AVAYA

Avaya Solution & Interoperability Test Lab

Application Notes for Posh Voice with Avaya Session Border Controller for Enterprise – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate Posh Voice with Avaya Session Border Controller for Enterprise 10.1. Posh Voice is a conversational AI IVR that interfaces to a SIP service provider, which in turn connects to Avaya Session Border Controller for Enterprise via a SIP trunk. Avaya Session Border Controller for Enterprise provides access to a contact center on Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

Posh Voice is an adjunct to the contact center, not a replacement. Therefore, the initial call comes into the contact center and is then routed to Posh Voice via a SIP service provider. Posh Voice interacts with callers to answer their questions and perform banking transactions using their voice in a conversational style or DTMF using their telephone keypad. If required, Posh Voice can transfer the call to an agent and provide caller information via User-to-User Information (UUI).

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any 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 configuration steps required to integrate Posh Voice with Avaya Session Border Controller for Enterprise 10.1. Posh Voice is a conversational AI IVR that interfaces to a SIP service provider, which in turn connects to Avaya Session Border Controller for Enterprise via a SIP trunk. Avaya Session Border Controller for Enterprise (SBCE) provides access to a contact center on Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

Posh Voice is an adjunct to the contact center, not a replacement. Therefore, the initial call comes into the contact center and is then routed to Posh Voice via a SIP service provider. In the compliance test, there were two ingress points to the SIP service provider for redundancy corresponding to a primary and secondary site in different geographic locations Posh Voice interacts with callers to answer their questions and perform banking transactions using their voice in a conversational style or DTMF using their telephone keypad. If required, Posh Voice can transfer the call to a contact center agent and provide caller information via User-to-User Information (UUI).

The general call flow is as follow:

1. Caller places a call from the PSTN to the Avaya contact center, which arrives on SBCE.
2. The call is then routed to the Posh Voice staging or production environment via a SIP service provider using TLS-encrypted SIP signaling and SRTP media.
3. Caller interacts with Posh Voice using voice or DTMF to request or provide information (e.g., hear business hours, request account balances, and log in with credentials).
4. Upon request, Posh Voice can transfer the call to a live agent with a SIP REFER and send caller information (e.g., customer number and authentication status) in UUI. It is up to the client to use the UUI data, as needed, in the systems the agent uses.
5. Posh Voice can transfer the call to an agent by calling a VDN or any internal extension. This call transfer could also be forwarded to the PSTN.
6. Caller is connected to an agent (or other transferred-to party).
7. The call to Posh Voice is disconnected.

Note: In these Application Notes, "Posh Voice SIP service provider" refers to a third-party SIP service provider that is compatible with Posh Voice and connects directly to Avaya Session Border Controller for Enterprise via a SIP trunk. Posh Voice does not provide SIP trunking services. As such, all calls from the Avaya network to Posh Voice are routed through the SIP service provider.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on customer calls being routed to Posh Voice through a SIP service provider. Posh Voice then provided customer service via sample IVR application, which allowed customers to access information or be transferred to a call center agent on Communication Manager. Customers interacted with Posh Voice using speech and DTMF via a telephone keypad. For example, callers made verbal requests to hear the business hours, get account balance, or transfer to an agent after logging in using speech or DTMF. For calls routed to an agent, Posh Voice provided customer information via UUI. Calls to Posh Voice staging and production environments were verified.

The serviceability test cases focused on restarting SBCE and verifying that calls to Posh Voice were successful after SBCE came back into service (i.e., no adverse effects on Posh Voice or the SIP service provider).

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

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

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

For the testing associated with these Application Notes, the interface between Avaya systems and Posh Voice SIP service provider utilized encryption capabilities of TLS/SRTP.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP trunk between SBCE and Posh Voice SIP service provider using TLS transport and verifying the exchange of SIP OPTIONS messages (i.e., Posh Voice SIP service provider responded successfully to SIP OPTIONS from SBCE).
- Establishing SIP trunks and placing calls to both Posh Voice staging and production environments.
- PSTN calls routed from SBCE to Posh Voice staging and production environments through a SIP service provider with Direct IP Media (Shuffling) and Initial IP-IP Direct Media enabled and disabled.
- Posh Voice providing customer service via a sample IVR application and customers navigating the application using speech and DTMF.
- Posh Voice transferring call to an agent on Communication Manager using VDN.
- Posh Voice providing caller information (e.g., customer number and authentication status) in User-to-User (UUI) when transferring call to live agents.
- Posh Voice sending UUI with the SIP REFER when the call is transferred to an agent.
- Posh Voice transferring call to PSTN.
- Multiple simultaneous calls to Posh Voice.
- Telephony features, such as holding and resuming calls to Posh Voice, transferring calls to Posh Voice, joining Posh Voice in a conference, forwarding calls to Posh Voice, and calls to Posh Voice lasting more than 5 minutes.
- DTMF transmission using RFC2833.
- SIP signaling encrypted using TLS 1.2.
- Audio encrypted using SRTP.
- G.711mu-law codec support as supported by SIP service provider.
- Restarting SBCE and verifying no adverse effects on Posh Voice or SIP service provider.

2.2. Test Results

All test cases passed.

2.3. Support

Technical support on Posh Voice can be obtained through the following:

- **Phone:** (617) 457-5567
- **Email:** support@posh.tech
- **Web:** https://www.posh.tech

3. Reference Configuration

Figure 1 illustrates a sample configuration consisting of Posh Voice connected through a SIP service provider to an Avaya Aura® environment. All customer calls were routed through the Avaya Aura® environment and then to Posh Voice via the SIP service provider. The Avaya Aura® environment consisted of the following products:

- SBCE with SIP trunk connectivity to Session Manager, Posh Voice SIP service provider, and the PSTN.
- Session Manager connected to SBCE and Communication Manager via a SIP trunk and acting as a Registrar/Proxy for SIP telephones.
- Media resources in Avaya G430 Media Gateway and Avaya Aura® Media Server.
- System Manager used to configure Session Manager.
- Agents on Avaya 96x1 Series H.323 Deskphones, Avaya J100 Series SIP Deskphones, and Avaya Agent for Desktop.

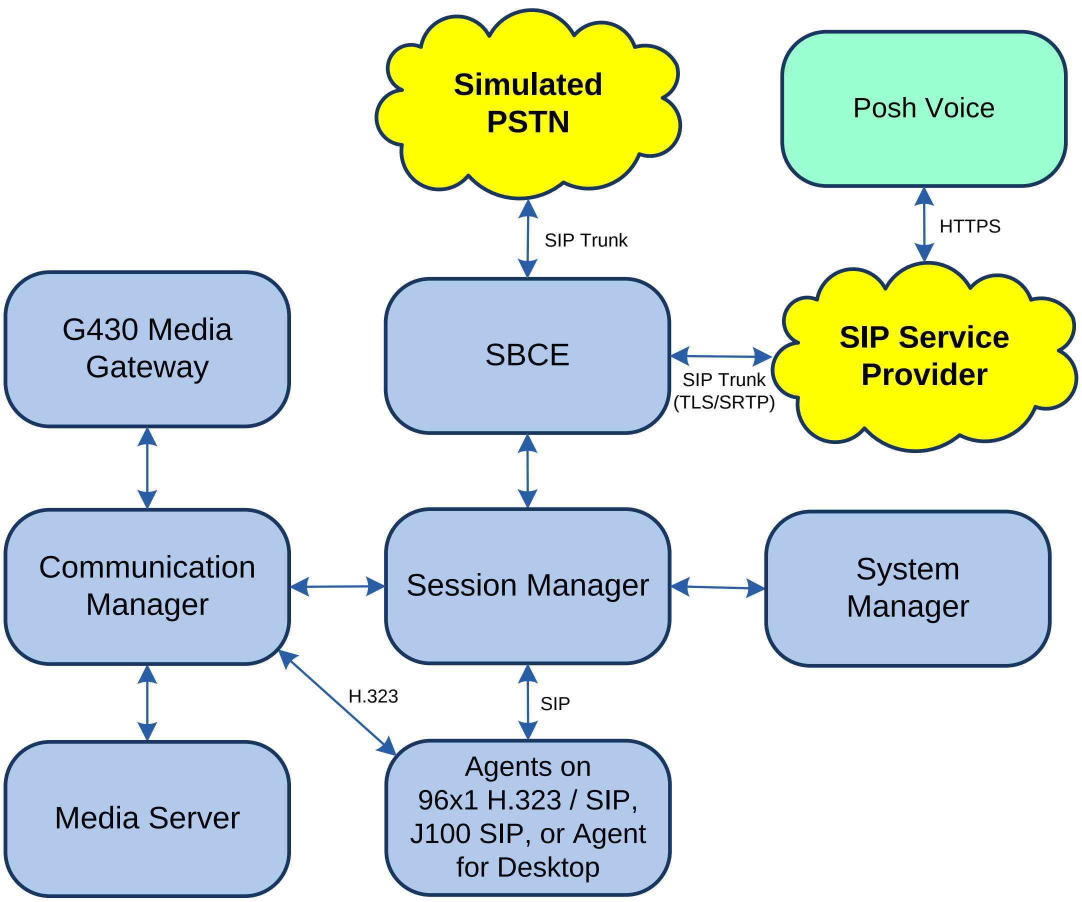


Figure 1: Avaya Aura® Environment with Posh Voice

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager	10.1.0.1.0-SP1
Avaya G430 Media Gateway	FW 42.8.0
Avaya Aura® Media Server	v.10.1.0.77
Avaya Aura® System Manager	10.1.0.1 Build No. – 10.1.0.0.537353 Software Update Revision No: 10.1.0.1.061394 Service Pack 1
Avaya Aura® Session Manager	10.1.0.1.1010105
Avaya Session Border Controller for Enterprise	10.1.1.0-35-21872
Avaya 96x1 Series IP Deskphones	6.8511 (H.323)
Avaya J100 Series IP Telephones	4.0.10.3.2 (SIP)
Avaya Agent for Desktop	2.0.6.0.10 (SIP)
Posh Voice	November 2022

5. Configure Avaya Aura® Communication Manager

This section covers the configuration steps required to establish a SIP trunk between Communication Manager and Session Manager, routing calls to Posh Voice, and a sample Vector that uses UUI and queue calls to a skill group. Communication Manager is configured through the System Access Terminal (SAT). The procedures include the following areas:

- Verify Licenses
- Administer IP Node Names
- Administer IP Codec Set
- Administer IP Network Region
- Administer SIP Trunk Group to Session Manager
- Administer AAR Call Routing
- Administer Call Center

Note: The configuration of the call center, including agents, skill/hunt group, and VDNs are outside the scope of these Application Notes, but note that Posh Voice transferred calls to a VDN, which pointed to a vector that queued the call to a split/hunt group, and eventually routed the call to an available agent or queued the call.

5.1. Verify Licenses

Using the SAT, enter the **display system-parameters customer-options** command to verify there is sufficient capacity for SIP trunks on **Page 2**. The license file installed on the system controls these options. If there is insufficient capacity of SIP Trunks or a required feature is not enabled, 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:  2400      2
         Maximum Administered Remote Office Trunks: 12000      0
Max Concurrently Registered Remote Office Stations:  2400      0
          Maximum Concurrently Registered IP eCons:  128       0
     Max Concur Reg Unauthenticated H.323 Stations:   100      0
                    Maximum Video Capable Stations:  36000     2
               Maximum Video Capable IP Softphones:   2400     2
                   Maximum Administered SIP Trunks: 12000     30
  Max Administered Ad-hoc Video Conferencing Ports: 12000      0
   Max Number of DS1 Boards with Echo Cancellation: 688    0

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

5.2. Administer IP Node Names

In the **IP Node Names** form, assign an IP address and host name for Communication Manager (*procr*) and Session Manager (*devcon-sm*). The host names will be used in other configuration screens of Communication Manager.

```
change node-names ip                                                Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
default             0.0.0.0
devcon-aes          10.64.102.119
devcon-ams          10.64.102.118
devcon-sm           10.64.102.117
procr               10.64.102.115
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
```

5.3. Administer IP Codec Set

In the **IP Codec Set** form, specify the audio codec to be used by Posh Voice SIP service provider. The form is accessed via the **change ip-codec-set** command. Note the codec set number since it will be used in the IP Network Region covered in the next section. For the compliance test, G.711MU was used. For all integrations, either G711MU or G.711A is required. For the compliance test, **Media Encryption** and **Encrypted SRTCP** was used as shown below, but it is not required between Communication Manager and SBCE.

```
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.711MU           n         2        20
 2:
 3:
 4:
 5:
 6:
 7:

     Media Encryption                   Encrypted SRTCP: best-effort
 1: 1-srtp-aescm128-hmac80
 2: 2-srtp-aescm128-hmac32
 3: none
 4:
 5:
```

5.4. Administer IP Network Region

In the **IP Network Region** form, specify the codec set to be used for Posh Voice SIP service provider and enable **IP-IP Direct Audio** (Shuffling), if desired. Shuffling allows audio traffic to be sent directly between IP endpoints and SBCE without using media resources in the Avaya Media Gateway or Avaya Aura® Media Server after call establishment. For this compliance test, shuffling was enabled. The **Authoritative Domain** for this configuration is *avaya.com*.

```
change ip-network-region 2                                         Page   1 of  20
                               IP NETWORK REGION
  Region: 2       NR Group: 2
Location: 1       Authoritative Domain: avaya.com
    Name: To Avaya SBCE         Stub Network Region: n
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 2              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

5.5. Administer SIP Trunk to Session Manager

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the **Signaling Group** form as follows:

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- The **Transport Method** field was set to *tls*, but *tcp* is also supported.
- Specify Communication Manager (*procr*) and the Session Manager (*devcon-sm*) as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form.
- Ensure that the appropriate TLS port value is specified (e.g., *5062)* in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*.
- **Direct IP-IP Audio Connections** is enabled to allow shuffling for calls routed over the trunk group associated with this signaling group.
- Set **Initial IP-IP Direct Media** field to *y*.

Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

```
add signaling-group 11                                      Page   1 of   2
                                SIGNALING GROUP

 Group Number: 11             Group Type: sip
  IMS Enabled? n       Transport Method: tls
        Q-SIP? n
     IP Video? y          Priority Video? n        Enforce SIPS URI for SRTP? n
  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: devcon-sm
 Near-end Listen Port: 5062                Far-end Listen Port: 5062
                                        Far-end Network Region: 2

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

Configure the **Trunk Group** form as shown below. This trunk group is used for SIP calls to the VoIP Service Provider. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Accept the default values for the remaining fields.

```
add trunk-group 11                                          Page   1 of   5
                                TRUNK GROUP

Group Number: 11                   Group Type: sip          CDR Reports: y
  Group Name: To SIP Service Provider     COR: 1        TN: 1        TAC: 1011
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 11
                                                     Number of Members: 10
```

On **Page 3** of the trunk group form, set the **Numbering Format** field to *private*. This field specifies the format of the calling party number sent to the far-end. Set **UUI Treatment** field to *shared*.

```
add trunk-group 11                                          Page   3 of   5
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                      Maintenance Tests? y



   Suppress # Outpulsing? n  Numbering Format: private
                                                UUI Treatment: shared
                                              Maximum Size of UUI Contents: 128
                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n



                                Modify Tandem Calling Number: no
               Send UCID? n



 Show ANSWERED BY on Display? y
```

On **Page 5** of the trunk group form, set the **Always Use re-INVITE for Display Updates** field to *y* so that Communication Manager sends a re-INVITE instead of an UPDATE when the Session Refresh interval expires. For the compliance test, the Posh Voice SIP service provider did not support the UPDATE method.

```
add trunk-group 11                                                Page   5 of   5
                              PROTOCOL VARIATIONS

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

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


                        Convert 180 to 183 for Early Media? n
                  Always Use re-INVITE for Display Updates? y
     Resend Display UPDATE Once on Receipt of 481 Response? 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
```

5.6. Administer AAR Call Routing

Configure the **Uniform Dial Plan** to steer calls to Posh Voice staging and production environments using extensions 78701 and 78702, respectively, to AAR as shown below.

```
change uniform-dialplan 7                                            Page   1 of   2
                       UNIFORM DIAL PLAN TABLE
                                                                   Percent Full: 0

  Matching                          Insert                 Node
  Pattern              Len Del      Digits       Net Conv Num
 78701                 5   0                     aar n
 78702                 5   0                     aar n
```

SIP calls to Session Manager are routed over a SIP trunk via AAR call routing. Configure the AAR analysis form and add entries to route calls to Posh Voice staging and production environments. 78701 and 78702 were used to route calls to Posh Voice staging and production environments, respectively. These calls were routed to route pattern 13 as shown below.

```
change aar analysis 78                                               Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all          Percent Full: 1

          Dialed          Total     Route     Call   Node  ANI
          String         Min  Max  Pattern    Type   Num   Reqd
    78701                5    5     13        lev0         n
    78702                5    5     13        lev0         n
```

Configure a preference in **Route Pattern** 13 to route calls over SIP trunk group 11 as shown below.

```
change route-pattern 13                                         Page   1 of   4
                    Pattern Number: 13     Pattern Name: Posh Voice
    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: 11   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                               lev0-pvt  none
 2: y y y y y n  n            rest                                         none
```

5.7. Administer Call Center

For the compliance test, a basic call center was configured on Communication Manager, consisting of agents, hunt/skill group, VDN, and vector. The call center configuration is outside the scope of these Application Notes and will not be covered, but the sample Vector used will be shown as it demonstrates the use of UUI data.

Device Type	Extension
VDN	77550
Skill Group	77500
Agent Stations	77301, 78004
Agent IDs	76301, 76302

Enter the **change variables** command to specify the variables that will be used to store the UUI data received from Posh Voice. In the following example, the UUI data was parsed into three variables of varying lengths. The number of variables and their length should be determined based on the UUI data.

```
change variables                                                Page   1 of  39
                              VARIABLES FOR VECTORS

Var Description                 Type    Scope Length Start Assignment       VAC
A                               asaiuui L     16     1
B                               asaiuui L     2      17
C                               asaiuui L     1      19
```

The following vector was invoked when VDN 77550 is called. The vector makes use of the aforementioned variables and then queues the call to skill group 50 to route the call to an available agent.

