Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura ® Communication Manager R8.1, Avaya Aura ® Session Manager R8.1 and Avaya Session Border Controller for Enterprise R8.1 to support DIDWW SIP Trunk Service - Issue 1.0

Abstract

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between the DIDWW SIP Trunk Service and an Avaya SIP enabled Enterprise Solution. The Avaya solution consists of Avaya Aura® Communication Manager R8.1, Avaya Aura® Session Manager R8.1 and Avaya Session Border Controller for Enterprise R8.1.

The DIDWW SIP Platform provides PSTN access via a SIP trunk connected to the DIDWW Voice over Internet Protocol (VoIP) network as an alternative to legacy analogue or digital trunks.

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.

DIDWW is a member of the DevConnect Service Provider program. Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between the DIDWW SIP Trunk Service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of the following: Avaya Aura ® Communication Manager R8.1 (Communication Manager); Avaya Aura ® Session Manager R8.1 (Session Manager) and Avaya Session Border Controller for Enterprise R8.1 (Avaya SBCE).

The DIDWW SIP Trunk platform offers a powerful two-way SIP trunking solution. The tested DIDWW configuration consisted of a separate inbound trunk dedicated for inbound traffic and a separate outbound trunk dedicated for outbound traffic. This ensures the DIDWW network is fully scalable with the possibility of virtually unlimited call capacity and advanced trunk configurations to support specific communication needs of corporate clients.

Customers using this Avaya SIP-enabled enterprise solution with the DIDWW SIP Trunk 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 and Avaya SBCE. The enterprise site was configured to connect to the DIDWW 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 via the inbound DIDWW SIP trunk, calls made to SIP and H.323 telephones at the enterprise.
- Outgoing calls from the enterprise site via the outbound DIDWW SIP trunk to PSTN destinations, calls made from SIP and H.323 telephones.
- Incoming and Outgoing PSTN calls to/from Avaya one-X® Communicator and Avaya Workplace for Windows soft phones via both inbound and outbound DIDWW SIP trunks.
- Calls using G.729, G.711A and G.711MU codecs.
- Fax calls via the DIDWW inbound SIP trunk from a PSTN-connected fax machine to a group 3 fax machine using T.38 fax transmissions.
- 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.
- Routing inbound vector call to call center agent queues.
- Inbound and outbound calls to toll-free numbers.
- Transmission and response of SIP OPTIONS messages sent by Avaya requiring 200 OK response by DIDWW.

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for the DIDWW SIP Trunking Service with the following observations:

- T.38 fax is not supported on the DIDWW outbound SIP trunk and therefore was not tested.
- Access to Emergency Services was not tested as no test call had been booked by the Service Provider with the Emergency Services 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 DIDWW products please contact the DIDWW support team:

- Website: https://www.didww.com/
- Email: support@didww.com
- Contact phone numbers: US 1-212-6600065, UK: 44-20-80995011.
- Documentation: https://doc.didww.com/

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an Enterprise site connected to the DIDWW SIP platform. Located at the Enterprise site is an Avaya SBCE, Session Manager and Communication Manager. Endpoints are Avaya 96x1 series IP telephones (with SIP and H.323 firmware), Avaya J179 series IP telephone (with SIP firmware), Avaya 16xx series IP telephones (with H.323 firmware), 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 for Windows running on laptop PCs.

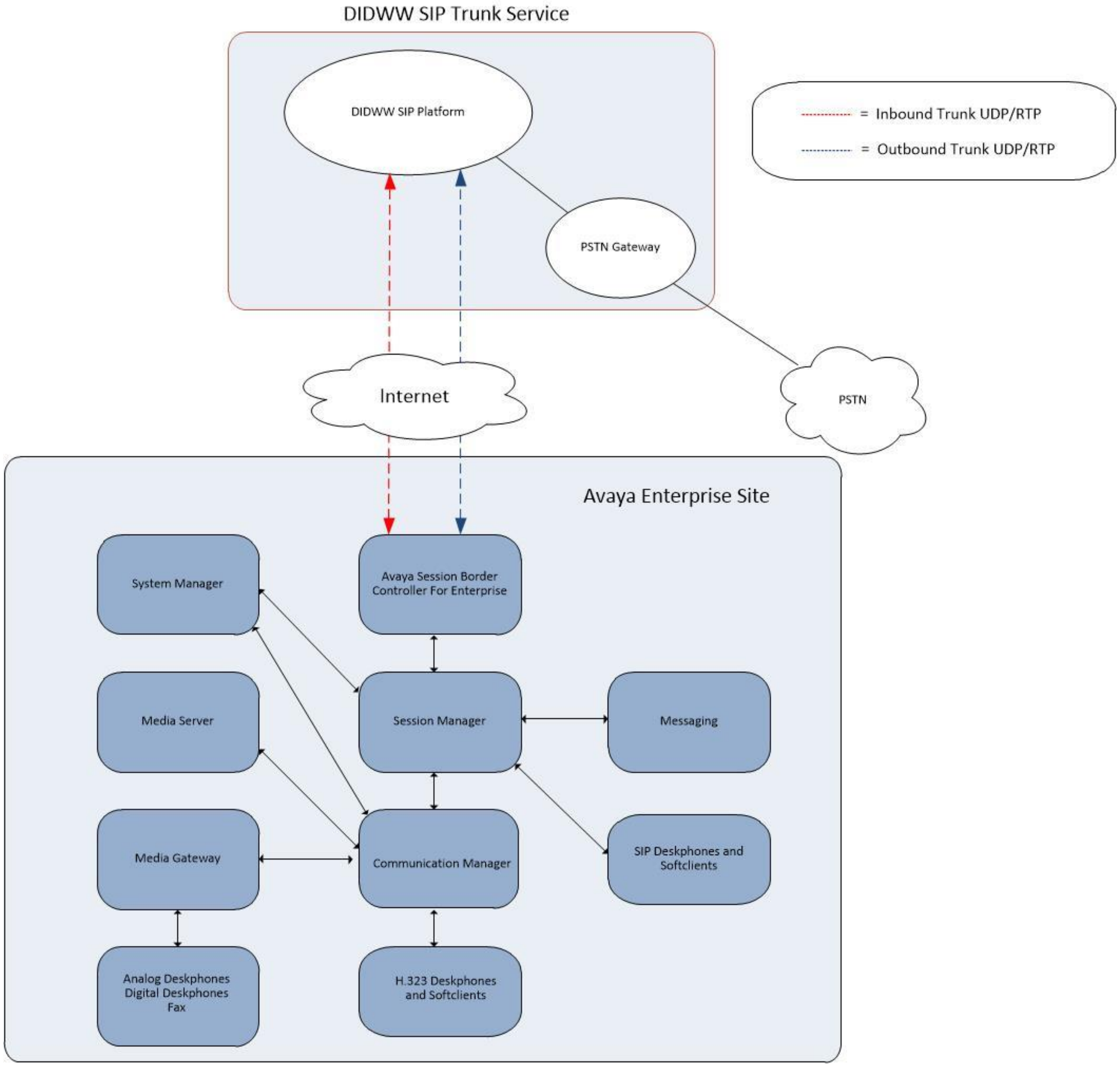


Figure 1: Test Setup DIDWW SIP Trunk Service to Avaya Enterprise

4. Equipment and Software Validated

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

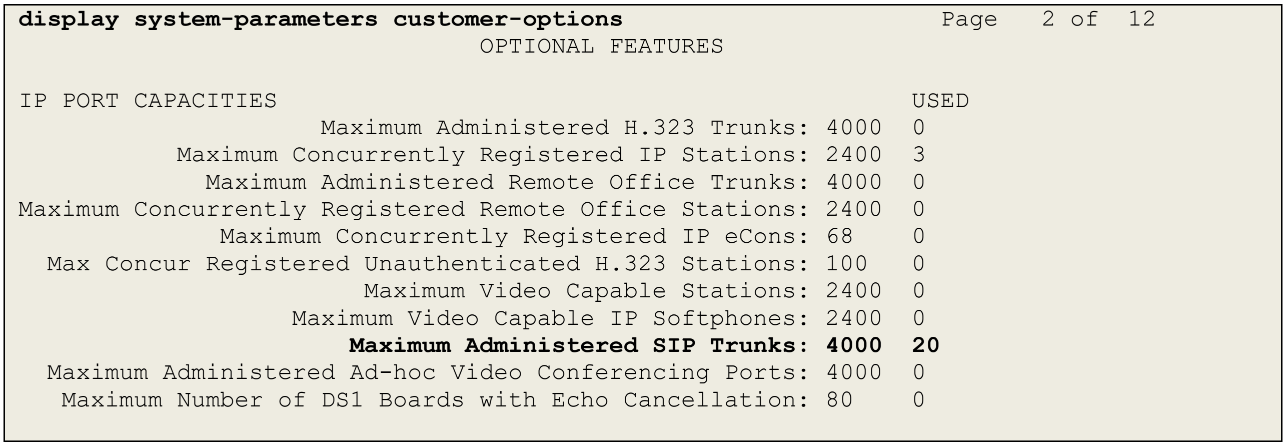
Equipment/Software	Release/Version
Avaya	
Avaya Aura® System Manager	8.1.3.3 Build No. – 8.1.0.0.733078 Software Update Revision No: 8.1.3.3.1013529 Service Pack 3
Avaya Aura® Session Manager	8.1.3.3.813310
Avaya Aura® Communication Manager	8.1.3.1 – 26766
Avaya Session Border Controller for Enterprise	8.1.3.0-31-21052
Avaya G430 Media Gateway	41.34.3
Avaya Aura® Media Server	v.8.0.2.SP8
Avaya 1600 IP Deskphone (H.323)	1.3.12
Avaya 96x1 IP DeskPhone (H.323)	6.8.5
Avaya 9611 IP DeskPhone (SIP)	7.1.14
Avaya 9608 IP DeskPhone (SIP)	7.1.14
Avaya J179 IP Deskphone (SIP)	4.0.10.2
Avaya one–X® Communicator (H.323 & SIP)	6.2.14.15 -SP14-Patch 7
Avaya Workplace for Windows (SIP)	3.23.0.64
Analogue Handset	N/A.
Analogue Fax	N/A
DIDWW SIP Platform	
DIDWW SBC	Version: 1.X

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 DIDWW SIP Trunking Service. For incoming calls, Session Manager receives SIP messages from the Avaya SBCE 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 SBCE at the enterprise site that then sends the SIP messages to the DIDWW 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 DIDWW SIP Trunking Service and any other SIP trunks used.



