Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager 8.1, Avaya Aura® Session Manager 8.1, Avaya Aura® Experience Portal 7.2 and Avaya Session Border Controller for Enterprise 8.0 with Alestra SIP Trunking Service – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking Service on an enterprise solution consisting of Avaya Aura® Communication Manager 8.1, Avaya Aura® Session Manager 8.1, Avaya Aura® Experience Portal 7.2 and Avaya Session Border Controller for Enterprise 8.0 to interoperate with Alestra SIP Trunking service. These Application Notes update previously published Application Notes with newer versions of Communication Manager, Session Manager, and Avaya Session Border Controller for Enterprise.

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.

The Alestra SIP Trunking service provides customers with PSTN access via a SIP trunk between the enterprise and the Alestra 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 procedures for configuring Session Initiation Protocol (SIP) Trunking Service between the Alestra network and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Aura® Communication Manager 8.1 (Communication Manager), Avaya Aura® Session Manager 8.1 (Session Manager), Avaya Aura® Experience Portal 7.2 (Experience Portal), Avaya Session Border Controller for Enterprise 8.0 (Avaya SBCE) and various Avaya endpoints, listed in **Section 4**.

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

The terms "Service Provider" or "Alestra" will be used interchangeably throughout these Application Notes.

2. General Test Approach and Test Results

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

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

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in this DevConnect Application Note included the enablement of supported encryption capabilities in the Avaya products only (private network side). Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

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

For the testing associated with this Application Note, the interface between Avaya systems and the Alestra SIP Trunking service did not include the use of any specific encryption features.

Encryption (TLS/SRTP) was used internal to the enterprise between Avaya products wherever possible.

2.1. Interoperability Compliance Testing

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

- SIP Trunk Registration (Dynamic Authentication).
- Response to SIP OPTIONS queries.
- Incoming calls from the PSTN were routed to DID numbers assigned by Alestra. Incoming PSTN calls were terminated to the following endpoints: Avaya 96x1 Series IP Deskphones (H.323 and SIP), Avaya J179 IP Deskphones (H.323), Avaya 2420 Digital Deskphones, Avaya one-X® Communicator softphone (H.323 and SIP), Avaya Equinox softphone (SIP) and analog Deskphones.
- Inbound and outbound PSTN calls to/from Remote Workers using Avaya 96x1 Deskphones (SIP).
- Outgoing calls to the PSTN were routed via Alestra network to various PSTN destinations.
- Proper disconnect when the caller abandons the call before the call is answered.
- Proper disconnect via normal call termination by the caller or the called parties.
- Proper disconnect by the network for calls that are not answered (with voicemail off).
- Proper response to busy endpoints.
- Proper response/error treatment when dialing invalid PSTN numbers.
- Proper Codec negotiation and two-way speech-path. Testing was performed with codecs: G.729, G.711MU and G.711A.
- No matching codecs.
- DTMF tone transmissions as out-of-band RTP events as per RFC2833:
    - Outbound call to PSTN application requiring DTMF (e.g., an IVR or voice mail system).
    - Inbound call from PSTN to Avaya CPE application requiring DTMF (e.g., Aura® Messaging, Experience Portal, Avaya vector digit collection steps.
- 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.
- Inbound caller interaction with Experience Portal applications, including prompting, caller DTMF input, wait treatment (e.g., announcements and/or music on hold).
- Experience Portal use of SIP REFER to redirect inbound calls, via the Avaya SBCE, to the appropriate Communication Manager agents and extensions.
- Call and two-way talk path establishment between callers and Communication Manager agents and extensions following redirection from Experience Portal.
- Routing inbound vector call to call center agent queues.

- Simultaneous active calls.
- Long duration calls (over one hour).
- Proper response/error treatment to all trunks busy.
- Proper response/error treatment when disabling SIP connection.

Note – Remote Worker was tested as part of this solution. The configuration necessary to support remote workers is beyond the scope of these Application Notes and is not included in these Application Notes. Consult reference [**11**] in the **References** section for additional information on this topic.

Items that are supported and that were not tested includes the following:

- Fax is not currently supported on Alestra's Broadsoft platform.
- Inbound toll free calls were not tested.
- 0, 0+10 digits, 911 Emergency and international calls were not tested.
- SIP NCR using SIP 302 Re-direction message. (Redirect before answer) was not tested.

2.2. Test Results

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

- **OPTIONS –** Alestra does not send OPTIONS messages to the Avaya enterprise network, but it does respond to OPTIONS messages it receives from the Avaya enterprise, this was enough to maintain the SIP trunk link up in service.
- "**483 Too Many Hops**" **response from Session Manager –** Inbound calls from the PSTN to SIP endpoints at the enterprise were failing with Session Manager responding with "483 Too Many Hops". This was caused by Alestra sending the "Max-Forward" header value in SIP messages set to "9". This issue was solved by adding a Sigma script to the Session Manager Server Configuration Profile to change the Max-Forward value from "9" to "69". Refer to **Section 8.8** and **13**.
- **Outbound Calling Party Number (CPN) Block** – Alestra did not allow outbound calls with privacy enabled. When a user activated "CPN Block" to enable user privacy on an outbound call, CM sent "anonymous" in the "From" header and the "Privacy: id" header, while the caller information was still being sent in the **"**P-Asserted-Identity" header. Alestra responded with a "500 Server Internal Error" message and the call was rejected. Alestra requires the Pilot number associated with the SIP trunk in the "From" and "Contact" headers, otherwise it will reject the call with "500 Server Internal Error", thus "anonymous" in the "From" header is not allowed.
- **Alestra requires the Pilot number associated with the SIP trunk in the "From" and "Contact" headers** – Alestra requires the Pilot number associated with the SIP trunk in the "From" and "Contact" headers of SIP messages it receives from the enterprise, otherwise it will reject the call with "500 Server Internal Error". A SigMa script was added to the Service Provider Server Configuration Profile to add the Pilot number to the "From" and "Contact" headers of SIP messages sent to Alestra. Refer to **Section 8.8** and **13**.

- **Caller ID on outbound calls** – On calls originating from the enterprise to the PSTN, the caller ID number displayed at the PSTN endpoints was always of the Pilot number associated with the SIP trunk, not of the DID number assigned to the Communication Manager extension originating the call. This included calls to "twinned" mobile phones (EC500), and calls that were forwarded or transferred back out on the SIP trunk to the PSTN. This is the expected behavior of the Alestra service for outbound calls on the SIP trunk; it is listed here simply as an observation.
- **TLS/SRTP used within the enterprise** – When TLS/SRTP is used within the enterprise; the SIP headers include the SIPS URI scheme for Secure SIP. The Avaya SBCE converts these header schemes from SIPS to SIP when it sends the SIP message toward Alestra. However, for call forward and EC500 calls, the Avaya SBCE was not changing the Diversion header scheme as expected. This anomaly is currently under investigation by the Avaya SBCE team. A workaround is to include a SigMa script for the Service Provider Server Configuration Profile on the Avaya SBCE to convert "sips" to "sip" in the Diversion header (**Sections 8.8** and **13**).
- **Removal of unwanted xml element information from the SDP in SIP messages sent to Alestra** – A Signaling Manipulation script (SigMa) was added to the Avaya SBCE to remove unwanted xml element information from the SDP in SIP messages sent to Alestra, the xml elements were causing Alestra to respond with "500 Error in IRP: processing UA response" to an UPDATE messages sent by Communication Manager. (**Sections 8.8** and **13**).
- **Call Transfers to the PSTN by Experience Portal** – With SIP REFER enabled in Communication Manager, calls from the PSTN to Experience Portal that were transferred back out to the PSTN by Experience Portal failed to complete with Alestra rejecting the REFER message sent by the Avaya SBCE with "403 Forbitten". This issue was observed with the Refer Handling feature in the Avaya SBCE and with the *URI Group* criteria as a discriminator enabled, whereby SIP REFER messages matching the URI Group criteria are processed by the Avaya SBCE, while SIP REFER messages that do not match the URI Group criteria, are passed through to the Service Provider. With SIP REFER enabled in Communication Manager the *URI Group* criteria method for SIP REFER handling in the Avaya SBCE is required since Experience Portal inbound call processing may include call redirection to Communication Manager agents, other CPE destinations or call to the PSTN. The work around to this issue is to disable SIP REFER in Communication Manager by setting "Network Call Redirection" to "n" on the Trunk Group (**Section 5.7**) and enabling the REFER handling feature in the Avaya SBCE **without** the *URI Group* criteria as a discriminator, as shown in **Appendix B**. With this setting call transfers to Communication Manager agents, to other CPE destinations or to the PSTN by Experience Portal were successful. The work around mentioned should only be used if the customer plans to deploy Experience Portal, otherwise SIP REFER should be left enabled. Note that the testing (except for the Experience Portal testing) was done with SIP REFER enabled in Communication Manager, as shown in **Section 5.7**.
- **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-Correlation-ID, Alert-Info, Endpoint-View, P-AV-Message-id, P-Charging-Vector, AV-Global-Session-ID and P-Location (Refer to **Section 7.4**). To help reduce the packet size further, the Avaya SBCE can remove the "*gsid*" and "*epv"* parameters that may be included within the Contact header by applying a Sigma script to the Alestra server configuration. Refer to **Section 8.8** and **13**.

2.3. Support

For support of Alestra SIP Trunking Service visit the corporate Web page at: http://www.alestra.com.mx/

For technical support on the Avaya products described in these Application Notes visit http://support.avaya.com

3. Reference Configuration

Figure 1 illustrates the sample Avaya SIP-enabled enterprise solution, connected to the Alestra SIP Trunking Service through the public Internet.

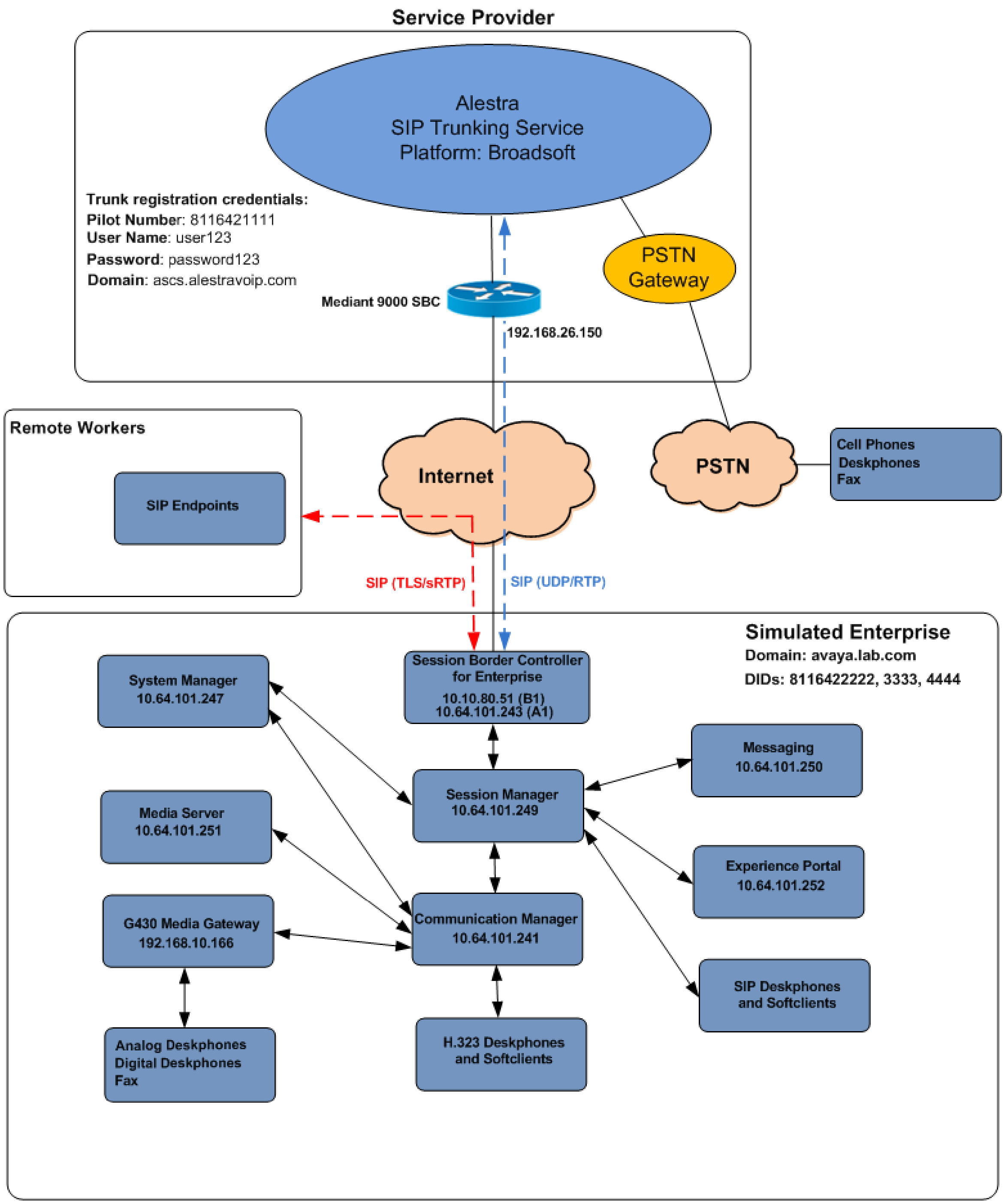


Figure 1: **Avaya SIP Enterprise Solution connected to Alestra SIP Trunking Service**

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

- Avaya Aura® Communication Manager.
- Avaya Aura® Session Manager.
- Avaya Aura® System Manager.
- Avaya Session Border Controller for Enterprise.
- Avaya Aura® Messaging.
- Avaya Aura® Media Server.
- Avaya Aura® Experience Portal.
- Avaya G430 Media Gateway.
- Avaya 96x1 Series IP Deskphones (H.323 and SIP).
- Avaya J179 IP Deskphones (H.323).
- Avaya one-X® Communicator softphones (H.323 and SIP).
- Avaya Equinox™ for Windows softphone (SIP).
- Avaya digital and analog telephones.

Additionally, the reference configuration included remote worker functionality. A remote worker is a SIP endpoint that resides in the untrusted network, registered to Session Manager at the enterprise via the Avaya SBCE. Remote workers offer the same functionality as any other endpoint at the enterprise. This functionality was successfully tested during the compliance test using only the Avaya 96x1 SIP Deskphones. For signaling, Transport Layer Security (TLS) and for media, Secure Real-time Transport Protocol (SRTP) was used on Avaya 96x1 SIP Deskphones used to test remote worker functionality. Other Avaya SIP endpoints that are supported in a Remote Worker configuration deployment were not tested.

The configuration tasks required to support remote workers are beyond the scope of these Application Notes; hence they are not discussed in this document. Consult reference [**11**] in the **References** section for additional information on this topic.

The Avaya SBCE was located at the edge of the enterprise. Its public side was connected to the public Internet, while its private side was connected to the enterprise infrastructure. All signaling and media traffic entering or leaving the enterprise flowed through the Avaya SBCE, protecting in this way the enterprise against any SIP-based attacks. The Avaya SBCE also performed network address translation at both the IP and SIP layers.

For inbound calls, the calls flowed from the service provider to the Avaya SBCE then to Session Manager. Session Manager used the configured dial patterns (or regular expressions) and routing policies to determine the recipient (Communication Manager or Experience Portal) and on which link to send the call.

Outbound calls to the PSTN were first processed by Communication Manager for outbound feature treatment such as automatic route selection and class of service restrictions. Once Communication Manager selected the proper SIP trunk, the call was routed to Session Manager.

Session Manager once again used the configured dial patterns (or regular expressions) and routing policies to determine the route to the Avaya SBCE for egress to the Alestra network.

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

As part of the Avaya Aura® version 8.0 release, Communication Manager incorporates the ability to use the Avaya Aura® Media Sever (AAMS) as a media resource. The AAMS is a software-based, high density media server that provides DSP resources for IP-based sessions. Media resources from both the AAMS and a G430 Media Gateway were utilized during the compliance test. The configuration of the AAMS is not discussed in this document. For more information on the installation and administration of the AAMS in Communication Manager refer to the AAMS documentation listed in the **References** section.

The Avaya Aura® Messaging was used during the compliance test to verify voice mail redirection and navigation, as well as the delivery of Message Waiting Indicator (MWI) messages to the enterprise telephones. Since the configuration tasks for Messaging are not directly related to the interoperability tests with the Alestra network SIP Trunking service, they are not included in these Application Notes.

The Avaya Aura® Experience Portal was also used during the compliance test to verify various SIP call flow scenarios with Alestra SIP trunking service.

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya	
Avaya Aura® Communication Manager	8.1.0.1.1 (01.0.890.0-25517)
Avaya Aura® Session Manager	8.1.0.0 (8.1.0.0.810007)
Avaya Aura® System Manager	8.1.0.0 Build No. 8.1.0.0.733078 Software Update Rev. No. 8.1.0.0.079880
Avaya Session Border Controller for Enterprise	ASBCE 8.0 8.0.0.0-19-16991
Avaya Aura® Messaging	7.1 Service Pack 1 (MSG-01.0.532.0-0100)
Avaya Aura® Media Server	8.0.1.121_2019.04.29
Avaya G430 Media Gateway	g430_sw_41_9_0
Avaya Aura® Experience Portal	7.2.2.0.2118
Avaya 96x1 Series IP Deskphones (SIP)	Version 7.1.5.0.11
Avaya 96x1 Series IP Deskphones (H.323)	Version 6.8202
Avaya J179 IP Deskphones (H.323)	Version 6.8202
Avaya one-X® Communicator (H.323, SIP)	6.2.14.1-SP14
Avaya Equinox for Windows (SIP)	3.5.7.30.1
Avaya 2420 Series Digital Deskphones	N/A
Avaya 6210 Analog Deskphones	N/A
Alestra	
Broadsoft Broadworks	21sp1
Lucent 5ESS	R16
AudioCodes Mediant 9000 SBC	7.20A.204-015

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

Note – The Avaya Aura® servers and the Avaya SBCE used in the reference configuration and shown on the previous table were deployed on a virtualized environment. These Avaya components ran as virtual machines over VMware® (ESXi 6.0.0) platforms. Consult the installation documentation on the **References** section for more information.

5. Configure Avaya Aura® Communication Manager

This section describes the procedure for configuring Communication Manager to work with the Alestra SIP Trunking Service. A SIP trunk is established between Communication Manager and Session Manager for use by signaling traffic to and from the service provider. It is assumed that the general installation of Communication Manager, the Avaya G430 Media Gateway and the Avaya Media Server has been previously completed and is not discussed here.

The Communication Manager configuration was performed using the System Access Terminal (SAT). Some screens in this section have been abridged and highlighted for brevity and clarity in presentation. Some screens capture 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 trunks to and from the service provider. The example shows that **40000** licenses are available and **120** are in use. The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative.

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

IP PORT CAPACITIES                                             USED
                 Maximum Administered H.323 Trunks: 12000       0
       Maximum Concurrently Registered IP Stations: 18000       2
         Maximum Administered Remote Office Trunks: 12000       0
Max Concurrently Registered Remote Office Stations: 18000       0
          Maximum Concurrently Registered IP eCons:  414        0
     Max Concur Reg Unauthenticated H.323 Stations:   100       0
                    Maximum Video Capable Stations:  41000      0
               Maximum Video Capable IP Softphones:  18000      6
                   Maximum Administered SIP Trunks: 40000     120
  Max Administered Ad-hoc Video Conferencing Ports: 24000       0
   Max Number of DS1 Boards with Echo Cancellation: 999     0




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

5.2. System Features

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

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

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




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

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

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

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

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

INTERNATIONAL CALL ROUTING PARAMETERS
                 Local Country Code:
          International Access Code:

SCCAN PARAMETERS
   Enable Enbloc Dialing without ARS FAC? n

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

5.3. IP Node Names

Use the **change node-names ip** command to verify that node names have been previously defined for the IP addresses of Communication Manager (**proc**r) and the Session Manager security module (**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              10.64.101.243
SM                    10.64.101.249
default               0.0.0.0
media server          10.64.101.251
procr                 10.64.101.241
procr6                ::

( 6  of 6    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. Enter the corresponding codec in the **Audio Codec** column of the table. Alestra supports audio codecs ***G.729***, ***G.711MU*** and ***G.711A***.

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

                          IP MEDIA PARAMETERS
    Codec Set: 2

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

     Media Encryption                     Encrypted SRTCP: best-effort
 1: 1-srtp-aescm128-hmac80
 2: none
 3:
 4:
 5:
```

On **Page 2**, set the **Fax Mode** to ***off*** (Refer to **Section 2.1**).

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

                          IP MEDIA PARAMETERS

                              Allow Direct-IP Multimedia? n



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


Media Connection IP Address Type Preferences
 1: IPv4
 2:
```

5.5. IP Network Regions

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

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

```
change ip-network-region 2                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 2       NR Group: 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 (the rest of the enterprise). Enter the desired IP codec set in the **codec set** column of the row with destination region (**dst rgn**) 1. Default values may be used for all other fields. The following example shows the settings used for the compliance test. It indicates that codec set ***2*** will be used for calls between region 2 (the service provider region) and region 1 (the rest of the enterprise).

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

 Source Region: 2     Inter Network Region Connection Management     I      M
                                                                     G  A   t
 dst codec direct   WAN-BW-limits   Video      Intervening       Dyn A  G   c
 rgn  set    WAN  Units      Total Norm  Prio Shr Regions        CAC R  L   e
 1    2      y    NoLimit                                            n      t
 2    2                                                                 all
 3
 4
 5
 6
 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 trunk. This signaling group is used for inbound and outbound calls between the service provider and the enterprise. For the compliance test, signaling group 2 was used and was configured using the parameters highlighted below, shown on the screen on the next page:

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

Note: Once the **Peer-Server** field is updated to ***SM***, the system changes the default values of the following fields, setting them to display–only:

- **Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers?** is changed to ***y***.
- **Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers?** is changed to ***n***.
- Set the **Near-end Node Name** to ***procr***. This node name maps to the IP address of the Communication Manager as defined in **Section 5.3**.
- Set the **Far-end Node Name** to ***SM***. This node name maps to the IP address of Session Manager, as defined in **Section 5.3**.
- Set the **Near-end Listen Port** and **Far-end Listen Port** to a valid unused port instead of the default well-known port value. (For TLS, the well-known port value is 5061). This is necessary so Session Manager can distinguish this trunk from the trunk used for other enterprise SIP traffic. The compliance test was conducted with the **Near-end Listen Port** and **Far-end Listen Port** set to ***5071***.
- Set the **Far-end Network Region** to the IP network region defined for the Service Provider in **Section 5.5**.
- Set the **Far-end Domain** to the domain of the enterprise.
- Set the **DTMF over IP** field to ***rtp-payload***. This value enables Communication Manager to send DTMF transmissions using RFC 2833.
- Set **Direct IP-IP Audio Connections** to ***y***. This field will enable media shuffling on the SIP trunk allowing Communication Manager to redirect media traffic directly between the Avaya SBCE and the enterprise endpoint. If this value is set to **n**, then the Avaya Media Gateway or Media Server will remain in the media path of all calls between the SIP trunk and the endpoint. Depending on the number of media resources available in the Avaya Media Gateway and Media Server, these resources may be depleted during high call volume preventing additional calls from completing.
- Default values may be used for all other fields.

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

 Group Number: 2             Group Type: sip
  IMS Enabled? n       Transport Method: tls
        Q-SIP? n
     IP Video? n                                 Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM                     Clustered? n
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                  Far-end Node Name: SM
 Near-end Listen Port: 5071                 Far-end Listen Port: 5071
                                         Far-end Network Region: 2

Far-end Domain: avaya.lab.com
                                          Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate               RFC 3389 Comfort Noise? n
        DTMF over IP: rtp-payload          Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                   IP Audio Hairpinning? n
        Enable Layer 3 Test? n                  Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n          Alternate Route Timer(sec): 6
```

5.7. Trunk Group

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

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

```
change trunk-group 2                                            Page   1 of   4
                                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** is set to a value acceptable to the service provider. This value defines the interval that re-INVITEs must be sent to keep the active session alive. The default value of ***600*** seconds was used.

```
change trunk-group 2                                            Page   2 of   4
      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. When ***public*** format is used, Communication Manager automatically inserts a "+" sign, preceding the numbers in the "From", "Contact" and "P-Asserted Identity" (PAI) headers. To keep uniformity with the format used by Alestra, the **Numbering Format** was set to ***public*** and the **Numbering Format** in the route pattern was set to ***pub-unk*** (see **Section 5.10**).
- Set the **Replace Restricted Numbers** and **Replace Unavailable Numbers** fields to ***y***. This will allow the CPN displayed on local endpoints to be replaced with the value set in **Section 5.2**, if the inbound call has enabled CPN block.

```
change trunk-group 2                                            Page   3 of   4
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y



   Suppress # Outpulsing? n  Numbering Format: public
                                                 UUI Treatment: service-provider

                                                  Replace Restricted Numbers? y
                                                 Replace Unavailable Numbers? y

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




 Show ANSWERED BY on Display? y
```

On **Page 4**:

- Set the **Network Call Redirection** field to ***y***. With this setting, Communication Manager will use the SIP REFER method for the redirection of PSTN calls that are transferred back to the SIP trunk (Refer to **Section 2.2** for issues related to Experience Portal).
- Set the **Send Diversion Header** field to ***y*** and **Support Request History** to ***n***.
- Set the **Telephone Event Payload Type** to **101**, the value preferred by Alestra.
- Verify that **Identity for Calling Party Display** is set to ***P-Asserted-Identity***.
- Default values were used for all other fields.

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

                                        Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                       Send Transferring Party Information? n
                                 Network Call Redirection? y
          Build Refer-To URI of REFER From Contact For NCR? n
                                     Send Diversion Header? y
                                   Support Request History? n
                              Telephone Event Payload Type: 101


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

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

5.8. Calling Party Information

The calling party number is sent in the SIP "From", "Contact" and "PAI" headers. Since public numbering 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. DID numbers are provided by the SIP service provider. Each DID number is assigned in this table to one enterprise internal extension or Vector Directory Numbers (VDNs). In the example below, three DID numbers assigned by the service provider are shown. These DID numbers were used as the outbound calling party information on the service provider trunk when calls were originated from the mapped extensions.

```
change 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: 3
 4  3                                              4      Maximum Entries: 9999
 4  5                                              4
 4  3041                 2          8116422222     10  Note: If an entry applies to
 4  3042                 2          8116423333     10  a SIP connection to Avaya
 4  3044                 2          8116424444     10  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

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

```
change inc-call-handling-trmt trunk-group 2                          Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number       Del  Insert
 Feature         Len      Digits
 public-ntwrk     10 8116422222       10  3041
 public-ntwrk     10 8116423333       10  3042
 public-ntwrk     10 8116424444       10  3044
 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: ____
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 group to the service provider.

For international call to the U.S. (e.g., dialing: 90017863311234):

```
change ars analysis 001                                      Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                 Location: all            Percent Full: 1
```

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd
001	13	18	2	intl		n
01	12	12	2	natl		n
011	10	18	2	intl		n
040	3	3	2	svcl		n
045	13	13	2	natl		n
101xxxx0	8	8	deny	op		n
101xxxx0	18	18	deny	op		n
101xxxx01	16	24	deny	iop		n
101xxxx011	17	25	deny	intl		n
101xxxx1	18	18	deny	fnpa		n
10xxx0	6	6	deny	op		n
10xxx0	16	16	deny	op		n
10xxx01	14	22	deny	iop		n
10xxx011	15	23	deny	intl		n
10xxx1	16	16	deny	fnpa		n

For calls within Mexico (e.g., dialing: 928811011):

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

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd
2	8	8	2	hnpa		n
3	7	7	1	hnpa		n
4	7	7	1	hnpa		n
407	10	10	2	hnpa		n
411	3	3	2	svcl		n
443	10	10	2	hnpa		n
5	7	7	2	hnpa		n
555	7	7	deny	hnpa		n
6	7	7	2	hnpa		n
611	3	3	2	svcl		n
63	8	8	2	hnpa		n
631	10	10	2	hnpa		n
7	10	10	2	hnpa		n
808	10	10	2	hnpa		n
809	10	10	2	hnpa		n