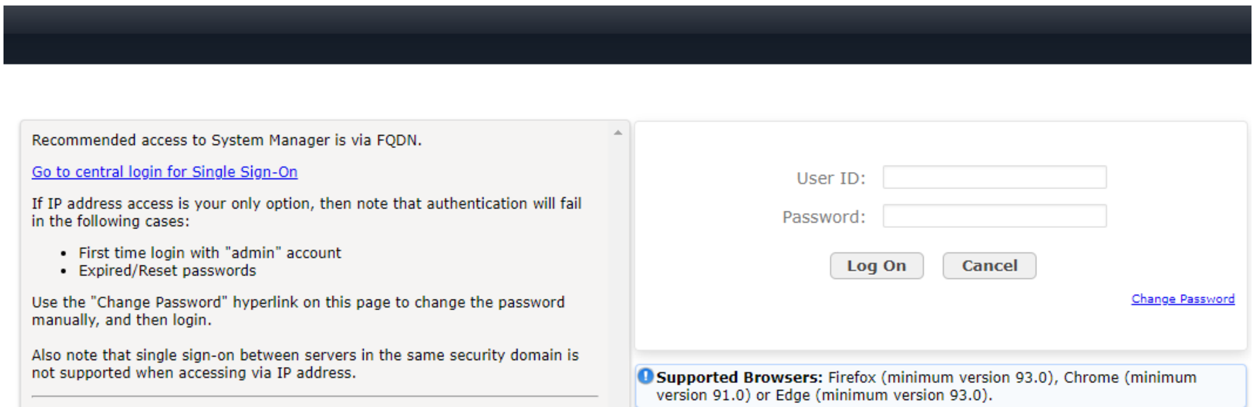
```
change vector 50                                                Page   1 of   6
                                  CALL VECTOR

    Number: 50               Name: Call Center
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    2   secs hearing ringback
02 goto step    5             if A                =      0
03 goto step    5             if B                =      0
04 goto step    5             if C                =      0
05 queue-to     skill 50   pri m
```

6. Configure Avaya Aura® Session Manager

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

- Adaptation
- SIP Entities for Communication Manager and SBCE
- Entity Links, which defines the SIP trunk parameters used by Session Manager when routing calls to/from Communication Manager and SBCE
- Routing Policies and Dial Patterns
- Session Manager, corresponding to Session Manager server to be managed by System 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.

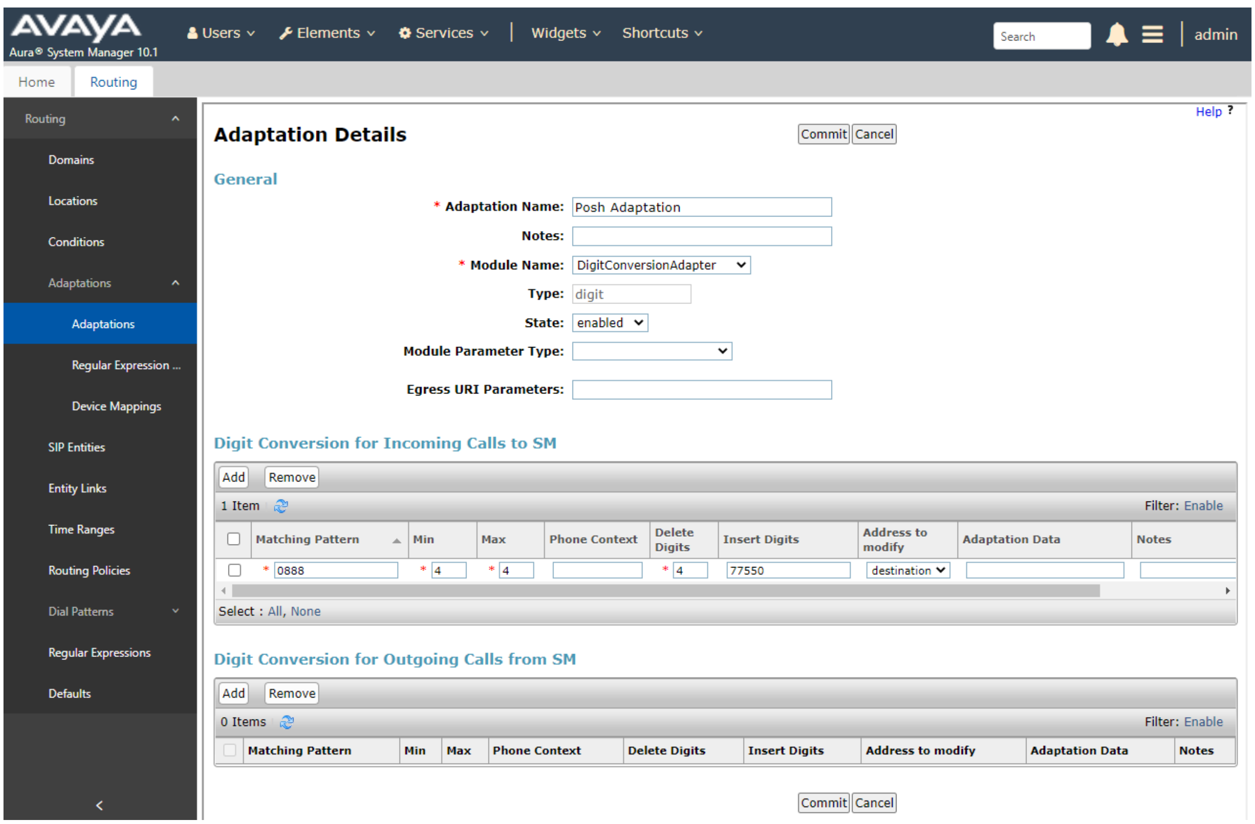


6.1. Add Adaptation

Posh Voice can transfer a call to a specific number, such as a VDN that routes call to an agent in a skill group or a PSTN number. If the number used by Posh Voice matches the destination exactly, then no Adaptation is required; otherwise, that number can be modified via an Adaptation. The following example modifies the digits sent by Posh Voice (e.g., 0974) to a VDN (e.g., 77550). This modifies the number in the Refer-To header of the SIP REFER.

To create an **Adaptation** that will be applied to the SBCE SIP entity in **Section 6.2.2**, navigate to **Elements → Routing → Adaptations** and click on the **New** button (not shown). In the **General** section, enter the following values. Use default values for all remaining fields.

- **Adaptation Name:** Enter a descriptive name for the Adaptation (e.g., *Posh Adaptation*).
- **Module Name:** Select *DigitConversionAdapter*.
- **Digit Conversion for Incoming Calls to SM:** Delete the incoming digits (e.g., *0888*) and replace with the appropriate number (e.g., *77550*) as shown below.



6.2. Add SIP Entities and Entity Links

In the sample configuration, two SIP Entities were added for Communication Manager and SBCE. This section also covers the configuration of the Entity Links.

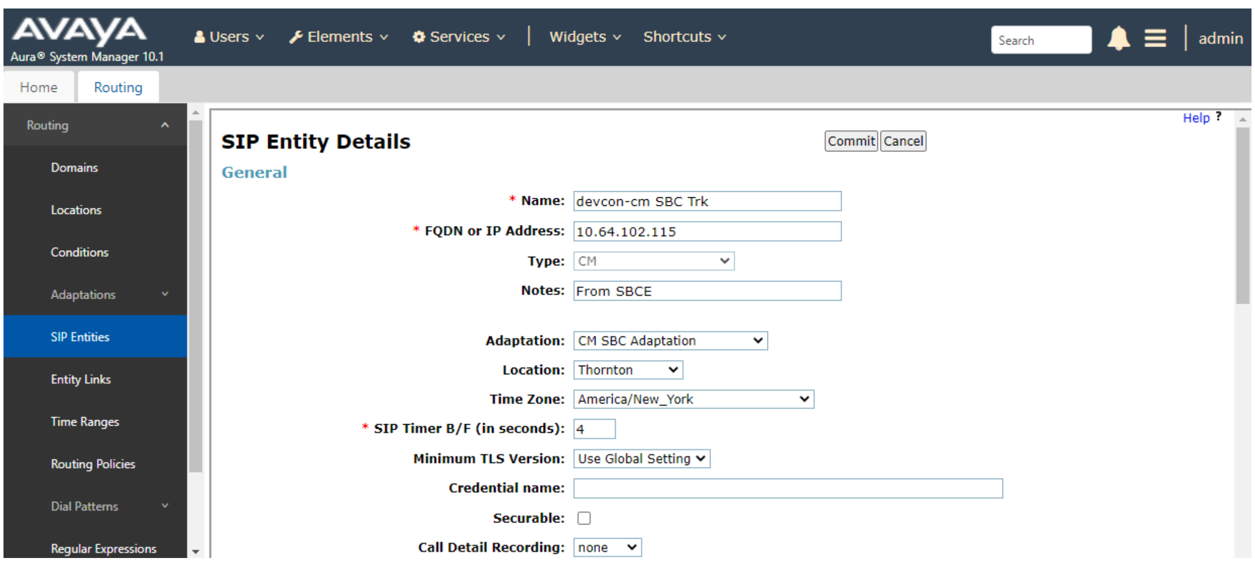
6.2.1. SIP Entity and Entity Link for Communication Manager

A SIP Entity must be added for Communication Manager. To add a SIP Entity, select **Elements → Routing → SIP Entities** from the top menu, followed by **New** in the subsequent screen (not shown) to add a new SIP entity for Communication Manager.

The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields and retain the default values for the remaining fields.

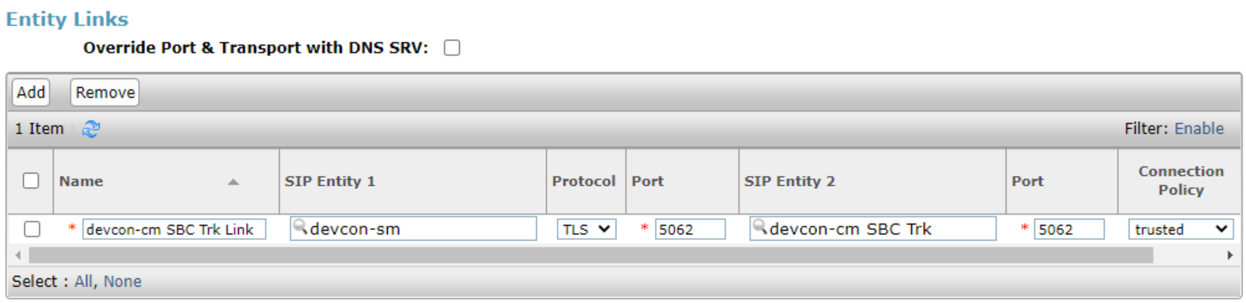
- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface (e.g., procr) on Communication Manager.
- **Type:** Select *CM*.
- **Location:** Select the appropriate pre-existing location name.
- **Time Zone:** Time zone for this location.

Default values can be used for the remaining fields.



Scroll down to the **Entity Links** sub-section and click **Add** to add an entity link. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name (e.g., *devcon-sm*).
- **Protocol:** Set to *TLS*. TCP may also be used between Communication Manager and Session Manager.
- **Port:** Set to appropriate TLS port (e.g., *5062)*.
- **SIP Entity 2:** The Communication Manager entity name from this section.
- **Port:** Set to appropriate TLS port (e.g., *5062)*.
- **Connection Policy:** Set to *trusted*.

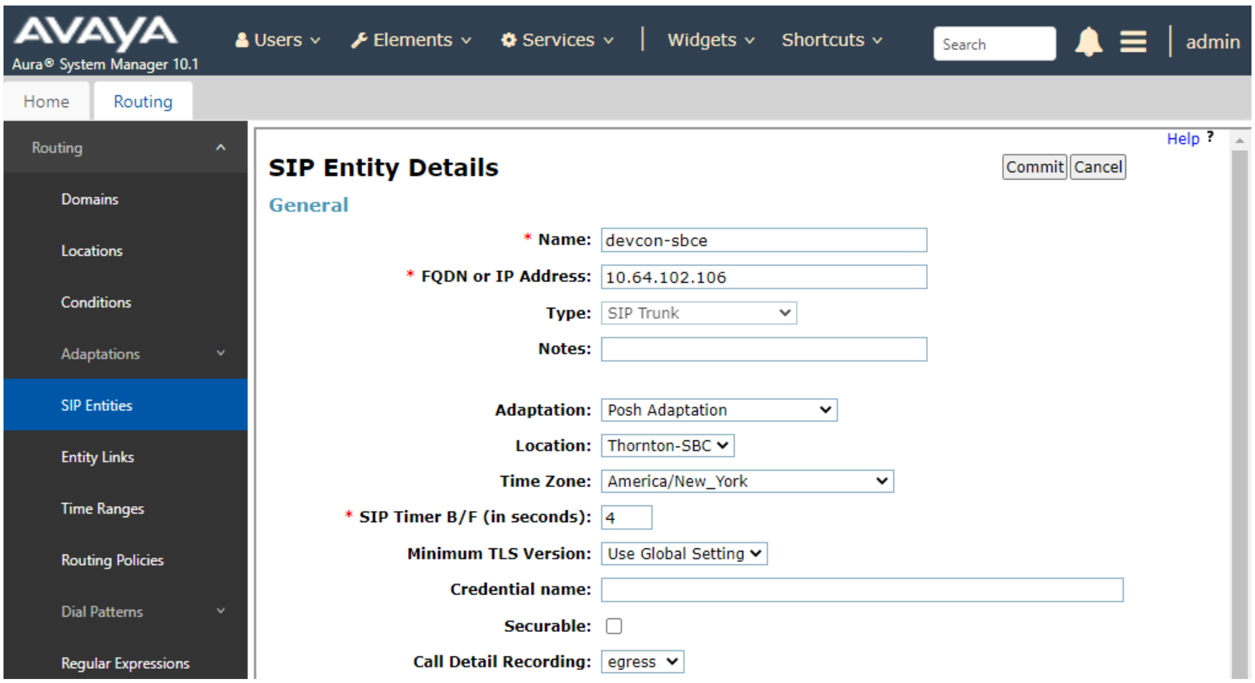


6.2.2. SIP Entity and Entity Link for SBCE

A SIP Entity must be added for SBCE. To add a SIP Entity, select **Elements → Routing → SIP Entities** from the top menu, followed by **New** in the subsequent screen (not shown) to add a new SIP entity for SBCE.

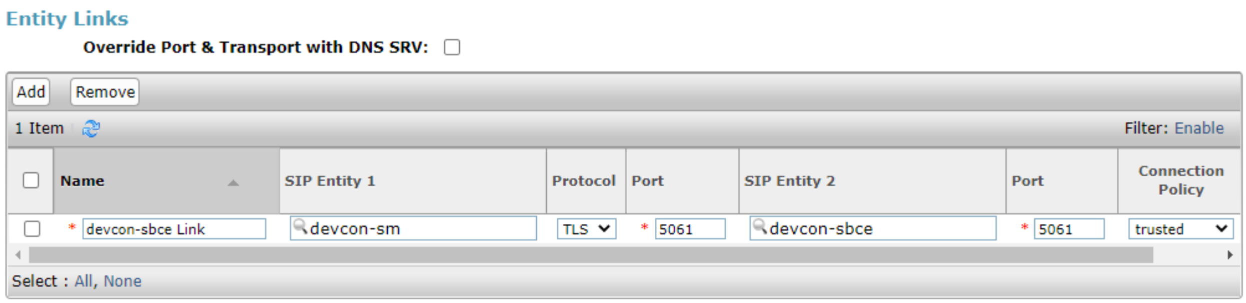
The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of the SBCE internal interface.
- **Type:** Select *SIP Trunk*.
- **Adaptation :** Select the Adaptation configured in **Section 6.1**.
- **Location:** Select the appropriate pre-existing location name.
- **Time Zone:** Time zone for this location.



Scroll down to the **Entity Links** sub-section and click **Add** to add an entity link. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name (e.g., *devcon-sm*).
- **Protocol:** Set to *TLS*. TCP may also be used between Session Manager and SBCE.
- **Port:** Set to appropriate TLS port (e.g., *5061)*.
- **SIP Entity 2:** The SBCE entity name from this section.
- **Port:** Set to appropriate TLS port (e.g., *5061)*.
- **Connection Policy:** Set to *trusted*.



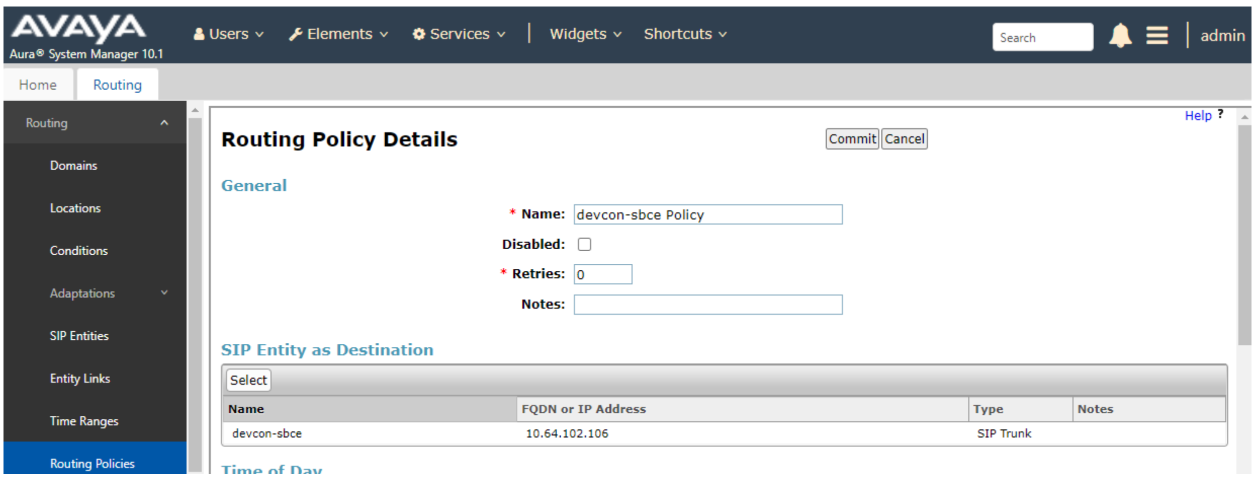
6.3. Add Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.2**. A routing policy was added for SBCE to route outgoing calls to the Posh Voice SIP service provider. To add a routing policy, select **Routing Policies** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:
Enter a descriptive name in **Name**.

Under *SIP Entity as Destination*:
Click **Select**, and then select the appropriate SIP entity to which this routing policy applies.

Defaults can be used for the remaining fields. Click **Commit** to save the Routing Policy definition.



6.4. Add Dial Patterns

Dial patterns are defined to direct calls to the appropriate SIP Entity. In the sample configuration, extensions 78701 and 78702 were routed to Posh Voice staging and production environments, respectively, through SBCE.

