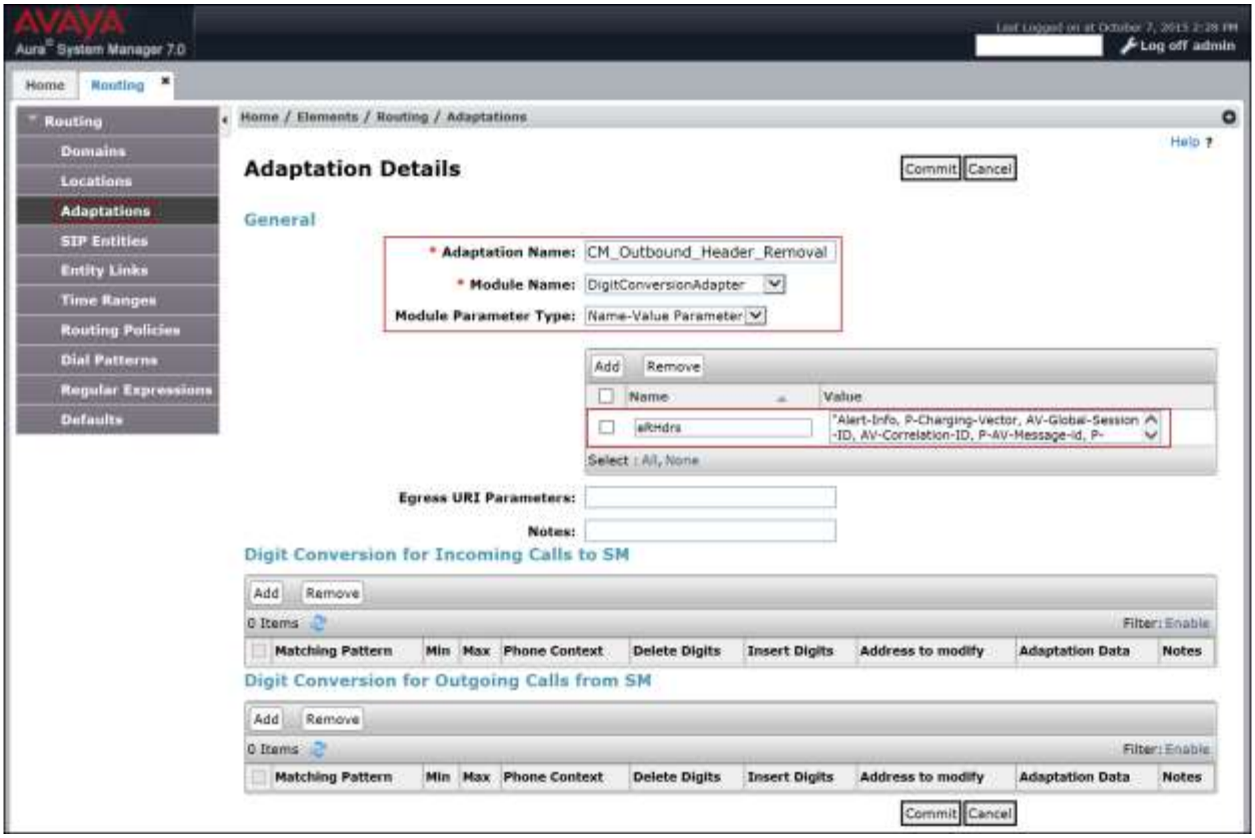
Navigate to **Routing → Adaptations** in the left-hand navigation pane and click the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

- **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.

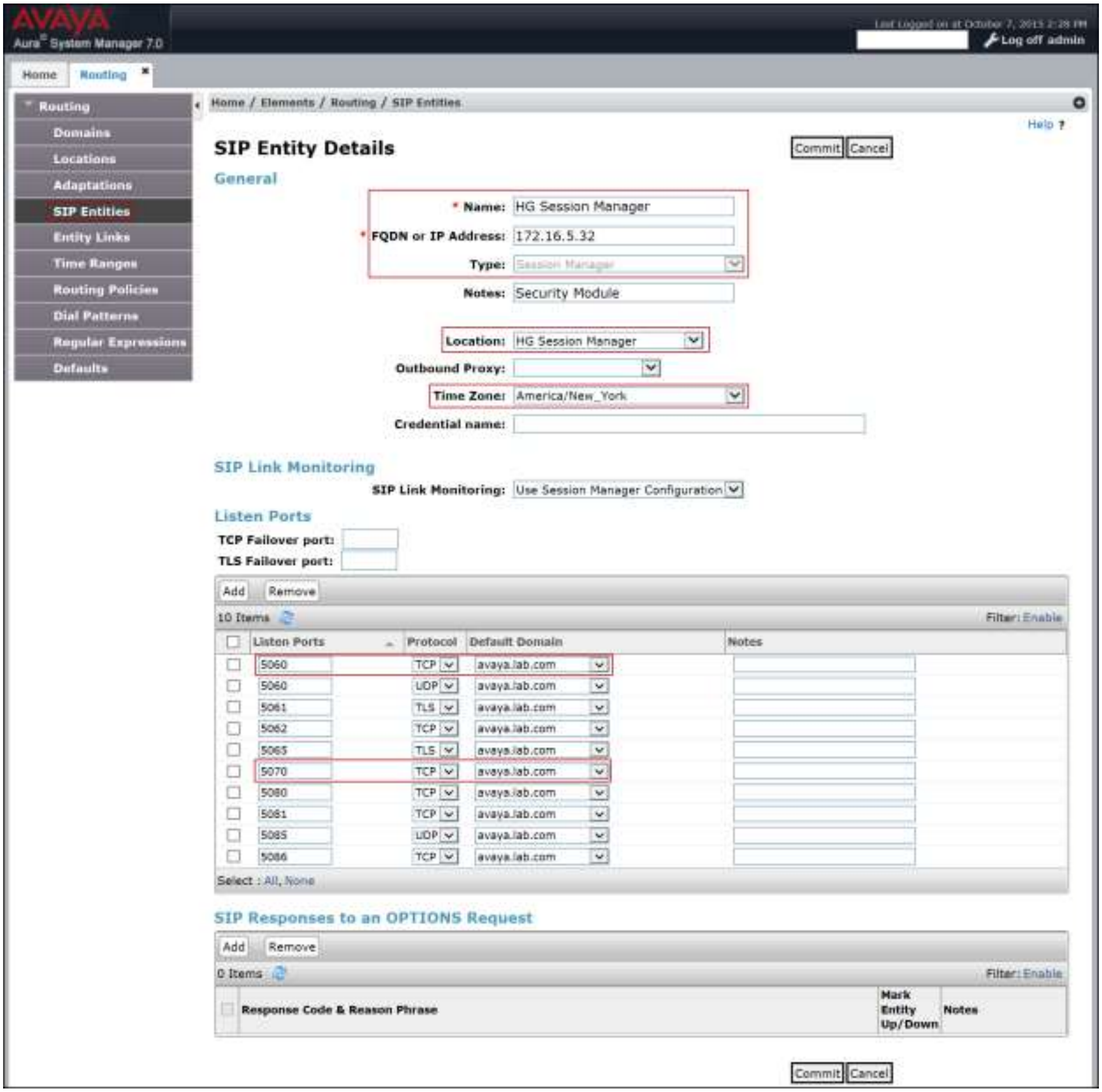
In the **Port** section, click **Add** and enter the following values. Use default values for all remaining fields:

- **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:

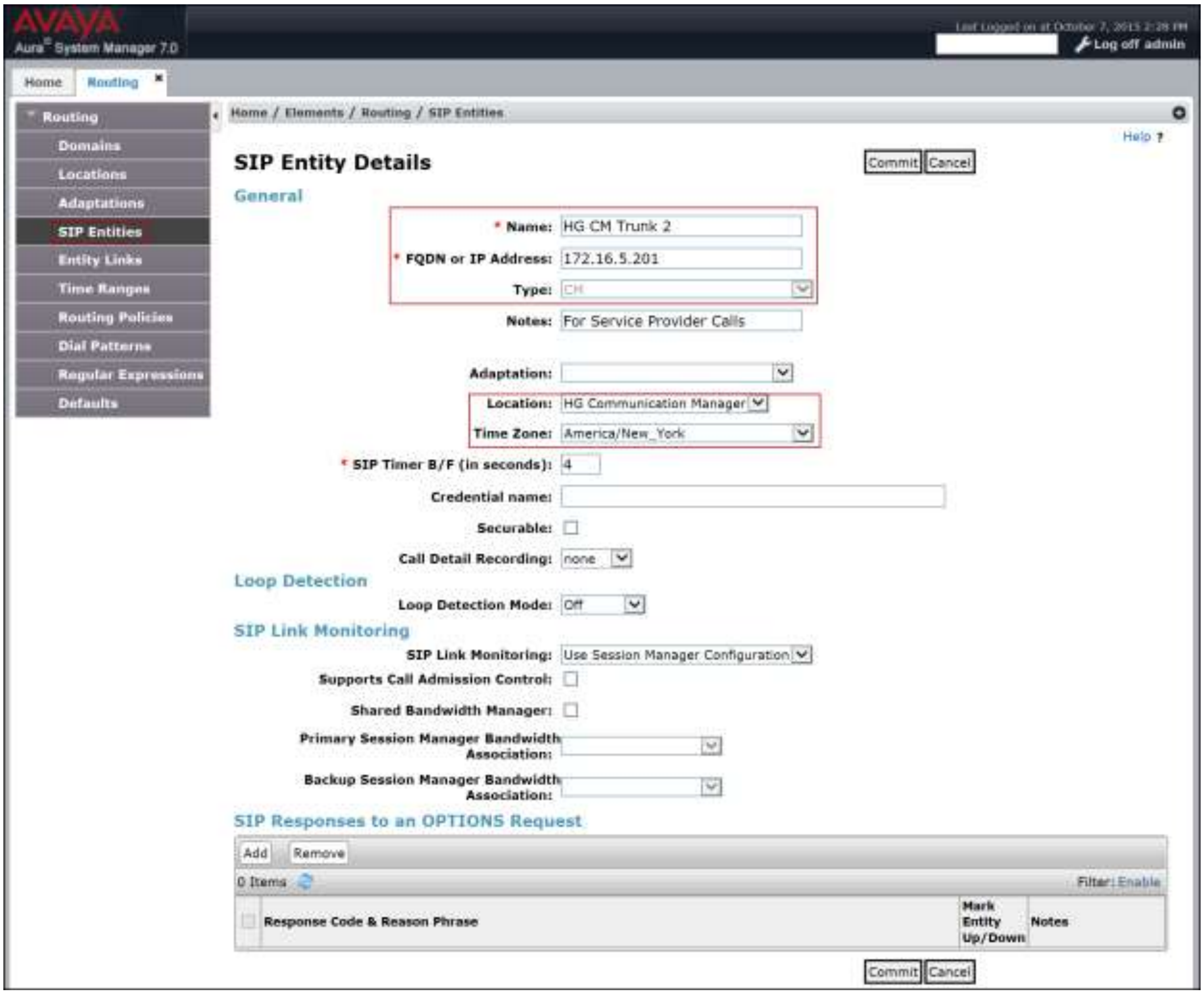
- ***5060*** with ***TCP*** for connecting to the Avaya SBCE.
- ***5070*** with ***TCP*** for connecting to Communication Manager.



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.

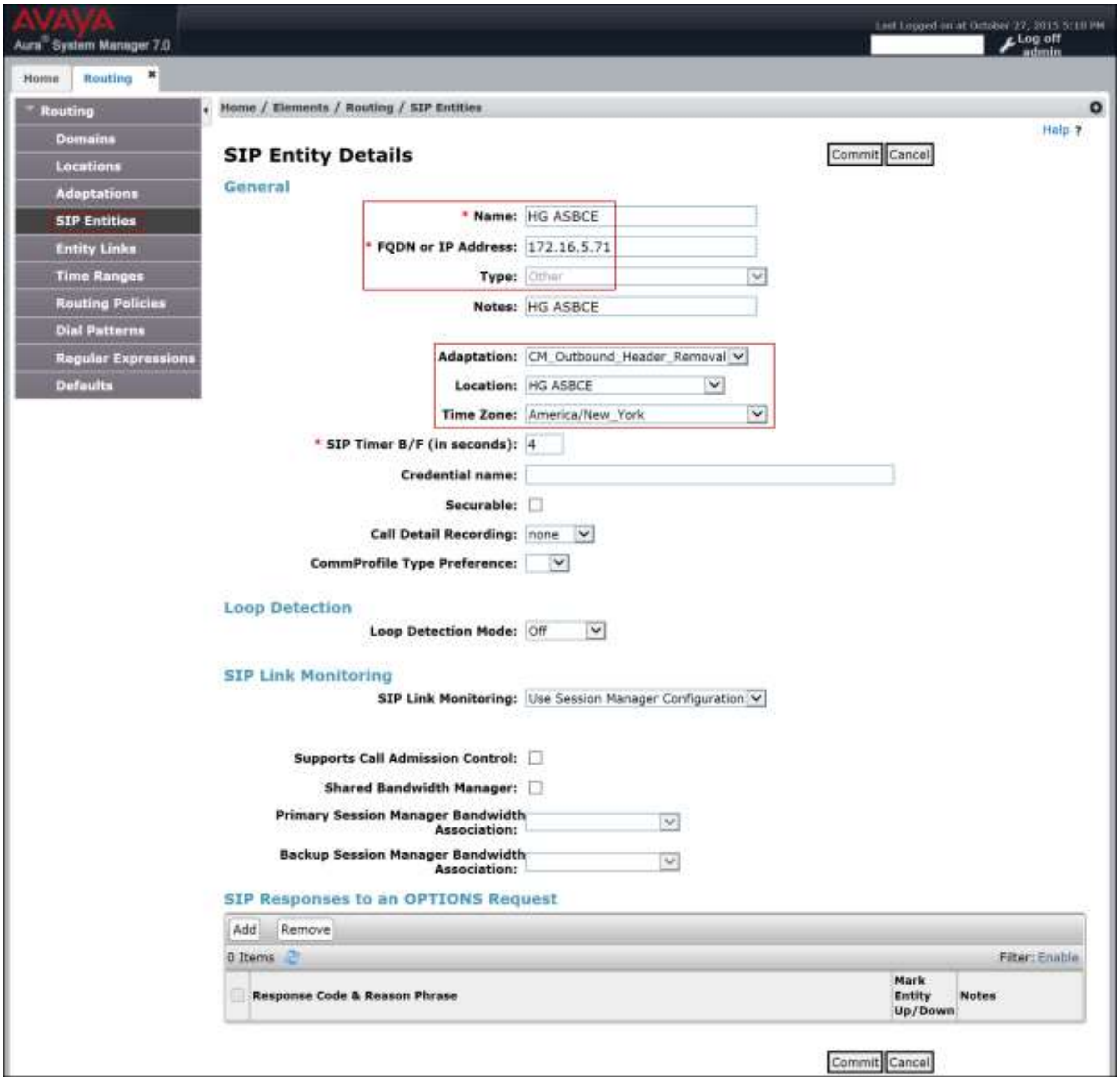
The name ***HG CM Trunk 2***, the IP of the Communication Manager **procr** interface in **Section 5.3**, the **Type** of ***CM*** for Communication Manager, the Location ***HG Communication Manager*** created in **Section 6.3** and the **Time Zone** were used.



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:

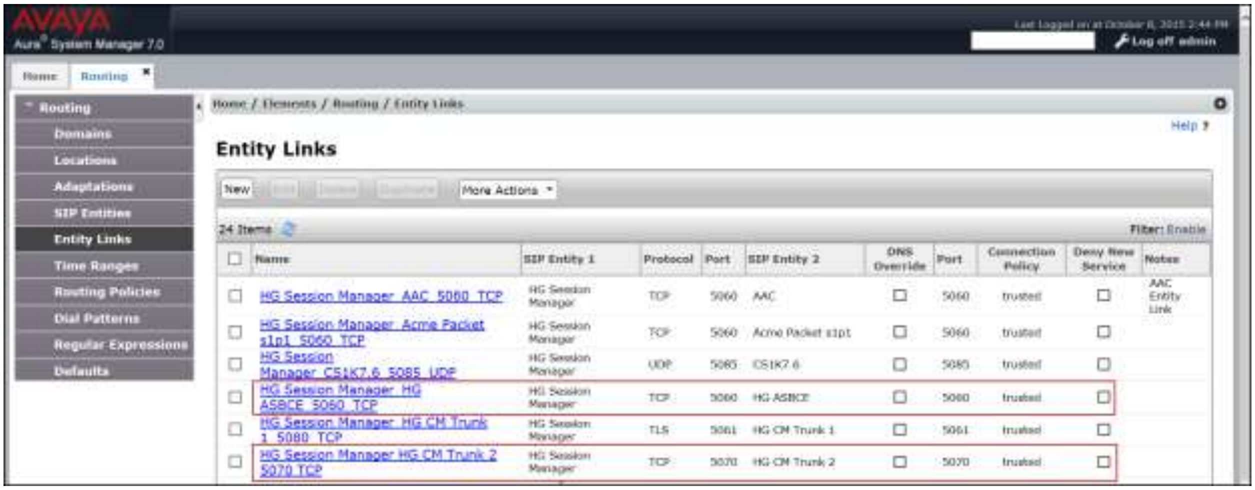
- **Name:** Enter a descriptive name.
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select the transport protocol used for this link. For Communication Manager this was matched to the **Transport Method** defined on the Communication Manager signaling group in **Section 5.6**. For the Avaya SBCE, this was matched to the **Transport** defined on the **Server Configuration** for Session Manager (Call Server) in **Section 7.2.4**.
- **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 7.2.4**.
- **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 **TCP Port** defined for the private **Signaling Interface** on the Avaya SBCE in **Section 7.4.3**.
- **Connection Policy:** Select ***Trusted***.
- Click **Commit** to save.