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

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 4000  0
           Maximum Concurrently Registered IP Stations: 2400  3
             Maximum Administered Remote Office Trunks: 4000  0
Maximum Concurrently Registered Remote Office Stations: 2400  0
              Maximum Concurrently Registered IP eCons: 68    0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 2400  0
                   Maximum Video Capable IP Softphones: 2400  0
                       Maximum Administered SIP Trunks: 4000  20
  Maximum Administered Ad-hoc Video Conferencing Ports: 4000  0
   Maximum Number of DS1 Boards with Echo Cancellation: 80    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:

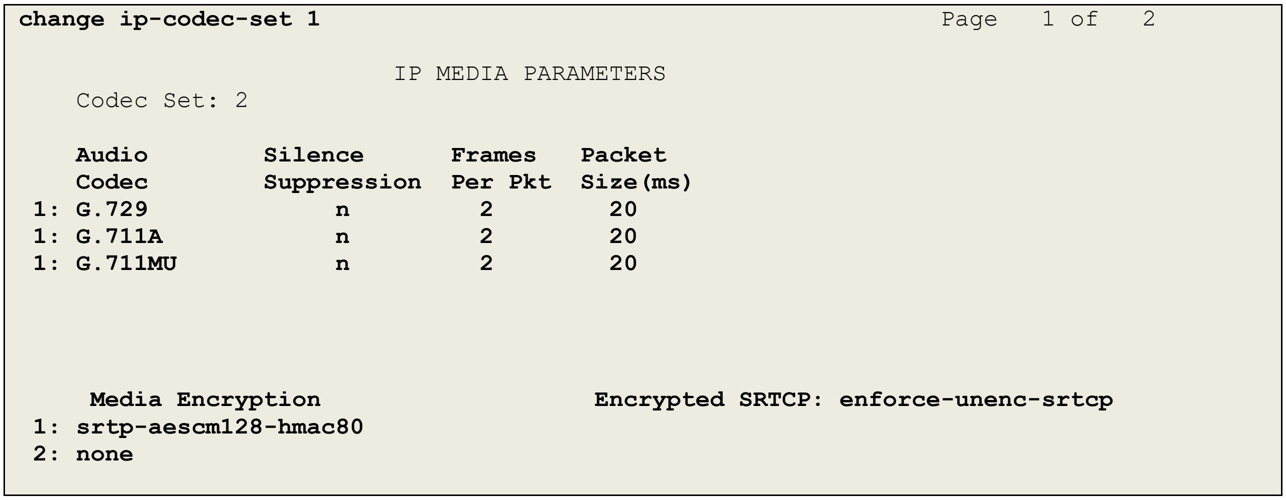
- The **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is **avaya.com**.
- By default, **IP-IP Direct Audio** (both **Intra**- and **Inter-Region**) is enabled (**yes**) to allow audio traffic to be sent directly 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 SBCE.
- 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: 2
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 codecs supported by DIDWW were configured, namely **G.729**, **G.711A** and **G.711MU**.

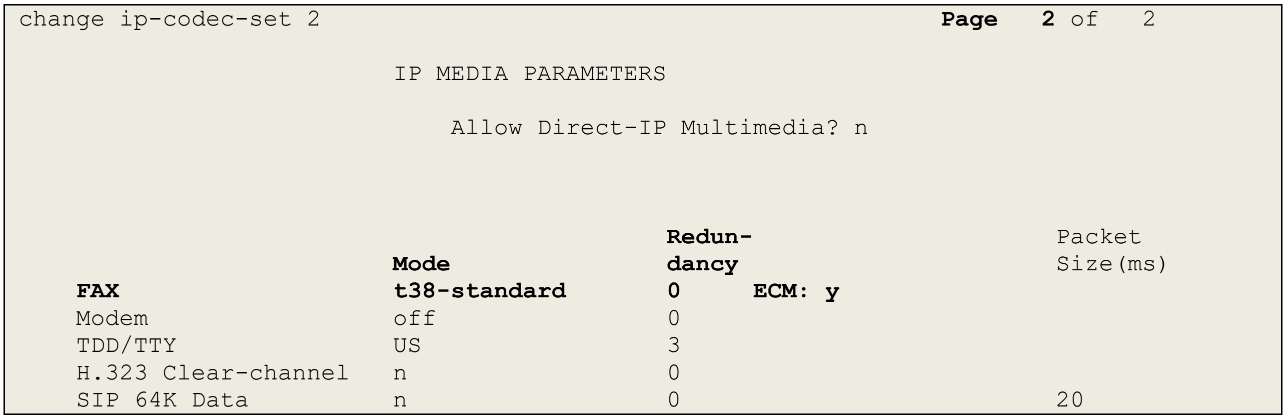
In addition to the codec's, the **Media Encryption** is defined here. For the compliance test, a value of **srtp-aescm128-hmac80** was used.



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

                          IP MEDIA PARAMETERS
    Codec Set: 2

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




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

DIDWW SIP trunk supports T.38 for transmission of fax on the DIDWW Inbound SIP trunk only as per **Section 2.2**. 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**.



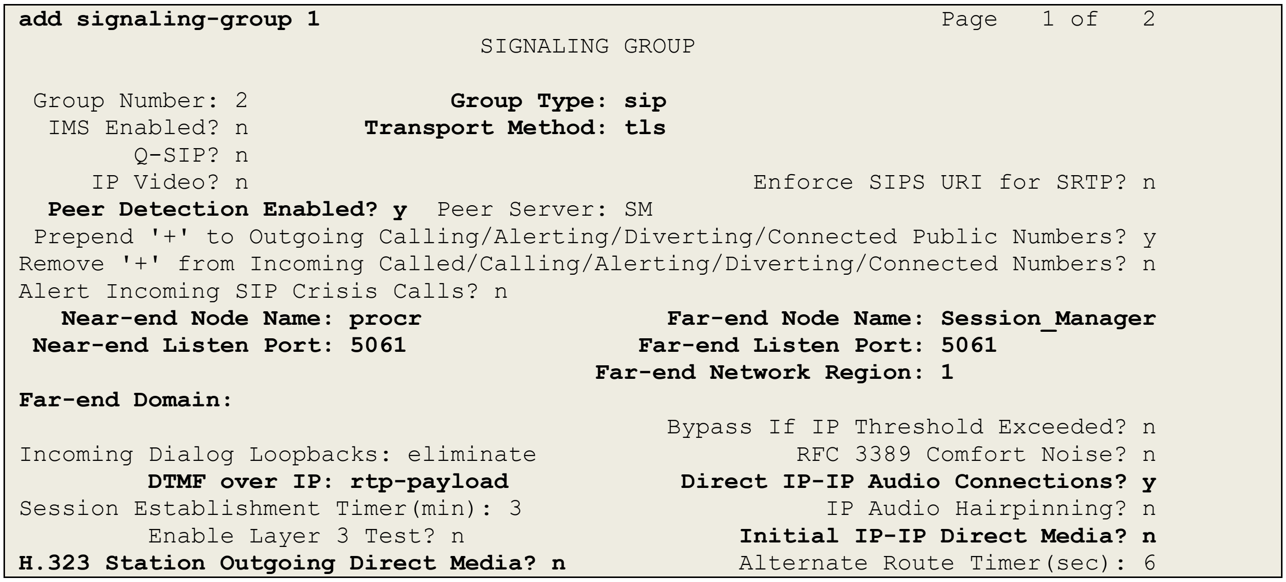
```
change ip-codec-set 2                                           Page   2 of   2

                          IP MEDIA PARAMETERS

                              Allow Direct-IP Multimedia? n



                                             Redun-                      Packet
                          Mode               dancy                       Size(ms)
    FAX                   t38-standard       0     ECM: y
    Modem                 off                0
    TDD/TTY               US                 3
    H.323 Clear-channel   n                  0
    SIP 64K Data          n                  0                           20
```

5.5. Administer SIP Signaling Groups

This signalling group (and trunk group) will be used for inbound and outbound PSTN calls to the DIDWW SIP Trunking Service. 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**.
- Set **H.323 Station Outgoing Direct Media** to **n**.

The default values for the other fields may be used.



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

 Group Number: 2              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                    Enforce SIPS URI for SRTP? 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? n               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:

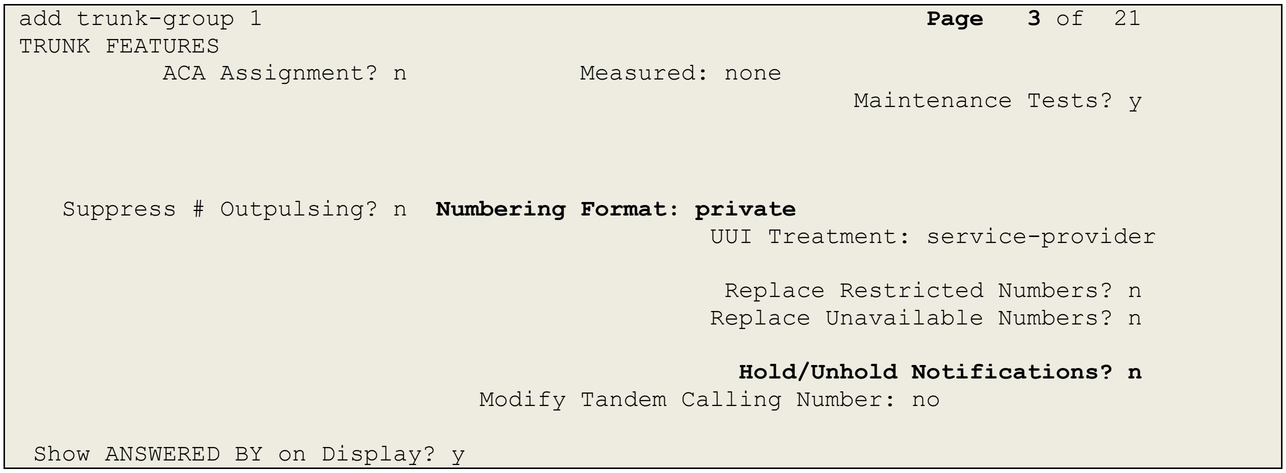
- Set the **Group Type** field to **sip**.
- Choose a descriptive **Group Name**.
- Specify a trunk access code (**TAC**) consistent with the dial plan.
- The **Direction** is set to **two-way** to allow incoming and outgoing calls.
- Set the **Service Type** field to **public-netwrk**.
- Specify the 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 DIDWW to prevent unnecessary SIP messages during call setup. During testing, a value of **180** was used.

```
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): 180

 Disconnect Supervision - In? y  Out? y


             XOIP Treatment: auto    Delay Call Setup When Accessed Via IGAR? n

 Caller ID for Service Link Call to H.323 1xC: station-extension
```

On **Page 3**, set the **Numbering Format** field to **private**. Also, set the **Hold/Unhold Notifications** to **n**.