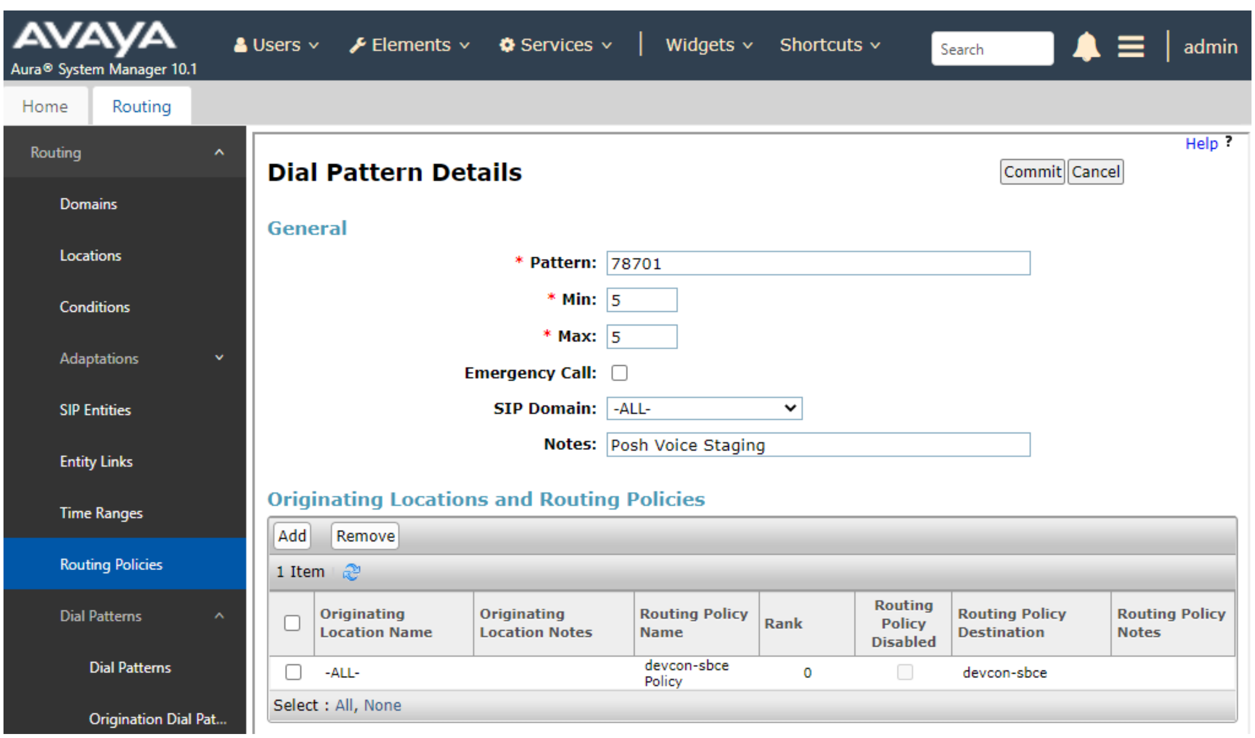
To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button (not shown) on the right. Fill in the following:

Under *General*:

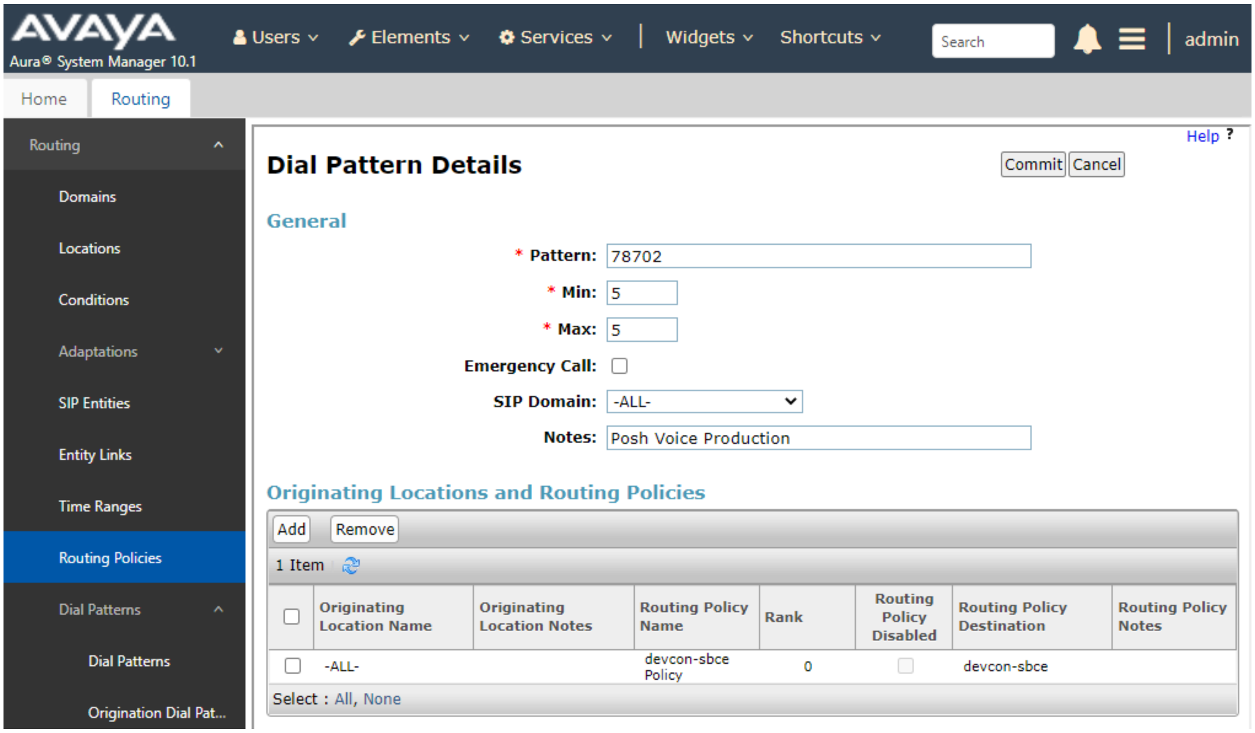
- **Pattern:** Dialed number or prefix.
- **Min** Minimum length of dialed number.
- **Max** Maximum length of dialed number.
- **SIP Domain** SIP domain of dial pattern.
- **Notes** Comment on purpose of dial pattern (optional).

Under *Originating Locations and Routing Policies*:
Click **Add**, and then select the appropriate location and routing policy from the list.

Default values can be used for the remaining fields. Click **Commit** to save this dial pattern. The following screen shows the dial pattern definition for routing calls to Posh Voice staging.



The following screen shows the dial pattern definition for routing calls to Posh Voice production.



7. Configure Avaya Session Border Controller for Enterprise

This section covers the configuration of Avaya SBCE. Avaya SBCE provides SIP connectivity to Session Manager, POSH Voice SIP service provider, and the PSTN.

This section covers the following SBCE configuration:

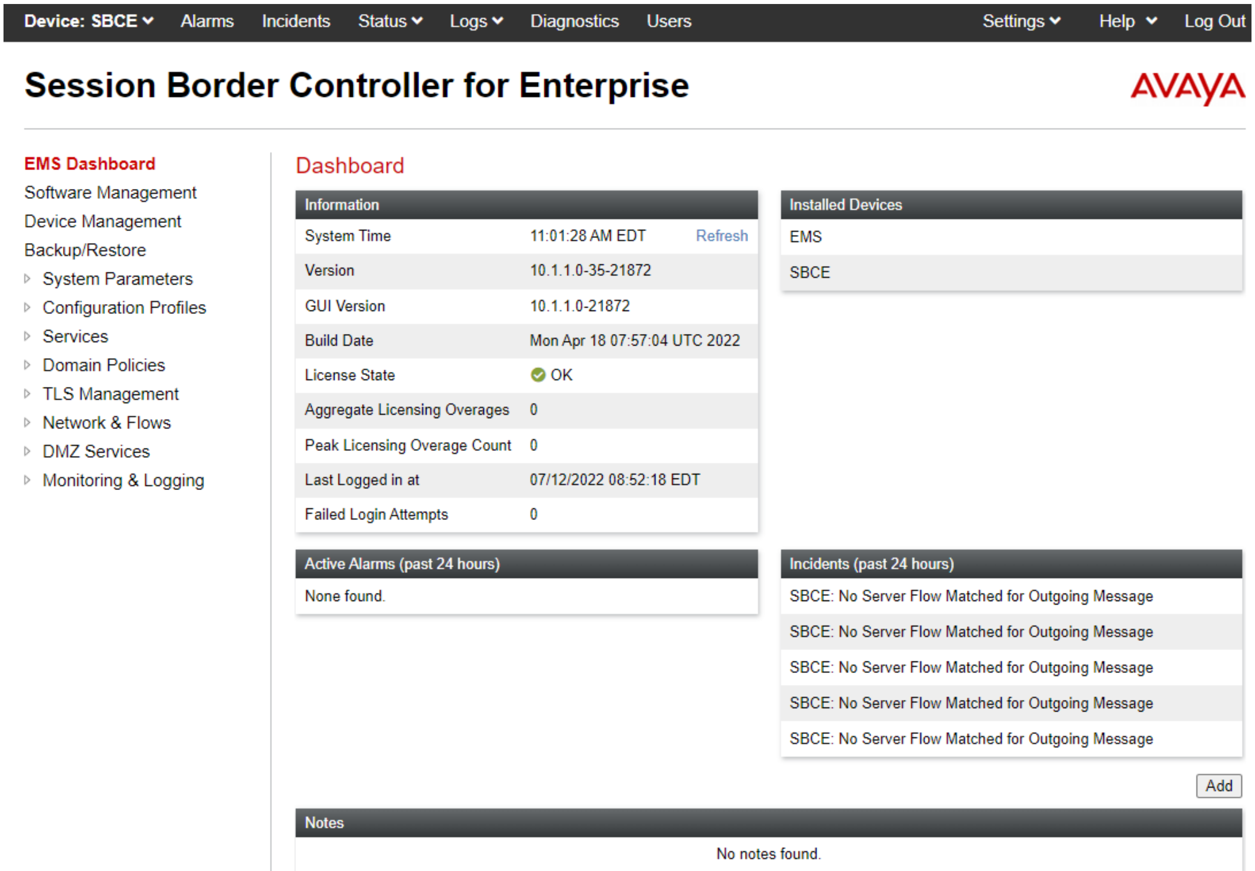
- Launch SBCE Web Interface
- Administer SIP Servers
- Administer Routing Profiles
- Administer URI Groups
- Administer Media Rules
- Administer End Point Policy Groups
- Administer TLS Management
- Administer Media Interfaces
- Administer Signaling Interfaces
- Administer End Point Flows

Note: This section will focus on routing and connectivity for Session Manager and Posh Voice. The configuration for the PSTN is not covered. For security reasons, public IP addresses and FQDNs will be redacted in these Application Notes.

7.1. Launch SBCE Web Interface

Access the SBCE web interface by using the URL **https://<*ip-address*>/sbc** in an Internet browser window, where **<*ip-address*>** is the IP address of the SBCE management interface. The screen below is displayed. Log in using the appropriate credentials.

After logging in, the Dashboard will appear as shown below. All configuration screens of the SBCE are accessed by navigating the menu tree in the left pane. Select **Device → SBCE** from the top menu.



7.2. Administer Server Interworking Profiles

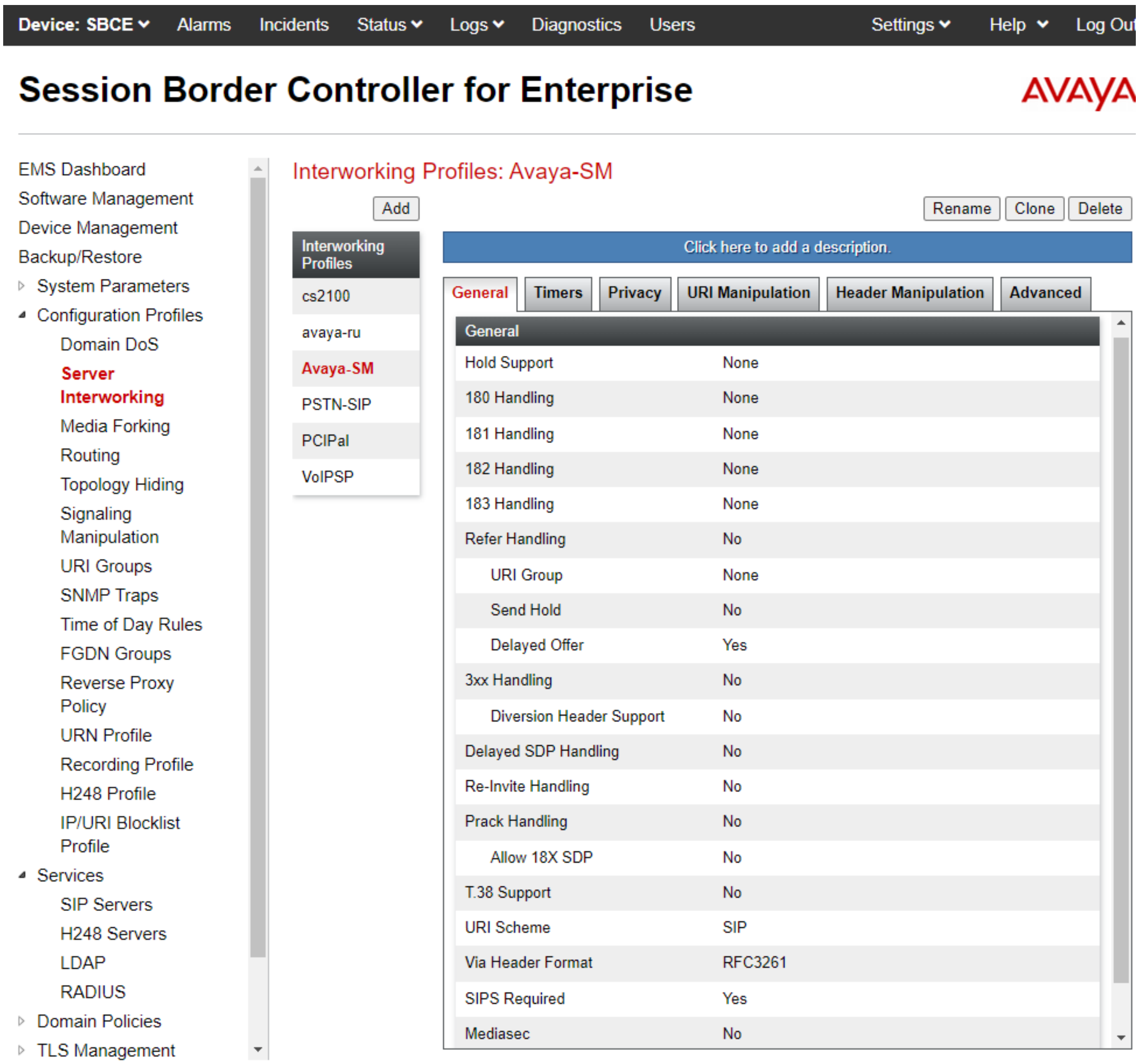
A server interworking profile defines a set of parameters that aid in interworking between the SBCE and a connected server. **Server Interworking** profiles were added for Session Manager and a default Server Interworking profile was used for the Posh Voice IP service provider.

7.2.1. Server Interworking Profile for Posh Voice SIP Service Provider

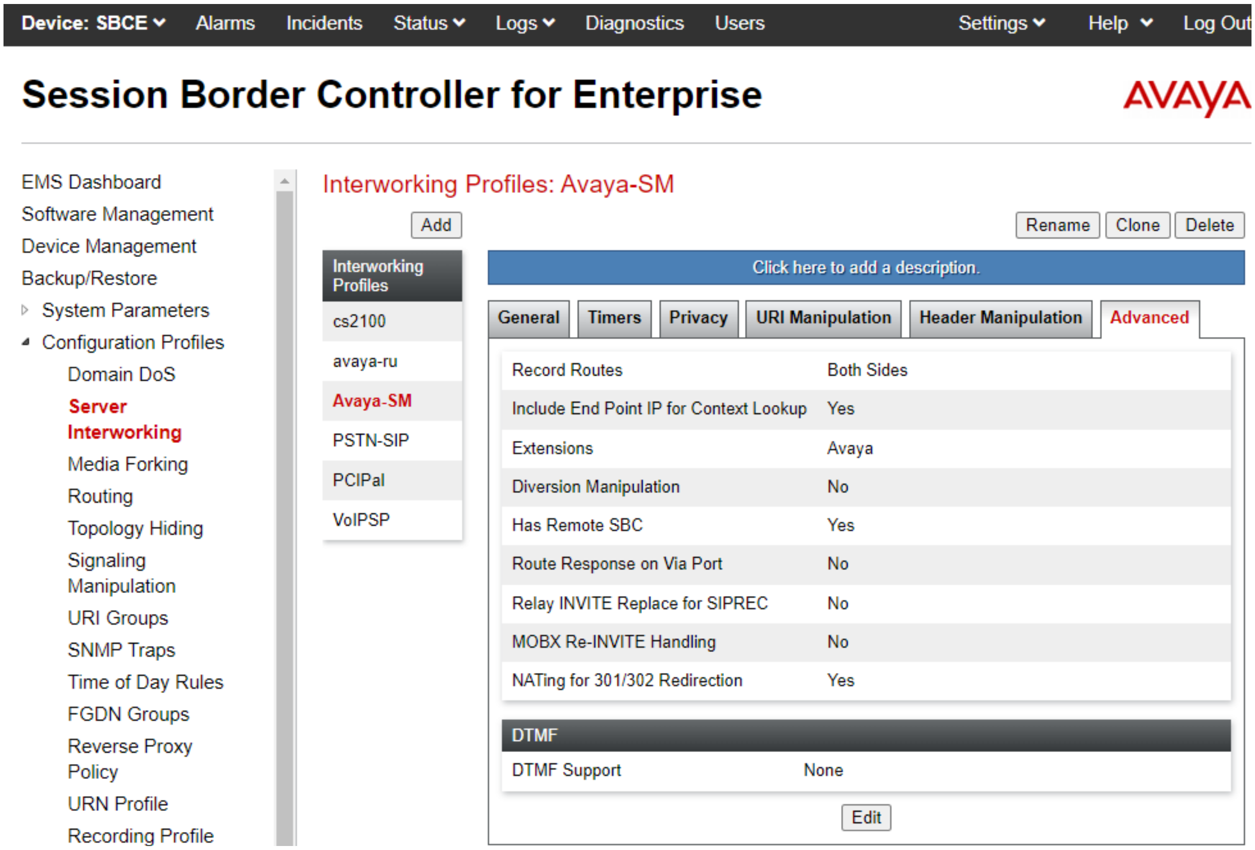
No server interworking profile was required for the Posh Voice SIP service provider.

7.2.2. Server Interworking Profile for Session Manager

Session Manager profile was cloned from the same **avaya-ru** profile. The **General** tab below shows the default settings.



Select the **Advanced** tab and configure as shown in the screen capture below.