The following screens illustrate the entity links to Communication Manager and to the Avaya SBCE. It should be noted that in a customer environment the entity link to Communication Manager would normally use TLS. For the compliance test, TCP was used to aid in troubleshooting since the signaling traffic is not encrypted.

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:

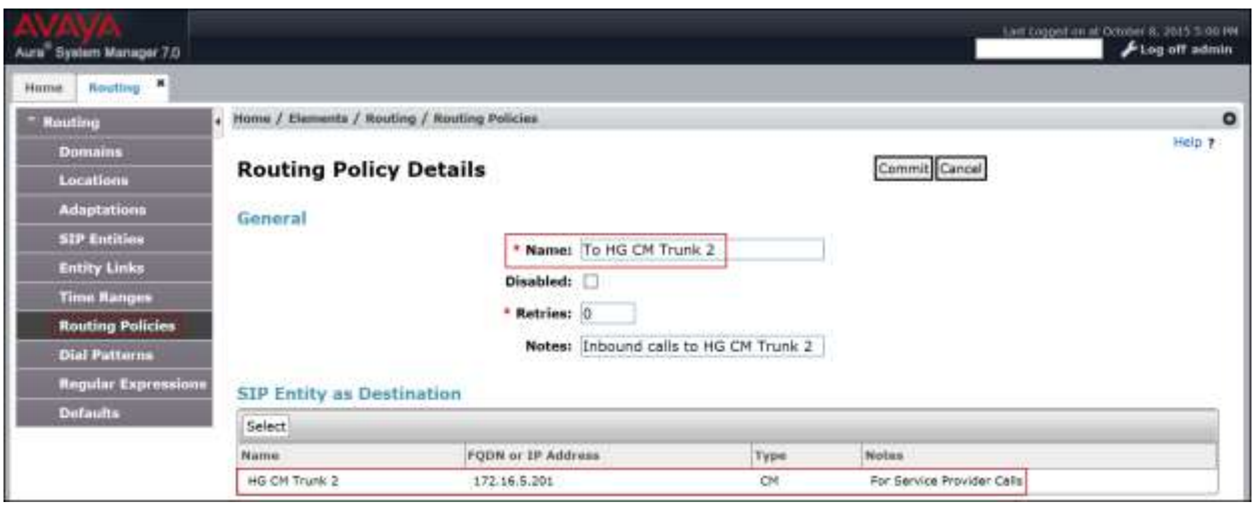
In the **General** section, enter the following values. Use default values for all remaining fields:

- **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 Claro 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 following screens:

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).

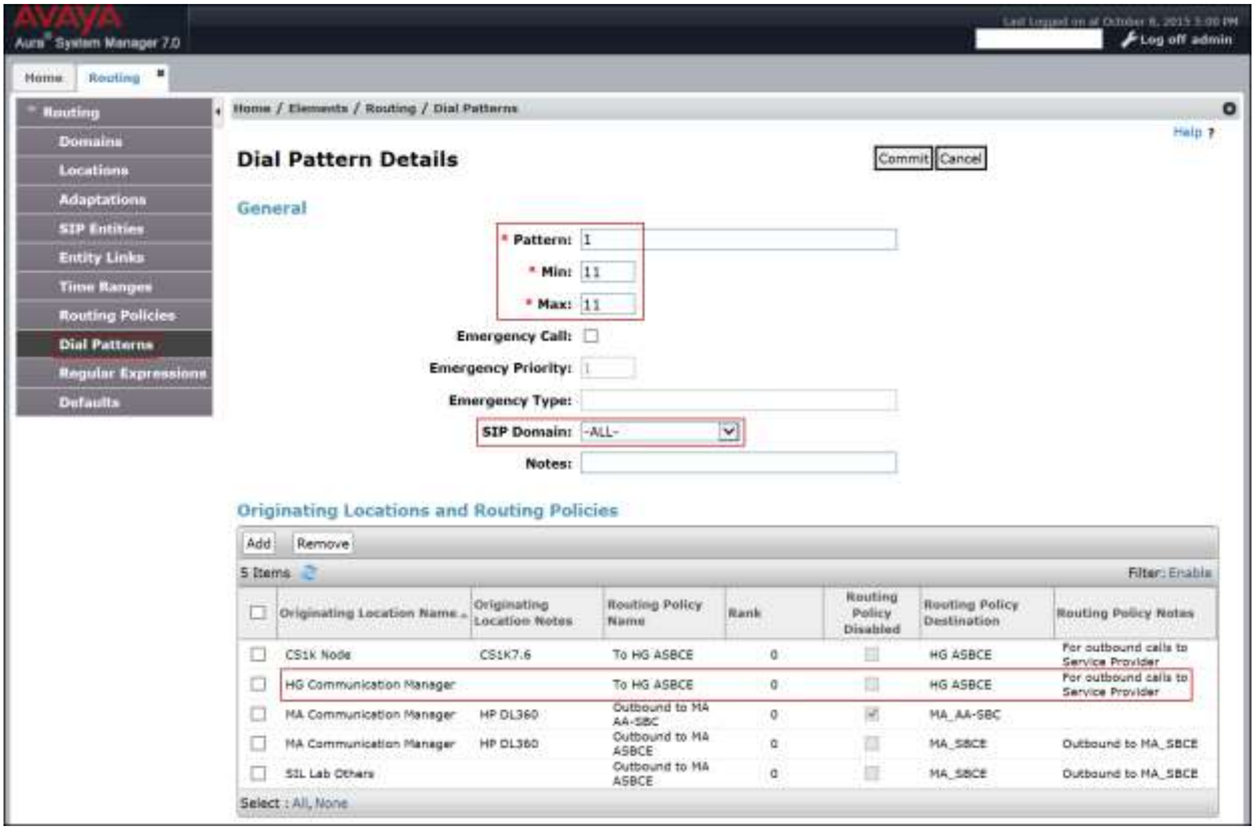
In the **Originating Locations and Routing Policies** section, click **Add**. From the **Originating Locations and Routing Policy List** that appears (not shown), select the appropriate originating location for use in the match criteria. Lastly, select the routing policy from the list that will be used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

- Click **Commit** to save.

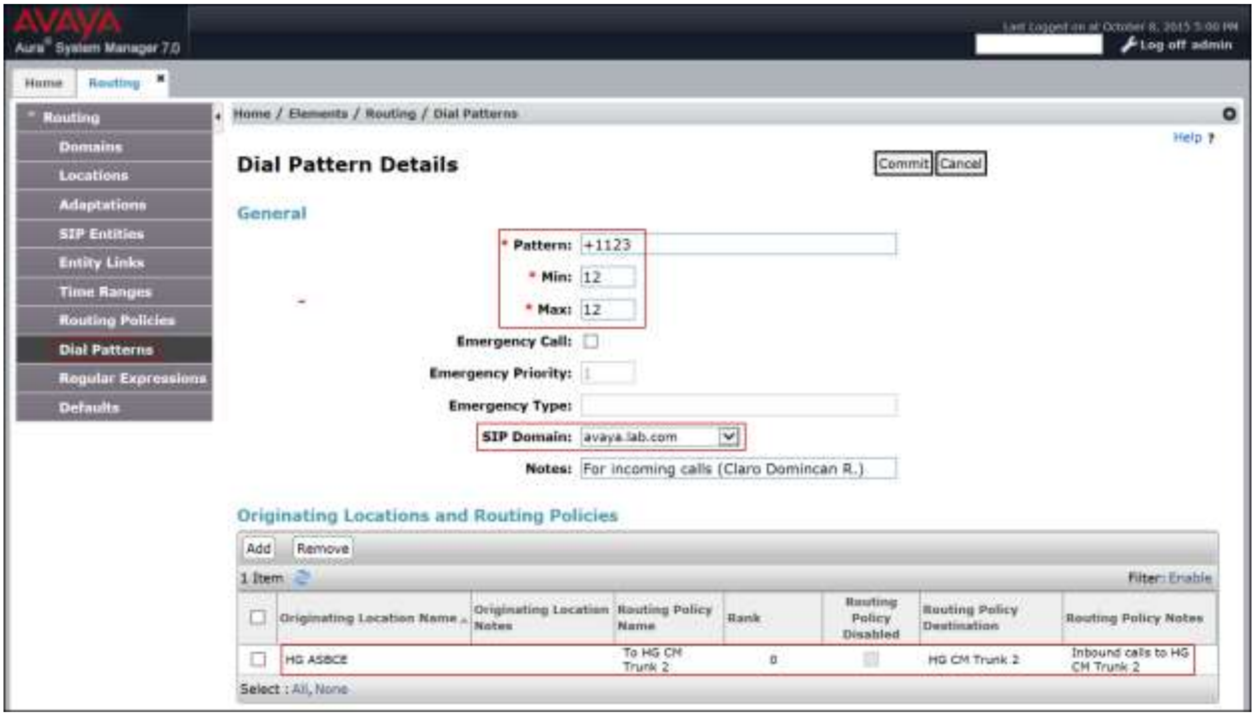
Examples of dial patterns used for the compliance testing are shown below.

The first example shows dial pattern ***1***, 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 outbound calls to the PSTN.

Note: The SIP Domain was set to –ALL- since dial pattern 1 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 ***+1123*** matching the “+” and the first four digits sent by Claro on inbound calls to the enterprise. The pattern also matches the “+” and the first four digits of DID numbers assigned to Communication Manager in **Section 5.9 Inbound Routing**. This dial pattern was configured with the destination SIP Domain of ***avaya.lab.com***, Originating Location Name ***HG ASBCE,*** and Routing Policy name ***To HG CM Trunk 2.***



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.

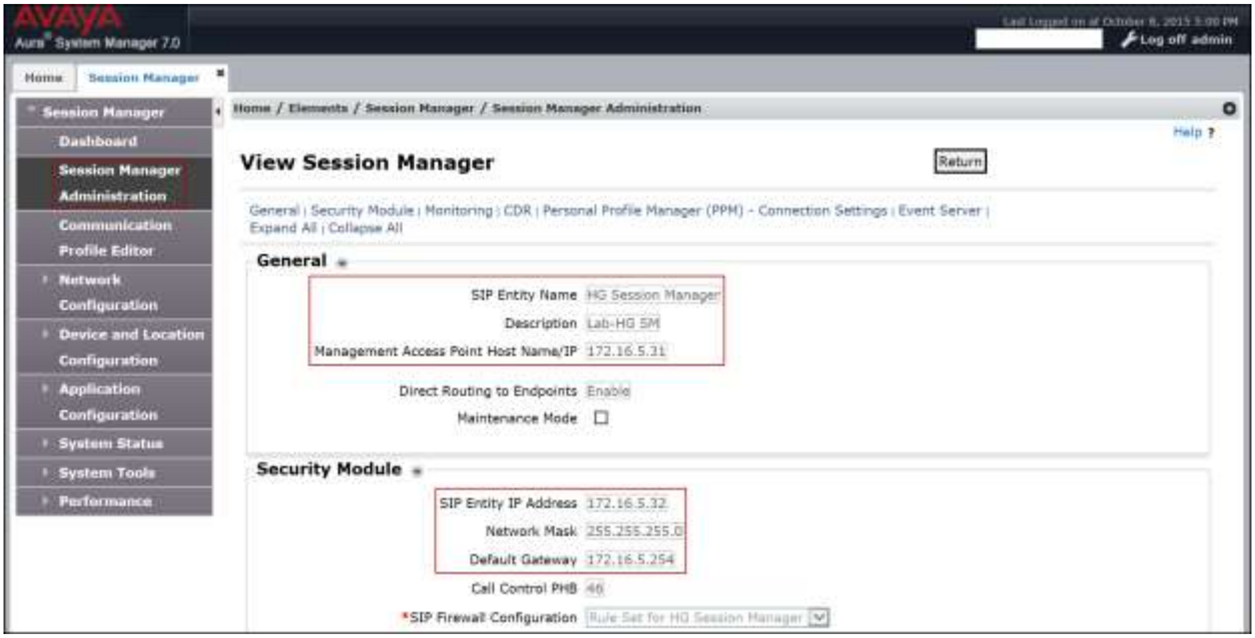
In the **Security Module** section, enter the following values:

- **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 Claro's SIP Trunking service.

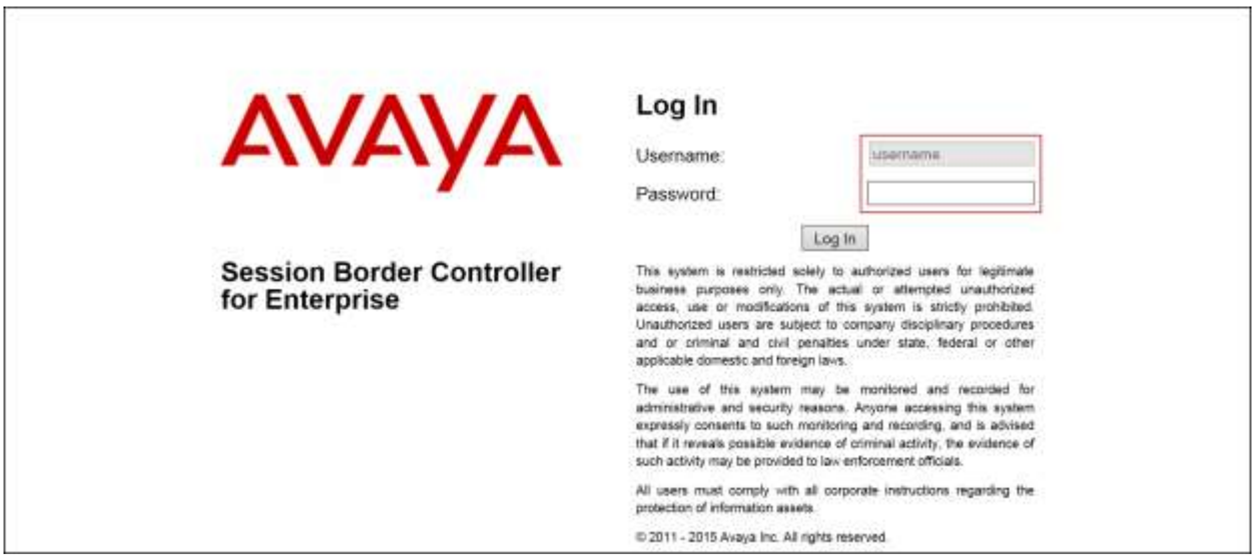
It is assumed that the Avaya SBCE was provisioned and is ready to be used; the configuration shown here is accomplished using the Avaya SBCE web interface.

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.

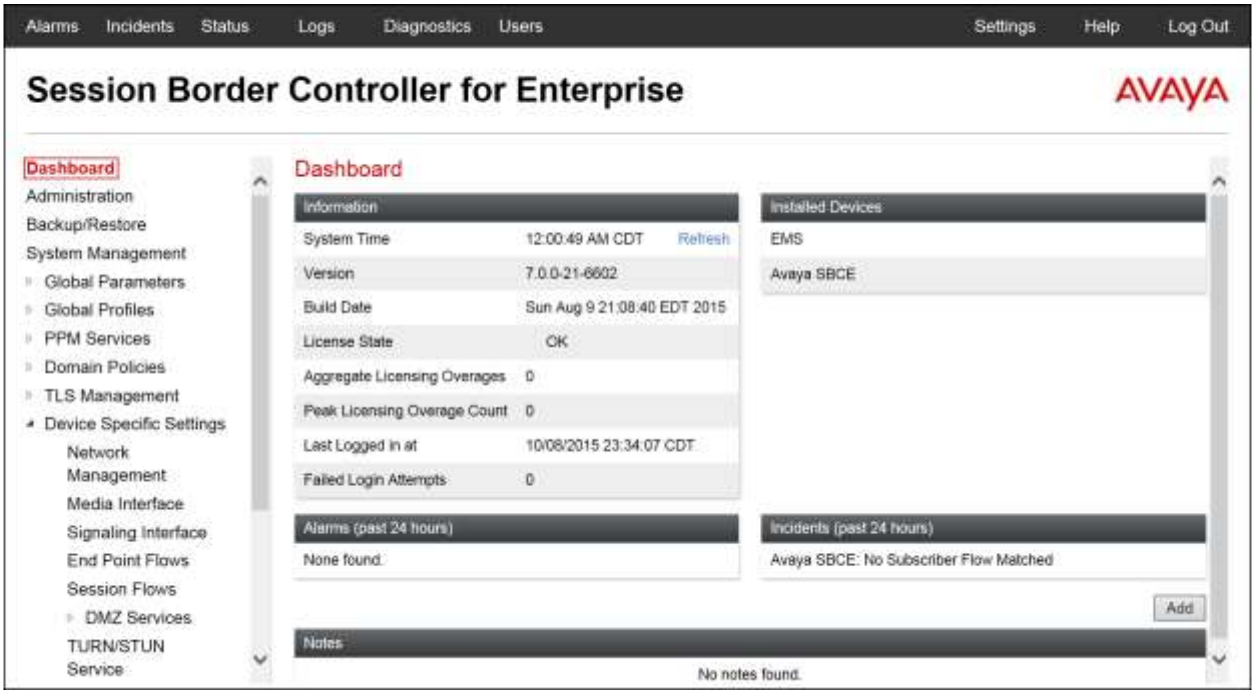
7.1. Log in Avaya SBCE

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.