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

- **Pattern Name**: Enter a descriptive name.
- **Grp No**: Enter the outbound trunk group for the SIP service provider.
- **FRL**: Set the Facility Restriction Level (**FRL**) field to a level that allows access to this trunk for all users that require it. The value of **0** is the least restrictive level.
- **Numbering Format**: Set to ***pub-unk***. All calls using this route pattern will use the public numbering table. See setting of the **Numbering Format** in the trunk group form for full details in **Section 5.7**.

```
change route-pattern 2                                          Page   1 of   4
                    Pattern Number: 2      Pattern Name: Serv. Provider
    SCCAN? n     Secure SIP? n     Used for SIP stations? n

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

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

Note - Enter the **save translation** command (not shown) to save all the changes made to the Communication Manager configuration in the previous sections.

6. Configure Avaya Aura® Experience Portal

These Application Notes assume that the necessary Experience Portal licenses have been installed and basic Experience Portal administration has already been performed. Consult [**9**] in the **References** section for further details if necessary.

6.1. Background

Experience Portal consists of one or more Media Processing Platform (MPP) servers and an Experience Portal Manager (EPM) server. A single "server configuration" was used in the reference configuration. This consisted of a single MPP and EPM, running on a VMware environment, including an Apache Tomcat Application Server (hosting the Voice XML (VXML) and/or Call Control XML (CCXML) application scripts), that provide the directives to Experience Portal for handling the inbound calls.

References to the Voice XML and/or Call Control XML applications are administered on Experience Portal, along with one or more called numbers for each application reference. When an inbound call arrives at Experience Portal, the called party DID number is matched against those administered called numbers. If a match is found, then the corresponding application is accessed to handle the call. If no match is found, Experience Portal informs the caller that the call cannot be handled and disconnects the call[1].

For the sample configuration described in these Application Notes, a simple VXML test application was used to exercise various SIP call flow scenarios with Alestra SIP Trunking service. In production, enterprises can develop their own VXML and/or CCXML applications to meet specific customer self-service needs or consult Avaya Professional Services and/or authorized Avaya Business Partners. The development and deployment of VXML and CCXML applications is beyond the scope of these Application Notes.

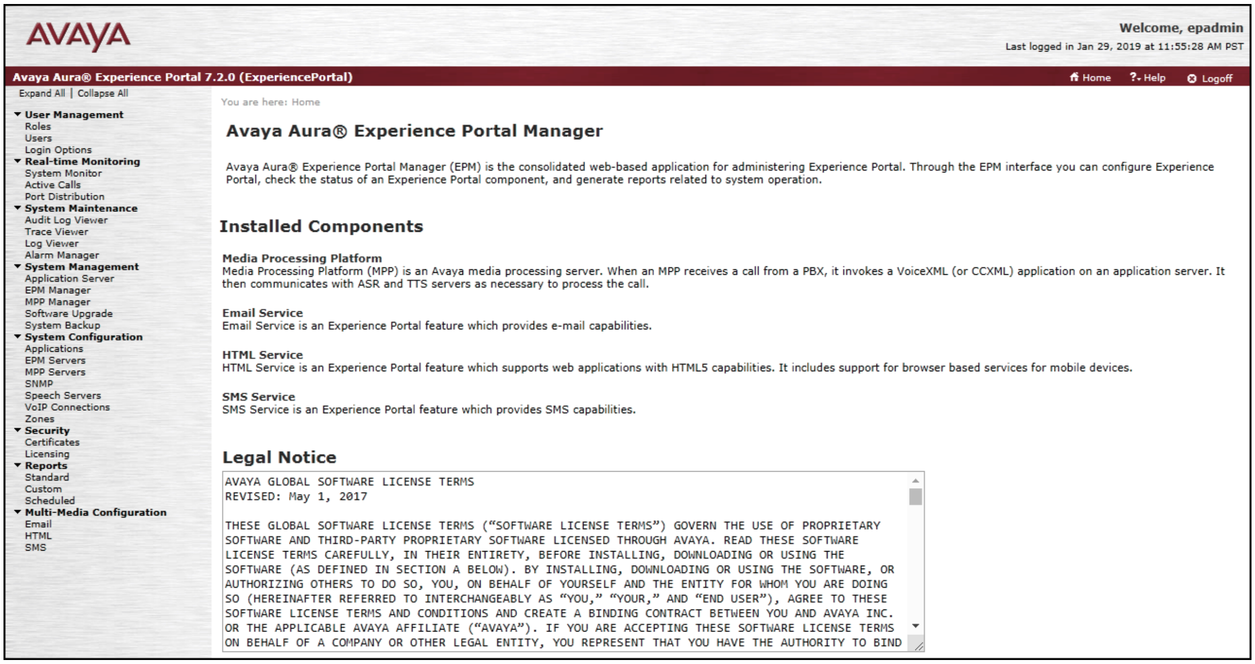
[1] An application may be configured with "inbound default" as the called number, to process all inbound calls that do not match any other application references.

6.2. Logging in and Licensing

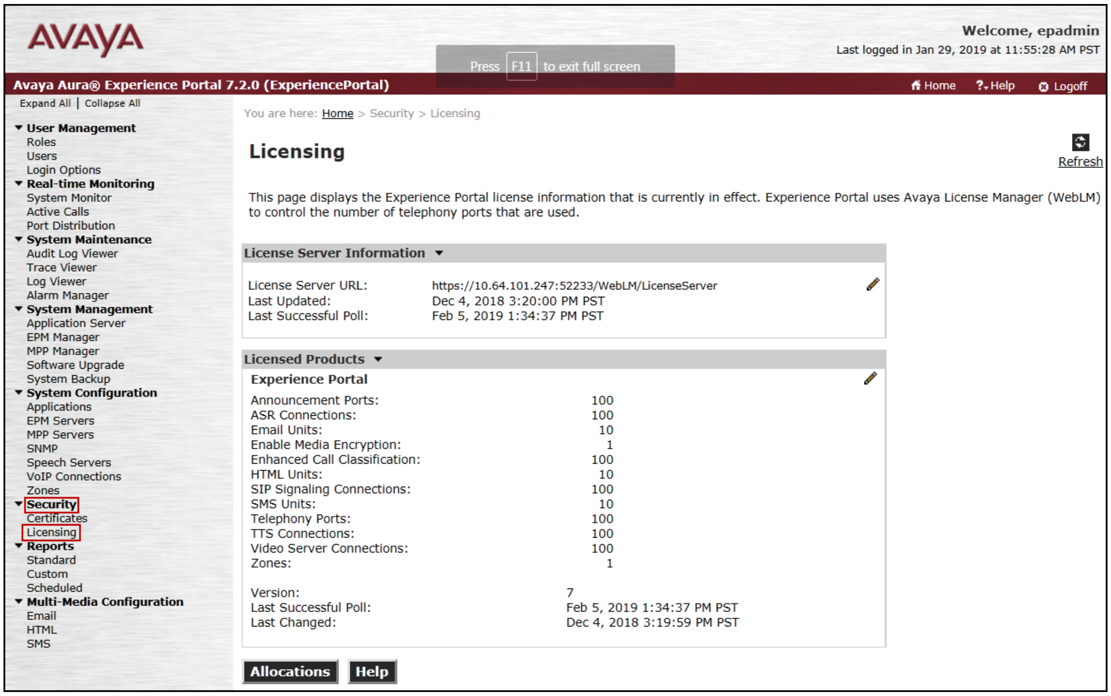
This section describes the steps on Experience Portal for administering a SIP connection to the Session Manager.

Step 1 - Launch a web browser, enter http://<IP address of the Avaya EPM server>/ in the URL, log in with the appropriate credentials and the following screen is displayed.

Note – All page navigation described in the following sections will utilize the menu shown on the left pane of the screenshot below.



Step 2 - In the left pane, navigate to **Security→Licensing**. On the **Licensing** page, verify that Experience Portal is properly licensed. If required licenses are not enabled, contact an authorized Avaya account representative to obtain the licenses.

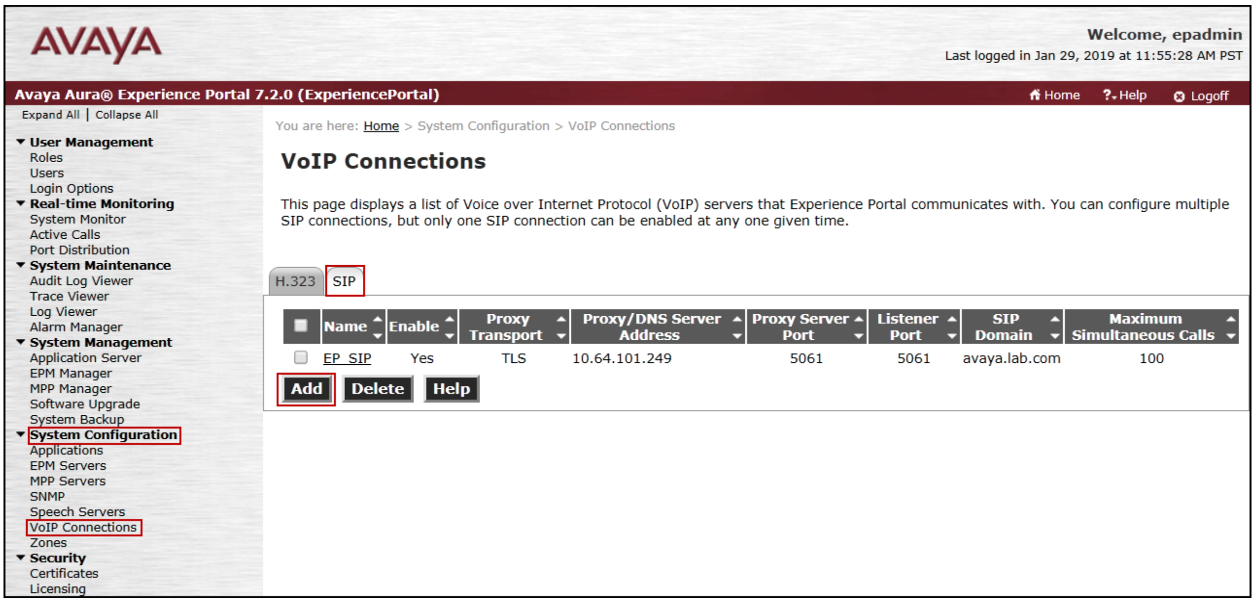


6.3. VoIP Connection

This section defines a SIP trunk between Experience Portal and Session Manager (**Sections 7.5 and 7.6**).

Step 1 - In the left pane, navigate to **System Configuration→VoIP Connections**. On the **VoIP Connections** page, select the **SIP** tab and click **Add** to add a SIP trunk.

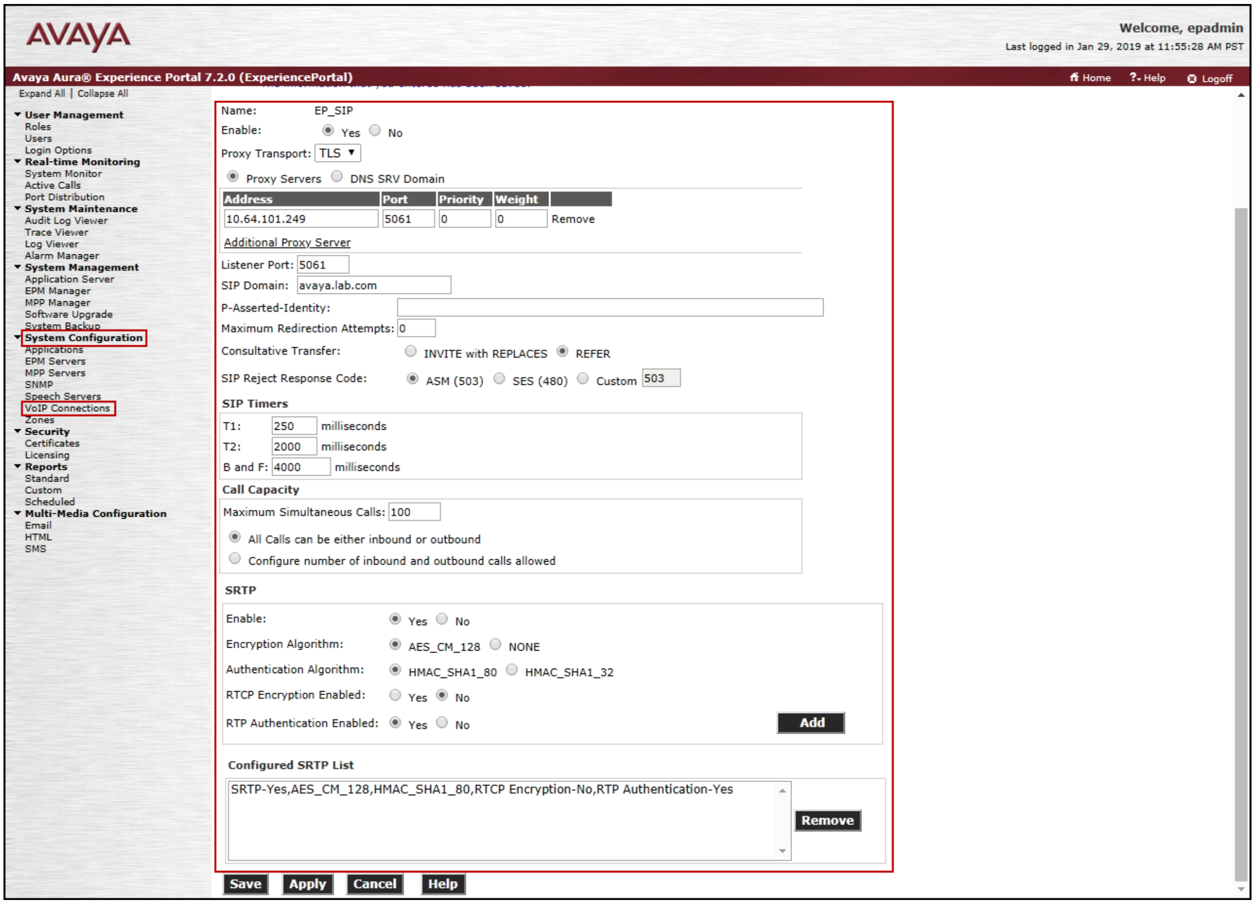
Note – Only ***one*** SIP trunk can be active at any given time on Experience Portal.



Step 2 - Configure a SIP connection as follows:

- **Name** – Set to a descriptive name (e.g., **EP_SIP**).
- **Enable** – Set to **Yes**.
- **Proxy Server Transport** – Set to **TLS**.
- Select **Proxy Servers**, and enter:
  - **Proxy Server Address** = **10.64.101.249** (the IP address of the Session Manager signaling interface defined in **Section 7.5**).
  - **Port** = **5061**
  - **Priority** = **0** (default)
  - **Weight** = **0** (default)
- **Listener Port** – Set to **5061**.
- **SIP Domain** – Set to **avaya.lab.com** (see **Section 7.2)**.
- **Consultative Transfer** – Select **REFER**.
- **SIP Reject Response Code** – Select **ASM (503)**.
- **Maximum Simultaneous Calls** – Set to a number in accordance with licensed capacity. In the reference configuration a value of **100** was used.
- Select **All Calls can be either inbound or outbound**.

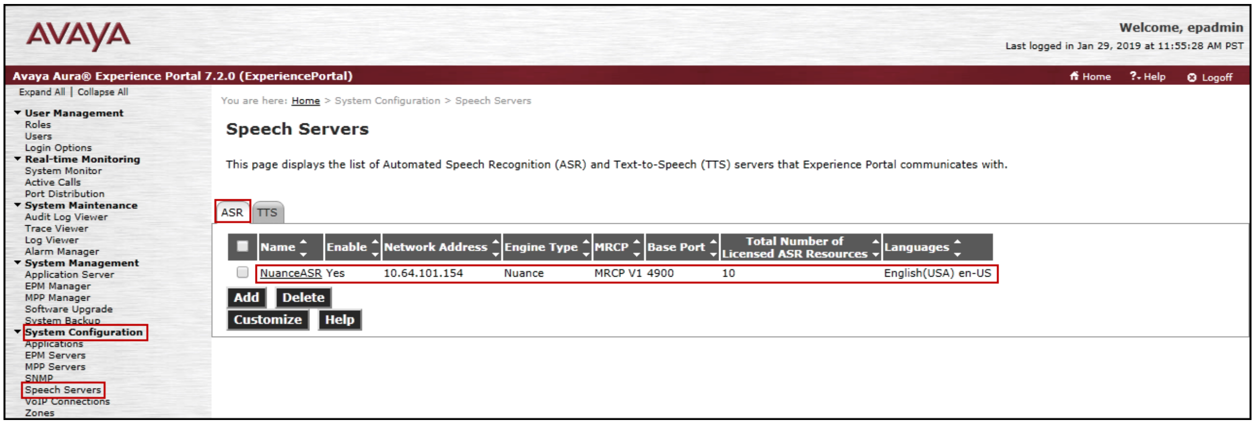
- **SRTP Enable** = **Yes**
- **Encryption Algorithm** = **AES_CM_128**
- **Authentication Algorithm** = **HMAC_SHA1_80**
- **RTCP Encryption Enabled** = **No**
- **RTP Authentication Enabled** = **Yes**
- Click on **Add** to add SRTP settings to the **Configured SRTP List**
- Use default values for all other fields.
- Click **Save**.



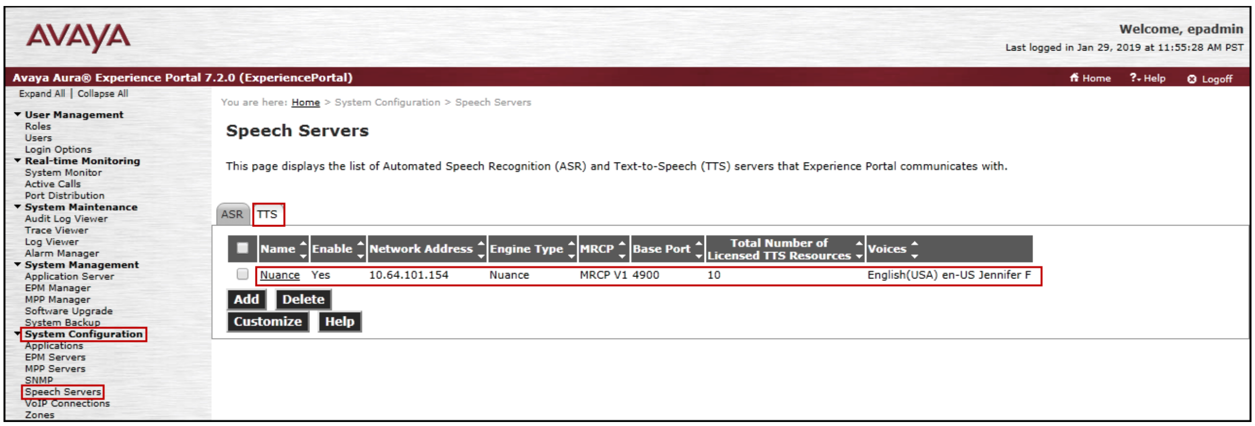
6.4. Speech Servers

The installation and administration of the ASR and TSR Speech Servers are beyond the scope of this document. Some of the values shown below were defined during the Speech Server installations. Note that in the reference configuration the ASR and TTS servers used the same IP address.

ASR speech server:



TTS speech server:



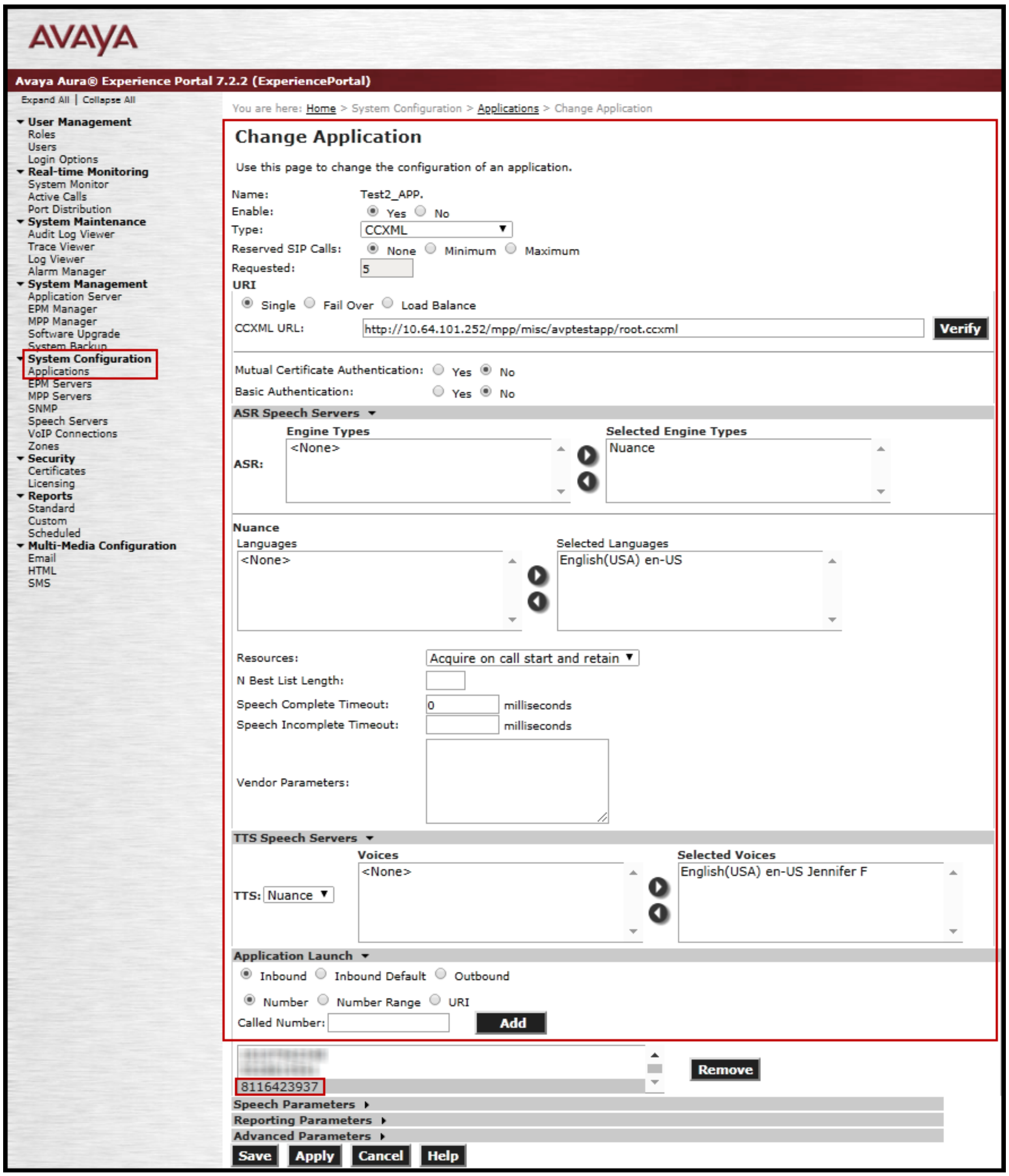
6.5. Application References

This section describes the steps for administering a reference to the VXML and/or CCXML applications residing on the application server. In the sample configuration, the applications were co-resident on one Experience Portal server, with IP Address 10.64.101.252.

Step 1 - In the left pane, navigate to **System Configuration→Applications**. On the **Applications** page (not shown), click **Add** to add an application and configure as follows:

- **Name** – Set to a descriptive name (e.g., **Test2_APP**).
- **Enable** – Set to **Yes**. This field determines which application(s) will be executed based on their defined criteria.
- **Type** – Select **VoiceXML**, **CCXML**, or **CCXML/VoiceXML** according to the application type.
- **VoiceXML** and/or **CCXML URL** – Enter the necessary URL(s) to access the VXML and/or CCXML application(s) on the application server. In the sample screen below, the Experience Portal test application on a single server is referenced.

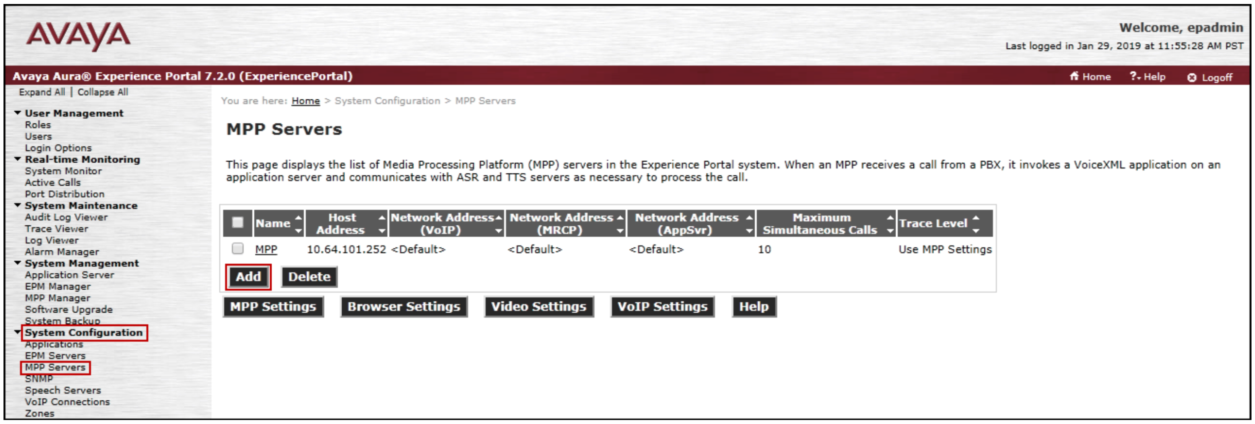
- **Speech Servers ASR** and **TTS** – Select the appropriate ASR and/or TTS servers as necessary.
- **Application Launch** – Set to **Inbound**.
- **Called Number** – Enter the number to match against an inbound SIP INVITE message and click **Add**. In the sample configuration illustrated in these Application Notes, the dialed DID number 8116423937 provided by Alestra was used. Repeat to define additional called party numbers as needed. Inbound calls with these called party numbers will be handled by the application defined in this section.



6.6. MPP Servers and VoIP Settings

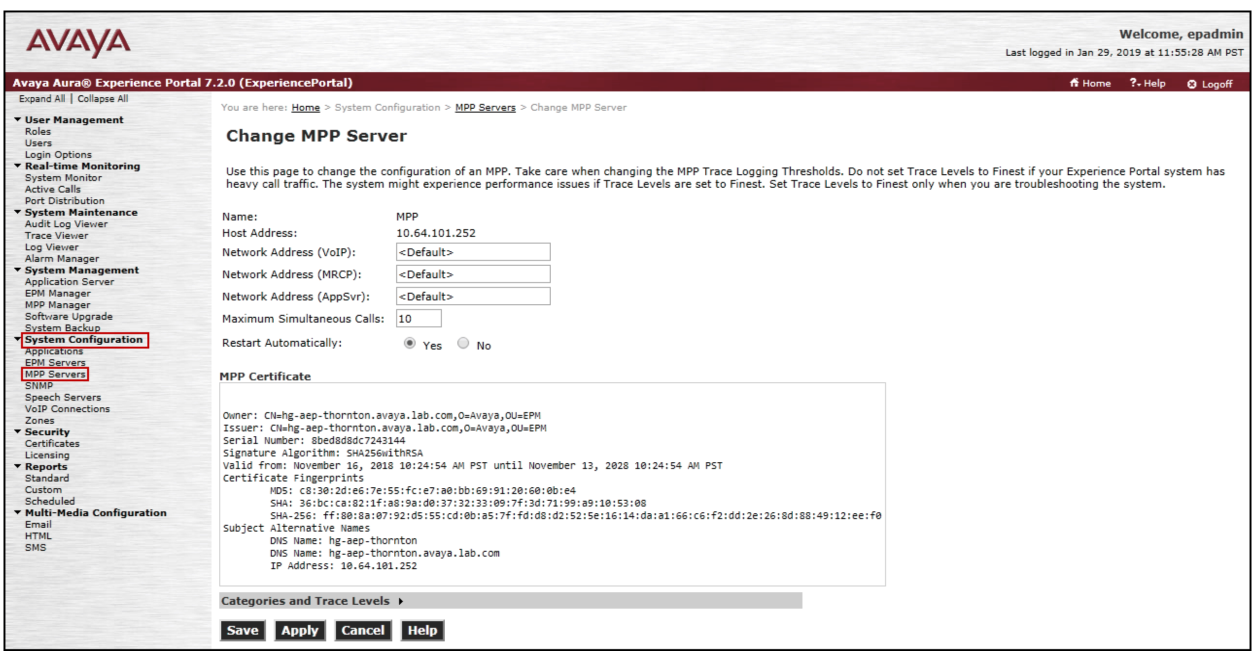
This section illustrates the procedure for viewing or changing the MPP Settings. In the sample configuration, the MPP Server is co-resident on a single server with the Experience Portal Management server (EPM).

Step 1 - In the left pane, navigate to **System Configuration→MPP Servers** and the following screen is displayed. Click **Add**.



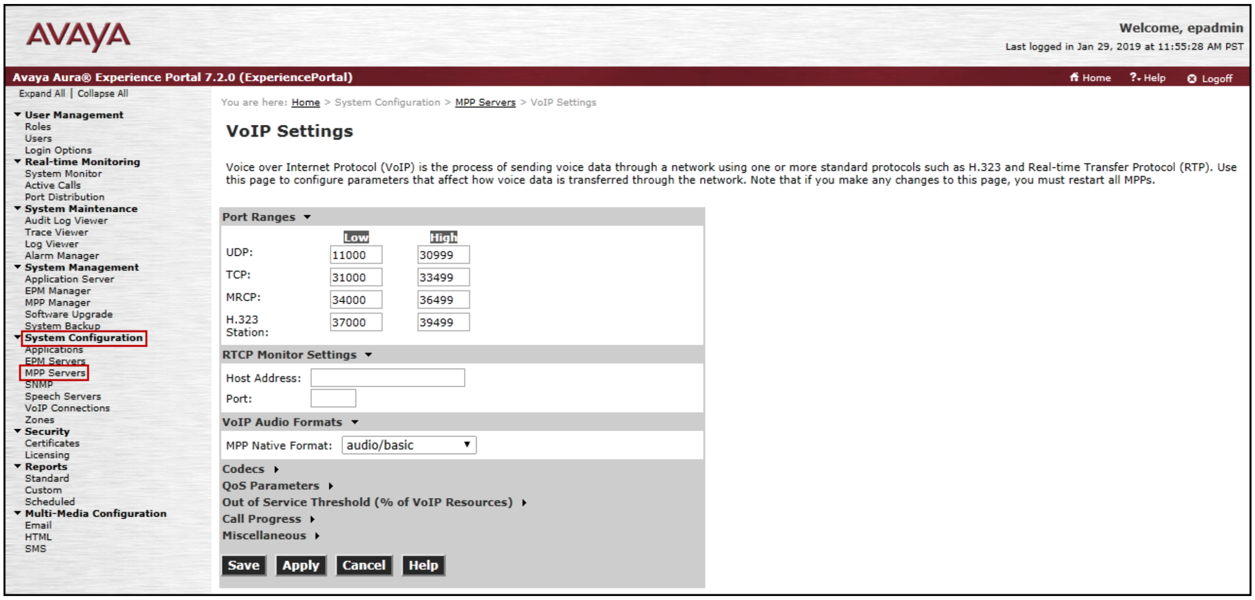
Step 2 - Enter any descriptive name in the **Name** field (e.g., **MPP**) and the IP address of the MPP server in the **Host Address** field and click **Continue** (not shown). Note that the Host Address used is the same IP address assigned to Experience Portal.

Step 3 - The certificate page will open. Check the **Trust this certificate** box (not shown). Once complete, click **Save**.



Step 4 - Click **VoIP Settings** tab on the screen displayed in **Step 1**, and the following screen is displayed.

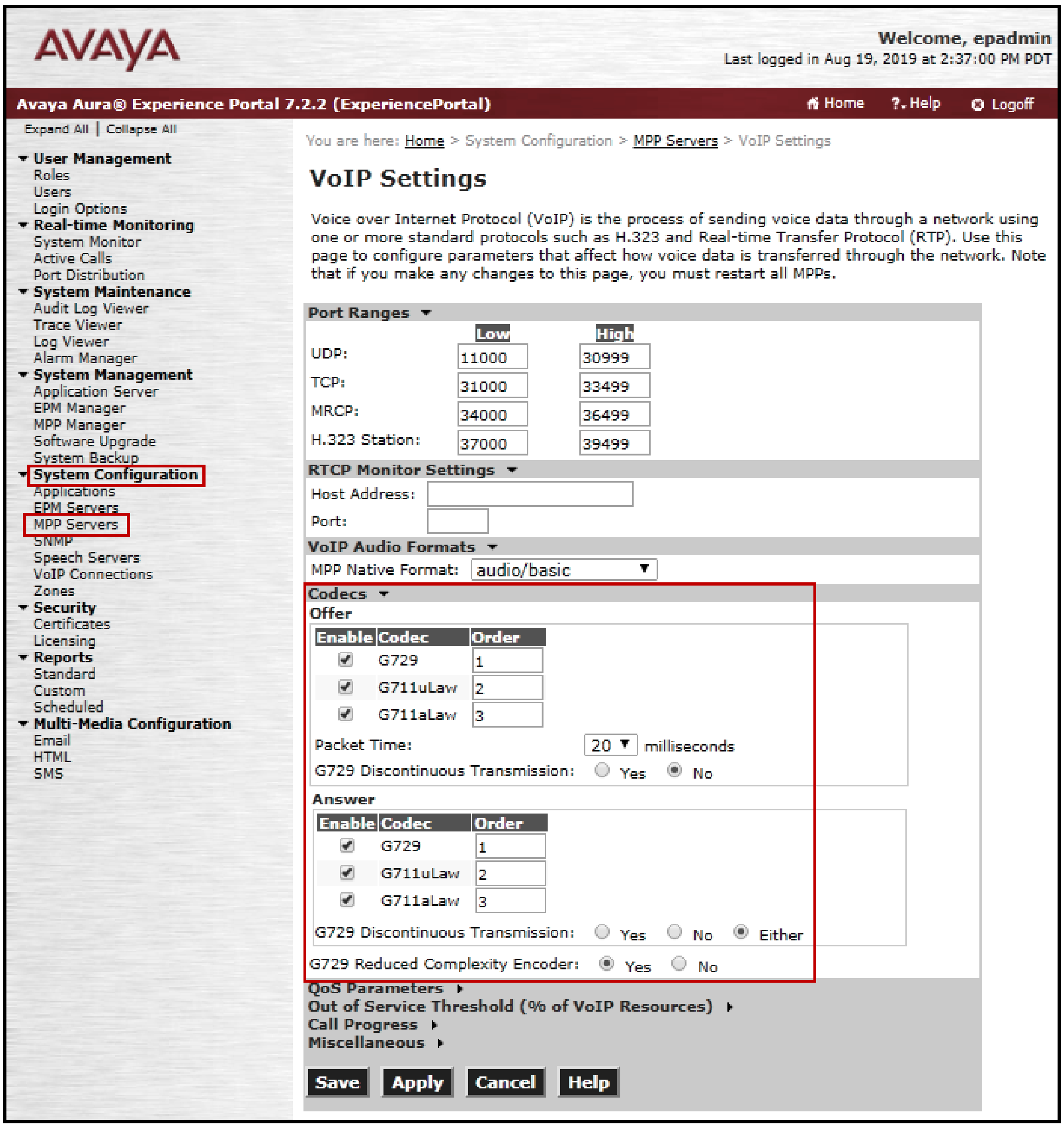
- In the Port Ranges section, default ports were used.



- In the Codecs section set:
  - Set **Packet Time** to **20**.
  - Verify Codecs **G729**, **G711uLaw** and **G711aLaw** are enabled (check marks). Set the **Offer** and Answer **Order** as shown. In the sample configuration **G729** is the preferred codec, with **Order 1**, followed by **G711uLaw** with **Order 2** and **G711aLaw** with **Order 3**.
  - On the codec Offer set **G729 Discontinuous Transmission** to **No** (for G.729A).

- Use default values for all other fields.

Step 5 - Click on **Save** (not shown).



6.7. Configuring RFC2833 Event Value Offered by Experience Portal

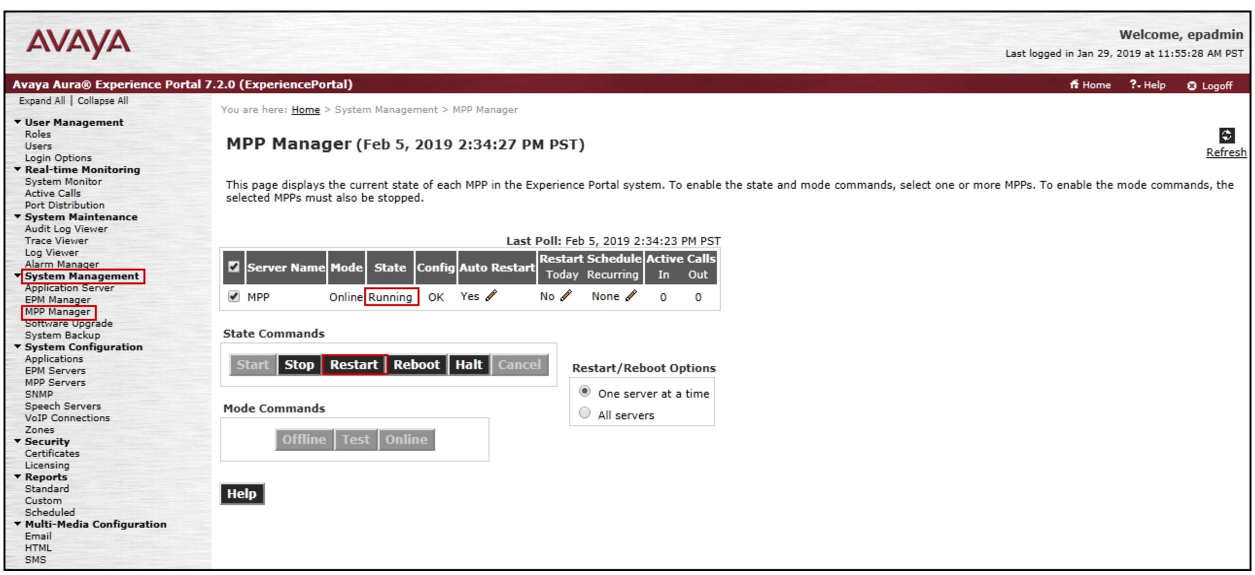
The configuration change example noted in this section was not required for any of the call flows illustrated in these Application Notes. For incoming calls from Alestra to Experience Portal, Alestra specifies the value 101 for the RFC2833 telephone-events that signal DTMF digits entered by the user. When Experience Portal answers, the SDP from Experience Portal matches this Alestra offered value.

When Experience Portal sends an INVITE with SDP as part of an INVITE-based transfer (e.g., bridged transfer), Experience Portal offers the SDP. By default, Experience Portal specifies the value 127 for the RFC2833 telephone-events. Optionally, the value that is offered by Experience Portal can be changed, and this section outlines the procedure that can be performed by an Avaya authorized representative.

- Access Experience Portal via the command line interface.
- Navigate to the following directory: /opt/Avaya/ ExperiencePortal/MPP/config
- Edit the file mppconfig.xml.
- Search for the parameter “mpp.sip.rfc2833.payload”. If there is no such parameter specified add a line such as the following to the file, where the value 101 is the value to be used for the RFC2833 events. If the parameter is already specified in the file, simply edit the value assigned to the parameter.
