DevConnect Program

Application Notes for Configuring Avaya Aura® Communication Manager R10.1, Avaya Aura® Session Manager R10.1, Avaya Experience Portal R8.1 and Avaya Session Border Controller R10.1 to support VTX Connect PBX-IP SIP Trunk Service - Issue 1.0

Abstract

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between the VTX Connect PBX-IP SIP Service and an Avaya SIP enabled Enterprise Solution. The Avaya solution consists of Avaya Aura® Communication Manager R10.1, Avaya Aura® Session Manager R10.1, Avaya Experience Portal R8.1 and Avaya Session Border Controller R10.1.

The VTX Connect PBX-IP SIP Platform provides PSTN access via a SIP trunk connected to the VTX Voice over Internet Protocol (VoIP) network as an alternative to legacy analogue or digital trunks. VTX is a member of the DevConnect Service Provider program.

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.

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1. Introduction

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between the VTX Connect PBX-IP SIP Service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of the following: Avaya Aura ® Communication Manager R10.1 (Communication Manager), Avaya Aura ® Session Manager R10.1 (Session Manager), Avaya Experience Portal R8.1 (Experience Portal) and Avaya Session Border Controller R10.1 (Avaya SBC).

Customers using this Avaya SIP-enabled enterprise solution with the VTX Connect PBX-IP SIP Service are able to place and receive PSTN calls via a dedicated Internet connection and the SIP protocol. This approach generally results in lower cost for the enterprise customer.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using an Avaya SIP telephony solution consisting of Communication Manager, Session Manager, Experience Portal and Avaya SBC. The enterprise site was configured to connect to the VTX SIP platform.

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.

2.1. Interoperability Compliance Testing

The interoperability test included the following:

- Incoming calls to the enterprise site from PSTN phones using the VTX SIP platform, calls made to SIP, H.323, Digital and Analogue telephones at the enterprise.
- Outgoing calls from the enterprise site completed via the VTX SIP platform to PSTN destinations, calls made from SIP, H.323, Digital and Analogue telephones.
- Incoming and Outgoing PSTN calls to/from Avaya one-X® Communicator and Avaya Workplace Client for Windows soft phones.
- Calls using G.711A and G.729 codecs.
- Fax calls to/from Client a group 3 fax machine to a PSTN-connected fax machine using T.38 transmission.
- DTMF transmission using RFC 2833 with successful Voice Mail/Vector navigation for inbound and outbound calls.
- User features such as hold and resume, transfer, conference, call forwarding, etc.
- Caller ID Presentation and Caller ID Restriction.
- Call coverage and call forwarding for endpoints at the enterprise site.
- 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 SBC, 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.
- Transmission and response of SIP OPTIONS messages sent by VTX requiring Avaya response and sent by Avaya requiring VTX response.

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for the VTX Connect PBX-IP SIP Trunking Service with the following observations:

- It was observed during testing that certain Call Forwarding All Calls and Call Forwarding No Answer scenarios from Communication Manager to a number of IP-PBX services hosted on the VTX SIP platform were failing. As a workaround, direct media setting "Initial IP-IP Direct Media" was set to "**n**" in the Communication Manager SIP Signalling Group as per **Section 5.5**. Once Direct Media was disabled on Communication Manager, all Call Forwarding All Calls and Call Forwarding No Answer calls terminated successfully on the multiple IP-PBX services hosted within the VTX SIP platform.
- It was observed during testing that Experience Portal uses REFER to complete Blind and Consultative transfers to internal Contact Center/ACD applications, such as agent routing, which led to signalling issues and transfer failures between Avaya and the VTX SIP trunk. In order to complete Blind and Consultative transfers successfully within Experience Portal, REFER Handling needs to be enabled on the VTX Server Interworking profile (**Section 8.5.2**) on the Avaya SBC. When the REFER message comes from an Avaya enterprise element such as Experience Portal, the Avaya SBC translates that REFER into a reINVITE which will then be routed towards the trunk server (i.e., VTX) based on the trunk server interworking profile configuration.
- No Inbound Toll-Free access available for test.
- No Emergency Services test call booked with Operator.

2.3. Support

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

For technical support on VTX Services products please contact the VTX Services support team:
https://www.vtx.ch/fr/support/contact
https://www.vtx.ch/de/support/kontakt

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an Enterprise site connected to the VTX SIP platform. Located at the Enterprise site is an Avaya SBC, Session Manager and Communication Manager. Endpoints are Avaya 96x1 series IP telephones (with SIP and H.323 firmware), Avaya J179 series SIP telephones, Avaya digital telephone, Avaya analogue telephones and an analogue fax machine. Also included in the test configuration was an Avaya one-X® Communicator soft phone and Avaya Workplace Client for Windows running on laptop PCs.

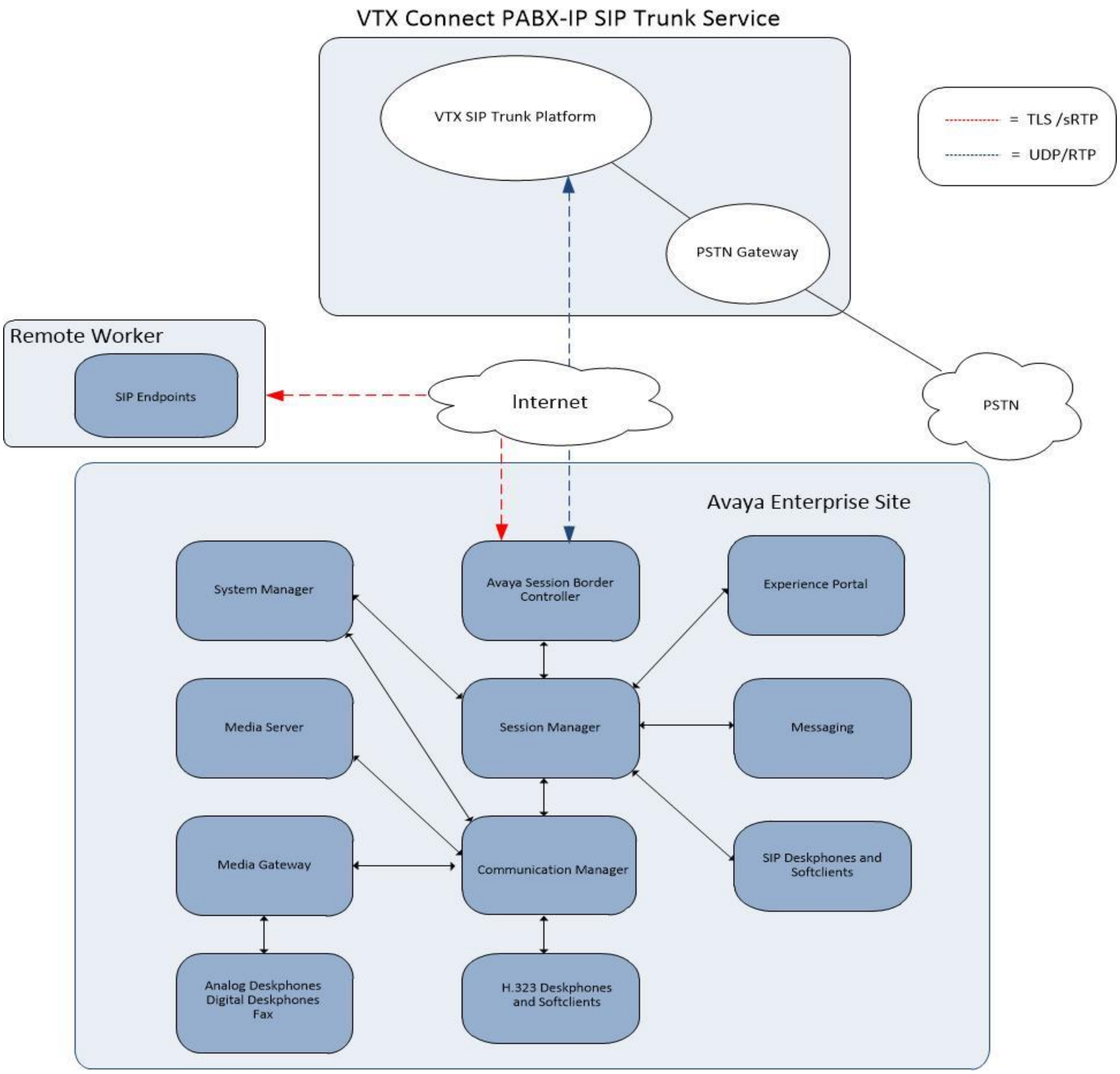


Figure 1: VTX Connect PBX-IP SIP Service to Avaya Enterprise

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya	
Avaya Aura® System Manager	10.1.3.1 Build No. – 10.1.0.0.537353 Software Update Revision No: 10.1.3.1.0716149 – Service Pack 1
Avaya Aura® Session Manager	10.1.3.1.1013103
Avaya Aura® Communication Manager	10.1.3.1 Service Pack 1 - 27937
Avaya Session Border Controller	10.1.2.0-64-23285
Avaya Experience Portal	8.1.1
Avaya G430 Media Gateway	42.24.0
Avaya Aura® Media Server	v.10.1.0.125
Avaya Aura® Messaging	7.2 SP3
Avaya 1600 IP Deskphone (H.323)	1.3.12
Avaya 96x1 IP DeskPhone (H.323)	6.8.5.
Avaya 9611 IP DeskPhone (SIP)	7.1.15.
Avaya 9608 IP DeskPhone (SIP)	7.1.15.
Avaya J179 IP Deskphone (SIP)	4.1.1.0
Avaya one–X® Communicator (H.323 & SIP)	6.2.14.18 -SP14-Patch 9
Avaya Workplace for Windows (SIP)	3.34.1.
Avaya 1408 Digital Telephone	R48
Analogue Handset	N/A.
VTX	
VTX CONNECT PBX IP	VTX CONNECT PBX-IP <X> (<X>: depending on number of channels: 4, 8, 15, 30, 60, etc.)
Platform: Communi5, VAS C5	7.4 udp7
SBC: Audiocodes	Mediant VE SBC, Version 7.40A.250.541

5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring Communication Manager for SIP Trunking. SIP trunks are established between Communication Manager and Session Manager. These SIP trunks will carry SIP signalling associated with the VTX Connect PBX-IP SIP Trunking Service. For incoming calls, Session Manager receives SIP messages from the Avaya SBC and directs the incoming SIP messages to Communication Manager. Once the message arrives at Communication Manager further incoming call treatment, such as incoming digit translations and class of service restrictions may be performed. All outgoing calls to the PSTN are processed within Communication Manager and may be first subject to outbound features such as automatic route selection, digit manipulation and class of service restrictions. Once Communication Manager selects a SIP trunk, the SIP signalling is routed to Session Manager. The Session Manager directs the outbound SIP messages to the Avaya SBC at the enterprise site that then sends the SIP messages to the VTX network. 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. The general installation of the Servers and Avaya G430 Media Gateway is presumed to have been previously completed and is not discussed here.

5.1. Confirm System Features

The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity. Use the **display system-parameters customer-options** command and on **Page 2**, verify that the **Maximum Administered SIP Trunks** supported by the system is sufficient for the combination of trunks to the VTX SIP Trunk network, and any other SIP trunks used.

```
display system-parameters customer-options                      Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 12000 0
           Maximum Concurrently Registered IP Stations: 18000 3
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 41000 0
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 24000 10
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   0
                     Maximum Media Gateway VAL Sources: 250   1
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0
```

On **Page 5**, verify that **IP Trunks** field is set to **y.**

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

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

           IP Attendant Consoles? y
```

5.2. Administer IP Node Names

The node names defined here will be used in other configuration screens to define a SIP signalling group between Communication Manager and Session Manager. In the **IP Node Names** form, assign the node **Name** and **IP Address** for Session Manager. In this case, **Session Manager** and **10.10.3.42** are the **Name** and **IP Address** for the Session Manager SIP interface. Also note the **procr** IP address as this is the processor interface that Communication Manager will use as the SIP signalling interface to Session Manager.

```
display node-names ip
                                  IP NODE NAMES
    Name              IP Address
AMS                 10.10.3.45
Session_Manager     10.10.3.42
default             0.0.0.0
procr               10.10.3.44
procr6              ::
```

5.3. Administer IP Network Region

Use the **change ip-network-region n** command where **n** is the chosen value of the configuration for the SIP Trunk. Set the following values:

- The **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is **avaya.com**.
- By default, **IP-IP Direct Audio** (both **Intra**- and **Inter-Region**) is enabled (**yes**) to allow audio traffic to be sent directly between endpoints without using gateway VoIP resources. When a PSTN call is shuffled or the call is set up with initial IP-IP direct media, the media stream is established directly between the enterprise end-point and the internal media interface of the Avaya SBC.
- The **Codec Set** is set to the number of the IP codec set to be used for calls within the IP network region. In this case, codec set **1** is used.
- The rest of the fields can be left at default values.

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

5.4. Administer IP Codec Set

Open the IP Codec Set form for the codec set specified in the IP Network Region form in **Section 5.3** by typing **change ip-codec set n w**here **n** is the chosen value of the configuration for the SIP Trunk. Enter the list of audio codec's eligible to be used in order of preference. For the interoperability test the codec supported by VTX were configured, namely **G.711A** and **G.729**.

In addition to the codecs, the **Media Encryption** is defined here. For the compliance test, a value of **srtp-aescm128-hmac80** was used.

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

                          IP MEDIA PARAMETERS
    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
1: G.711A         n         2        20
2: G.729          n         2        20




     Media Encryption                  Encrypted SRTCP: enforce-unenc-srtcp
 1: srtp-aescm128-hmac80
 2: none
```

VTX SIP Trunk supports T.38 for transmission of fax. Navigate to **Page 2** and define fax properties as follows:

- Set the **FAX - Mode** to **t.38-standard**.
- Leave **ECM** at default value of **y**.

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

                          IP MEDIA PARAMETERS

                              Allow Direct-IP Multimedia? n



                                             Redun-                     Packet
                          Mode               dancy                      Size(ms)
    FAX                   t.38-standard      0   ECM: y
    Modem                 off                0
    TDD/TTY               US                 3
    H.323 Clear-channel   n                  0
    SIP 64K Data          n                  0                          20

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

5.5. Administer SIP Signaling Groups

This signalling group (and trunk group) will be used for inbound and outbound PSTN calls to the VTX SIP platform. Configure the **Signaling Group** using the **add signaling-group n** command as follows:

- Set **Group Type** to **sip**.
- Set **Transport Method** to **tls**.
- Set **Peer Detection Enabled** to **y** allowing Communication Manager to automatically detect if the peer server is a Session Manager.
- Set **Near-end Node Name** to the processor interface (node name **procr** as defined in the **IP Node Names** form shown in **Section 5.2**).
- Set **Far-end Node Name** to Session Manager interface (node name **Session_Manager** as defined in the **IP Node Names** form shown in **Section 5.2**).
- Set **Near-end Listen Port** and **Far-end Listen Port** as required. The standard value for TLS is **5061**.
- Set **Far-end Network Region** to the IP Network Region configured in **Section 5.3** (logically establishes the far-end for calls using this signalling group as region **1**).
- Leave **Far-end Domain** blank to allow Communication Manager to accept calls from any SIP domain on the associated trunk.
- Leave **DTMF over IP** at default value of **rtp-payload** (Enables **RFC2833** for DTMF transmission from Communication Manager).
- Set **Direct IP-IP Audio Connections** to **y**.
- Set **Initial IP-IP Direct Media** to **n** as per **Section 2.2**.
- Set **H.323 Station Outgoing Direct Media** to **y**.

The default values for the other fields may be used.

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

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                  Enforce SIPS URI for SRTP? n
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                Far-end Node Name: Session_Manager
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                       Far-end Network Region: 1
Far-end Domain:
                                            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? y           Alternate Route Timer(sec): 6
```

5.6. Administer SIP Trunk Groups

A trunk group is associated with the signalling group described in **Section 5.5**. Configure the trunk group using the **add trunk-group n** command, where **n** is an available trunk group for the SIP Trunk. On **Page 1** of this form:

- Set the **Group Type** field to **sip**.
- Choose a descriptive **Group Name**.
- Specify a trunk access code (**TAC**) consistent with the dial plan.
- The **Direction** is set to **two-way** to allow incoming and outgoing calls.
- Set the **Service Type** field to **public-netwrk**.
- Specify the signalling group associated with this trunk group in the **Signaling Group** field as previously configured in **Section 5.5**.
- Specify the **Number of Members** administered for this SIP trunk group.

```
add trunk-group 1                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 1                   Group Type: sip           CDR Reports: y
  Group Name: OUTSIDE CALL               COR: 1        TN: 1        TAC: 101
   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: 1
                                                     Number of Members: 10
```

On **Page 2** of the trunk-group form, the Preferred **Minimum Session Refresh Interval (sec)** field should be set to a value mutually agreed with VTX to prevent unnecessary SIP messages during call setup.

```
add trunk-group 1                                                 Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 5000

            SCCAN? n                               Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 900

 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 allows delivery of CLI in format of E.164 with leading "+". Also, set the **Hold/Unhold Notifications** to **n**.

```
add trunk-group 1                                               Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y



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

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n

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

 Show ANSWERED BY on Display? y
```

On **Page 4** of this form:

- Set **Mark Users as Phone** to **y**.
- Set **Send Transferring Party Information** to **n**.
- Set **Network Call Redirection** to **n**.
- Set **Send Diversion Header** to **y**.
- Set **Support Request History** to **n**.
- Set the **Telephone Event Payload Type** to **101** as requested by VTX.
- Set **Always Use re-INVITE for Display Updates** to **y**.
- Set the **Identity for Calling Party Display** to **P-Asserted-Identity**.

```
add trunk-group 1                                                Page   4 of  21
                              PROTOCOL VARIATIONS

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

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


                        Convert 180 to 183 for Early Media? n
                  Always Use re-INVITE for Display Updates? y
                        Identity for Calling Party Display: P-Asserted-Identity
            Block Sending Calling Party Location in INVITE? n
                 Accept Redirect to Blank User Destination? n
                                              Enable Q-SIP? N
          Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
                                Request URI Contents: may-have-extra-digits
```

5.7. Administer Calling Party Number Information

Use the **change public-unknown-numbering** command to configure Communication Manager to send the calling party number in the format required. These calling party numbers are sent in the SIP From, Contact and PAI headers as well as the Diversion header for forwarded calls. The numbers are displayed on display-equipped PSTN telephones with any reformatting performed in the network. The public numbering table is used for numbers in E.164 format.

```
change public-unknown-numbering 0                                   Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                         Total
Ext           Trk       CPN             CPN
Len Code          Grp(s)    Prefix          Len
                                                  Total Administered: 4
 4  6100          1         4121xxxxx06      11       Maximum Entries: 240
 4  6102          1         4121xxxxx07      11
 4  6010          1         4121xxxxx08      11
 4  6030          1         4121xxxxx09      11     Note: If an entry applies to
 4  6104          1         4121xxxxx10      11     a SIP connection to Avaya
                                                  Aura(R) Session Manager,
                                                  the resulting number must
                                                  be a complete E.164 number.

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

5.8. Administer Route Selection for Outbound Calls

In the test environment, the Automatic Route Selection (ARS) feature was used to route outbound calls via the SIP trunk to the VTX SIP Trunking Service. The single digit **9** was used as the ARS access code providing a facility for telephone users to dial 9 to invoke ARS directly. Use the **change feature-access-codes** command to configure a digit 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: *69
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 7
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
```

Use the **change ars analysis** command to configure the routing of dialled digits following the first digit 9. A small sample of dial patterns are shown here as an example. Further administration of ARS is beyond the scope of this document. The example entries shown will match outgoing calls to numbers beginning **0**. Note that exact maximum number lengths should be used where possible to reduce post-dial delay. Calls are sent to **Route Pattern 1**.

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

          Dialed           Total      Route     Call   Node  ANI
          String          Min  Max  Pattern    Type   Num   Reqd
    0                     11   14     1        pubu         n
    00                    13   15     1        pubu         n
    0035391               13   13     1        pubu         n
    030                   10   10     1        pubu         n
    0800                  8    10     1        pubu         n
    0900                  8    8      1        pubu         n
                                                            n
```

Use the **change route-pattern x** command, where **x** is an available route pattern, to add the SIP trunk group to the route pattern that ARS selects. In this configuration, route pattern **1** is used to route calls to trunk group **1**. **Numbering Format** is applied to CLI and is used to set TDM signalling parameters such as type of number and numbering plan indicator. This does not have the same significance in SIP calls and during testing it was set to **pub-unk**.

```
change route-pattern 1                                          Page   1 of   3
                    Pattern Number: 1      Pattern Name:
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 1    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  No. Numbering LAR
    0 1 2 M 4 W     Request                                Dgts Format
                                                        Subaddress
 1: y y y y y n  n            rest                              pub-unk   none
 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
```

5.9. Administer Incoming Digit Translation

This step configures the settings necessary to map incoming DID calls to the proper Communication Manager extension(s). The incoming digits sent in the INVITE message from VTX can be manipulated as necessary to route calls to the desired extension. In the examples used in the compliance testing, the incoming DID numbers provided by VTX correlate to the internal extensions assigned within Communication Manager. The entries displayed below translates incoming DID numbers **+4121xxxxxxx** to a 4 digit extension by deleting all of the incoming digits and inserting an extension. Public DID numbers have been masked for security purposes.

```
change inc-call-handling-trmt trunk-group 1                     Page   1 of   3
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature         Len      Digits
 public-ntwrk    12 +4121xxxxx06       all 6010
 public-ntwrk    12 +4121xxxxx07       all 6102
 public-ntwrk    12 +4121xxxxx08       all 6020
 public-ntwrk    12 +4121xxxxx09       all 6030
 public-ntwrk    12 +4121xxxxx10       all 6104
```

5.10. EC500 Configuration

When EC500 is enabled on a Communication Manager station, a call to that station will generate a new outbound call from Communication Manager to the configured EC500 destination, typically a mobile phone.

The following screen shows an example EC500 configuration for the user with station extension 6102. Use the command **change off-pbx-telephone station-mapping x** where **x** is Communication Manager station.

- The **Station Extension** field will automatically populate with station extension.
- For **Application** enter **EC500**.
- Enter a **Dial Prefix** (e.g., 9) if required by the routing configuration.
- For the **Phone Number** enter the phone that will also be called (e.g. **0035389434xxxx**).
- Set the **Trunk Selection** to **ARS**.
- Set the **Config Set** to **1**.