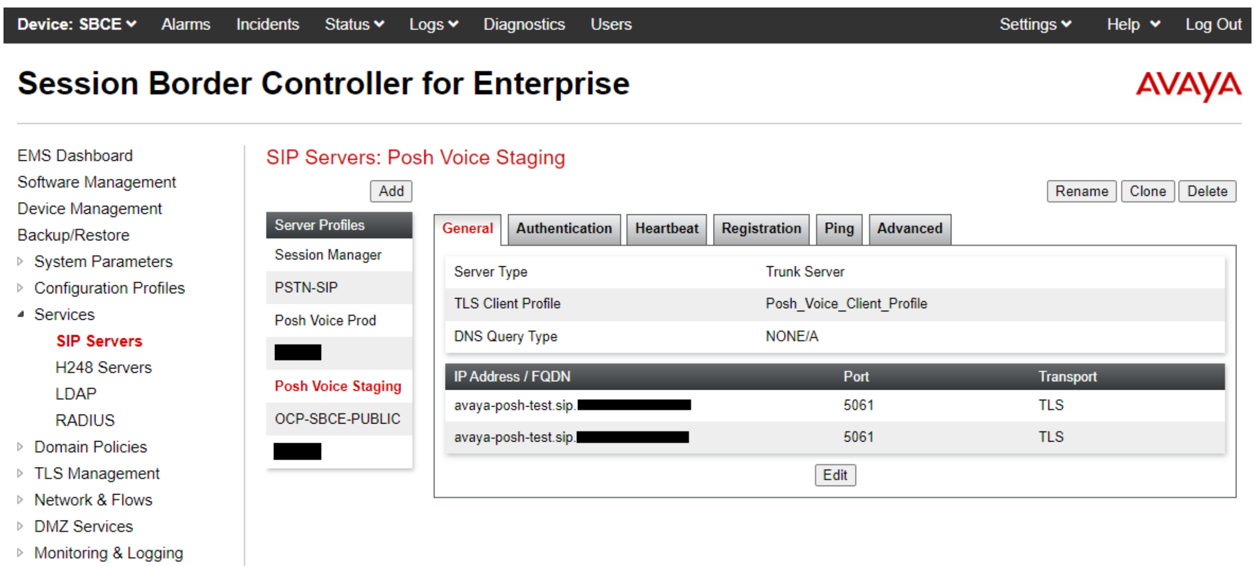
7.3. Administer SIP Servers

A SIP server definition is required for each server connected to SBCE. Two SIP servers were configured for Posh Voice: one for Posh Voice staging and one for Posh Voice production. Add **SIP Servers** for Posh Voice and Session Manager. TLS transport was used for the SIP trunks to Session Manager and the Posh Voice SIP service provider.

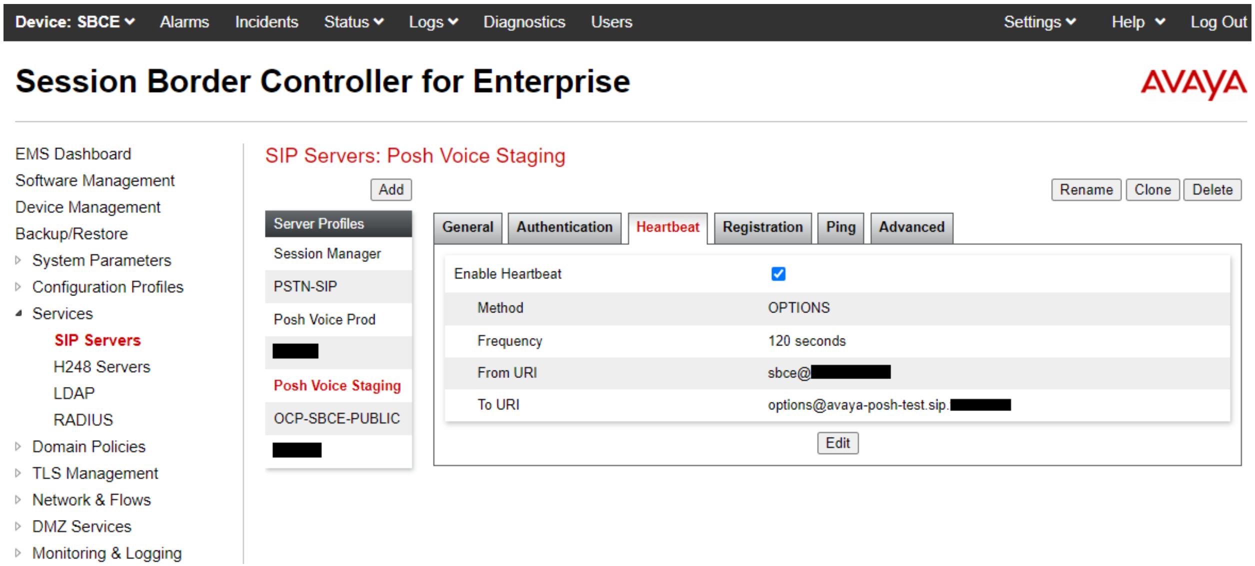
Note: TLS profiles were preconfigured for Session Manager and are not shown in these Application Notes. However, TLS profile configuration for Posh Voice SIP service provider is shown in **Section 7.8**.

7.3.1. SIP Server for Posh Voice Staging SIP Service Provider

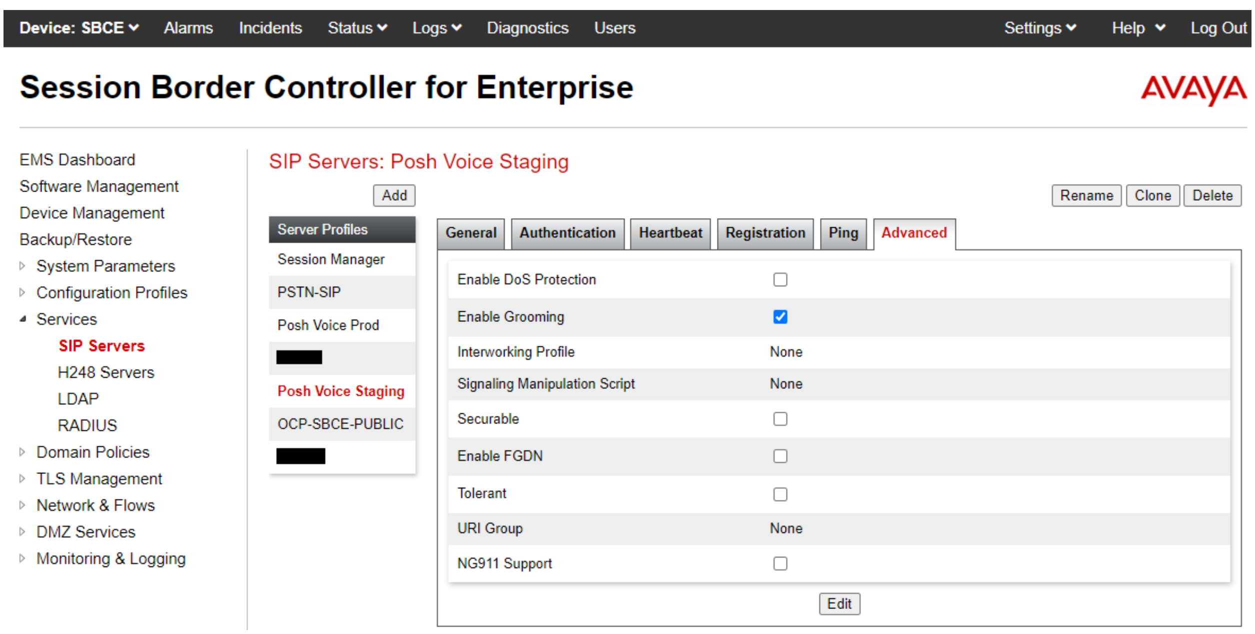
The **General** tab of the Posh Voice staging SIP server was configured as shown below. The *Posh Voice Staging* SIP server consists of a primary and a secondary site; hence, the two FQDNs. TLS transport was used for the Posh Voice SIP service provider SIP trunk. The configuration of the **TLS Client Profile** is shown in **Section 7.8**.



The **Heartbeat** tab was configured as shown below for Posh Voice staging. This allows SBCE to send SIP OPTIONS to the Posh Voice SIP service provider.

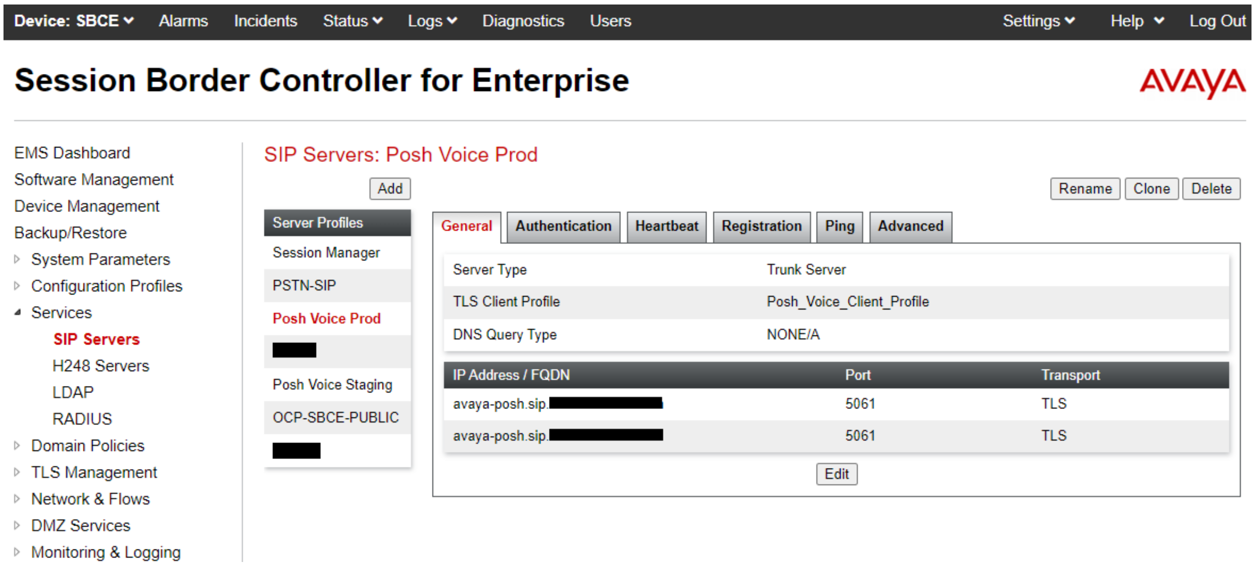


The **Advanced** tab was configured as shown below for Posh Voice staging. Grooming was enabled. All other tabs were left with their default values.

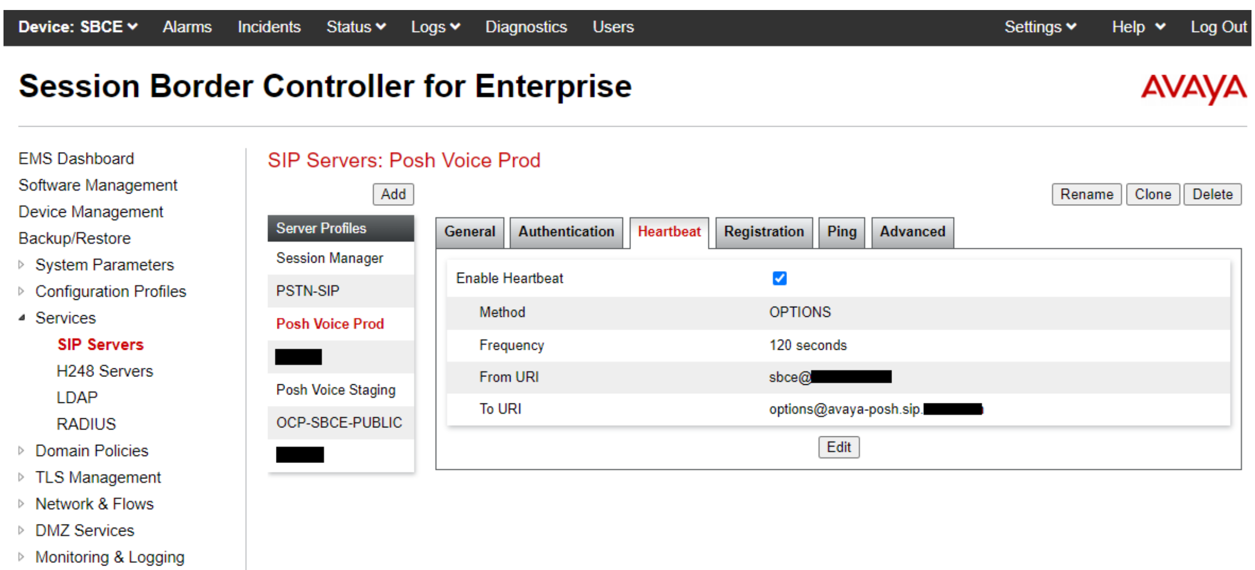


7.3.2. SIP Server for Posh Voice Production SIP Service Provider

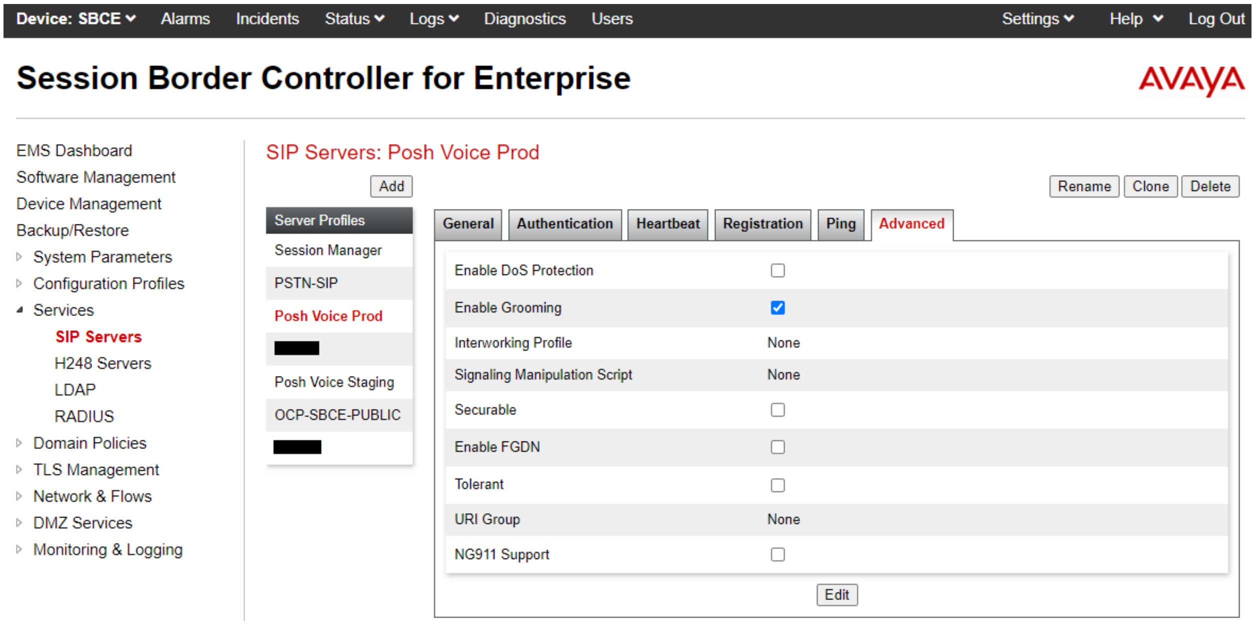
The **General** tab of the Posh Voice production SIP server was configured as shown below. The *Posh Voice Prod* SIP server consists of a primary and secondary site; hence, the two FQDNs. TLS transport was used for the Posh Voice SIP service provider SIP trunk. The configuration of the **TLS Client Profile** is shown in **Section 7.8**.



The **Heartbeat** tab was configured as shown below for Posh Voice staging. This allows SBCE to send SIP OPTIONS to the Posh Voice SIP service provider.

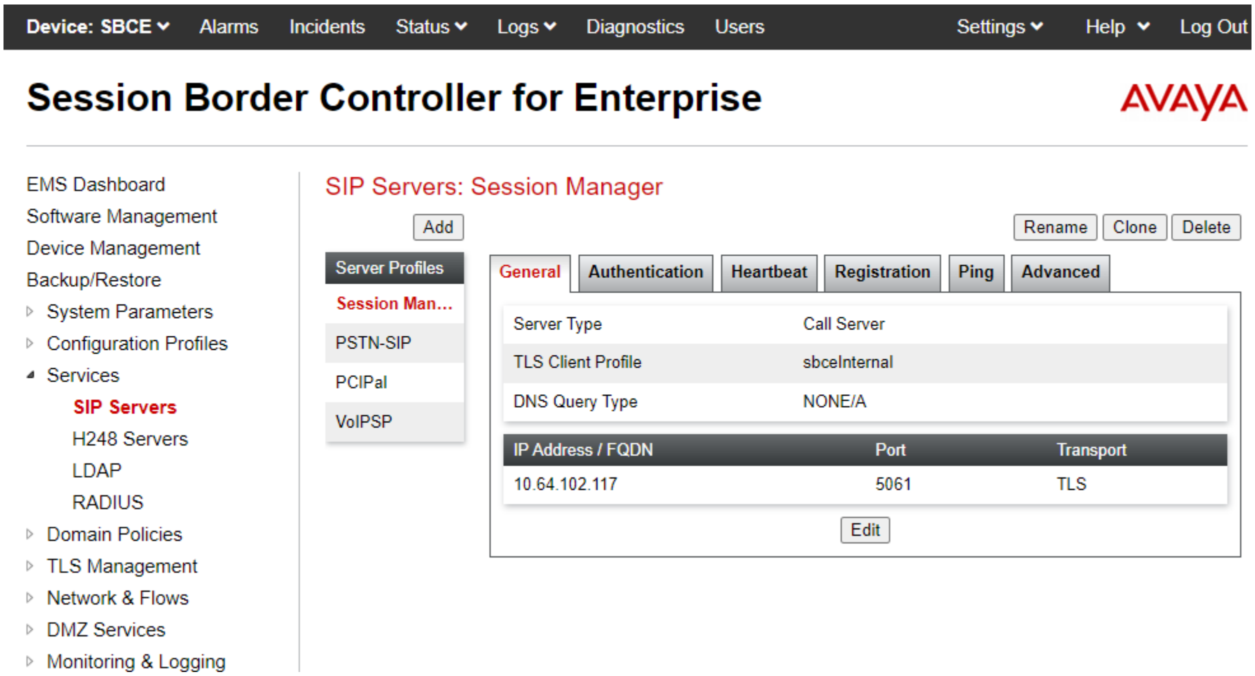


The **Advanced** tab was configured as shown below for Posh Voice staging. Grooming was enabled. All other tabs were left with their default values.