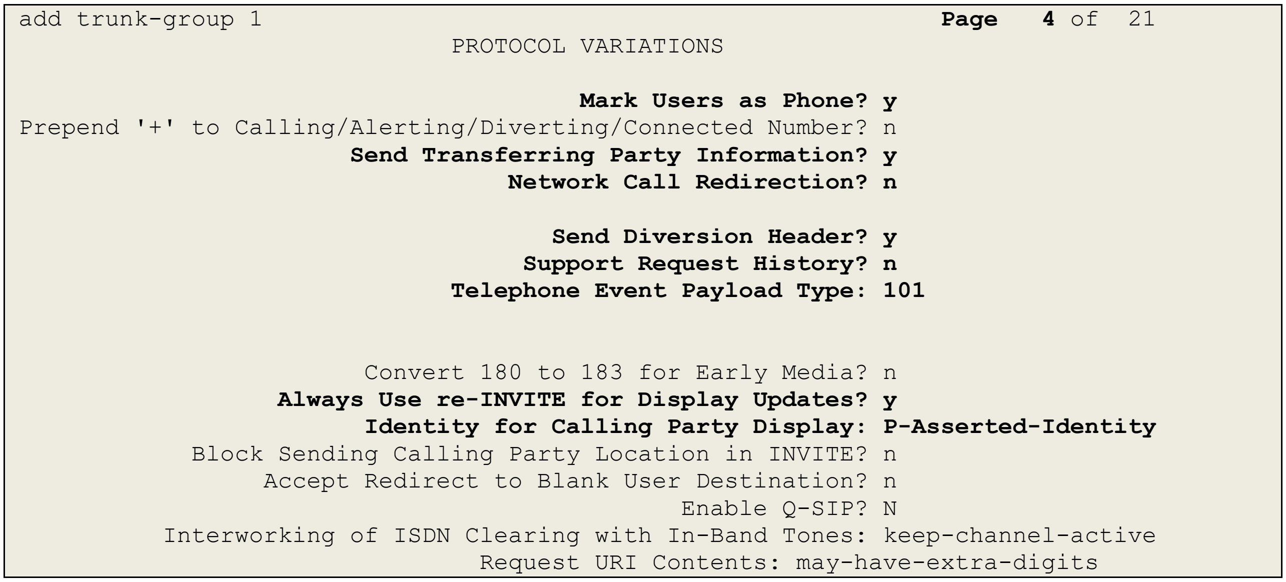


```
add trunk-group 1                                              Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y



   Suppress # Outpulsing? n  Numbering Format: private
                                                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 **y**.
- Set **Network Call Direction** to **y**.
- Set **Send Diversion Header** to **y**.
- Set **Support Request History** to **n**.
- Set the **Telephone Event Payload Type** to **101**.
- Set **Always Use re-INVITE for Display Updates** to **y**.
- Set the **Identity for Calling Party Display** to **P-Asserted-Identity**.

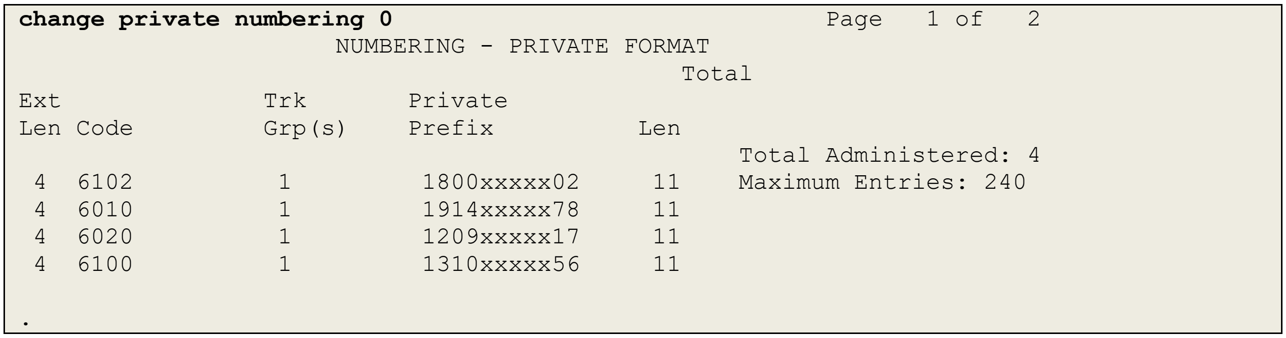
All other setting can be left at default.



```
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? y
                                  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

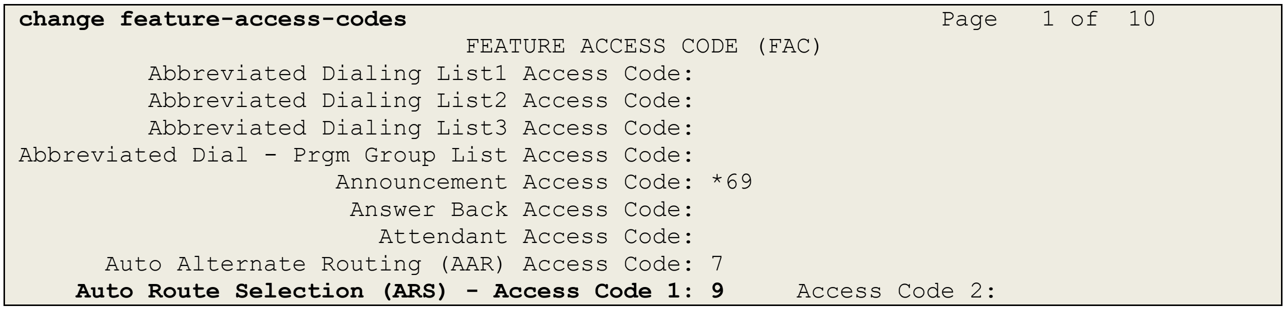
Use the **change private-numbering** command to configure Communication Manager to send the calling party number in the format required. These calling party numbers are sent in the SIP From, Contact and PAI headers as well as the Diversion header for forwarded calls. The numbers are displayed on display-equipped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



```
change private numbering 0                              Page   1 of   2
                      NUMBERING - PRIVATE FORMAT
                                              Total
Ext              Trk       Private
Len Code         Grp(s)    Prefix          Len
                                                  Total Administered: 4
 4  6102          1         1800xxxxx02     11    Maximum Entries: 240
 4  6010          1         1914xxxxx78     11
 4  6020          1         1209xxxxx17     11
 4  6100          1         1310xxxxx56     11

.
```

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 DIDWW 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 doesn't have the same significance in SIP calls and during testing it was set to **unk-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

 1: y y y y y n  n            rest                               unk-unk   none
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
 6: y y y y y n  n            rest                                         none
```

5.9. Administer Incoming Digit Translation

This step configures the settings necessary to map incoming DDI calls to the proper Communication Manager extension(s). The incoming digits sent in the INVITE message from DIDWW can be manipulated as necessary to route calls to the desired extension. In the examples used in the compliance testing, the incoming DDI numbers provided by DIDWW SIP platform correlate to the internal extensions assigned within Communication Manager. The entries displayed below translate incoming DDI numbers **1800xxxxx02**, **1914xxxxx78**, **1209xxxxx17** and **1310xxxxx56** to a 4-digit extension by deleting all of the incoming digits and inserting an extension.

```
change inc-call-handling-trmt trunk-group 1                    Page   1 of   3
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number         Del Insert
 Feature        Len     Digits
 public-ntwrk    11 1800xxxxx02     all  6102
 public-ntwrk    11 1914xxxxx78     all  6010
 public-ntwrk    11 1209xxxxx17     all  6020
 public-ntwrk    11 1310xxxxx56     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** if required by the routing configuration, none was required during testing.
- For the **Phone Number** enter the phone that will also be called (e.g. **0035389434xxxx**).
- Set the **Trunk Selection** to **ars** so that the ARS table will be used for routing.
- 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

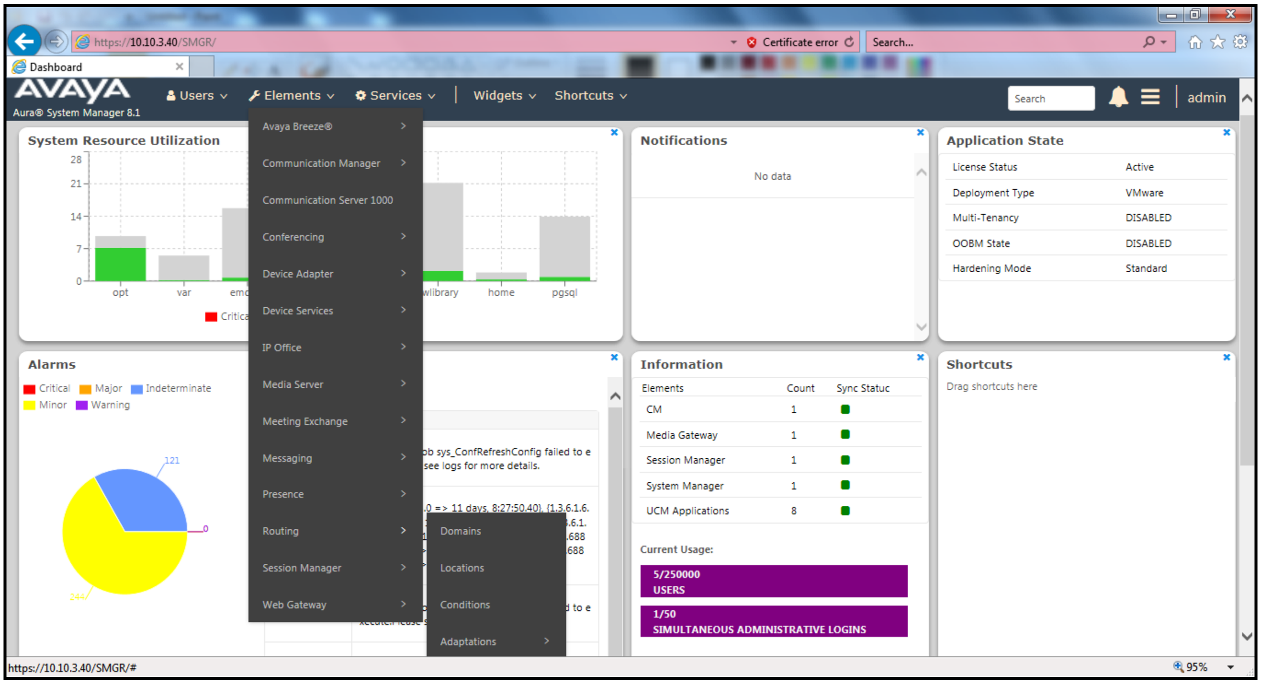
This section provides the procedures for configuring Session Manager. Session Manager is configured via System Manager. The procedures include the following areas:

- 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

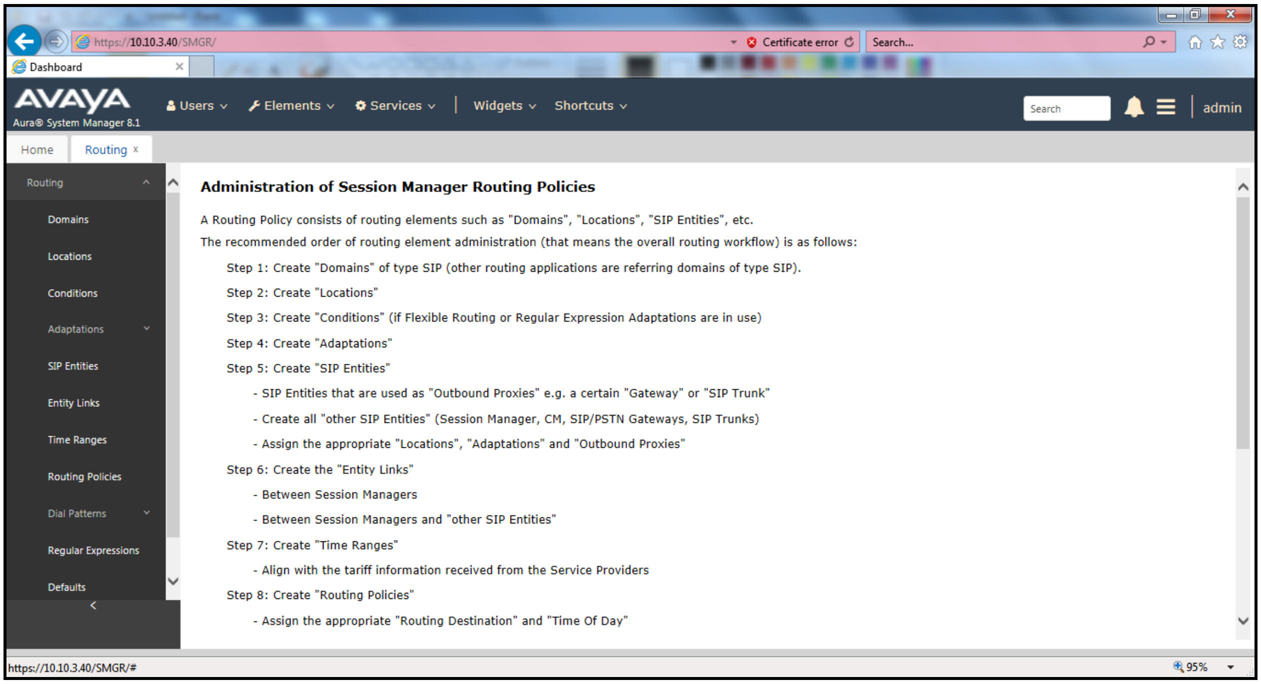
Access the System Manager using a web browser and entering **http://<FQDN >/SMGR**, where <**FQDN**> is the fully qualified domain name of System Manager. Log in using appropriate credentials (not shown) and the Dashboard tab will be presented with menu options shown below.