```
change off-pbx-telephone station-mapping 6102                  Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 6102             EC500        -      0035389434xxxx      ARS          1
```

Note: The phone number shown is for a mobile phone in the Avaya Lab. To use facilities for calls coming in from EC500 mobile phones, the calling party number received in Communication Manager must exactly match the number specified in the above table.

Save Communication Manager configuration by entering **save translation**.

6. Configuring Avaya Aura® Session Manager

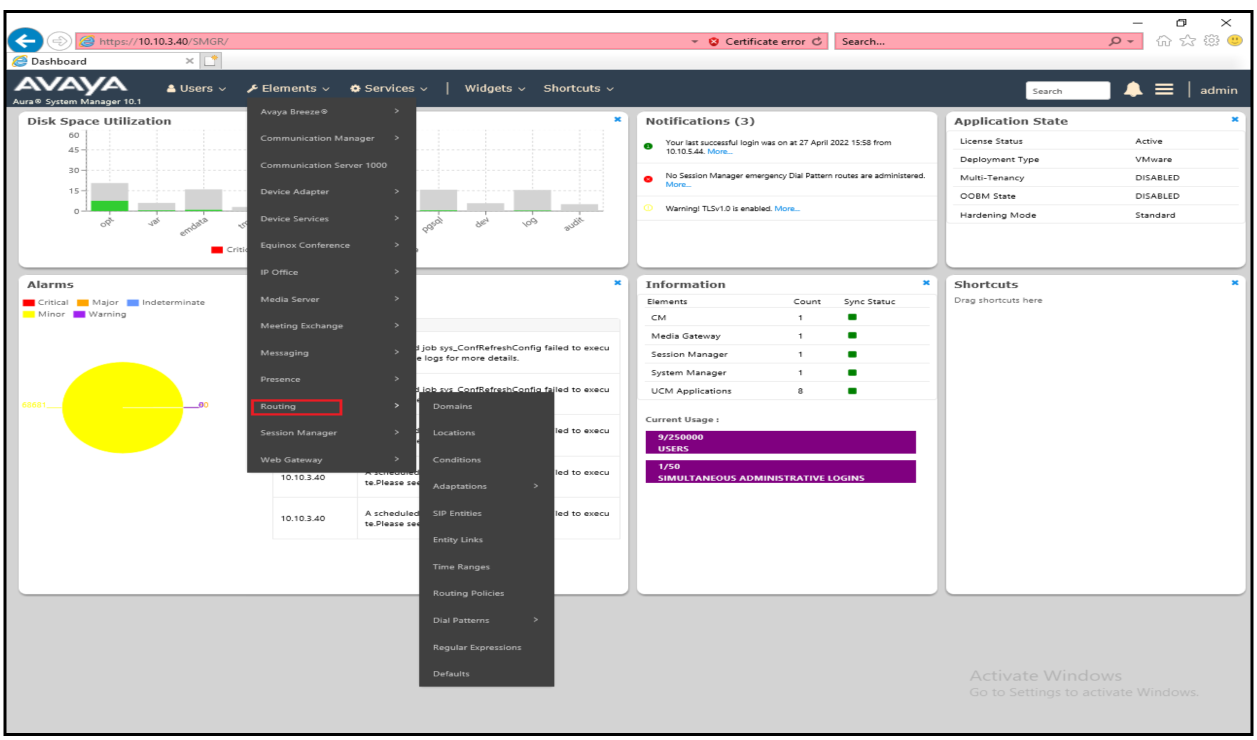
This section provides the procedures for configuring Session Manager. Session Manager is configured via System Manager. The procedures include the following areas:

- Log in to Avaya Aura® System Manager.
- Administer SIP Domain.
- Administer SIP Location.
- Administer Conditions.
- Administer Adaptations.
- Administer SIP Entities.
- Administer Entity Links.
- Administer Routing Policies.
- Administer Dial Patterns.

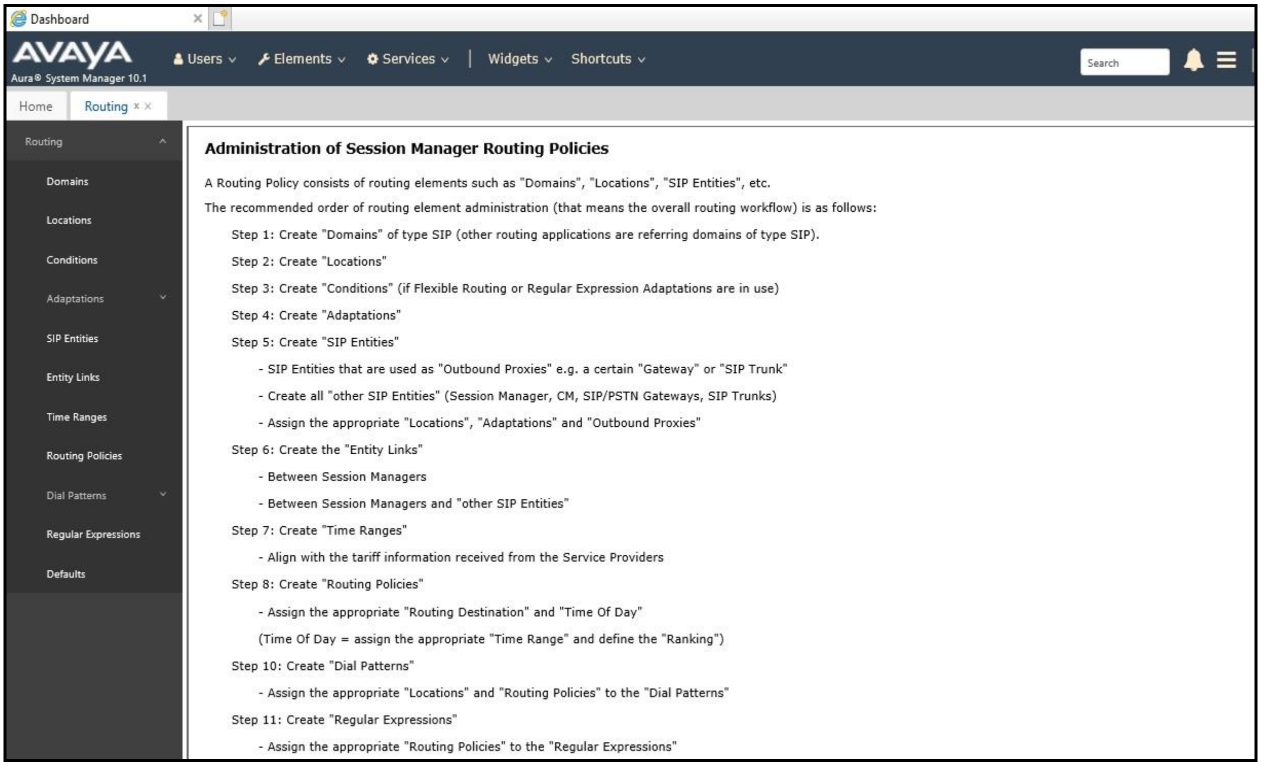
It may not be necessary to create all the items above when creating a connection to the service provider since some of these items would have already been defined as part of the initial Session Manager installation. This includes items such as certain SIP domains, locations, SIP entities, and Session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

6.1. Log in to Avaya Aura® System Manager

Access the System Manager using a web browser and entering **http://<FQDN >/SMGR**, where **<FQDN>** is the fully qualified domain name of System Manager. Log in using appropriate credentials (not shown) and the Dashboard tab will be presented with menu options shown below.



Most of the configuration items are performed in the Routing Element. Click on **Routing** in the Elements column shown above to bring up the **Administration of Session Manager Routing Policies** screen.

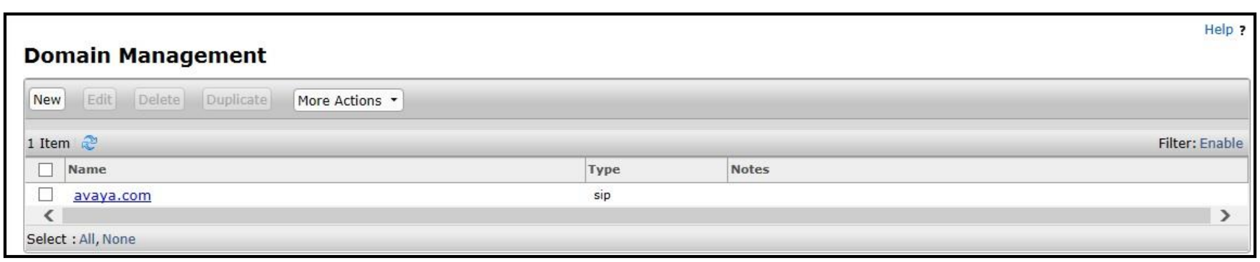


6.2. Administer SIP Domain

Create a SIP domain for each domain for which Session Manager will need to be aware in order to route calls. Expand **Elements → Routing** and select **Domains** from the left navigation menu, click **New** (not shown)**.** Enter the following values and use default values for remaining fields.

- **Name** Enter a Domain Name**.** In the sample configuration, **avaya.com** was used.
- **Type** Verify **SIP** is selected.
- **Notes** Add a brief description [Optional].

Click **Commit** to save. The screen below shows the SIP Domain defined for the sample configuration.

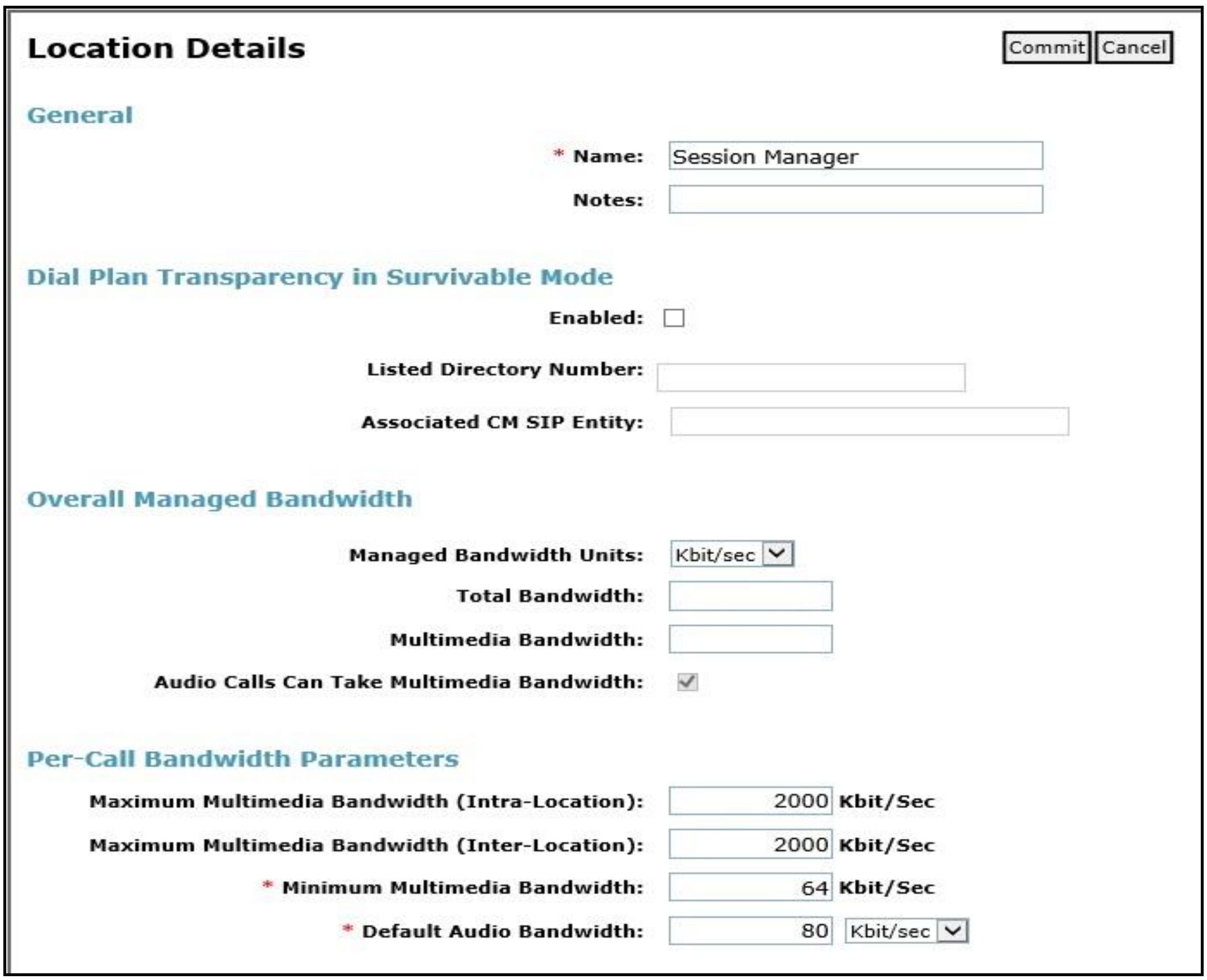


6.3. Administer Locations

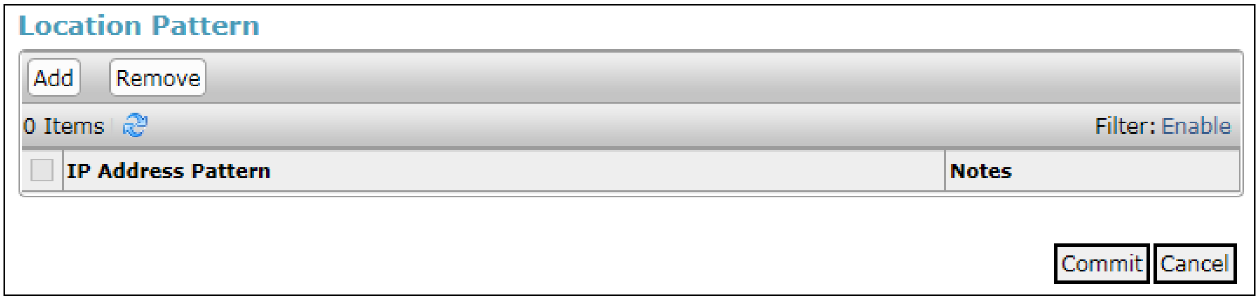
Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management and call admission control. To add a location, navigate to **Routing →Locations** in the left-hand navigation pane and click the **New** button in the right pane (not shown).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

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

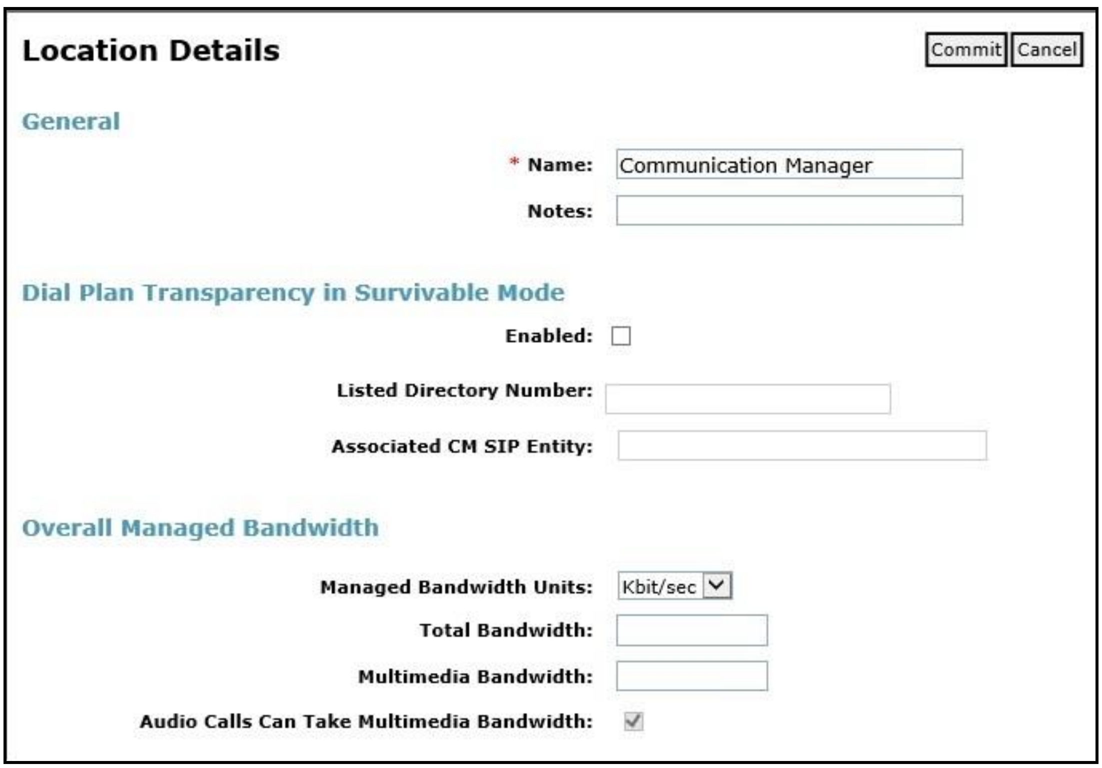
The following screenshot shows the location details named **Session Manager**. This location is assigned to the SIP Entity called Session Manager in **Section 6.5.1**.



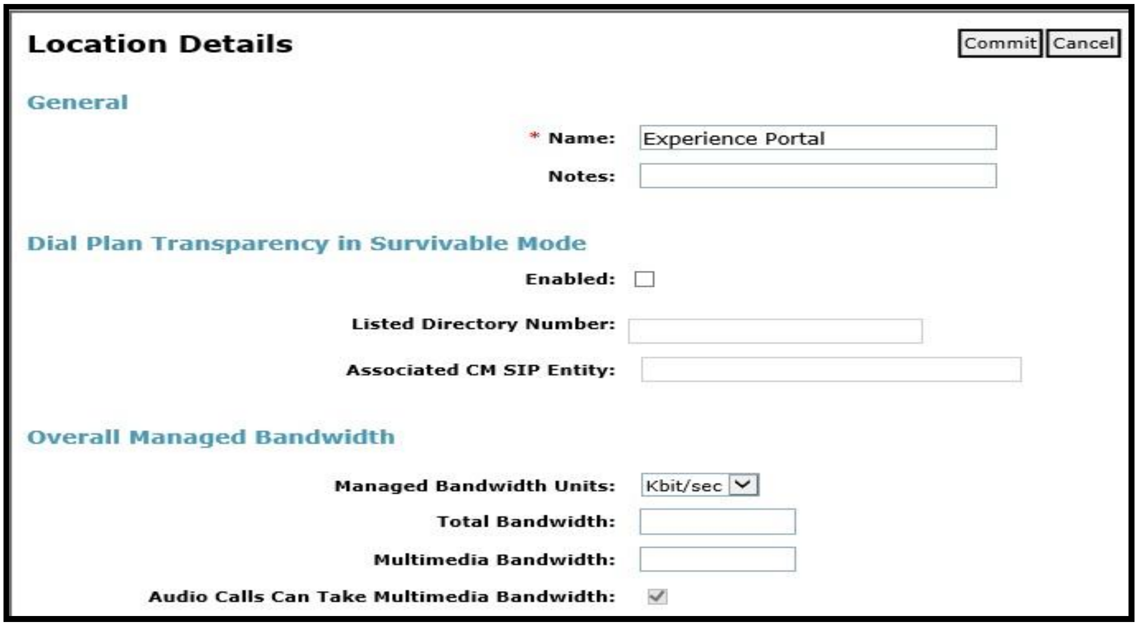
The location pattern is a way of using subnets to further refine the location information, this may be useful for endpoints that could be logged in from different subnets. If required, scroll to the bottom of the page and under **Location Pattern**, click **Add**, then enter an **IP Address Pattern** in the resulting new row, * is used to specify any number of allowed characters at the end of the string.



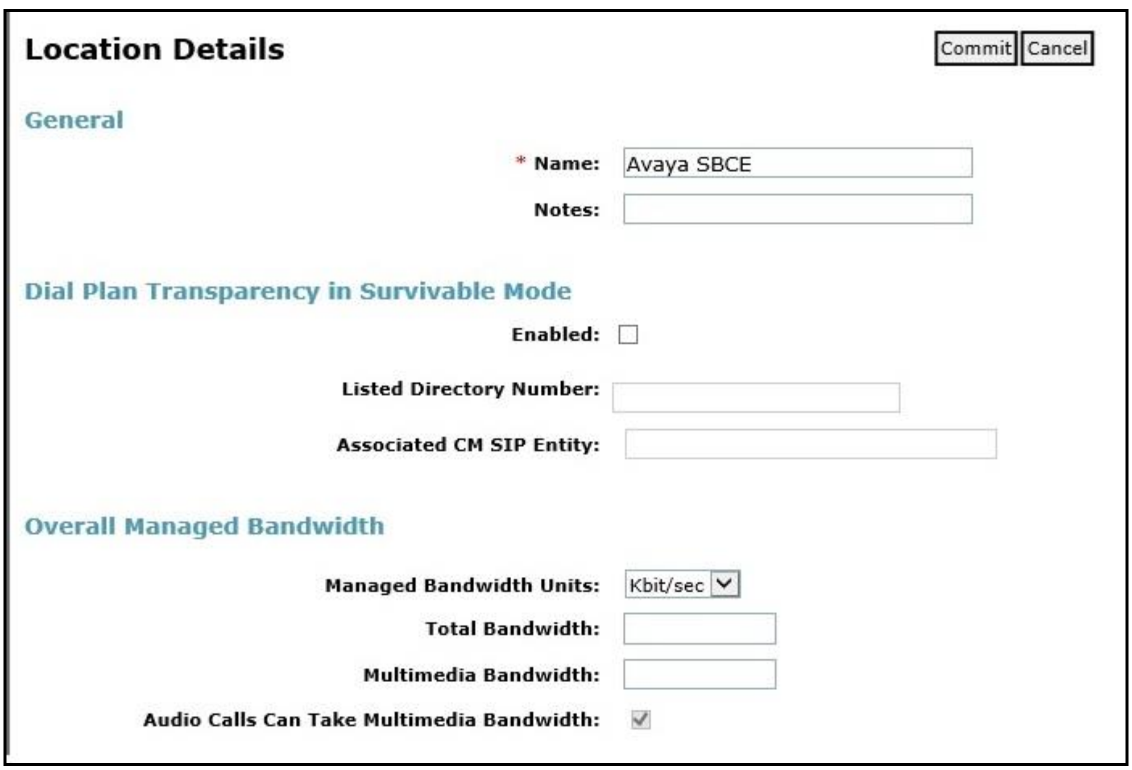
The following screenshot shows the location details named **Communication Manager**. This location is assigned to the SIP Entity called Communication Manager in **Section 6.5.2**.



The following screenshot shows the location details named **Experience Portal**. This location is assigned to the SIP Entity called Experience Portal in **Section 6.5.3**.



The following screenshot shows the location details named **Avaya SBC**. This location is assigned to the SIP Entity called Avaya SBC in **Section 6.5.4**.



6.4. Administer Adaptations

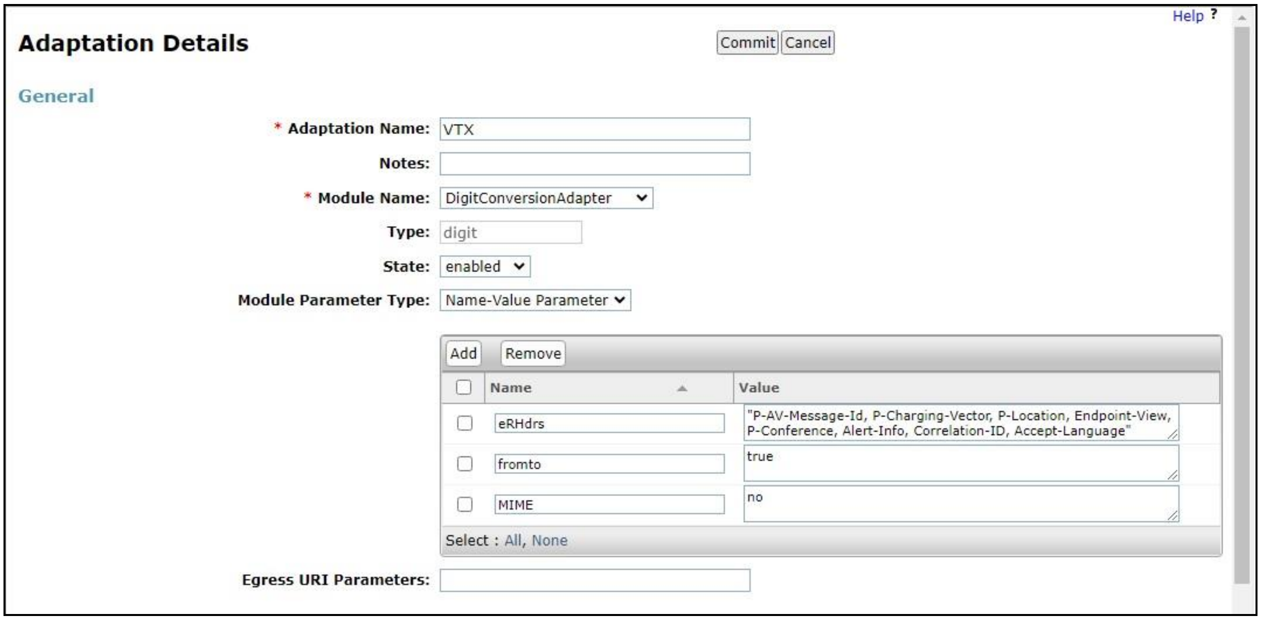
Session Manager Adaptations can be used to alter parameters in the SIP message headers. An Adaptation was used during testing to remove Avaya proprietary headers from messages sent. Adaptations can be used to modify the called and calling party numbers to meet the requirements of the service. The called party number present in the SIP INVITE Request URI is modified by the **DigitConversionAdapter** in the Adaptation. In order to improve interoperability with third party elements, Session Manager R10.1 incorporates the ability to use Adaptation modules to remove specific SIP headers that are either Avaya proprietary unnecessary for non-Avaya elements
For the compliance test, an Adaptation named "**VTX**" was created to block the following headers from outbound messages, before they were forwarded to the Avaya SBC: AV-Global-Session-ID, 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 and also add unnecessary size to outbound messages, while they have no significance to the service provider.

To add an adaptation, under the **Routing** tab select **Adaptations** on the left-hand menu and then click on the **New** button (not shown). Under **Adaptation Details** →**General**:

- **Adaption Name**: Enter an appropriate name such as **VTX**.
- **Module Name**: Select **DigitConversionAdapter**.
- **Modular Parameter Type**: Select **Name-Value Parameter**.

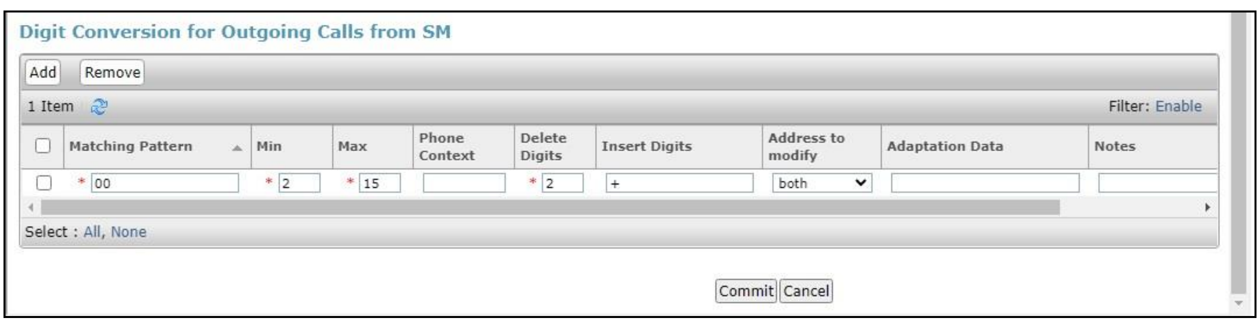
Click **Add** to add the name and value parameters.

- **Name**: Enter **eRHdrs**. This parameter will remove the specific headers from messages in the egress direction.
- **Value**: Enter **AV-Global-Session-ID**, **AV-Correlation-ID**, **Alert-Info**, **Endpoint-View**, **P-AV-Message-ID**, **P-Charging-Vector**, **P-Location**.
- **Name**: Enter **fromto**. Modifies From and To header of a message.
- **Value**: Enter **true**.
- **Name**: Enter **MIME**. Remove MIME message bodies from Session Manager.
- **Value**: Enter **no**.



Scroll down the page and under **Digit Conversion for Outgoing Calls from SM**, click the **Add** button and specify the digit manipulation to be performed as follows:

- Enter the leading digits that will be matched in the Matching Pattern field.
- In the **Min** and **Max** fields set the minimum and maximum digits allowed in the digit string to be matched.
- In the **Delete Digits** field enter the number of leading digits to be removed.
- In the **Insert Digits** field specify the digits to be prefixed to the digit string.
- In the **Address to modify** field specify the digits to manipulate by the adaptation. In this configuration the dialed number is the target so **both** have been selected.



This will ensure any outgoing national numbers matching 0 will have a prefix 0 inserted and any outgoing international numbers matching 00 will have prefix's 00 deleted and have + inserted being converted to E.164 format before being forwarded to the Avaya SBC.

6.5. Administer SIP Entities

A SIP Entity must be added for each SIP-based telephony system supported by a SIP connection to Session Manager. To add a SIP Entity, select **SIP Entities** on the left panel menu and then click on the **New** button (not shown). The following will need to be entered for each SIP Entity. Under **General:**

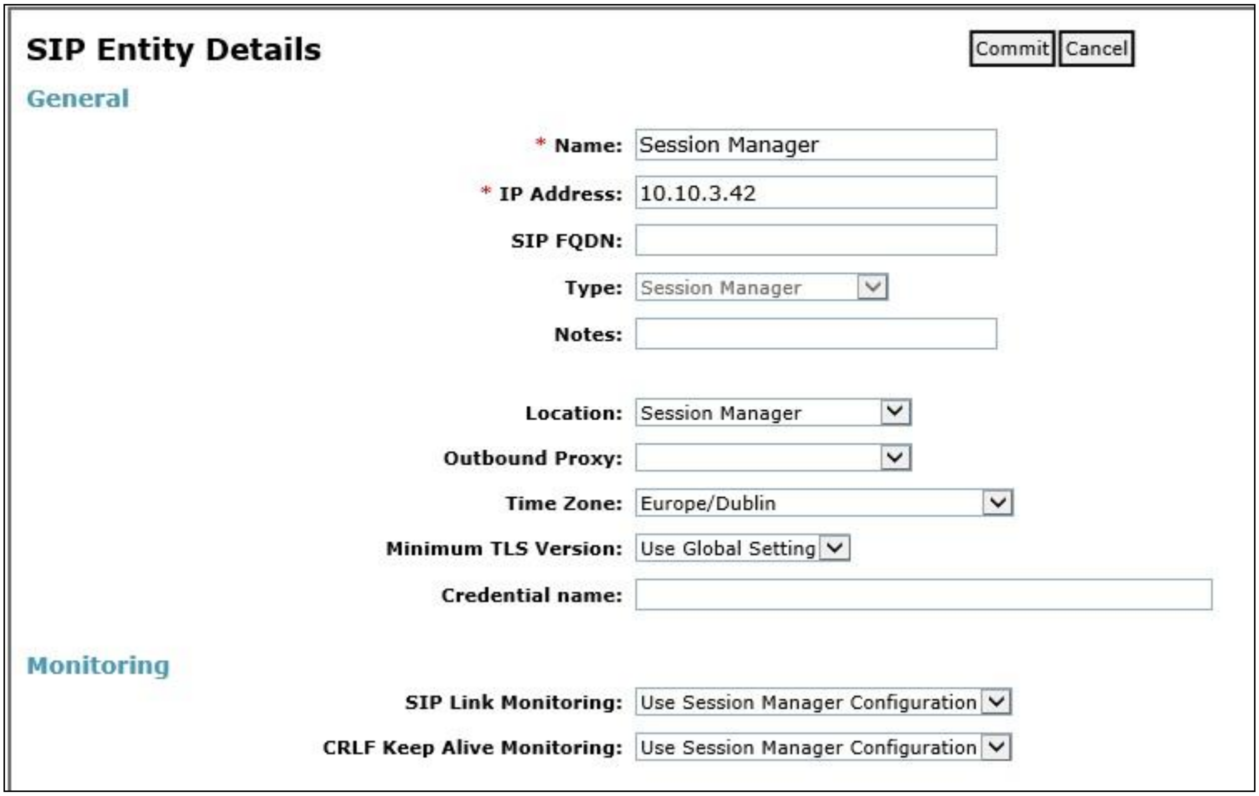
- In the **Name** field enter an informative name.