  ```
  <parameter name="mpp.sip.rfc2833.payload">101</parameter>
  ```
- In the verification of these Application Notes, the line was added directly above the line where the sip.session.expires parameter is configured.

After saving the file with the change, restart the MPP server for the change to take effect. As shown below, the MPP may be restarted using the **Restart** button available via the Experience Portal GUI at **System Management → MPP Manager**.

Note that the **State** column shows when the MPP is running after the restart.



7. Configure Avaya Aura® Session Manager

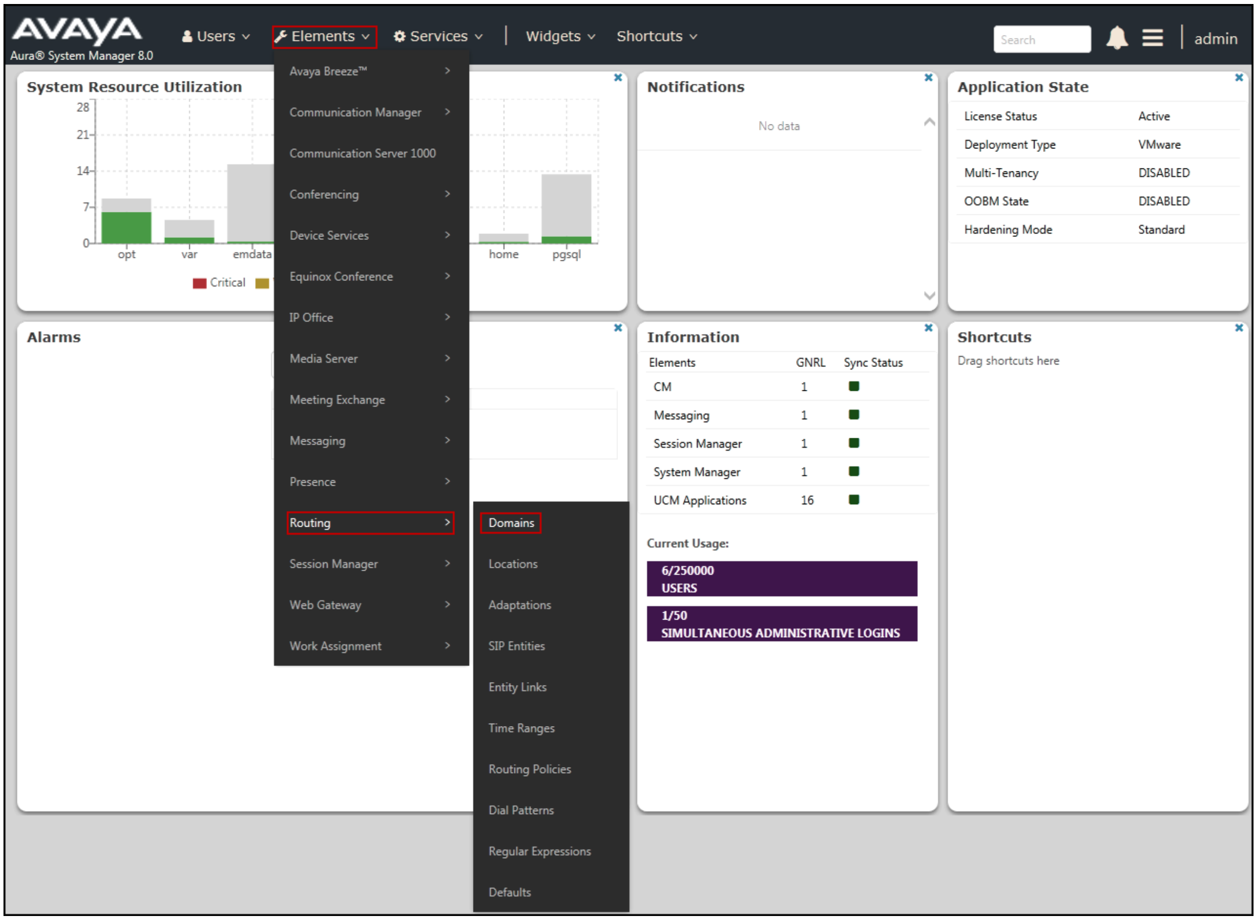
This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

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

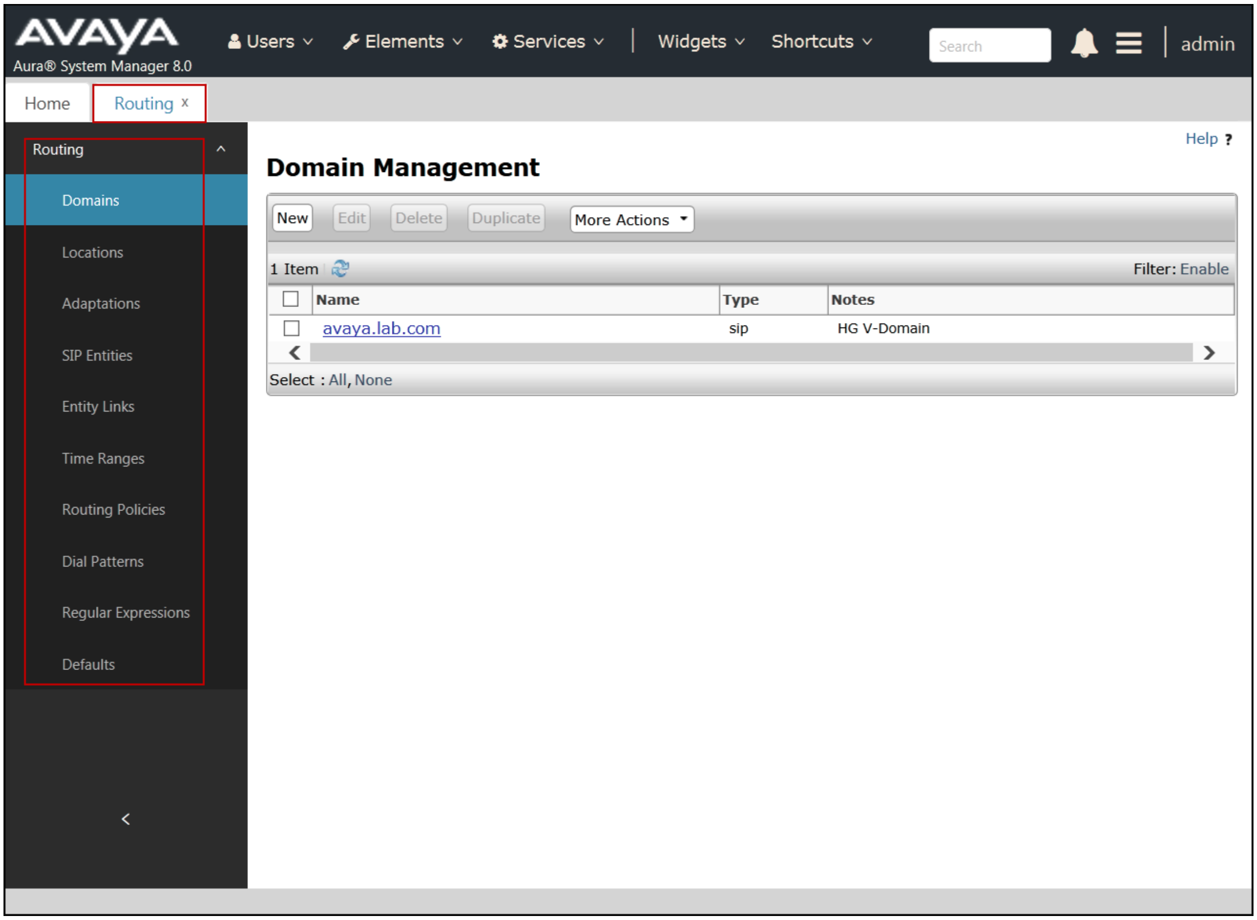
The following sections assume that the initial configuration of Session Manager and System Manager has already been completed, and that network connectivity exists between System Manager and Session Manager.

7.1. System Manager Login and Navigation

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



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

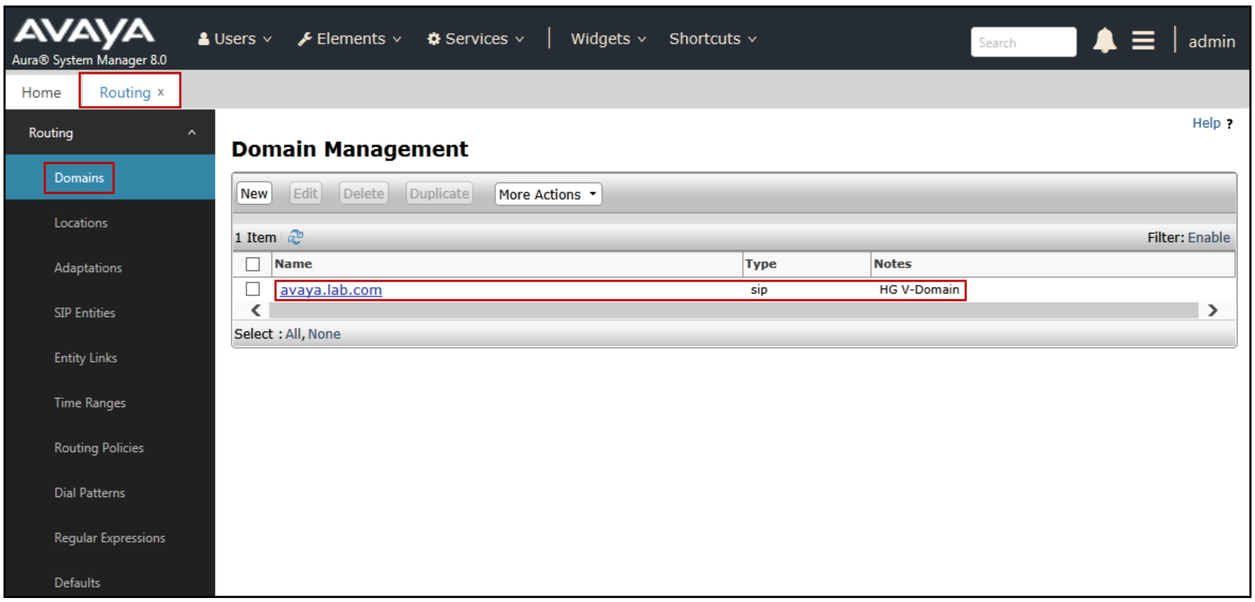


7.2. SIP Domain

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

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

The screen below shows the entry for the enterprise domain.

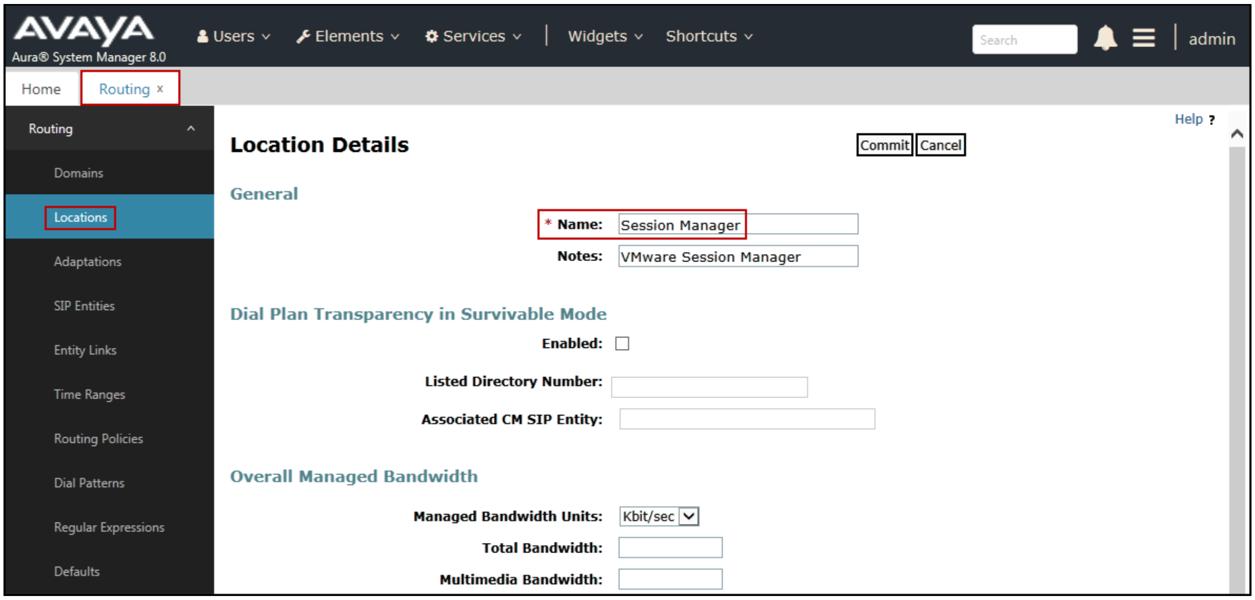


7.3. Locations

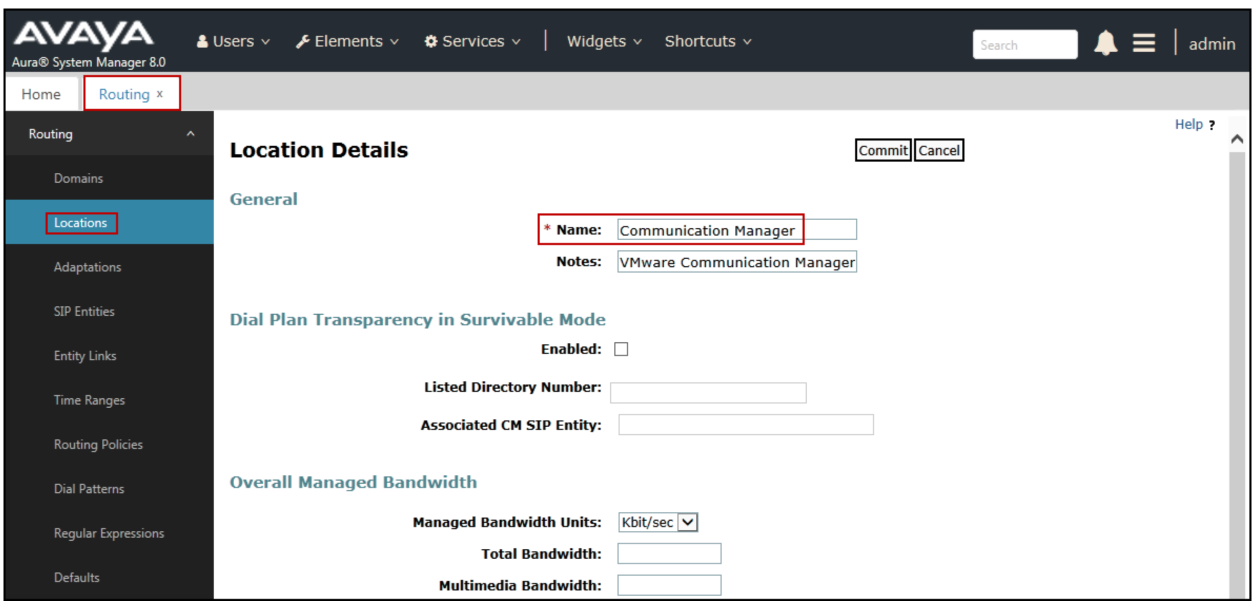
Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management, call admission control and location-based routing. To add a location, navigate to **Routing → Locations** in the left-hand navigation pane and click the **New** button in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

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

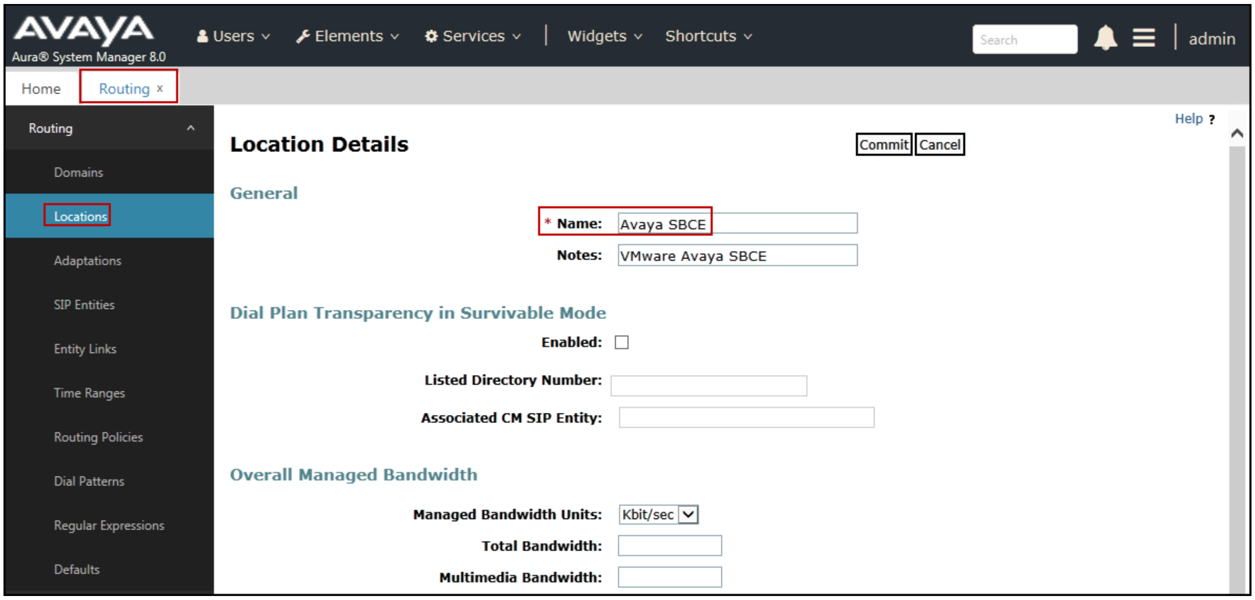
The following screen shows the location details for the location named ***Session Manager***. Later, this location will be assigned to the SIP Entity corresponding to Session Manager. Other location parameters (not shown) retained the default values.



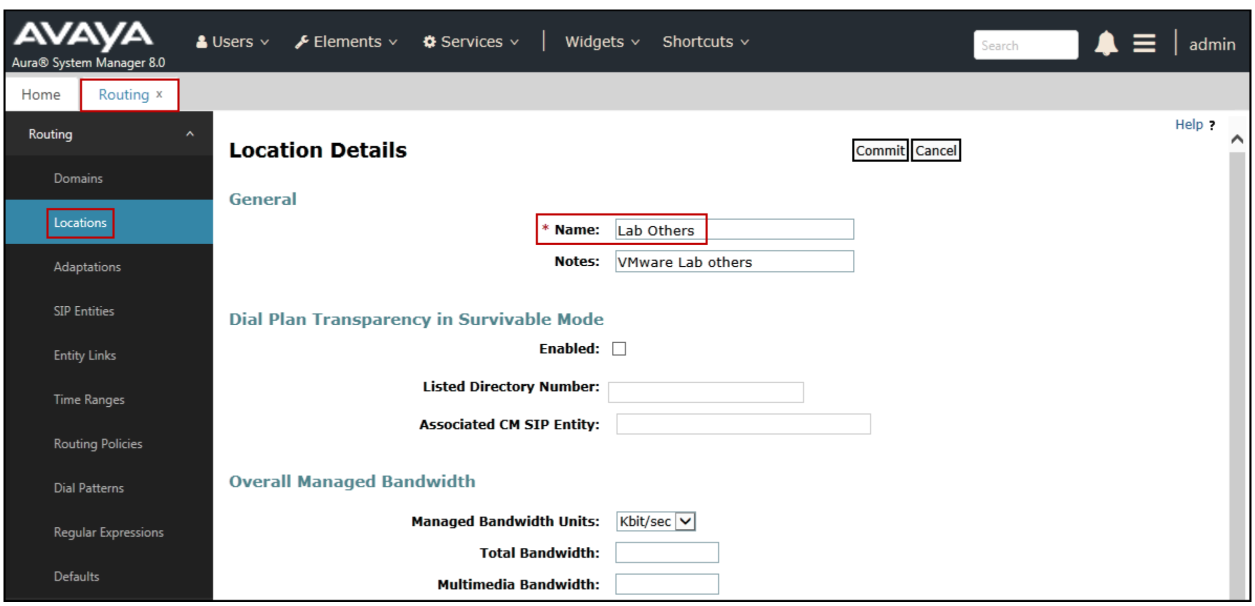
The following screen shows the location details for the location named ***Communication Manager***. Later, this location will be assigned to the SIP Entity corresponding to Communication Manager. Other location parameters (not shown) retained the default values.



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



The following screen shows the location details for the location named ***Lab Others***. Later, this location will be assigned to the SIP Entity corresponding to the Experience Portal. Other location parameters (not shown) retained the default values.



7.4. Adaptations

In order to improve interoperability with third party elements, Session Manager 8.1 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: AV-Correlation-ID, Alert-Info, Endpoint-View, P-AV-Message-ID, P-Charging-Vector and P-Location. 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.

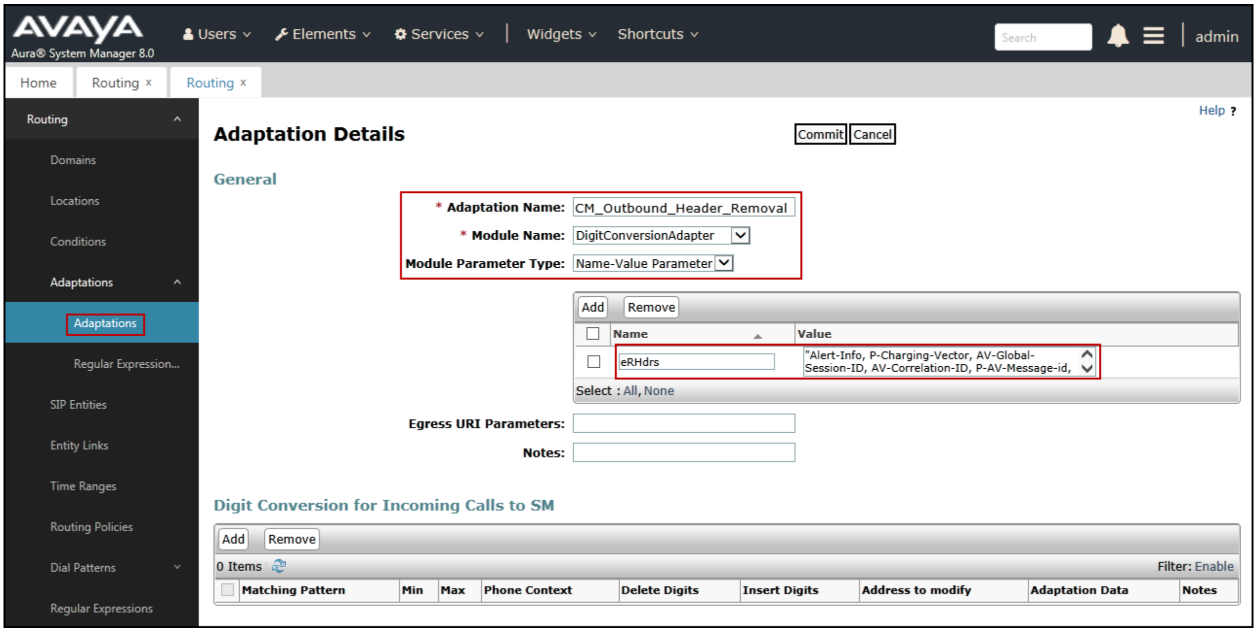
Navigate to **Routing → Adaptations** in the left-hand navigation pane and click the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

- **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, as follows:

- **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"***
- Click **Commit** to save.

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 at their default values.

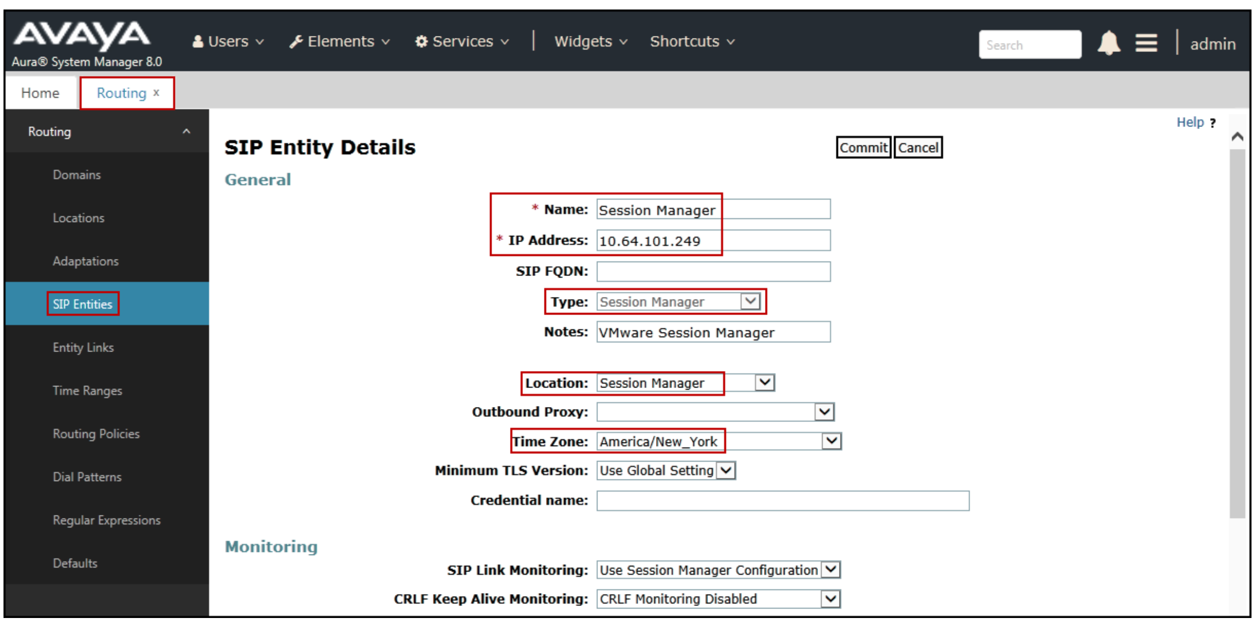


7.5. SIP Entities

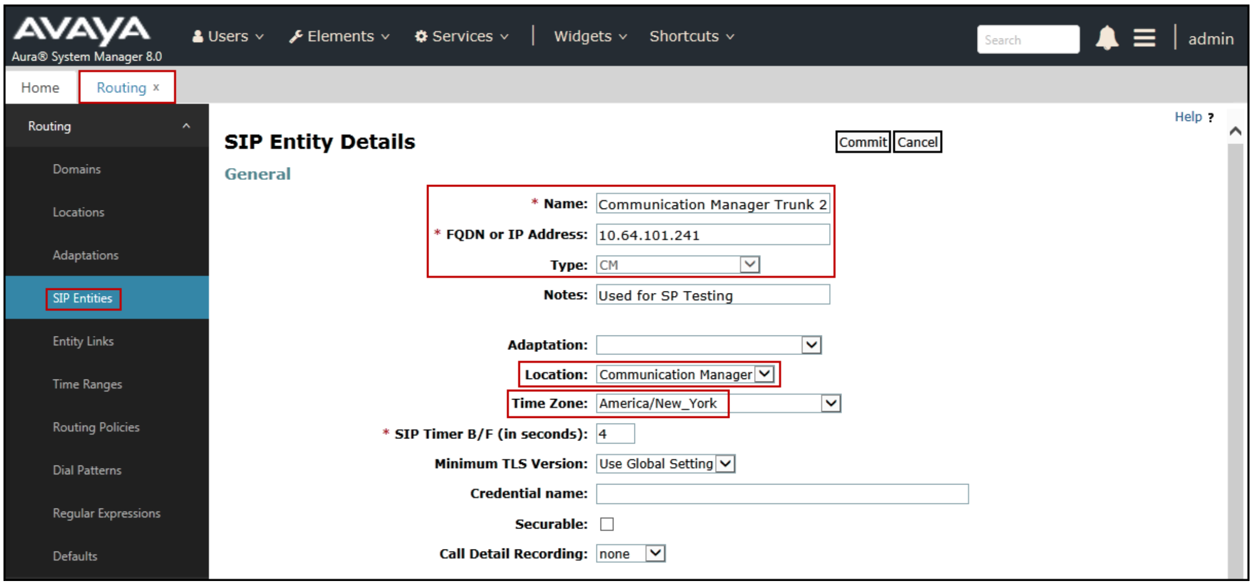
A SIP Entity must be added for Session Manager and for each SIP telephony system connected to it, which includes Communication Manager, Avaya SBCE and Experience Portal. Navigate to **Routing → SIP Entities** in the left navigation pane and click on the **New** button in the right pane (not shown). In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

- **Name:** Enter a descriptive name.
- **FQDN or IP Address:** Enter the FQDN or IP address of the SIP Entity that is used for SIP signaling (see **Figure 1**).
- **Type:** Select ***Session Manager*** for Session Manager, ***CM*** for Communication Manager, ***SIP Trunk*** (or ***Other***) for the Avaya SBCE and ***Voice Portal*** for the Experience Portal.
- **Adaptation:** This field is only present if **Type** is not set to **Session Manager** If Adaptations were to be created, here is where they would be applied to the entity.
- **Location:** Select the location that applies to the SIP Entity being created, defined in **Section 7.3**.
- **Time Zone:** Select the time zone for the location above.
- Click **Commit** to save.

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

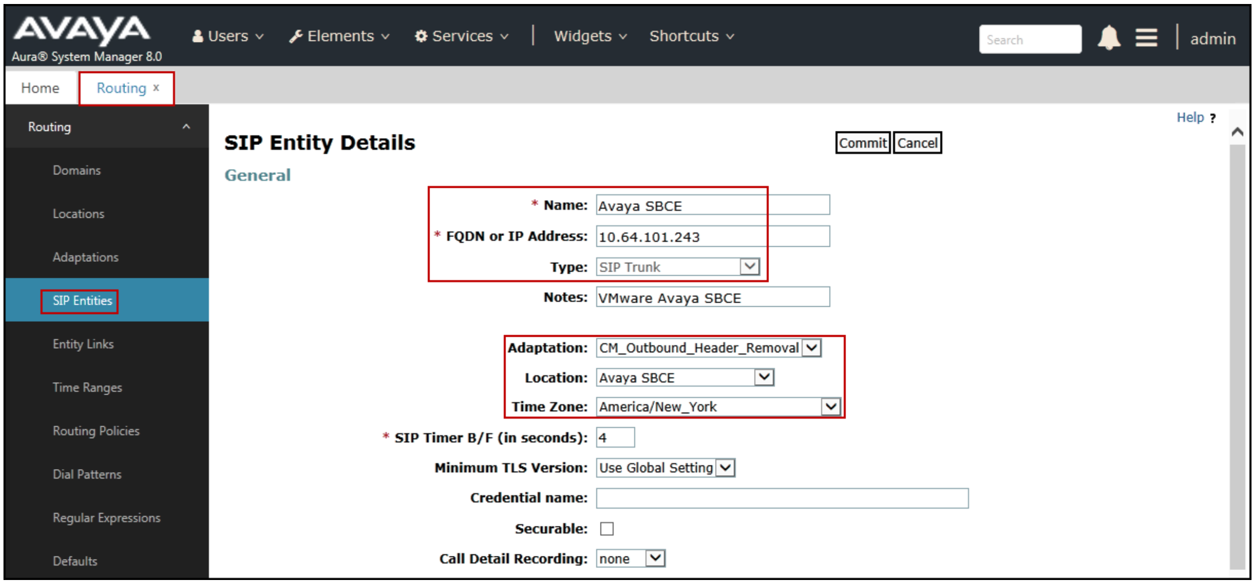


The following screen shows the addition of the ***Communication Manager Trunk 2*** SIP Entity for Communication Manager. In order for Session Manager to send SIP service provider traffic on a separate entity link to Communication Manager, the creation of a separate SIP entity for Communication Manager is required. This SIP Entity should be different than the one created during the Session Manager installation, used by all other enterprise SIP traffic. The **FQDN or IP Address** field is set to the IP address of the "**procr**" interface in Communication Manager, as seen in **Section 5.3**. Select the location that applies to the SIP Entity being created, defined in **Section 7.3**. Select the **Time Zone**.



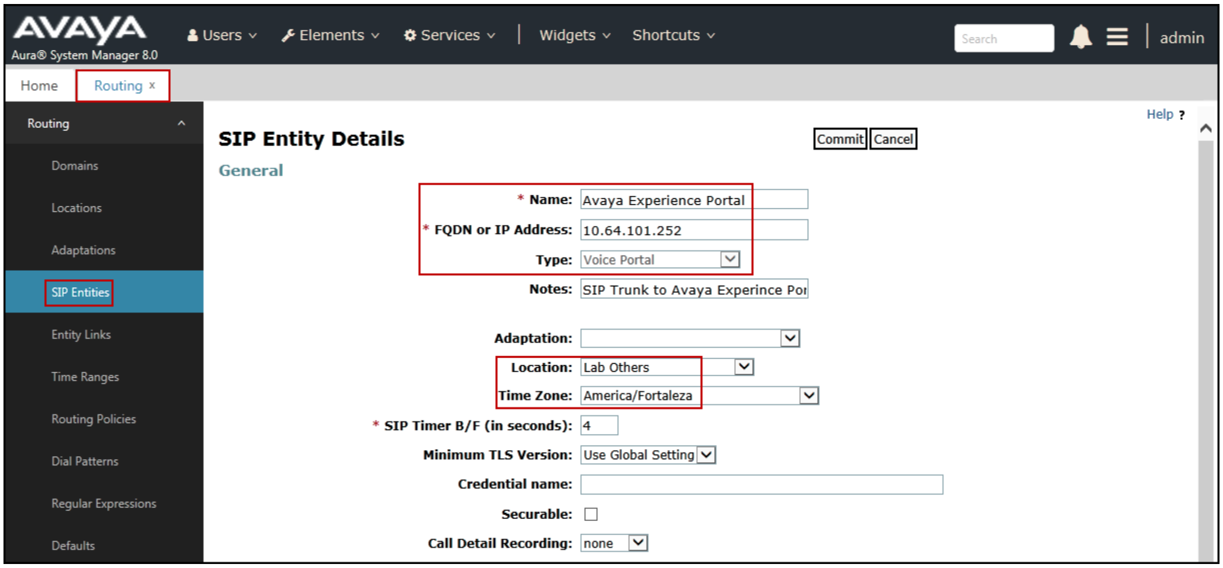
The following screen shows the addition of the ***Avaya SBCE*** SIP Entity for the Avaya SBCE:

- The **FQDN or IP Address** field is set to the IP address of the SBC private network interface (see **Figure 1**).
- On the **Adaptation** field, the adaptation module ***CM_Outbound_Header_Removal*** previously defined in **Section 7.4** was selected.
- Select the location that applies to the SIP Entity being created, defined in **Section 7.3**.
- Select the **Time Zone**.



The following screen shows the addition of the ***Avaya Experience Portal*** SIP Entity:

- The **FQDN or IP Address** field is set to the IP address of the Experience Portal (see **Figure 1**).
- Select the location that applies to the SIP Entity being created, defined in **Section 7.3**.
- Select the **Time Zone**.

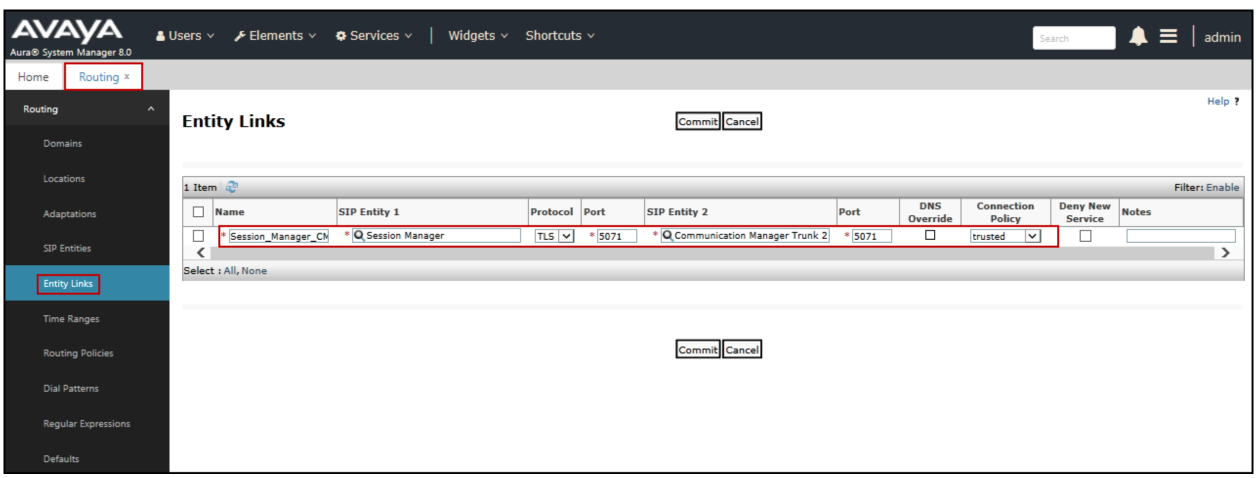


7.6. Entity Links

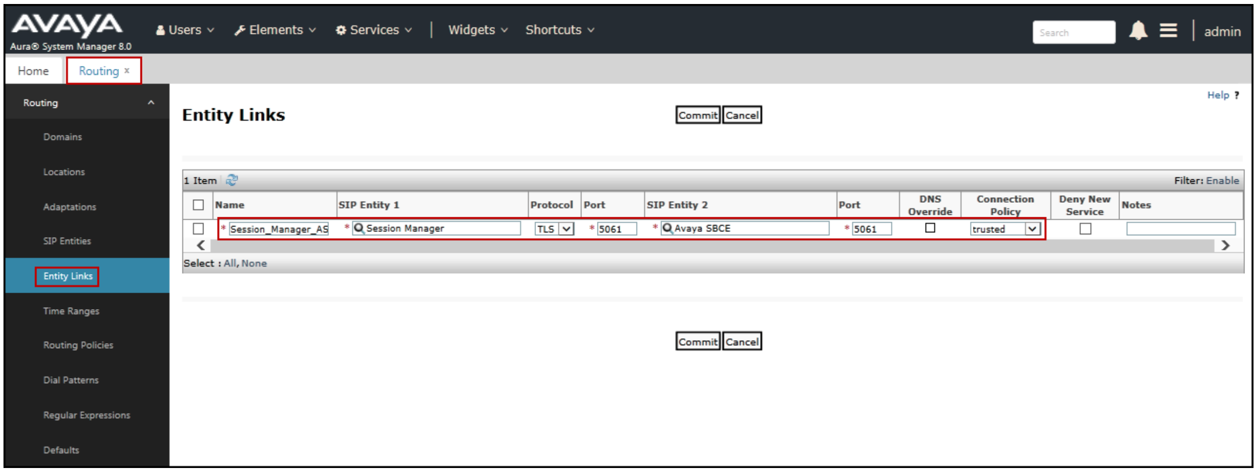
A SIP trunk between Session Manager and a telephony system is described by an Entity Link. Three Entity Links were created; an entity link to Communication Manager for use only by service provider traffic, an entity link to the Avaya SBCE and an entity link to Experience Portal. To add an Entity Link,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

- **Name:** Enter a descriptive name.
- **SIP Entity 1:** Select the Session Manager from the drop-down menu (**Section 7.5**).
- **Protocol:** Select the transport protocol used for this link (**Section 5.6**).
- **Port:** Port number on which Session Manager will receive SIP requests from the far-end (**Section 5.6**).
- **SIP Entity 2:** Select the name of the other system from the drop-down menu (**Section 7.5**).
- **Port:** Port number on which the other system receives SIP requests from Session Manager (**Section 5.6**).
- **Connection Policy:** Select **Trusted** to allow calls from the associated SIP Entity.
- Click **Commit** to save.

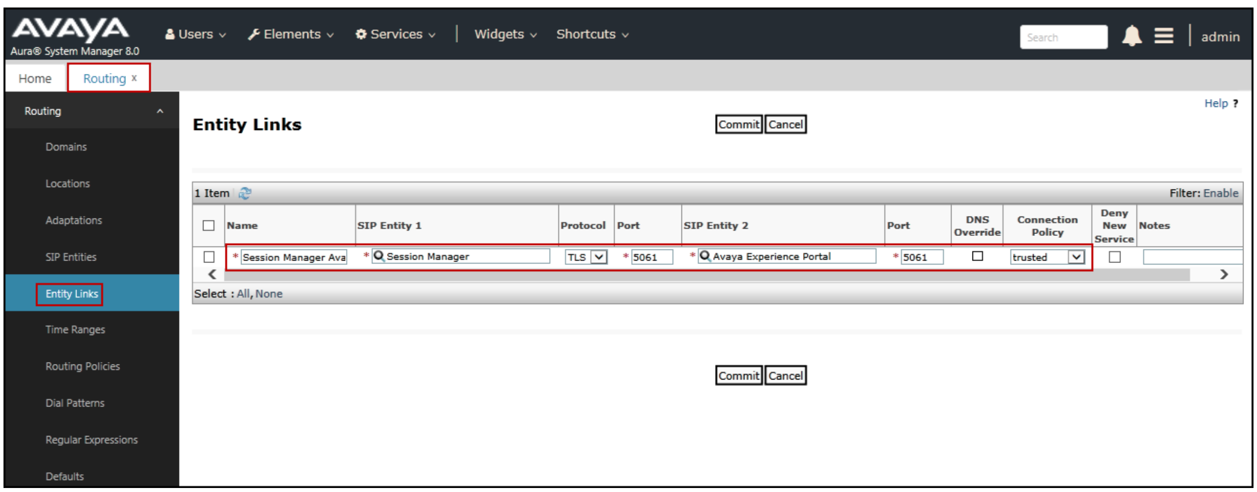
The screen below shows the Entity Link to Communication Manager. The protocol and ports defined here must match the values used on the Communication Manager signaling group form in **Section 5.6**. ***TLS*** transport and port ***5071*** were used.



The Entity Link to the Avaya SBCE is shown below; ***TLS*** transport and port ***5061*** were used.



The Entity Link to the Experience Portal is shown below; ***TLS*** transport and port ***5061*** were used.

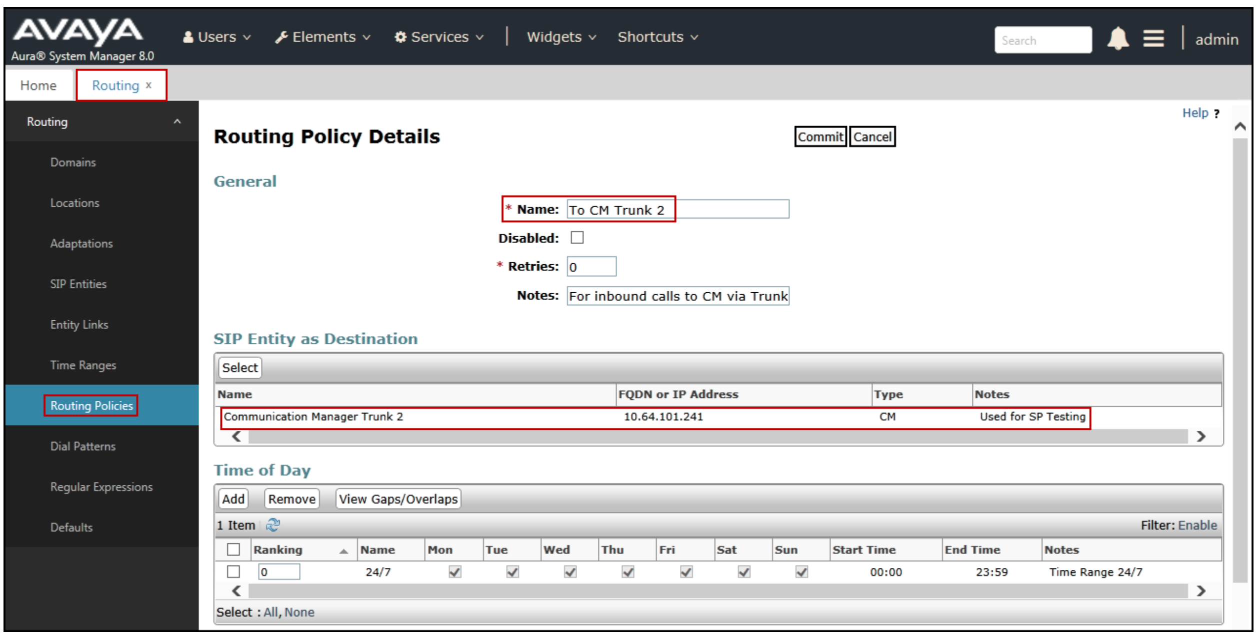


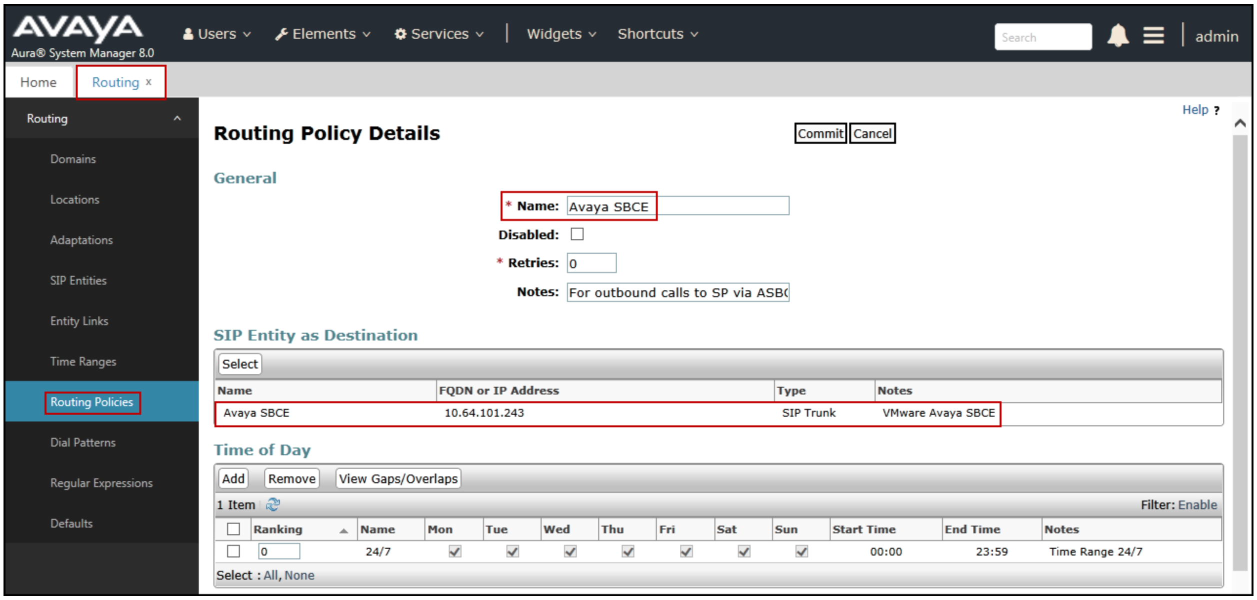
7.7. Routing Policies

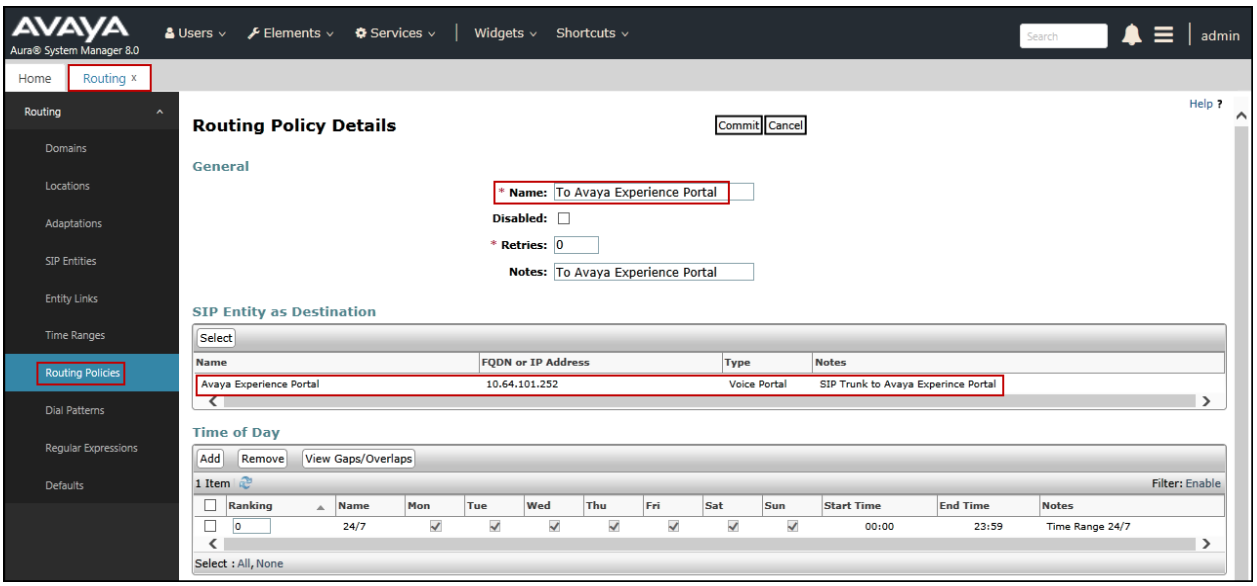
Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 7.5**. Three routing policies were added; an incoming policy with Communication Manager as the destination, an outbound policy with the Avaya SBCE as the destination and an incoming policy with Experience Portal as the destination. To add a routing policy, navigate to **Routing → Routing Policies** in the left navigation pane and click on the **New** button in the right pane (not shown). The following screen is displayed:

- In the **General** section, enter a descriptive **Name** and add a brief description under **Notes** (optional).
- In the **SIP Entity as Destination** section, click **Select**. The **SIP Entity List** page opens (not shown). Choose the appropriate SIP entity to which this routing policy applies (**Section 7.5**) and click **Select**. The selected SIP Entity displays on the **Routing Policy Details** page as shown below.
- Use default values for remaining fields.
- Click **Commit** to save.

The following screens show the Routing Policies for Communication Manager, the Avaya SBCE and the Experience Portal.







7.8. Dial Patterns

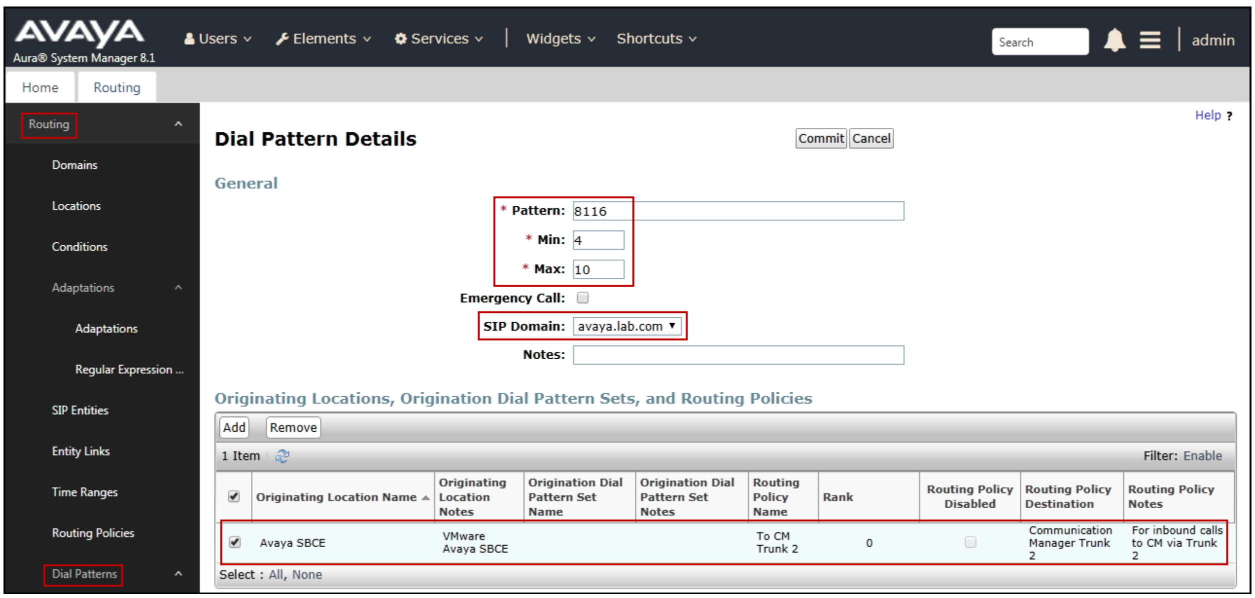
Dial Patterns are needed to route specific calls through Session Manager. For the compliance test, dial patterns were needed to route calls from Communication Manager to the service provider and vice versa. Also, a dial pattern was created to route calls from service provider to Experience Portal. Dial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

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

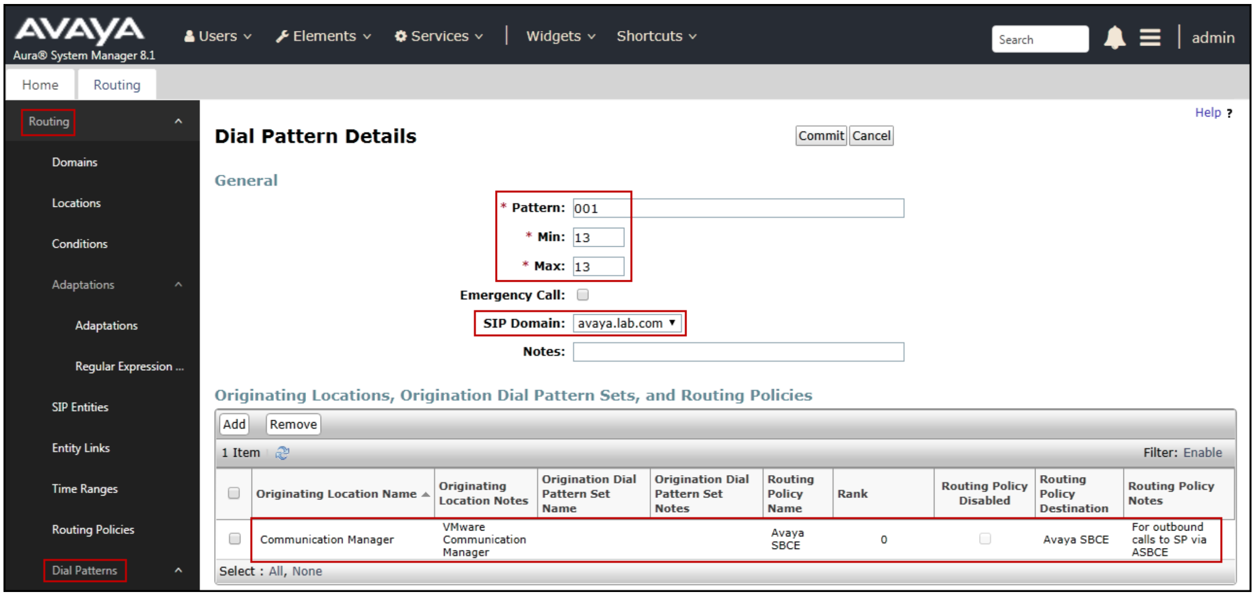
- **Pattern:** Enter a dial string that will be matched against the Request-URI of the call.

- **Min:** Enter a minimum length used in the match criteria.
- **Max:** Enter a maximum length used in the match criteria.
- **SIP Domain:** Enter the destination domain used in the match criteria, or select "**ALL**" to route incoming calls to all SIP domains.
- **Notes:** Add a brief description (optional).
- In the **Originating Locations and Routing Policies** section, click **Add**. From the **Originating Locations and Routing Policy List** that appears (not shown), select the appropriate originating location for use in the match criteria (**Section 7.3**).