Application State	
License Status	Active
Deployment Type	VMware
Multi-Tenancy	DISABLED
OOBM State	DISABLED
Hardening Mode	Standard

Elements	Count	Sync Status
CM	1	🟩
Media Gateway	1	🟩
Session Manager	1	🟩
System Manager	1	🟩
UCM Applications	8	🟩

Most of the configuration items are performed in the Routing Element. Click on **Routing** in the Elements column shown above to bring up the **Introduction to Network Routing Policy** screen.



Administration of Session Manager Routing Policies

A Routing Policy consists of routing elements such as "Domains", "Locations", "SIP Entities", etc.

The recommended order of routing element administration (that means the overall routing workflow) is as follows:

Step 1: Create "Domains" of type SIP (other routing applications are referring domains of type SIP).

Step 2: Create "Locations"

Step 3: Create "Conditions" (if Flexible Routing or Regular Expression Adaptations are in use)

Step 4: Create "Adaptations"

Step 5: Create "SIP Entities"

- SIP Entities that are used as "Outbound Proxies" e.g. a certain "Gateway" or "SIP Trunk"
- Create all "other SIP Entities" (Session Manager, CM, SIP/PSTN Gateways, SIP Trunks)
- Assign the appropriate "Locations", "Adaptations" and "Outbound Proxies"

Step 6: Create the "Entity Links"

- Between Session Managers
- Between Session Managers and "other SIP Entities"

Step 7: Create "Time Ranges"

- Align with the tariff information received from the Service Providers

Step 8: Create "Routing Policies"

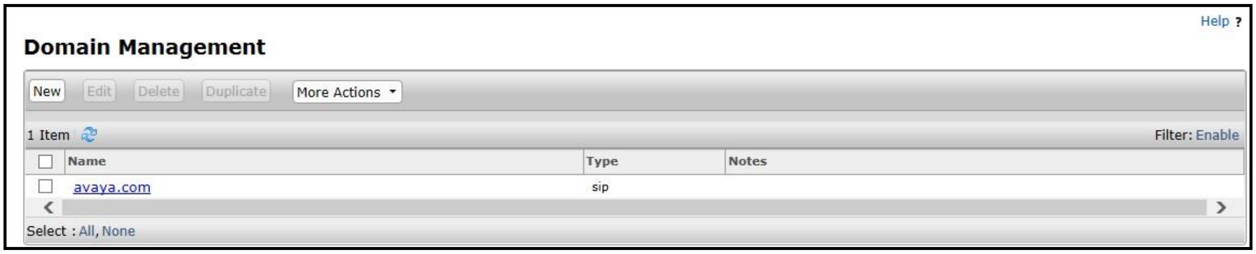
- Assign the appropriate "Routing Destination" and "Time Of Day"

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.



Domain Management

1 Item

□	Name	Type	Notes
□	avaya.com	sip	

6.3. Administer Locations

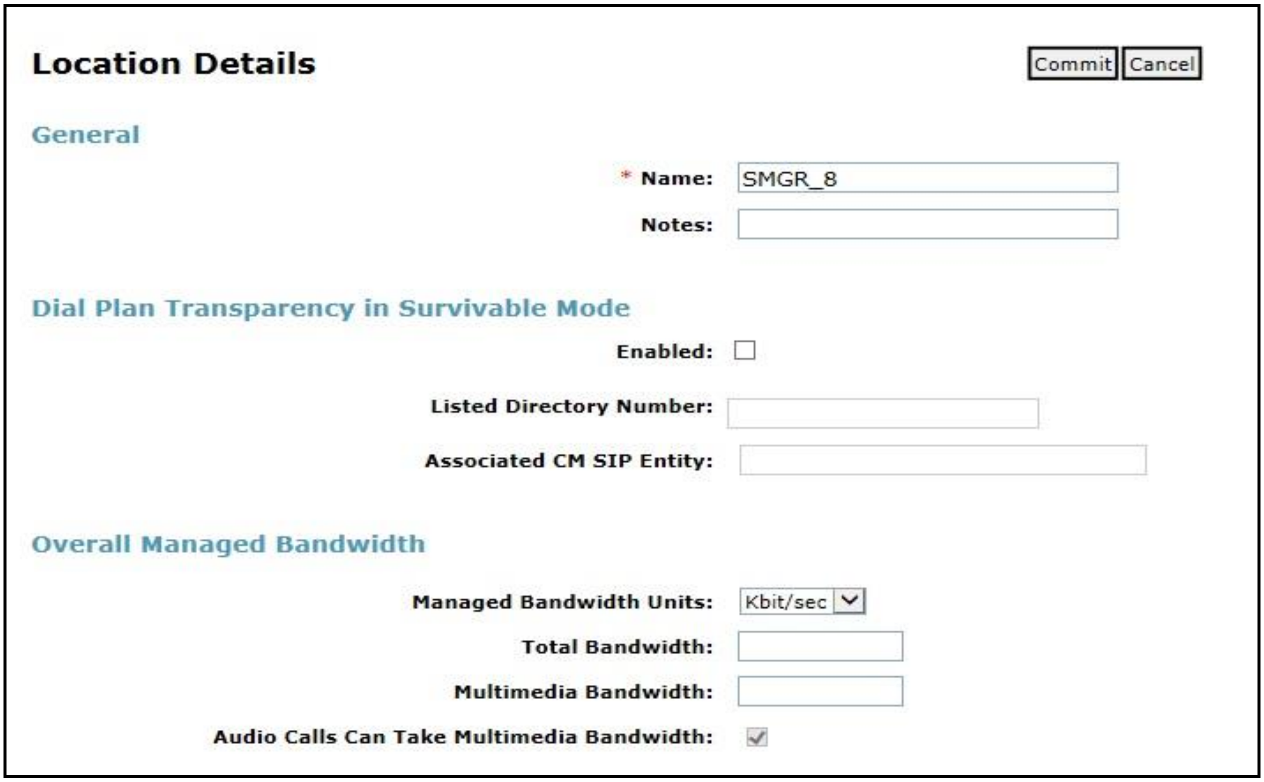
Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management and call admission control. To add a location, navigate to **Routing →Locations** in the left-hand navigation pane and click the **New** button in the right pane (not shown). 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 Location Pattern is used to identify call routing based on IP address. Session Manager matches the IP address against the patterns defined in this section. If a call is from a SIP Entity that does not match the IP address pattern, then Session Manager uses the location administered for the SIP Entity.
In the **Location Pattern** section, click **Add** and enter the following values.

- **IP Address Pattern** Enter the logical pattern used to identify the location.
- **Notes** Add a brief description [Optional].

Click **Commit** to save. The screenshot below shows the Location **SMGR_8** defined for the compliance testing.



General

* Name: SMGR_8

Notes:

Dial Plan Transparency in Survivable Mode

Enabled: □

Listed Directory Number:

Associated CM SIP Entity:

Overall Managed Bandwidth

Managed Bandwidth Units: Kbit/sec

Total Bandwidth:

Multimedia Bandwidth:

Audio Calls Can Take Multimedia Bandwidth: ☑

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 and remove headers from messages received from DIDWW. 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 **Digit Conversion** in the Adaptation. In order to improve interoperability with third party elements, Session Manager R8.1 incorporates the ability to use Adaptation modules to remove specific SIP headers that are either Avaya proprietary unnecessary for non-Avaya elements

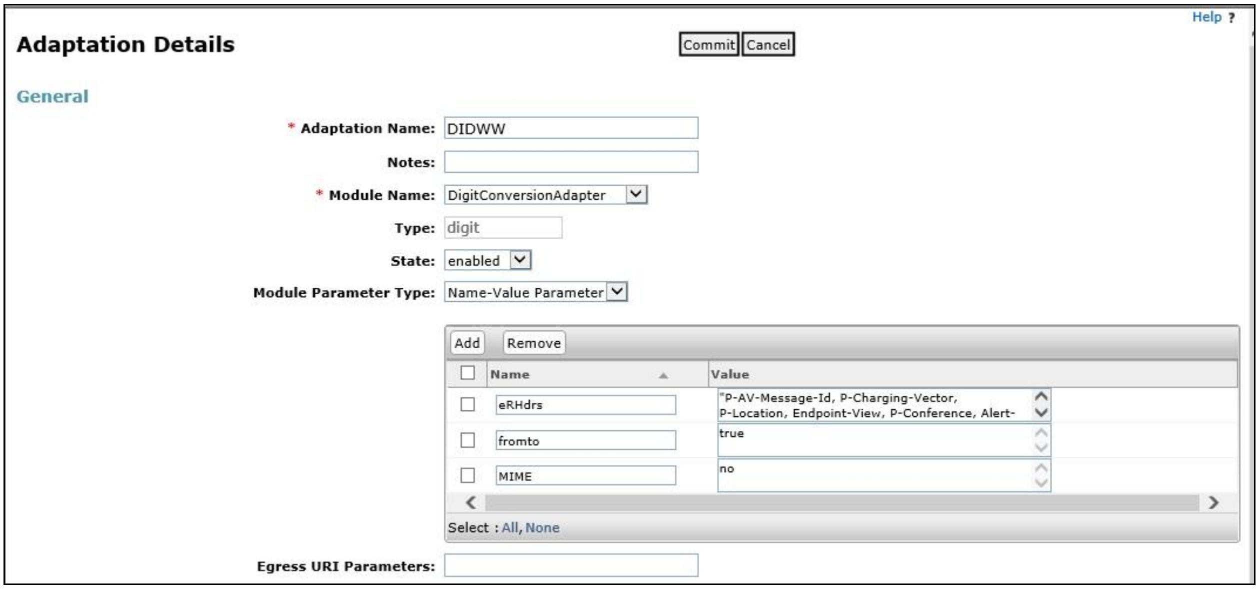
For the compliance test, an Adaptation named "**DIDWW**" was created to block the following headers from outbound messages, before they were forwarded to the Avaya SBCE: AV-Global-Session-ID, AV-Correlation-ID, Alert-Info, Endpoint-View, P-AV-Message-ID, P-Charging-Vector, and P-Location. These headers 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 **DIDWW**.
- **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**.



Adaptation Details

General

* Adaptation Name:	DIDWW
Notes:	
* Module Name:	DigitConversionAdapter
Type:	digit
State:	enabled
Module Parameter Type:	Name-Value Parameter

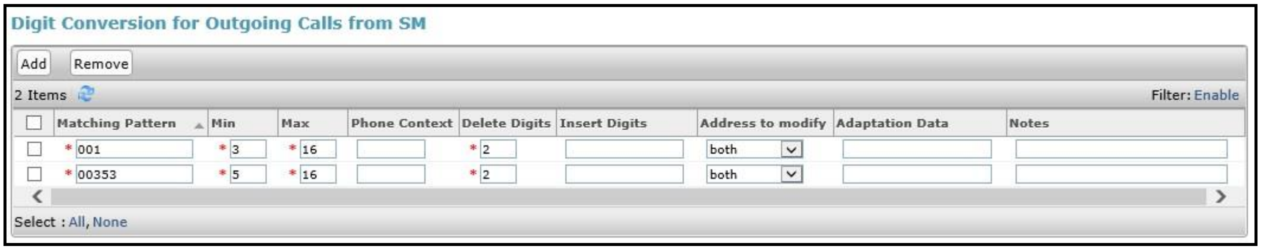
□	Name	Value
□	eRHdrs	"P-AV-Message-Id, P-Charging-Vector, P-Location, Endpoint-View, P-Conference, Alert-
□	fromto	true
□	MIME	no

Select : All, None

Egress URI Parameters:

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.