- In the **FQDN or IP Address** field enter the IP address of Session Manager or the signalling interface on the connecting system.
- In the **Type** field use **Session Manager** for a Session Manager SIP Entity, **CM** for a Communication Manager SIP Entity and **SIP Trunk** for the Avaya SBC SIP Entities.
- In the **Location** field select the appropriate location from the drop-down menu.
- In the **Time Zone** field enter the time zone for the SIP Entity.

In this configuration there are four SIP Entities.

- Session Manager SIP Entity.
- Communication Manager SIP Entity.
- Experience Portal SIP Entity.
- Avaya SBC SIP Entity.

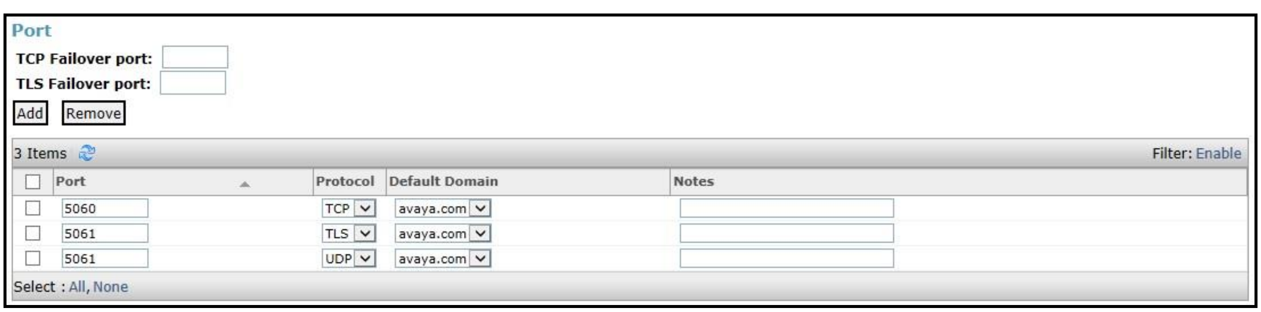
6.5.1. Avaya Aura® Session Manager SIP Entity

The following screens show the SIP entity for Session Manager. The **FQDN or IP Address** field is set to the IP address of the Session Manager SIP signalling interface and **Type** is **Session Manager**. Set the **Location** to that defined for Session Manager in **Section 6.3** and the **Time Zone** to the appropriate time zone.



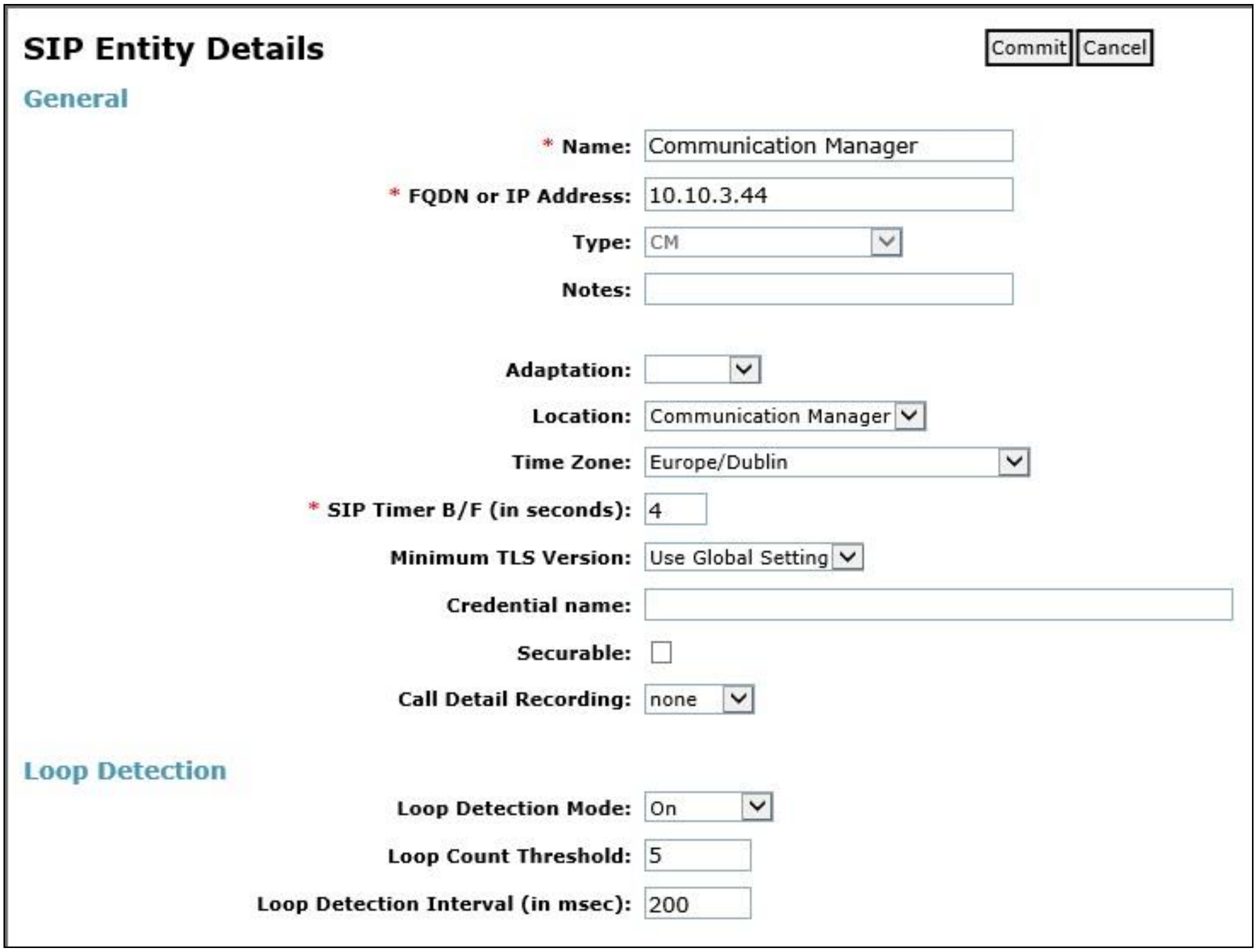
Session Manager must be configured with the port numbers on the protocols that will be used by the other SIP entities. To configure these scroll to the bottom of the page and under **Port**, click **Add**, then edit the fields in the resulting new row.

- In the **Port** field enter the port number on which the system listens for SIP requests.
- In the **Protocol** field enter the transport protocol to be used for SIP requests.
- In the **Default Domain** field, from the drop-down menu select the domain added in **Section 6.2** as the default domain.



6.5.2. Avaya Aura® Communication Manager SIP Entity

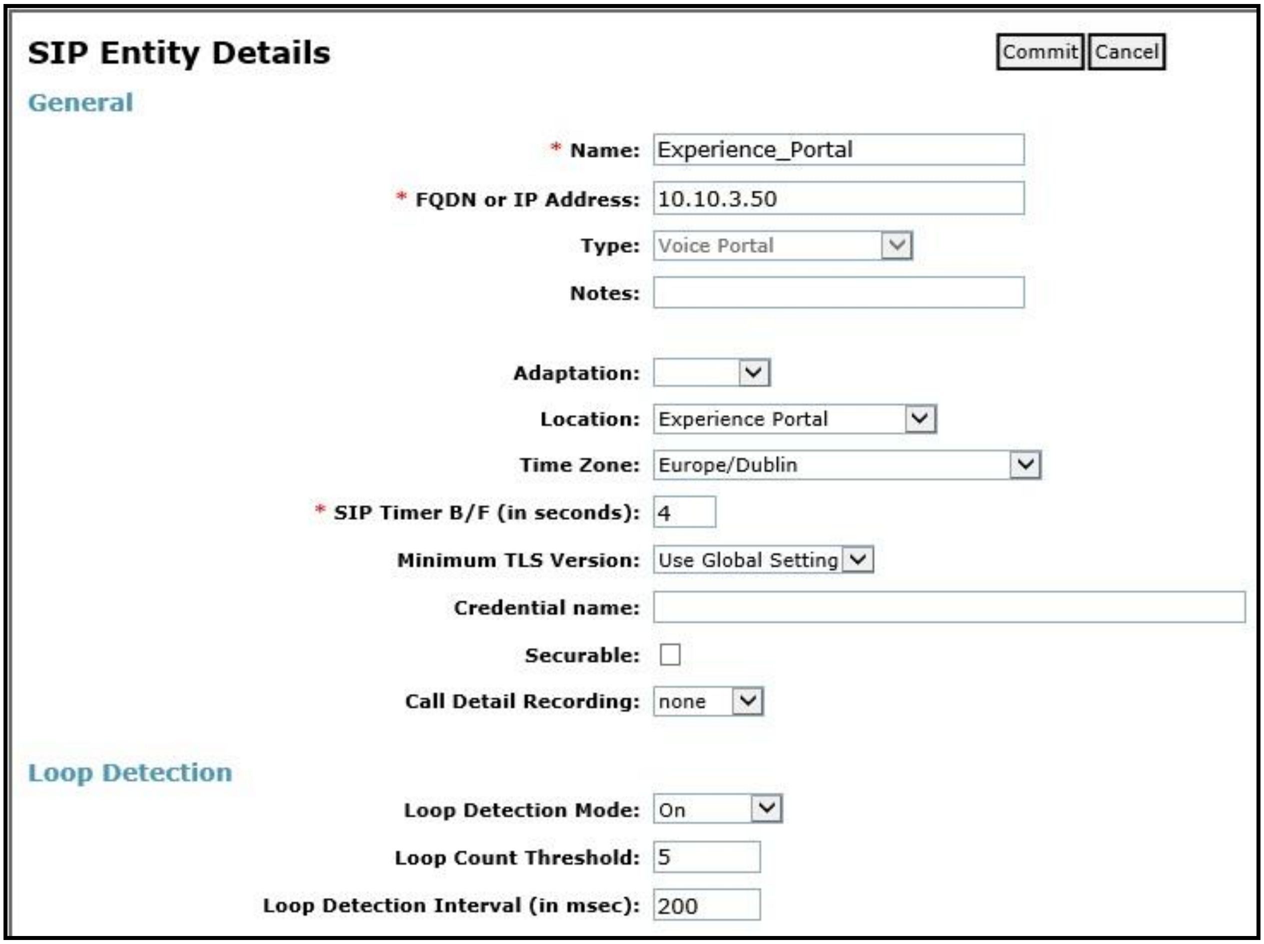
The following screen shows the SIP entity for Communication Manager. This SIP Entity is used for the SIP Trunk. The **FQDN or IP Address** field is set to the IP address of the interface on Communication Manager that will be providing SIP signalling. Set the **Location** to that defined for Communication Manager in **Section 6.3** and the **Time Zone** to the appropriate time zone.



Other parameters can be set for the SIP Entity as shown in the following screenshot, but for test, these were left at default values.

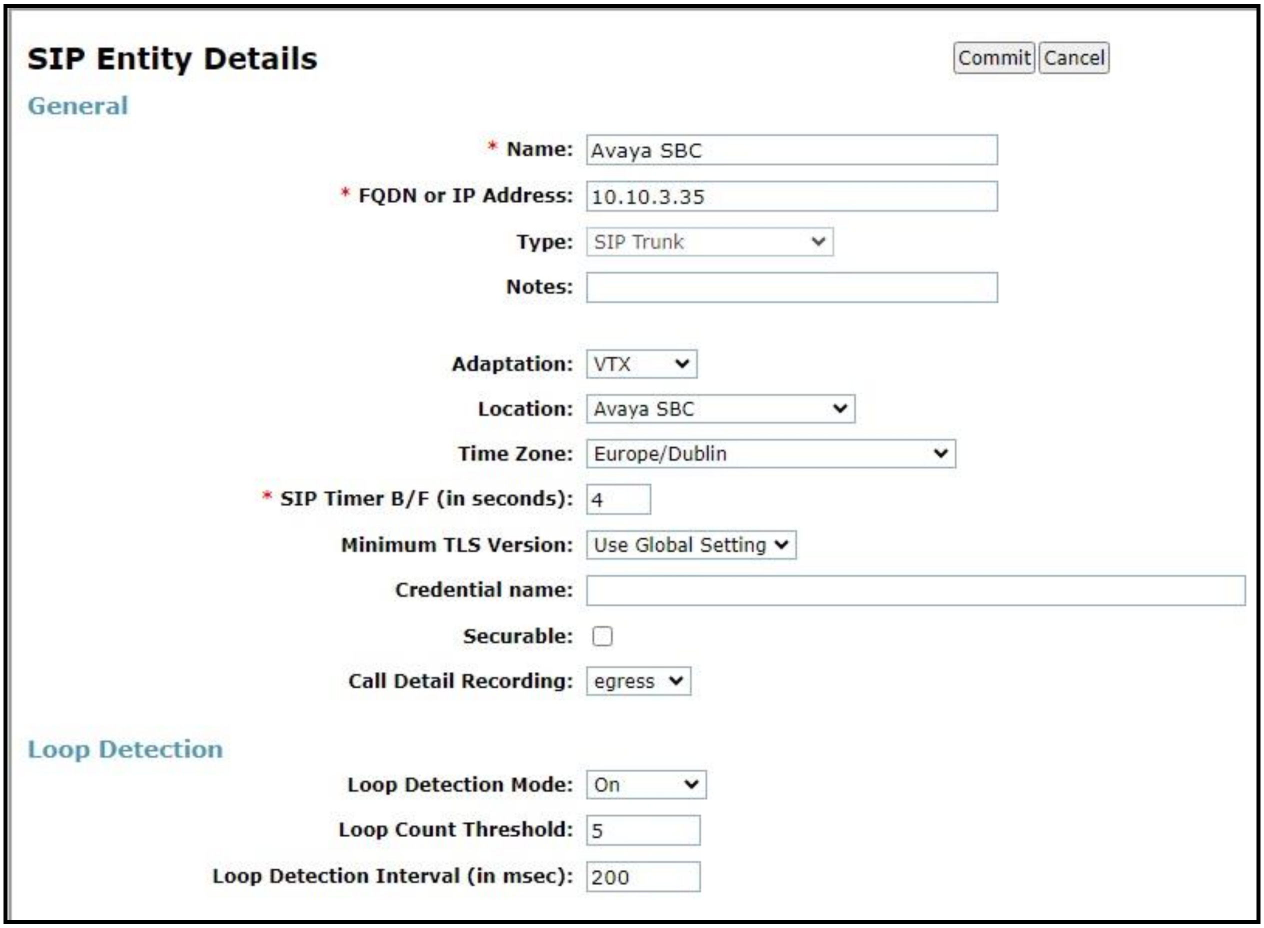
6.5.3. Avaya Experience Portal SIP Entity

The following screen shows the SIP entity for Experience Portal. The **FQDN or IP Address** field is set to the IP address of the Experience Portal. Set the **Location** to that defined for Experience Portal in **Section 6.3** and the **Time Zone** to the appropriate time zone.



6.5.4. Avaya Session Border Controller SIP Entity

The following screen shows the SIP Entity for the Avaya SBC used for PSTN destinations. The **FQDN or IP Address** field is set to the IP address of the Avaya SBC private network interface (See **Section 8.4.1**). Set the **Adaptation** to that defined in **Section 6.4**, the **Location** to that defined for Avaya SBC in **Section 6.3** and the **Time Zone** to the appropriate time zone.



6.6. Administer Entity Links

A SIP trunk between a Session Manager and another system is described by an Entity Link. To add an Entity Link, select **Entity Links** on the left panel menu and click on the **New** button (not shown). Fill in the following fields in the new row that is displayed.

- In the **Name** field enter an informative name.
- In the **SIP Entity 1** field select **Session Manager**.
- In the **Protocol** field enter the transport protocol to be used to send SIP requests.
- In the **Port** field enter the port number to which the other system sends its SIP requests.
- In the **SIP Entity 2** field enter the other SIP Entity for this link, created in **Section 6.5**.
- In the **Port** field enter the port number to which the other system expects to receive SIP requests.
- Select **Trusted** from the drop-down menu to make the other system trusted.

Click **Commit** to save changes. The following screenshot shows the Entity Links used in this configuration.

Help ?

Entity Links

New | Edit | Delete | Duplicate | More Actions

5 Items | Filter: Enable

	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	DNS Override	Connection Policy	Deny New Service	Notes
☐	Communication_Manager	Session Manager	TLS	5061	Communication Manager	5061	☐	trusted	☐	
☐	Experience_Portal	Session Manager	TLS	5061	Experience Portal	5061	☐	trusted	☐	
☐	IX_Messaging	Session Manager	TLS	5061	IX_Messaging	5061	☐	trusted	☐	
☐	Messaging	Session Manager	TLS	5061	Aura_Messaging	5061	☐	trusted	☐	
☐	to_Avaya_SBC	Session Manager	TLS	5061	Avaya SBC	5061	☐	trusted	☐	

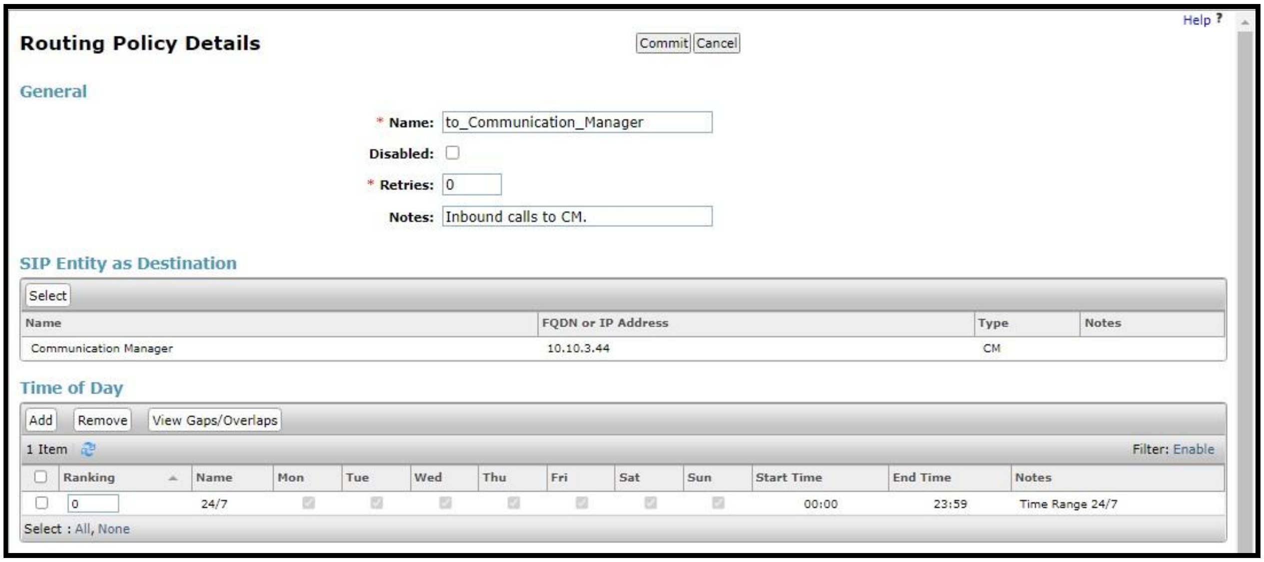
Select : All, None

6.7. Administer Routing Policies

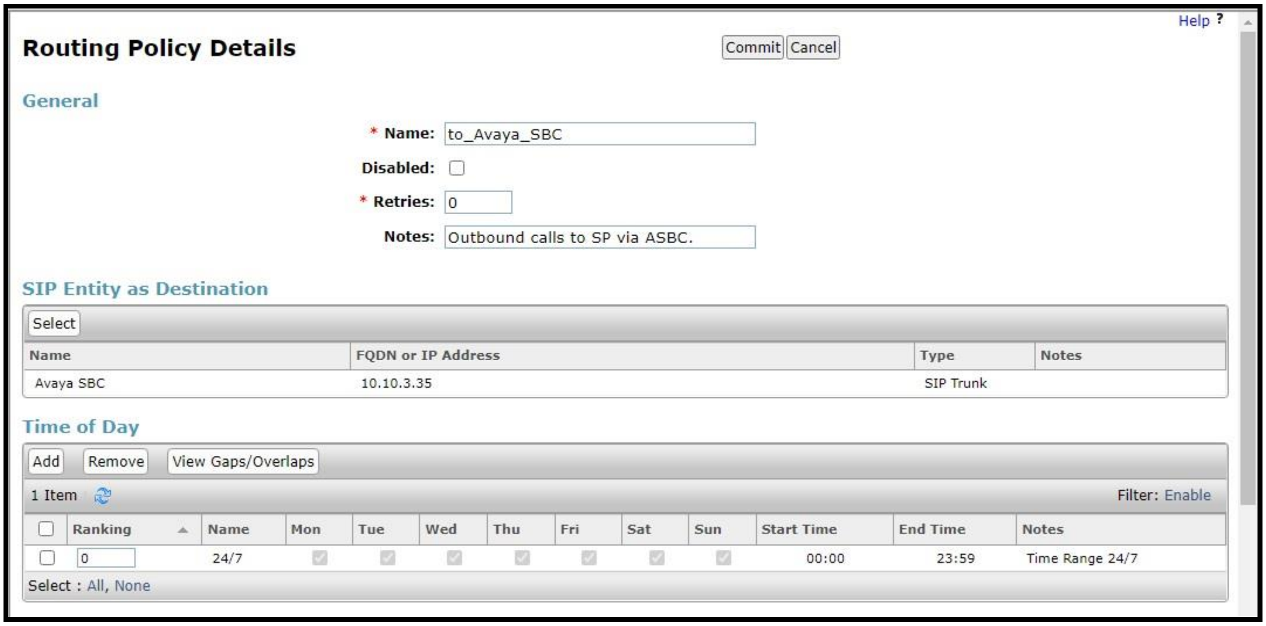
Routing policies must be created to direct how calls will be routed to a system. To add a routing policy, select **Routing Policies** on the left panel menu and then click on the **New** button (not shown). Under **General**:

- Enter an informative name in the **Name** field.
- Under **SIP Entity as Destination**, click **Select**, and then select the appropriate SIP entity to which this routing policy applies.
- Under **Time of Day**, click **Add**, and then select the time range.

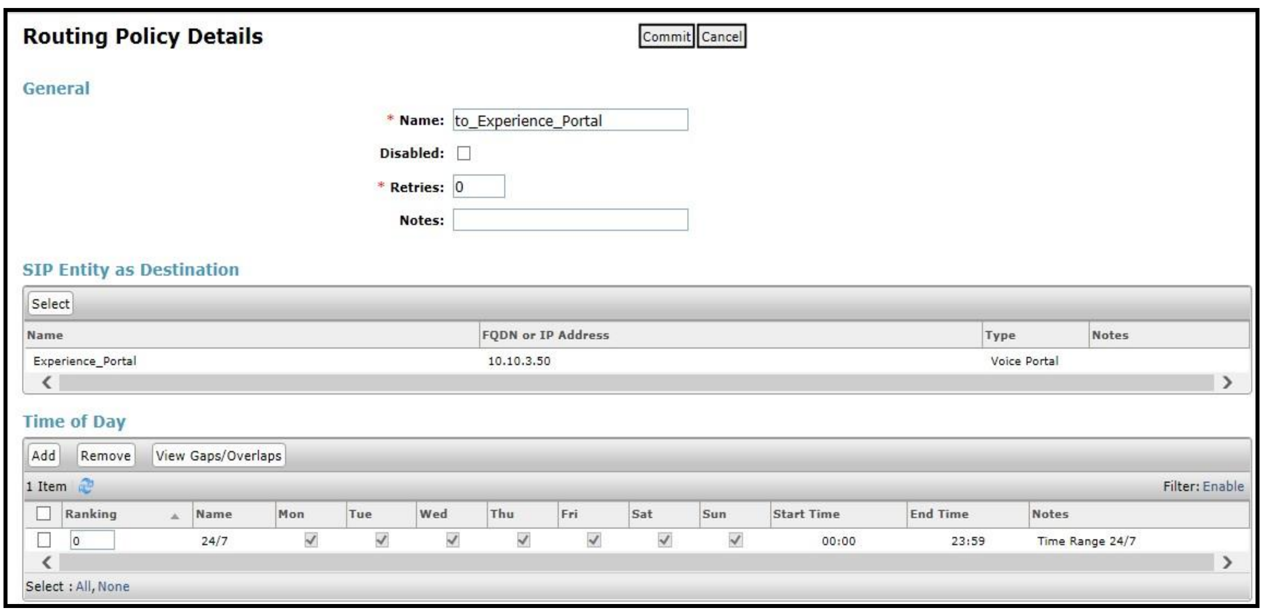
The following screen shows the routing policy for inbound calls from the VTX SIP Trunk to Communication Manager.



The following screen shows the routing policy for outbound calls from Communication Manager via Avaya SBC to the VTX SIP trunk.



The following screen shows the routing policy for calls inbound from the SIP Trunk to Experience Portal.



6.8. Administer Dial Patterns

A dial pattern must be defined to direct calls to the appropriate telephony system. To configure a dial pattern, select **Dial Patterns** on the left panel menu and then click on the **New** button (not shown).

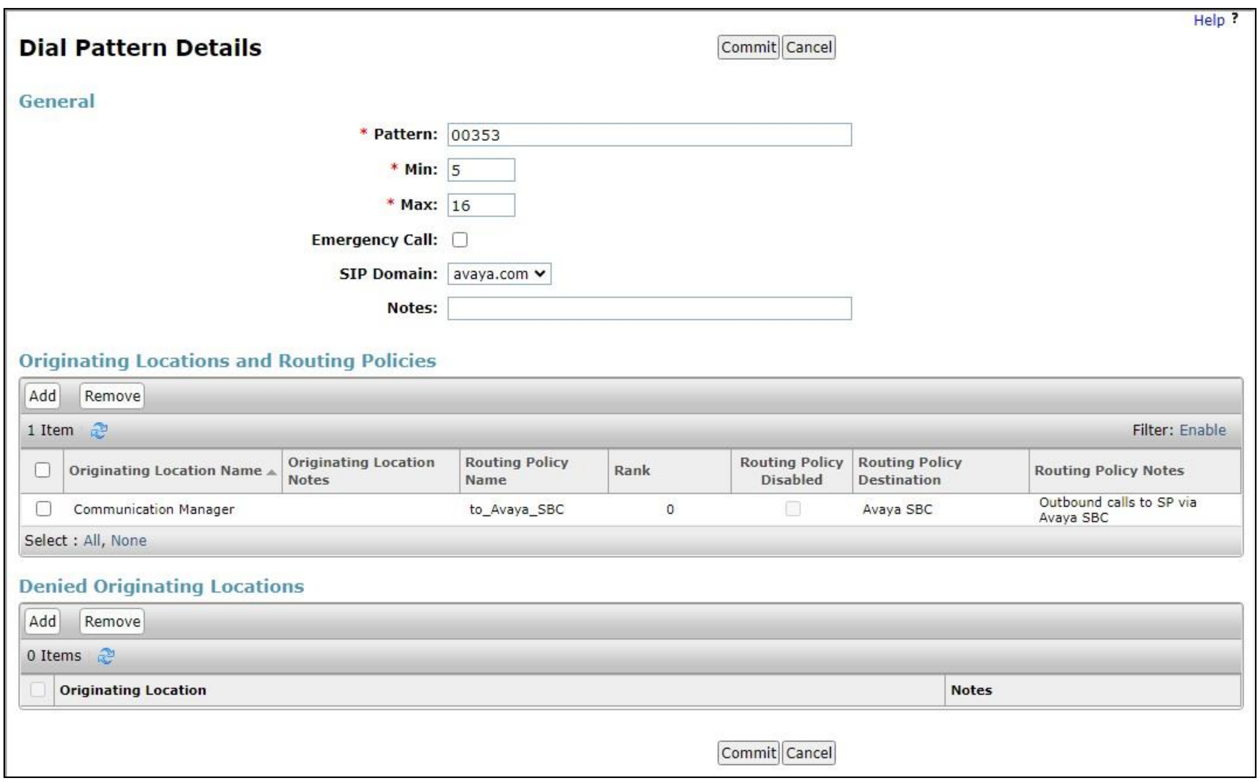
Under **General**:

- In the **Pattern** field enter a dialled number or prefix to be matched.
- In the **Min** field enter the minimum length of the dialled number.
- In the **Max** field enter the maximum length of the dialled number.
- In the **SIP Domain** field select **ALL** or alternatively one of those configured in **Section 6.2**.

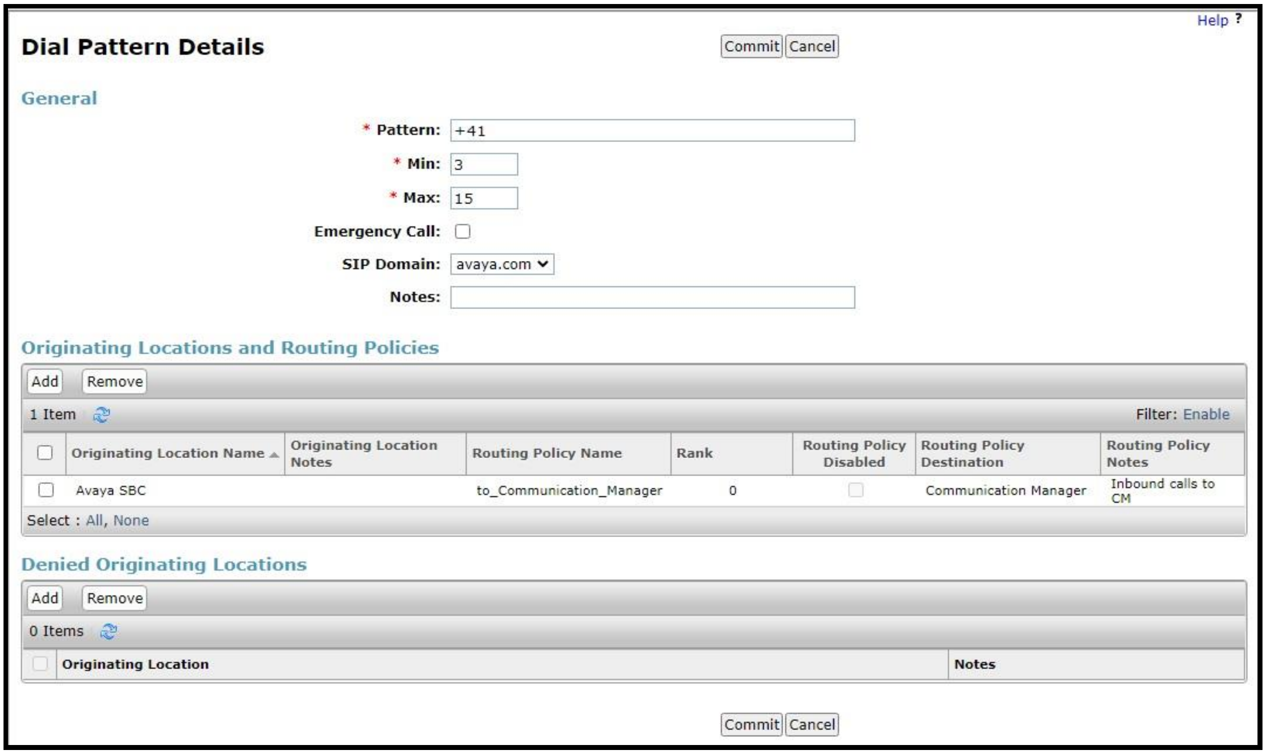
Under **Originating Locations and Routing Policies**:

- Click **Add**, in the resulting screen (not shown).
- Under **Originating Location**, select the location defined in **Section 6.3** or **ALL**.
- Under **Routing Policies** select one of the routing policies defined in **Section 6.7**.
- Click **Select** button to save.

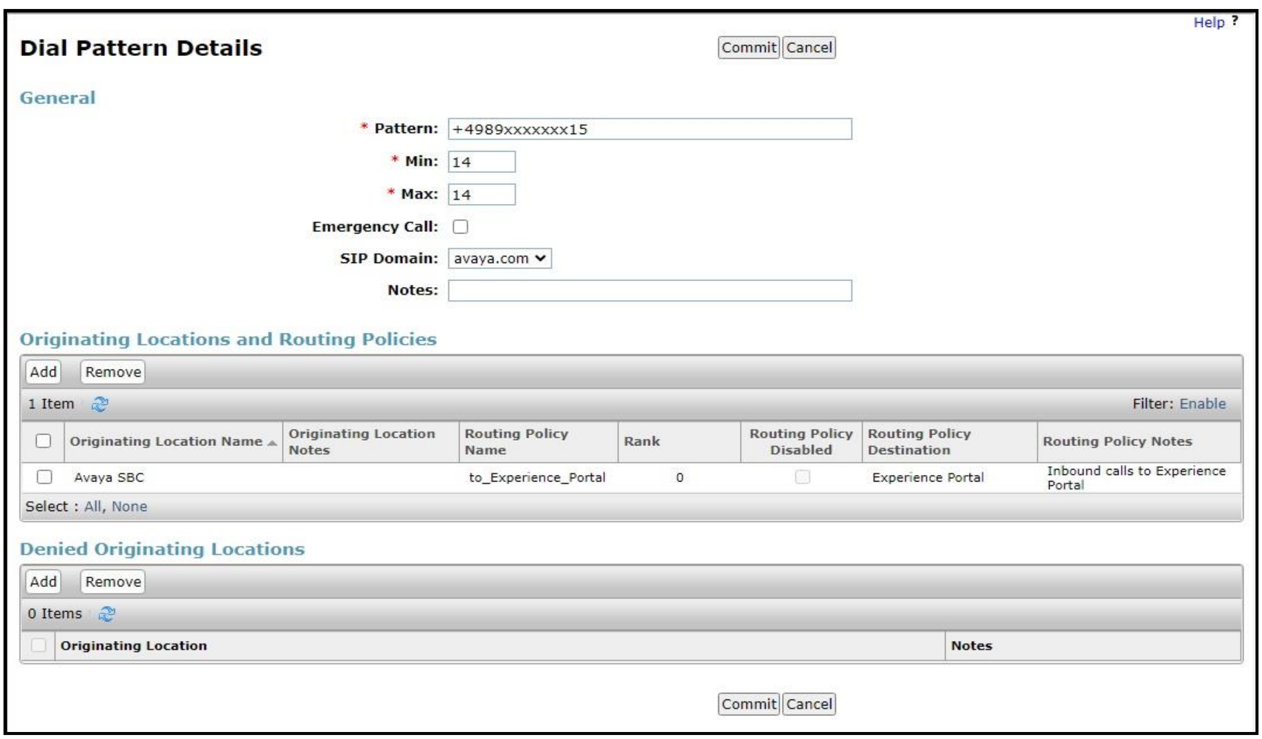
The following screen shows an example dial pattern configured for outbound calls to the VTX SIP Trunk.



The following screen shows the dial pattern configured for inbound calls to Communication Manager.



The following screen shows the dial pattern configured for inbound calls to Experience Portal.



7. Configure Avaya 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 [**16**] in the **References** section for further details if necessary.

7.1. Background

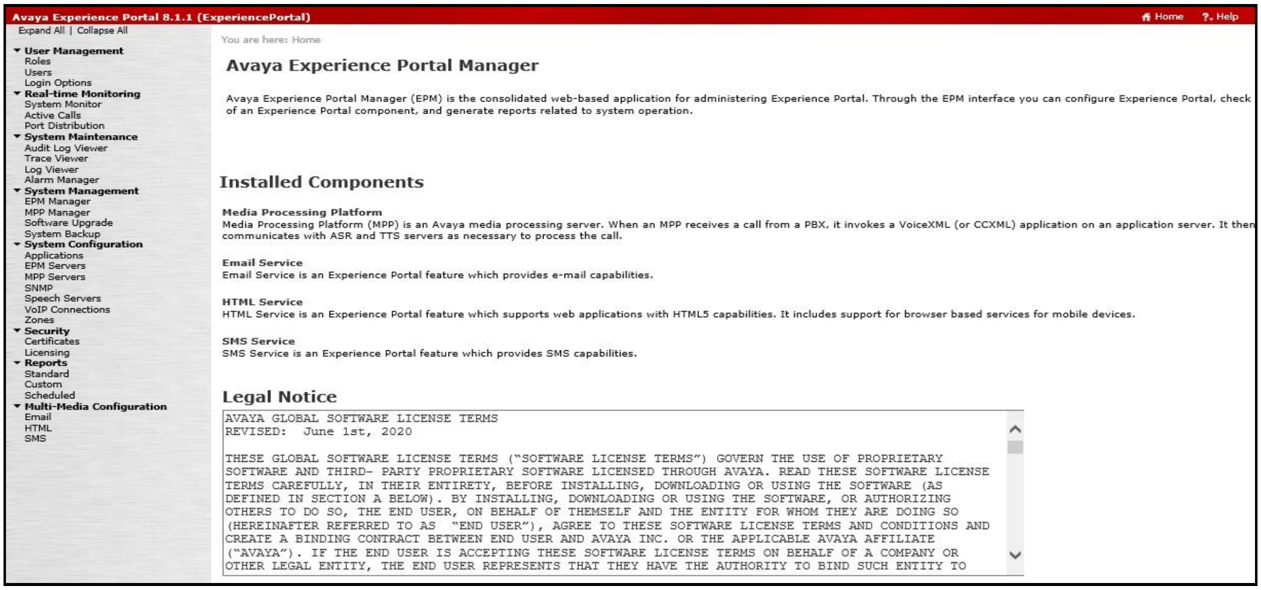
Experience Portal consists of one or more Media Processing Platform (MPP) servers and an Experience Portal Manager (EPM) server. A single “server configuration” was used in the reference configuration. This consisted of a single MPP and EPM, running on a VMware environment, including an Apache Tomcat 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 DDI 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 sample configuration described in these Application Notes. A simple VXML test application was used to exercise various SIP call flow scenarios with the VTX SIP Trunk 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.