- Lastly, select the routing policy from the list that will be used to route all calls that match the specified criteria (**Section 7.7**). Click **Select** (not shown).
- Click **Commit** to save.

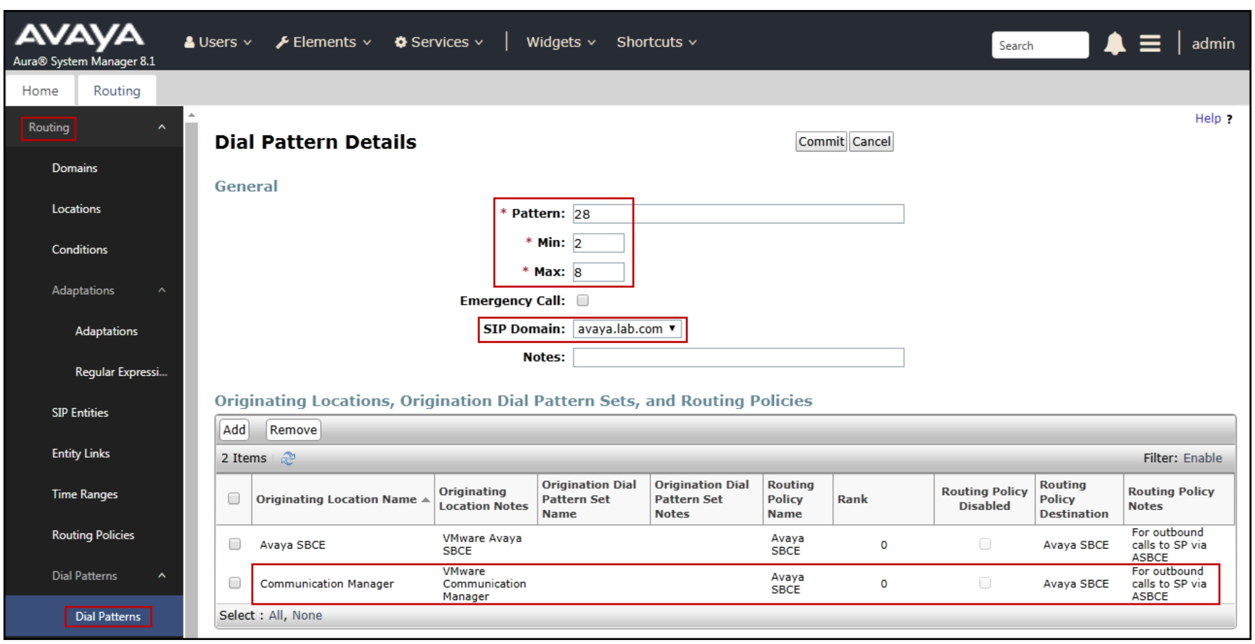
The following screen illustrates an example dial pattern used to verify inbound PSTN calls to Communication Manager. In the example, calls to 10-digit numbers starting with ***8116***, arriving from location ***Avaya SBCE***, used route policy ***To CM Trunk 2*** to Communication Manager. The SIP Domain was set to ***avaya.lab.com***.



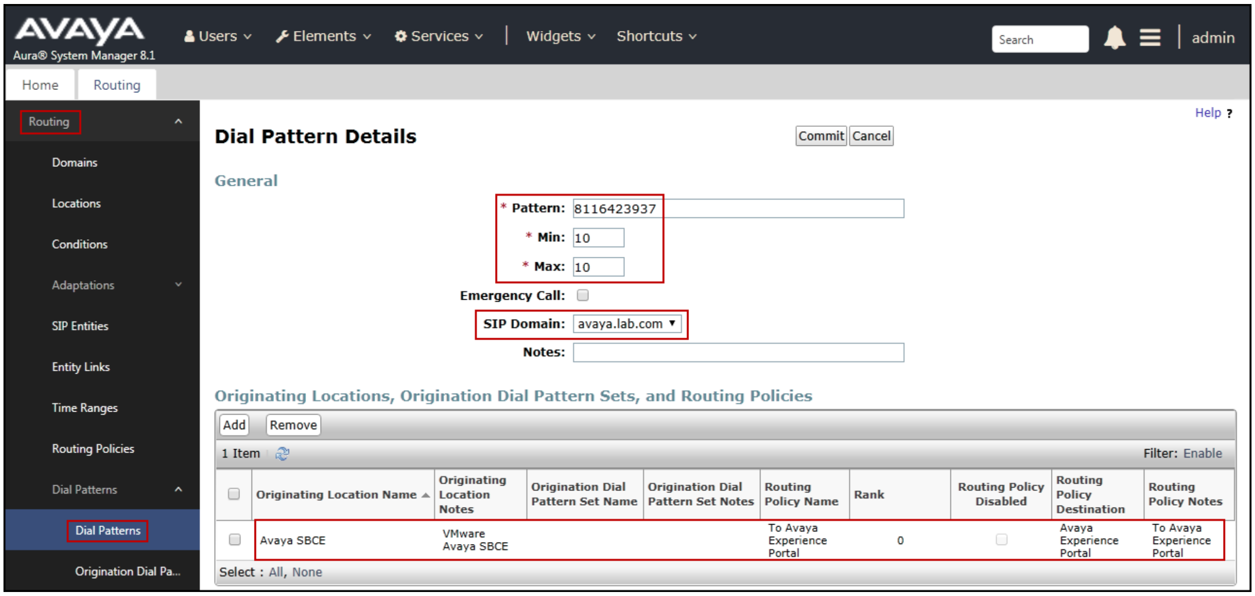
The example in this screen shows the 13-digit dialed numbers for international calls to the U.S., beginning with ***001***, arriving from the ***Communication Manager*** location, will use route policy ***Avaya SBCE***, which sends the call out to the PSTN via the Avaya SBCE and the service provider SIP trunk. The SIP Domain was set to ***avaya.lab.com***.



The example in this screen shows the 8-digit dialed numbers for calls within Mexico, beginning with ***28***, arriving from the ***Communication Manager*** location, will use route policy ***Avaya SBCE***, which sends the call out to the PSTN (Mexico) via the Avaya SBCE and the service provider SIP trunk. The SIP Domain was set to ***avaya.lab.com***.



The following screen illustrates an example dial pattern used to verify inbound PSTN calls to Experience Portal. In the sample configuration one of the DID numbers provided by Alestra was used as a test number to route calls from the PSTN to Experience Portal, arriving from location ***Avaya SBCE***, used routing policy ***To Avaya Experience Portal***. The SIP Domain was set to ***avaya.lab.com***.



Repeat the above procedures as needed to define additional dial patterns.

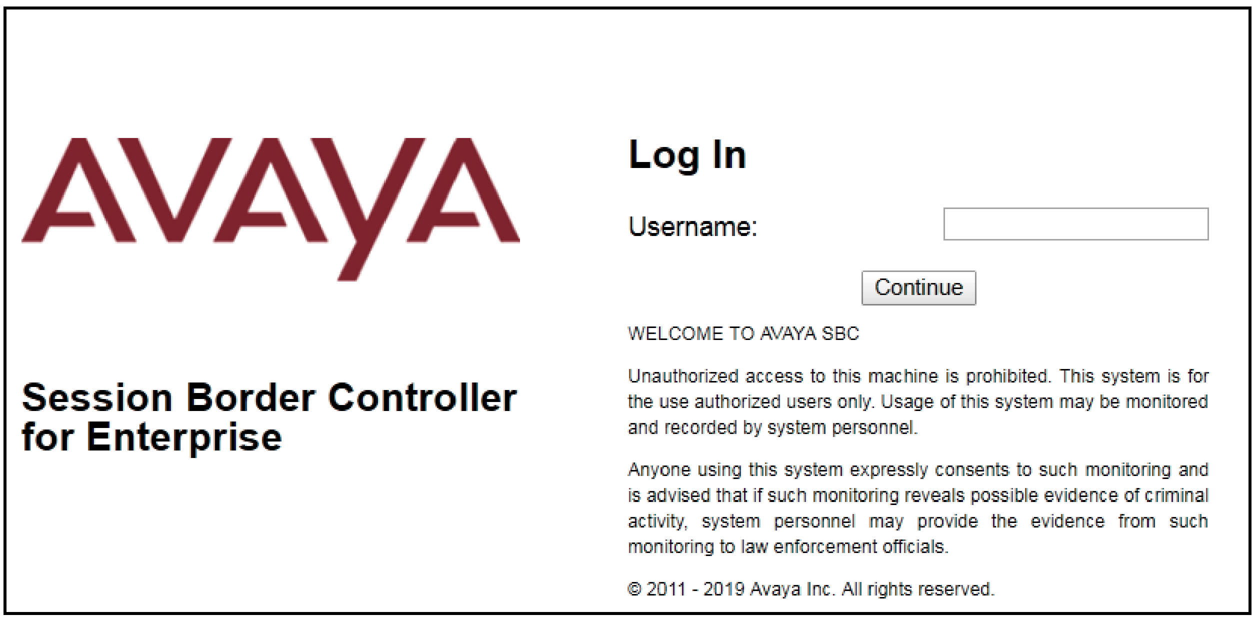
8. Configure Avaya Session Border Controller for Enterprise

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

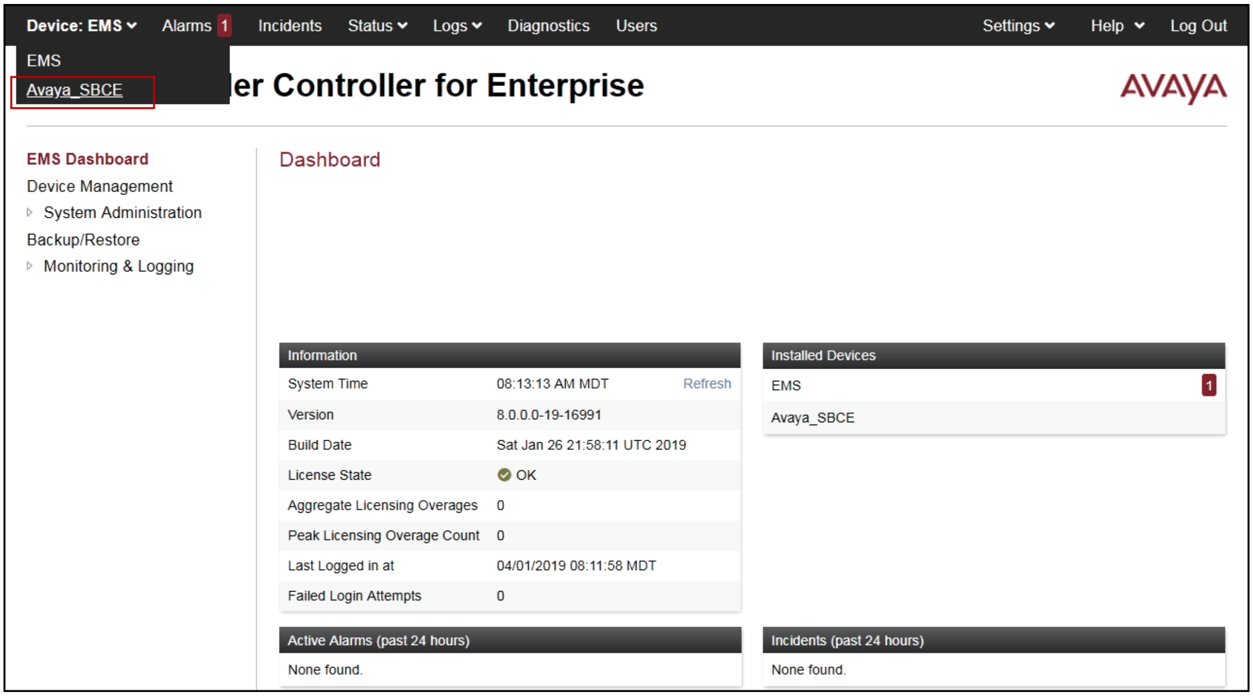
Note - The configuration tasks required to support TLS transport for signaling and SRTP for media are beyond the scope of these Application Notes; hence it's not discussed in detail in this document. Consult reference [**8**] in the **References** section for additional information on this topic.

8.1. System Access

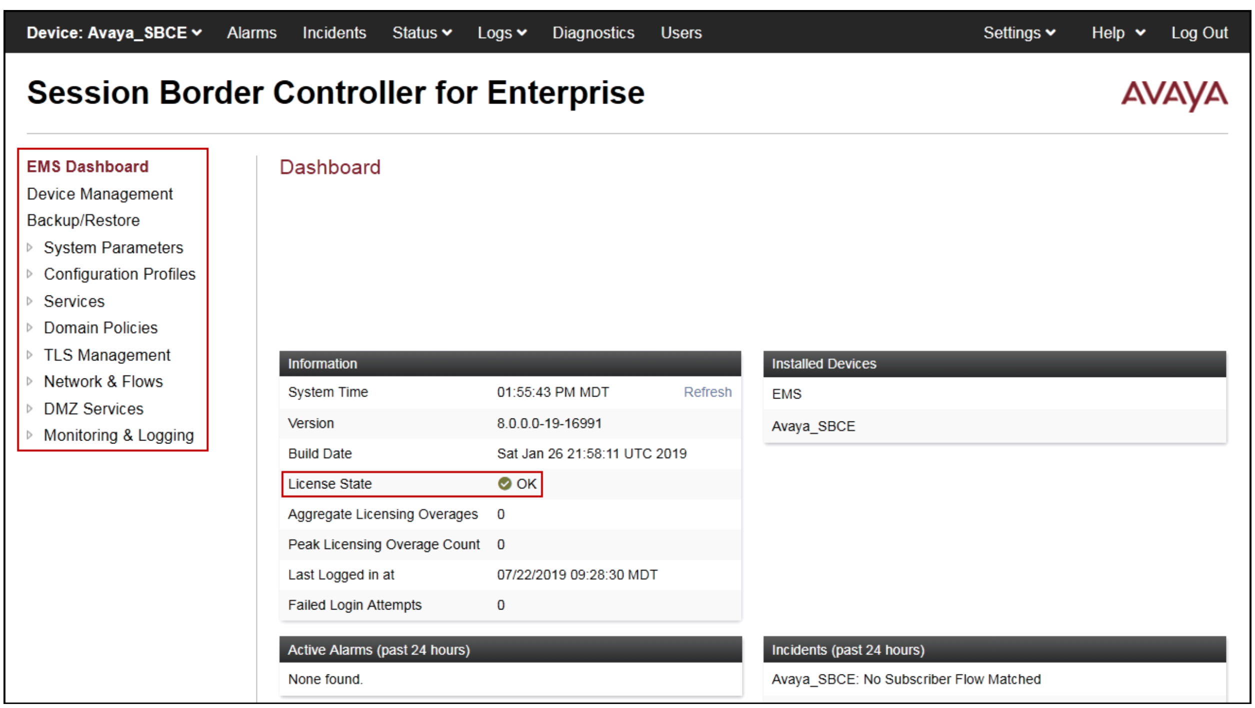
Access the Session Border Controller web management interface by using a web browser and entering the URL **https://<ip-address>**, where **<ip-address>** is the management IP address configured at installation. Log in using the appropriate credentials.



Once logged in, on the top left of the screen, under **Device:** select the device being managed, ***Avaya_SBCE*** in the sample configuration.

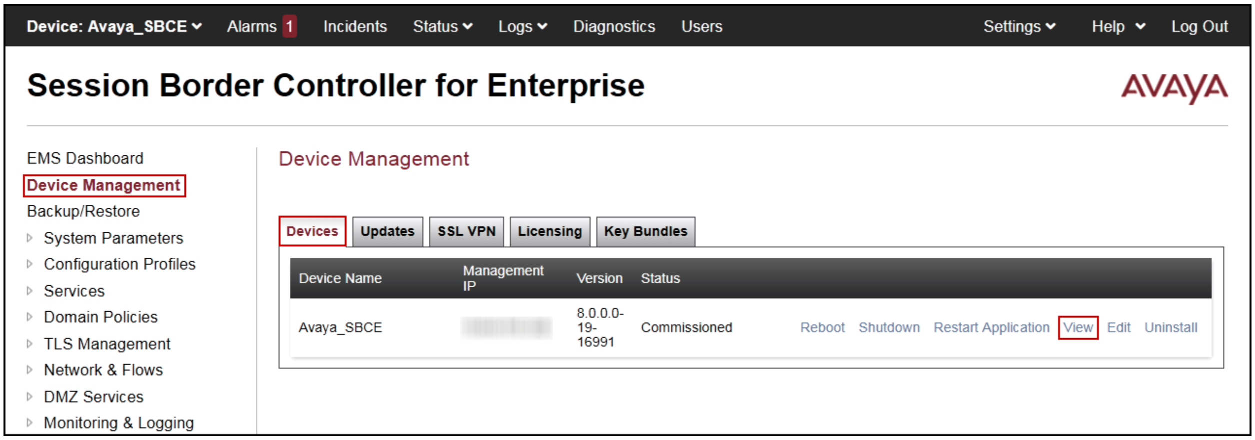


The left navigation pane contains the different available menu items used for the configuration of the Avaya SBCE. Verify that the status of the **License State** field is **OK**, indicating that a valid license is present. Contact an authorized Avaya sales representative if a license is needed.

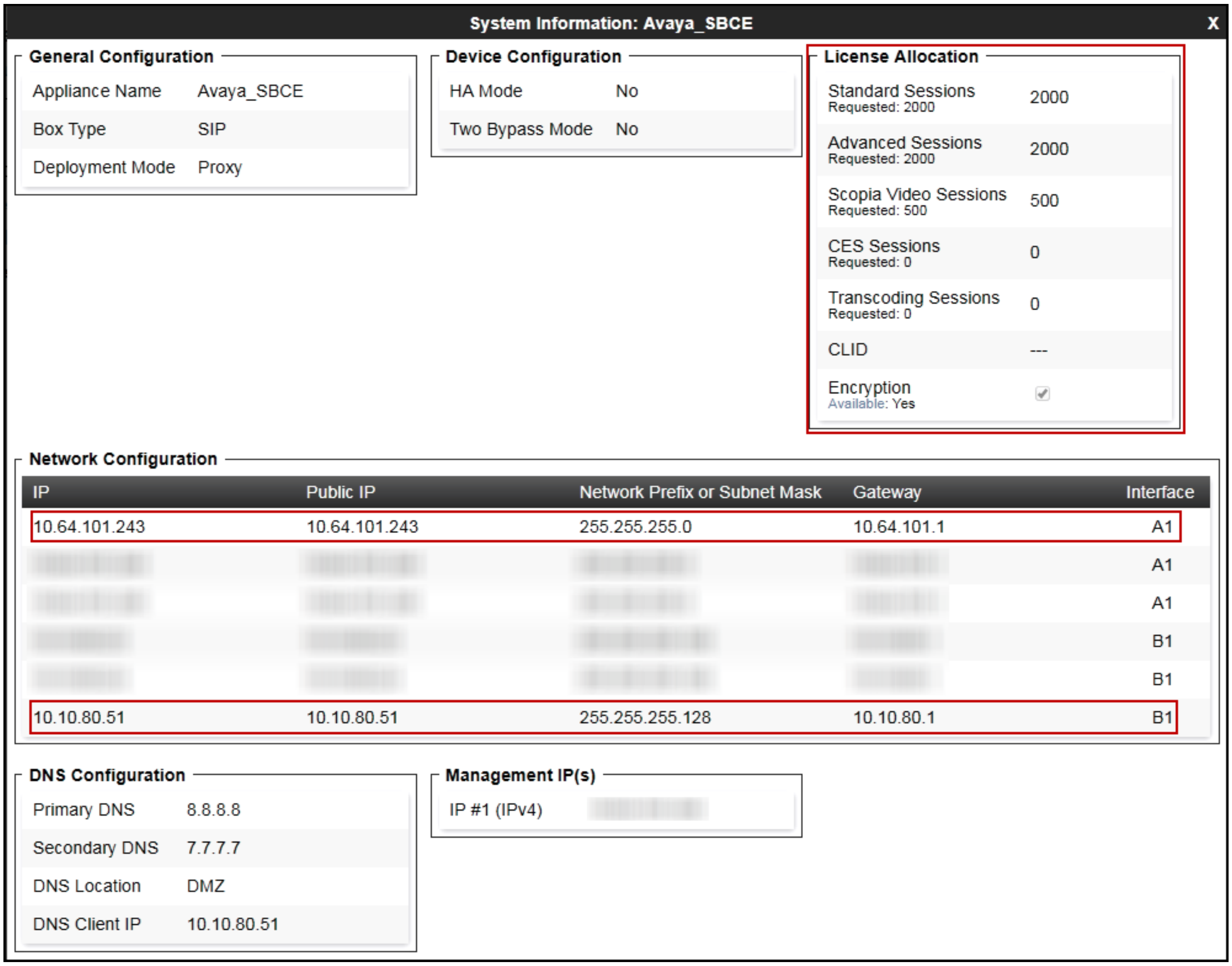


8.2. Device Management

To view current system information, select **Device Management** on the left navigation pane. In the reference configuration, the device named ***Avaya_SBCE*** is shown. The management IP address that was configured during installation is blurred out for security reasons, the current software version is shown. The management IP address needs to be on a subnet separate from the ones used in all other interfaces of the Avaya SBCE, segmented from all VoIP traffic. Verify that the **Status** is ***Commissioned***, indicating that the initial installation process of the device has been previously completed, as shown on the screen below.



To view the network configuration assigned to the Avaya SBCE, click **View** on the screen above. The **System Information** window is displayed, containing the current device configuration and network settings.



The highlighted IP addresses in the **System Information** screen shown above are the ones used for the SIP trunk to Alestra and are the ones relevant to these Application Notes. Other IP addresses assigned to the Avaya SBCE **A1** and **B1** interfaces are used to support remote workers and other SIP trunks, and they are not discussed in this document. Also note that for security purposes, any public IP addresses used during the compliance test have been masked in this document.

In the reference configuration, the private interface of the Avaya SBCE (10.64.101.243) was used to connect to the enterprise network, while its public interface (10.10.80.51) was used to connect to the public network. See **Figure 1**.

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

8.3. TLS Management

Transport Layer Security (TLS) is a standard protocol that is used extensively to provide a secure channel by encrypting communications over IP networks. It enables clients to authenticate servers or, optionally, servers to authenticate clients. UC-Sec security products utilize TLS primarily to facilitate secure communications with remote servers.

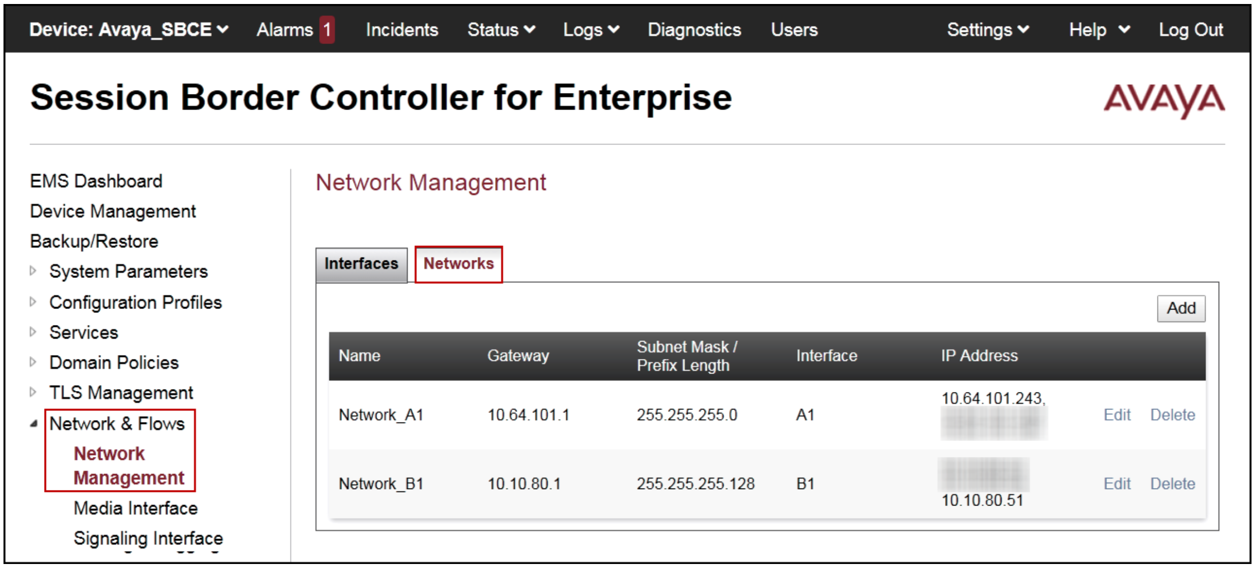
It is assumed that generation and installation of certificates and the creation of TLS Profiles on the Avaya SBCE have been previously completed, as it's not discussed in this document. Refer to item [**8**] in **Section 12**.

8.4. Network Management

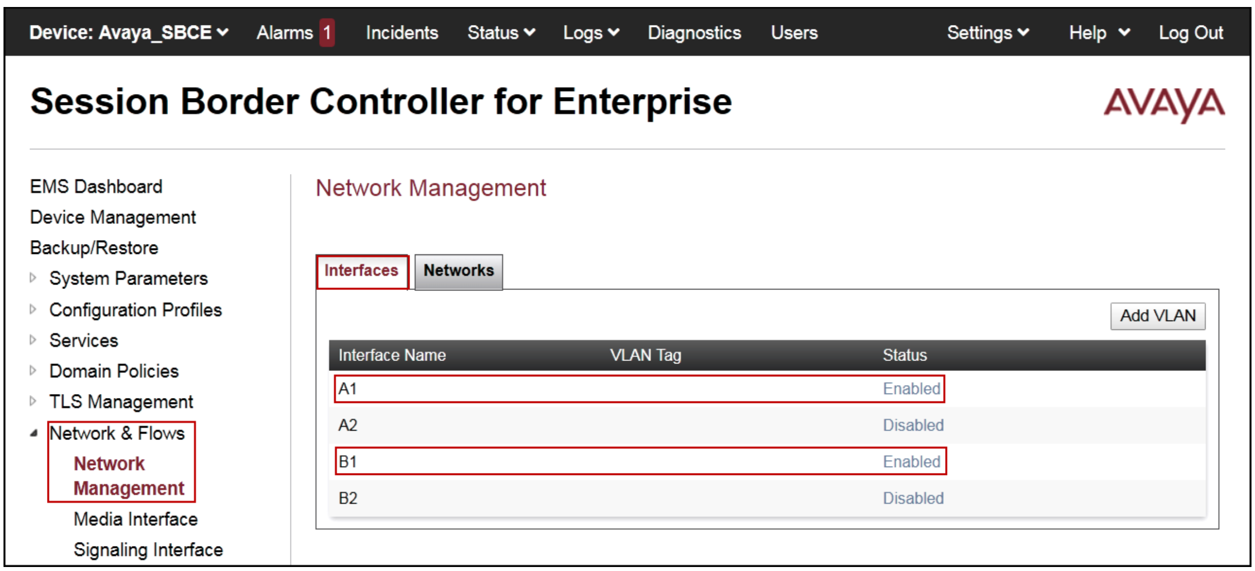
The network configuration parameters should have been previously specified during installation of the Avaya SBCE. In the event that changes need to be made to the network configuration, they can be entered here.

Select **Network Management** from the **Network & Flows** on the left-side menu. On the **Networks** tab, verify or enter the network information as needed.

Note that in the configuration used during the compliance test, the IP addresses assigned to the private (***10.64.101.243***) and public (***10.10.80.51***) sides of the Avaya SBCE are the ones relevant to these Application Notes.



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



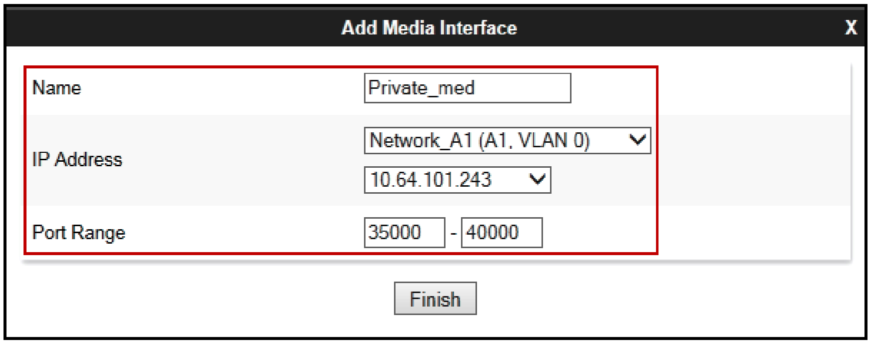
8.5. Media Interfaces

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

To add the Media Interface in the enterprise direction, select **Media Interface** from the **Network & Flows** menu on the left-hand side, click the **Add** button (not shown).

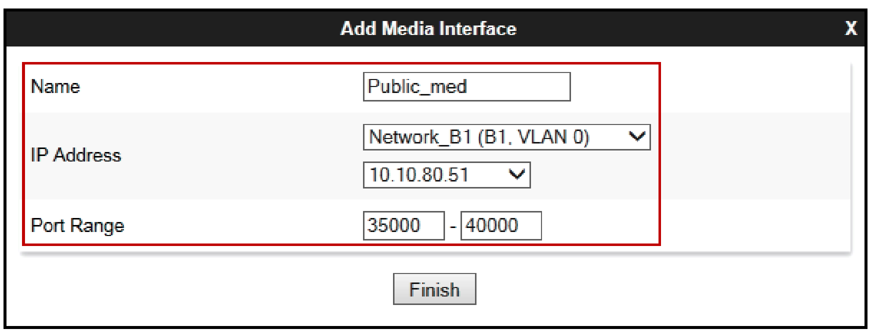
- On the **Add Media Interface** screen, enter an appropriate **Name** for the Media Interface.

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



A Media Interface facing the public side was similarly created with the name ***Public_med***, as shown below.

- Under **IP Address**, the network and IP address to be associated with this interface was selected.
- The **Port Range** was left at the default values.
- Click **Finish**.

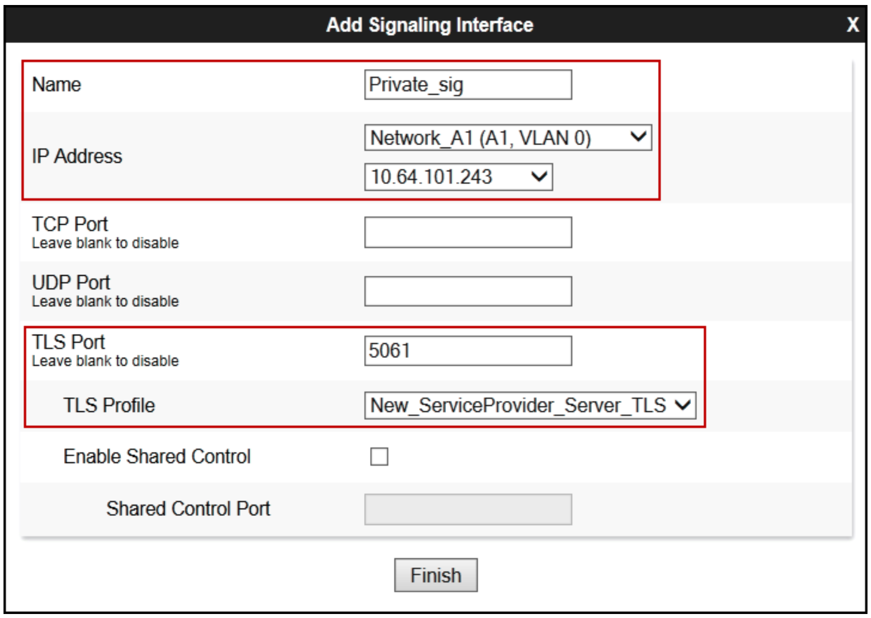


8.6. Signaling Interfaces

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

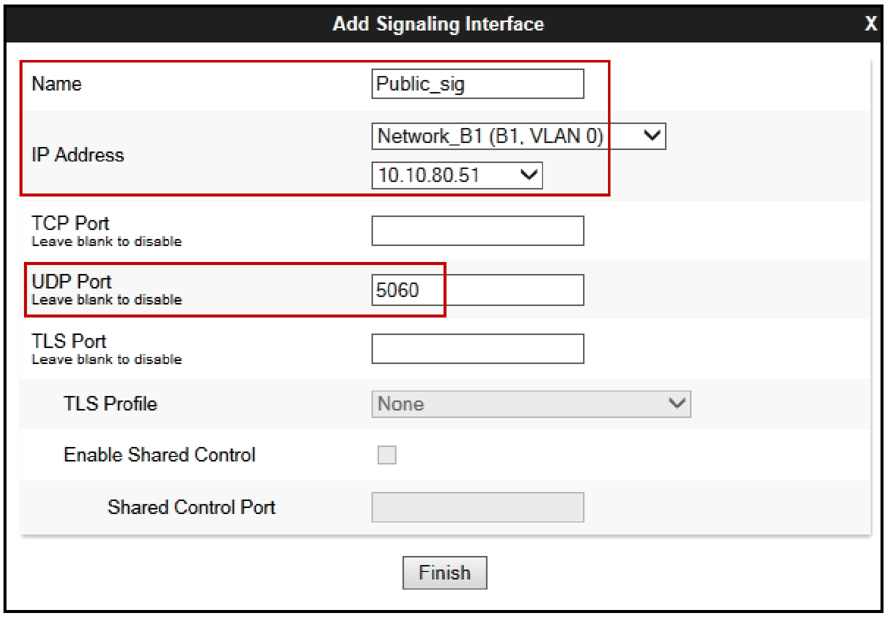
To add the Signaling Interface in the enterprise direction, select **Signaling Interface** from the **Network & Flows** menu on the left-hand side, click the **Add** button (not shown).

- On the **Add Signaling Interface** screen, enter an appropriate **Name** for the interface.
- Under **IP Address**, select from the drop-down menus the network and IP address to be associated with this interface.
- Enter ***5061*** for **TLS Port**, since TLS port 5061 is used to listen for signaling traffic from Session Manager in the sample configuration, as defined in **Section 7.6**.
- Select a **TLS Profile**.
- Click **Finish**.



A second Signaling Interface with the name ***Public_sig*** was similarly created in the service provider's direction.

- Under **IP Address**, select from the drop-down menus the network and IP address to be associated with this interface.
- Enter ***5060*** for **UDP Port**, since UDP port 5060 is used to listen for signaling traffic from Alestra in the sample configuration.
- Click **Finish**.

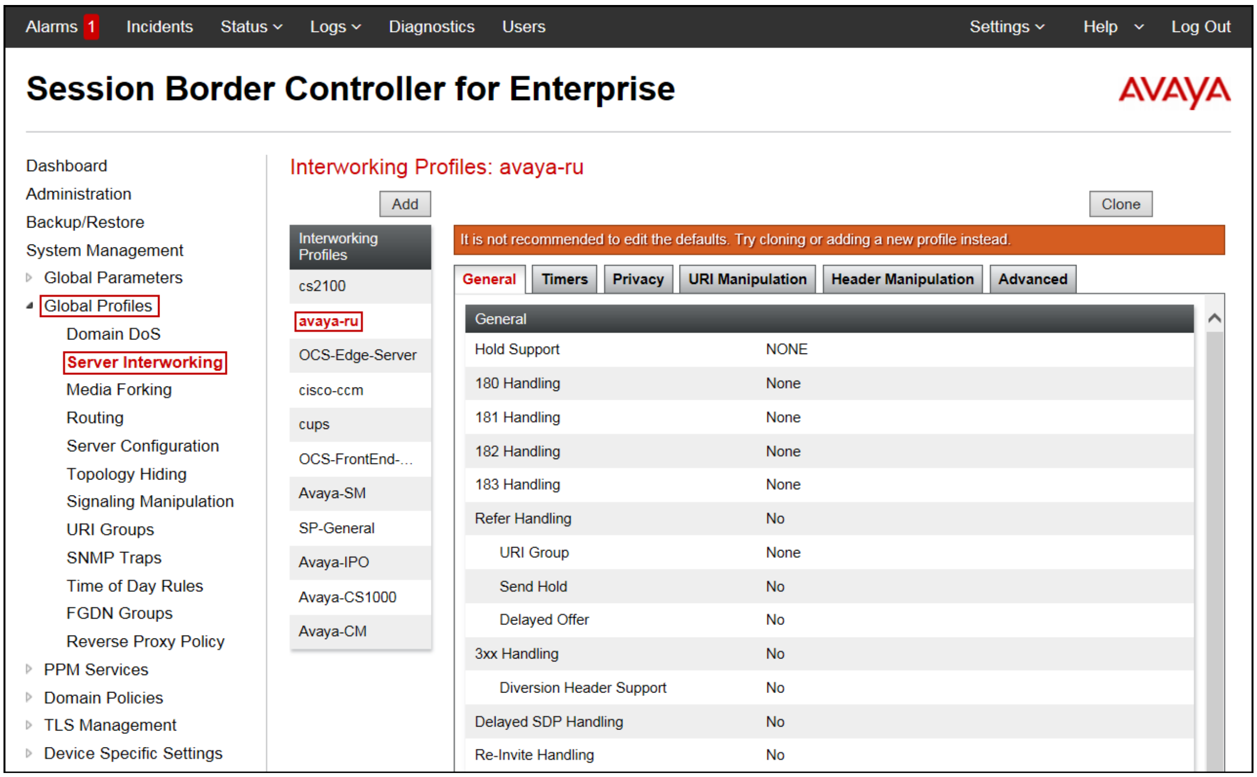


8.7. Server Interworking

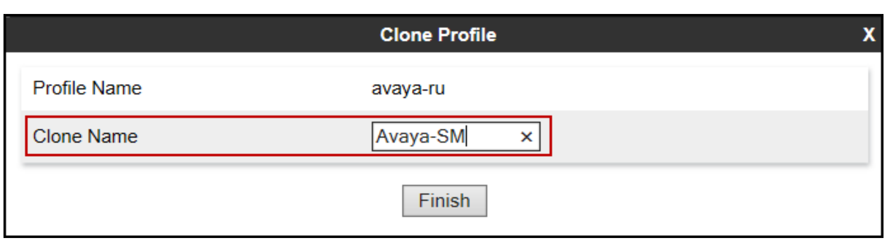
Interworking Profile features are configured to facilitate the interoperability between the enterprise SIP-enabled solution (Call Server) and the SIP trunk service provider (Trunk Server).

8.7.1. Server Interworking Profile – Enterprise

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

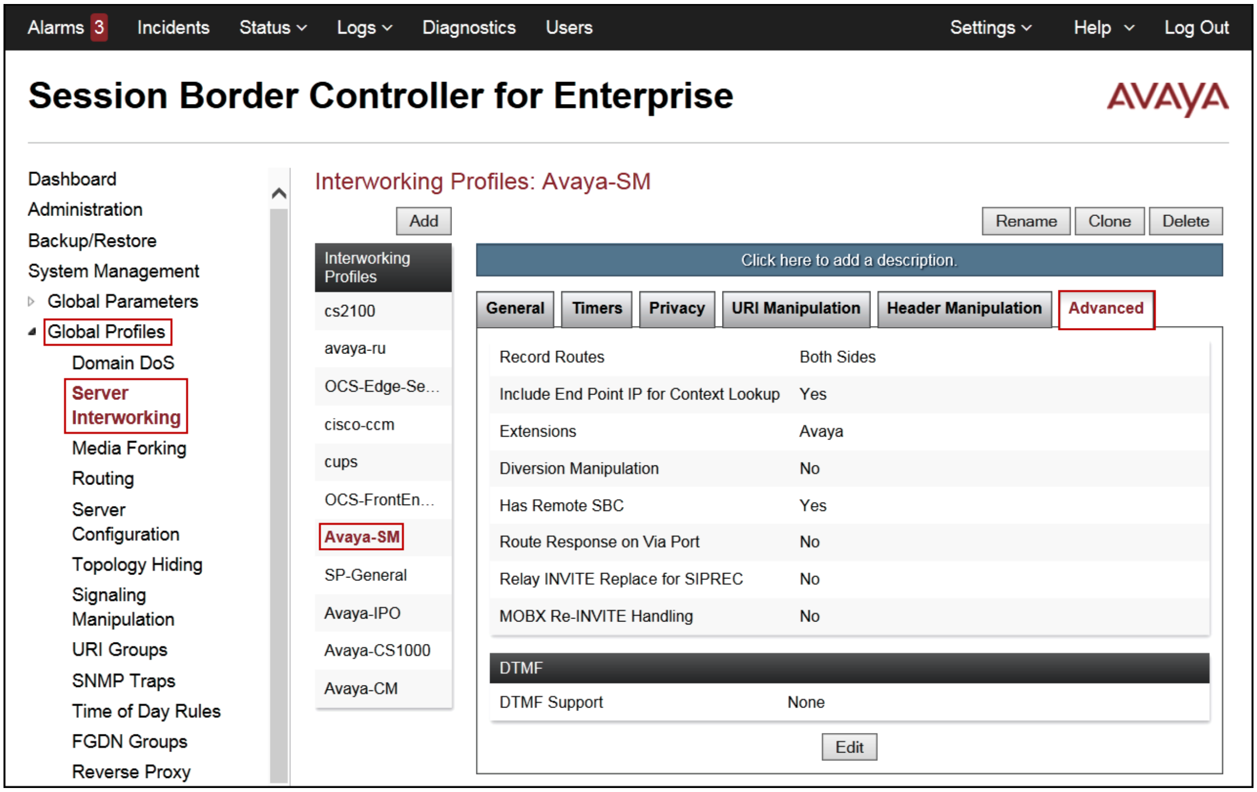


- Enter a descriptive name for the cloned profile.
- Click **Finish**.



The **Timers**, **Privacy**, **URI Manipulation** and **Header Manipulation** tabs contain no entries.

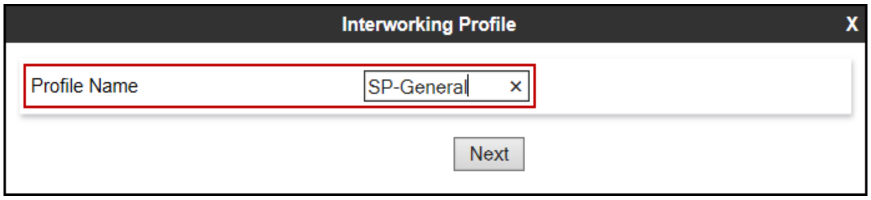
The **Advanced** tab settings are shown on the screen below:



8.7.2. Server Interworking Profile – Service Provider

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

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



- Click **Next** until the last tab is reached then click **Finish** on the last tab leaving remaining fields with default values (not shown).

8.8. 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 reference **[8]** in the **References** section for more information on this topic.

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

- Change the Max-Forwards in INVITES SIP messages received from Alestra from 9 to 69.
- Insert the Pilot number associated with the SIP trunk in the “From” and “Contact” headers of SIP messages sent to Alestra.
- Remove unwanted “gsid” and “epv” parameter from being sent to Alestra in the Contact header.
- Remove the P-Location parameter from being sent to Alestra.
- Change the Diversion header scheme from SIPS to SIP in SIP messages sent to Alestra.
- Remove unwanted xml element information from the SDP in SIP messages sent to Alestra.

The scripts will later be applied to the Server Configuration Profiles corresponding to Session Manager in **Section 8.9.1** and the Service Provider (toward Alestra) in **Section 8.9.2**.

To create the SigMa script to be applied to the Server Configuration Profile corresponding to Session Manager, on the left navigation pane, select **Configuration Profiles → Signaling Manipulation**. From the **Signaling Manipulation Scripts** list, select **Add**.

- For **Title** enter a name, the name ***Change Max-Forward*** was chosen in this example.
- Copy and paste the script shown below or from Appendix A.
- Click **Save**.