Digit Conversion for Outgoing Calls from SM

2 Items Filter: Enable

□	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
□	* 001	* 3	* 16		* 2		both		
□	* 00353	* 5	* 16		* 2		both		

Select : All, None

The above screenshot will ensure any outgoing numbers matching 001 and 00353 will have the first two digits 00 deleted from CLI before being presented to the DIDWW Outbound SIP Trunk.

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 SBCE 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.

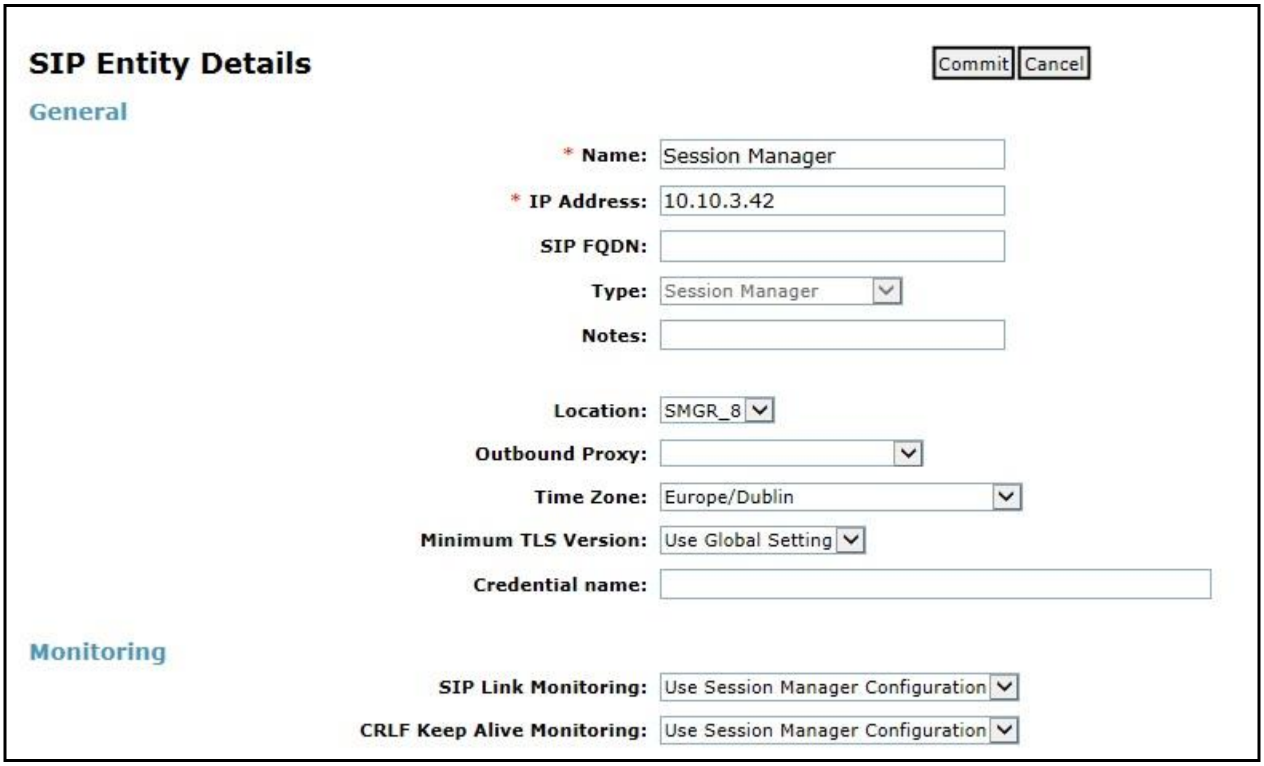
In this configuration there are four SIP Entities.

- Session Manager SIP Entity.
- Communication Manager SIP Entity.
- Avaya SBCE / DIDWW Inbound SIP Entity.
- Avaya SBCE / DIDWW Outbound SIP Entity.

In the DIDWW network, two network SBCs are provided as the interface to the enterprise equipment. For the purposes of this document have been designated as DIDWW Inbound trunk and DIDWW Outbound trunk. The routing for both these DIDWW inbound and outbound SIP trunks is configured on Session Manager, with two server flows configured on the Avaya SBCE for routing to each SIP trunk. There is an interface configured on the Avaya SBCE for each of these server flows, and a corresponding SIP Entity, Entity Link and Routing Policy is required on the Session Manager for each of these interfaces.

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 in **Section 6.3** and the **Time Zone** to the appropriate time.



SIP Entity Details

General

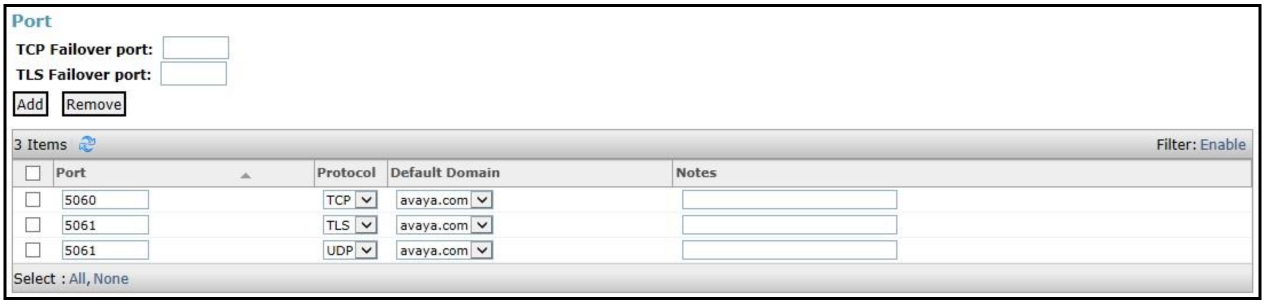
* Name:	Session Manager
* IP Address:	10.10.3.42
SIP FQDN:	
Type:	Session Manager
Notes:	
Location:	SMGR_8
Outbound Proxy:	
Time Zone:	Europe/Dublin
Minimum TLS Version:	Use Global Setting
Credential name:	

Monitoring

SIP Link Monitoring:	Use Session Manager Configuration
CRLF Keep Alive Monitoring:	Use Session Manager Configuration

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.



Port

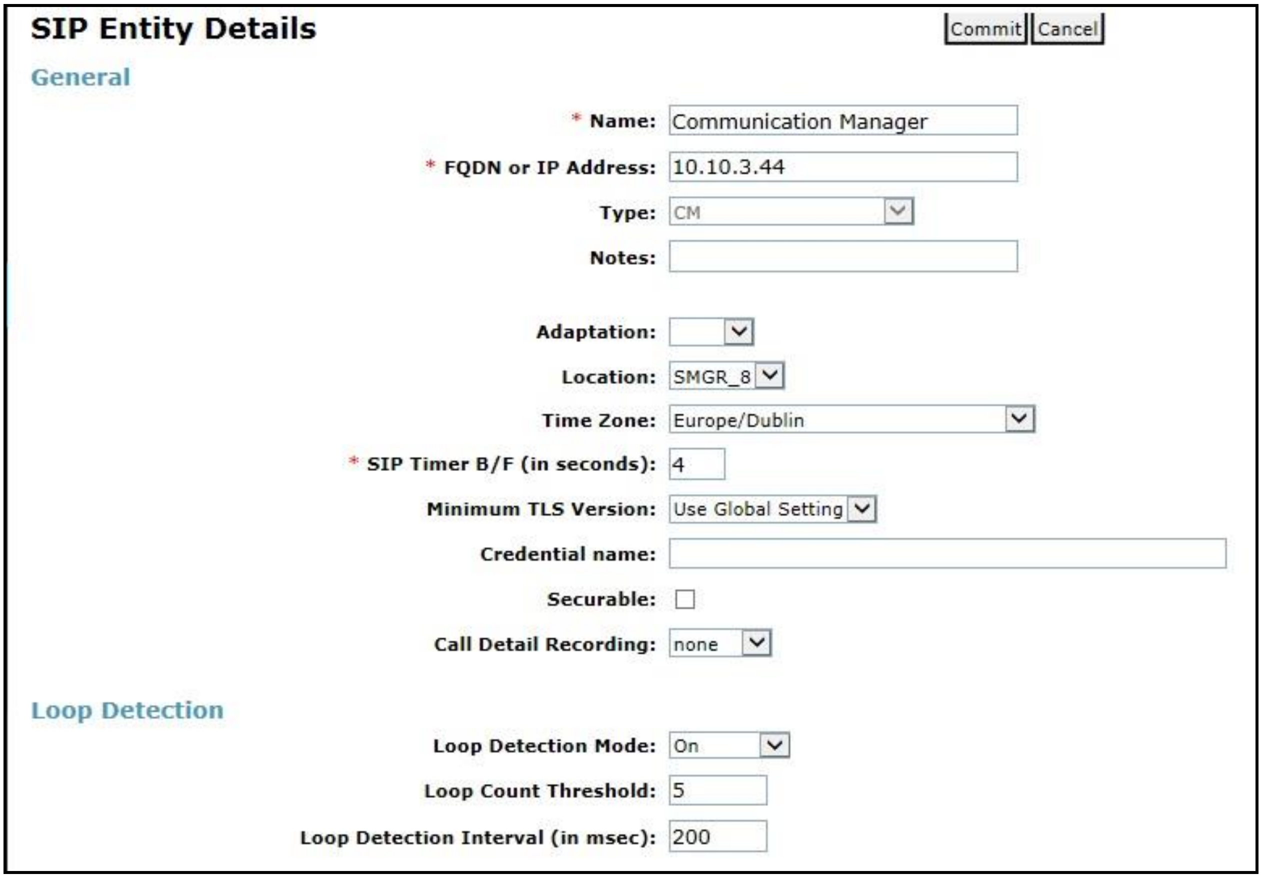
TCP Failover port:
TLS Failover port:

3 Items

□	Port	Protocol	Default Domain	Notes
□	5060	TCP	avaya.com	
□	5061	TLS	avaya.com	
□	5061	UDP	avaya.com	

6.5.2. Avaya Aura® Communication Manager SIP Entity

The following screen shows the SIP entity for Communication Manager which is configured as an Evolution Server. 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 in **Section 6.3**.