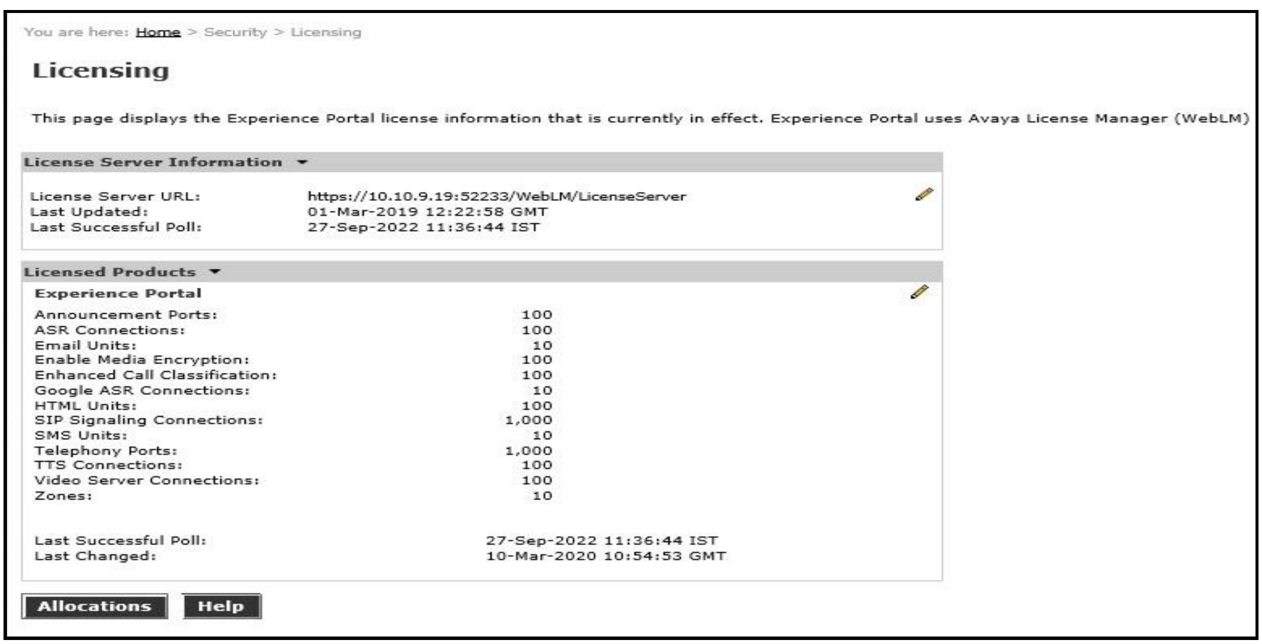
7.2. Logging In and Licensing

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.



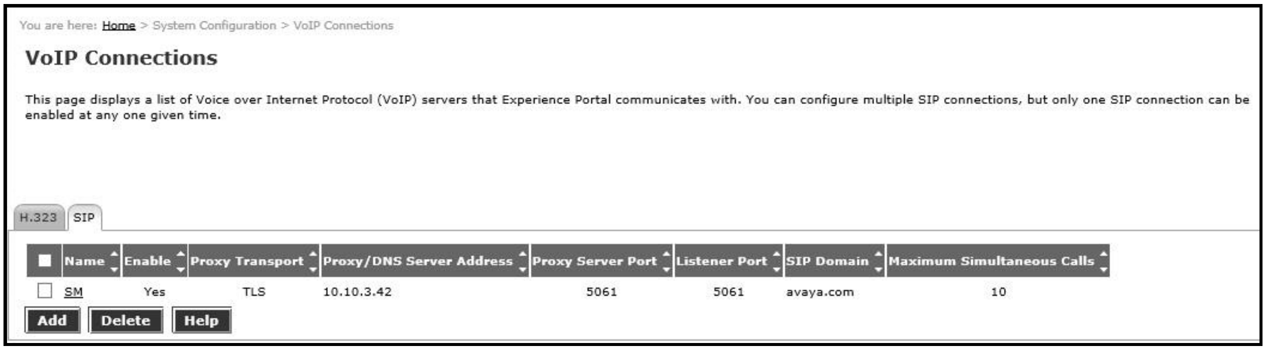
Step 2 - In the left pane, navigate to **Security→Licensing**. On the **Licensing** page, verify that Experience Portal is properly licensed. If required licenses are not enabled, contact an authorized Avaya representative to obtain the licenses.



7.3. VoIP Connection

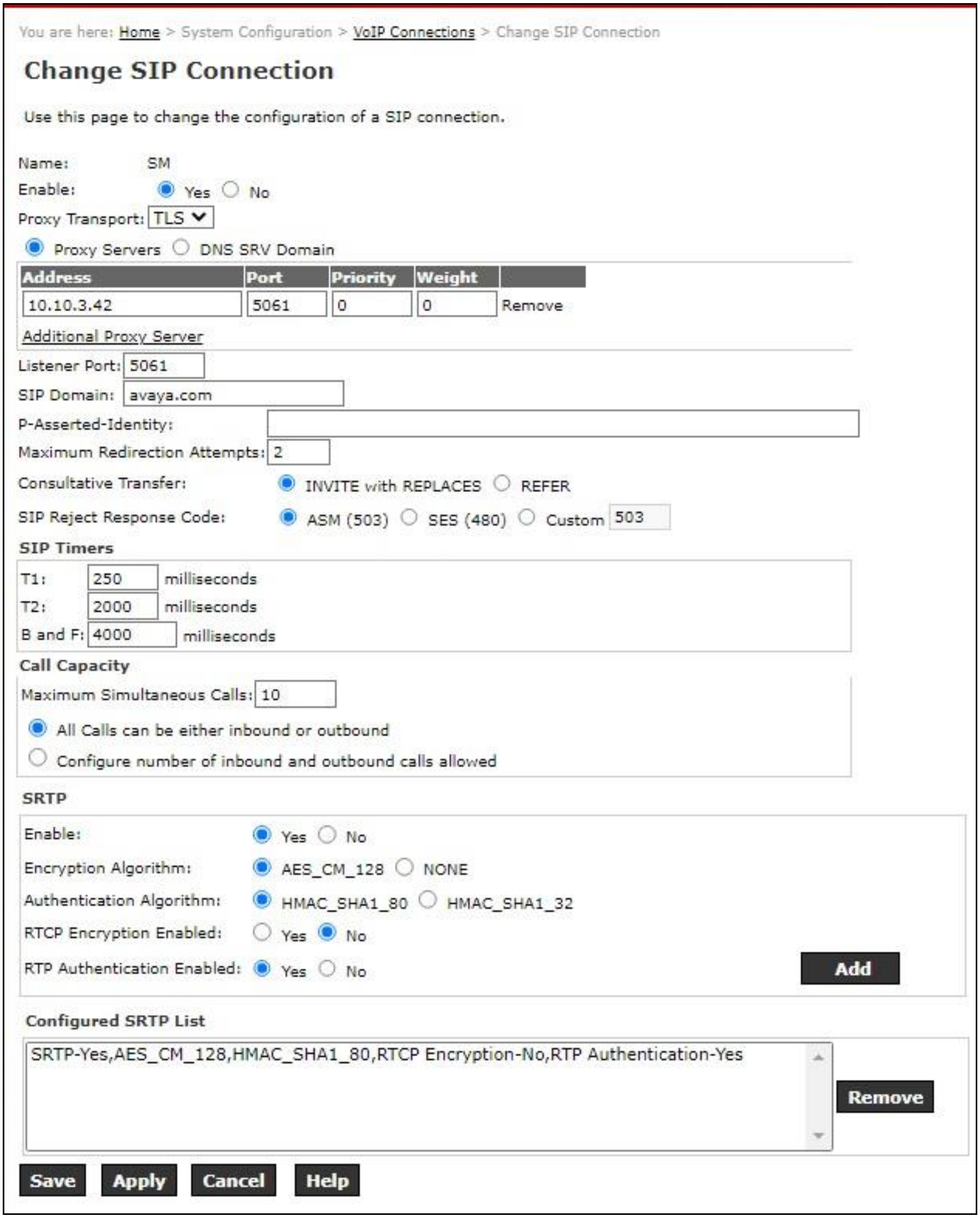
This section defines a SIP trunk between Experience Portal and Session Manager.

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., **SM**).
- **Enable** – Set to **Yes**.
- **Proxy Server Transport** – Set to **TLS**.
- Select **Proxy Servers**, and enter:
  - **Proxy Server Address** = **10.10.3.42** (the IP address of the Session Manager signaling interface defined in **Section 6.5.1**).
  - **Port** = **5061**
  - **Priority** = **0** (default)
  - **Weight** = **0** (default)
- **Listener Port** – Set to **5061**.
- **SIP Domain** – Set to **avaya.com** (see **Section 6.2)**.
- **Consultative Transfer** – Select **INVITE with REPLACES**.
- **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 **10** 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 **Add**.
- Use default values for all other fields and click **Save**.

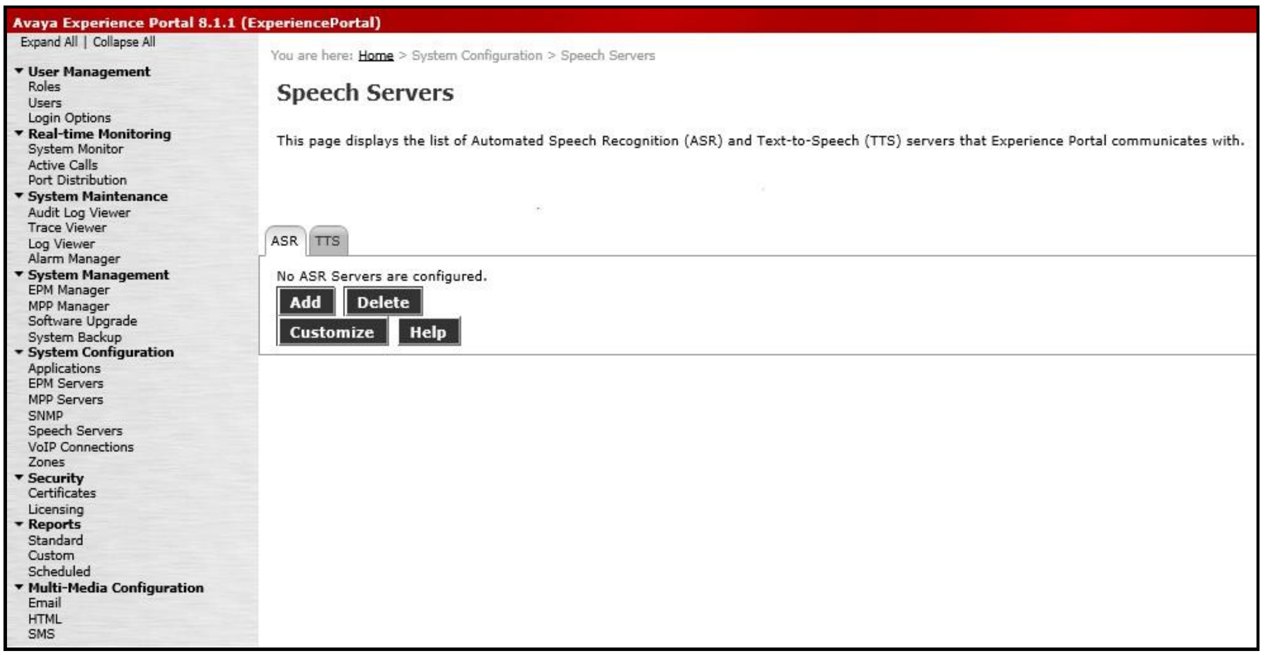

You are here: Home > System Configuration > VoIP Connections > Change SIP Connection
Change SIP Connection
Use this page to change the configuration of a SIP connection.
Name: SM
Enable: Yes No
Proxy Transport: TLS
Proxy Servers DNS SRV Domain
Address Port Priority Weight
10.10.3.42 5061 0 0 Remove
Additional Proxy Server
Listener Port: 5061
SIP Domain: avaya.com
P-Asserted-Identity:
Maximum Redirection Attempts: 2
Consultative Transfer: INVITE with REPLACES REFER
SIP Reject Response Code: ASM (503) SES (480) Custom 503
SIP Timers
T1: 250 milliseconds
T2: 2000 milliseconds
B and F: 4000 milliseconds
Call Capacity
Maximum Simultaneous Calls: 10
All Calls can be either inbound or outbound
Configure number of inbound and outbound calls allowed
SRTP
Enable: Yes No
Encryption Algorithm: AES_CM_128 NONE
Authentication Algorithm: HMAC_SHA1_80 HMAC_SHA1_32
RTCP Encryption Enabled: Yes No
RTP Authentication Enabled: Yes No
Add
Configured SRTP List
SRTP-Yes,AES_CM_128,HMAC_SHA1_80,RTCP Encryption-No,RTP Authentication-Yes
Remove
Save Apply Cancel Help

7.4. Speech Servers

Avaya Experience Portal system integrates with two types of third-party speech servers:

- Automatic Speech Recognition (ASR): This technology enables an interactive voice response (IVR) system to collect verbal responses from callers.
- Text-to-Speech (TTS): This technology enables an IVR system to render text content into synthesized speech output according to algorithms within the TTS software.

No speech servers were required as part of the test configuration. The installation and administration of the ASR and TTS Speech Servers are beyond the scope of this document

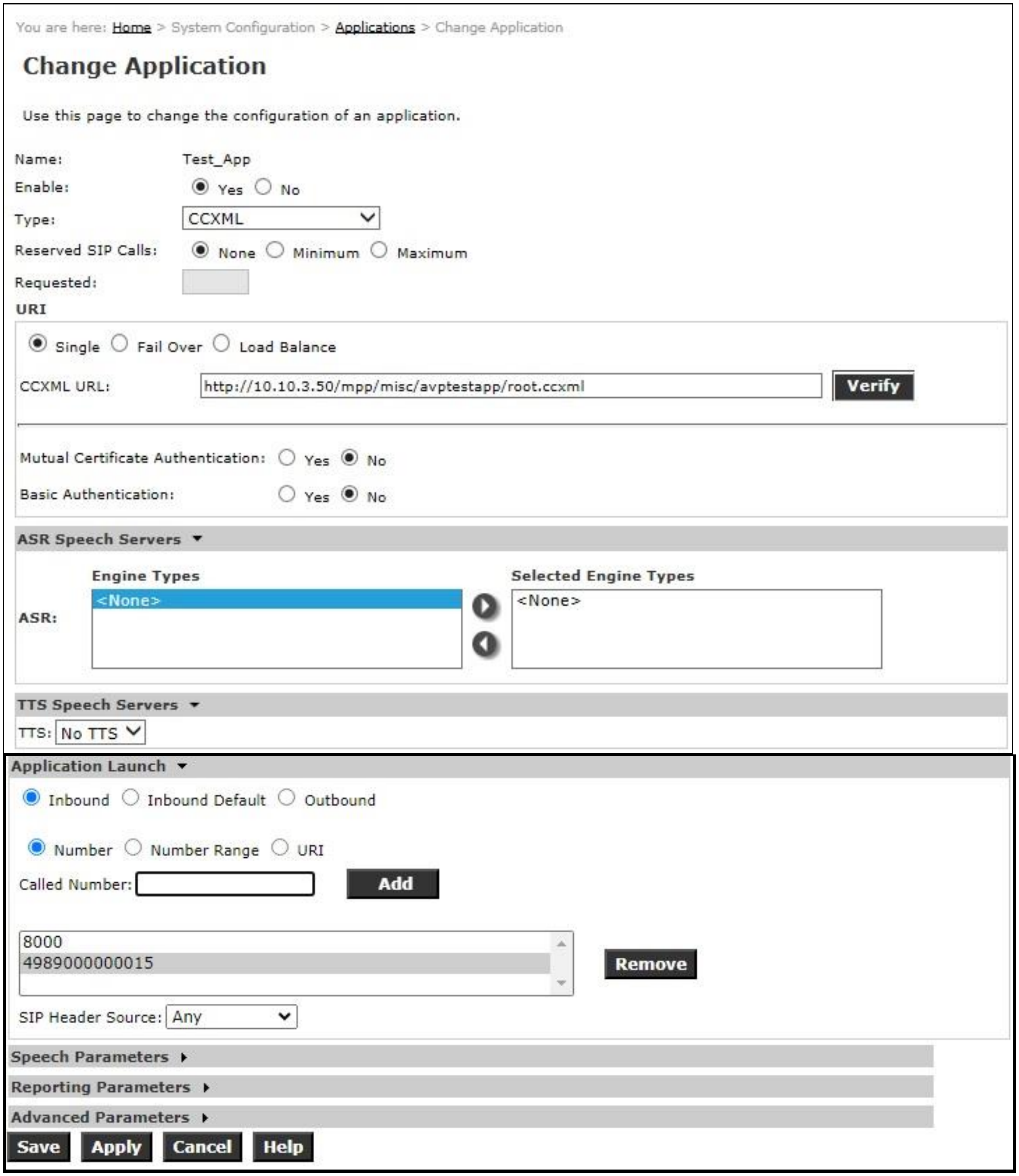


7.5. Application References

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 10.10.3.50.

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:

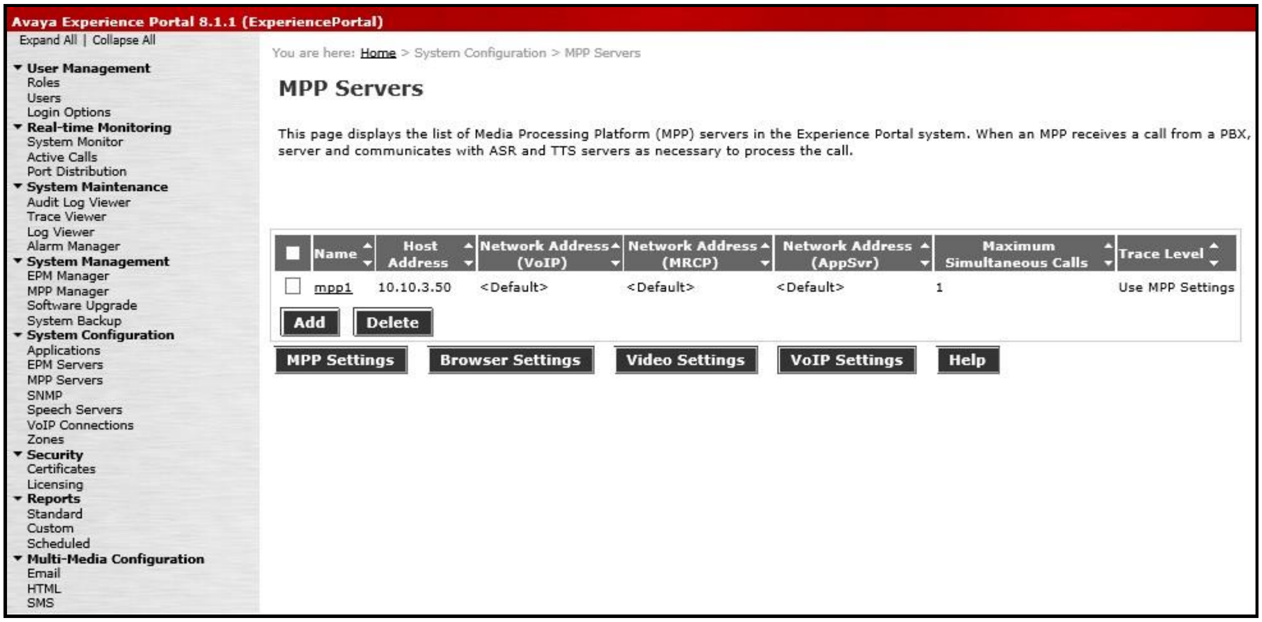
- **Name** – Set to a descriptive name (e.g., **Test_App**).
- **Enable** – Set to **Yes**. This field determines which application(s) will be executed based on their defined criteria.
- **Type** – Select **VoiceXML**, **CCXML**, or **CCXML/VoiceXML** according to the application type. CCXML was used in the test configuration.
- **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. CCXML was used in the test configuration.
- **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**.


You are here: Home > System Configuration > Applications > Change Application
Change Application
Use this page to change the configuration of an application.
Name: Test_App
Enable: Yes No
Type: CCXML
Reserved SIP Calls: None Minimum Maximum
Requested:
URI
Single Fail Over Load Balance
CCXML URL: http://10.10.3.50/mpp/misc/avptestapp/root.ccxml
Verify
Mutual Certificate Authentication: Yes No
Basic Authentication: Yes No
ASR Speech Servers
Engine Types
Selected Engine Types
ASR:
<None>
<None>
TTS Speech Servers
TTS: No TTS
Application Launch
Inbound Inbound Default Outbound
Number Number Range URI
Called Number:
Add
8000
4989000000015
Remove
SIP Header Source: Any
Speech Parameters
Reporting Parameters
Advanced Parameters
Save
Apply
Cancel
Help

7.6. 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**.

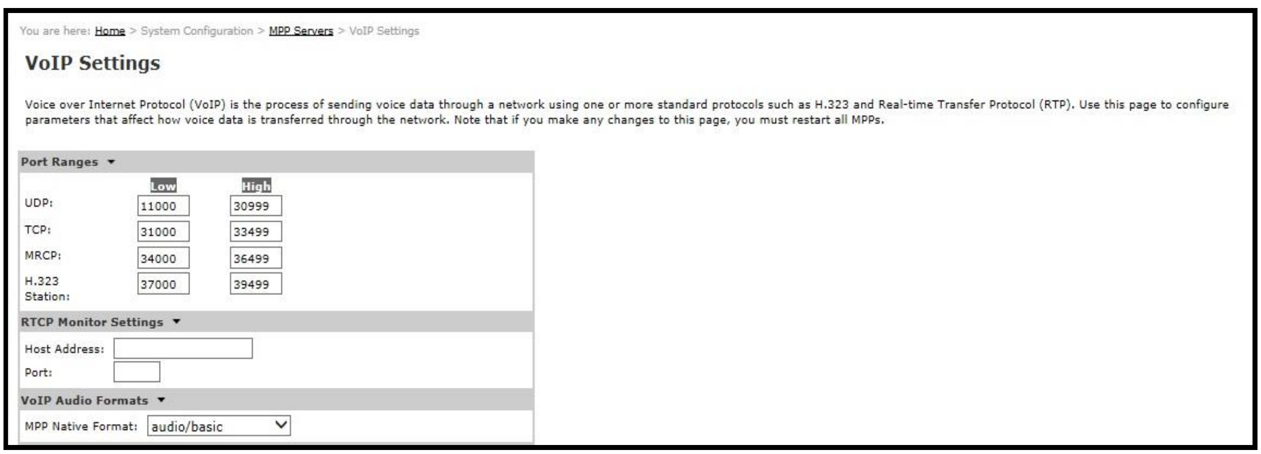


Step 2 - Enter any descriptive name in the **Name** field (e.g., **mpp1**) and the IP address of the MPP server in the **Host Address** field and click **Continue** (not shown).

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.

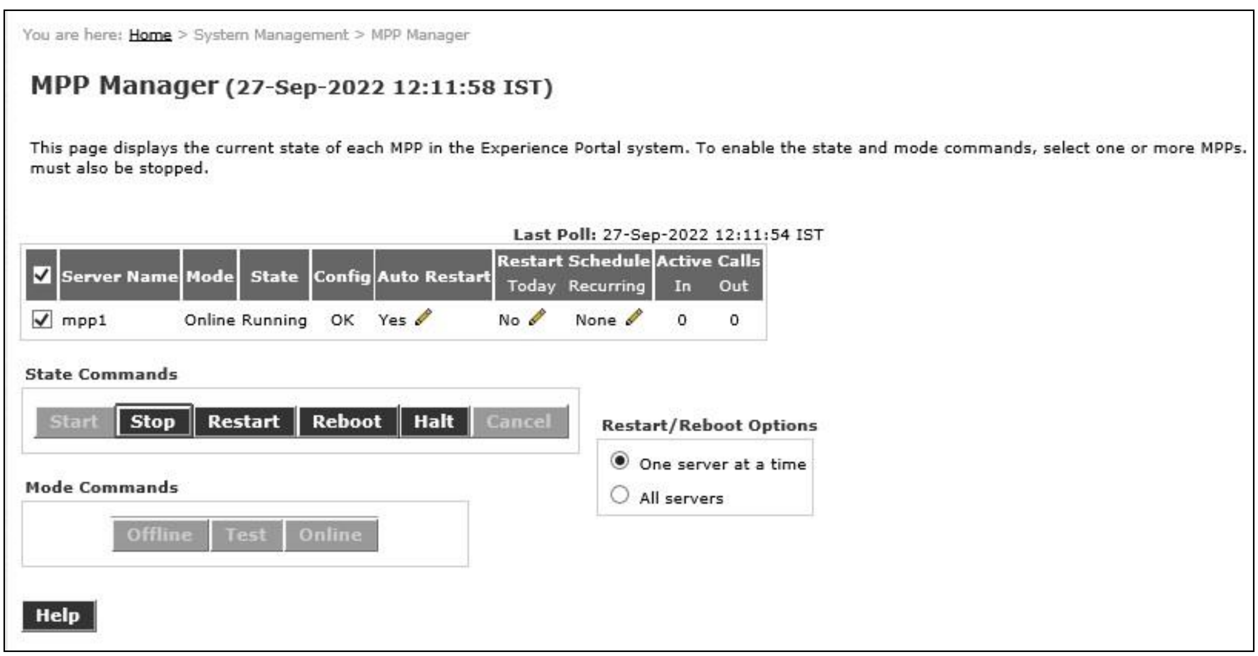


In the **Codecs** section set:

- Set **Packet Time** to **20**.
- Verify the **G711alaw**, **G729** and **G711ulaw** codecs are enabled.
- Set **G729 Discontinuous Transmission** to **No** (G.729A).
- Set the **Offer Order** to the preferred codec.
- Use default values for all other fields.

Step 5 - Click on **Save**.

After saving the configuration changes, 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.

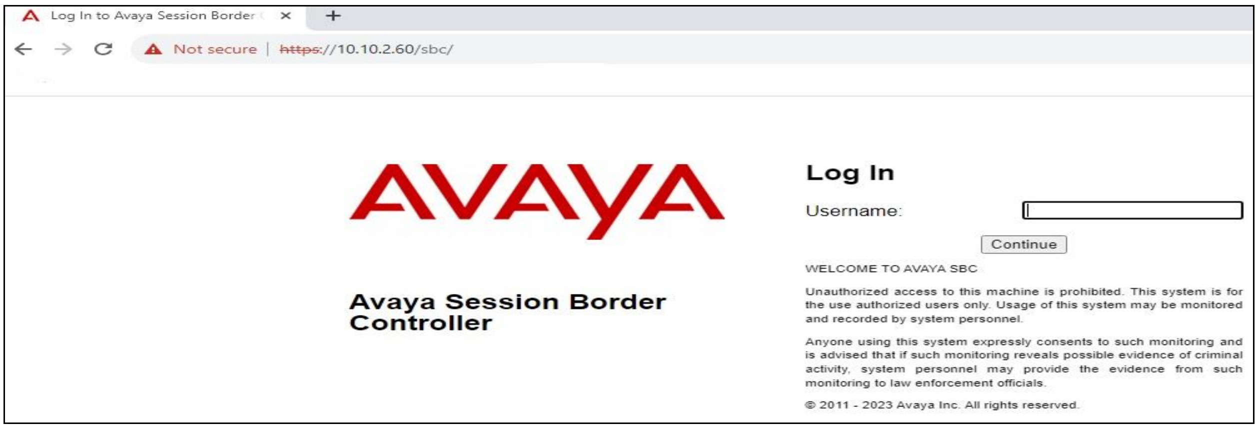


8. Configure Avaya Session Border Controller

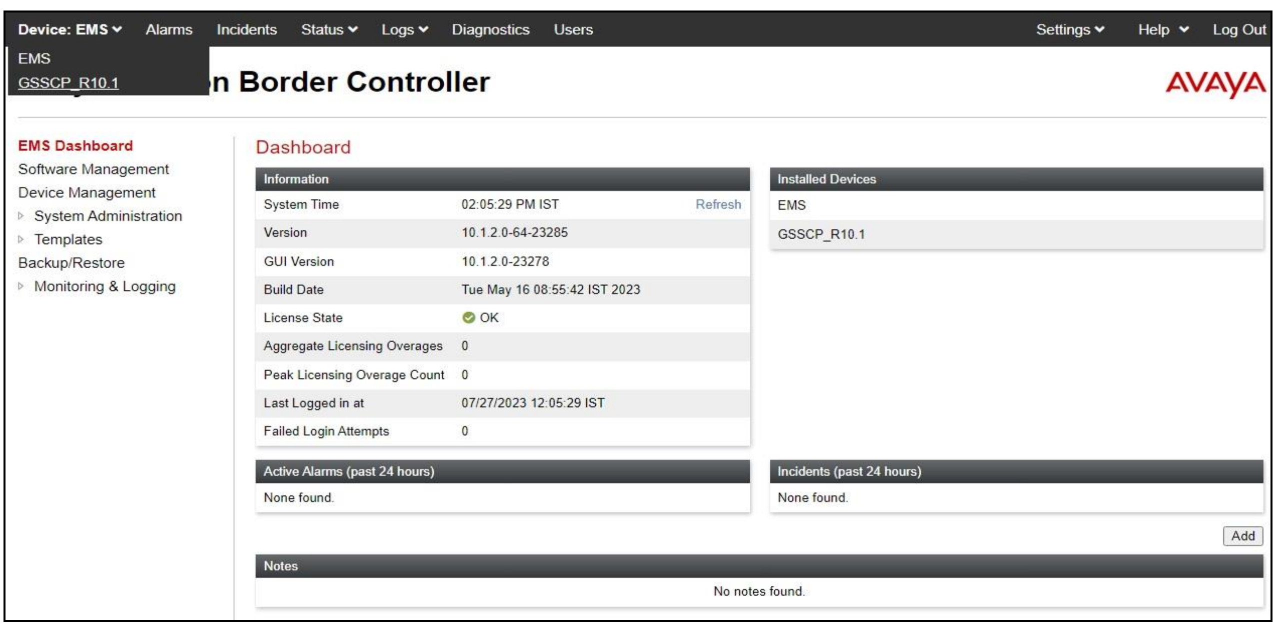
This section describes the configuration of the Session Border Controller (Avaya SBC). The Avaya SBC provides security and manipulation of signalling to deliver an interface to the Service Provider's SIP Trunk that is standard where possible and adapted to the Service Provider's SIP implementation where necessary.

8.1. Access Avaya Session Border Controller

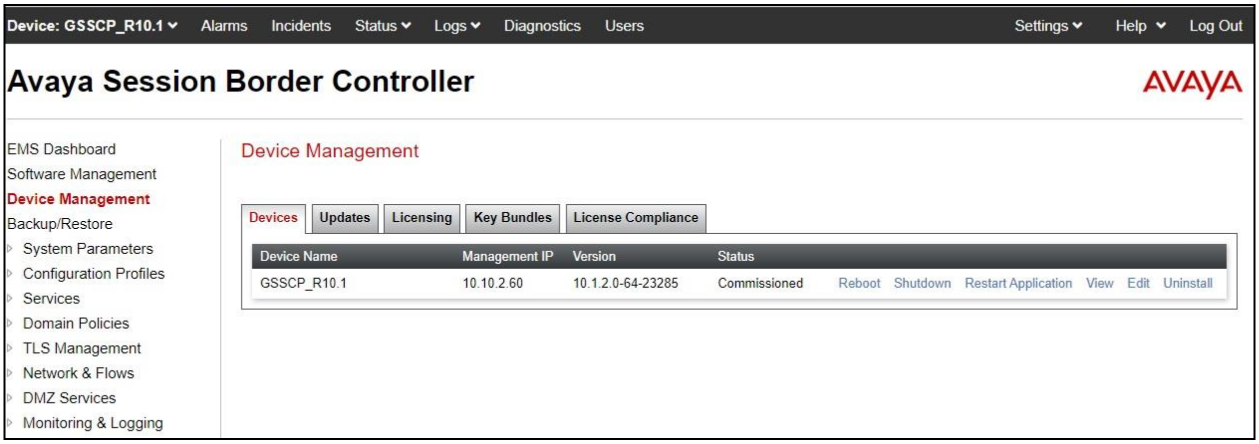
Access the Avaya SBC using a web browser by entering the URL **https://<ip-address>**, where **<ip-address>** is the management IP address configured at installation and enter the **Username** and **Password**.



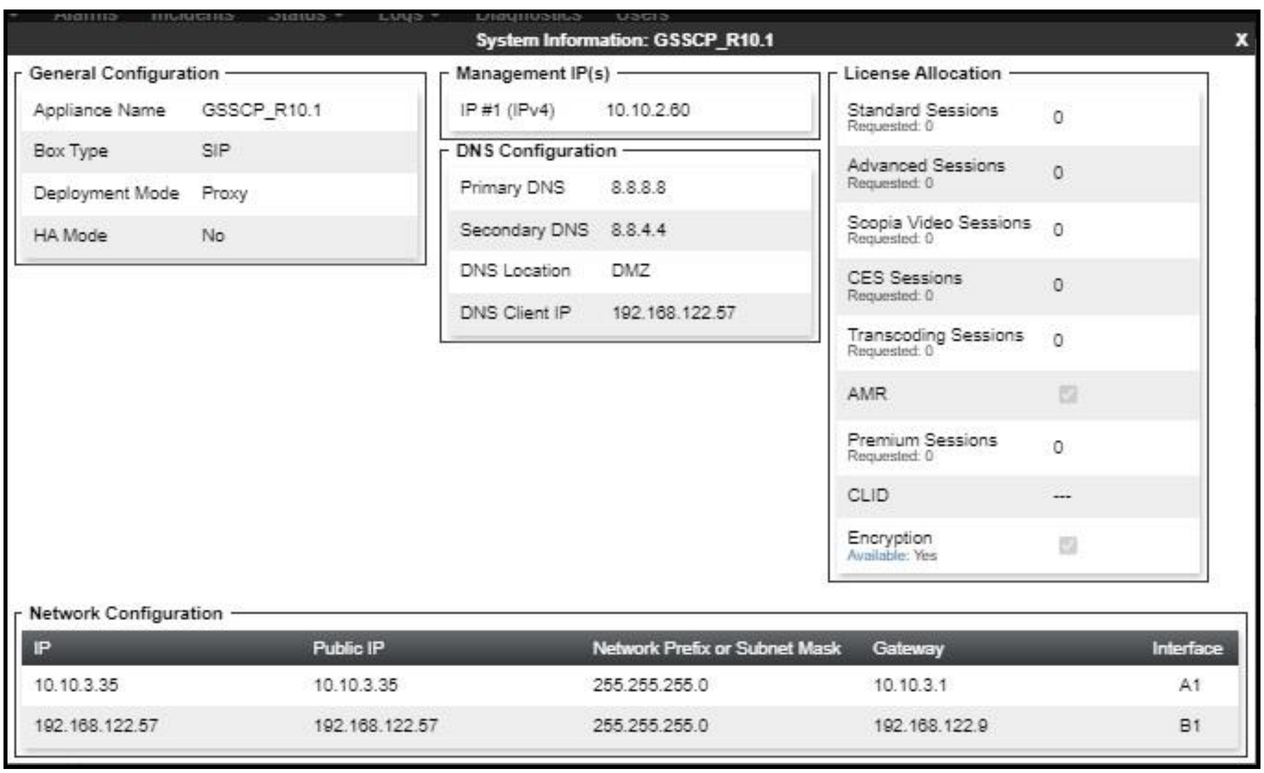
Once logged in, on the top-left of the screen, under **Device:** select the required device from the drop-down menu. with a menu on the left-hand side. In this case, **GSSCP_R10.1** is used as a starting point for all configuration of the Avaya SBC.



To view system information that was configured during installation, navigate to **Device Management**. A list of installed devices is shown in the right pane. In the case of the sample configuration, a single device named **GSSCP_R10.1** is shown. To view the configuration of this device, click **View** (the third option from the right).



The **System Information** screen shows the **General Configuration**, **Device Configuration**, **License Allocation**, **Network Configuration**, **DNS Configuration** and **Management IP** information.

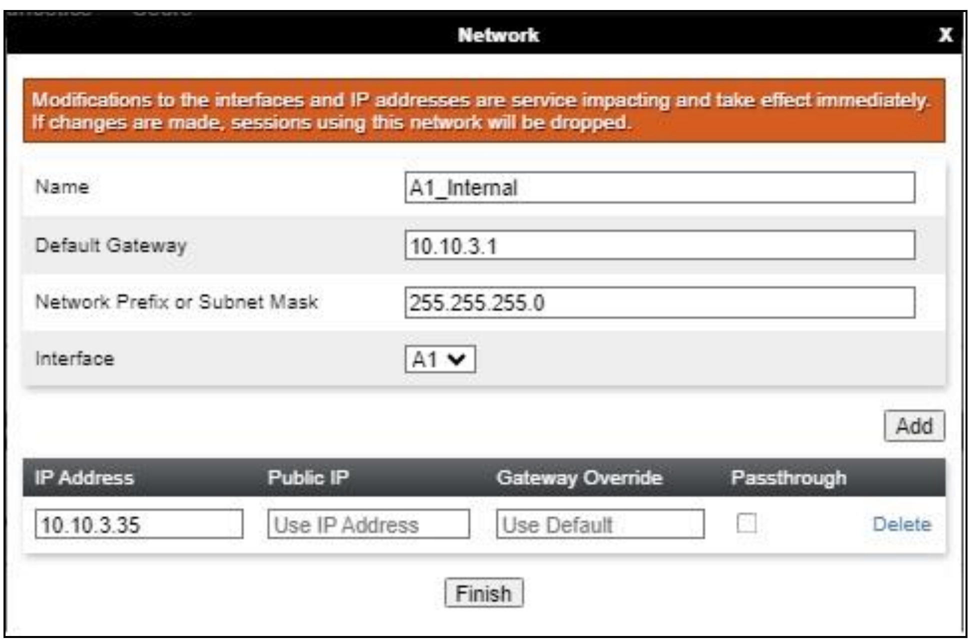


8.2. Define Network Management

Network information is required on the Avaya SBC to allocate IP addresses and masks to the interfaces. Note that only the **A1** and **B1** interfaces are used, typically the **A1** interface is used for the internal side and **B1** is used for external. Each side of the Avaya SBC can have only one physical interface assigned.

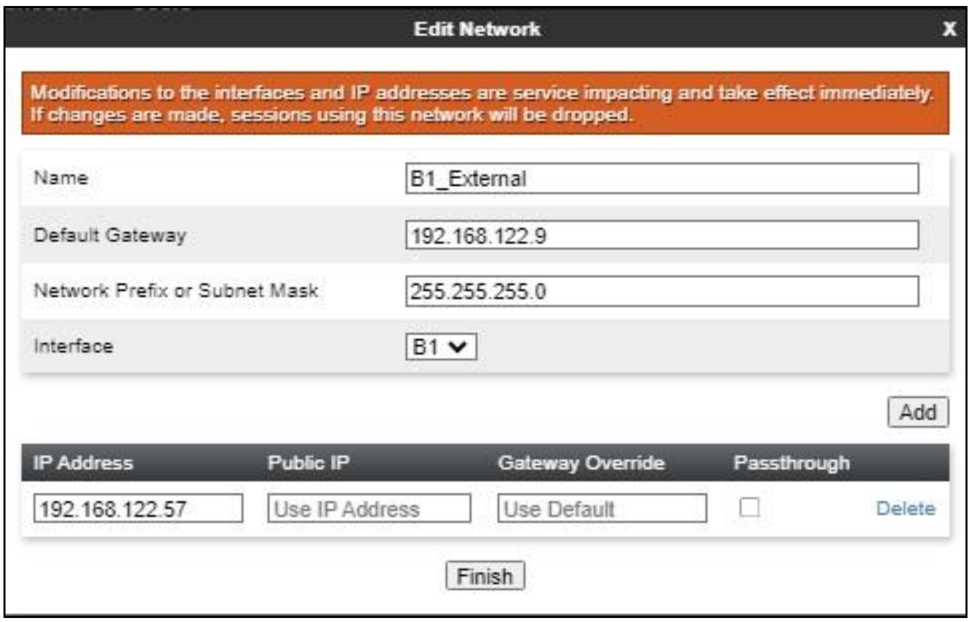
To define the network information, navigate to **Network & Flows → Network Management** in the main menu on the left-hand side and click on **Add**. Enter details for the external interfaces in the dialogue box:

- Enter a descriptive name in the **Name** field.