```
within session "INVITE"
{
   act on request where %DIRECTION="OUTBOUND" and
%ENTRY_POINT="POST_ROUTING"
   {
       if (exists(%HEADERS["Max-Forwards"][1])) then
       {
```

```
        %HEADERS["Max-Forwards"][1] = "69";
      }

  }
}
```

To create the SigMa script to be applied to the Server Configuration Profile corresponding to the Service Provider (Alestra), on the left navigation pane, select **Configuration Profiles → Signaling Manipulation**. From the **Signaling Manipulation Scripts** list, select **Add**.

- For **Title** enter a name, the name ***AlestraSigma*** was chosen in this example.
- Copy and paste the script shown below or from Appendix A.
- Click **Save**.

```
//Insert the Pilot number associated with the SIP Trunk in the FROM and CONTACT headers of
Outbound calls.
within session "ALL"

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

%fromuser = %HEADERS["From"][1].URI.USER;
%HEADERS["From"][1].URI.USER = "8116421111";

%contact = %HEADERS["Contact"][1].URI.USER;
%HEADERS["Contact"][1].URI.USER = "8116421111";

//Remove gsid and epv parameters from Contact header.
remove(%HEADERS["Contact"][1].URI.PARAMS["gsid"]);
remove(%HEADERS["Contact"][1].URI.PARAMS["epv"]);

//Remove P-Location parameter.
remove(%HEADERS["P-Location"][1]);

//Changes the Diversion header scheme from SIPS to SIP.
%HEADERS["Diversion"][1].regex_replace("sips","sip");

//Remove unwanted xml element information from the SDP in SIP messages sent to the Service
Provider.
remove(%BODY[1]);