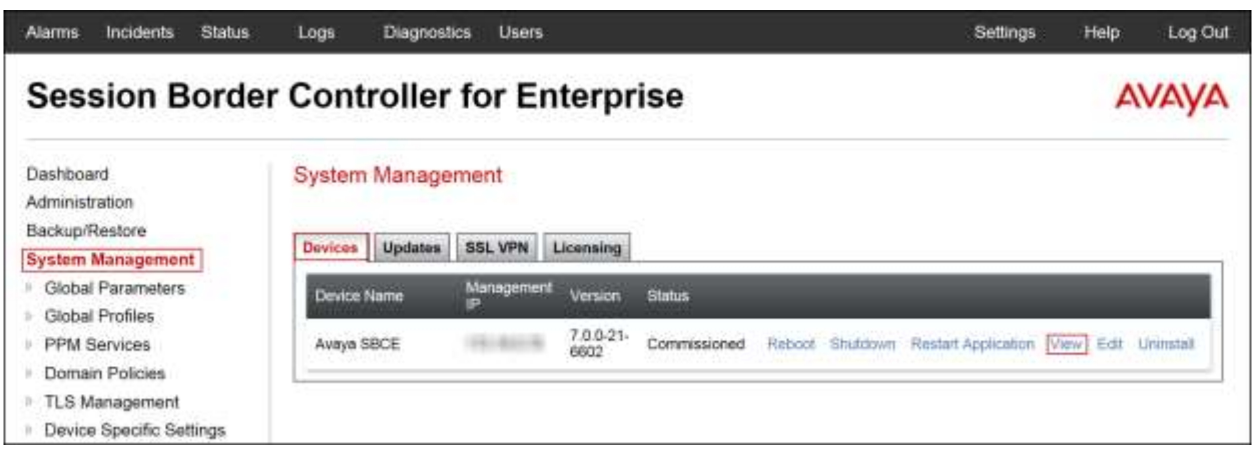
Enter the appropriate credentials and then click **Log In**.



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

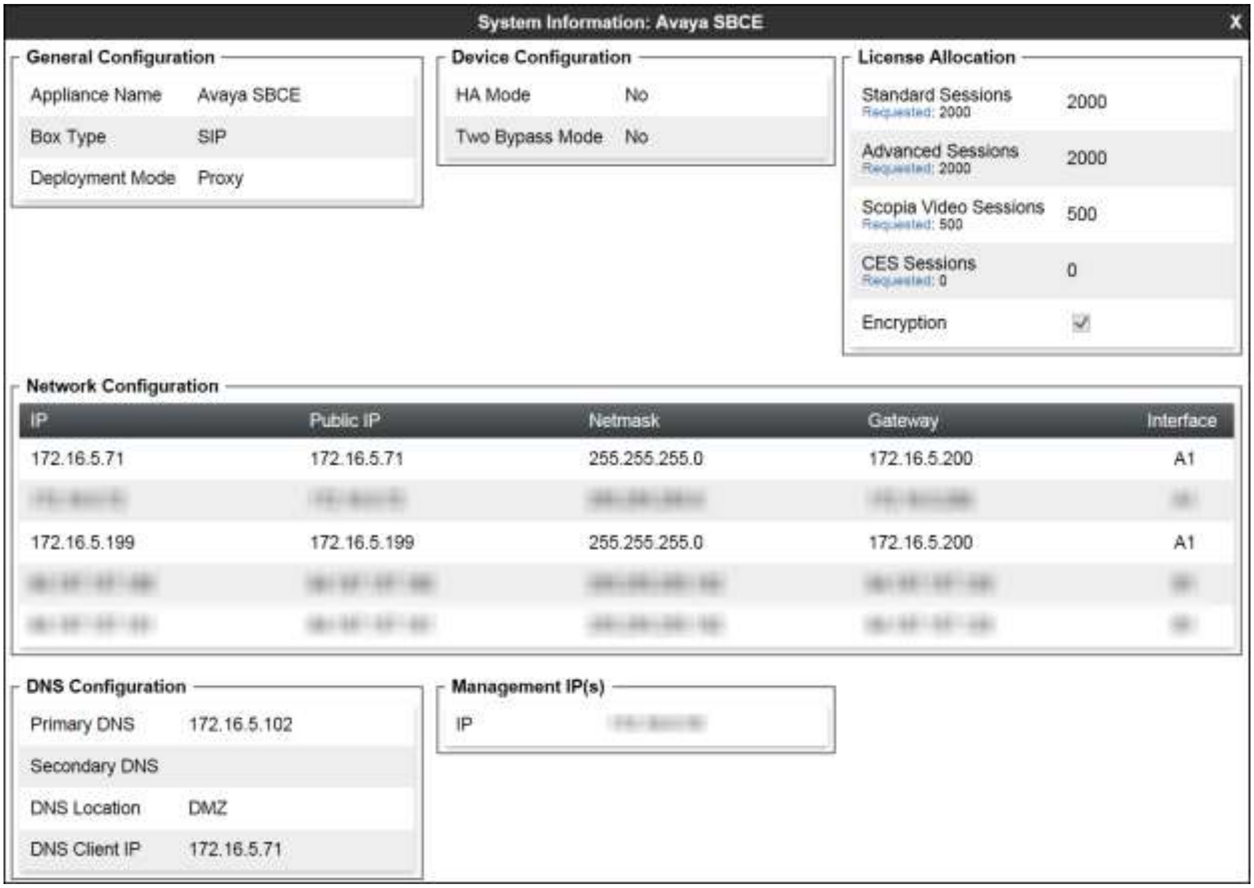


To view the system information that has been configured during installation, navigate to **System Management**. A list of installed devices is shown in the right pane. In the compliance testing, a single Device Name **Avaya SBCE** was already added.



To view the network configuration assigned to the Avaya SBCE, click **View** as shown on the previous screen. 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**. The **Box Type** was set to ***SIP*** and the **Deployment Mode** was set to ***Proxy***. Default values were used for all other fields.



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. Since a VPN connection was used with this solution to connect Claro's network to the enterprise network, the **A1** interface was used for access to the private enterprise network and to route calls to Claro's network across the VPN tunnel. In this solution, the **B1** interface was used for remote workers access to the enterprise. The configuration required for the **B1** interface is not discussed in this document. Refer to **Figure 1** for the IP addresses for both the A1 and B1 interfaces on the Avaya SBCE.

When a VPN connection is not used, the **B1** interface is normally used for remote workers access to the private network as well as to route calls to the Service Provider across the public Internet.

The management IP was blurred out for security reasons. The IP addresses used for the remote worker configuration was also blurred out since the remote worker configuration is beyond the scope of these Application Notes and is not discussed in these Application Notes.

IMPORTANT! – During the Avaya SBCE installation, the Management interface, (labeled "M1"), of the Avaya SBCE must be provisioned on a different subnet than either of the Avaya SBCE private and public network interfaces (e.g., A1 and B1). If this is not the case, contact your Avaya representative to have this resolved.

7.2. Global Profiles

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

7.2.1. 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 on the next page. 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.

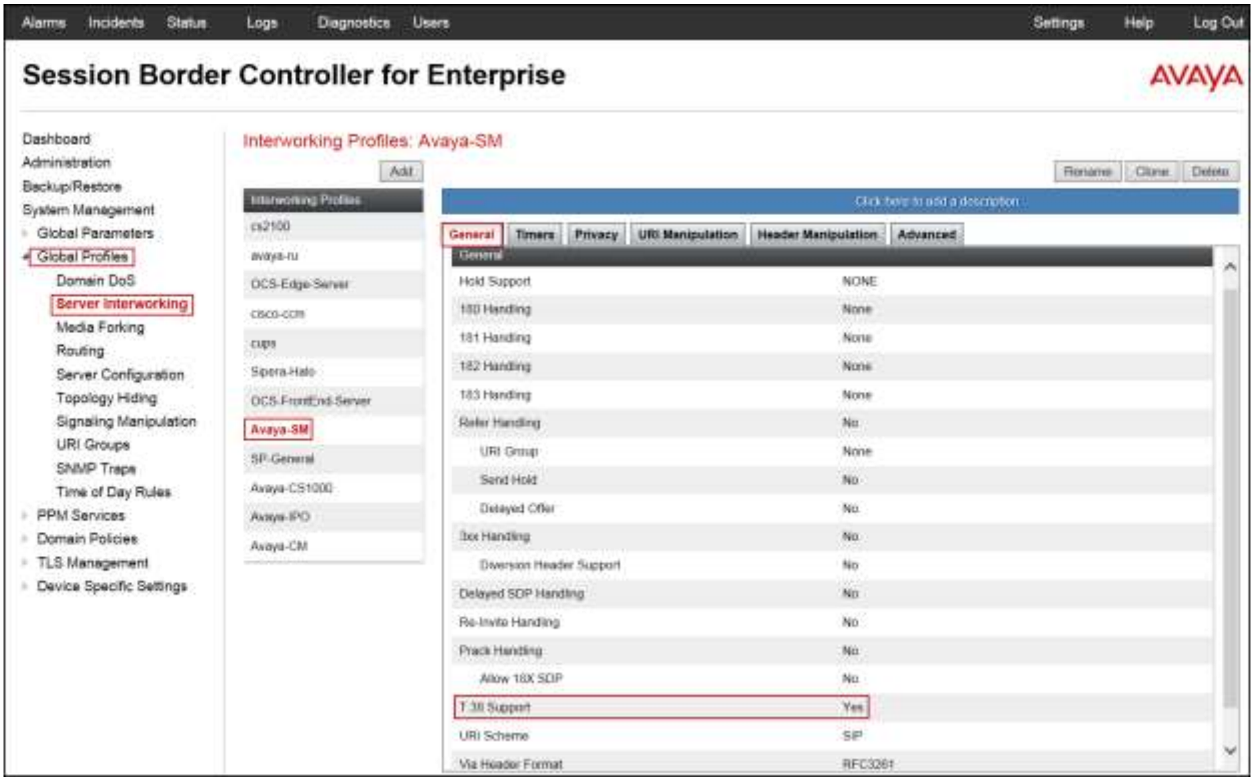
On the left navigation pane, select **Global Profiles → Server Interworking**. From the **Interworking Profiles** list, select **avaya-ru**. Click **Clone** on top right of the screen.

Enter the new profile name in the **Clone Name** field, the name of ***Avaya-SM*** was chosen in this example. Click **Finish** (nor shown).

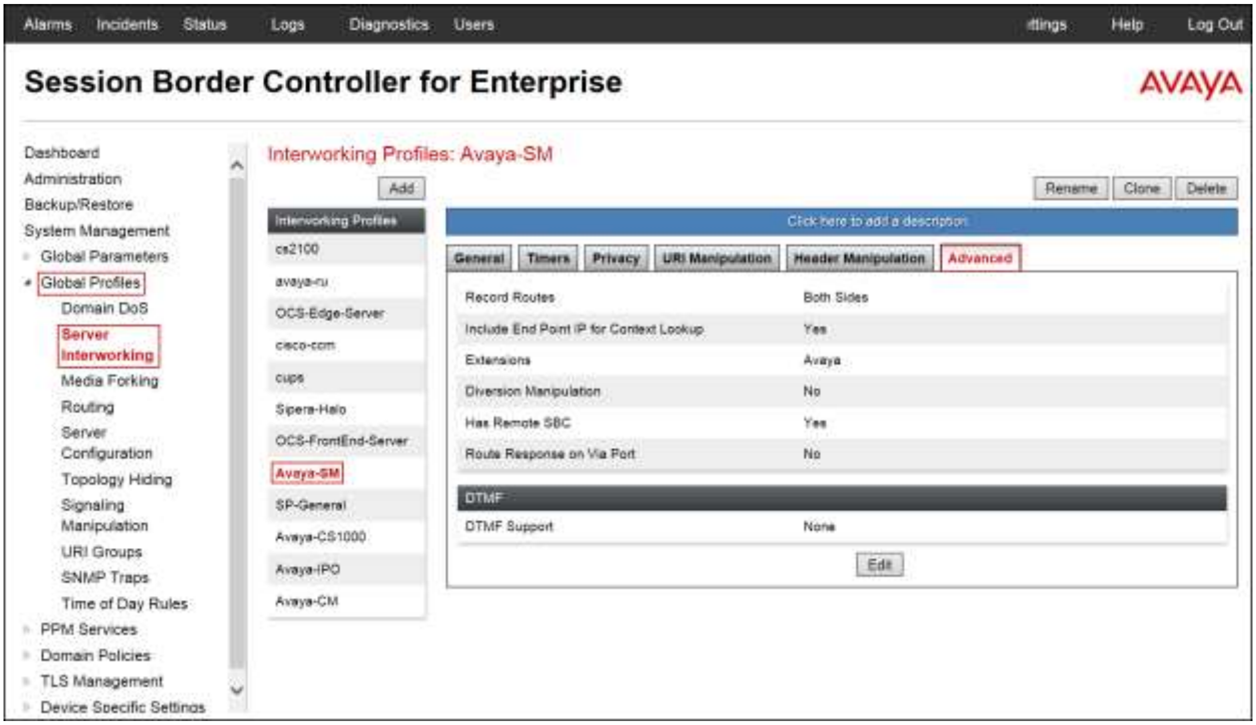
For the newly created **Avaya-SM** profile, click **Edit** (not shown) at the bottom of the **General** tab:

- Check ***T.38 Support***.
- Leave other fields with their default values.
- Click **Finish** in the **Editing Profile** window.

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.



7.2.2. Server Interworking SP-General

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

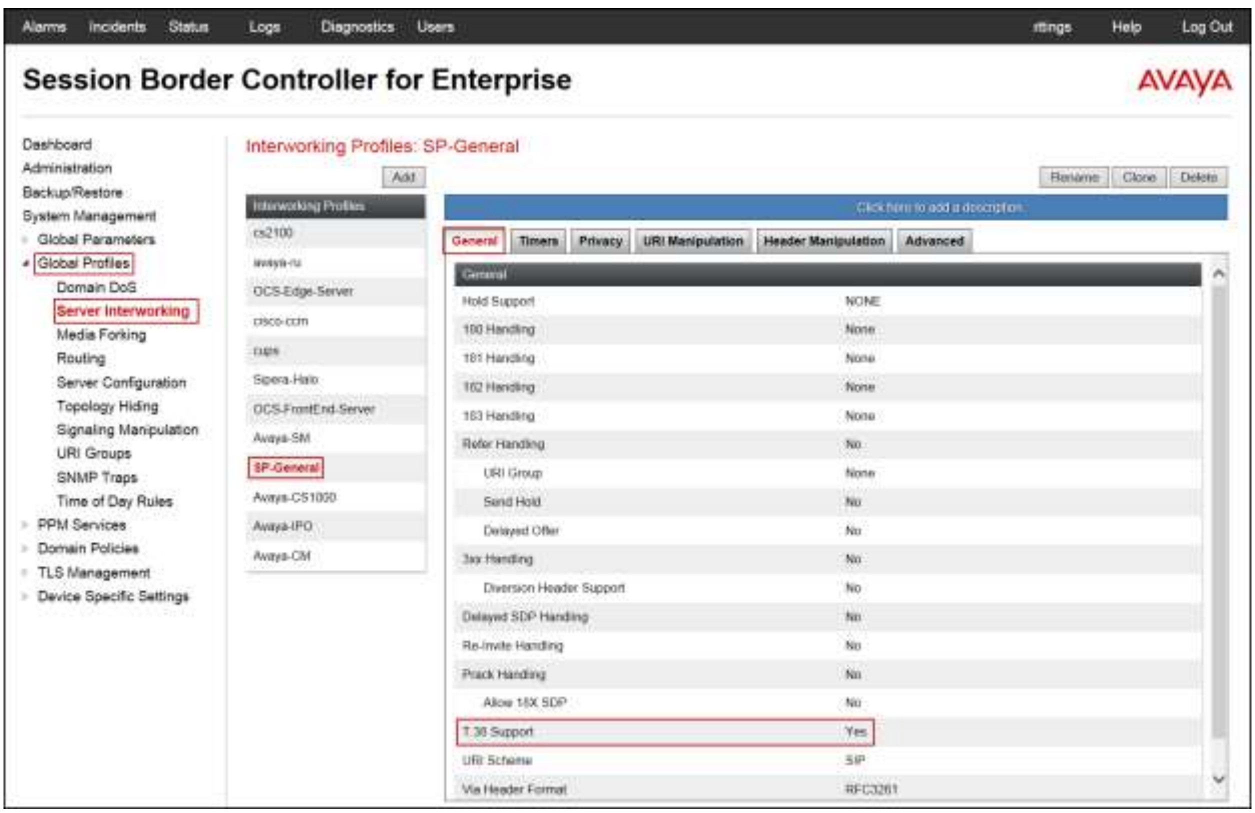
On the left navigation pane, select **Global Profiles → Server Interworking**. From the **Interworking Profiles** list, select **Add**.

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

On the **General** tab:

- Check ***T.38 Support***.
- Leave 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.