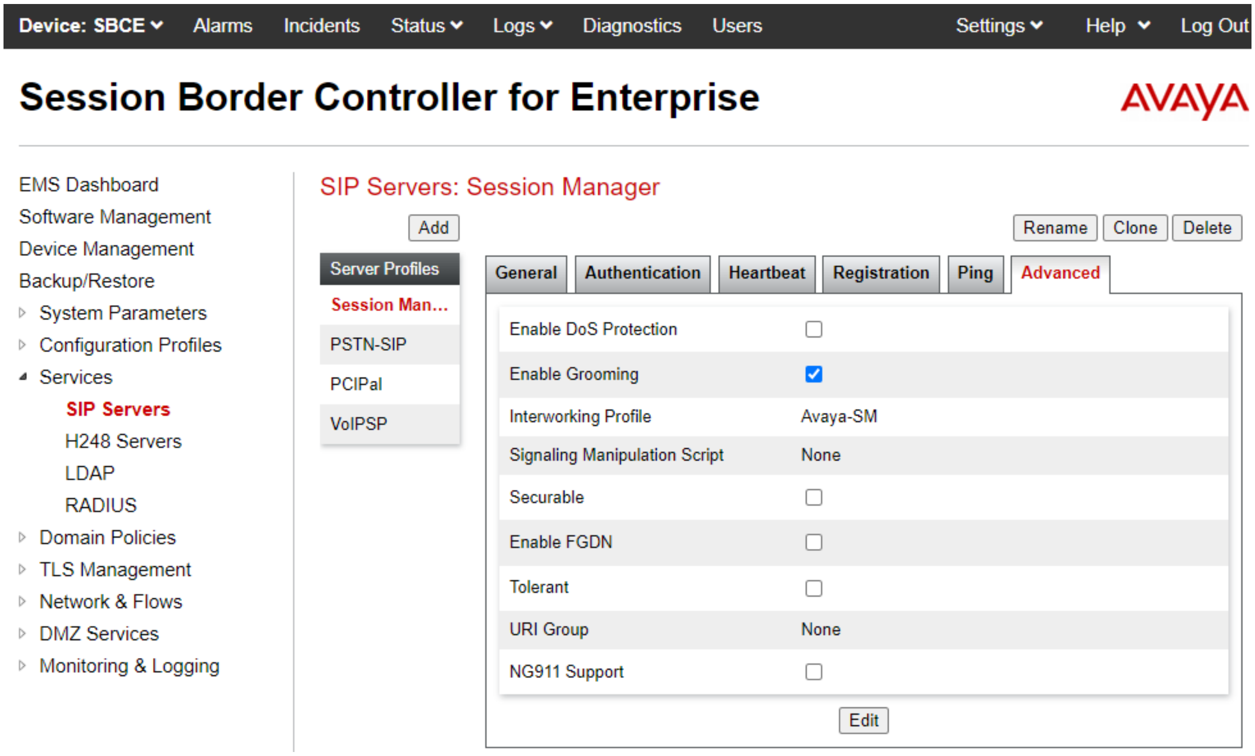


7.3.3. SIP Server for Session Manager

To define a SIP server, navigate to **Services → SIP Servers** from the left pane to display the existing SIP server profiles. Click **Add** to create a new SIP Server or select a pre-configured SIP server to view its settings. The **General** tab of the Session Manager SIP Server was configured as follows. TLS transport was used for the Session Manager SIP trunk.



The **Advanced** tab was configured as follows. Note that **Interworking Profile** was set to the one configured in **Section 7.2.2**. All other tabs were left with their default values.

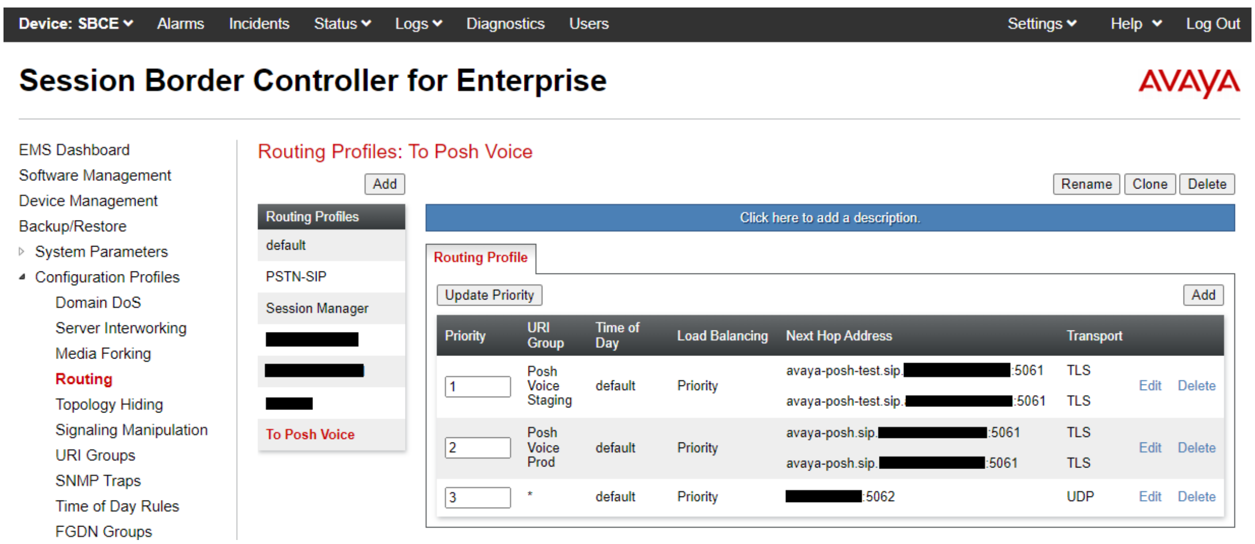


7.4. Administer Routing Profiles

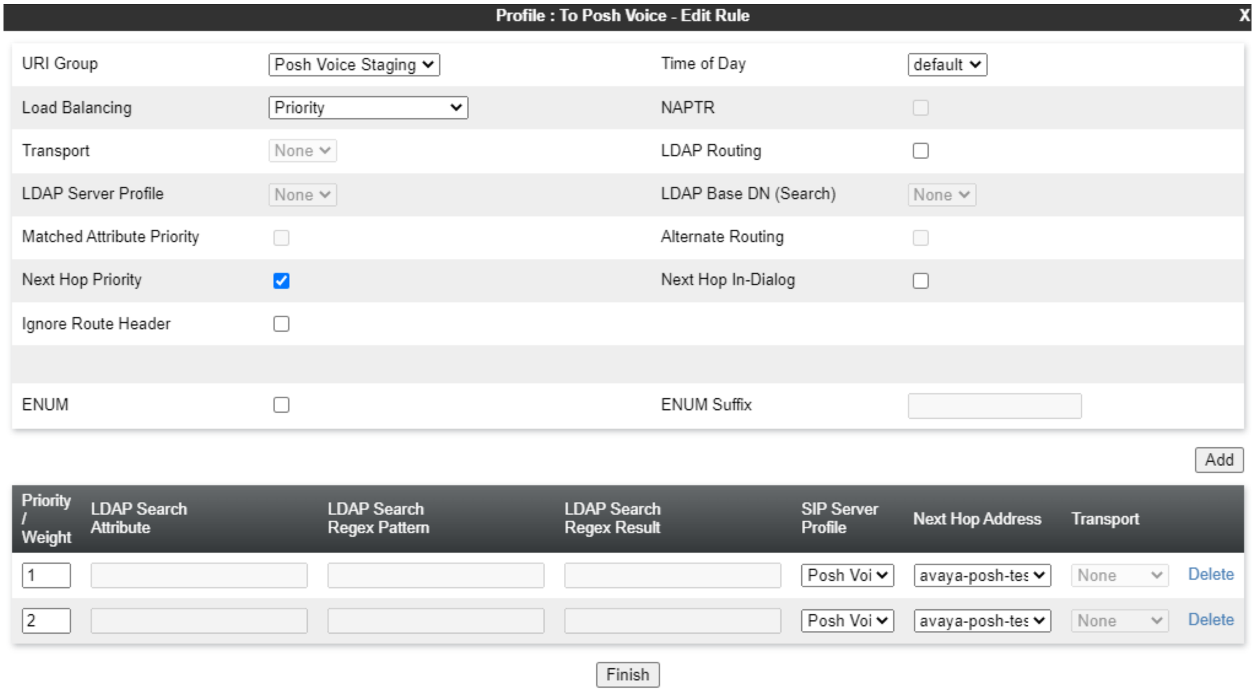
A routing profile is used to specify the next-hop for a SIP message. A routing profile is applied only after the traffic has matched an End Point Flow defined in **Section 7.11**. The IP addresses and ports defined here will be used as destination addresses for signaling. Create a routing profile for Posh Voice and Session Manager.

7.4.1. Routing Profile for Posh Voice

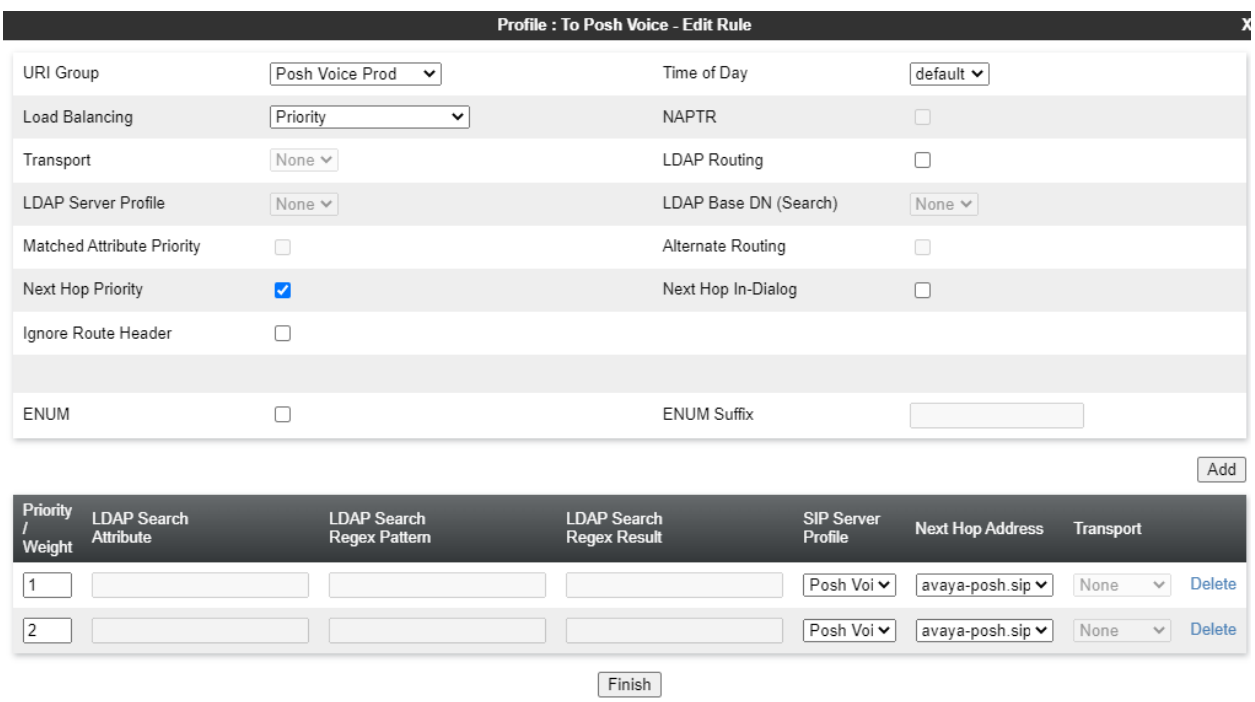
A routing profile was added for routing calls to Posh Voice staging and Posh Voice production based on the URI group. The routing profile was named *To Posh Voice*. This routing profile contains two routing rules. The first routing rule with **Priority** of *1* is used to route calls to Posh Voice staging if the number in the To header of the SIP INVITE matches the URI Group *Posh Voice Staging* configured in **Section 7.5**. The second routing rule with **Priority** of *2* is used to route calls directly to Posh Voice production if the number in the To header of the SIP INVITE matches the URI Group *Posh Voice Production* configured in **Section 7.5**. The third routing rule with **Priority** of 3 routes the call to the PSTN.



The details of the first routing rule of Routing Profile, *To Posh Voice*, is shown below. It contains two routing preferences corresponding to two different Posh Voice staging SIP service provider locations.



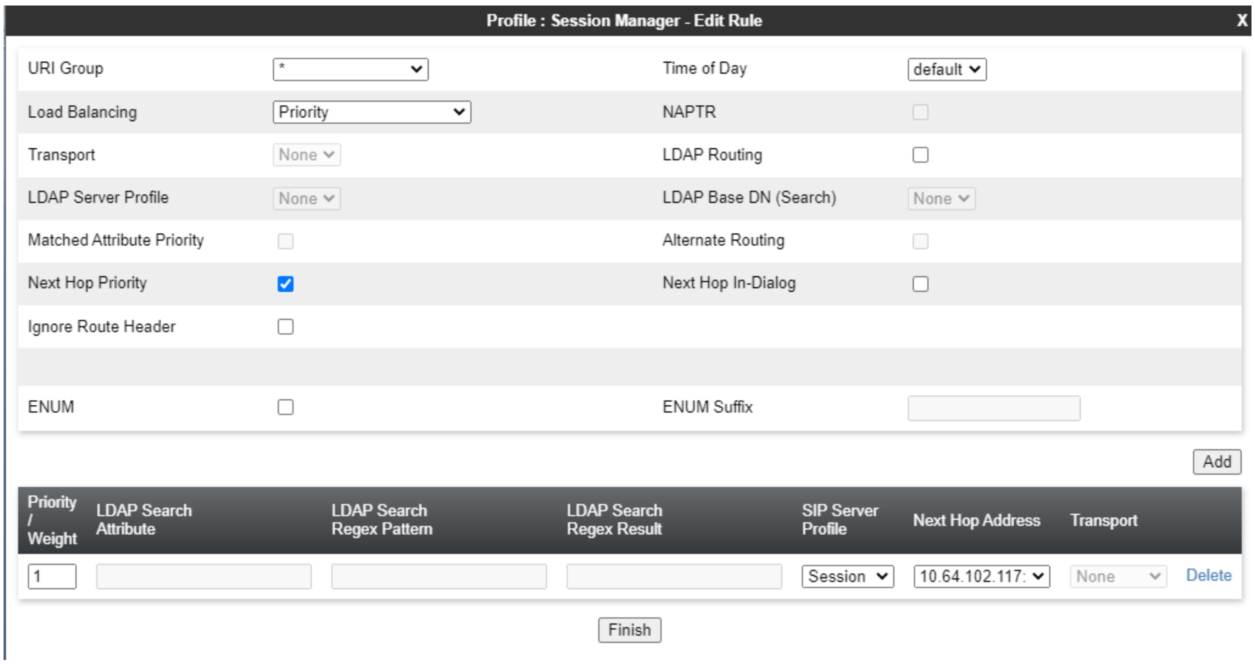
The details of the second routing rule are shown below. It contains two routing preferences corresponding to two different Posh Voice production SIP service provider locations.



7.4.2. Routing Profile for Session Manager

To create a new profile, navigate to **Configuration Profiles → Routing** in the left pane. In the center pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new profile, followed by series of pop-up windows in which the profile parameters can be configured. To view the settings of an existing profile, select the profile from the center pane.

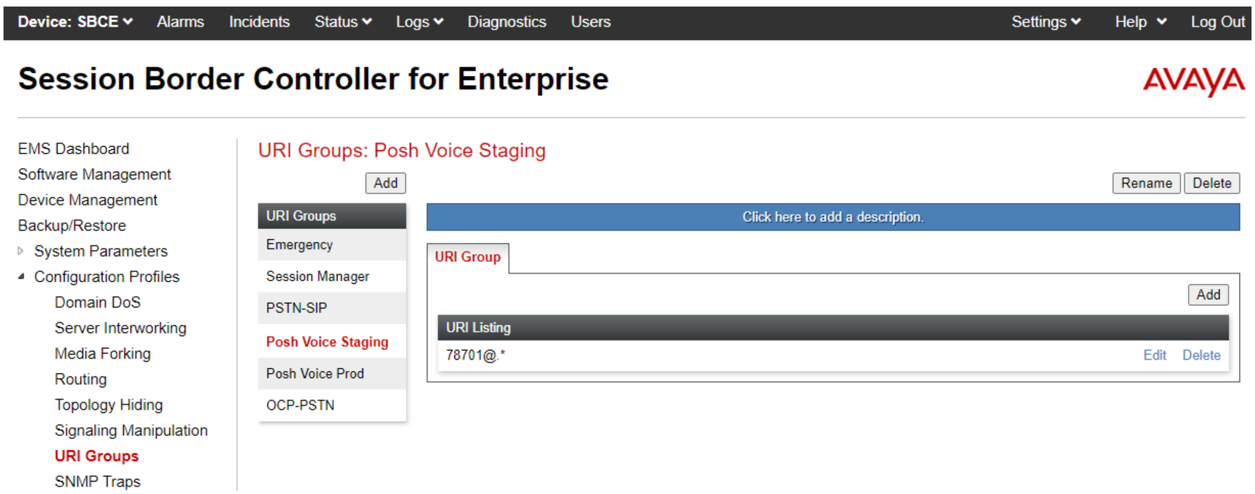
The routing profile for calls to Session Manager is shown below. The routing profile was named *SessionManager*. This routing profile contains the IP address of the signaling interface of Session Manager.



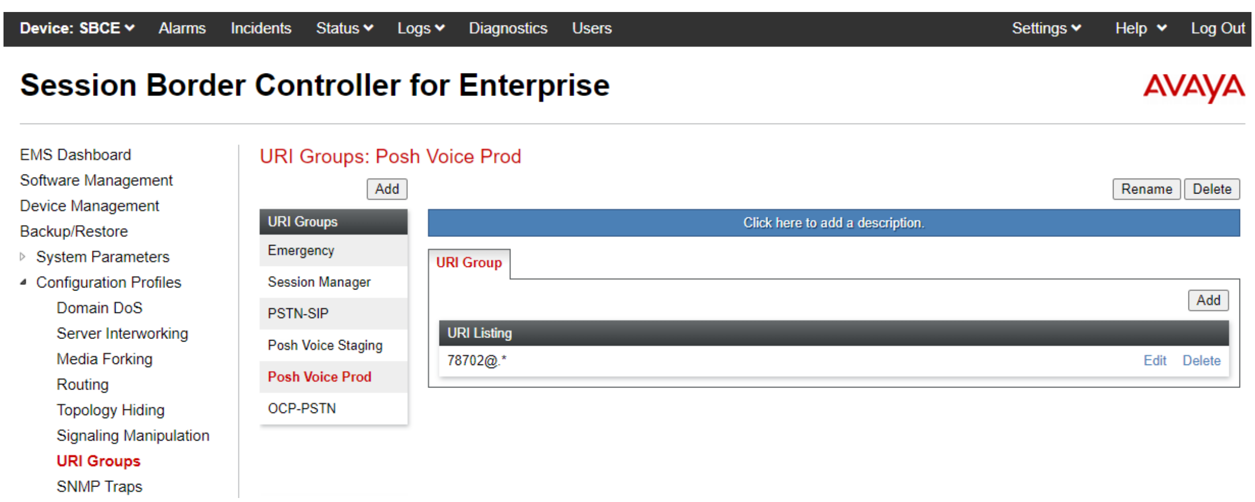
7.5. Administer URI Groups

URI Groups were used to aid in routing calls to the Posh Voice staging and production environments. For this solution, two **URI Groups** named *Posh Voice Staging* and *Posh Voice Prod* were created as shown below. *Posh Voice Staging* URI group specified a URI with 78701, and the *Posh Voice Prod* URI group specified a URI with 78702. These URI groups were specified in the routing profile configured in **Section 7.4.1**. If the To header in the SIP INVITE matched *78701*, the call would be routed to Posh Voice staging, and if it matched *78702*, the call would be routed to Posh Voice production. If it didn't match either URIs, the call would be routed to Session Manager.