}
```

}

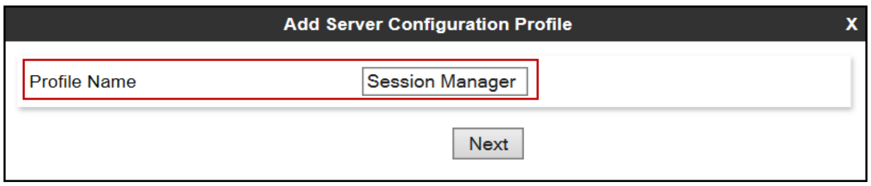
8.9. Server Configuration

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

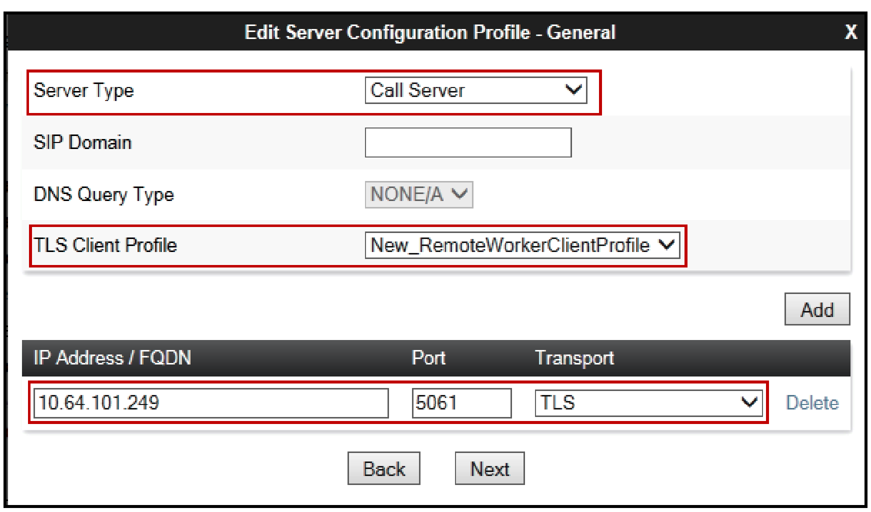
8.9.1. Server Configuration Profile – Enterprise

From the **Services** menu on the left-hand navigation pane, select **SIP Servers** and click the **Add** button (not shown) to add a new profile for the Call Server.

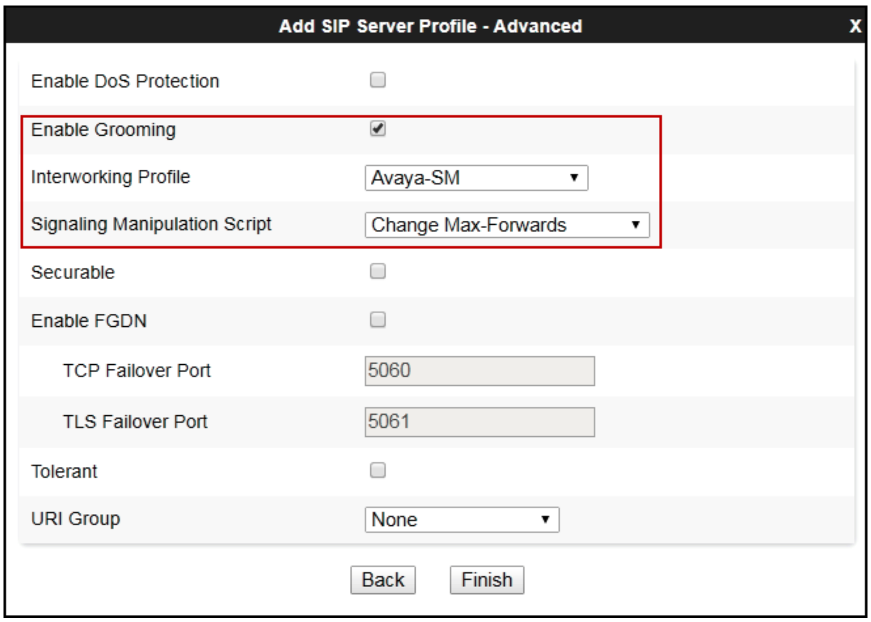
- Enter an appropriate **Profile Name** similar to the screen below.
- Click **Next**.



- On the **Edit SIP Server Profile – General** tab select ***Call Server*** from the drop-down menu under the **Server Type**.
- On the **IP Addresses / FQDN** field**,** enter the IP address of the Session Manager Security Module (**Section 7.5)**.
- Enter ***5061*** under **Port** and select ***TLS*** for **Transport**. The transport protocol and port selected here must match the values defined for the Entity Link to the Session Manager previously created in **Section 7.6**.
- Select a **TLS Profile**.
- Click **Next**.



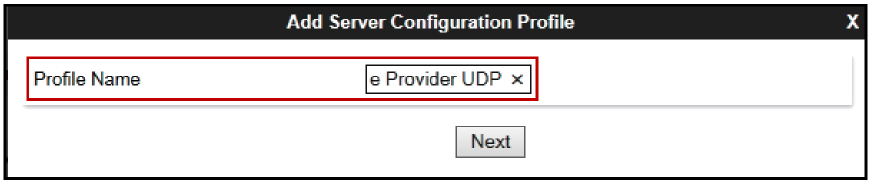
- Click **Next** until the **Add Server Configuration Profile – Advanced** tab is reached (not shown).
- On the **Add Server Configuration Profile – Advanced** tab:
  - Check ***Enable Grooming***.
  - Select ***Avaya-SM*** from the **Interworking Profile** drop-down menu (**Section 8.7.1**).
  - Select ***Change Max-Forwards*** from the **Signaling Manipulation Script** drop down menu (**Sections 8.8** and **Section 13**).
- Click **Finish**.



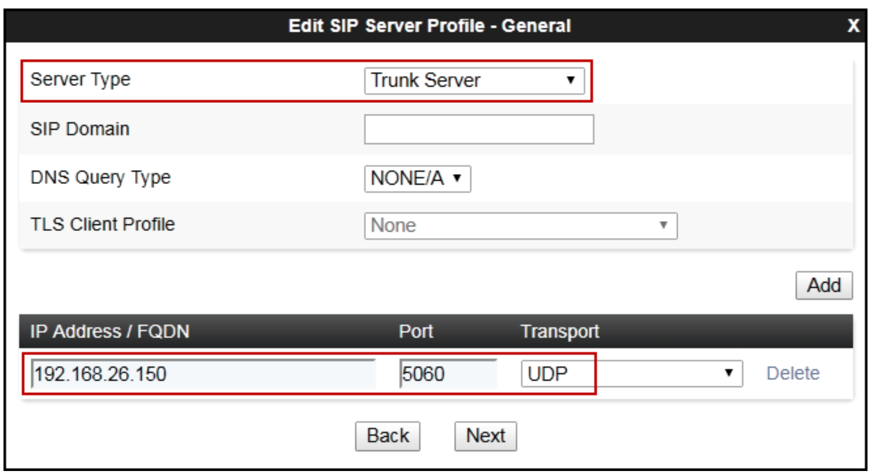
8.9.2. Server Configuration Profile – Service Provider

Similarly, to add the profile for the Trunk Server, click the **Add** button on the **Server Configuration** screen (not shown).

- Enter an appropriate **Profile Name** similar to the screen below (***Service Provider UDP*** was used).
- Click **Next**.

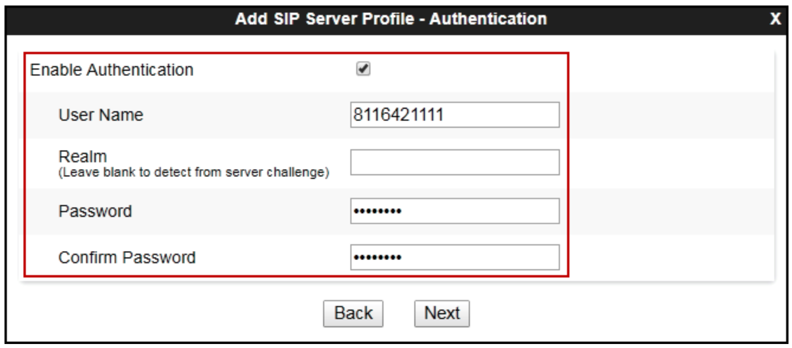


- On the **Edit Server Configuration Profile - General** Tab select ***Trunk Server*** from the drop-down menu for the **Server Type**.
- On the **IP Addresses / FQDN** field, enter ***192.168.26.150*** (Alestra's SIP proxy server IP address). This information was provided by Alestra.
- Enter ***5060*** under **Port** and select **UDP** for **Transport**.
- Click **Next**.



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

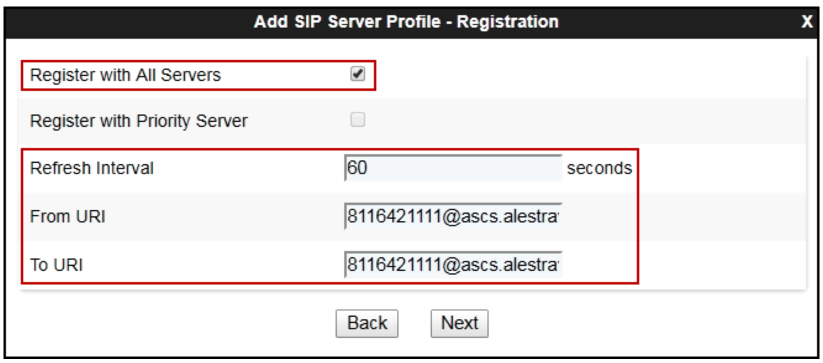
- Check the **Enable Authentication** box.
- Enter the **User Name** credential provided by the service provider for SIP trunk registration, the pilot number provided by Alestra was used for registration purpose.
- Leave the **Realm** blank.
- Enter **Password** credential provided by the service provider for SIP trunk registration.
- Click **Next**.



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

On the **Add Server Configuration Profile - Registration** tab:

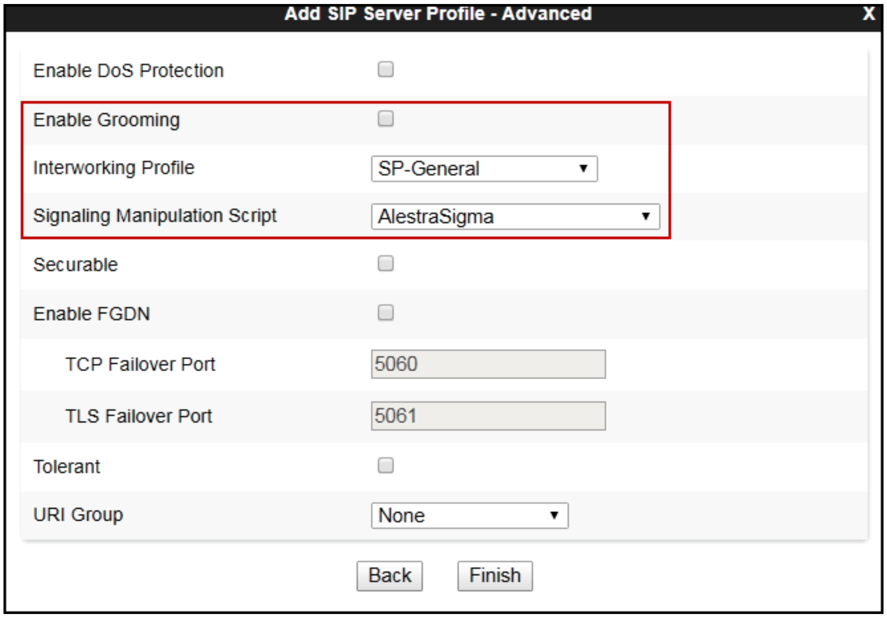
- Check the **Register with All Servers** box.
- **Frequency**: Enter the amount of time (in seconds) between REGISTER messages that will be sent from the enterprise to the Service Provider Proxy Server to refresh the registration binding of the SIP trunk. This value should be chosen in consultation with the service provider. **60** seconds was the value used during the compliance test.
- The **From URI** and **To URI** entries for the REGISTER messages are built using the following:
  - **From URI**: Enter the **User Name/Pilot number** entered above in the **Authentication** screen (**8116421111**) and Alestra's domain name (**ascs.alestravoip.com**), as shown below.
  - **To URI**: Enter the **User Name/Pilot number** entered above in the **Authentication** screen (**8116421111**) and Alestra's domain name (**ascs.alestravoip.com**), as shown below.
  - Click **Next**.



Click **Next** on the **Add Server Configuration Profile - Ping** window (not shown).

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

- Uncheck **Enable Grooming**.
- Select ***SP-General*** from the **Interworking Profile** drop-down menu (**Section 8.7.2)**.
- Select the ***AlestraSigma*** from the **Signaling Manipulation Script** drop down menu (**Sections 8.8** and **Section 13**).
- Click **Finish**.



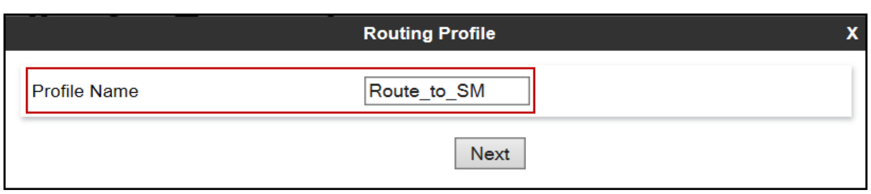
8.10.Routing

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

8.10.1. Routing Profile – Enterprise

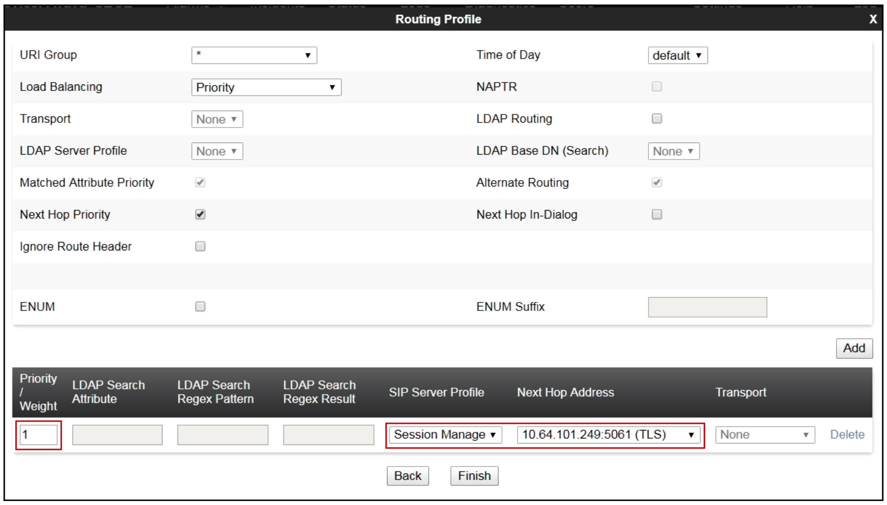
To create the inbound route, select the **Routing** tab from the **Configuration Profiles** menu on the left-hand side and select **Add** (not shown).

- Enter an appropriate **Profile Name** similar to the example below.
- Click **Next**.



- On the **Routing Profile** tab, click the **Add** button to enter the next-hop address.

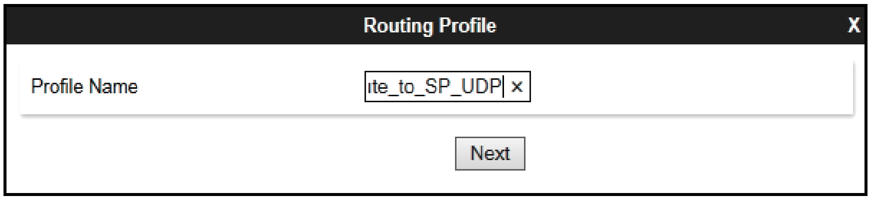
- Under **Priority/Weight** enter ***1***.
- Under **SIP Server Profile**, select ***Session Manager***. The **Next Hop Address** field will be populated with the IP address, port and protocol defined for the Session Manager Server Configuration Profile in **Section 8.9.1**.
- Defaults were used for all other parameters.
- Click **Finish**.



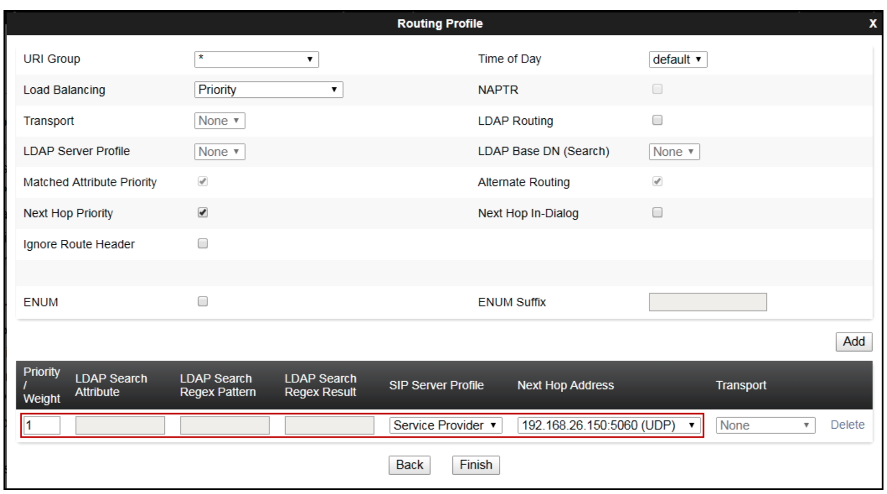
8.10.2. Routing Profile – Service Provider

Back at the **Routing** tab, select **Add** (not shown) to repeat the process in order to create the outbound route.

- Enter an appropriate **Profile Name** similar to the example below (***Route_to_SP_UDP*** was used).
- Click **Next**.



- Click the **Add** button to enter the next-hop address.
- Under **Priority/Weight** enter ***1***.
- Under **SIP Server Profile**, select ***Service Provider UDP***.
- The **Next Hop Address** is populated automatically with ***192.168.26.150:5060 (UDP)*** Alestra's SIP Proxy IP address, Port and Transport, Server Configuration Profile defined in **Section 8.9.2**.
- Click **Finish**



8.11.Topology Hiding

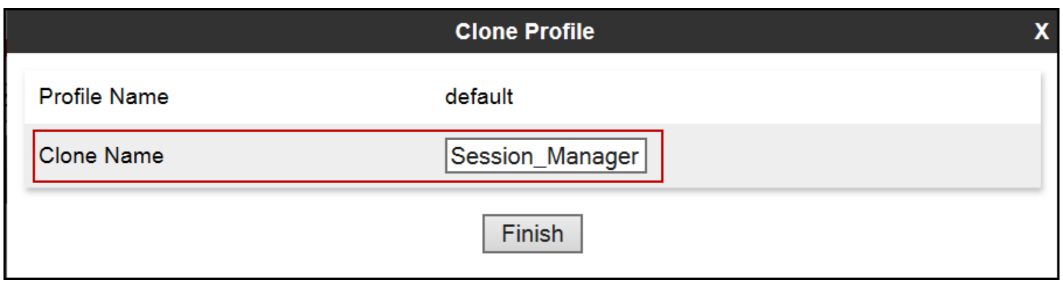
Topology Hiding is a security feature that allows the modification of several SIP headers, preventing private enterprise network information from being propagated to the untrusted public network.

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

8.11.1. Topology Hiding Profile – Enterprise

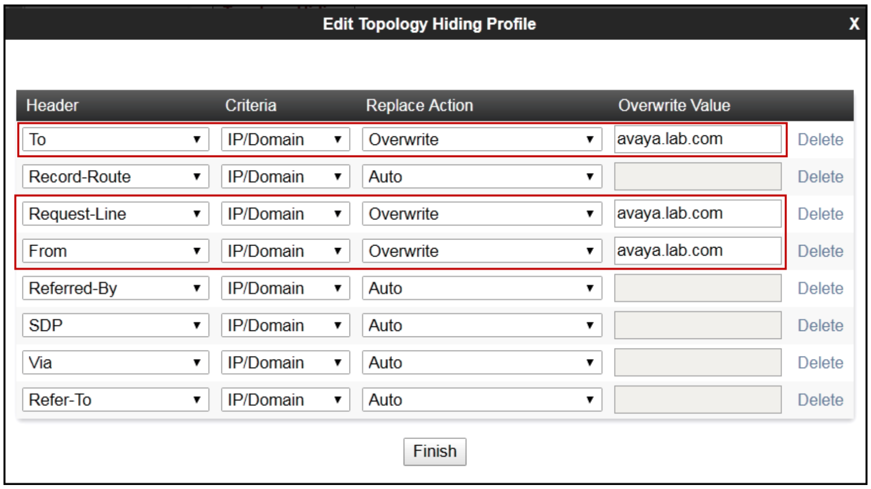
To add the Topology Hiding Profile in the enterprise direction, select **Topology Hiding** from the **Global Profiles** menu on the left-hand side, select ***default*** from the list of pre-defined profiles and click the **Clone** button (not shown).

- Enter a **Clone Name** such as the one shown below.
- Click **Finish**.



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

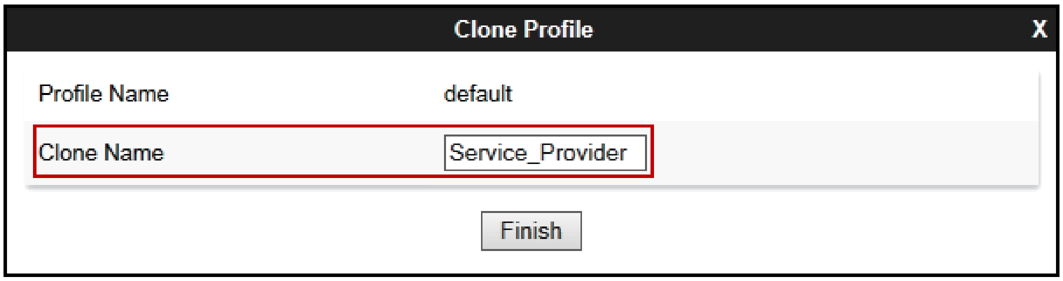
- For the, **From**, **To** and **Request-Line** headers, select ***Overwrite*** in the **Replace Action** column and enter the enterprise SIP domain ***avaya.lab.com***, in the **Overwrite Value** column of these headers, as shown below. This is the domain known by Session Manager, defined in **Section 7.2**.
- Default values were used for all other fields.
- Click **Finish**.



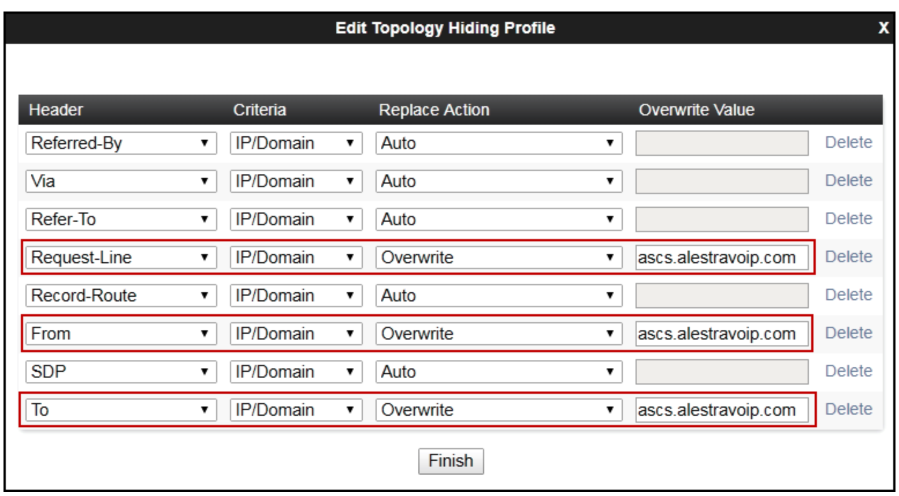
8.11.2. Topology Hiding Profile – Service Provider

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 ***default*** from the list of pre-defined profiles and click the **Clone** button (not shown).

- Enter a **Clone Name** such as the one shown below.
- Click **Finish**.



- Click **Edit** on the newly created **Service_Provider** Topology Hiding profile.
- On the **Request-Line** choose **Overwrite** from the pull-down menu under **Replace Action;** enter the domain name for the service provider (***ascs.alestravoip.com***) under **Overwrite Value**.
- On the **From** choose **Overwrite** from the pull-down menu under **Replace Action,** enter the domain name for the service provider (***ascs.alestravoip.com***) under **Overwrite Value**.
- On the **To** choose **Overwrite** from the pull-down menu under **Replace Action,** enter the domain name for the service provider (***ascs.alestravoip.com***) under **Overwrite Value**.
- Click **Finish**.



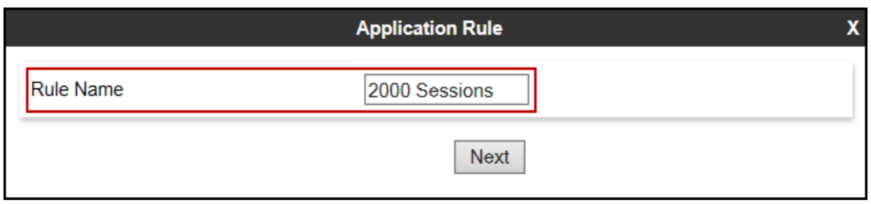
8.12. Domain Policies

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.

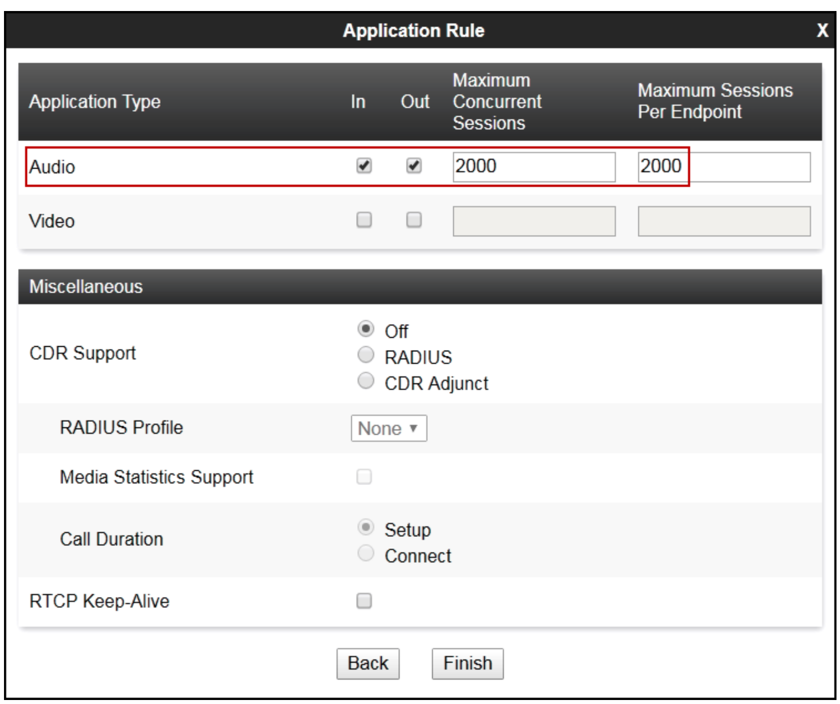
8.12.1. Application Rules

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 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.

- Under **Rule Name** enter the name of the profile, e.g., ***2000 Sessions***.
- Click **Next**.



- Under **Audio** check ***In*** and ***Out*** and set the **Maximum Concurrent Sessions** and **Maximum Sessions Per Endpoint** to recommended values, the value of ***2000*** for Audio. Repeat for video if needed.
- Click **Finish**.

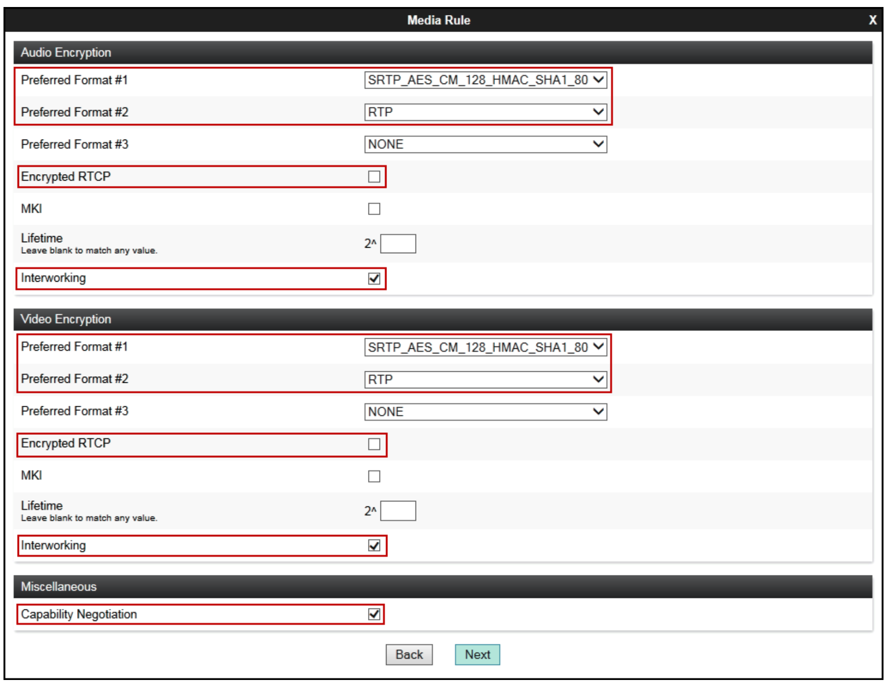


8.12.2. Media Rules

Media Rules allow one to define RTP media packet parameters such as prioritizing encryption techniques and packet encryption techniques. Together these media-related parameters define a strict profile that is associated with other SIP-specific policies to determine how media packets matching these criteria will be handled by the Avaya SBCE security product. For the compliance test, one media rule (shown below) was created toward Session Manager and a default media rule was used toward the Service Provider.

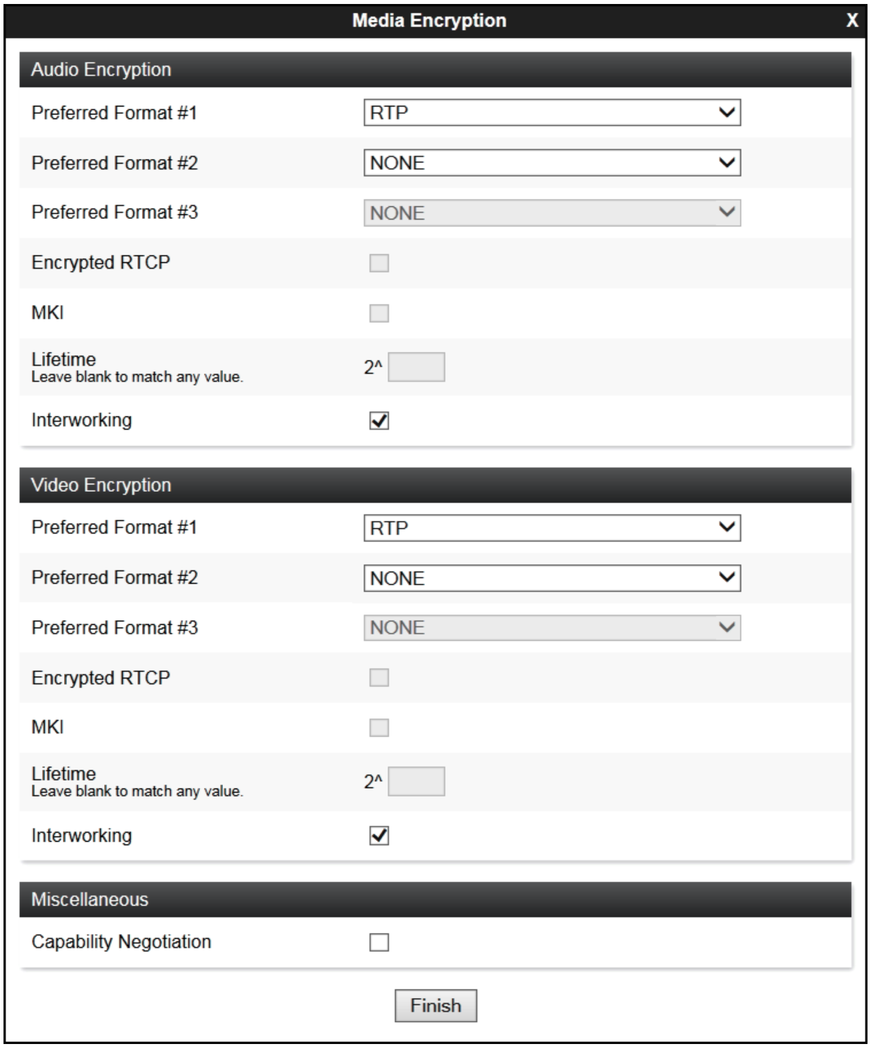
To add a media rule in the Session Manager direction, from the menu on the left-hand side, select **Domain Policies → Media Rules**.

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



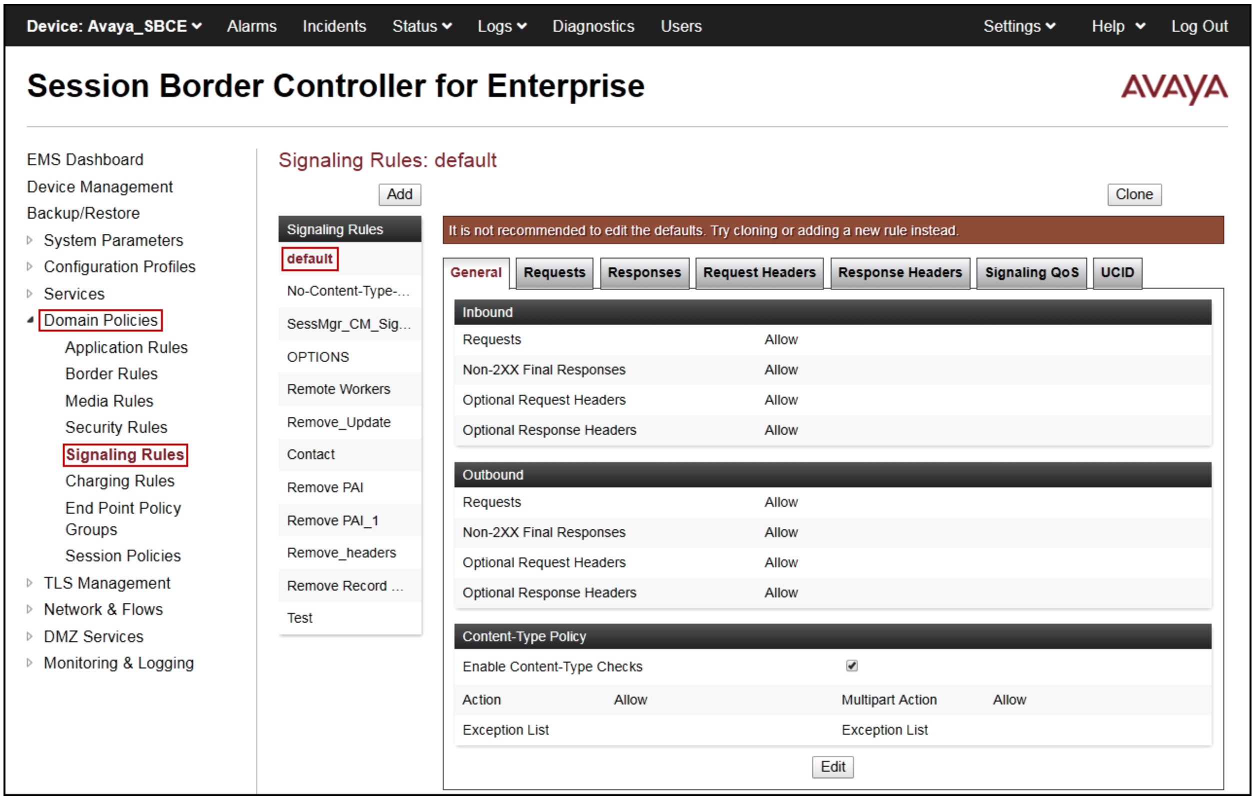
- Accept default values in the remaining sections by clicking **Next** (not shown), and then click **Finish** (not shown).

- For the compliance test, the **default-low-med** Media Rule was used in the Service Provider direction.



8.12.3. Signaling Rules