7.2.3. 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 **[6]** 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**):

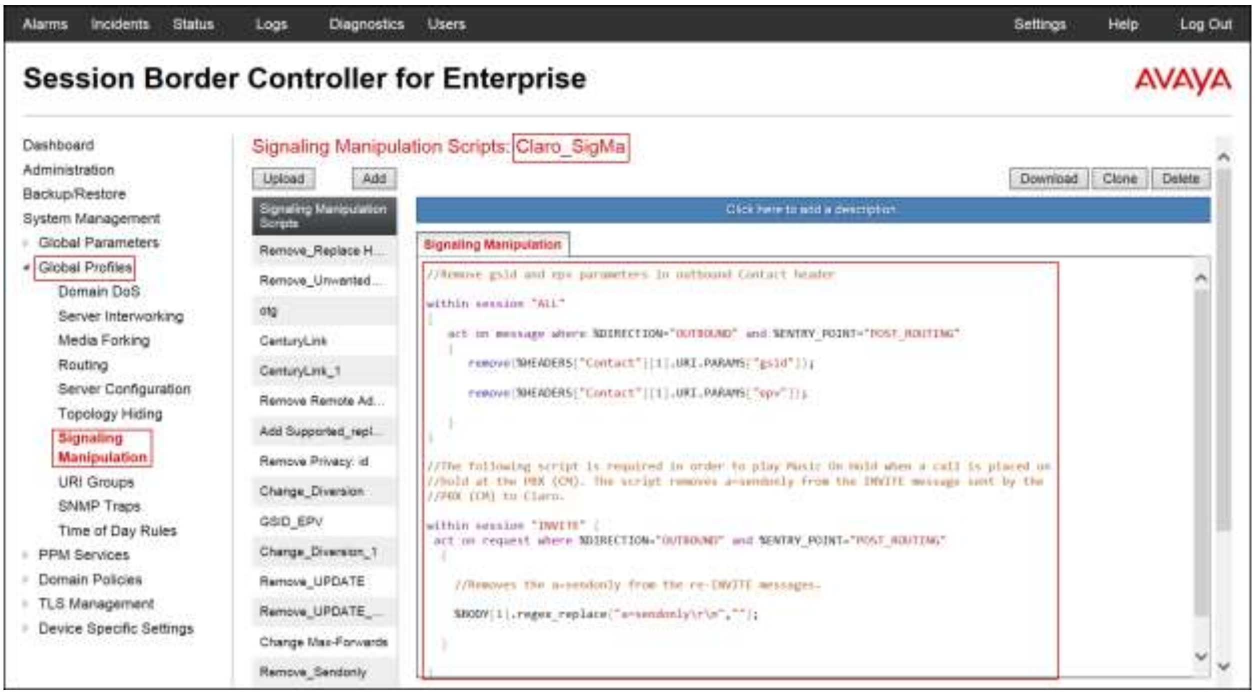
- **Remove unwanted headers**: Remove the “gsid” and “epv” parameters from outbound “Contact” headers. These parameters have no significance to the service provider and add unnecessary size to the outbound messages.
- **Music on hold**: When calls from/to the PSTN were placed on-hold by Communication Manager users, the PSTN users did not hear Music while on-hold. A SigMa script was created to remove the “sendonly**”** message Communication Manager includes in the SDP of re-INVITEs when calls from/to the PSTN are placed on-hold, this allowed the PSTN users to hear Music while on-hold. The script was latter applied to the Service

Provider side of the server configuration profile (refer to **Section 7.2.4**). The script was included under the **Claro_Sigma** script shown below.

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

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



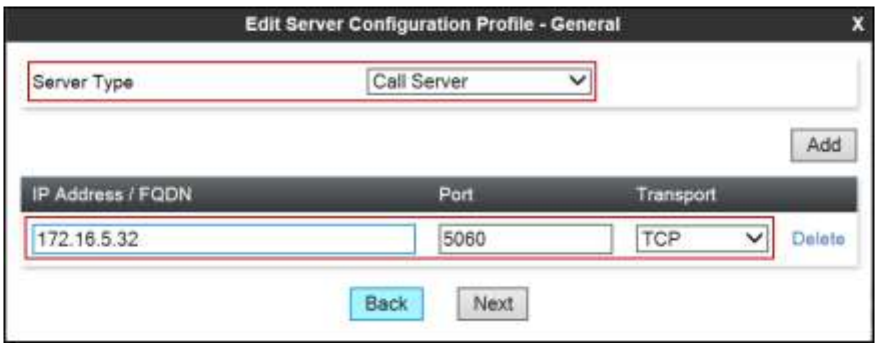
7.2.4. Server Configuration

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***.

In the **Add Server Configuration Profile** window:

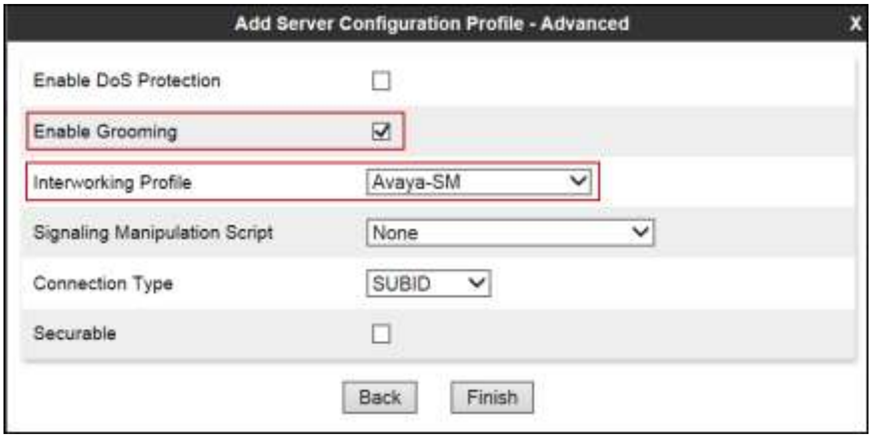
- **Server Type**: select ***Call Server***.
- **IP Address / FQDN:** ***172.16.5.32*** (IP Address of the Session Manager SIP entity).
- **Port:** ***5060*** (This port must match the port number defined in **Section 6.6**).
- **Transports:** Select ***TCP***.
- Click **Next**.



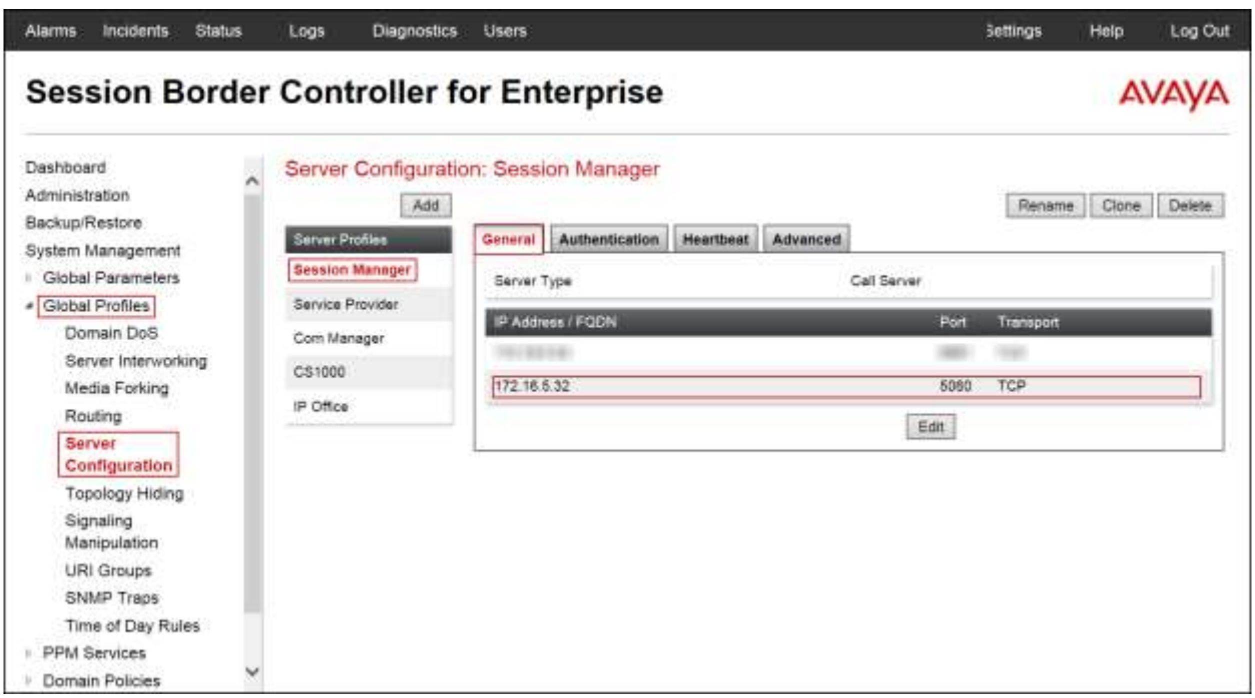
- Click **Next** in the **Add Server Configuration Profile** - **Authentication** window (not shown).
- Click **Next** in the **Add Server Configuration Profile - Heartbeat** window (not shown).

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

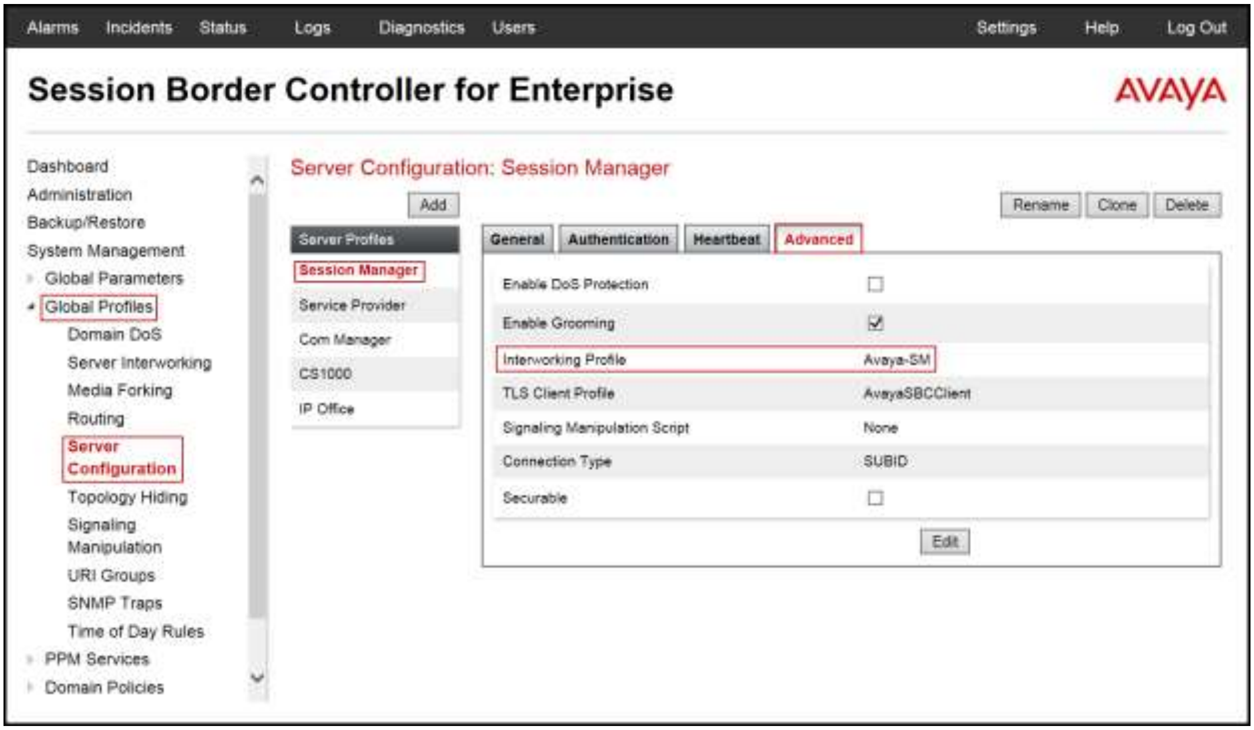
- Check ***Enable Grooming***.
- Select ***Avaya-SM*** from the **Interworking Profile** drop down menu.
- Leave the **Signaling Manipulation Script** at the default ***None***.
- Click **Finish**.



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



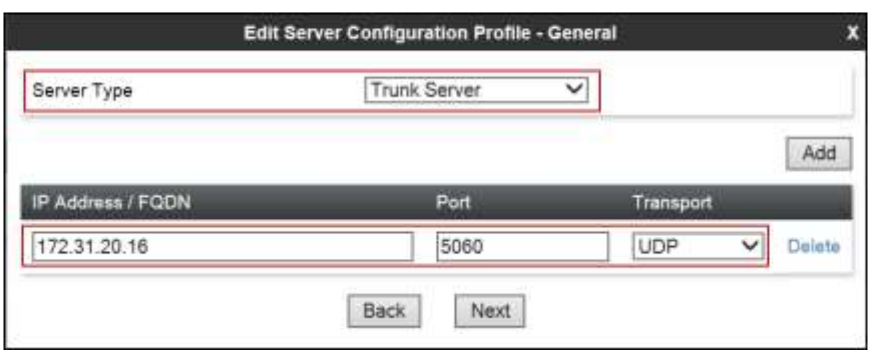
The following screen capture shows the **Advanced** tab of the newly created **Session Manager** Server Profile.



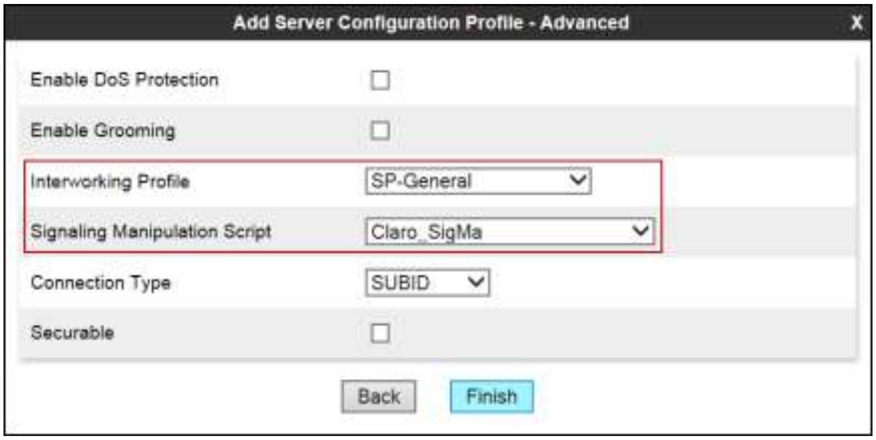
To add the profile for the Trunk Server, from the **Server Configuration** screen, click **Add** in the **Server Profiles** section and enter the profile name: ***Service Provider***.

In the **Add Server Configuration Profile** window

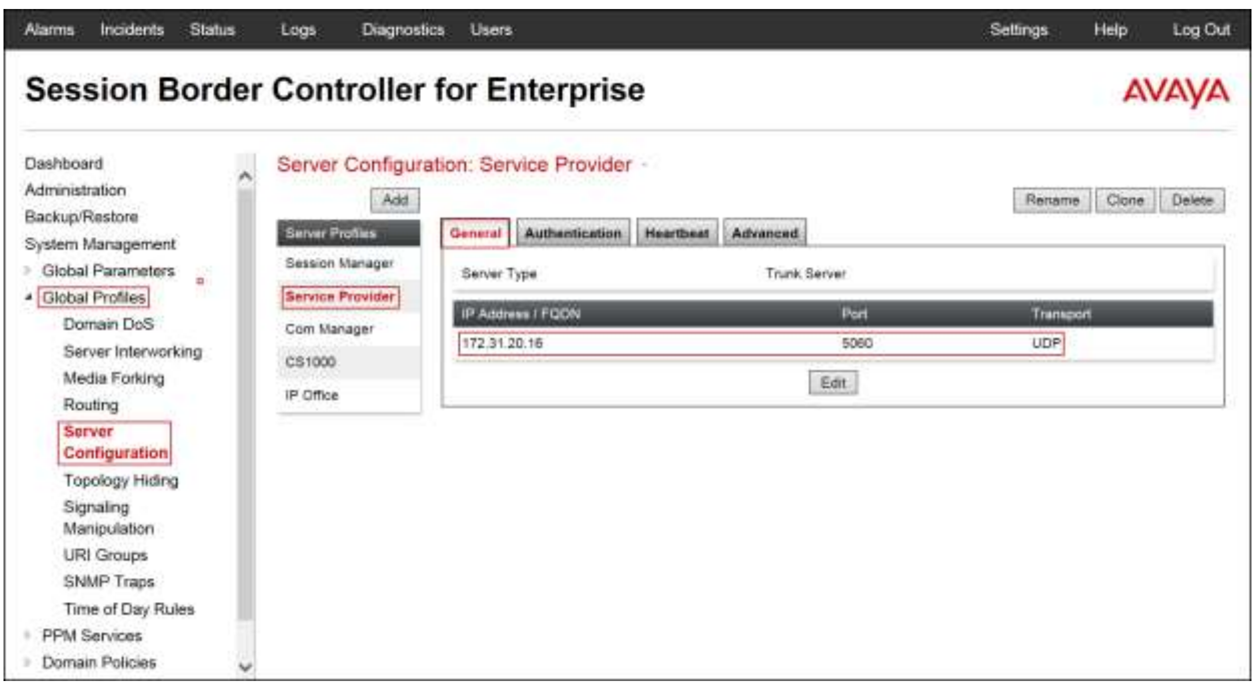
- **Server Type:** select ***Trunk Server***.
- **IP Address/FQDN:** ***172.31.20.16*** (Service Provider's SIP Proxy IP address).
- **Port:** ***5060***.
- **Transports**: Select ***UDP***.
- Click **Next**.



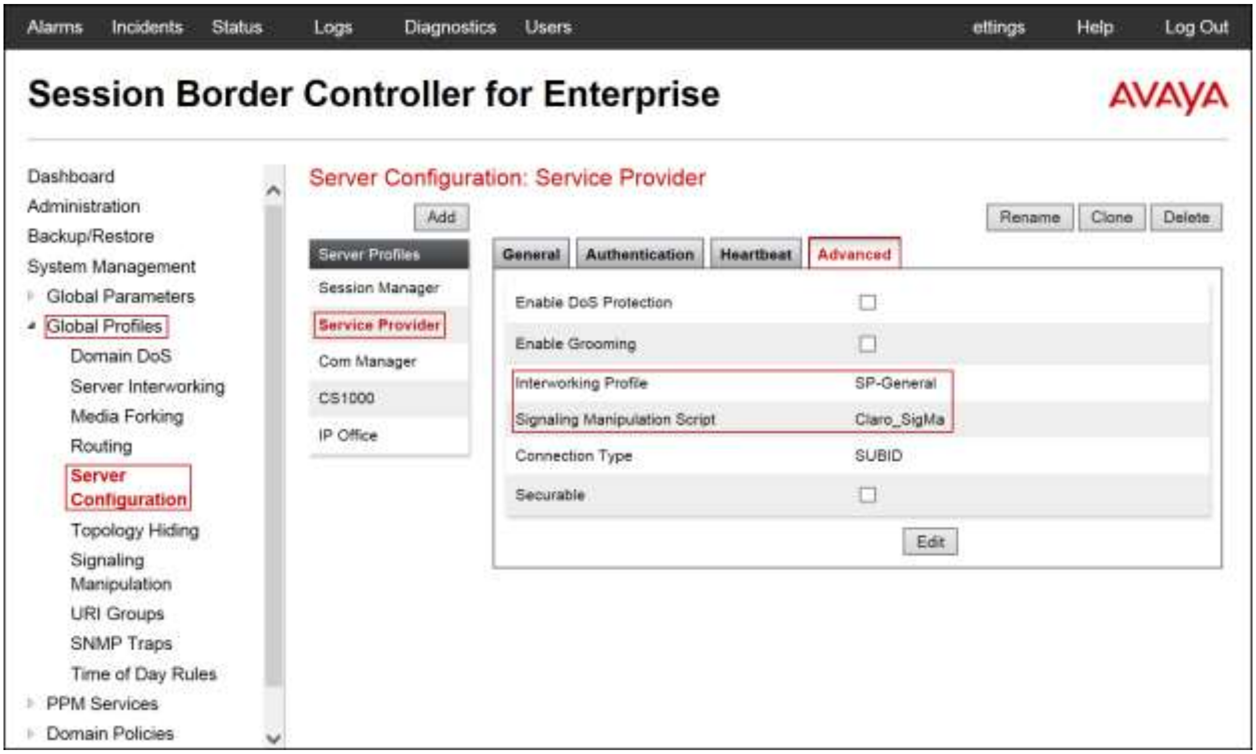
- Click **Next** in the **Add Server Configuration Profile - Authentication** window (not shown).
- Click **Next** in the **Add Server Configuration Profile - Heartbeat** window (not shown). In the **Add Server Configuration Profile - Advanced** window:
- Select ***SP-General*** from the **Interworking Profile**.
- Select ***Claro_SigMa*** from the **Signaling Manipulation Script**, script created in **Section 7.2.3**.
- Click **Finish**.