SIP Entity Details

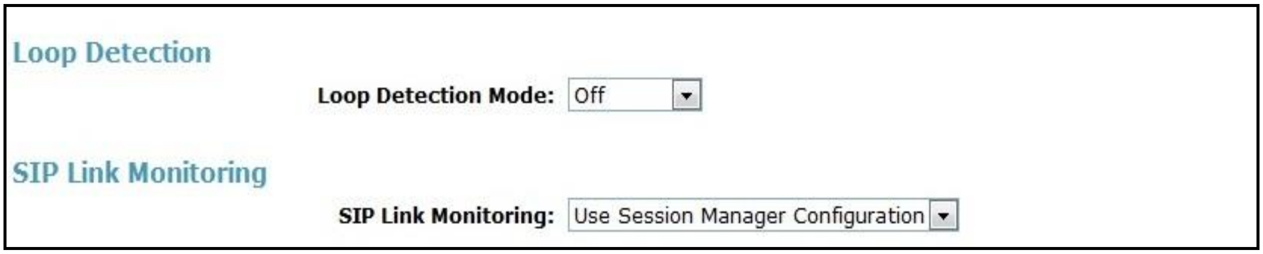
General

* Name:	Communication Manager
* FQDN or IP Address:	10.10.3.44
Type:	CM
Notes:	
Adaptation:	
Location:	SMGR_8
Time Zone:	Europe/Dublin
* SIP Timer B/F (in seconds):	4
Minimum TLS Version:	Use Global Setting
Credential name:	
Securable:	□
Call Detail Recording:	none

Loop Detection

Loop Detection Mode:	On
Loop Count Threshold:	5
Loop Detection Interval (in msec):	200

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



Loop Detection

Loop Detection Mode: Off

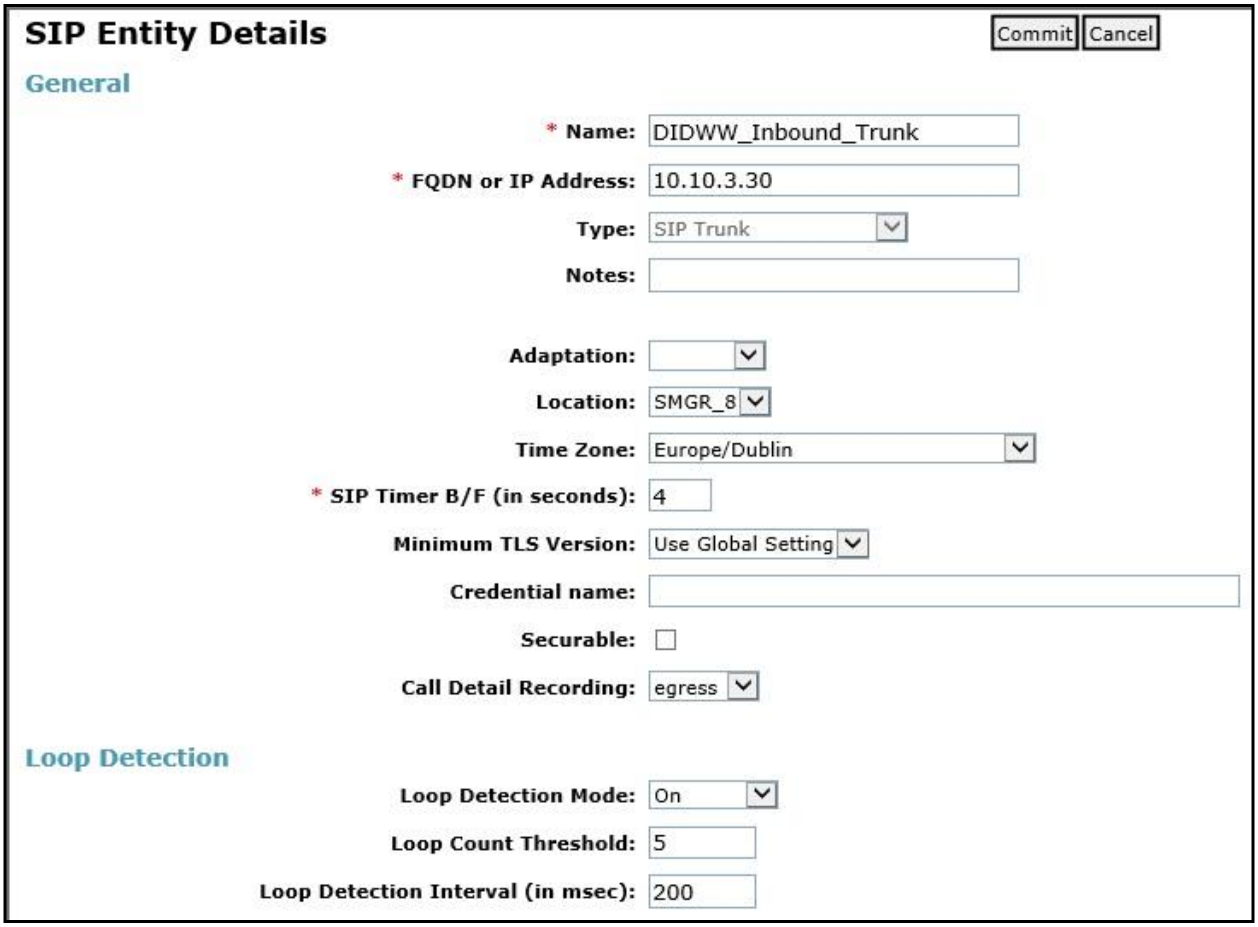
SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

6.5.3. Avaya Session Border Controller for Enterprise SIP Entities

Two SIP Entities were used for the two interfaces established so that separate routing could take place to both the inbound and outbound DIDWW SIP trunks.

The following screen shows the SIP Entity for the DIDWW Inbound trunk. The **FQDN or IP Address** field is set to the IP address of the Avaya SBCE private network interface (See **Section 7.4.1**). Set the **Location** to that defined in **Section 6.3** and the **Time Zone** to the appropriate time zone.



SIP Entity Details

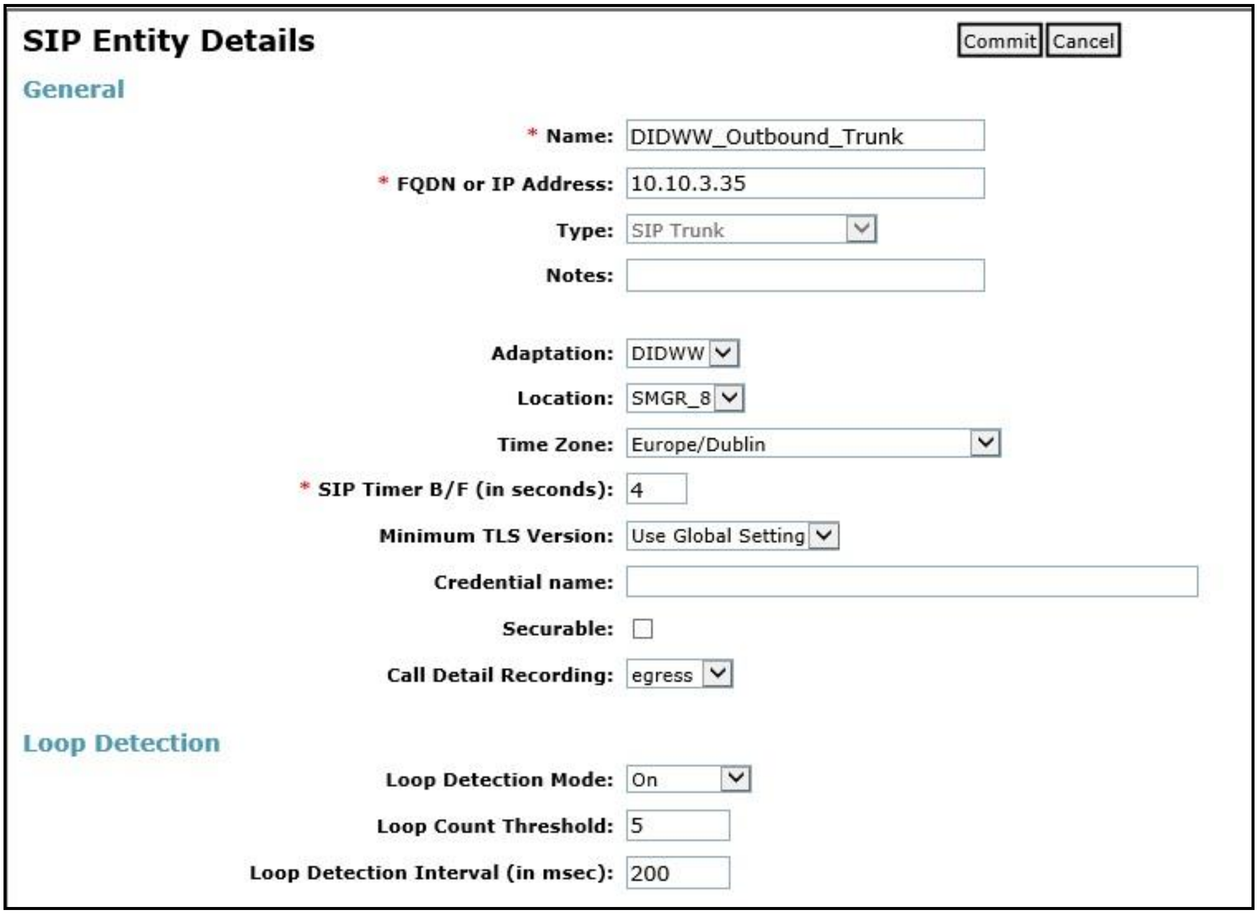
General

* Name:	DIDWW_Inbound_Trunk
* FQDN or IP Address:	10.10.3.30
Type:	SIP Trunk
Notes:	
Adaptation:	
Location:	SMGR_8
Time Zone:	Europe/Dublin
* SIP Timer B/F (in seconds):	4
Minimum TLS Version:	Use Global Setting
Credential name:	
Securable:	□
Call Detail Recording:	egress

Loop Detection

Loop Detection Mode:	On
Loop Count Threshold:	5
Loop Detection Interval (in msec):	200

The following screen shows the SIP Entity for the DIDWW Outbound trunk. The **FQDN or IP Address** field is set to the IP address of the Avaya SBCE private network interface (See **Section 7.4.1**). Set the **Adaptation** to that defined in **Section 6.4**, the **Location** to that defined in **Section 6.3** and the **Time Zone** to the appropriate time zone.



SIP Entity Details

General

* Name:	DIDWW_Outbound_Trunk
* FQDN or IP Address:	10.10.3.35
Type:	SIP Trunk
Notes:	
Adaptation:	DIDWW
Location:	SMGR_8
Time Zone:	Europe/Dublin
* SIP Timer B/F (in seconds):	4
Minimum TLS Version:	Use Global Setting
Credential name:	
Securable:	□
Call Detail Recording:	egress

Loop Detection

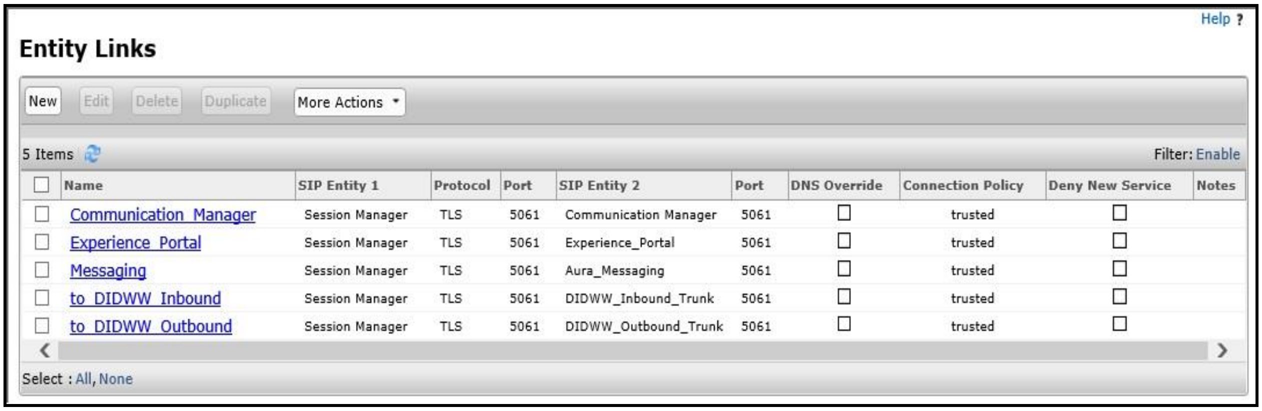
Loop Detection Mode:	On
Loop Count Threshold:	5
Loop Detection Interval (in msec):	200

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.