- Enter the default gateway IP address for the external interfaces in the **Default Gateway** field.
- Enter the subnet mask in the **Network Prefix or Subnet Mask** field.
- Select the external physical interface to be used from the **Interface** drop down menu. In the test environment, this was **B1**.
- Click on **Add** and an additional row will appear allowing an IP address to be entered.
- Enter the external IP address of the Avaya SBC on the SIP trunk in the **IP Address** field and leave the **Public IP** and **Gateway Override** fields blank.
- Click on **Finish** to complete the interface definition.

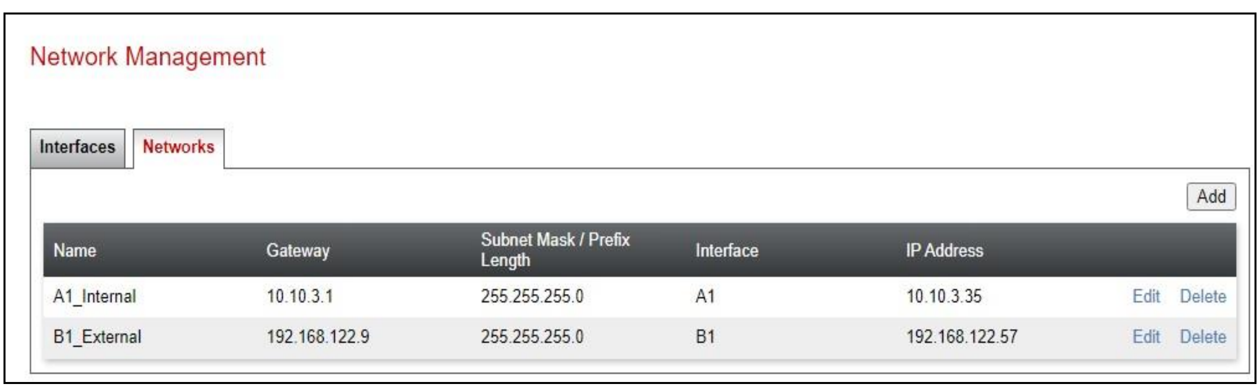


Click on **Add** to define the internal interfaces or Edit if it was defined during installation of the Avaya SBC. Enter details in the dialogue box:

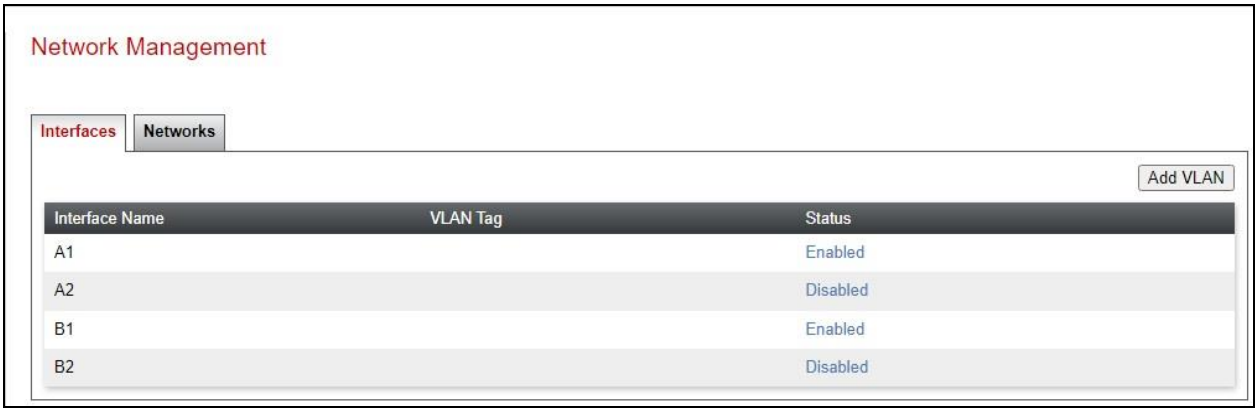
- Enter a descriptive name in the **Name** field.
- Enter the default gateway IP address for the internal interfaces in the **Default Gateway** field.
- Enter the subnet mask in the **Network Prefix or Subnet Mask** field.
- Select the internal physical interface to be used from the **Interface** drop down menu. In the test environment, this was **A1**.
- Click on **Add** and an additional row will appear allowing an IP address to be entered.
- Enter the internal IP address of the Avaya SBC on the SIP trunk in the **IP Address** field and leave the **Public IP** and **Gateway Override** fields blank.
- Click on **Finish** to complete the interface definition.



The following screenshot shows the completed Network Management configuration:



Select the **Interfaces** tab and click on the **Status** of the physical interface to toggle the state. Change the state to **Enabled** where required.



Note: to ensure that the Avaya SBC uses the interfaces defined, the Application must be restarted.

- Click on **Device Management** in the main menu (not shown).
- Select **Restart Application** indicated by an icon in the status bar (not shown).

A status box will appear that will indicate when the restart is complete.

8.3. Define TLS Profiles

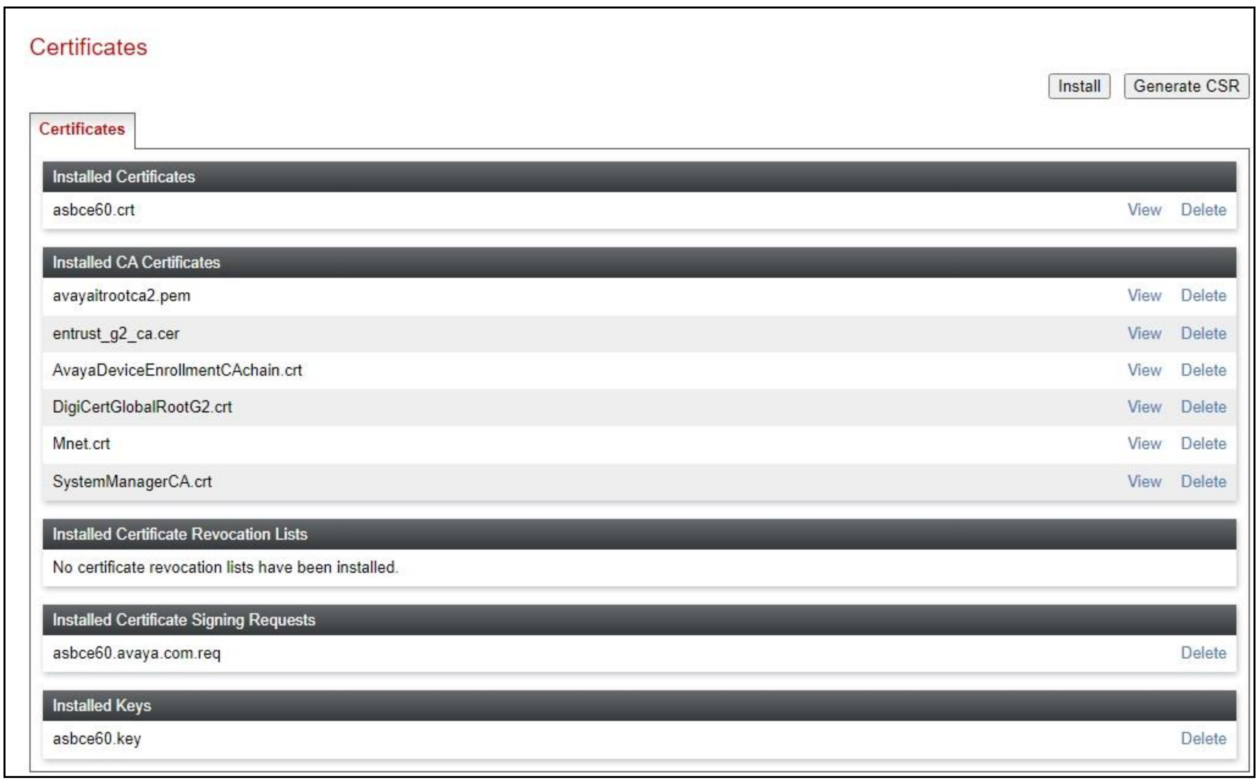
For the compliance test, TLS transport is used for signalling on the SIP trunk between Session Manager and the Avaya SBC. Compliance testing was done using identity certificates signed by a local certificate authority. The generation and installation of these certificates are beyond the scope of these Application Notes.

The following procedures show how to view the certificates and configure the Client and Server profiles to support the TLS connection.

8.3.1. Certificates

To view the certificates currently installed on the Avaya SBC, navigate to **TLS Management → Certificates**:

- Verify that an Avaya SBC identity certificate (**asbce60.crt**) is present under **Installed Certificates**.
- Verify that certificate authority root certificate (**SystemManagerCA.crt**) is present under **Installed CA certificates**.
- Verify that private key associated with the identity certificate (**asbce60.key**) is present under **Installed Keys**.



8.3.2. Client Profile

To create a new client profile, navigate to **TLS Management → Client Profile** in the left pane and click **Add** (not shown).

- Set **Profile Name** to a descriptive name. **GSSCP_Client** was used in the compliance testing.
- Set **Certificate** to the identity certificate **asbce60.crt** used in the compliance testing.
- **Peer Verification** is automatically set to **Required**.
- Set **Peer Certificate Authorities** to the **SystemManagerCA.pem** identity certificate.
- Set **Verification Depth** to **1**.

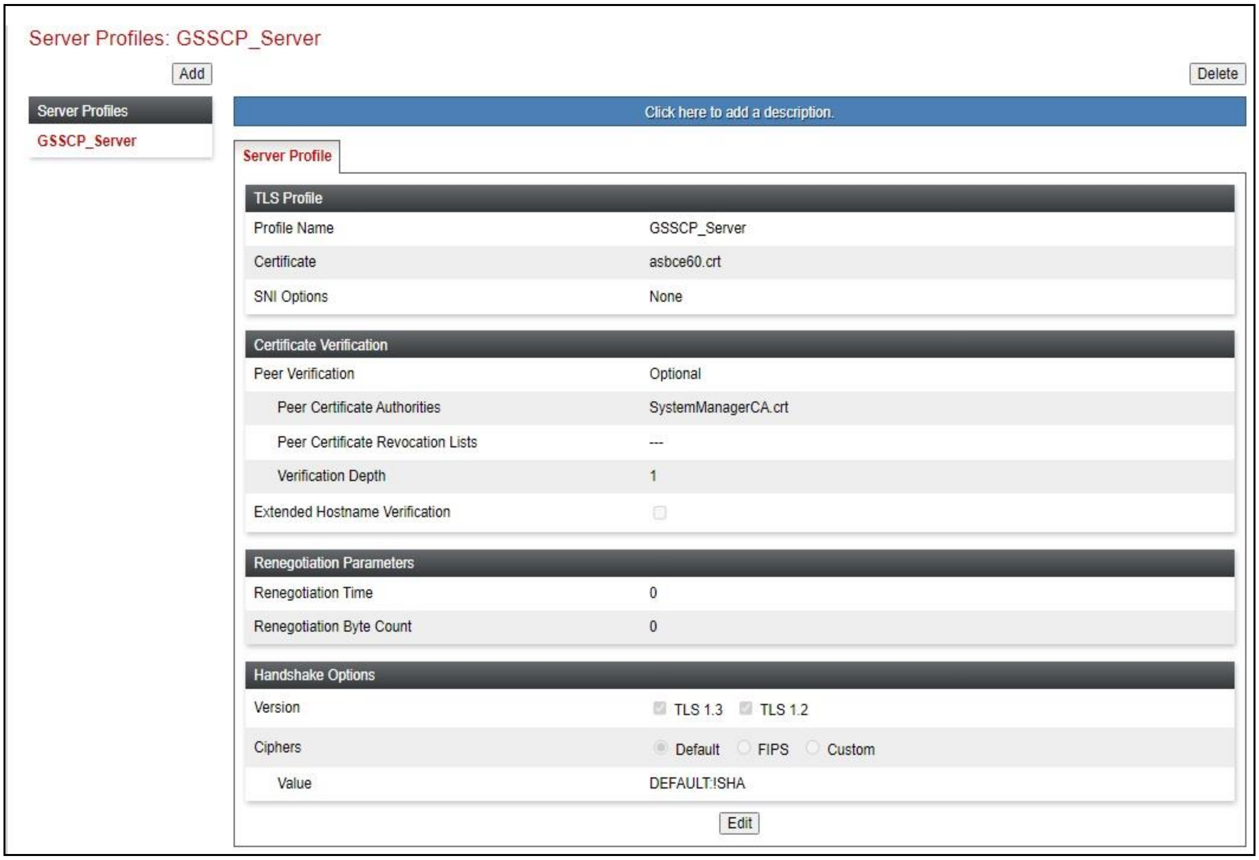
Click **Next** to accept default values for the next screen and click **Finish** (not shown).

8.3.3. Server Profile

To create a new server profile, navigate to **TLS Management → Server Profile** in the left pane and click **Add** (not shown).

- Set **Profile Name** to a descriptive name. **GSSCP_Server** was used in the compliance testing
- Set **Certificate** to the identity certificate **asbce60.crt** used in the compliance testing.
- Set **Peer Verification** to **Optional**.

Click **Next** to accept default values for the next screen and click **Finish** (not shown).



8.4. Define Interfaces

When the IP addresses and masks are assigned to the interfaces, these are then configured as signalling and media interfaces.

8.4.1. Signalling Interfaces

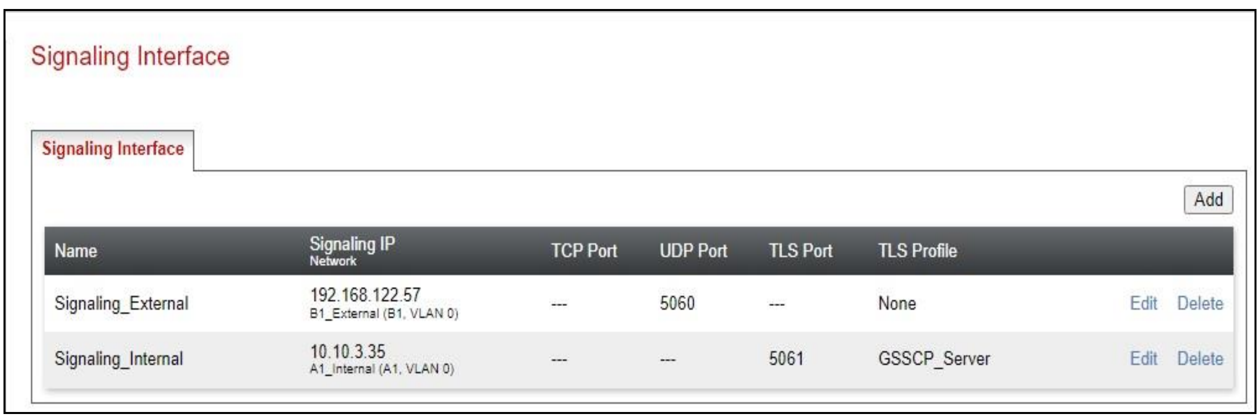
To define the signalling interfaces on the Avaya SBC, navigate to **Network & Flows → Signaling Interface** from the menu on the left-hand side. Details of transport protocol and ports for the internal and external SIP signalling are entered here.

To enter details of transport protocol and ports for the SIP signalling on the internal interface:

- Select **Add** and enter details of the internal signalling interface in the pop-up menu (not shown).
- In the **Name** field enter a descriptive name for the interface.
- For **Signaling IP**, select the **A1_Internal** signalling interface IP addresses defined in **Section 8.2**.
- Select **TLS** port number, **5061** is used for Session Manager.
- Select a **TLS Profile** defined in **Section 8.3.3** from the drop-down menu.
- Click **Finish**.

To enter details of transport protocol and ports for the SIP signalling on the external interface:

- Select **Add** and enter details of the external signalling interface in the pop-up menu (not shown).
- In the **Name** field enter a descriptive name for the external signalling interface.
- For **Signaling IP**, select the **B1_external** signalling interface IP address defined in **Section 8.2**.
- Select **UDP** port number, **5060** is used for the VTX SIP Trunk.
- Click **Finish**.


Signaling Interface

Signaling Interface

Add

Name	Signaling IP Network	TCP Port	UDP Port	TLS Port	TLS Profile		
Signaling_External	192.168.122.57 B1_External (B1, VLAN 0)	---	5060	---	None	Edit	Delete
Signaling_Internal	10.10.3.35 A1_Internal (A1, VLAN 0)	---	---	5061	GSSCP_Server	Edit	Delete

8.4.2. Media Interfaces

To define the media interfaces on the Avaya SBC, navigate to **Network & Flows → Media Interface** from the menu on the left-hand side. Details of the RTP and SRTP port ranges for the internal and external media streams are entered here. The IP addresses for media can be the same as those used for signalling.

To enter details of the media IP and RTP port range for the internal interface to be used in the server flow:

- Select **Add Media Interface** and enter details in the pop-up menu.
- In the **Name** field enter a descriptive name for the internal media interface.
- For **Media IP**, select the **A1_Internal** media interface IP address defined in **Section 8.2**.
- For **Port Range**, enter **35000-40000**.
- Click **Finish**.

To enter details of the media IP and RTP port range on the external interface to be used in the server flow.

- Select **Add Media Interface** and enter details in the pop-up menu.
- In the **Name** field enter a descriptive name for the external media interface.
- For **Media IP**, select the **B1_External** media interface IP address defined in **Section 8.2**.
- Select **Port Range**, enter **35000-40000**.
- Click **Finish**.


Media Interface

Media Interface

Add

Name	Media IP Network	Port Range		
Media_Internal	10.10.3.35 A1_Internal (A1, VLAN 0)	35000 - 40000	Edit	Delete
Media_External	192.168.122.57 B1_External (B1, VLAN 0)	35000 - 40000	Edit	Delete

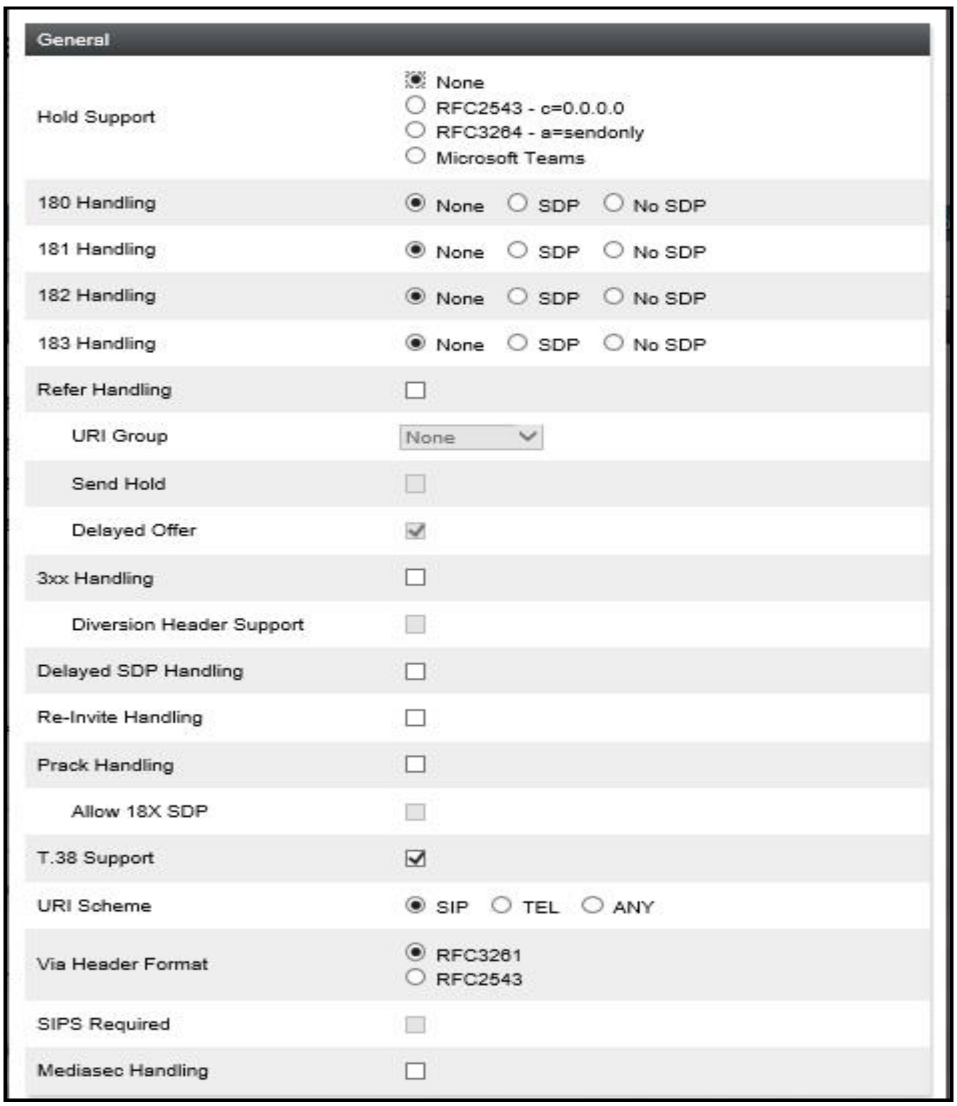
8.5. Define Server Interworking

Server interworking is defined for each server connected to the Avaya SBC. In this case, VTX is connected as the Trunk Server and Session Manager is connected as the Call Server.