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



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



7.2.5. 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.

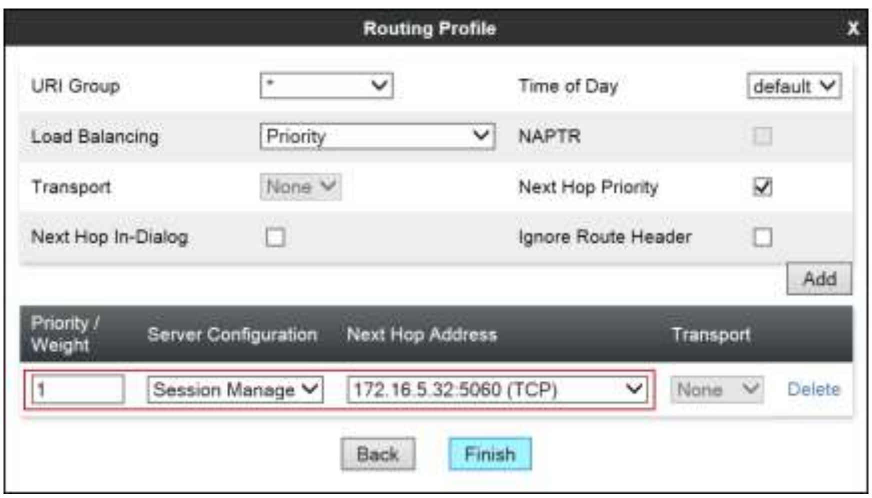
Two Routing profiles were created; one for inbound calls, with Session Manager as the destination, and the second one for outbound calls, which are sent to the service provider.

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

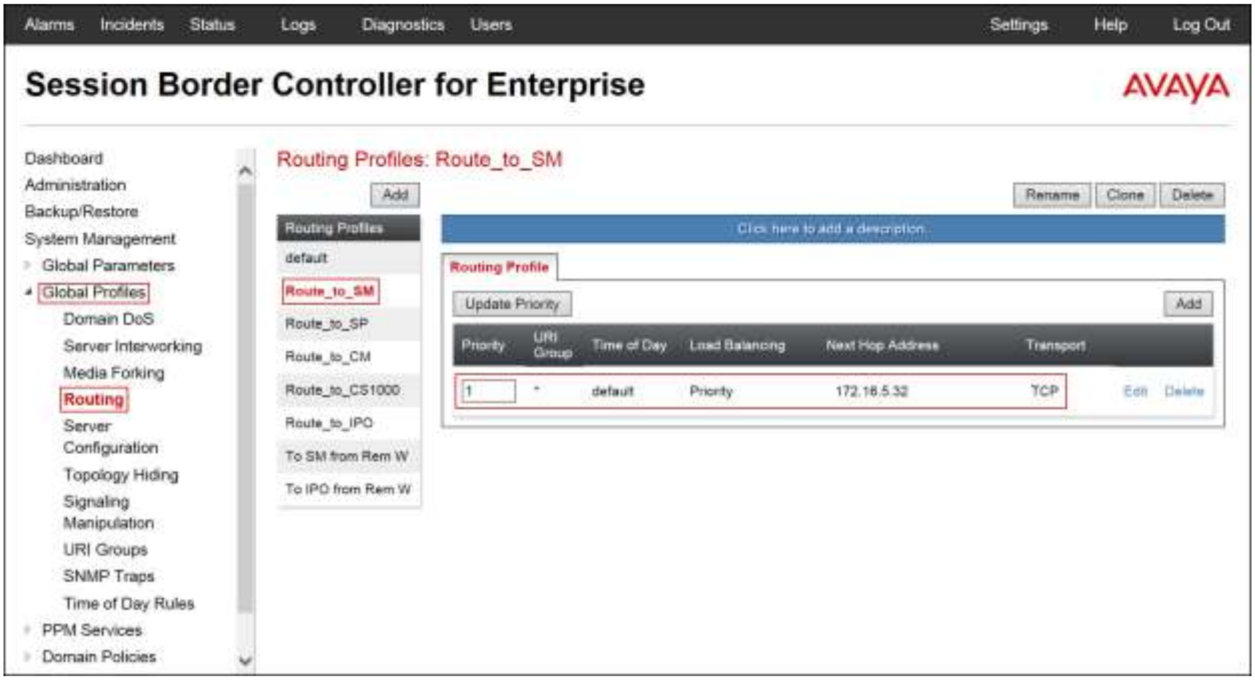
- Select **Routing**.
- Click **Add** in the **Routing Profiles** section**.**
- Enter Profile Name: ***Route_to_SM***.
- Click **Next**.

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 ***172.16.5.32:5060 (TCP)*** (Session Manager IP address, Port and Transport).
- Click **Finish**.



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

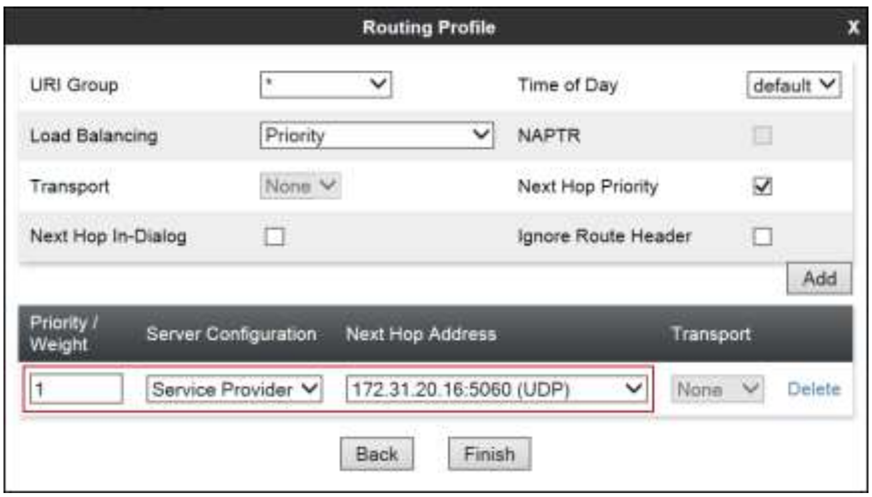


Similarly, for the outbound route:

- Select **Routing**.
- Click **Add** in the **Routing Profiles** section.
- Enter Profile Name: ***Route_to_SP***.
- 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 ***Service Provider***.
- The **Next Hop Address** is populated automatically with ***172.31.20.16:5060 (UDP)*** (Service Provider SIP Proxy IP address, Port and Transport).
- Click **Finish**.



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

7.2.6. 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.

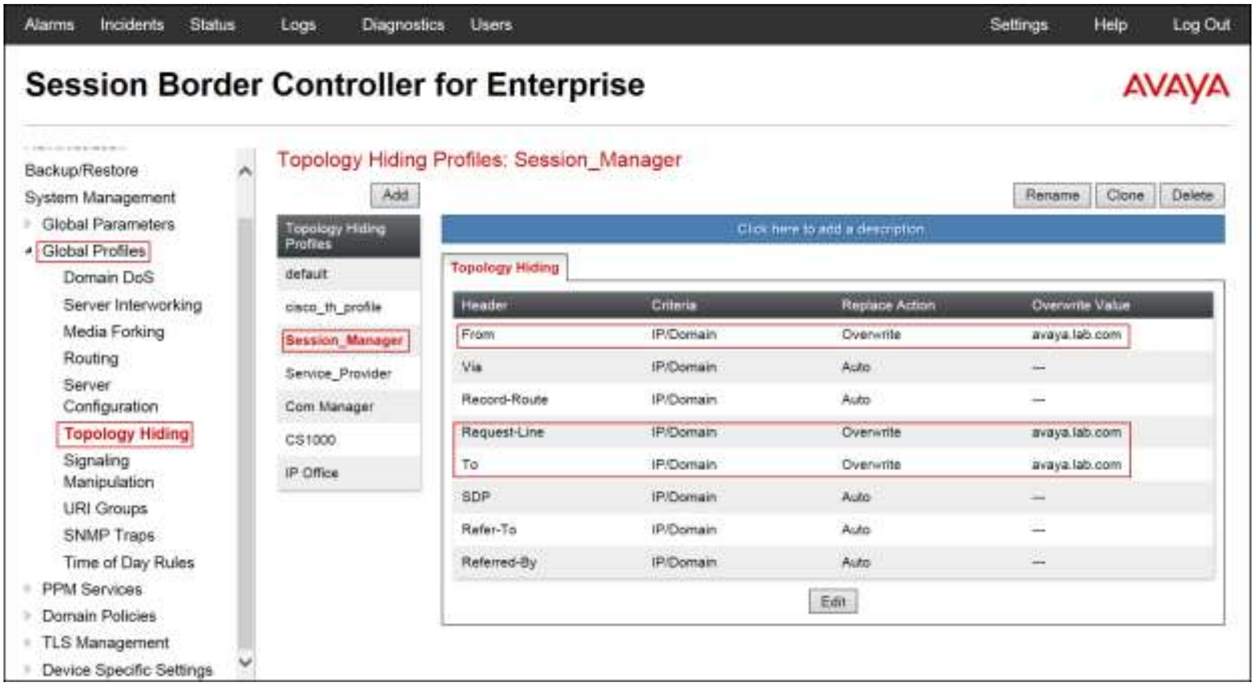
Topology Hiding can also be used as an interoperability tool to adapt the host portion in SIP headers like To, From, Request-URI, Via, Record-Route and SDP to the IP addresses or domains expected by Session Manager and the SIP trunk Service Provider, allowing the call to be accepted in each case.

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:

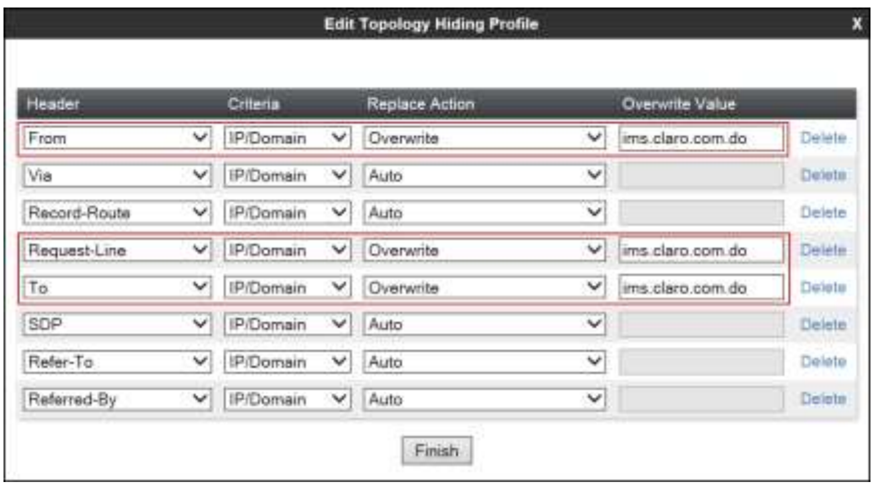
- Select the **default** profile in the **Topology Hiding Profiles** list, then click **Clone** on top right of the screen.
- Enter the **Profile Name**: ***Session_Manager***.
- Click **Finish**.
- Click **Edit** on the newly added **Session_Manager** Topology Hiding profile.
- For **Request-Line** 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**.
- 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 **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**.

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

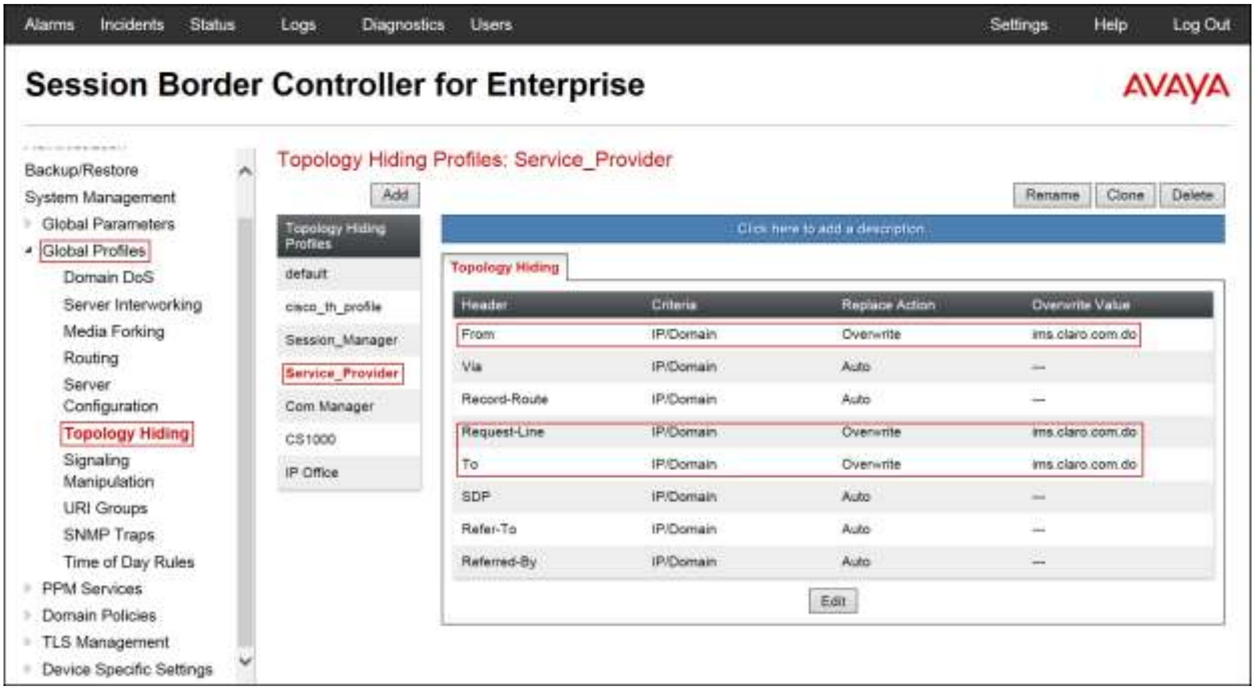


To add the Topology Hiding profile in the service provider direction, select **Topology Hiding** from the **Global Profiles** menu on the left-hand side:

- 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 **Request-Line** under **Header**, choose ***Overwrite*** from the pull-down menu under **Replace Action;** enter the domain name for the service provider **(*ims.claro.com.do*)** 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 **(*ims.claro.com.do*)** under **Overwrite Value**.
- For **To** under **Header**, choose ***Overwrite*** from the pull-down menu under **Replace Action,** enter the domain name for the service provider **(*ims.claro.com.do*)** under **Overwrite Value**.



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



7.3. Domain Policies

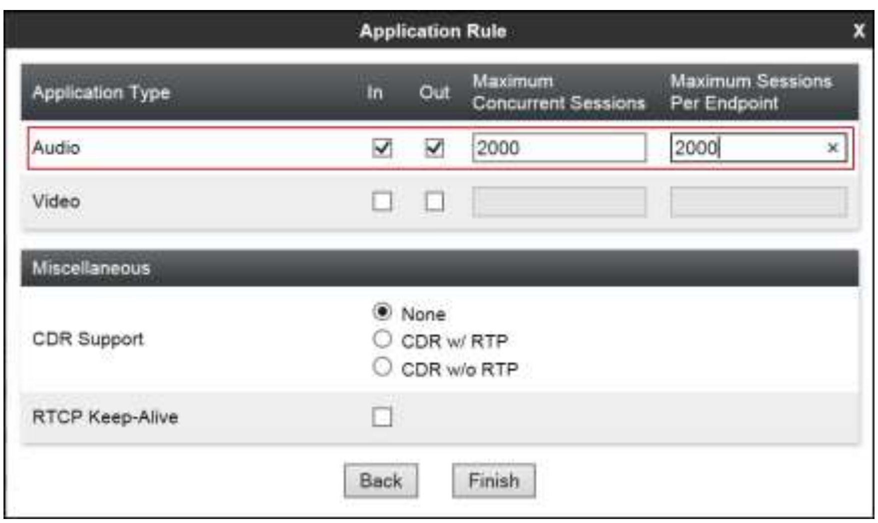
Domain Policies allow the configuration of sets of rules designed to control and normalize the behavior of call flows, based upon various criteria of communication sessions originating from or terminating in the enterprise. Domain Policies include rules for Application, Media, Signaling, Security, etc.