The *Posh Voice Staging* URI group is shown below. It was configured with **Type** set to *Regular Expression*.



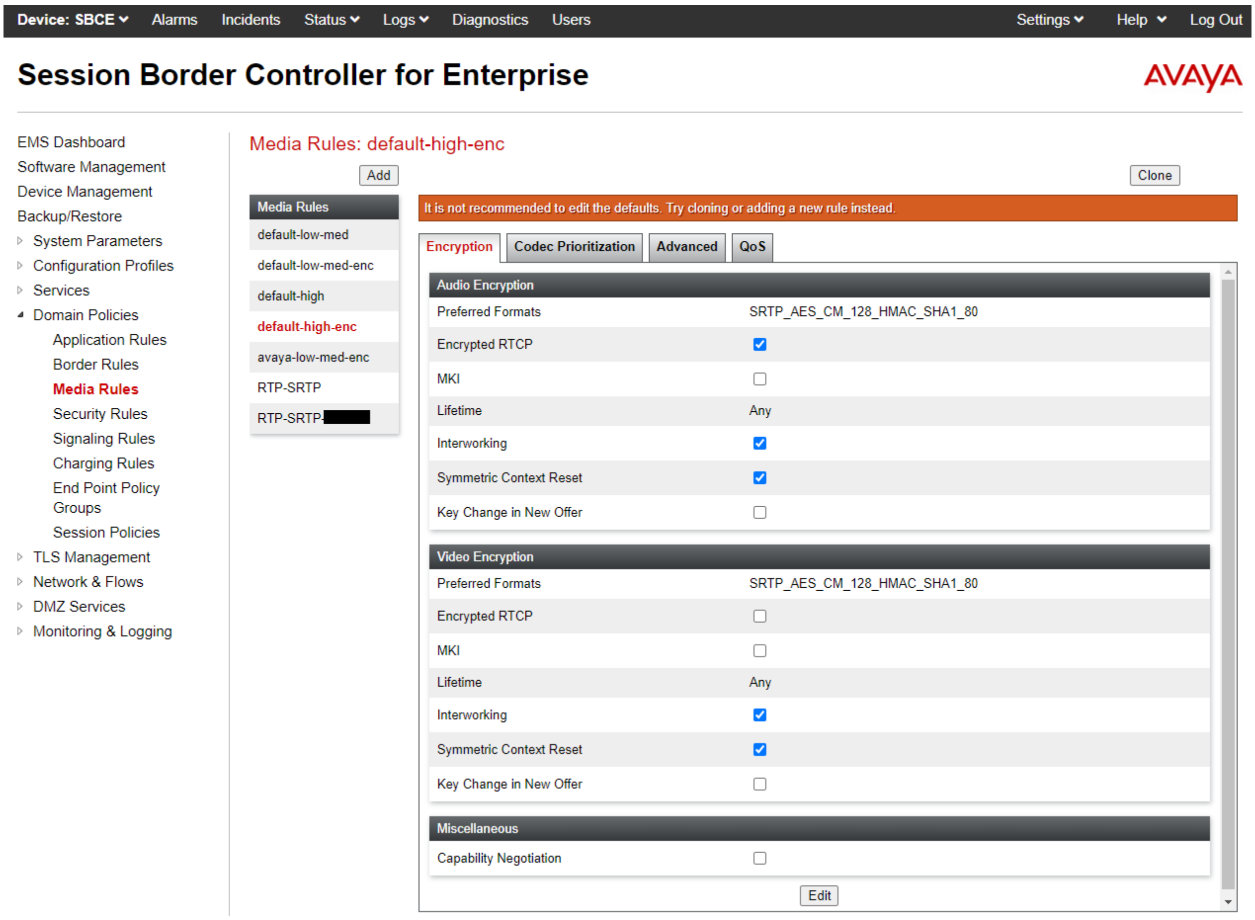
The *Posh Voice Prod* URI group is shown below. It was configured with **Type** set to *Regular Expression*.



7.6. Administer Media Rules

A media rule defines the processing to be applied to the selected media. A media rule is one component of the larger endpoint policy group defined in **Section 7.7**.

To view an existing rule, navigate to **Domain Policies → Media Rules** in the left pane. For the compliance test, a pre-existing media rule was used, *default-high-enc*. In the center pane, select the rule (e.g., *default-high-enc*) to view its default settings. The **Encryption** tab was configured as shown below.

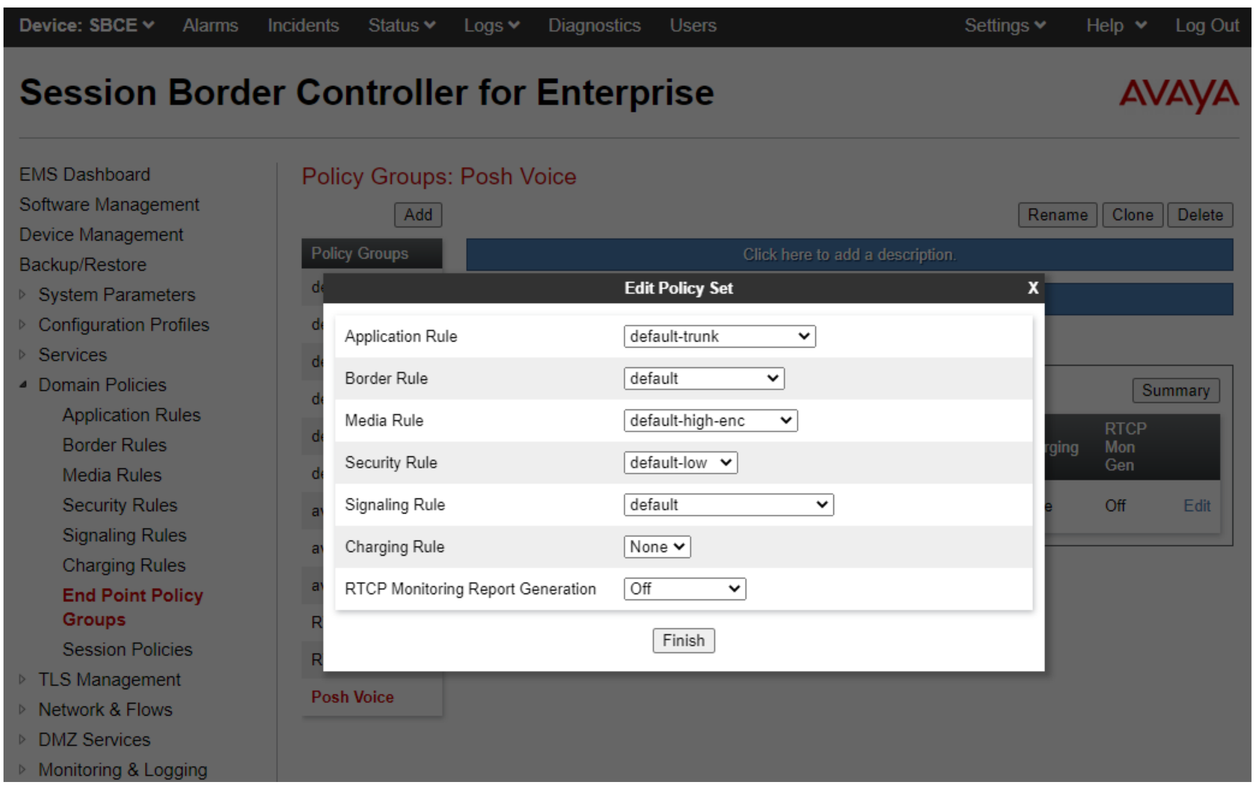


7.7. Administer End Point Policy Groups

An endpoint policy group is a set of policies that will be applied to traffic between the SBCE and an endpoint (connected server). The endpoint policy group is applied to the traffic as part of the endpoint flow defined in **Section 7.11**.

To create a new group, navigate to **Domain Policies → End Point Policy Groups** in the left pane. In the right pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new group, followed by the **Policy Group** window (not shown) to configure the group parameters. Once complete, the settings will be displayed. To view the settings of an existing group, select the group from the list. The settings will appear in the right pane.

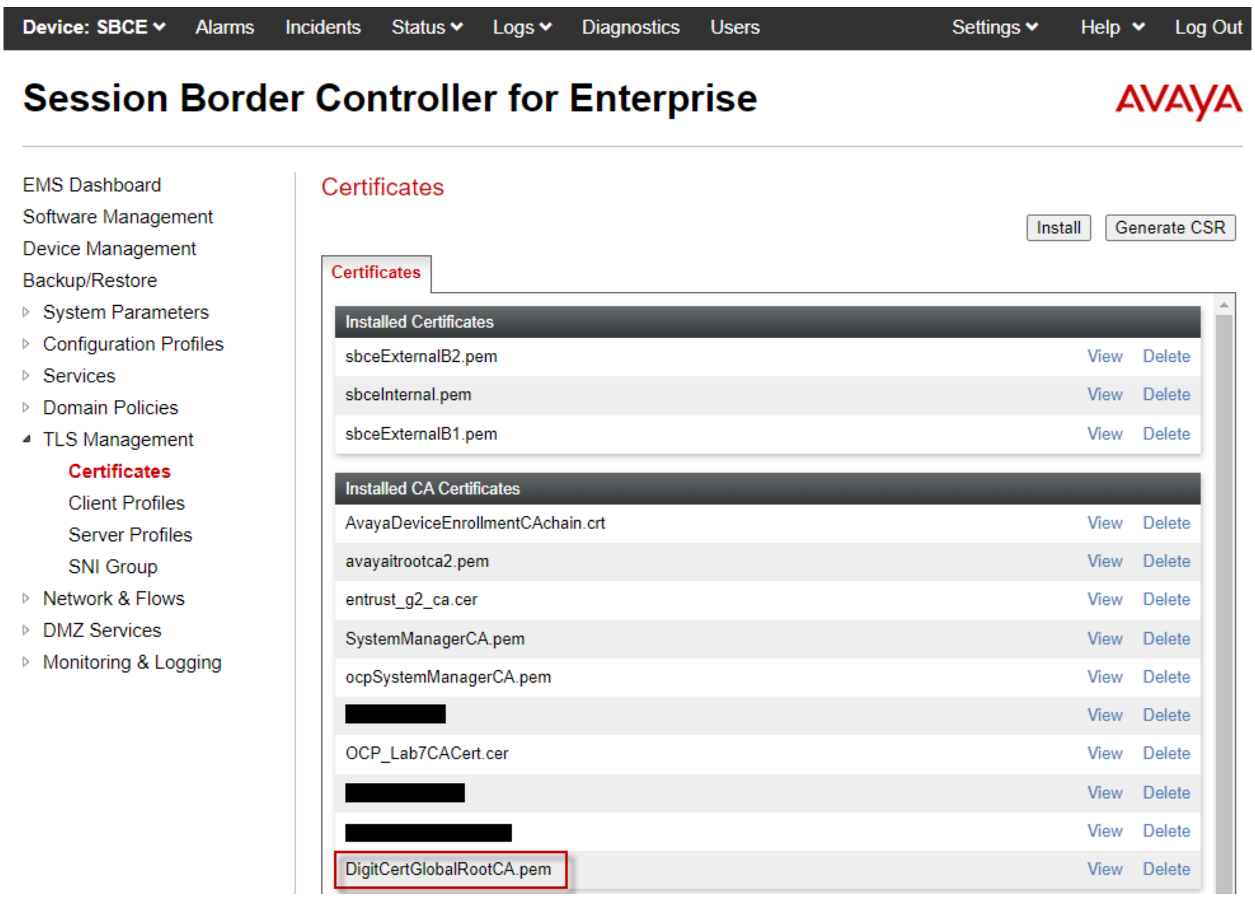
The new endpoint policy group, named *Posh Voice*, is shown below and is assigned the *RTP-SRTP* media rule configured above. This endpoint policy group is used for Posh Voice and Session Manager.



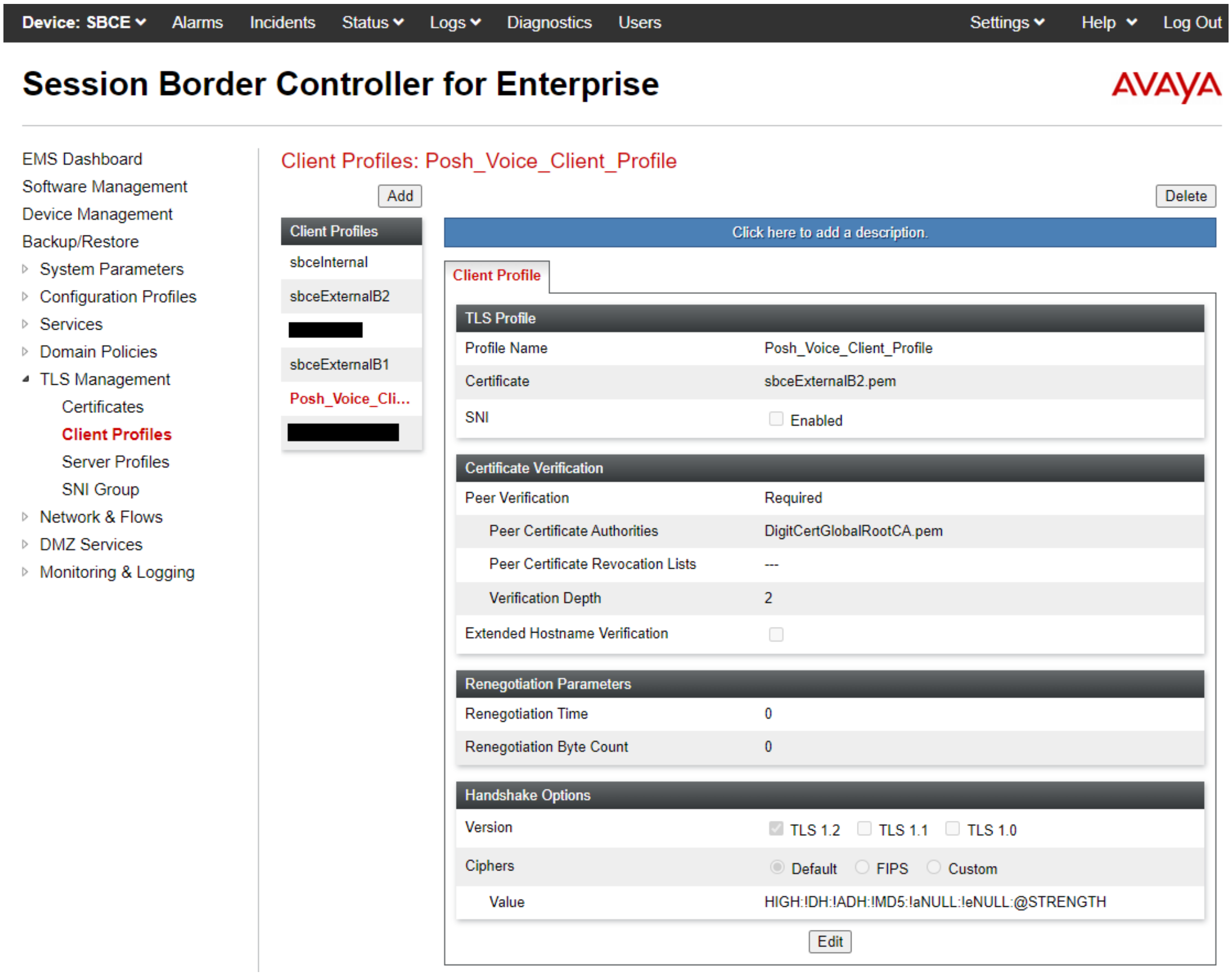
7.8. Administer TLS Management

This section covers installing the Posh Voice SIP service provider certificate, configuring the Posh Voice client profile, and configuring the server profile for the B2 public interface, which connects to the Posh Voice SIP service provider, to set up secure communications using TLS. The TLS configuration for Session Manager is assumed to already be in place and is not shown in these Application Notes.

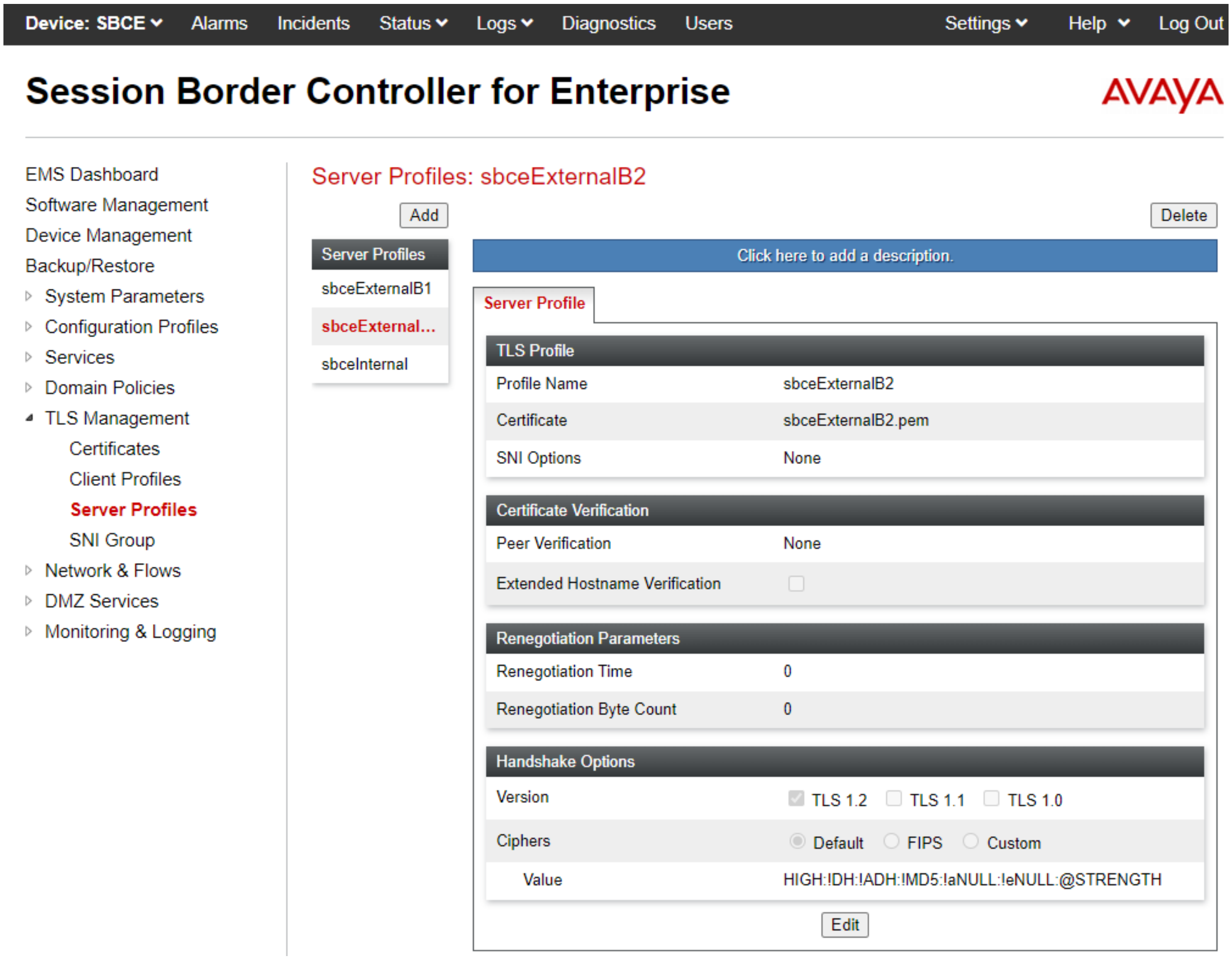
Navigate to **TLS Management → Certificates** and install the Posh Voice SIP service provider certificate. For the compliance test, the certificate was named *DigitCertGlobalRootCA.pem* as shown below.