8.5.1. Server Interworking Avaya

Server Interworking allows the configuration and management of various SIP call server-specific capabilities such as call hold and T.38. From the left-hand menu select **Configuration Profiles → Server Interworking** and click on **Add**.

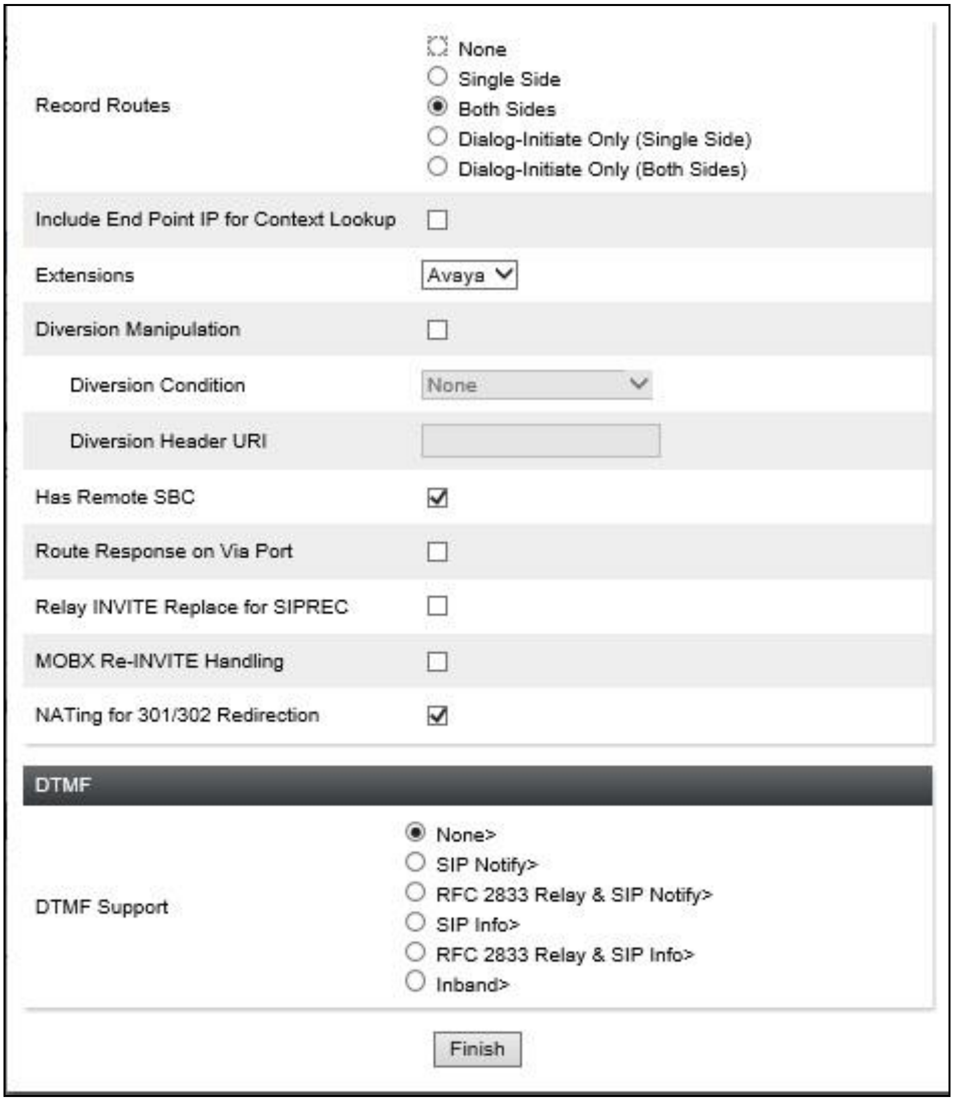
- Enter profile name such as Avaya and click **Next** (Not Shown).
- Check **Hold Support = None**.
- Check **T.38 Support**.
- Uncheck **SIPS Required**.
- All other options on the **General** Tab can be left at default.



On the **Advanced** Tab:

- Check **Record Routes** = **Both Sides**.
- Ensure **Extensions** = **Avaya**.
- Check **Has Remote SBC**.
- All other options on the **Advanced** Tab can be left at default.

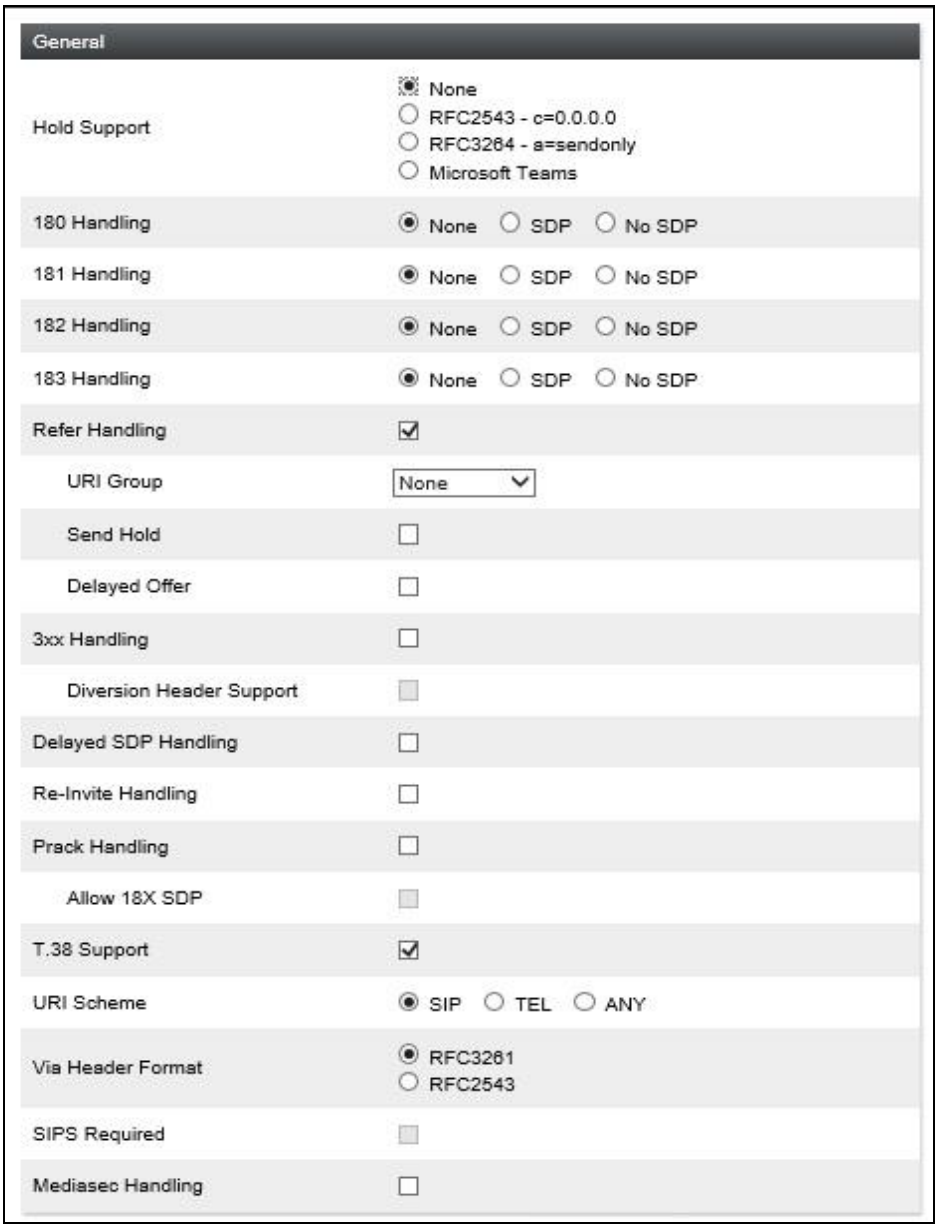
Click **Finish**.



8.5.2. Server Interworking – VTX

Server Interworking allows the configuration and management of various SIP call server-specific capabilities such as call hold and T.38. From the left-hand menu select **Configuration Profiles → Server Interworking** and click on **Add**.

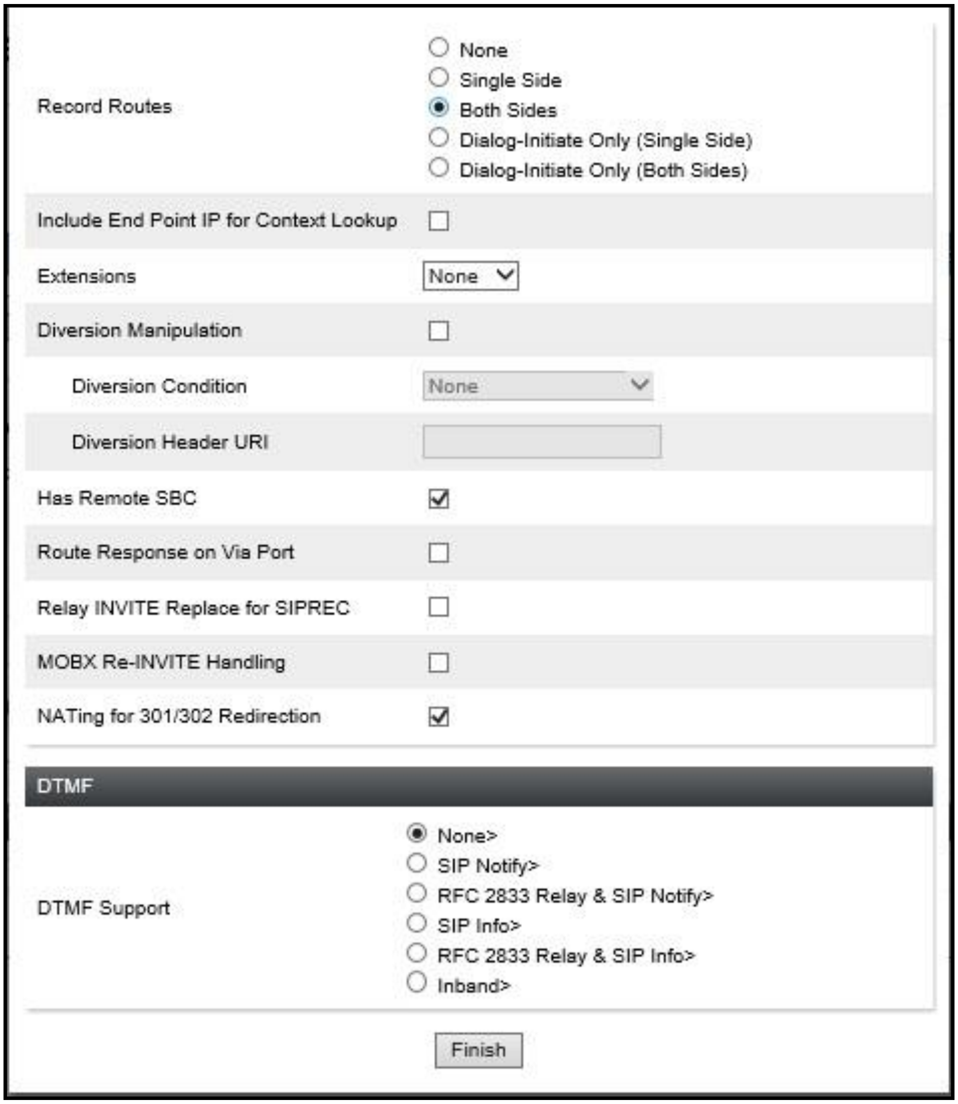
- Enter profile name such as **VTX** and click **Next** (Not Shown).
- Check **Hold Support = None**.
- Check **Refer Handling** as per **Section 2.2**.
- Check **T.38 Support**.
- Uncheck **SIPS Required**.
- All other options on the **General** Tab can be left at default.



On the **Advanced** Tab:

- Check **Record Routes** = **Both Sides**.
- Ensure **Extensions** = **None**.
- Check **Has Remote SBC**.
- All other options on the **Advanced** Tab can be left at default.

Click **Finish**.



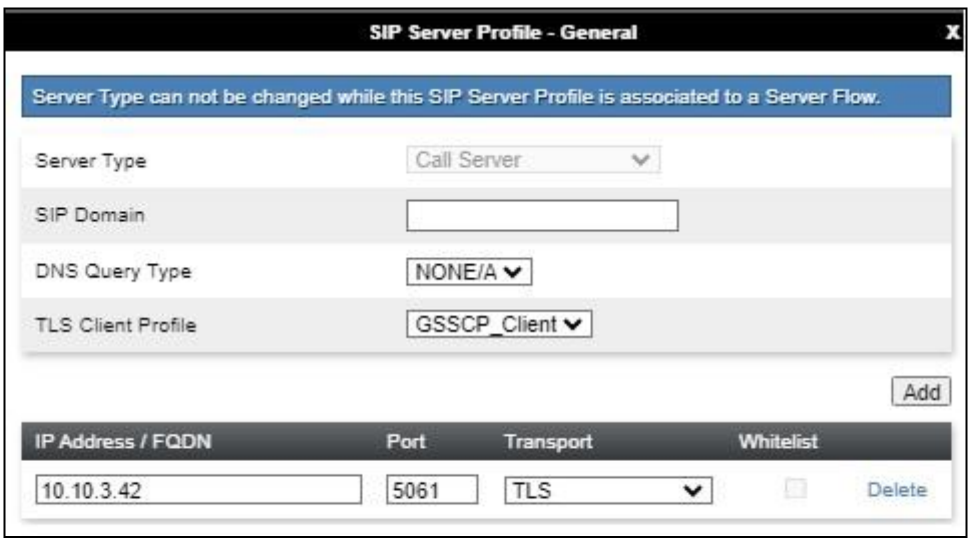
8.6. Define Servers

Servers are defined for each server connected to the Avaya SBC. In this case, VTX is connected as the Trunk Server and Session Manager is connected as the Call Server.

8.6.1. Server Configuration – Avaya

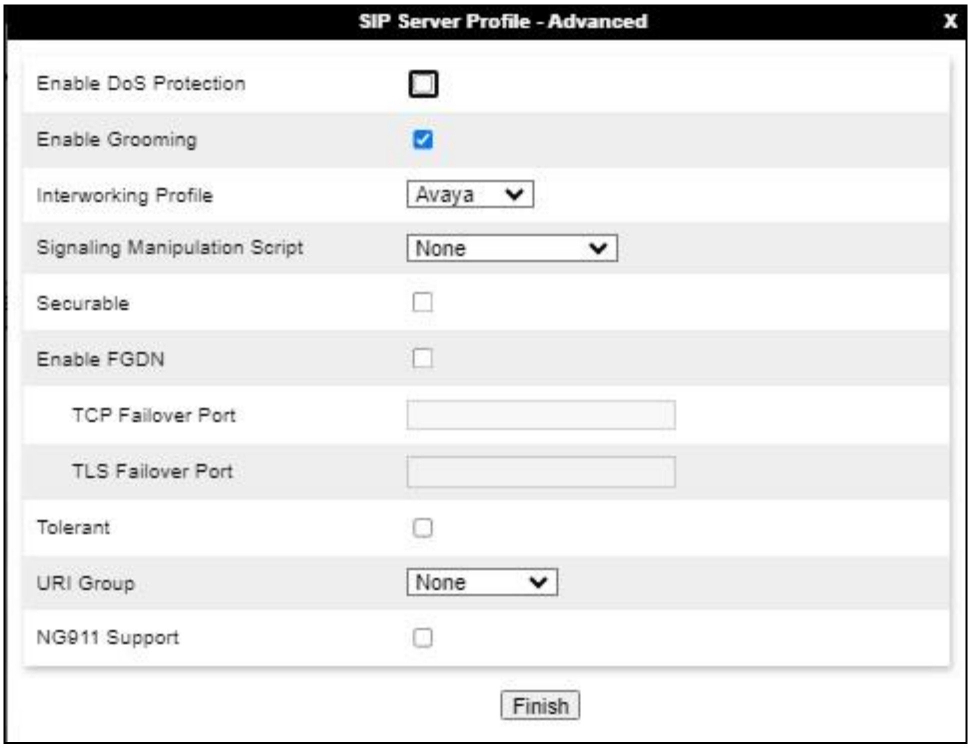
From the left-hand menu select **Services → SIP Servers** and click on **Add** and enter a descriptive name. On the **Add Server Configuration Profiles** tab, set the following:

- Select **Server Type** to be **Call Server**.
- Select **TLS Client Profile** to be **GSSCP_Client** as defined in **Section 8.3.2**.
- Enter **IP Address / FQDN** to **10.10.3.42** (Session Manager IP Address).
- For **Port**, enter **5061**.
- For **Transport,** select **TLS**.
- Click on **Next** (not shown) to use default entries on the **Authentication** and **Heartbeat** tabs.



On the **Advanced** tab:

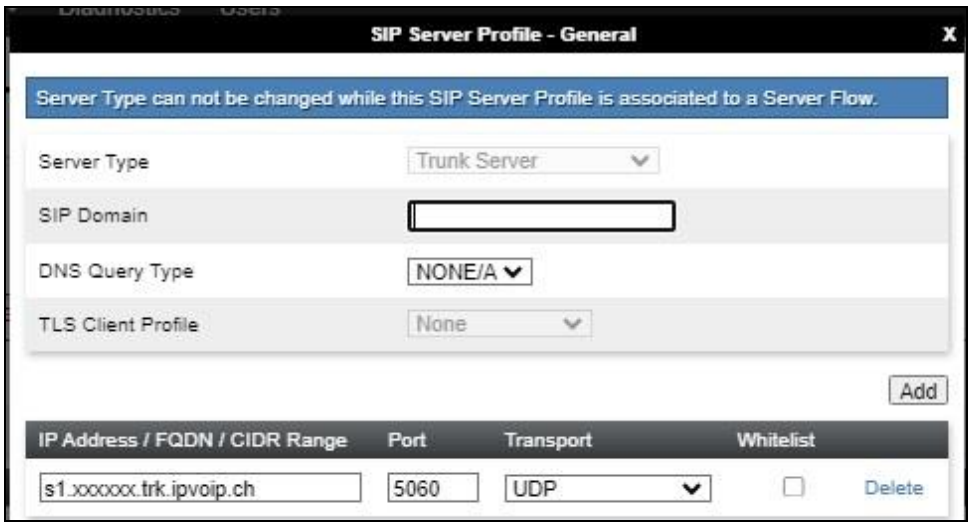
- Check **Enable Grooming**.
- Select **Avaya** for **Interworking Profile**.
- Click **Finish**.



8.6.2. Server Configuration – VTX

To define the VTX Trunk Server, navigate to **Services → SIP Servers** and click on **Add** and enter a descriptive name. On the **Add Server Configuration Profile** tab, set the following:

- Select **Server Type** to be **Trunk Server**.
- Enter **IP Address / FQDN** to **s1.xxxxxx.trk.ipvoip.ch** (VTX SIP Platform).
- For **Port**, enter **5060**.
- For **Transport,** select **UDP**.
- Click on **Next** (not shown).



In the new Authentication window that appears, enter the following values as VTX require authentication to connect to their network:

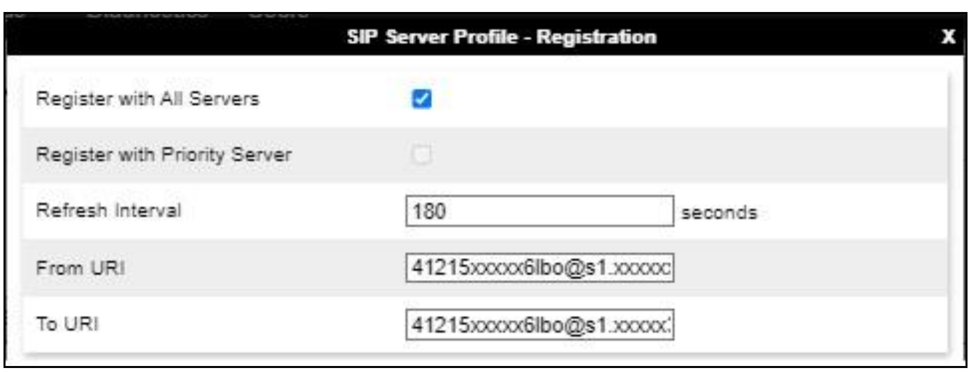
- **Enabled Authentication:** Checked
- **User Name:** Enter username provided by the Service Provider.
- **Realm:** Enter realm details provided by the Service Provider or leave blank to be detected by the server challenge.
- **Password** Enter password provided by the Service Provider.
- **Confirm Password** Re-enter password provided by the Service Provider.

Click **Next** to continue (not shown).

In the new Registration window that appears, enter the following values.

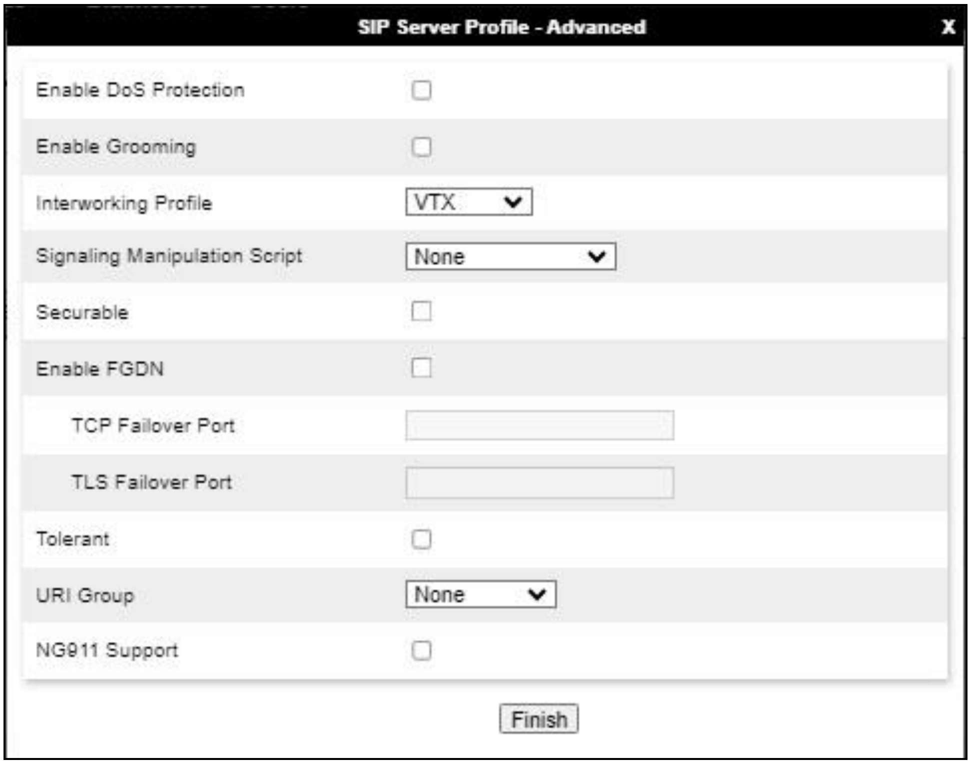
- **Register with Priority Server:** Check.
- **Refresh Interval** Choose the desired frequency in seconds the Avaya SBC will send SIP REGISTERS.
- **From URI:** Enter an URI to be sent in the FROM header for SIP REGISTERS.
- **TO URI:** Enter an URI to be sent in the TO header for SIP REGISTERS.

Click **Next** to continue (not shown).



On the Advanced tab:

- Select **VTX** for **Interworking Profile**.
- Click **Finish**.



8.7. Routing

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

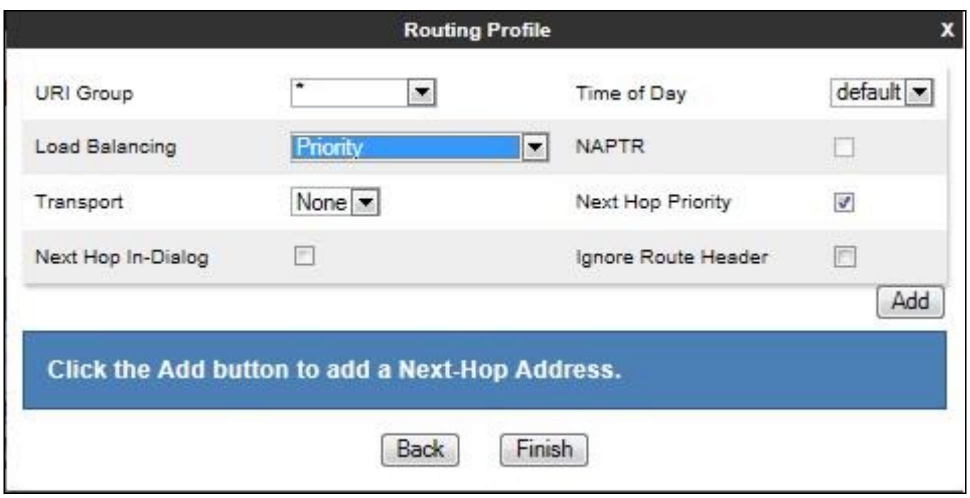
Routing information is required for routing to Session Manager on the internal side and VTX address on the external side. The IP addresses and ports defined here will be used as the destination addresses for signalling. If no port is specified in the **Next Hop IP Address**, default 5060 is used.

8.7.1. Routing – Avaya

Create a Routing Profile for Session Manager.

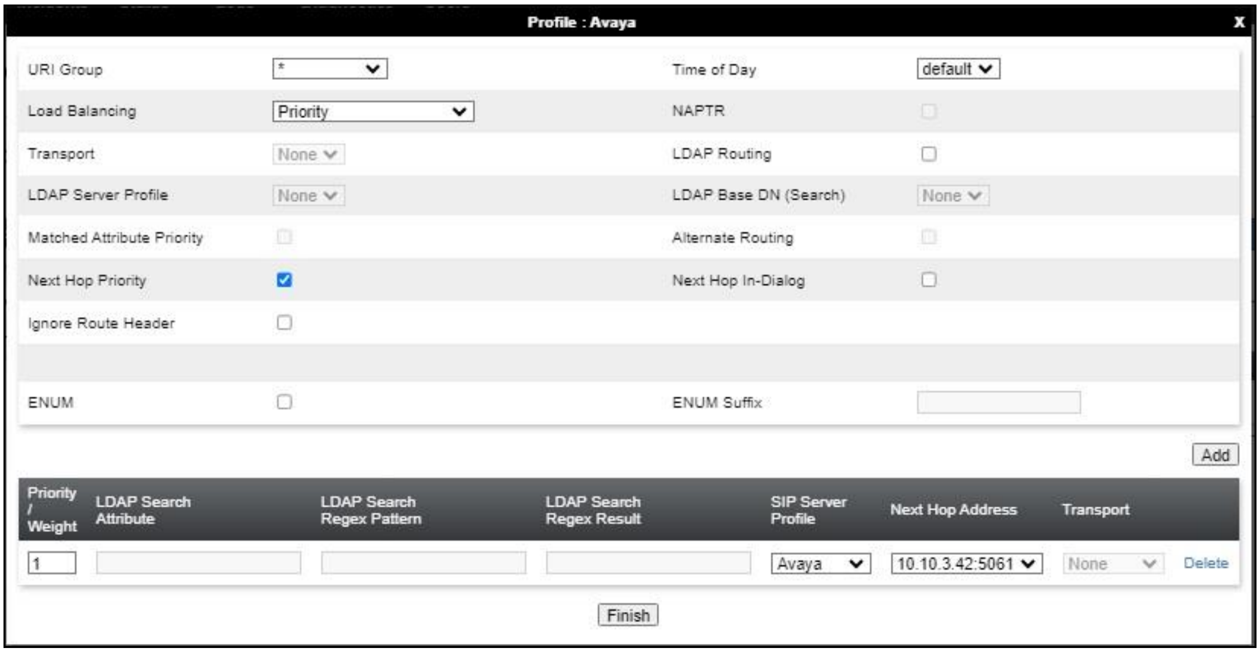
- Navigate to **Configuration Profiles → Routing** and select **Add Profile**.
- Enter a **Profile Name** and click **Next**.

The Routing Profile window will open. Use the default values displayed and click **Add**.



On the **Next Hop Address** window, set the following:

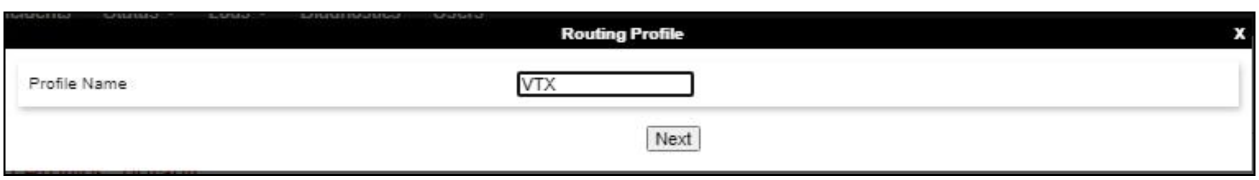
- **Priority/Weight** = **1**.
- **SIP Server Profile** = **Avaya** (**Section 8.6.1**) from drop down menu.
- **Next Hop Address** = Select **10.10.3.42:5061 (TLS)** from drop down menu.
- Click **Finish**.



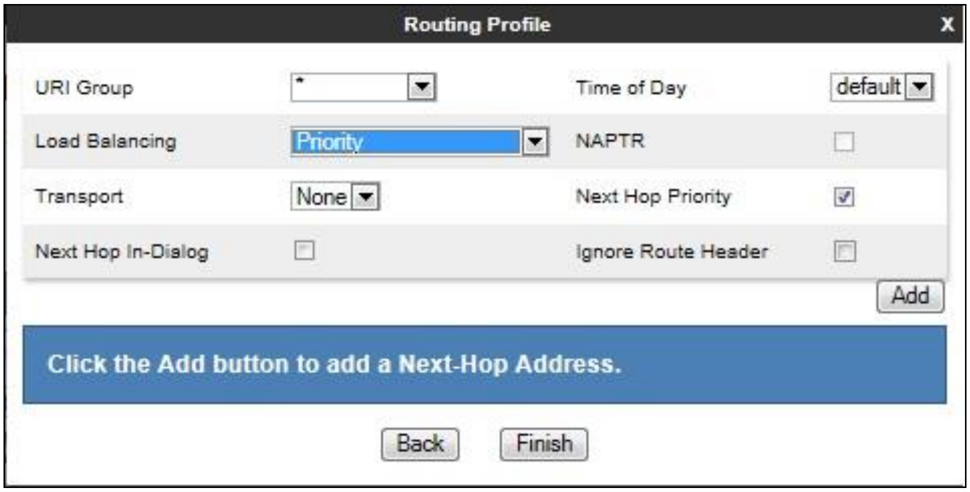
8.7.2. Routing – VTX

Create a Routing Profile for VTX SIP network.

- Navigate to **Configuration Profiles → Routing** and select **Add Profile**.
- Enter a **Profile Name** and click **Next**.

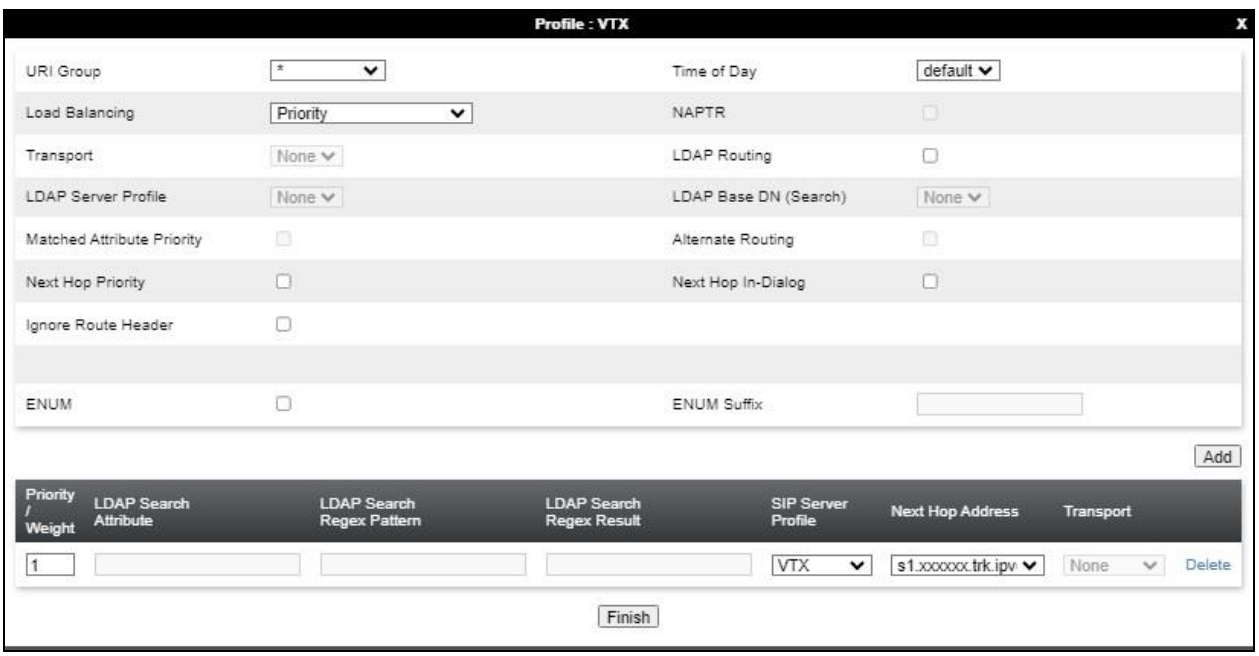


The Routing Profile window will open. Use the default values displayed and click **Add**.



On the **Next Hop Address** window, set the following:

- **Priority/Weight** = **1**.
- **SIP Server Profile** = **VTX** (**Section 8.6.2**) from drop down menu.
- **Next Hop Address** = Select **s1.xxxxxx.trk.ipvoip.ch (UDP)** from drop down menu.
- Click **Finish**.

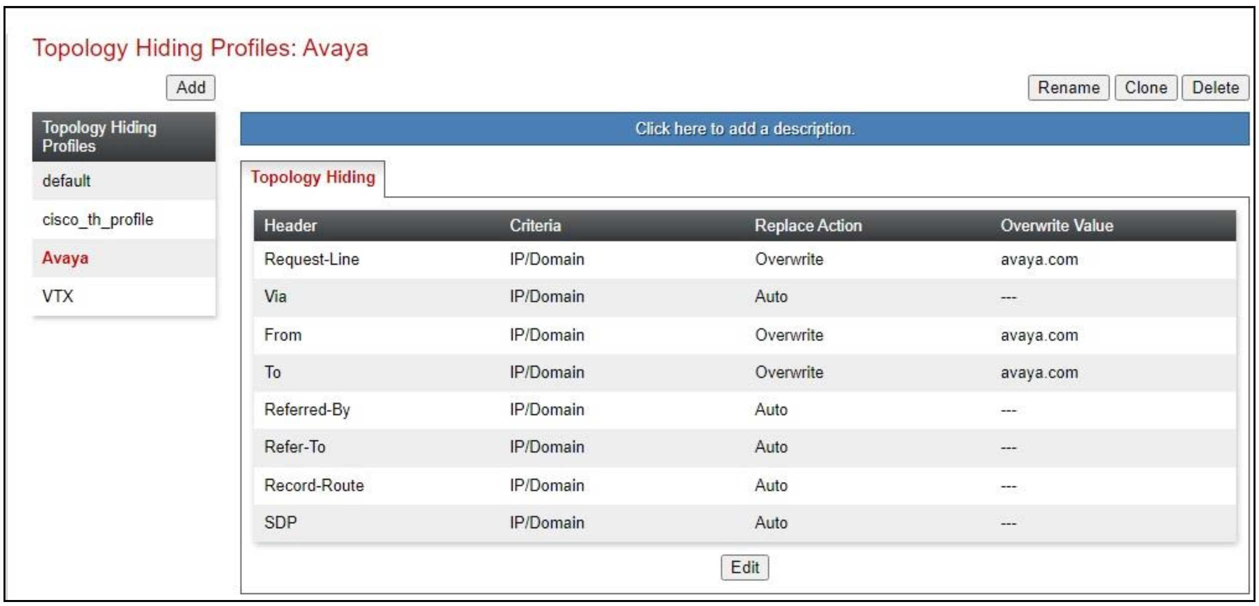


8.8. Topology Hiding

Topology hiding is used to hide local information such as private IP addresses and local domain names. The local information can be overwritten with a domain name or IP addresses. The default **Replace Action** is **Auto**, this replaces local information with IP addresses, generally the next hop. Topology hiding has the advantage of presenting single Via and Record-Route headers externally where multiple headers may be received from the enterprise. In some cases where Topology Hiding cannot be applied, in particular the Contact header, IP addresses are translated to the Avaya SBC external addresses using NAT.

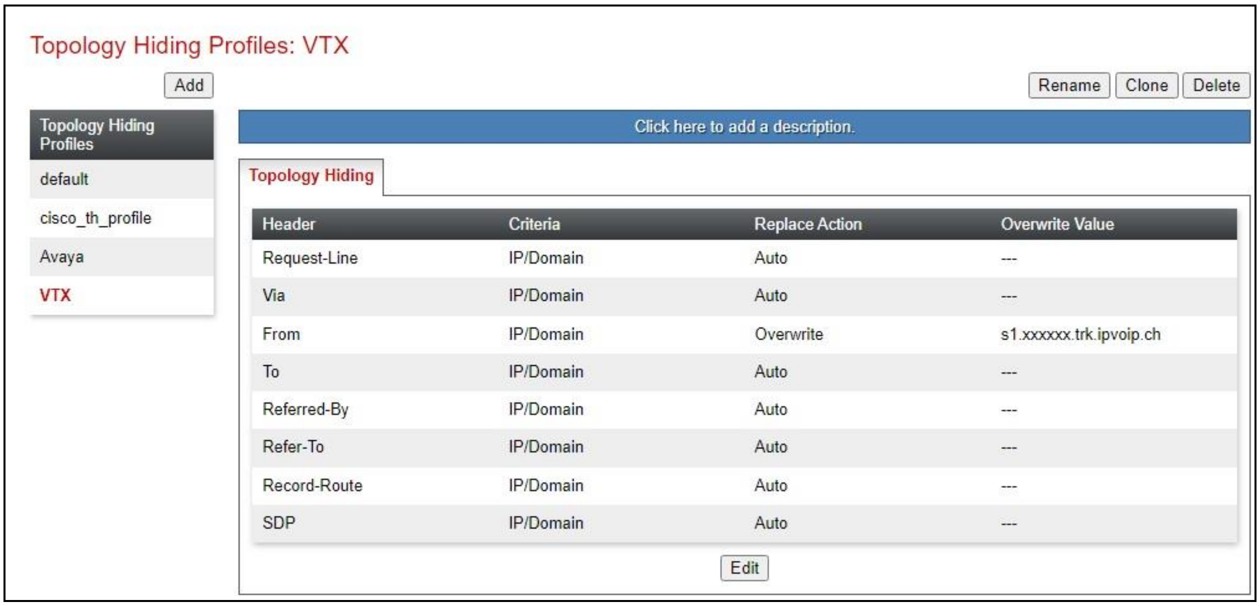
To define Topology Hiding for Session Manager, navigate to **Configuration Profiles → Topology Hiding** from menu on the left-hand side. Click on **Add** and enter details in the **Topology Hiding Profile** pop-up menu (not shown).

- Enter a descriptive Profile Name such as **Avaya**.
- If the required Header is not shown, click on **Add Header**.
- Under the **Header** field for **To**, **From** and **Request Line**, select **IP/Domain** under **Criteria** and **Overwrite** under **Replace Action**. For Overwrite value, insert **avaya.com**.
- Click **Finish** (not shown).



To define Topology Hiding for VTX, navigate to **Configuration Profiles → Topology Hiding** from the menu on the left-hand side. Click on **Add** and enter details in the **Topology Hiding Profile** pop-up menu (not shown).

- In the **Profile Name** field enter a descriptive name for VTX and click **Next**.
- If the required Header is not shown, click on **Add Header**.