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.

7.3.1. 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**.



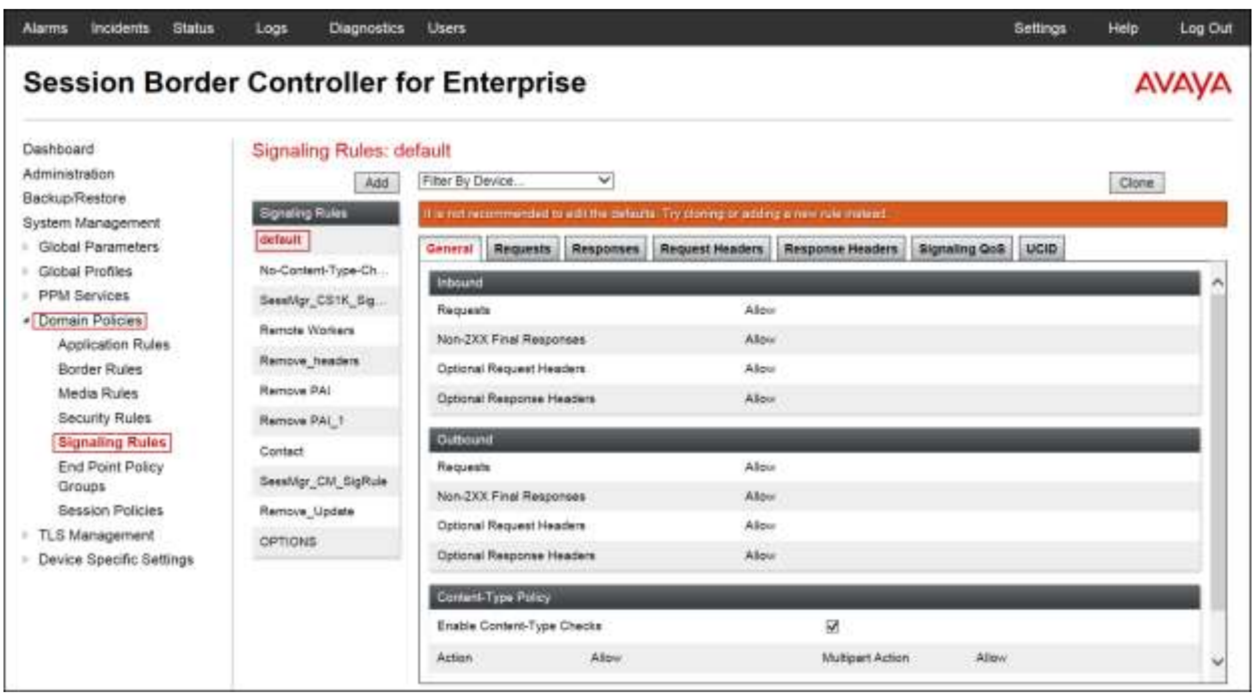
The following screen capture shows the newly created **2000 Sessions** Application Rule.

7.3.2. Media Rules

For the compliance test, the **default-low-med** Media Rule was used.

7.3.3. Signaling Rules

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

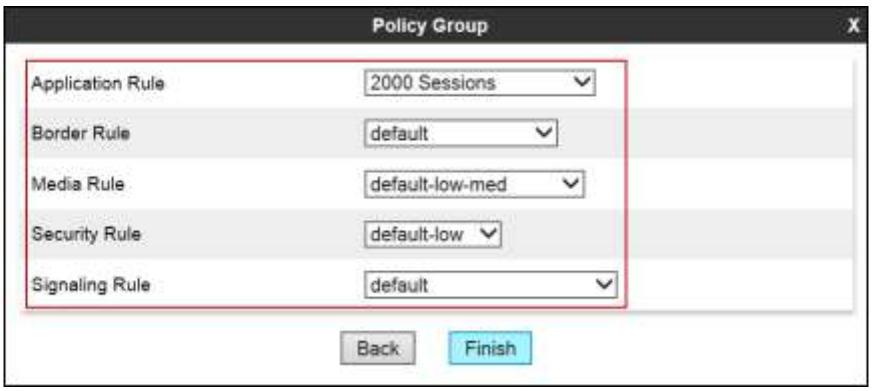


7.3.4. End Point Policy Groups

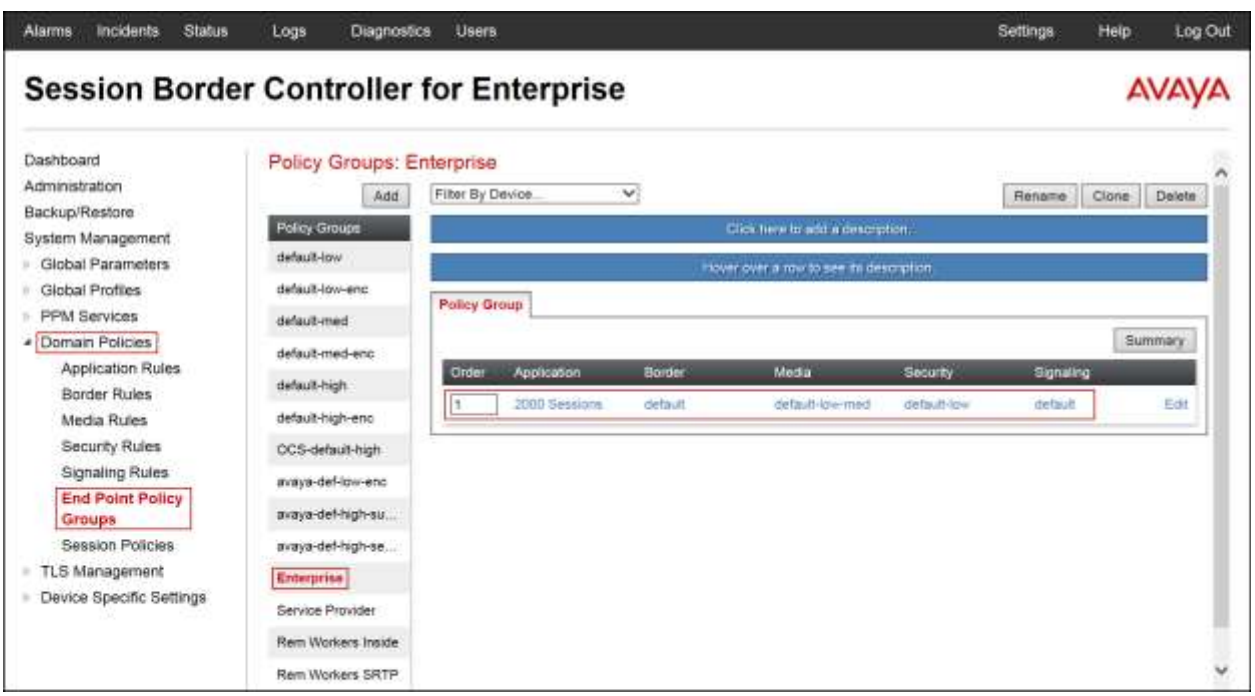
End Point Policy Groups are associations of different sets of rules (Media, Signaling, Security, etc.) to be applied to specific SIP messages traversing through the Avaya SBCE.

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 ***Enterprise***.

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

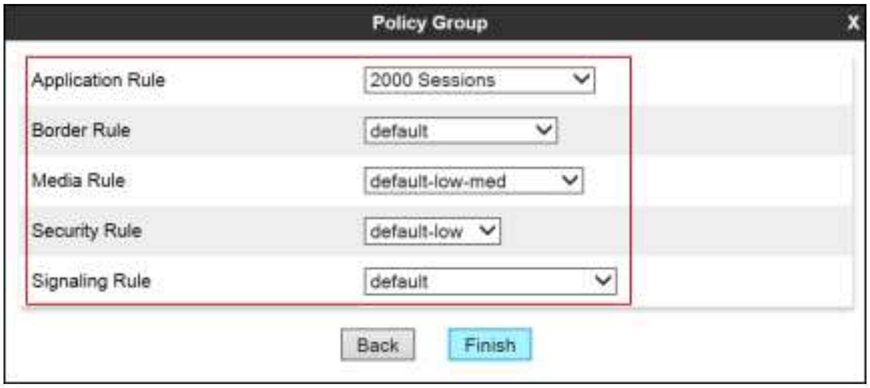


The following screen capture shows the newly created **Enterprise** End Point Policy Group.

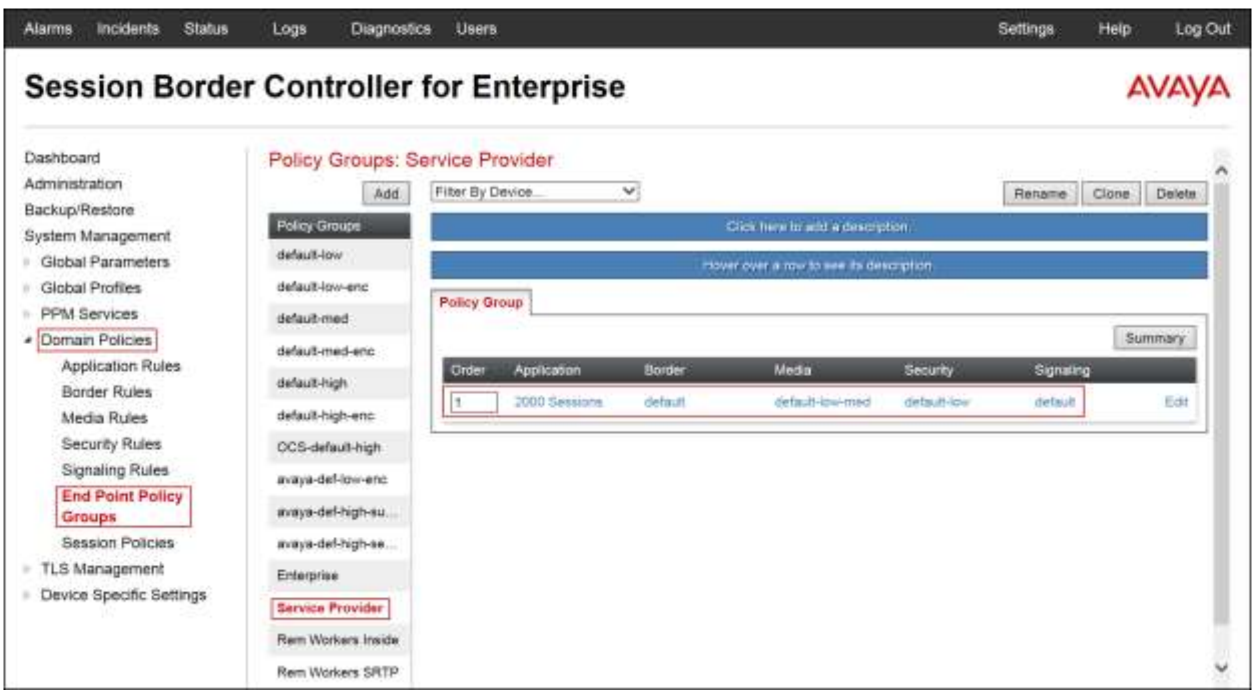


Similarly, to create an End Point Policy Group for the Service Provider SIP Trunk, select **Add Group**, under **Group Name** enter ***Service Provider***.

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



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



7.4. 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.

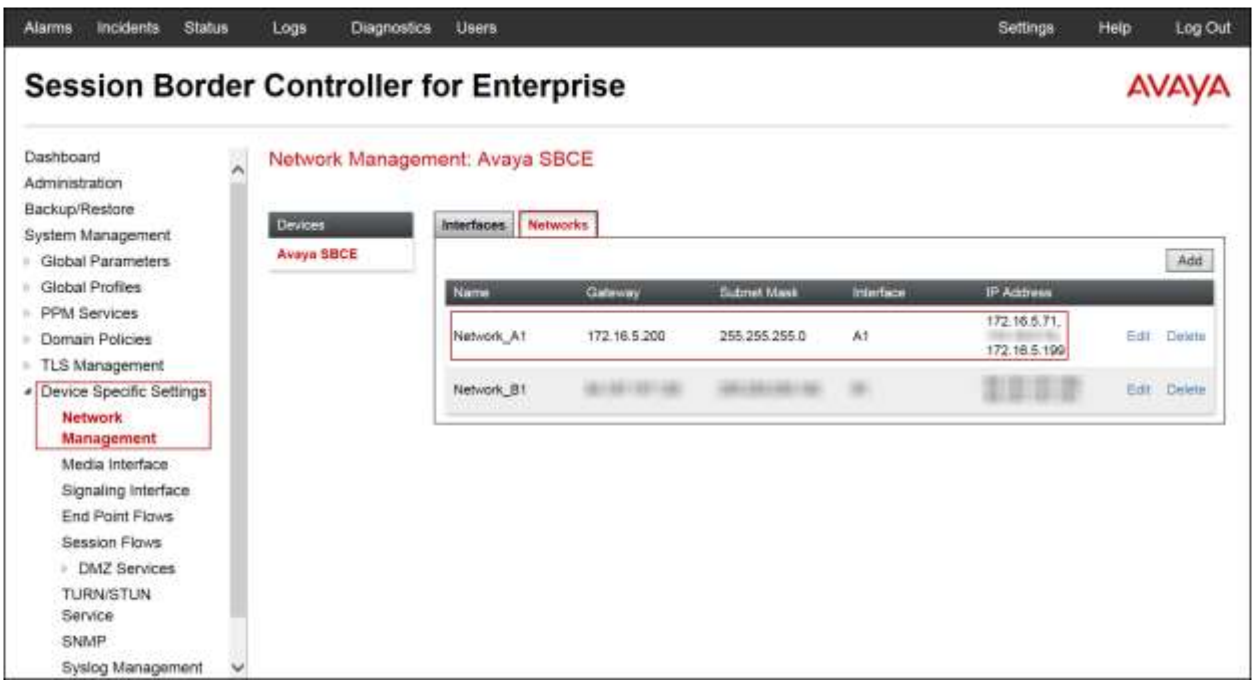
7.4.1. Network Management

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.

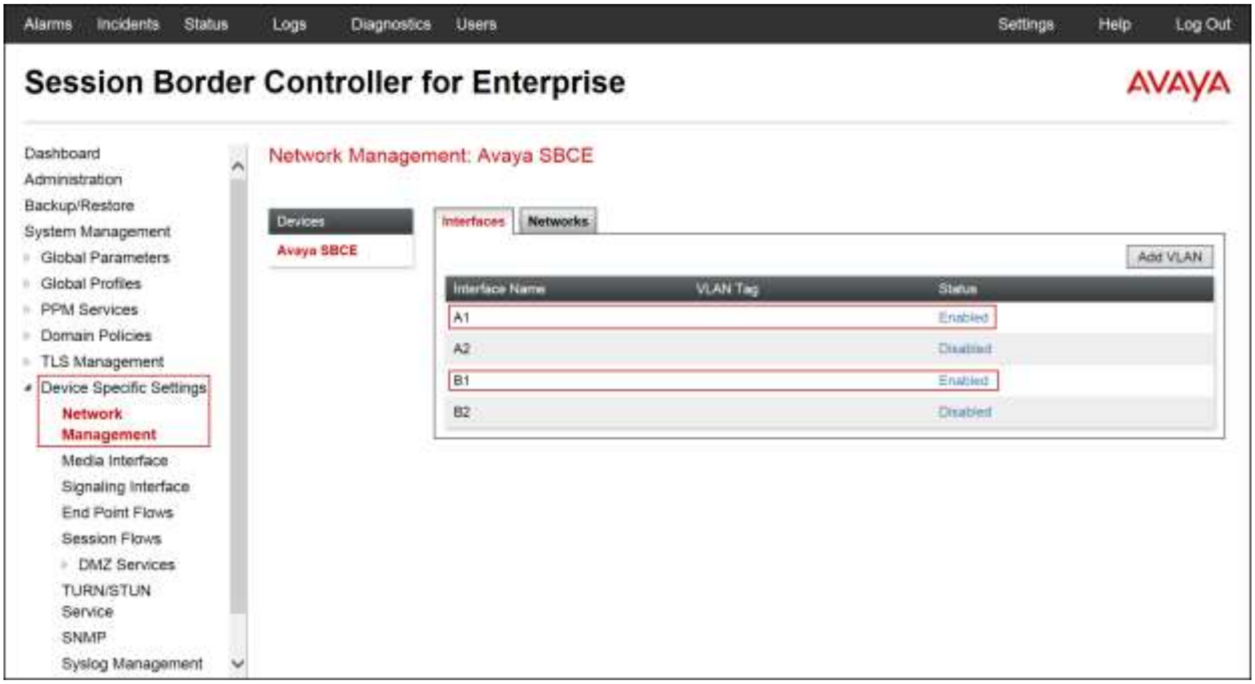
Use **Figure 1** as reference for IP address assignments.

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 this Application Notes.



Note: In cases where the private interface (A1) of the Avaya SBCE is on a different IP subnet as Session Manager or the VPN gateway, it will be required to configure "Gateway Override" under **Device Specific Settings → Network Management → Networks** in order to separate and route the traffic to the correct destinations (Session Manager or the VPN gateway). In the sample configuration shown, the private interface (A1) of the Avaya SBCE was in the same IP subnet as Session Manager and the VPN gateway, thus configuration of "Gateway Override" was not required.

On the Interface Configuration tab, click the **Toggle** control 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.

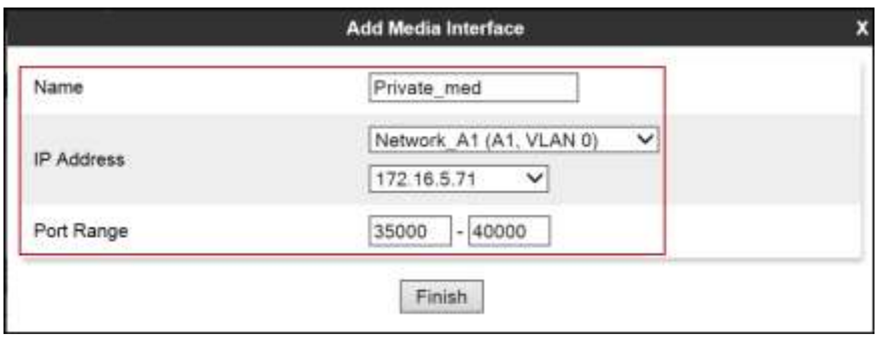


7.4.2. Media Interface

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.