For the compliance test, the **default** signaling rule was used.



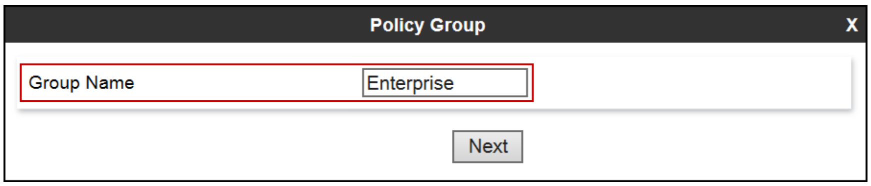
8.13.End Point Policy Groups

End Point Policy Groups associate the different sets of rules under Domain Policies (Media, Signaling, Security, etc.) to be applied to specific SIP messages traversing through the Avaya SBCE. Please note that changes should not be made to any of the default rules used in these End Point Policy Groups.

8.13.1. End Point Policy Group – Enterprise

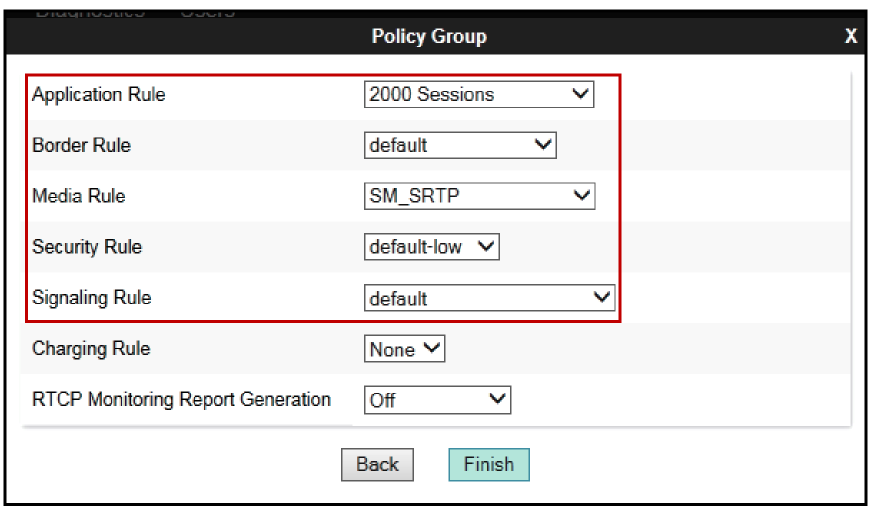
To create an End Point Policy Group for the enterprise, select **End Point Policy Groups** under the **Domain Policies** menu and select **Add** (not shown).

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



Under the **Policy Group** tab enter the following:

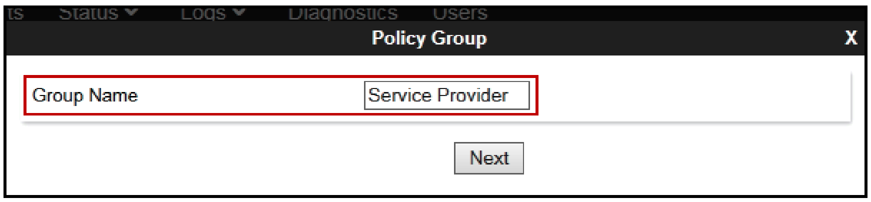
- **Application Rule: *2000 Sessions*** (**Section 8.12.1**).
- **Border Rule: *default*.**
- **Media Rule: *SM_SRTP*** (**Section 8.12.2**).
- **Security Rule: *default-low***.
- **Signaling Rule: *default*** (**Section 8.12.3**).
- Click **Finish**.



8.13.2. End Point Policy Group – Service Provider

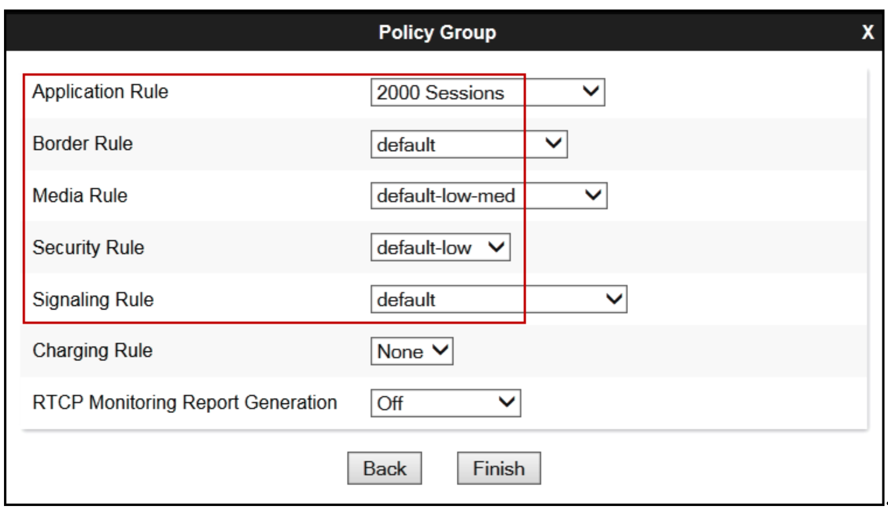
To create an End Point Policy Group for the Service Provider, select **End Point Policy Groups** under the **Domain Policies** menu and select **Add** (not shown).

- Enter an appropriate name in the **Group Name** field (***Service Provider*** was used).
- Click **Next**.



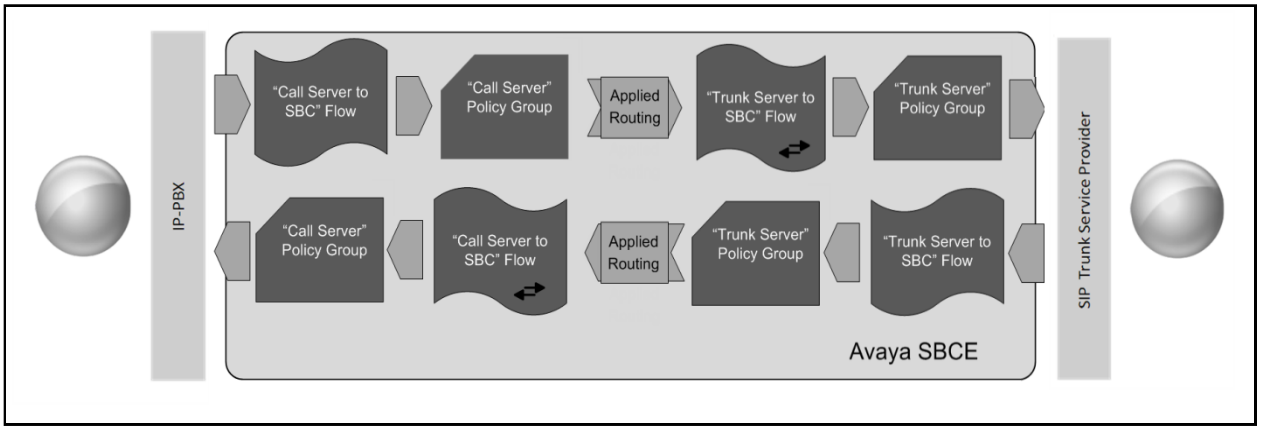
Under the **Policy Group** tab enter the following:

- **Application Rule:** ***2000 Sessions*** (**Section 8.12.1**).
- **Border Rule:** ***default***.
- **Media Rule:** ***default-low-med*** (**Section 8.12.2**).
- **Security Rule:** ***default-low***.
- **Signaling Rule:** ***default*** (**Section 8.12.3**).
- Click **Finish**.



8.14.End Point Flows

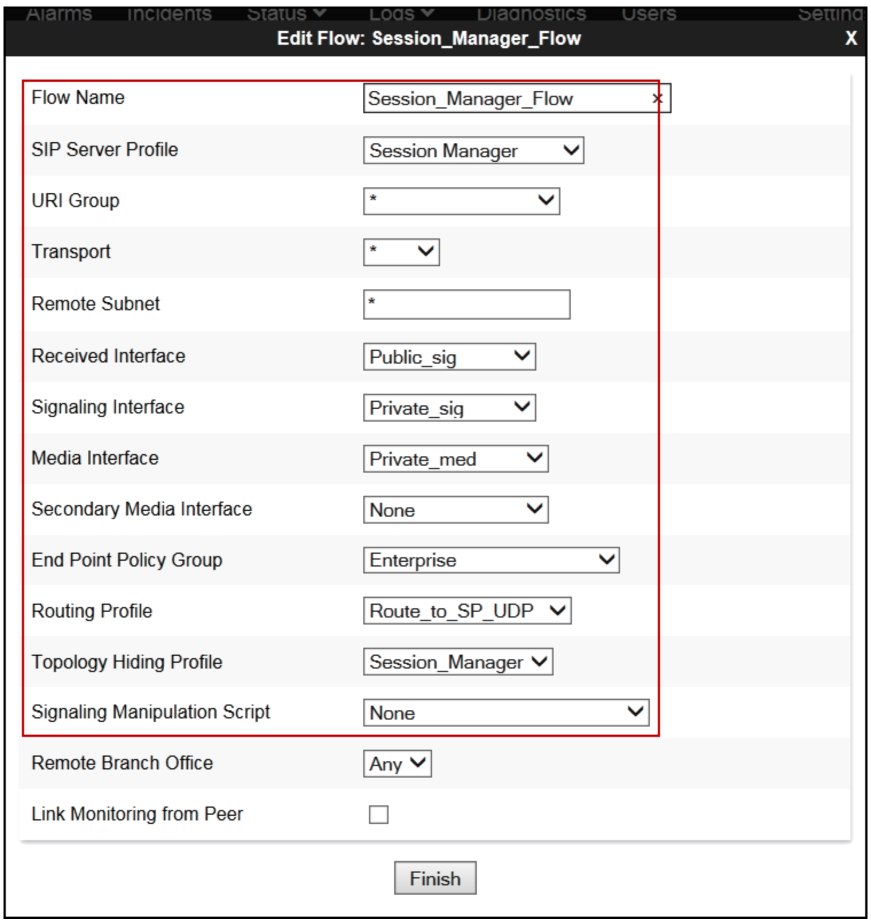
When a packet is received by Avaya SBCE, the content of the packet (IP addresses, URIs, etc.) is used to determine which flow it matches. Once the flow is determined, the flow points to a policy group which contains several rules concerning processing, privileges, authentication, routing, etc. Once routing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

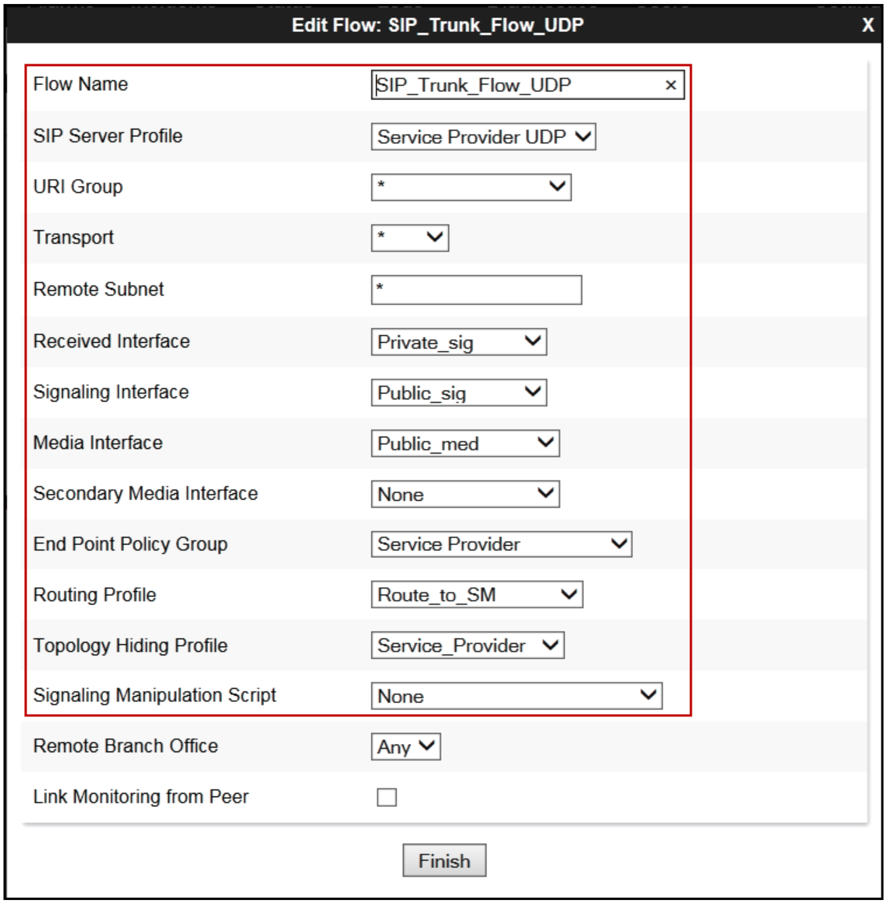
8.14.1. End Point Flow – Enterprise

To create the call flow toward the enterprise, from the **Device Specific** menu, select **End Point Flows**, then select the **Server Flows** tab. Click **Add** (not shown). The screen below shows the flow named ***Session_Manager_Flow*** created in the sample configuration. The flow uses the interfaces, policies, and profiles defined in previous sections. Note that the **Routing Profile** selection is the profile created for the Service Provider in **Section 8.10.2**, which is the reverse route of the flow. Click **Finish**.



8.14.2. End Point Flow – Service Provider

A second Server Flow with the name ***SIP_Trunk_Flow_UDP*** was similarly created in the Service Provider direction. The flow uses the interfaces, policies, and profiles defined in previous sections. Note that the **Routing Profile** selection is the profile created for Session Manager in **Section 8.10.1**, which is the reverse route of the flow. Also note that there is no selection under the **Signaling Manipulation Script** field. Click **Finish**.



9. Alestra SIP Trunking Service Configuration

To use Alestra SIP Trunking Service, a customer must request the service from Alestra using the established sales processes. The process can be started by contacting Alestra via the corporate web site at: http://www.alestra.com.mx/

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

Alestra will provide the following information:

- SIP Trunk registration credentials (User Name, Password, etc.).
- Domain name.
- DID numbers.
- Etc.

10. Verification and Troubleshooting

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

10.1. General Verification Steps

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

10.2. Communication Manager Verification

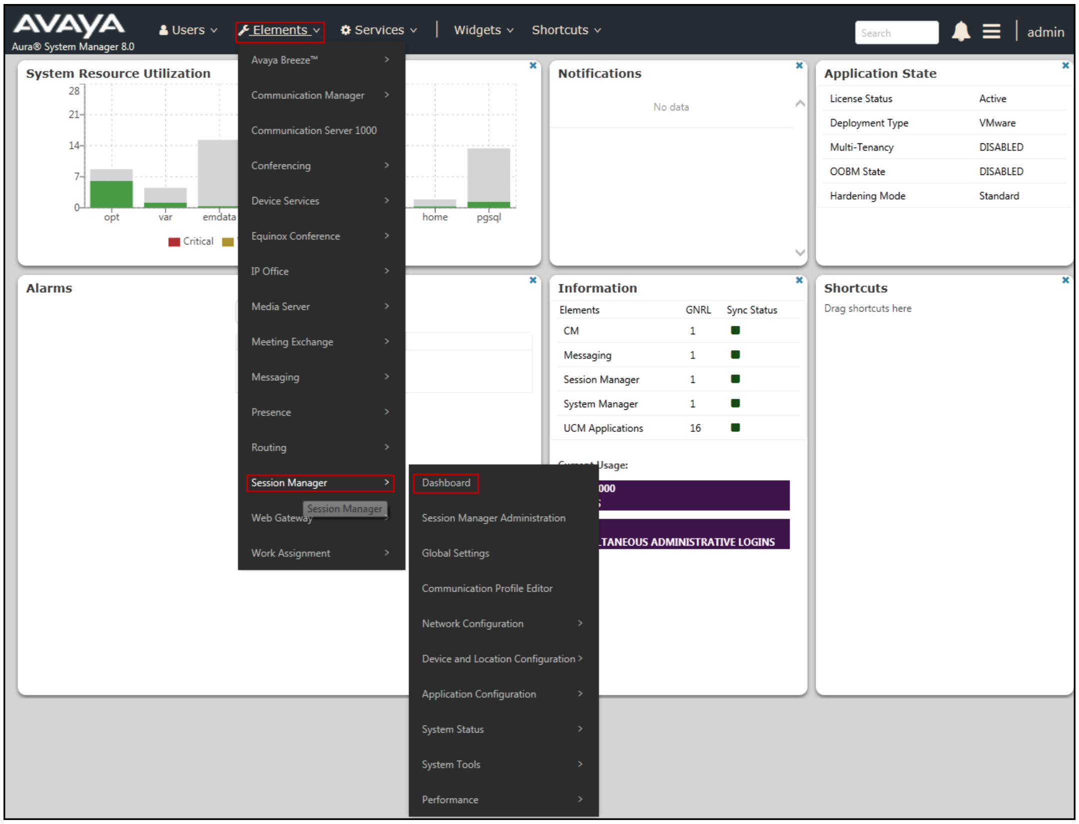
The following commands can be entered in the Communication Manager SAT terminal to verify the SIP trunk functionality:

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

10.3.Session Manager Verification

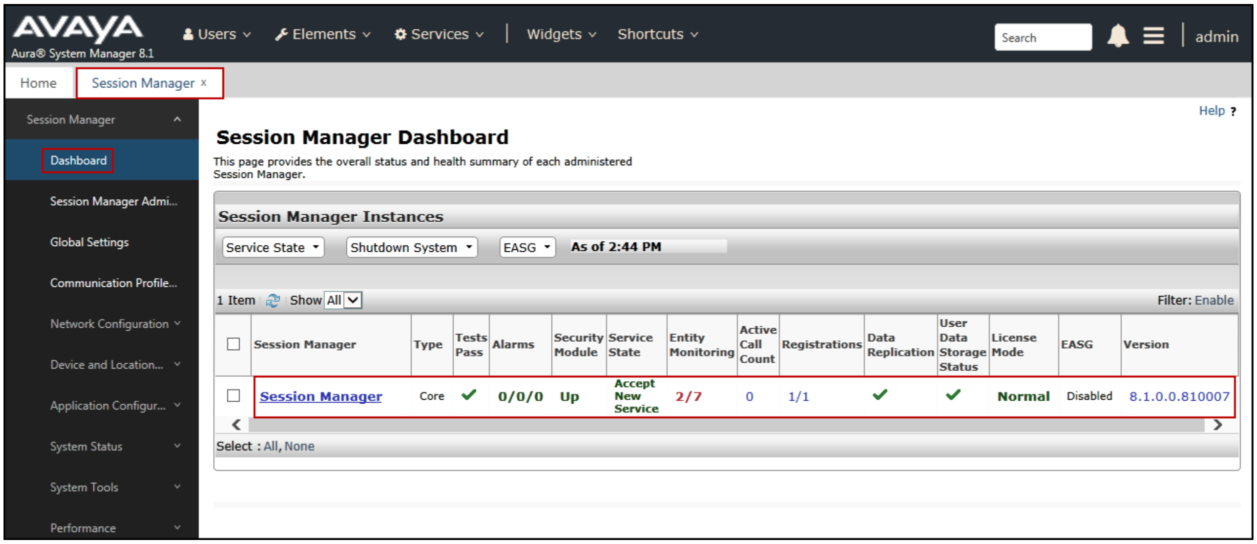
The Session Manager configuration may be verified via System Manager.

Step 1 - Using the procedures described in **Section 7**, access the System Manager GUI. From the **Home** screen, under the **Elements** heading, select **Session Manager,** then select **Dashboard**.

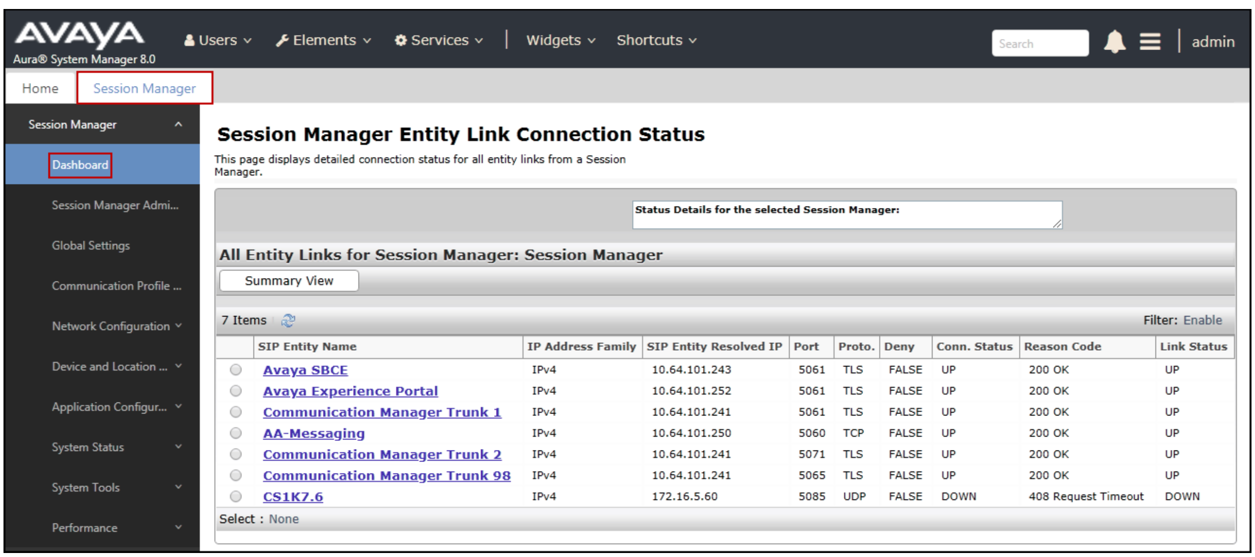


Step 2 - The Session Manager Dashboard is displayed. Note that the **Test Passed**, **Alarms**, **Service State**, and **Data Replication** columns all show good status.

In the **Entity Monitoring** column, Session Manager shows that there are **2** alarms out of the **7** Entities defined.



Verify that the state of the Session Manager links under the **Conn. Status** and **Link Status** columns are ***UP***, like shown on the screen below



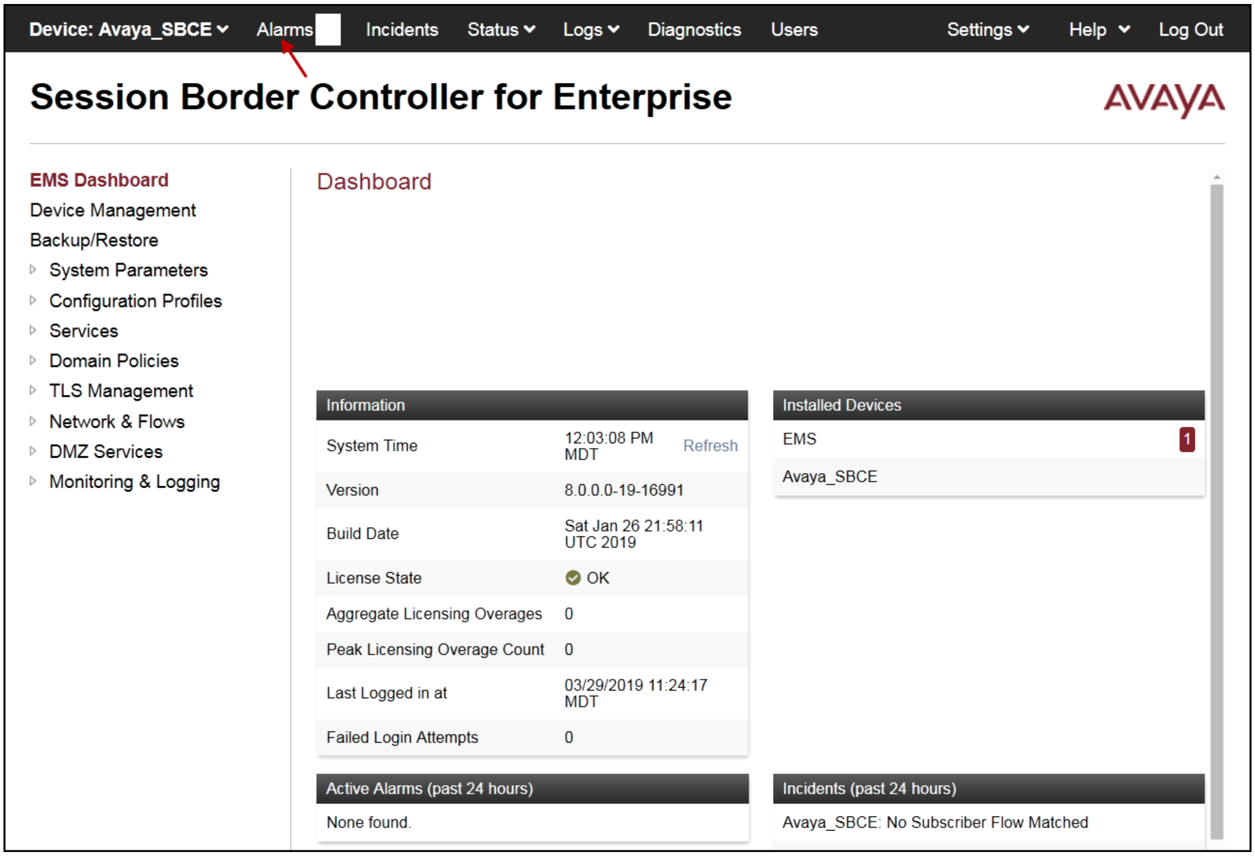
Other Session Manager useful verification and troubleshooting tools include:

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

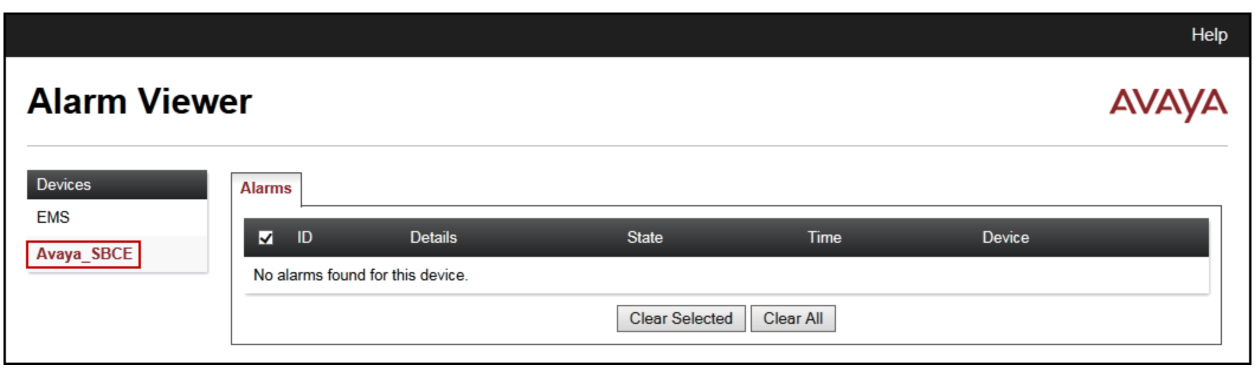
10.4. Avaya SBCE Verification

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

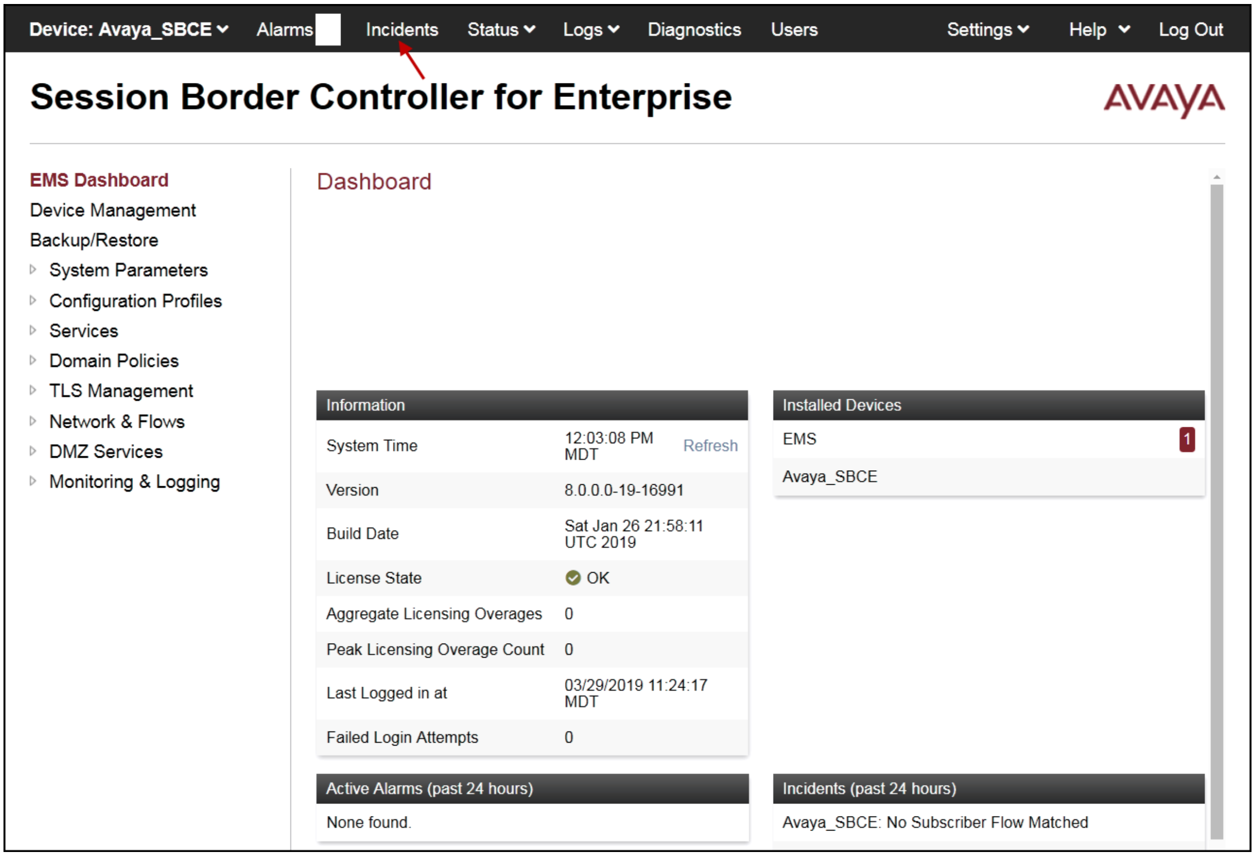
Alarms: This screen provides information about the health of the SBC.



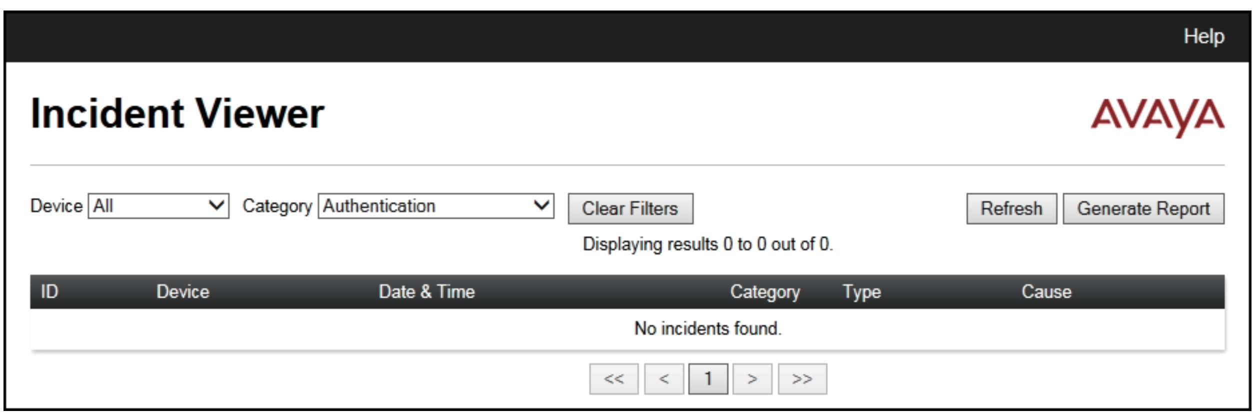
The following screen shows the **Alarm Viewer** page.



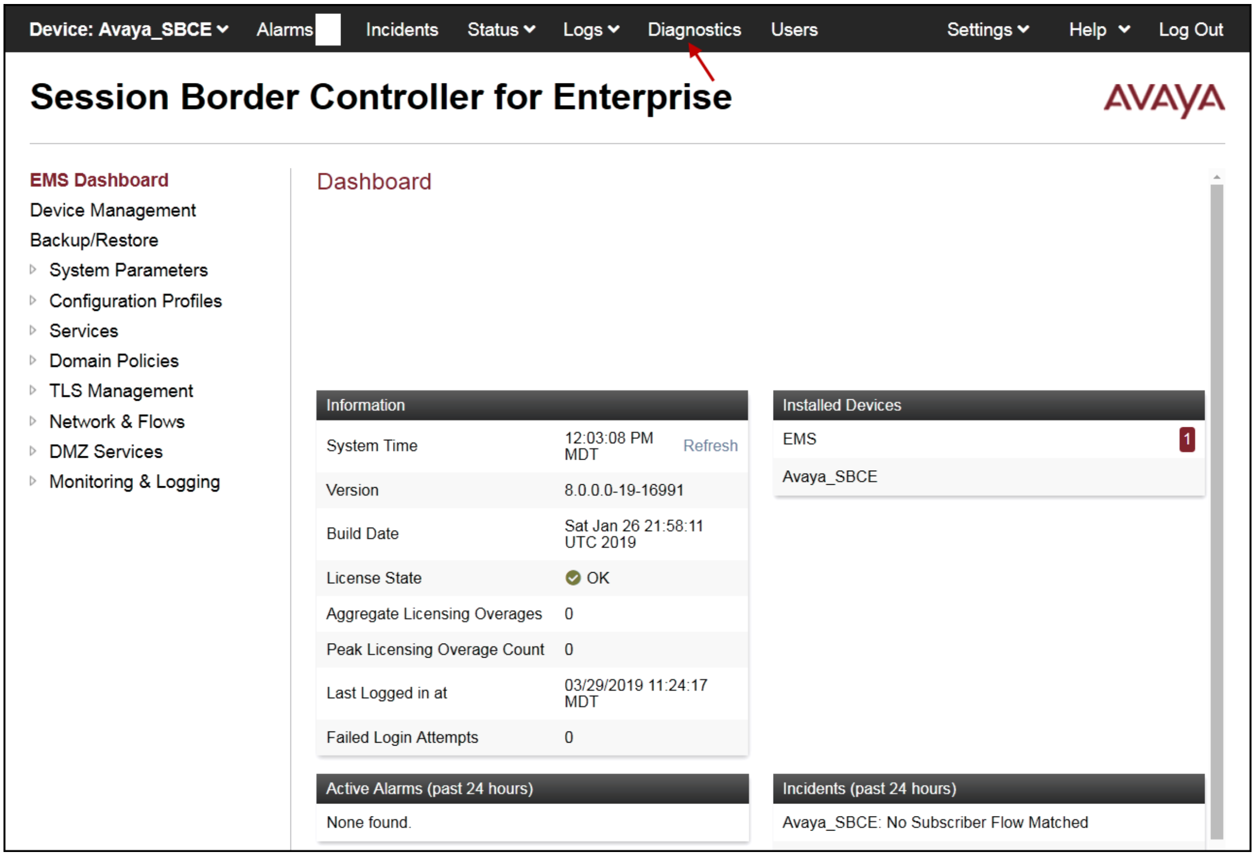
Incidents : Provides detailed reports of anomalies, errors, policies violations, etc.



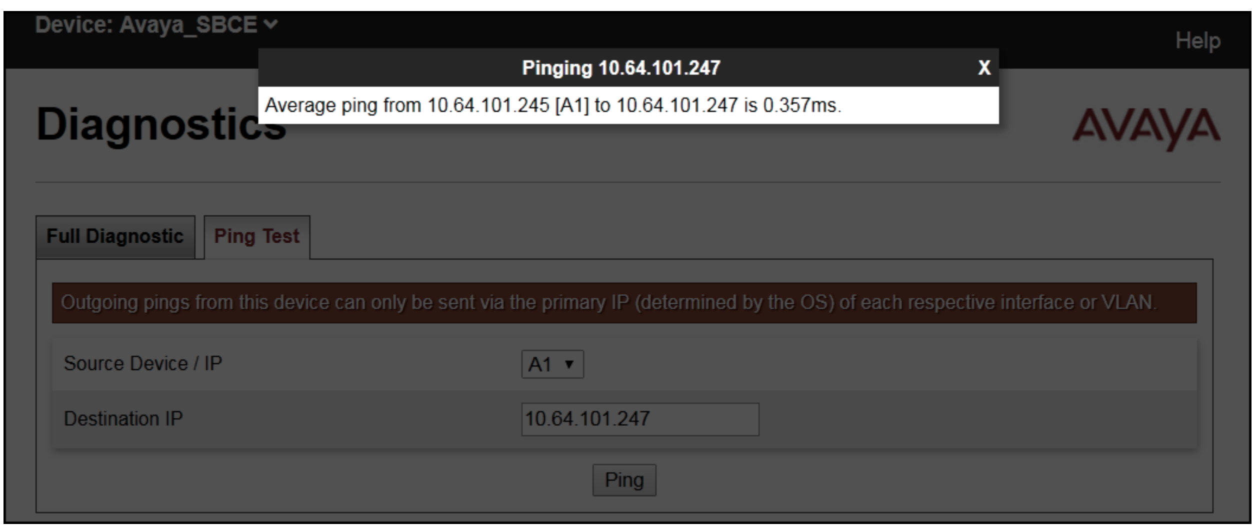
The following screen shows the **Incident Viewer** page.



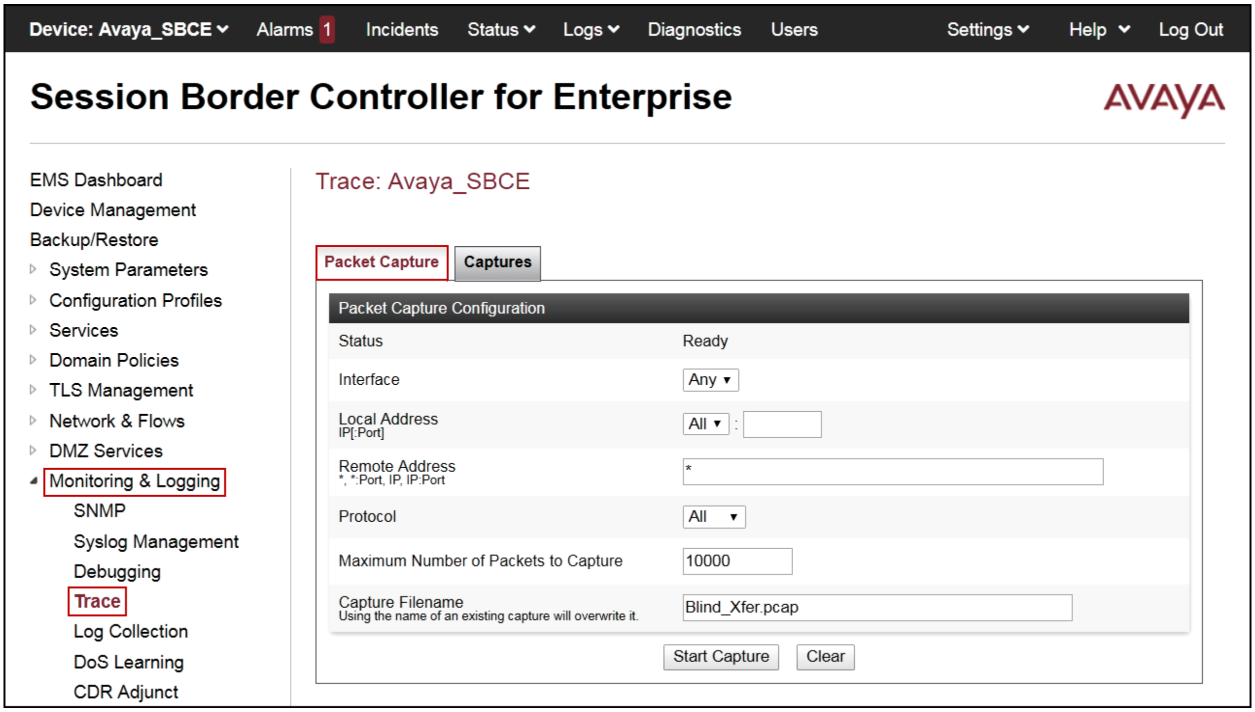
Diagnostics: This screen provides a variety of tools to test and troubleshoot the Avaya SBCE network connectivity.



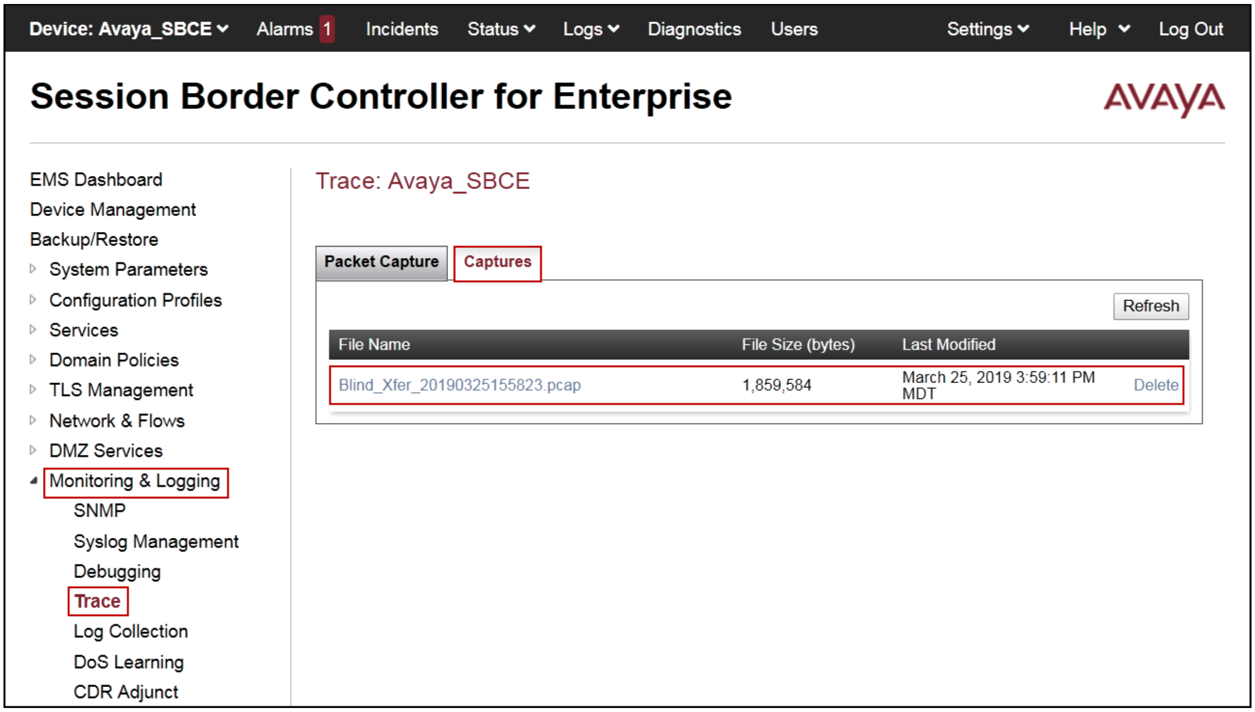
The following screen shows the Diagnostics page with the results of a ping test.



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



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



Also, the **traceSBC** tool can be used to monitor the SIP signaling messages between the Service provider and the Avaya SBCE.

11. Conclusion

These Application Notes describe the procedures required to configure Avaya Aura® Communication Manager 8.1, Avaya Aura® Session Manager 8.1, Avaya Aura® Experience Portal 7.2 and Avaya Session Border Controller for Enterprise 8.0, to connect to the Alestra SIP Trunking service, as shown in **Figure 1**.

Interoperability testing of the sample configuration was completed with successful results for all test cases with the observations/limitations described in **Sections 2.1** and **2.2**.

12. References

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

[1] *Deploying Avaya Aura® Communication Manager* in a Virtualized Environment, Release 8.1.x, Issue 2, August 2019.

[2] *Administering Avaya Aura® Communication Manager*, Release 8.1.x, Issue 3, August 2019.

[3] *Administering Avaya Aura® System Manager* for Release 8.1.x, Issue 3, July 2019.

[4] *Deploying Avaya Aura® System Manager* in a Virtualized Environment, Release 8.1.x, Issue 2, July 2019.

[5] *Deploying Avaya Aura® Session Manager and Avaya Aura® Branch Session Manager* in a Virtualized Environment , Release 8.1., Issue 1, June 2019.

[6] *Administering Avaya Aura® Session Manager*, Release 8.1, Issue 1, June 2019.

[7] *Deploying Avaya Session Border Controller for Enterprise*, Release 8.0, Issue 3, July 2019.

[8] *Administering Avaya Session Border Controller for Enterprise*, Release 8.0, Issue 1, February 2019.

[9] *Administering Avaya Aura® Experience Portal*, Release 7.2.2, Issue 1, March 2019

[10] *Implementing Avaya Aura® Experience Portal on a single server*, Release 7.2.2, Issue 1, July 2019

[11] *Configuring Remote Workers with Avaya Session Border Controller for Enterprise Rel. 7.0, Avaya Aura® Communication Manager Rel. 7.0 and Avaya Aura® Session Managers Rel. 7.0 - Issue 1.0.*

[12] *Deploying and Updating Avaya Aura® Media Server Appliance*, Release 8.0.x, Issue 7, June 2019.

[13] *Implementing and Administering Avaya Aura® Media Server*. Release 8.0.x, Issue 5, June 2019.

[14] *Planning for and Administering Avaya Equinox for Android, iOS, Mac, and Windows*. Release 3.6, Issue 1, July 2019.

[15] *Administering Avaya one-X® Communicator*. Release 6.2, Feature Pack 10, November 2015.

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

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

13. Appendix A: SigMa Scripts

Following are the two Signaling Manipulation (SigMa) scripts that were used during the compliance test, add the scripts as indicated in **Section 8.8**, enter a name for each script in the Title (e.g., ***Change Max-Forwards*** or ***AlestraSigma***) and copy/paste the text for each individual script shown below. Assign the scripts to the Server Configuration Profiles as instructed in **Sections 8.9.1** and **8.9.2**.

The following SigMa scripts will:

- Change the Max-Forwards in INVITES SIP messages received from Alestra from 9 to 69.
- Insert the Pilot number associated with the SIP trunk in the “From” and “Contact” headers of SIP messages sent to Alestra.
- Remove unwanted “gsid” and “epv” parameter from being sent to Alestra in the Contact header.
- Remove the P-Location parameter from being sent to Alestra.
- Change the Diversion header scheme from SIPS to SIP in SIP messages sent to Alestra.
- Remove unwanted xml element information from the SDP in SIP messages sent to Alestra.

Title: *Change Max-Forwards*