Next, create a **Client Profile** for Posh Voice as shown below. The **Profile Name** was *Posh_Voice_Client_Profile* and the certificate for B2 public interface was selected. **Peer Verification** was set to *Required* and the *DigitCertGlobalRootCA* certificate was selected for **Peer Certificate Authorities**. The **Verification Depth** was set to *2* and the **Version** was set to *TLS 1.2*. This client profile was assigned to the Posh Voice staging and production SIP servers in **Section 7.3**.



The following server profile is assigned to the B2 public interface covered in **Section 7.10**.



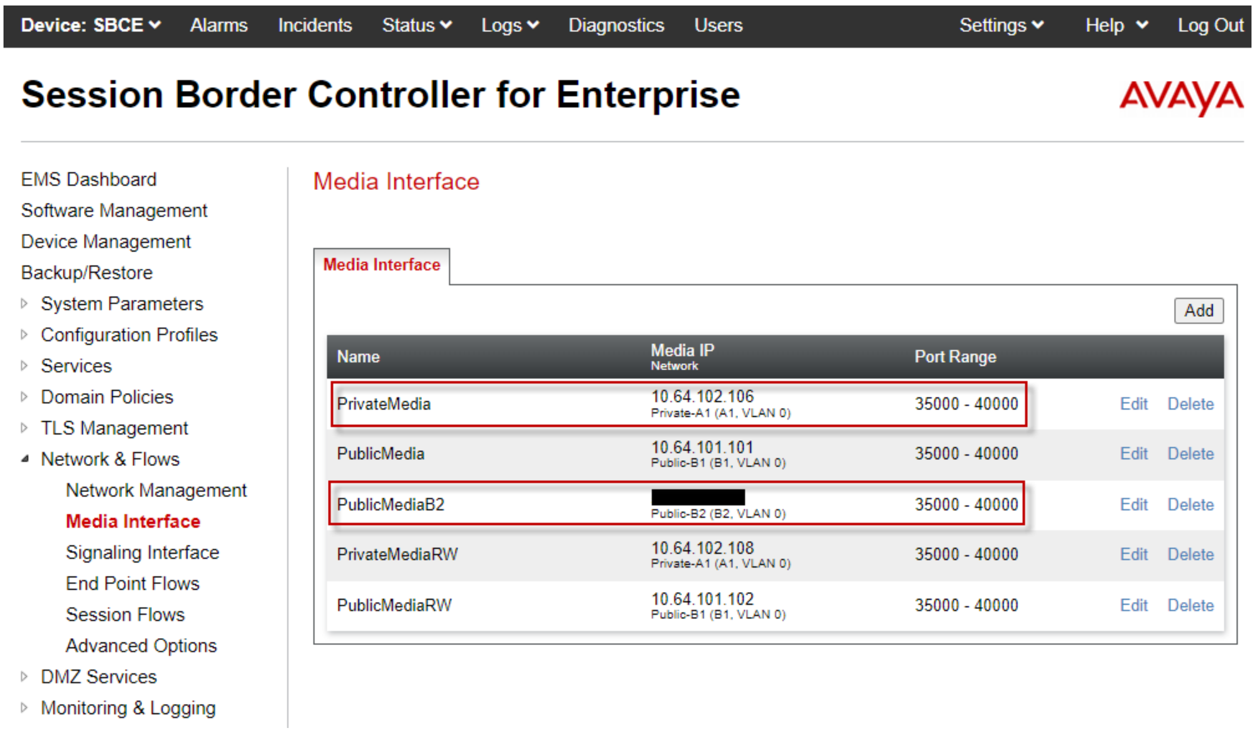
7.9. Administer Media Interfaces

A media interface defines an IP address and port range for transmitting media. Create a media interface for both the internal and external sides of the SBCE. Media Interfaces need to be defined for each SIP server to send and receive media (RTP or SRTP).

Navigate to **Networks & Flows → Media Interface** to define a new Media Interface. During the compliance test, the following interfaces were defined. For security reasons, public IP addresses have been masked. The media interfaces used for this solution are listed below.

- **PrivateMedia:** Interface used by Session Manager to send and receive media.
- **PublicMediaB2:** Interface used by Posh Voice SIP service provider to send and receive media.

Note: A Port Range of 10,000-20,000 may also be used.

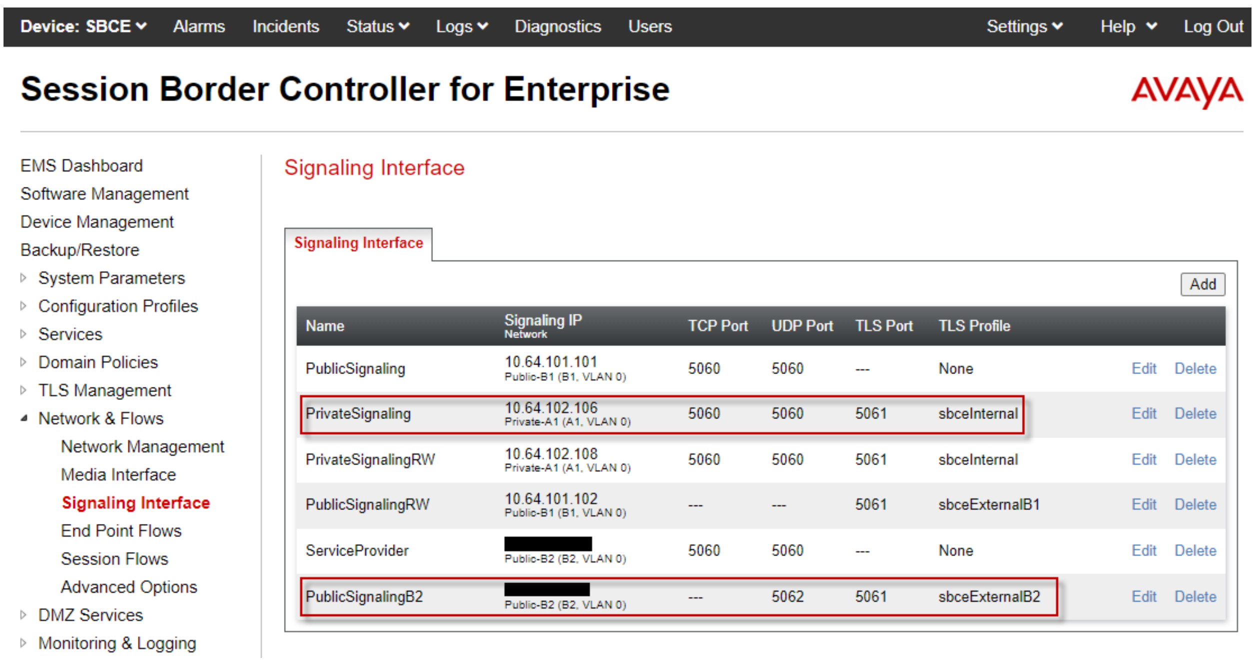


7.10. Administer Signaling Interfaces

A signaling interface defines an IP address, protocols and listen ports that the SBCE can use for signaling. Create a signaling interface for both the internal and external sides of the SBCE. Signaling Interface needs to be defined for each SIP server to send and receive SIP signaling messages.

Navigate to **Networks & Flows → Signaling Interface** to define a new **Signaling Interface**. During the Compliance Testing the following interfaces were defined. For security reasons, public IP addresses have been masked. The signaling interfaces used for this solution are listed below.

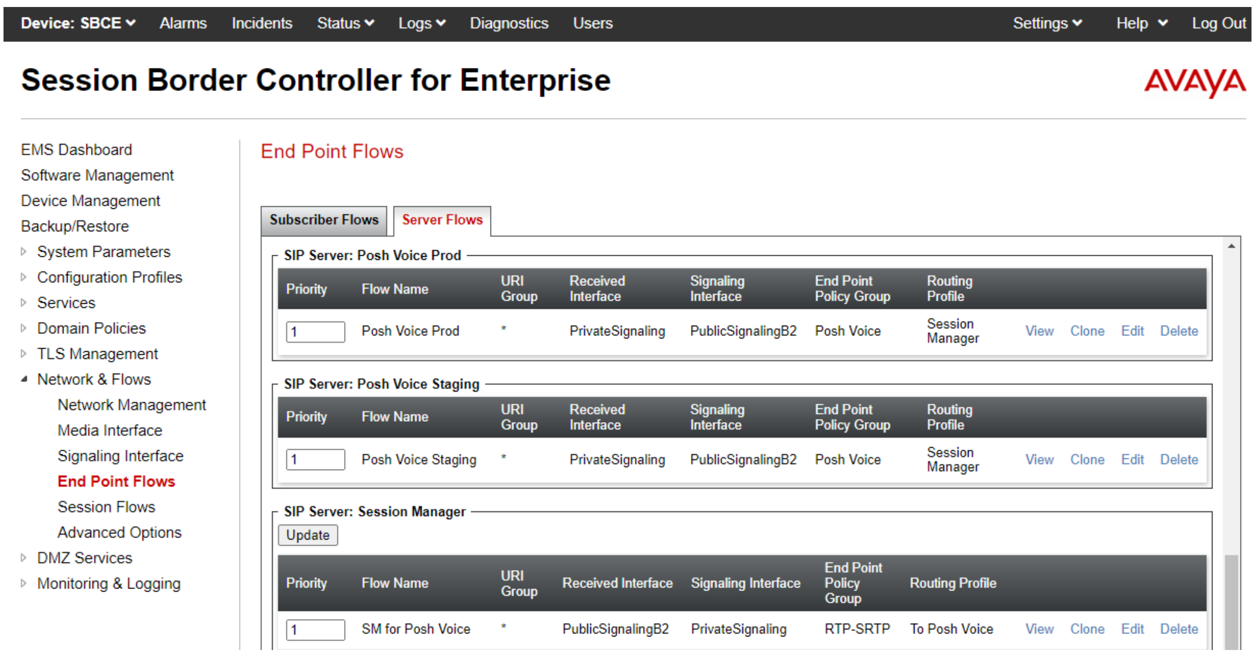
- **PrivateSignaling:** Interface used by Session Manager to send and receive calls.
- **PublicSignalingB2:** Interface used by Posh Voice SIP service provider to send and receive calls.



7.11. Administer End Point Flows

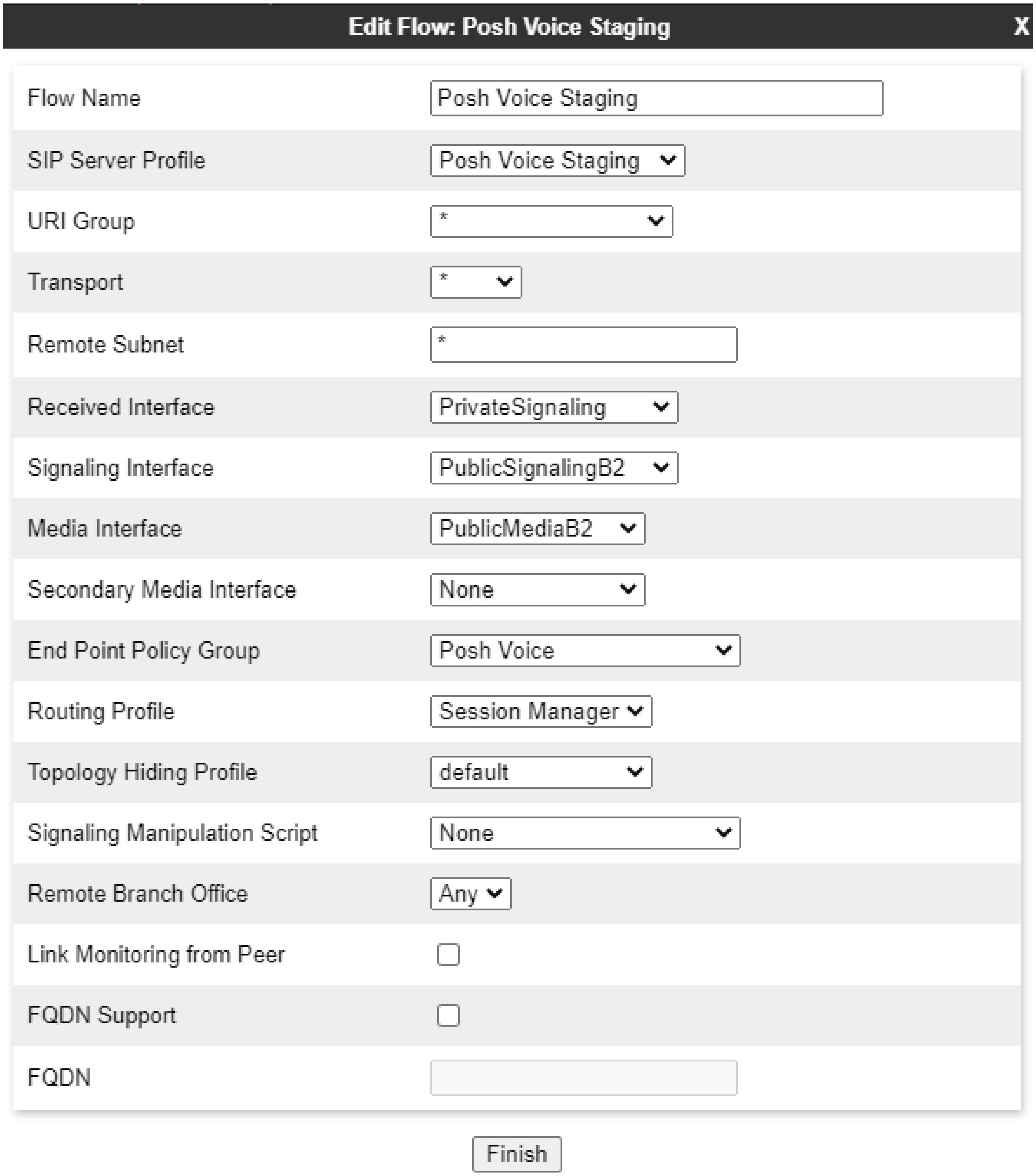
Endpoint flows are used to determine the endpoints (connected servers) involved in a call in order to apply the appropriate policies. When a packet arrives at the SBCE, the content of the packet (IP addresses, URIs, etc.) is used to determine which flow it matches. Once the flow is determined, the flow points to policies and profiles that control processing, privileges, authentication, routing, etc. Once routing is applied and the destination endpoint is determined, the policies for the destination endpoint are applied. Thus, two flows are involved in every call: the source endpoint flow and the destination endpoint flow. In the case of the compliance test, the endpoints are Posh Voice SIP service provider and Session Manager.

Navigate to **Network & Flows → End Point Flows → Server Flows** and select the **Server Flows** tab. The configured **Server Flows** used in the compliance test are shown below. The following subsections will review the settings for each server flow.

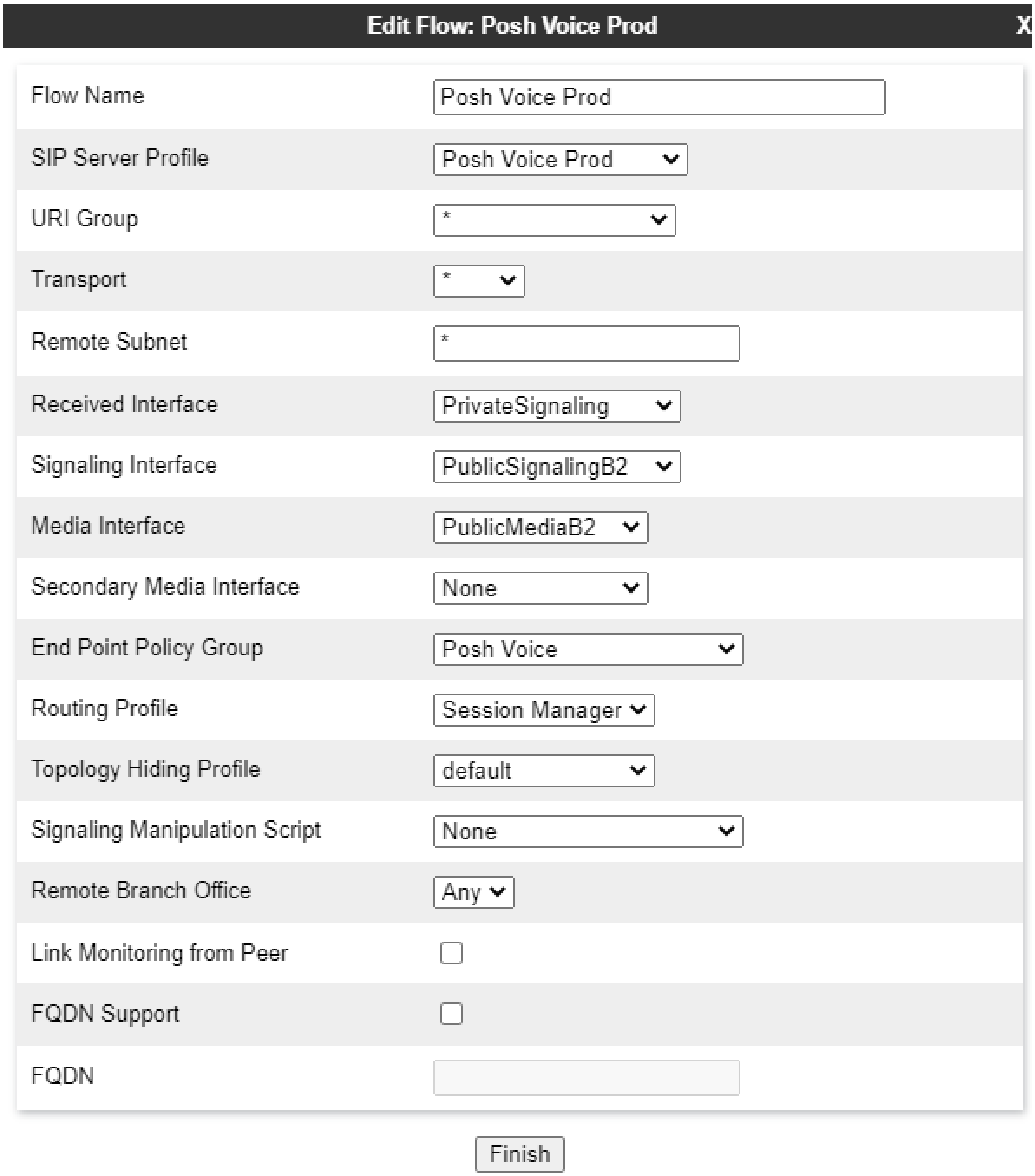


7.11.1. Server Flows for Posh Voice

For the compliance test, two server flows were created for Posh Voice: one for Posh Voice staging and one for Posh Voice production. The following server flow is for Posh Voice staging.