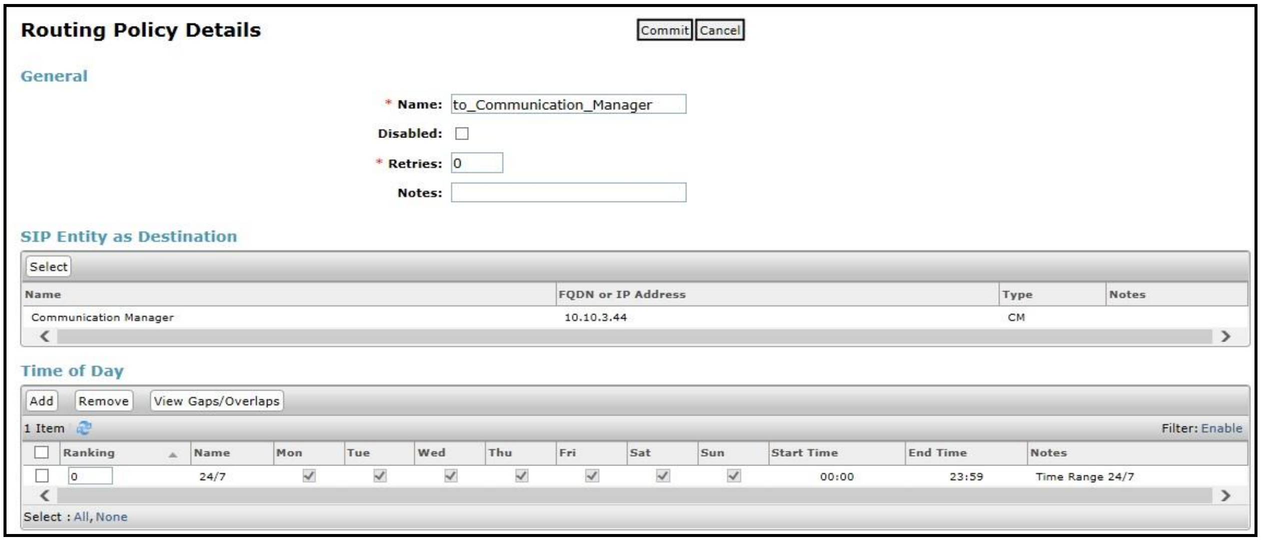
□	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	DNS Override	Connection Policy	Deny New Service	Notes
□	Communication_Manager	Session Manager	TLS	5061	Communication Manager	5061	□	trusted	□	
□	Experience_Portal	Session Manager	TLS	5061	Experience_Portal	5061	□	trusted	□	
□	Messaging	Session Manager	TLS	5061	Aura_Messaging	5061	□	trusted	□	
□	to_DIDWW_Inbound	Session Manager	TLS	5061	DIDWW_Inbound_Trunk	5061	□	trusted	□	
□	to_DIDWW_Outbound	Session Manager	TLS	5061	DIDWW_Outbound_Trunk	5061	□	trusted	□	

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 calls inbound from the SIP trunk to Communication Manager.



Routing Policy Details

General

* Name:	to_Communication_Manager
Disabled:	□
* Retries:	0
Notes:	

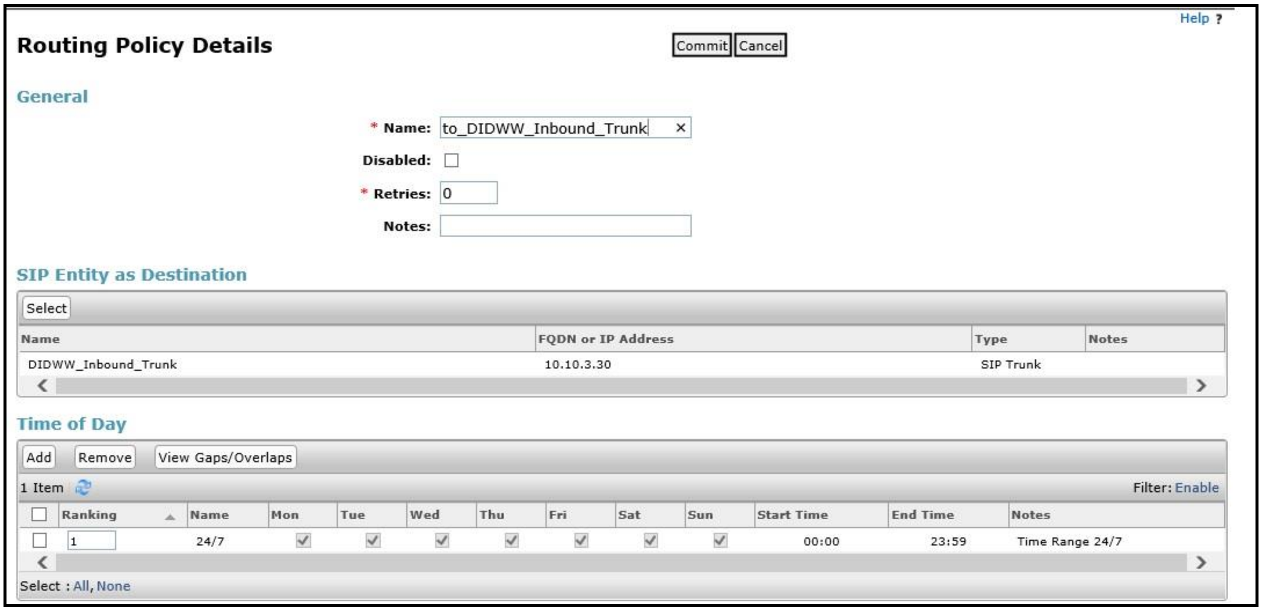
SIP Entity as Destination

Name	FQDN or IP Address	Type	Notes
Communication Manager	10.10.3.44	CM	

Time of Day

□	Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
□	0	24/7	☑	☑	☑	☑	☑	☑	☑	00:00	23:59	Time Range 24/7

The following screen shows the routing policy for Avaya SBCE for the DIDWW Inbound SIP trunk.



Routing Policy Details

General

* Name: to_DIDWW_Inbound_Trunk

Disabled: □

* Retries: 0

Notes:

SIP Entity as Destination

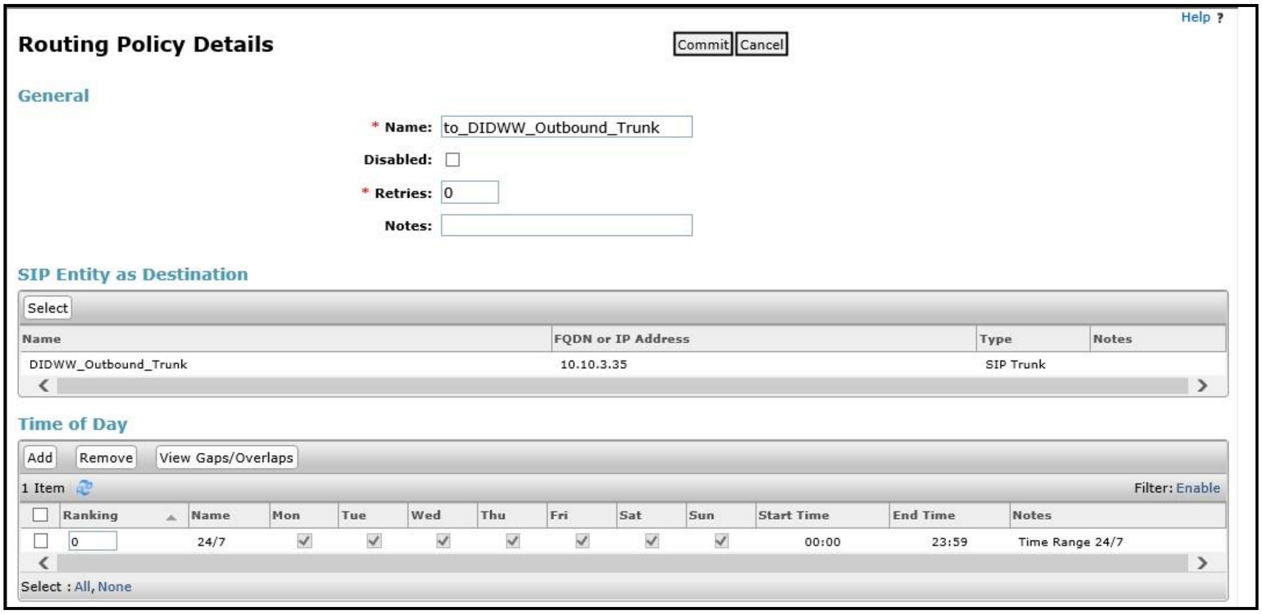
Name	FQDN or IP Address	Type	Notes
DIDWW_Inbound_Trunk	10.10.3.30	SIP Trunk	

Time of Day

□	Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
□	1	24/7	☑	☑	☑	☑	☑	☑	☑	00:00	23:59	Time Range 24/7

Select : All, None

The following screen shows the routing policy for Avaya SBCE for the DIDWW Outbound SIP trunk.



Routing Policy Details

General

* Name: to_DIDWW_Outbound_Trunk

Disabled: □

* Retries: 0

Notes:

SIP Entity as Destination

Name	FQDN or IP Address	Type	Notes
DIDWW_Outbound_Trunk	10.10.3.35	SIP Trunk	

Time of Day

□	Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
□	0	24/7	☑	☑	☑	☑	☑	☑	☑	00:00	23:59	Time Range 24/7

Select : All, None

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).

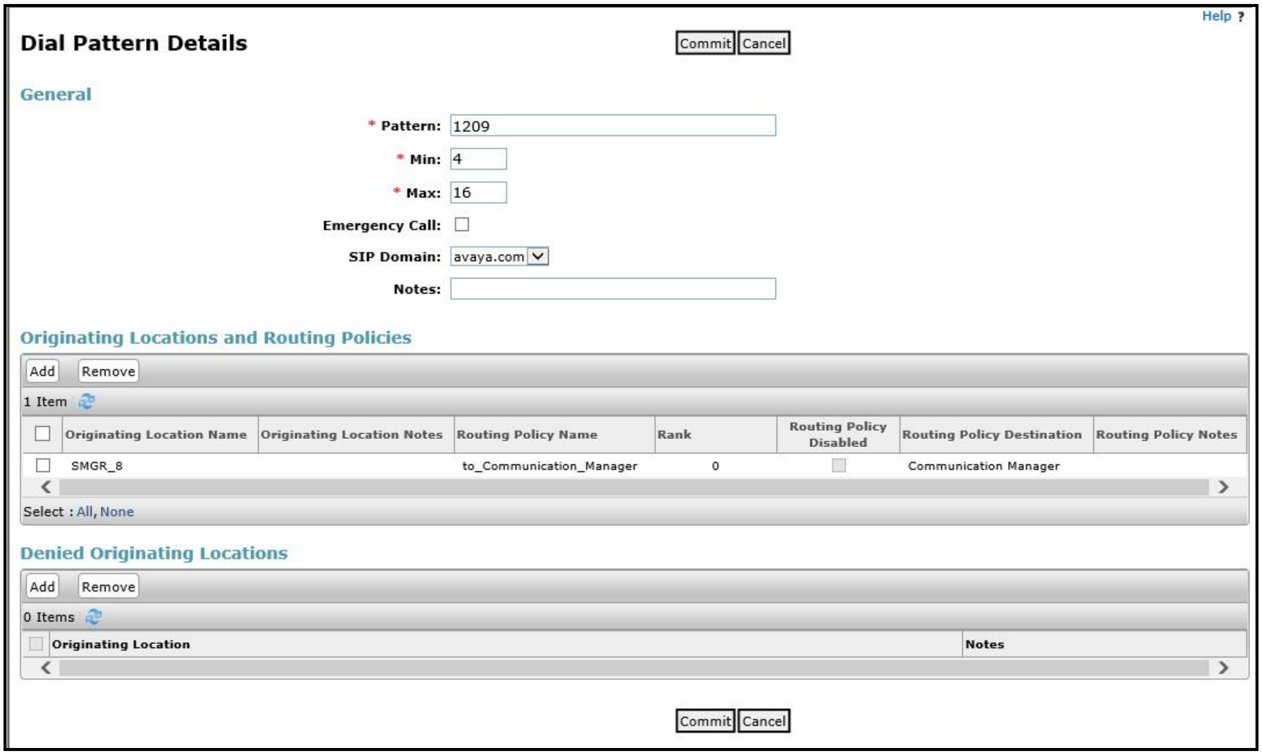
Under **General**:

- 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 the dial pattern configured towards Communication Manager.