- Under the **Header** field for, **From**, select **IP/Domain** under **Criteria** and **Overwrite** under **Replace Action**. For Overwrite value, insert **s1.xxxxxx.trk.ipvoip.ch**.
- Click **Finish** (not shown).



Header	Criteria	Replace Action	Overwrite Value
Request-Line	IP/Domain	Auto	---
Via	IP/Domain	Auto	---
From	IP/Domain	Overwrite	s1.xxxxxx.trk.ipvoip.ch
To	IP/Domain	Auto	---
Referred-By	IP/Domain	Auto	---
Refer-To	IP/Domain	Auto	---
Record-Route	IP/Domain	Auto	---
SDP	IP/Domain	Auto	---

8.9. 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.

In the reference configuration, only new Media and Signalling rules were defined. All other rules under Domain Policies, linked together on End Point Policy Groups later in this section, used one of the default sets already pre-defined in the configuration. Please note that changes should not be made to any of the defaults. If changes are needed, it is recommended to create a new rule by cloning one the defaults and then make the necessary changes to the new rule.

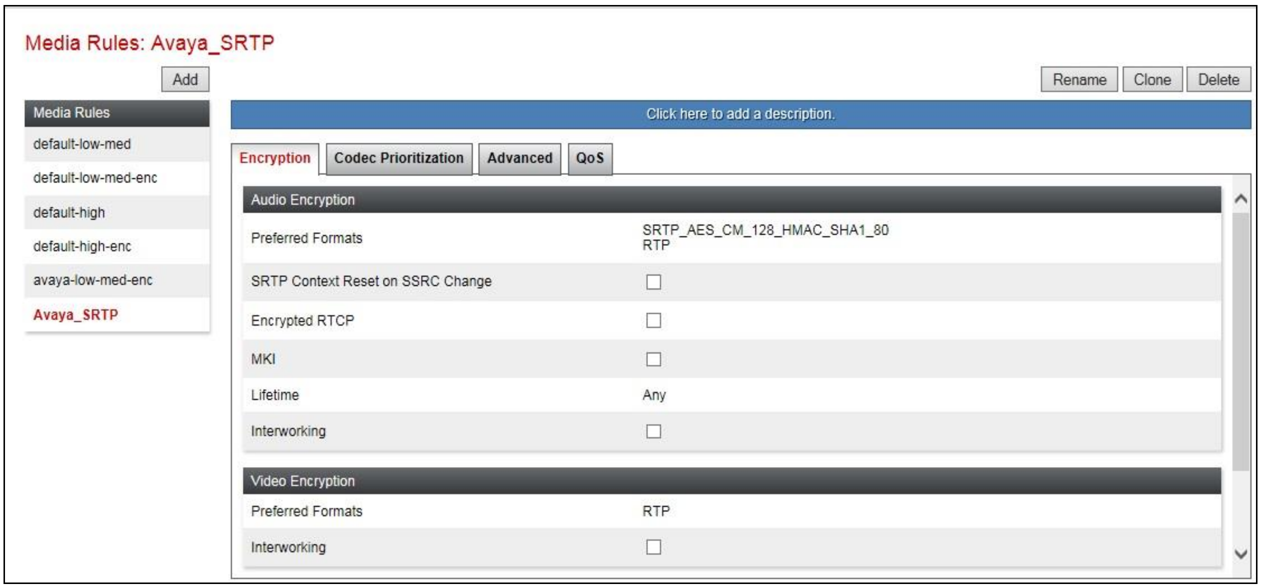
8.9.1. Media Rules

A media rule defines the processing to be applied to the selected media. For the compliance test, media rules were created for Session Manager to SRTP and VTX to use RTP.

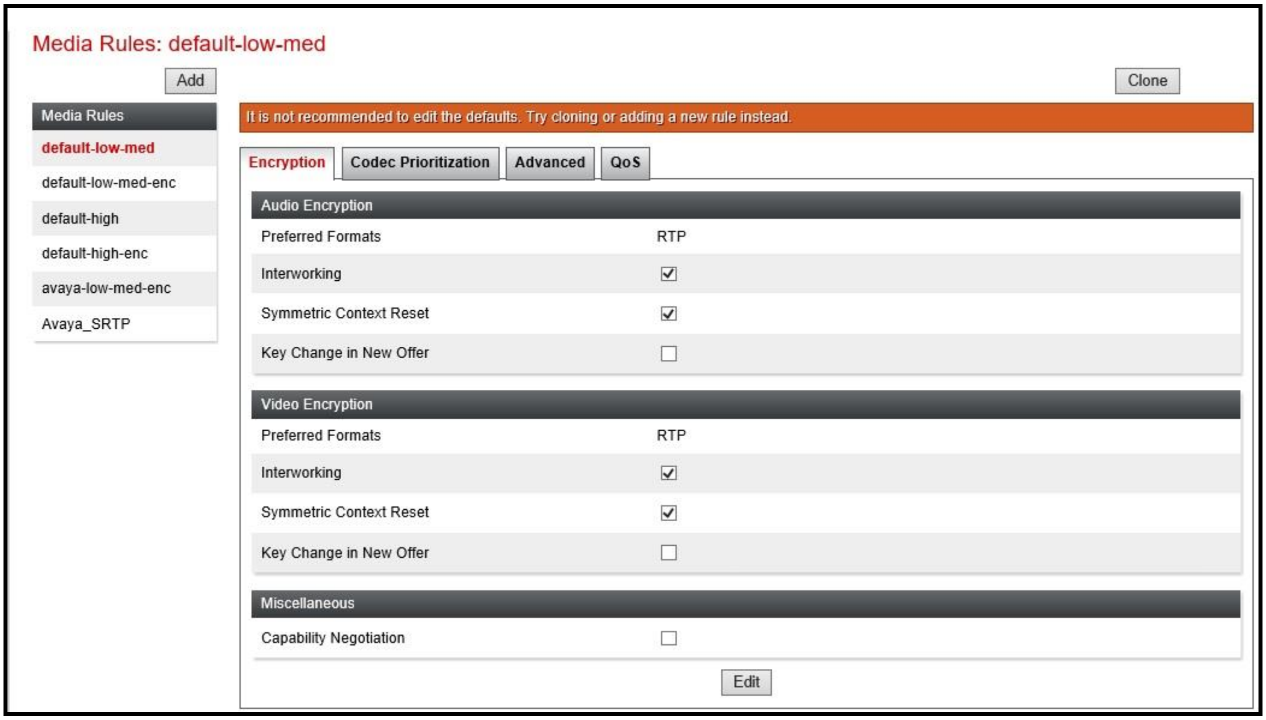
To define the Media Rule for Session Manager, navigate to **Domain Policies →Media Rules** in the main menu on the left-hand side. Click on **Add** and enter details in the Media Rule pop-up box (not shown)

- In the **Rule Name** field enter a descriptive name such as **Avaya_SRTP**.
- Set **Preferred Format #1** to **SRTP_AES_CM_128_HMAC_SHA1_80**.
- Set **Preferred Format #2** to **RTP**.
- Uncheck **Encrypted RTCP**.
- Check **Capability Negotiation** under **Miscellaneous** (not shown).

Default values were used for all other fields. Click **Finish** (not shown).



For the compliance test, the default media rule **default-low-med** was used for VTX.



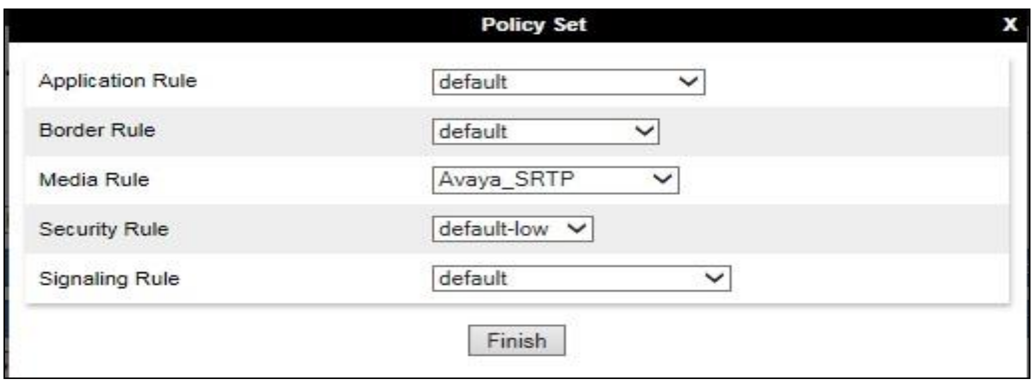
8.10. End Point Policy Groups

An end point policy group is a set of policies that will be applied to traffic between the Avaya SBC and a signaling endpoint (connected server). Thus, one end point policy group must be created for Session Manager and another for the VTX SIP trunk. The end point policy group is applied to the traffic as part of the end point flow defined in **Section 8.11**.

8.10.1. End Point Policy Group – Session Manager

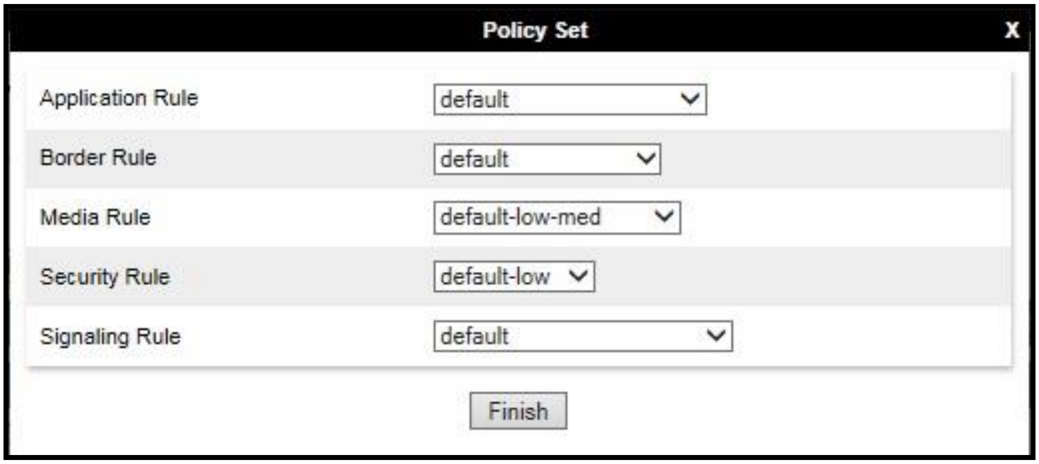
To define an End Point policy for Session Manager, navigate to **Domain Policies → End Point Policy Groups** in the main menu on the left-hand side. Click on **Add** and enter details in the Policy Group pop-up box (not shown).

- In the **Group Name** field enter a descriptive name, in this case **Avaya**, and click **Next** (not shown).
- Leave the **Application Rule**, **Border Rule**, **Security Rule** and **Signalling Rule** fields at their default values.
- In the **Media Rule** drop down menu, select the recently added Media Rule called **Avaya_SRTP**.
- Click **Finish**.



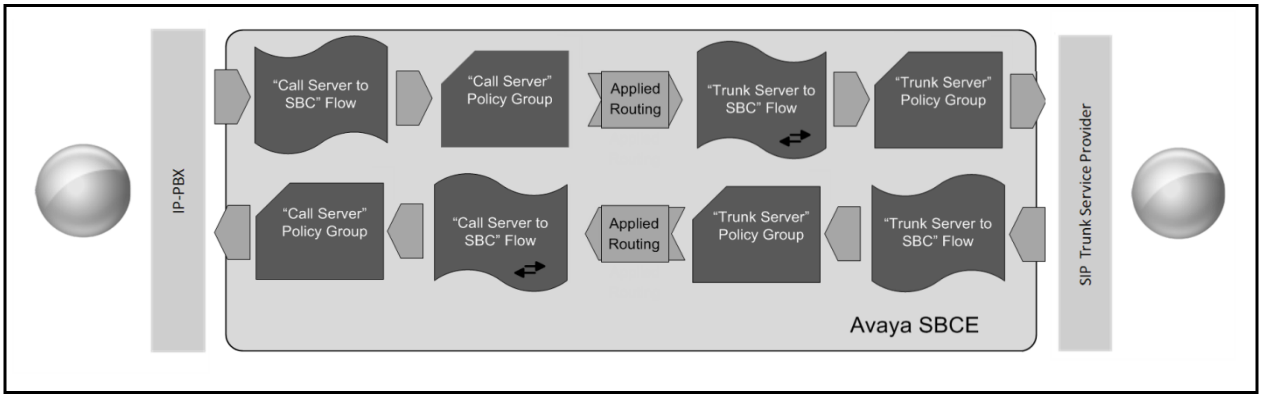
8.10.2. End Point Policy Group – VTX

For the compliance test, the predefined End Point Policy **default-low** was used for the VTX End Point Policy Group.

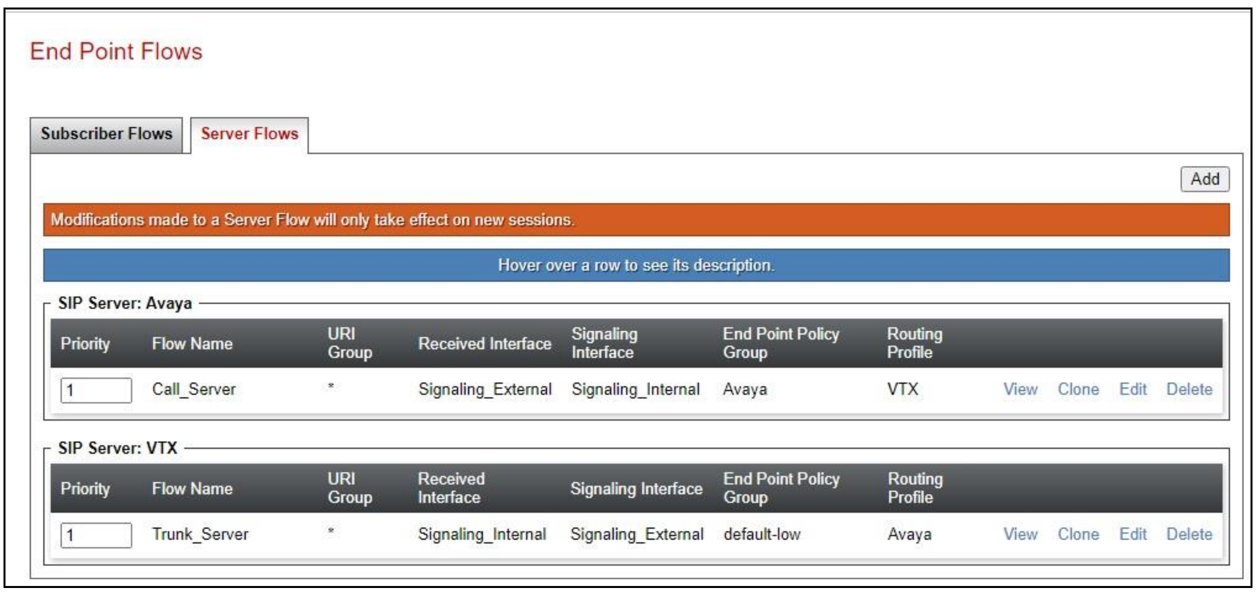


8.11. Server Flows

Server Flows combine the previously defined profiles into outgoing flows from Session Manager to VTX's SIP Trunk and incoming flows from VTX's SIP Trunk to Session Manager. The following screen illustrates the flow through the Avaya SBC to secure a SIP Trunk call.



This configuration ties all the previously entered information together so that calls can be routed from Session Manager to VTX SIP Trunk and vice versa. The following screenshot shows all configured flows.



End Point Flows

Subscriber Flows | Server Flows

Add

Modifications made to a Server Flow will only take effect on new sessions.

Hover over a row to see its description.

SIP Server: Avaya

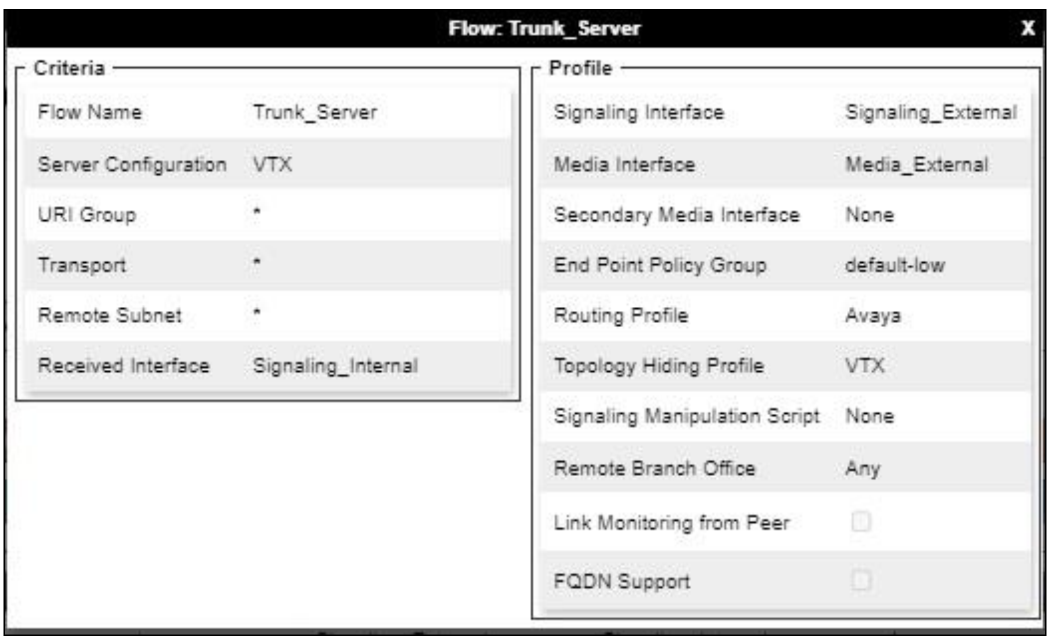
Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile				
1	Call_Server	*	Signaling_External	Signaling_Internal	Avaya	VTX	View	Clone	Edit	Delete

SIP Server: VTX

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile				
1	Trunk_Server	*	Signaling_Internal	Signaling_External	default-low	Avaya	View	Clone	Edit	Delete

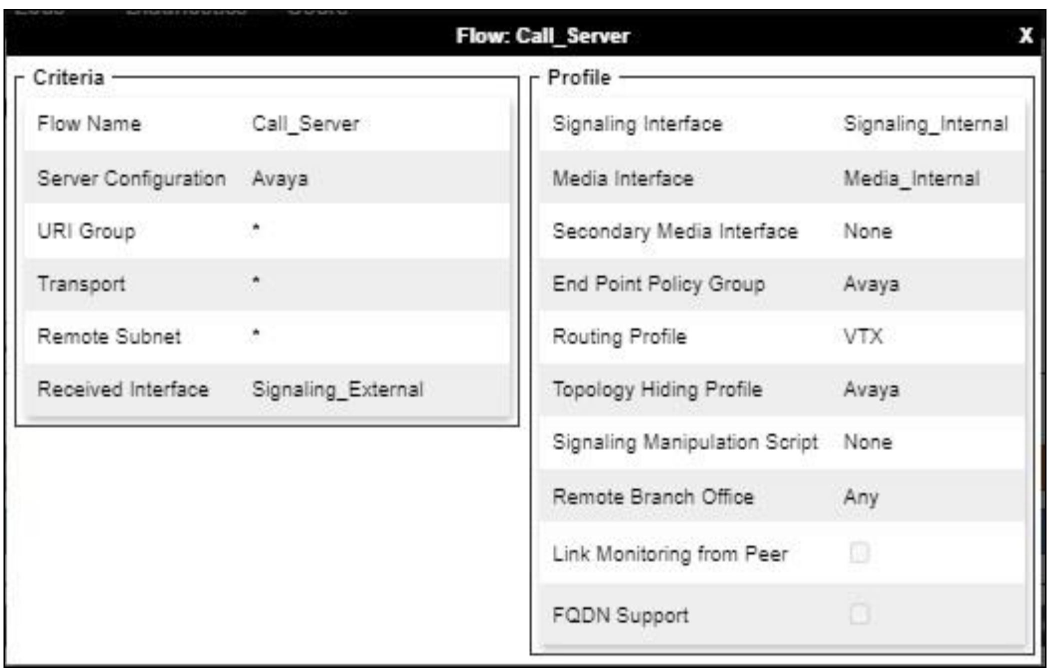
To define a Server Flow for the VTX SIP Trunk, navigate to **Network & Flows → End Point Flows**.

- Click on the **Server Flows** tab.
- Select **Add Flow** and enter details in the pop-up menu.
- In the **Name** field enter a descriptive name for the server flow for VTX SIP Trunk, in the test environment **Trunk_Server** was used.
- In the **Server Configuration** drop-down menu, select the VTX server configuration defined in **Section 8.7.2**.
- In the **Received Interface** drop-down menu, select the internal SIP signalling interface defined in **Section 8.4.1**.
- In the **Signaling Interface** drop-down menu, select the external SIP signalling interface defined in **Section 8.4.1**.
- In the **Media Interface** drop-down menu, select the external media interface defined in **Section 8.4.2**.
- Set the **End Point Policy Group** to the endpoint policy group **default-low**.
- In the **Routing Profile** drop-down menu, select the routing profile of the Session Manager defined in **Section 8.7.1**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of the VTX SIP Trunk defined in **Section 8.8** and click **Finish** (not shown).



To define an incoming server flow for Session Manager from the VTX network, navigate to **Network & Flows → End Point Flows**.

- Click on the **Server Flows** tab.
- Select **Add Flow** and enter details in the pop-up menu.
- In the **Name** field enter a descriptive name for the server flow for Session Manager, in the test environment **Call_Server** was used.
- In the **Server Configuration** drop-down menu, select the server configuration for Session Manager defined in **Section 8.7.1**.
- In the **Received Interface** drop-down menu, select the internal SIP signalling interface defined in **Section 8.4.1**.
- In the **Signaling Interface** drop-down menu, select the external SIP signalling interface defined in **Section 8.4.1**.
- In the **Media Interface** drop-down menu, select the external media interface defined in **Section 8.4.2**.
- Set the **End Point Policy Group** to the endpoint policy group **Avaya**.
- In the **Routing Profile** drop-down menu, select the routing profile of the VTX SIP Trunk defined in **Section 8.7.2**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of Session Manager defined in **Section 8.8** and click **Finish** (not shown).



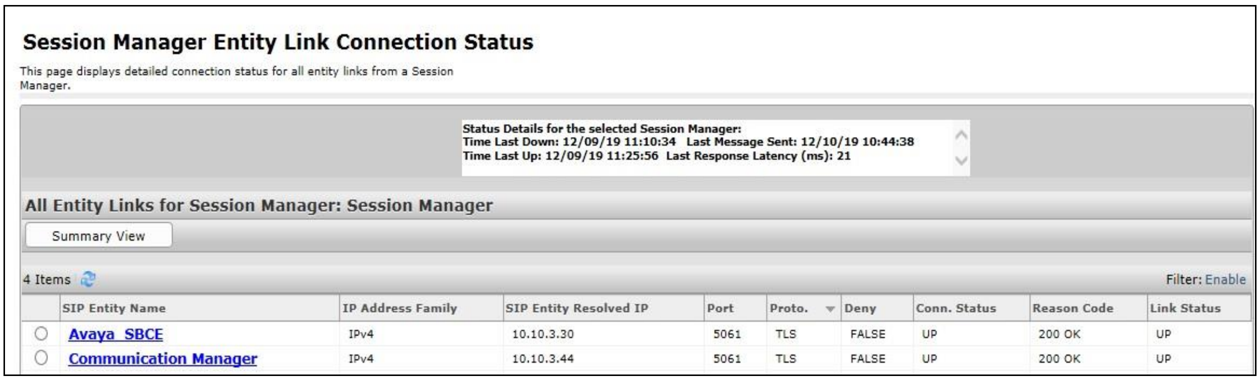
9. VTX Connect PBX-IP SIP Trunk Configuration

The configuration of the VTX equipment used to support VTX's SIP Trunk is outside of the scope of these Application Notes and will not be covered. To obtain further information on VTX equipment and system configuration please contact an authorized VTX representative as listed in **Section 2.3**.

10. Verification Steps

This section provides steps that may be performed to verify that the solution is configured correctly.

1. From System Manager **Home** tab click on **Session Manager** and navigate to **Session Manager → System Status → SIP Entity Monitoring**. Select the relevant SIP Entities from the list and observe if the **Conn Status** and **Link Status** are showing as **UP**.



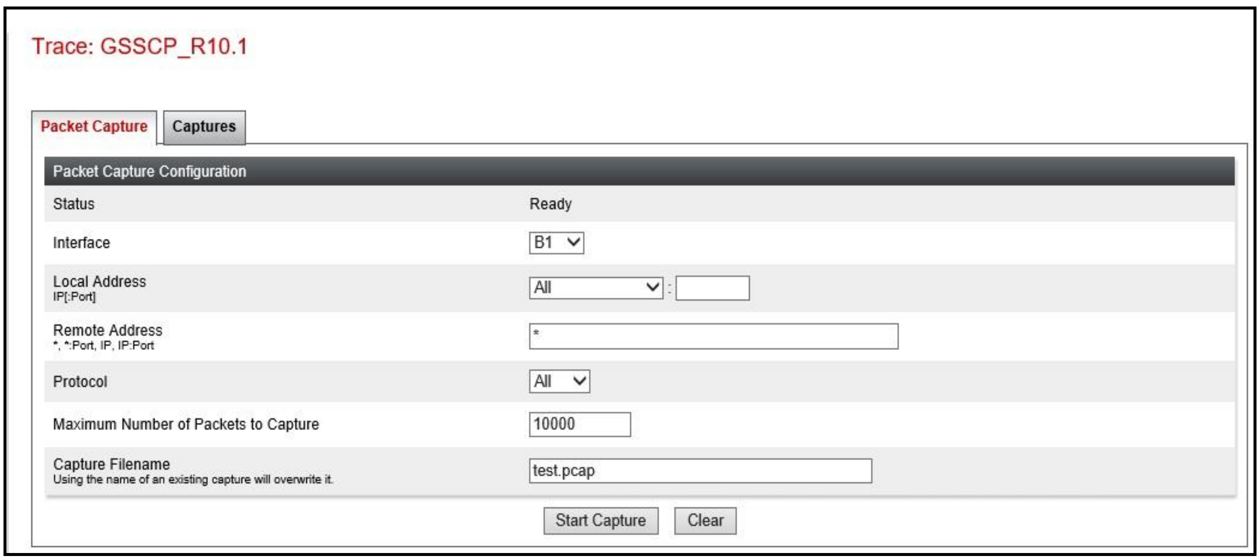
2. From Communication Manager SAT interface run the command **status trunk n** where **n** is a previously configured SIP trunk. Observe if all channels on the trunk group display **in-service/idle**.