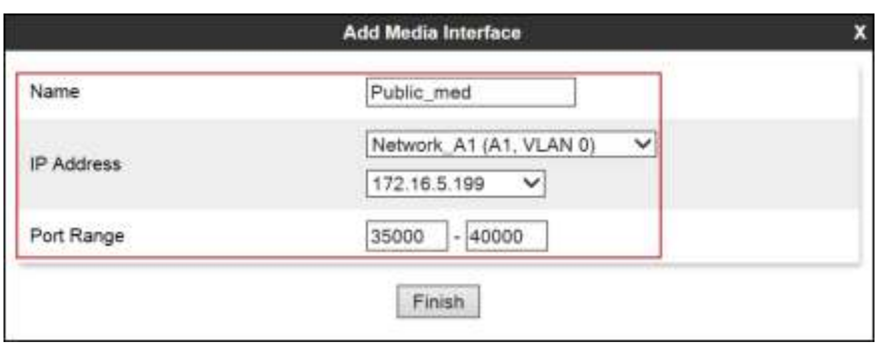
From the **Device Specific Settings** menu on the left-hand side, select **Media Interface**. Below is the configuration of the inside, private Media Interface of the Avaya SBCE.

- 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*** (Inside 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_A1 (A1, VLAN 0)**
  Select **IP Address**: ***172.16.5.199*** (IP Address of the Avaya SBCE toward the Service Provider via the VPN Tunnel).
- **Port Range:** ***35000-40000***.
- Click **Finish**.



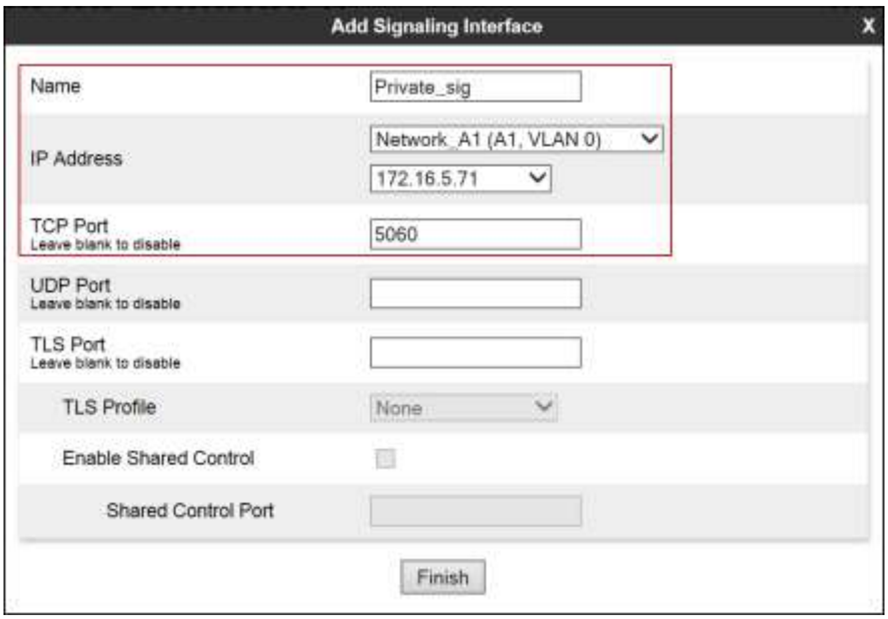
The following screen capture shows the newly created **Media Interfaces**.

7.4.3. Signaling Interface

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

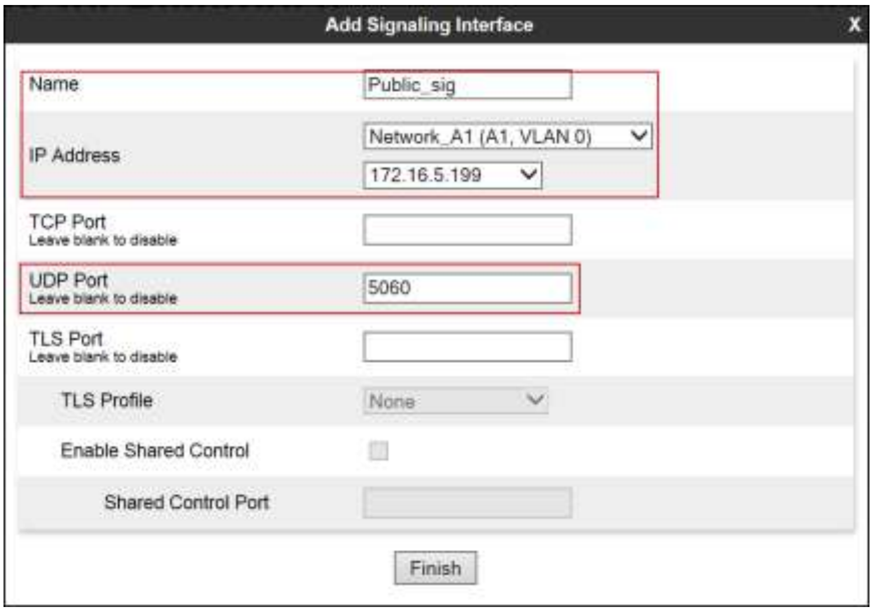
Below is the configuration of the inside private Signaling Interface of the Avaya SBCE.

- 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).
- **TCP Port**: ***5060***
- Click **Finish**.

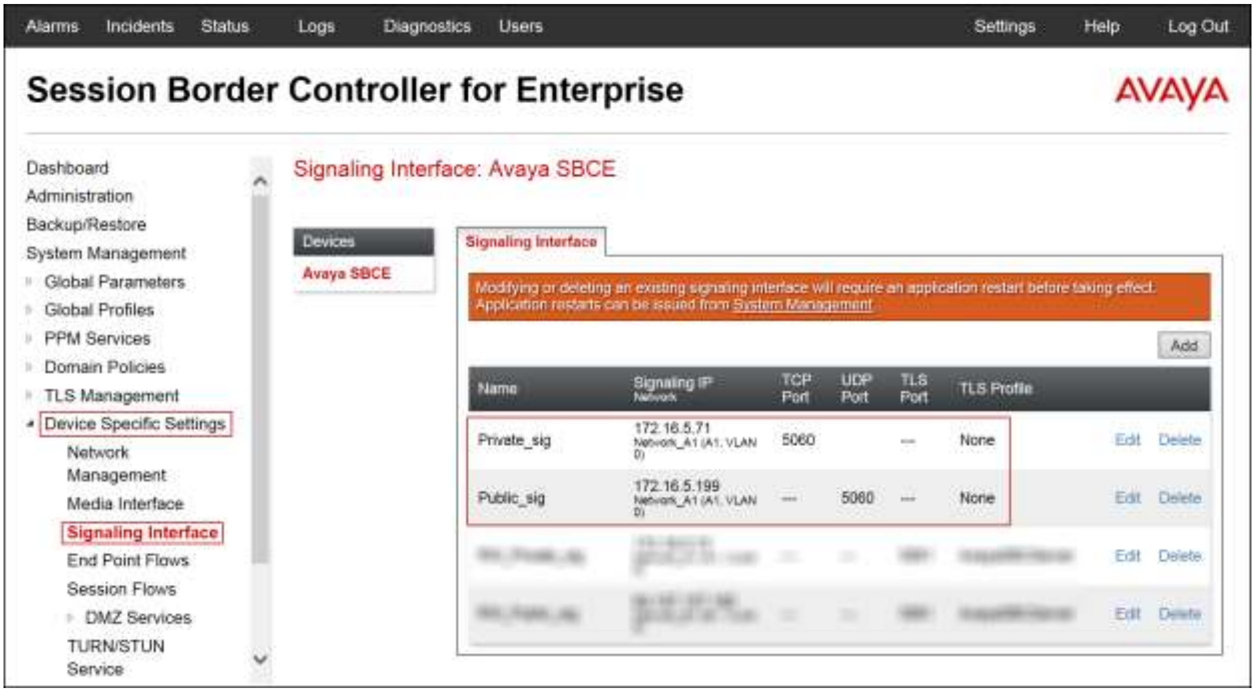


Below is the configuration of the outside, public signaling Interface of the Avaya SBCE.

- Select **Add** in the **Signaling Interface** area.
- **Name:** ***Public_sig***.
- Under **IP Address** select: **Network_A1 (A1, VLAN 0)**
- Select **IP Address**: ***172.16.5.199*** (IP Address of the Avaya SBCE toward the Service Provider via the VPN Tunnel).
- **UDP Port:** ***5060***.
- Click **Finish**.

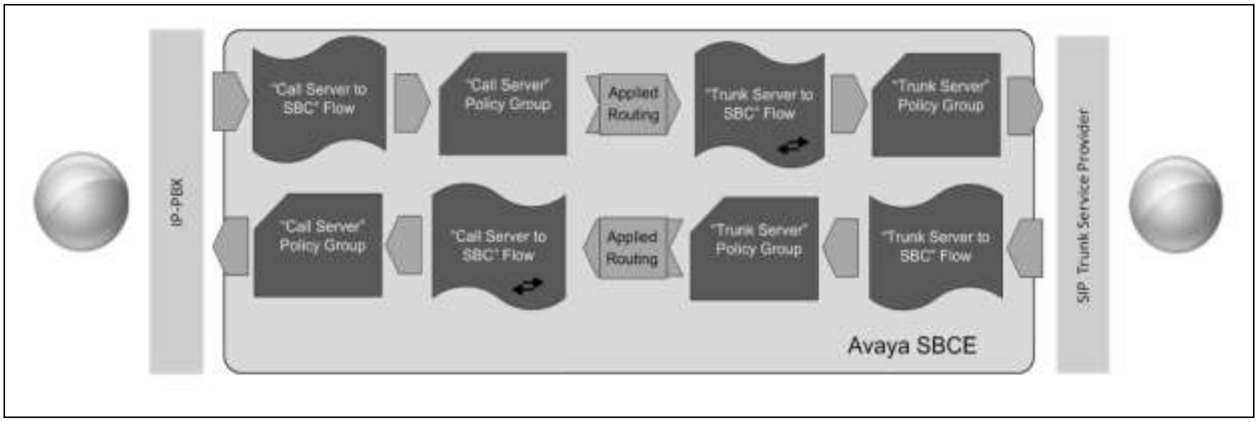


The following screen capture shows the newly created **Signaling Interfaces**.



7.4.4. End Point Flows

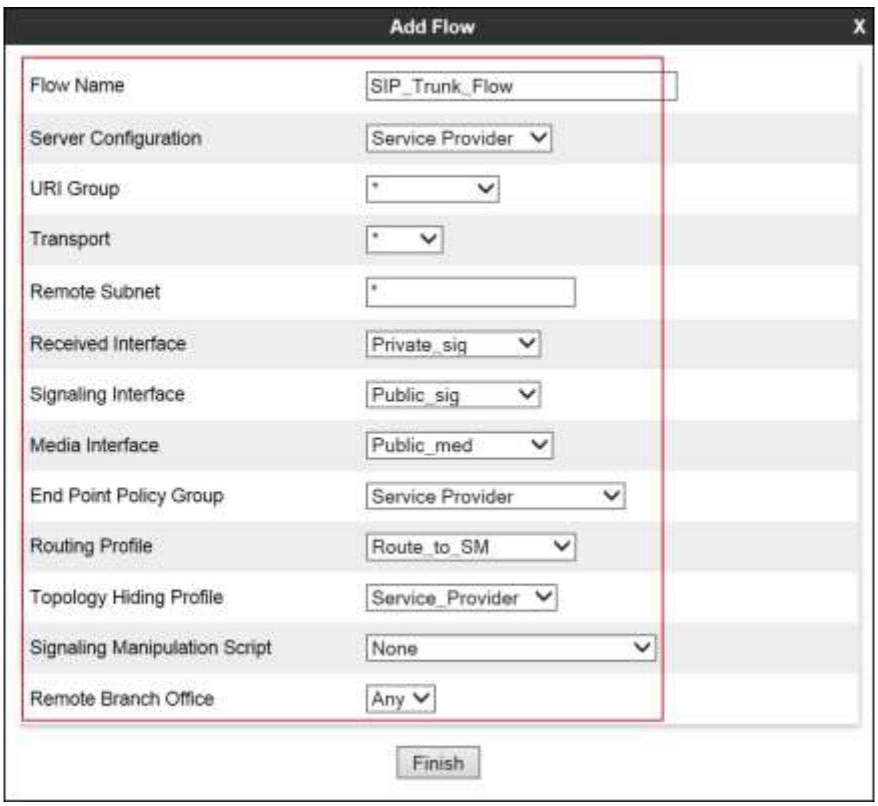
When a packet is received by Avaya SBCE, the content of the packet (IP addresses, URIs, etc.) is used to determine which flow it matches. Once the flow is determined, the flow points to a policy group which contains several rules concerning processing, privileges, authentication, routing, etc. Once routing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

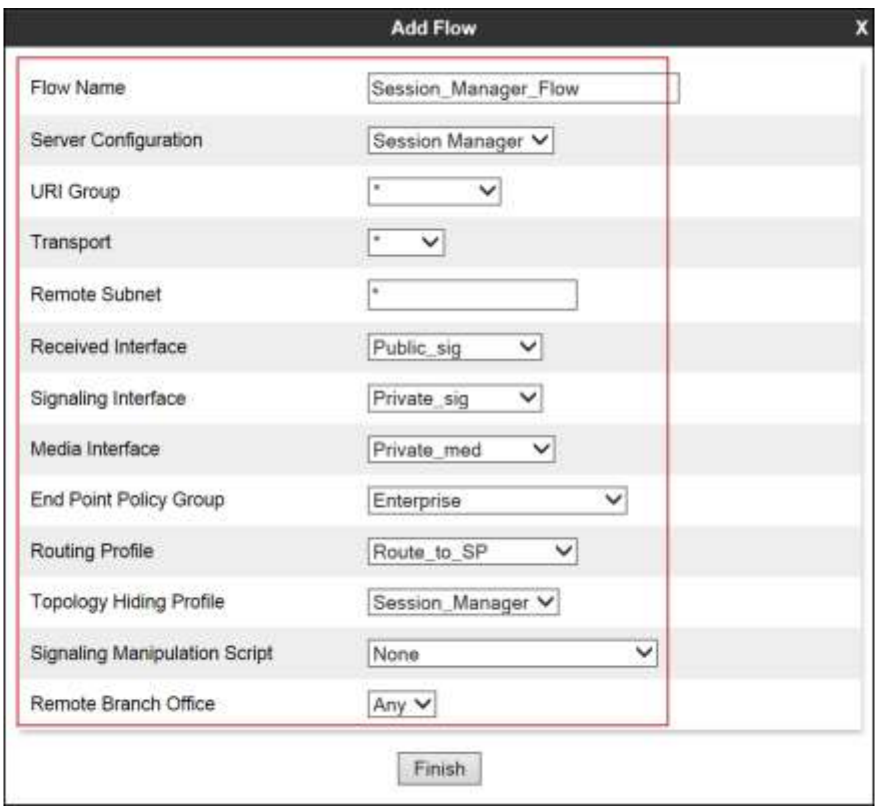
To create the call flow toward the Service Provider SIP trunk, from the **Device Specific Settings** menu, select **End Point Flows**, and then the **Server Flows** tab. Click **Add** (not shown).

- **Flow Name: *SIP_Trunk_Flow***.
- **Server Configuration**: ***Service Provider***.
- **URI Group: ***
- **Transport: ***
- **Remote Subnet: ***
- **Received Interface**: ***Private_sig***.
- **Signaling Interface: *Public_sig***.
- **Media Interface**: ***Public_med***.
- **End Point Policy Group: *Service Provider***.
- **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**.

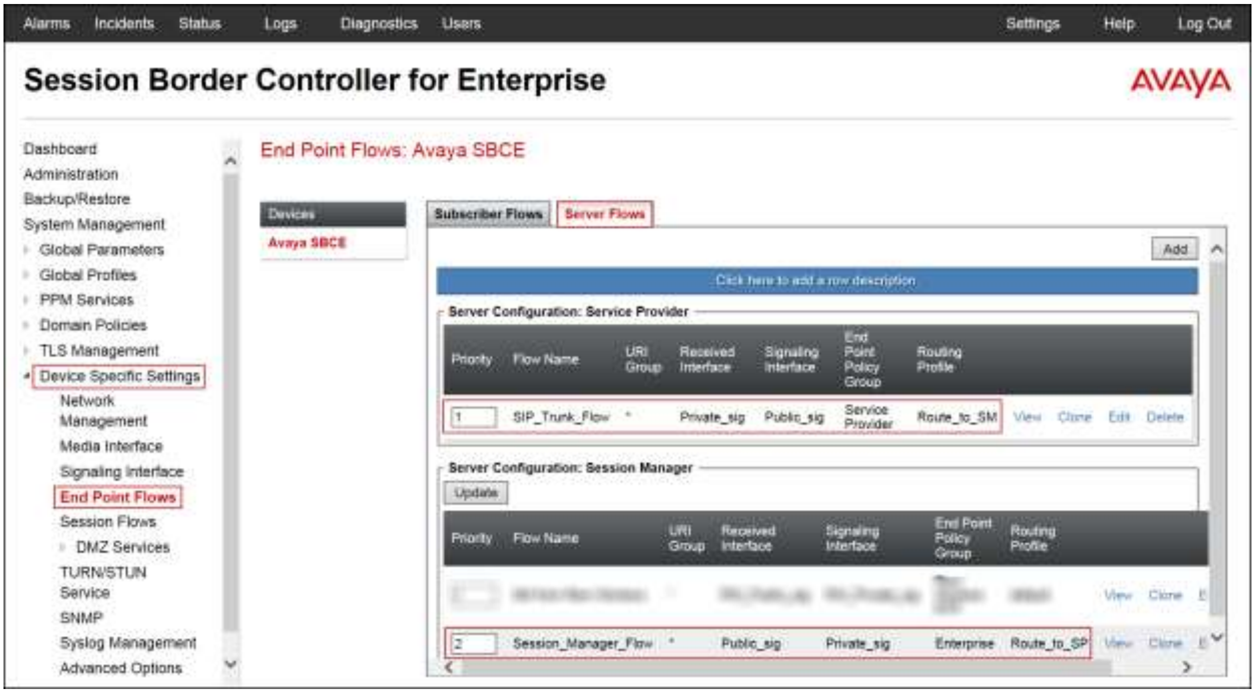


To create the call flow toward the Session Manager, click **Add**.