```
within session "INVITE"
{
   act on request where %DIRECTION="OUTBOUND" and
%ENTRY_POINT="POST_ROUTING"
   {
       if (exists(%HEADERS["Max-Forwards"][1])) then
       {
         %HEADERS["Max-Forwards"][1] = "69";
       }

   }
}
```

Title: *AlestraSigma*

```
//Insert the Pilot number associated with the SIP Trunk in the FROM and CONTACT headers of
Outbound calls.
within session "ALL"
```

```
{
act on request where %DIRECTION="OUTBOUND" and
%ENTRY_POINT="POST_ROUTING"
{

%fromuser = %HEADERS["From"][1].URI.USER;
%HEADERS["From"][1].URI.USER = "8116421111";

%contact = %HEADERS["Contact"][1].URI.USER;
%HEADERS["Contact"][1].URI.USER = "8116421111";

//Remove gsid and epv parameters from Contact header.
remove(%HEADERS["Contact"][1].URI.PARAMS["gsid"]);
remove(%HEADERS["Contact"][1].URI.PARAMS["epv"]);

//Remove P-Location parameter.
remove(%HEADERS["P-Location"][1]);

//Changes the Diversion header scheme from SIPS to SIP.
%HEADERS["Diversion"][1].regex_replace("sips","sip");

//Remove unwanted xml element information from the SDP in SIP messages sent to the Service
Provider.
remove(%BODY[1]);

}
}
```

14. Appendix B – Avaya Session Border Controller for Enterprise – Refer Handling

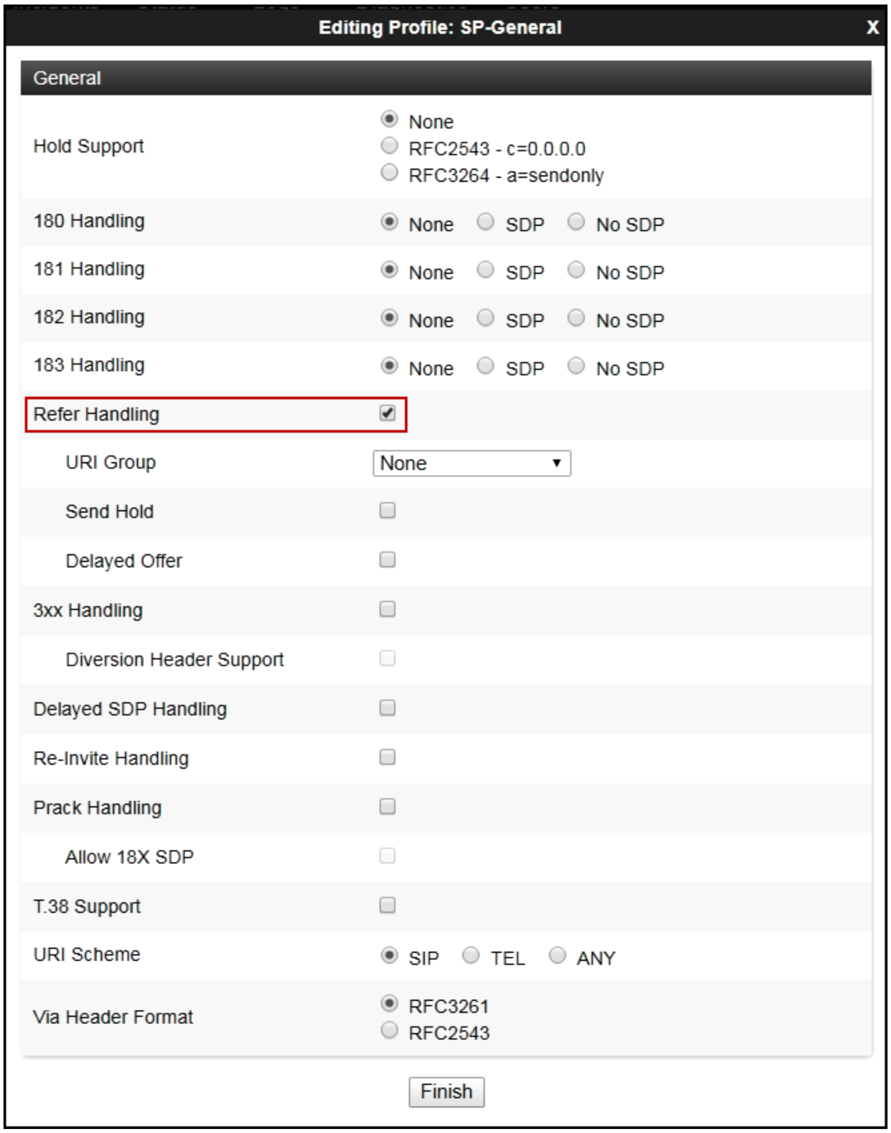
One of the capabilities important to the Experience Portal environment is the Avaya SBCE Refer Handling option. Experience Portal inbound call processing may include call redirection to Communication Manager agents, or other CPE destinations. This redirection is accomplished by having Experience Portal send SIP REFER messaging to the Avaya SBCE. Enabling the Refer Handling option causes the Avaya SBCE to intercept and process the REFER and generate a new SIP INVITE messages back to the CPE (e.g., Communication Manager).

As an additional option, the Refer Handling feature can also specify *URI Group* criteria as a discriminator, whereby SIP REFER messages matching the URI Group criteria are processed by the Avaya SBCE, while SIP REFER messages that do not match the URI Group criteria, are passed through to the Service Provider. The *URI Group* criteria method for SIP REFER handling was not used during the compliance test with Alestra, refer to **Section 2.2**.

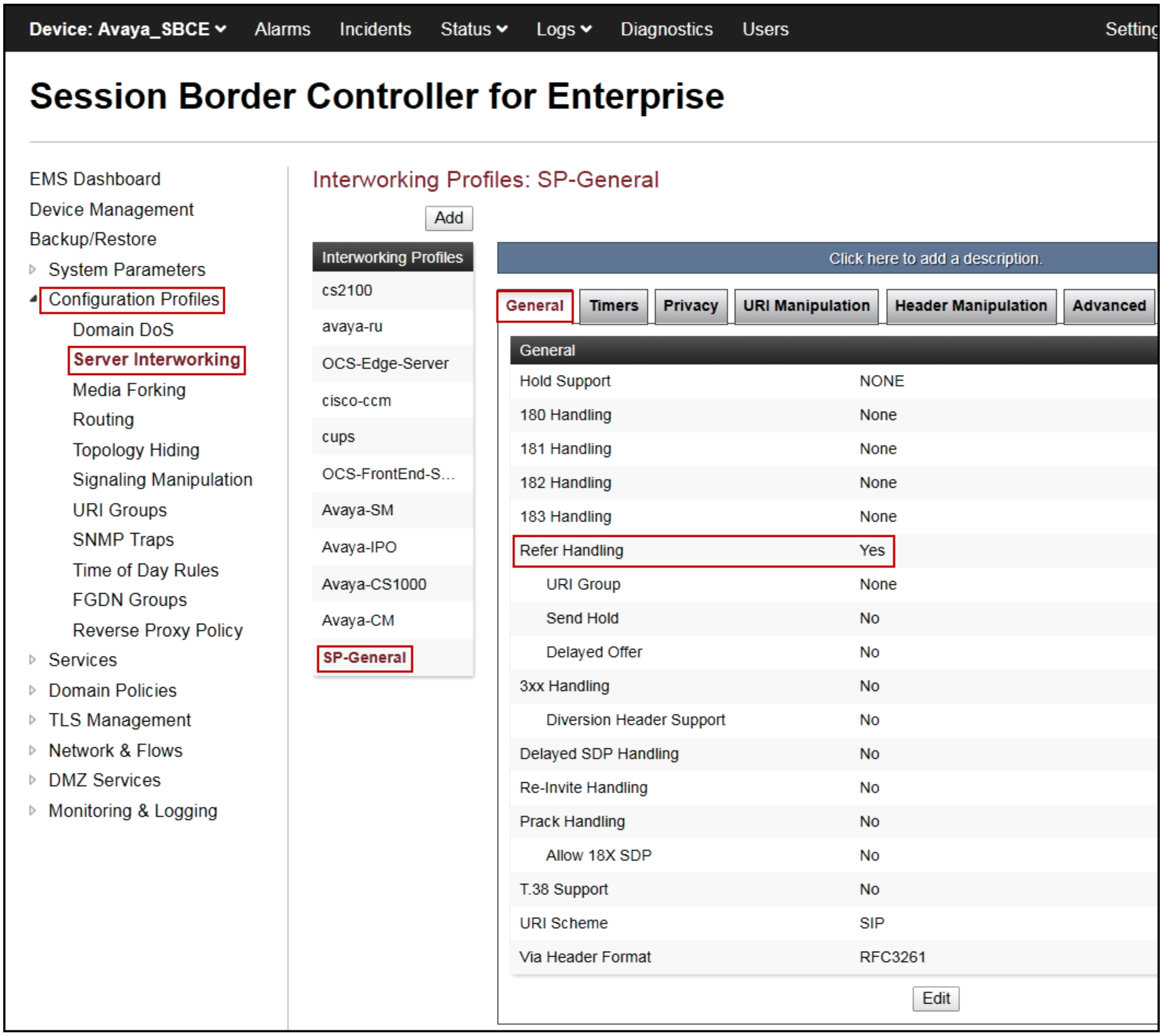
Edit the existing **SP-General** Server Interworking Profile to enable Refer Handling.

Step 1 - Select **Configuration Profiles → Server Interworking** from the left-hand menu (not shown).
Step 2 - Select the **SP-General** Server Interworking Profile created in **Section 8.7.2** and click **Edit**

- Check **Refer Handling**.
- Select **Finish**.



Following is the SP-General Server Interworking profile after editing.



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.