```
status trunk 1

                          TRUNK GROUP STATUS

Member   Port     Service State     Mtce Connected Ports
                                    Busy

0002/001 T00011   in-service/idle    no
0002/002 T00012   in-service/idle    no
0002/003 T00013   in-service/idle    no
0002/004 T00014   in-service/idle    no
0002/005 T00015   in-service/idle    no
0002/006 T00016   in-service/idle    no
```

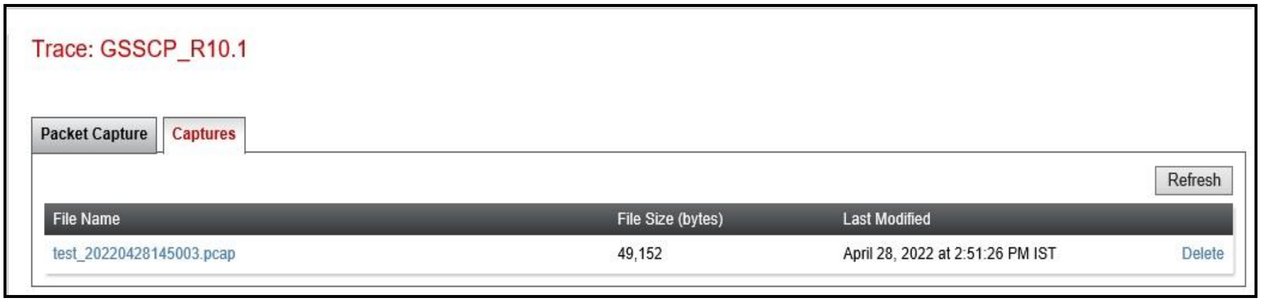
3. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active.
4. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active.
5. Verify that the user on the PSTN can end an active call by hanging up.
6. Verify that an endpoint at the enterprise site can end an active call by hanging up.
7. Should issues arise with the SIP trunk, use the Avaya SBC trace facility to check that the OPTIONS requests sent from Session Manager via the Avaya SBC to the network SBCs are receiving a response.

To define the trace, navigate to **Monitoring & Logging → Trace** in the main menu on the left-hand side and select the **Packet Capture** tab.

- Select the SIP Trunk interface from the **Interface** drop down menu.
- Select the signalling interface IP address or from the **Local Address** drop down menu.
- Enter the IP address of the network SBC in the **Remote Address** field or enter a ***** to capture all traffic.
- Specify the **Maximum Number of Packets to Capture**, **1000** is shown as an example.
- Specify the filename of the resultant .pcap file in the **Capture Filename** field.
- Click on **Start Capture**.



To view the trace, select the **Captures** tab and click on the relevant filename in the list of traces.



The trace is viewed as a standard .pcap file in Wireshark. If the SIP trunk is working correctly, a SIP response to OPTIONS in the form of a 200 OK will be seen from the VTX network.

11. Conclusion

These Application Notes describe the configuration necessary to successfully connect Avaya Aura ® Communication Manager R10.1, Avaya Aura ® Session Manager 10.1 and Avaya Session Border Controller R10.1 to the VTX Connect PBX-IP SIP platform.

The reference configuration shown in these Application Notes is representative of a basic enterprise customer configuration and demonstrates Avaya Aura ® Communication Manager R10.1, Avaya Aura ® Session Manager 10.1 with Avaya Session Border Controller can be configured to interoperate successfully with VTX Connect PBX-IP SIP Trunk Service. The VTX Connect PBX-IP SIP Service is a SIP-based Voice over IP solution providing businesses a flexible, cost-saving alternative to traditional hardwired telephony trunks. The service was successfully tested with a number of observations listed in **Section 2.2**.

12. Additional 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 Appliance Virtualization Platform*, Release 10.1, Sep 2023
[2] *Upgrading Avaya Aura® applications*, Release 10.1, Sep 2023
[3] *Deploying Avaya Aura® applications from System Manager*, Release 10.1, Feb 2023
[4] *Deploying Avaya Aura® Communication Manager*, Release 10.1, May 2023
[5] *Administering Avaya Aura® Communication Manager,* Release 10.1, May 2023
[6] *Upgrading Avaya Aura® Communication Manager*, Release 10.1, May 2023
[7] *Deploying Avaya Aura® System Manager*, Release 10.1, Feb 2023
[8] *Upgrading Avaya Aura® System Manager*, Release 10.1, Feb 2023
[9] *Administering Avaya Aura® System Manager*, Release 10.1, Feb 2023
[10] *Deploying Avaya Aura® Session Manager*, Release 10.1 Feb 2023
[11] *Upgrading Avaya Aura® Session Manager*, Release 10.1, Feb 2023
[12] *Administering Avaya Aura® Session Manager,* Release 10.1, Feb 2023
[13] *Deploying Avaya Session Border Controller*, Release 10.1, Jun 2023
[14] *Upgrading Avaya Session Border Controller*, Release 10.1 Jun 2023
[15] *Administering Avaya Session Border Controller*, Release 10.1, Jun 2023
[16] *Deploying Avaya Experience Portal*, Release 8.1.2, Oct 2022
[17] *Administering Avaya Experience Portal*, Release 8.1.2, Oct 2022
[18] *RFC 3261 SIP: Session Initiation Protocol*, http://www.ietf.org/

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.