- **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:** ***Enterprise***.
- **Routing Profile:** ***Route_to_SP*** (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. Claro SIP Trunking Service Configuration

To use Claro's SIP Trunking Service, a customer must request the service from Claro using the established sales processes. The process can be started by contacting Claro via the corporate web site at: http://www.claro.com.do/wps/portal/do/sc/empresas and requesting information.

During the signup process, Claro and the customer will discuss details about the preferred method to be used to connect the customer's enterprise network to Claro's network. Claro will provide IP addresses, Direct Inward Dialed (DID) numbers to be assigned to the enterprise, etc. This information is used to complete the Avaya Aura® Communication Manager, Avaya Aura® Session Manager and Avaya Session Border Controller for Enterprise configuration discussed in the previous sections.

During the interoperability testing, a VPN connection was used to connect the simulated enterprise site to Claro's network via the public Internet. The connection could also be done without the use of a VPN connection, by directly connecting the Avaya SBCE via the public Internet to a public facing SBC located in Claro's network. This is accomplished by assigning public IP addresses, capable of being reached across the public Internet, to the Avaya SBCE (interface **B1**) and to the Claro's SBC.

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.

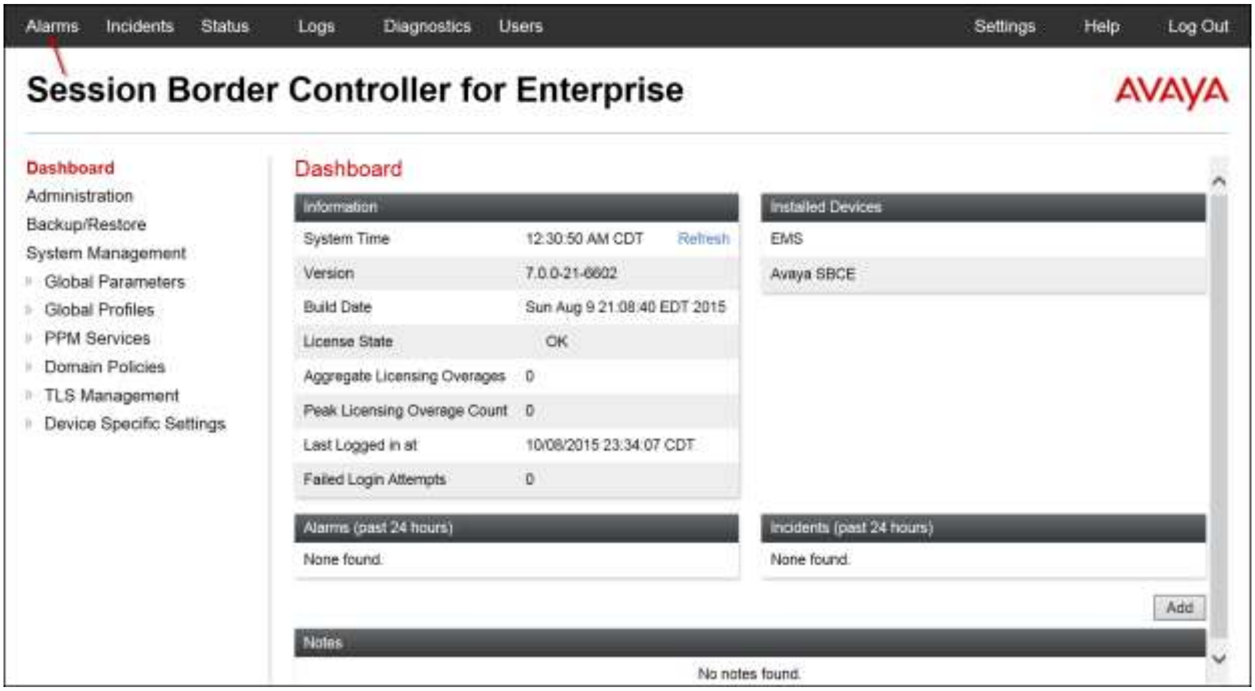
9.1.2. Session Manager

- **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 SBCE

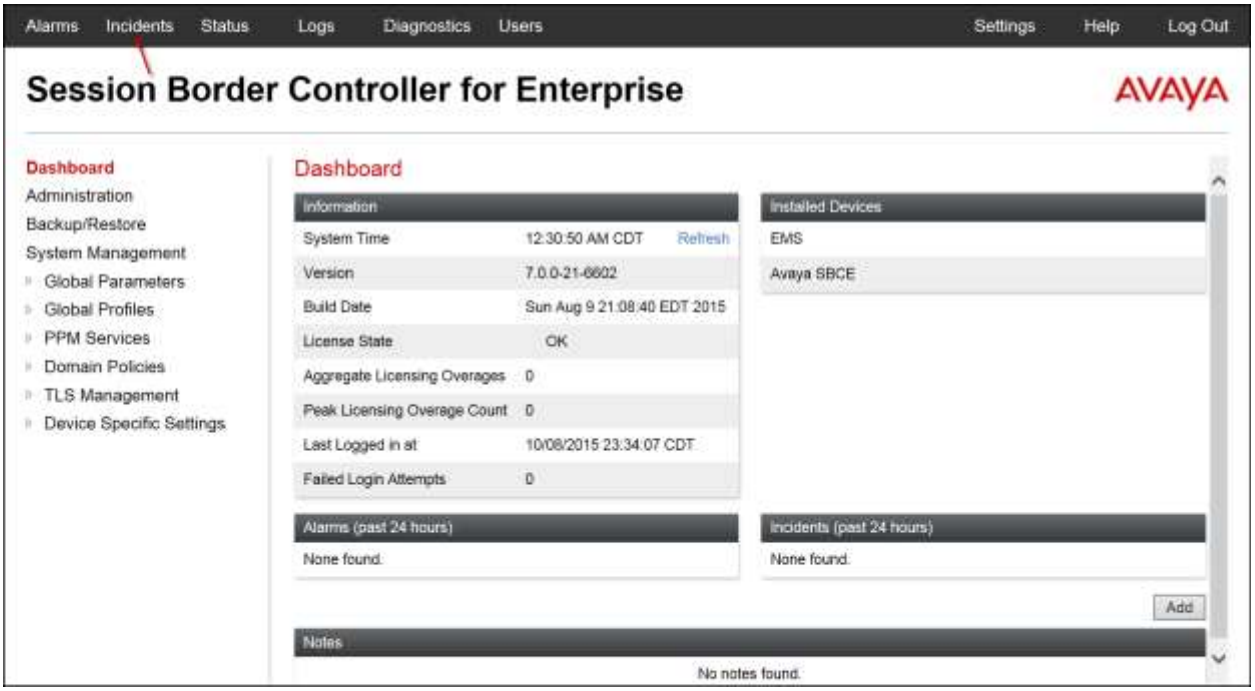
There are several links and menus located on the taskbar at the top of the screen of the web interface that can be used for diagnostic and troubleshooting.

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

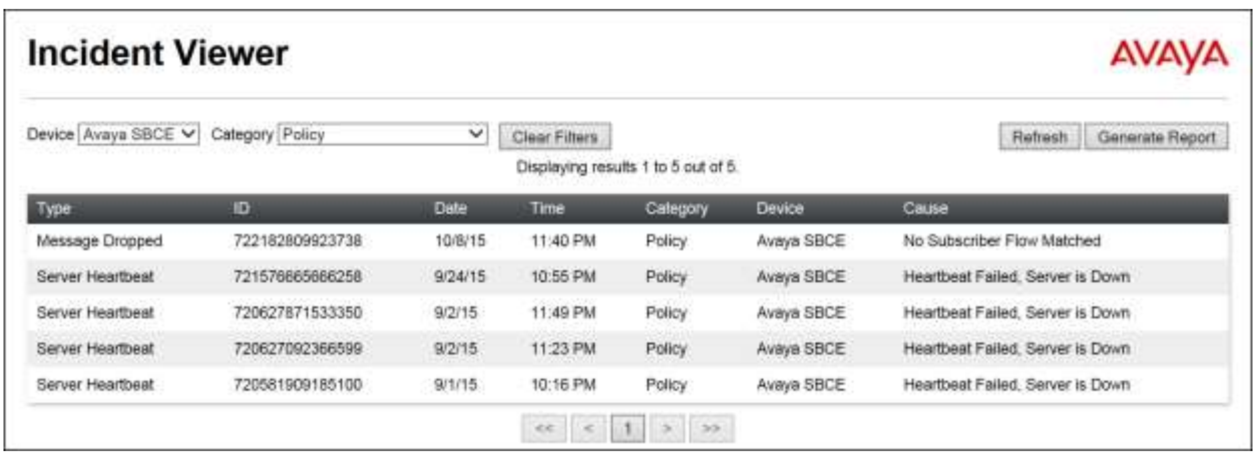


The following screen shows the **Alarm Viewer** page.

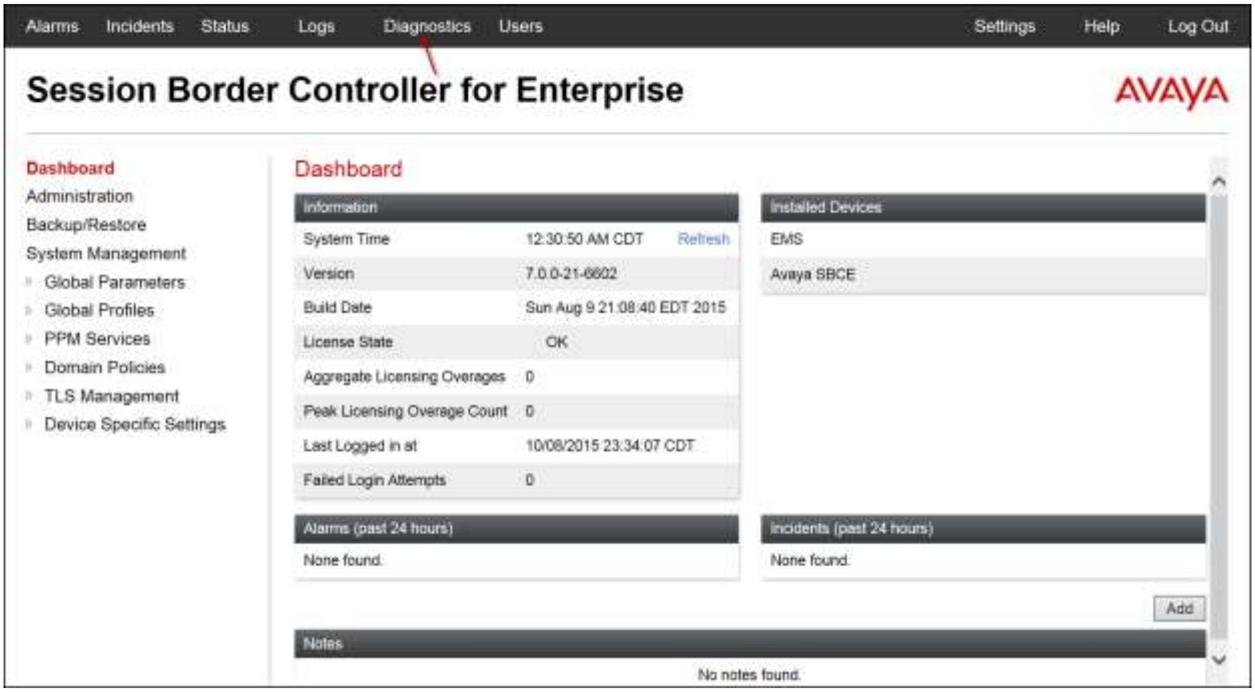
Incidents : Provides detailed reports of anomalies, errors, policies violations, etc.



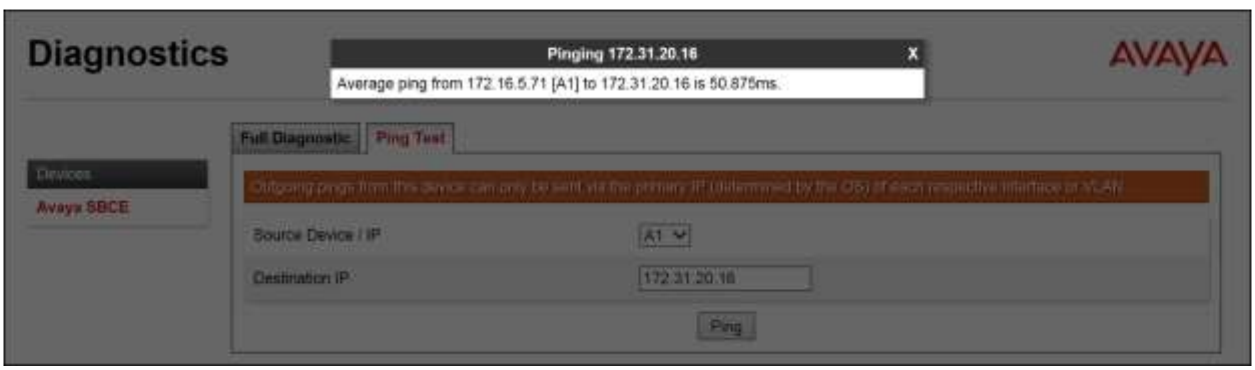
The following screen shows the Incident Viewer page.



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

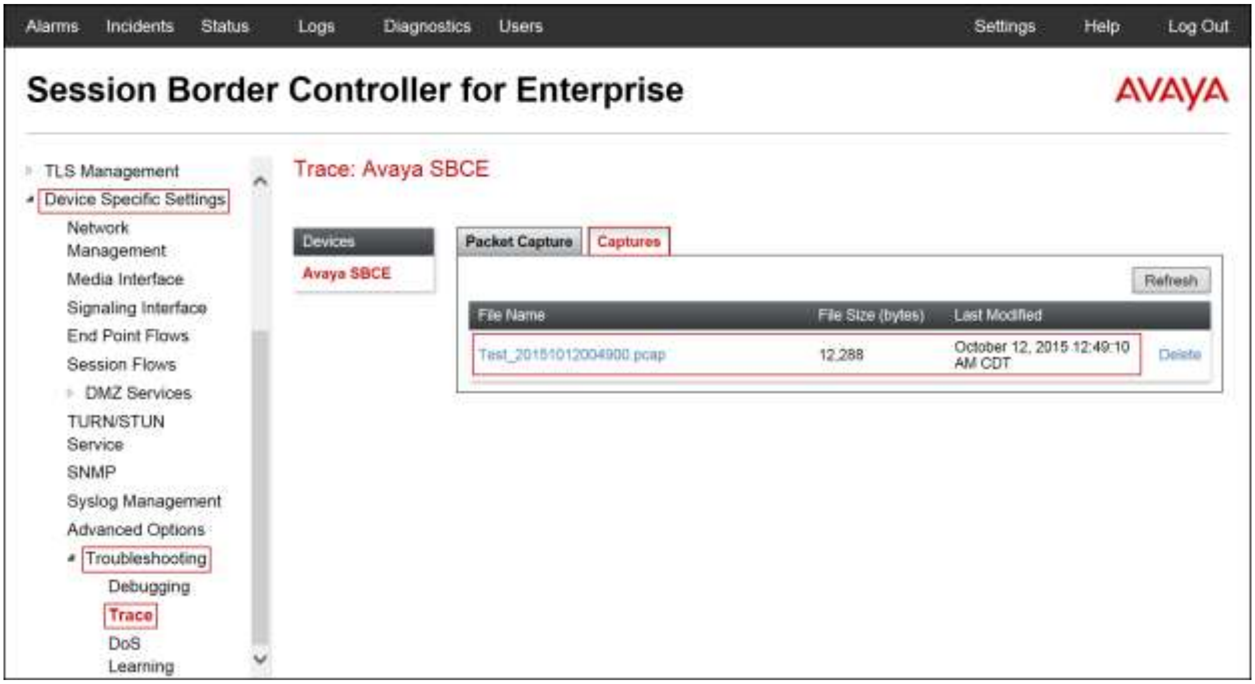


The following screen shows the Diagnostics page with the results of a ping test. Note that IP addresses have been blurred out for security reasons.



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.



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 Claro SIP Trunking Service, 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.

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

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

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

[4] *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/

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

[6] *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/

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

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

Other resources:

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

[10] *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 7.2.3**. When adding this script as instructed in **Section 7.2.3** enter a name for the script in the Title (e.g., **Claro_SigMa**) and copy/paste the entire script.

Title: Claro_SigMa

```
//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"]);

  }
}

//The following script is required in order to play Music On Hold when a call is placed on
//hold at the PBX (CM). The script removes a=sendonly from the INVITE message sent by the
//PBX (CM) to Claro.

within session "INVITE" {
 act on request where %DIRECTION="OUTBOUND" and
%ENTRY_POINT="POST_ROUTING"
  {

   //Removes the a=sendonly from the re-INVITE messages.

   %BODY[1].regex_replace("a=sendonly\r\n","");

  }

}
```

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.