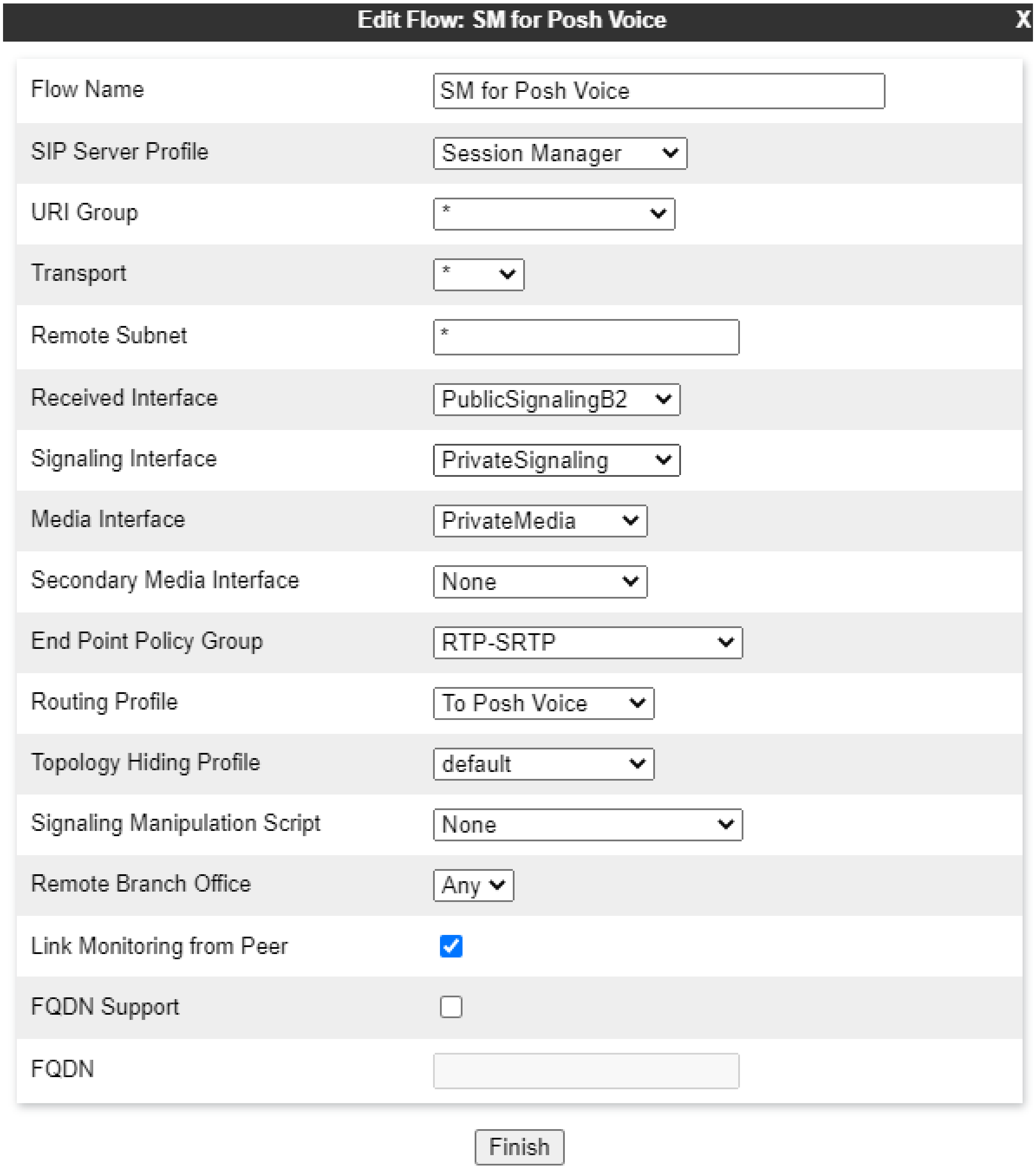


The following server flow is for Posh Voice production.



7.11.2. Server Flow for Session Manager

This section covers the server flow for Session Manager. Note that the Routing Profile, *To Posh Voice*, is used to route calls to Posh Voice staging, Posh Voice production, or the PSTN according to URI groups.



Edit Flow: SM for Posh Voice

Flow Name	SM for Posh Voice
SIP Server Profile	Session Manager
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	PublicSignalingB2
Signaling Interface	PrivateSignaling
Media Interface	PrivateMedia
Secondary Media Interface	None
End Point Policy Group	RTP-SRTP
Routing Profile	To Posh Voice
Topology Hiding Profile	default
Signaling Manipulation Script	None
Remote Branch Office	Any
Link Monitoring from Peer	☑
FQDN Support	☐
FQDN	

Finish

8. Configure Posh Voice

The configuration of Posh Voice is performed by Posh technical personnel. For provisioning, Posh will require the following information :

- SBCE public IP address.
- Agent queues (e.g., skill group or VDN extension) used by Posh Voice to transfer calls to contact center.

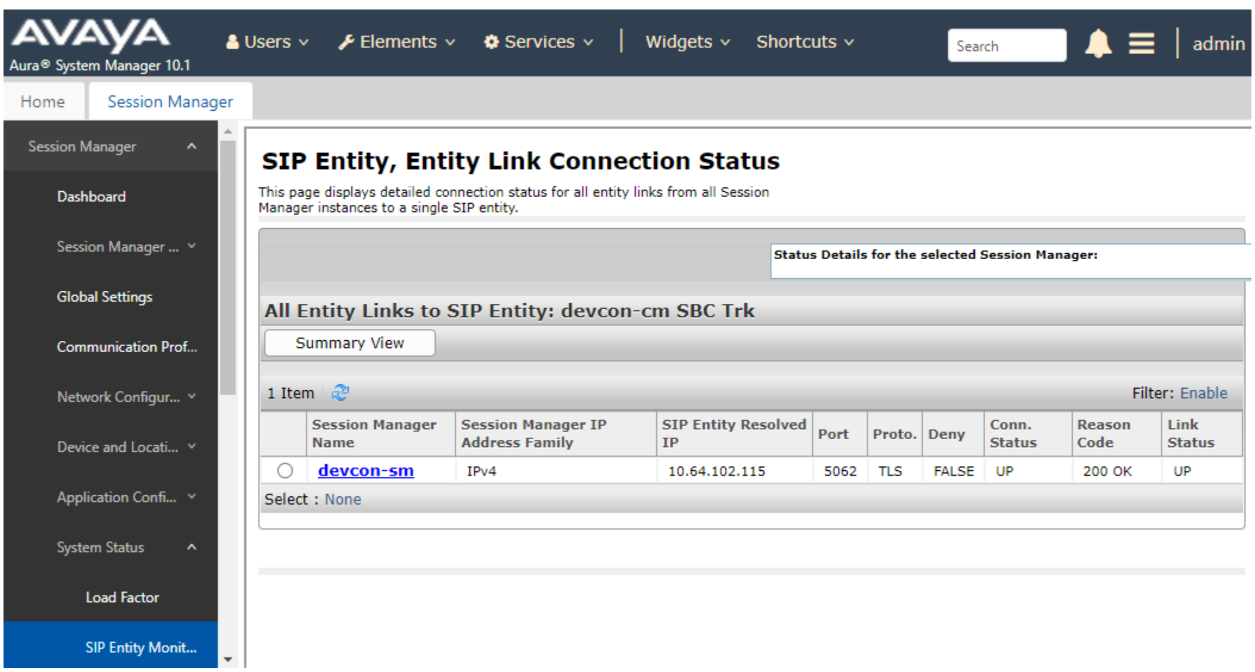
9. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Communication Manager, Session Manager, SBCE, and Posh Voice.

1. From the System Manager home page (not shown), select **Elements → Session Manager** from the top menu to display the **Session Manager Dashboard** screen (not shown).

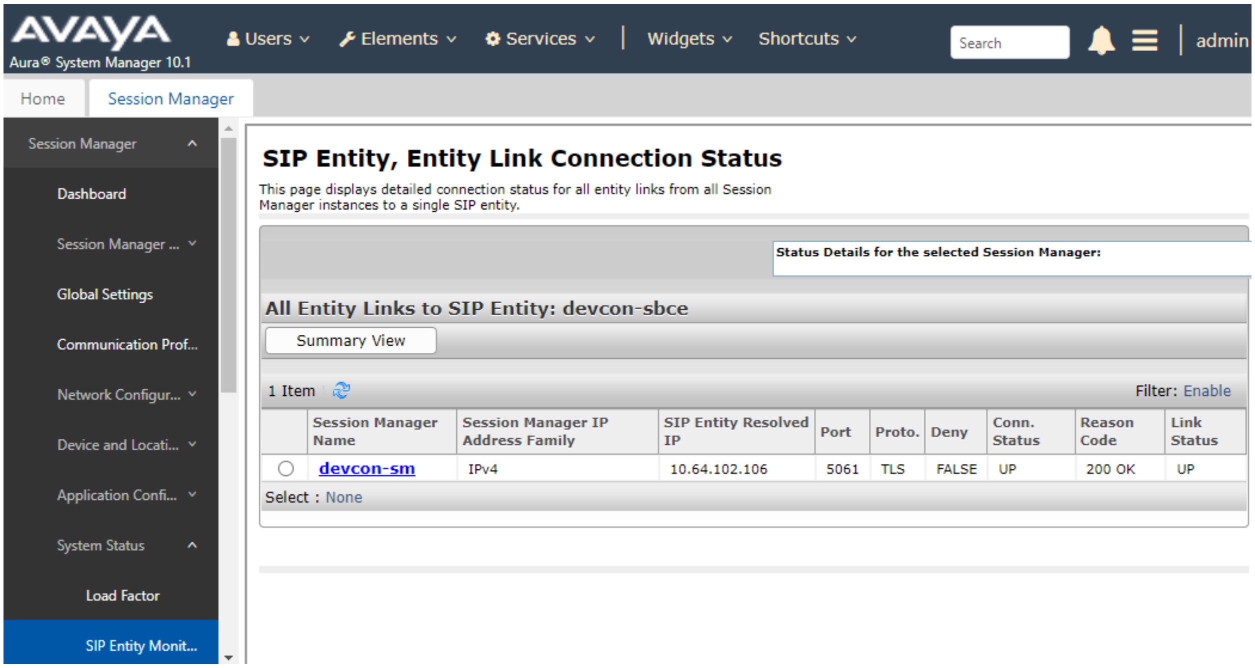
   Select **Session Manager → System Status → SIP Entity Monitoring** from the left pane to display the **SIP Entity Link Monitoring Status Summary** screen. Click on the Communication Manager entity name from **Section 6.2.1**.

   The **SIP Entity, Entity Link Connection Status** screen is displayed. Verify that the **Conn. Status** and **Link Status** are "UP", as shown below.

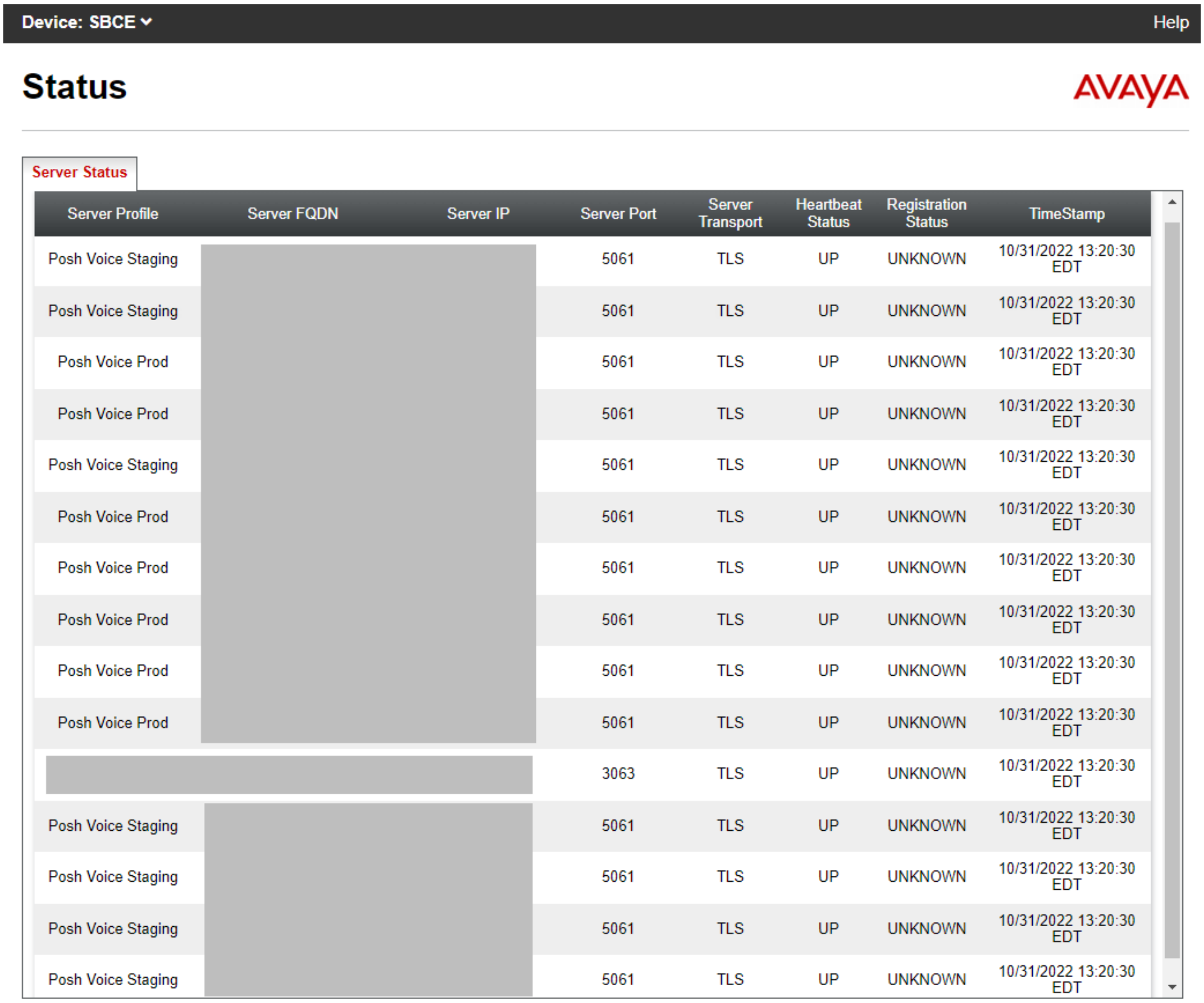


2. Select **Session Manager → System Status → SIP Entity Monitoring** from the left pane to display the **SIP Entity Link Monitoring Status Summary** screen. Click on the SBCE entity name from **Section 6.2.2**.

   The **SIP Entity, Entity Link Connection Status** screen is displayed. Verify that the **Conn. Status** and **Link Status** are "UP", as shown below.



3. To verify the SIP trunks between SBCE and the Posh Voice SIP service provider are in service, navigate to **Status → Server Status** in the SBCE web interface. The **Hearbeat Status** for each of the SIP trunks to the Posh Voice staging and production environments should be *UP* as shown below.



Server Profile	Server FQDN	Server IP	Server Port	Server Transport	Heartbeat Status	Registration Status	TimeStamp
Posh Voice Staging	[redacted]		5061	TLS	UP	UNKNOWN	10/31/2022 13:20:30 EDT
Posh Voice Staging			5061	TLS	UP	UNKNOWN	10/31/2022 13:20:30 EDT
Posh Voice Prod			5061	TLS	UP	UNKNOWN	10/31/2022 13:20:30 EDT
Posh Voice Prod			5061	TLS	UP	UNKNOWN	10/31/2022 13:20:30 EDT
Posh Voice Staging			5061	TLS	UP	UNKNOWN	10/31/2022 13:20:30 EDT
Posh Voice Prod			5061	TLS	UP	UNKNOWN	10/31/2022 13:20:30 EDT
Posh Voice Prod			5061	TLS	UP	UNKNOWN	10/31/2022 13:20:30 EDT
Posh Voice Prod			5061	TLS	UP	UNKNOWN	10/31/2022 13:20:30 EDT
Posh Voice Prod			5061	TLS	UP	UNKNOWN	10/31/2022 13:20:30 EDT
Posh Voice Prod			5061	TLS	UP	UNKNOWN	10/31/2022 13:20:30 EDT
[redacted]			3063	TLS	UP	UNKNOWN	10/31/2022 13:20:30 EDT
Posh Voice Staging	[redacted]		5061	TLS	UP	UNKNOWN	10/31/2022 13:20:30 EDT
Posh Voice Staging			5061	TLS	UP	UNKNOWN	10/31/2022 13:20:30 EDT
Posh Voice Staging			5061	TLS	UP	UNKNOWN	10/31/2022 13:20:30 EDT
Posh Voice Staging			5061	TLS	UP	UNKNOWN	10/31/2022 13:20:30 EDT

4. Place a call to Posh Voice and verify the application answers and the appropriate greeting is heard.

5. Caller navigates through the application using speech and DTMF. Verify Posh Voice provides the requested information.

6. Posh Voice transfers call to an agent or PSTN. Verify the transferred call is established with two-way audio.

7. Caller terminates the call successfully.

10. Conclusion

These Application Notes have described the configuration steps required to integrate Posh Voice with Avaya Session Border Controller for Enterprise. Posh Voice connected to an Avaya contact center via a SIP service provider. Callers were able to interact with Posh Voice using speech and DTMF to retrieve and provide information. In addition, Posh Voice was able to transfer the call to an agent and send caller information in UUI. All test cases passed.

11. Additional References

This section references the product documentation relevant to these Application Notes.

[1] *Administering Avaya Aura® Communication Manager*, Release 10.1.x, Issue 1, December 2021, available at http://support.avaya.com.

[2] *Administering Avaya Aura® System Manager*, Release 10.1.x, Issue 6, June 2022, available at http://support.avaya.com.

[3] *Administering Avaya Aura® Session Manager*, Release 10.1.x, Issue 3, April 2022, available at http://support.avaya.com.

[4] *Administering Avaya Session Border Controller for Enterprise*, Release 10.1.x, Issue 1, December 2021, available at http://support.avaya.com.

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.