Dial Pattern Details

General

* Pattern: 1209

* Min: 4

* Max: 16

Emergency Call: □

SIP Domain: avaya.com

Notes:

Originating Locations and Routing Policies

1 Item

□	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
□	SMGR_8		to_Communication_Manager	0	□	Communication Manager	

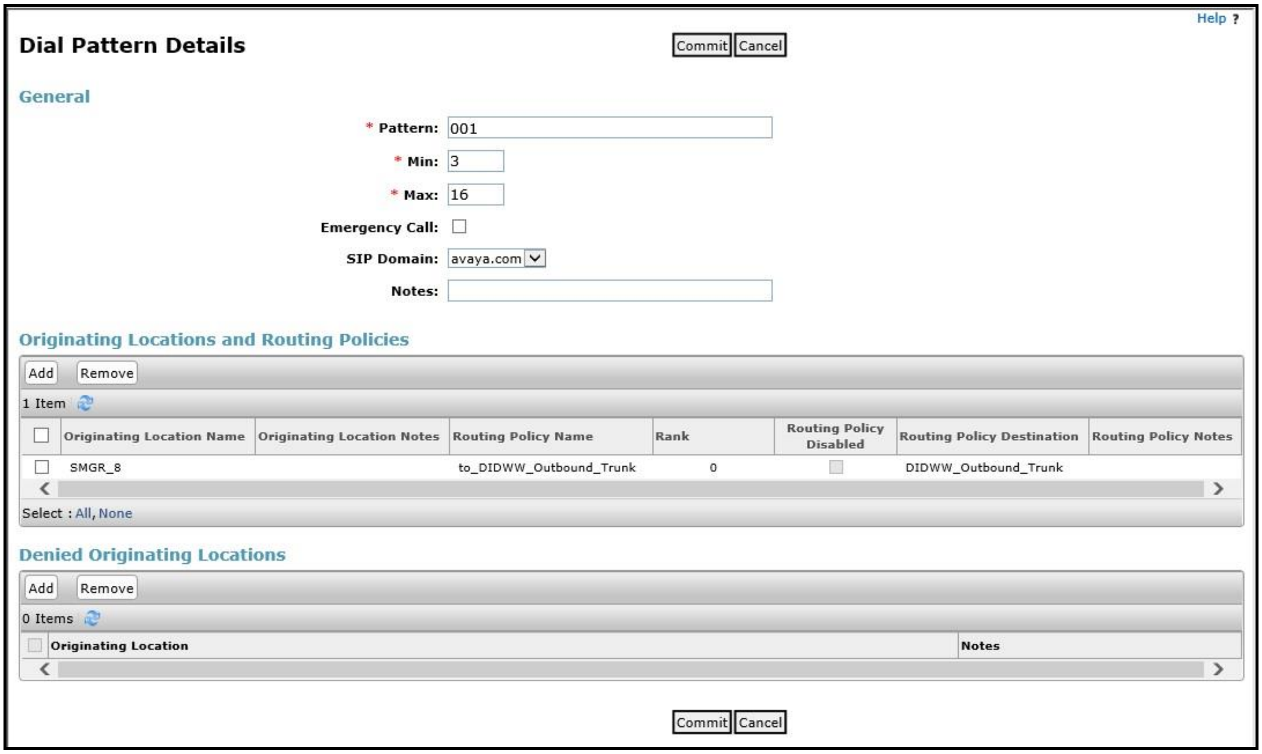
Select : All, None

Denied Originating Locations

0 Items

□	Originating Location	Notes

The following screen shows an example dial pattern for outbound traffic from Communication Manager towards the DIDWW Outbound SIP trunk.



Dial Pattern Details

General

* Pattern:	001
* Min:	3
* Max:	16
Emergency Call:	□
SIP Domain:	avaya.com
Notes:	

Originating Locations and Routing Policies

1 Item

□	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
□	SMGR_8		to_DIDWW_Outbound_Trunk	0	□	DIDWW_Outbound_Trunk	

Select : All, None

Denied Originating Locations

0 Items

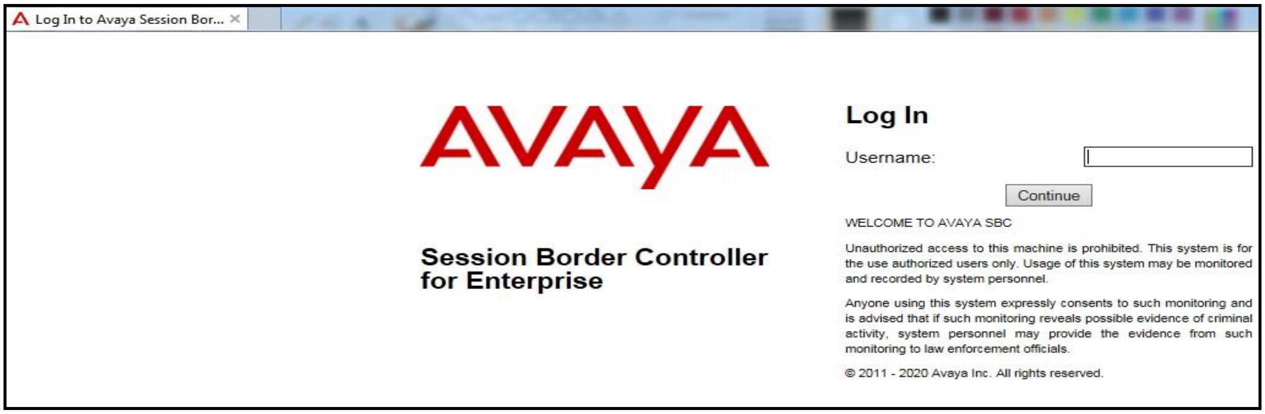
□	Originating Location	Notes

7. Configure Avaya Session Border Controller for Enterprise

This section describes the configuration of the Session Border Controller for Enterprise (Avaya SBCE). The Avaya SBCE provides security and manipulation of signalling to provide 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.

7.1. Access Avaya Session Border Controller for Enterprise

Access the Avaya SBCE 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**.


Log In to Avaya Session Bor...
AVAYA
Session Border Controller for Enterprise
Log In
Username:
Continue

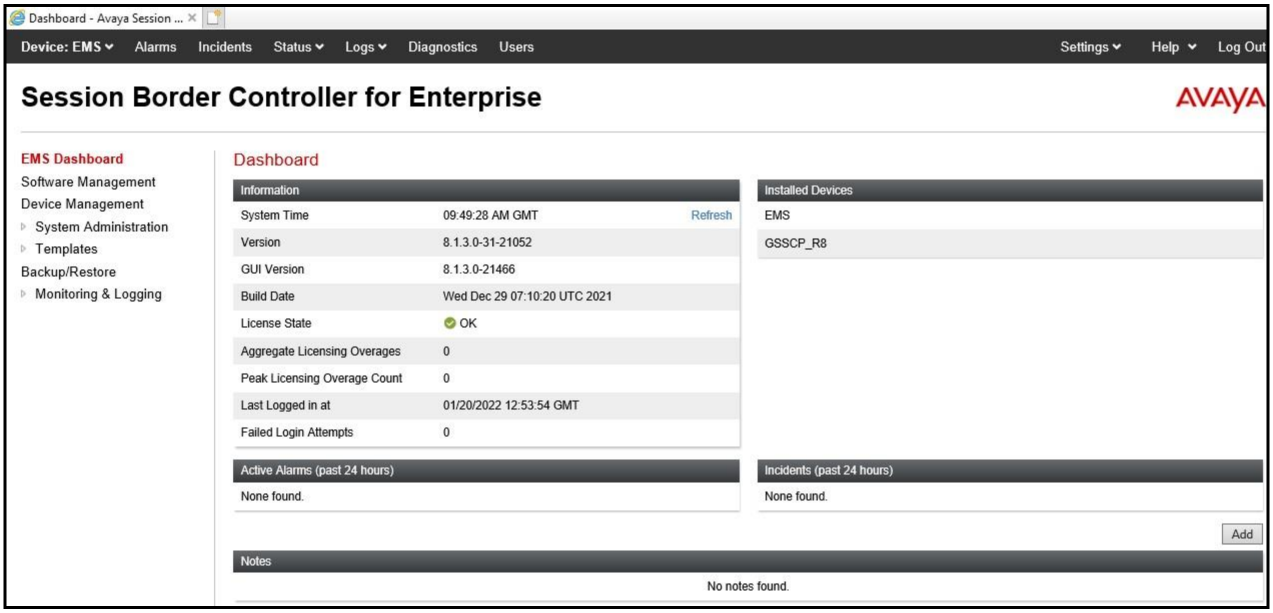
WELCOME TO AVAYA SBC

Unauthorized access to this machine is prohibited. This system is for the use authorized users only. Usage of this system may be monitored and recorded by system personnel.

Anyone using this system expressly consents to such monitoring and is advised that if such monitoring reveals possible evidence of criminal activity, system personnel may provide the evidence from such monitoring to law enforcement officials.

© 2011 - 2020 Avaya Inc. All rights reserved.

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_R8** is used as a starting point for all configuration of the Avaya SBCE.



Information	
System Time	09:49:28 AM GMT
Version	8.1.3.0-31-21052
GUI Version	8.1.3.0-21466
Build Date	Wed Dec 29 07:10:20 UTC 2021
License State	✓ OK
Aggregate Licensing Overages	0
Peak Licensing Overage Count	0
Last Logged in at	01/20/2022 12:53:54 GMT
Failed Login Attempts	0

Installed Devices
EMS
GSSCP_R8

Active Alarms (past 24 hours)

None found.

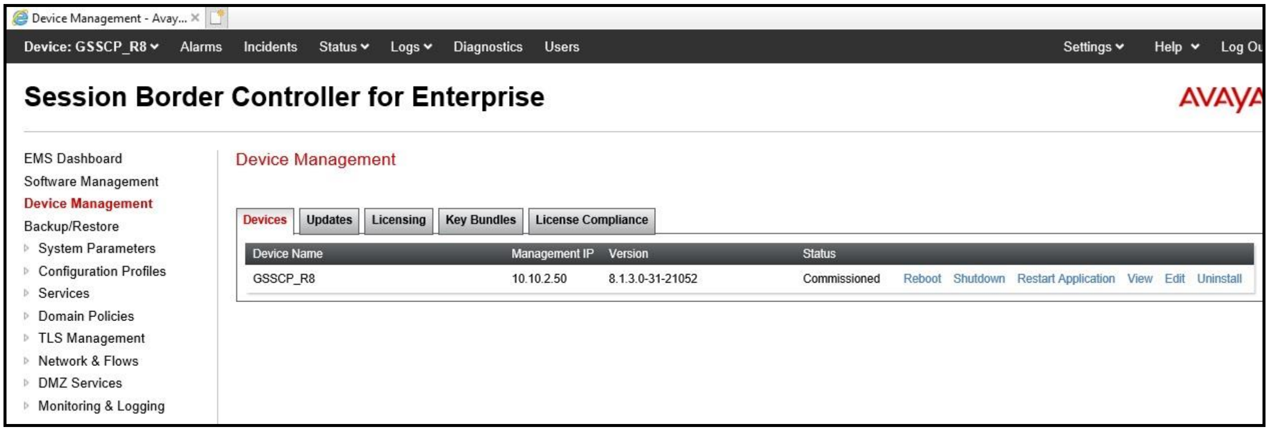
Incidents (past 24 hours)

None found.

Notes

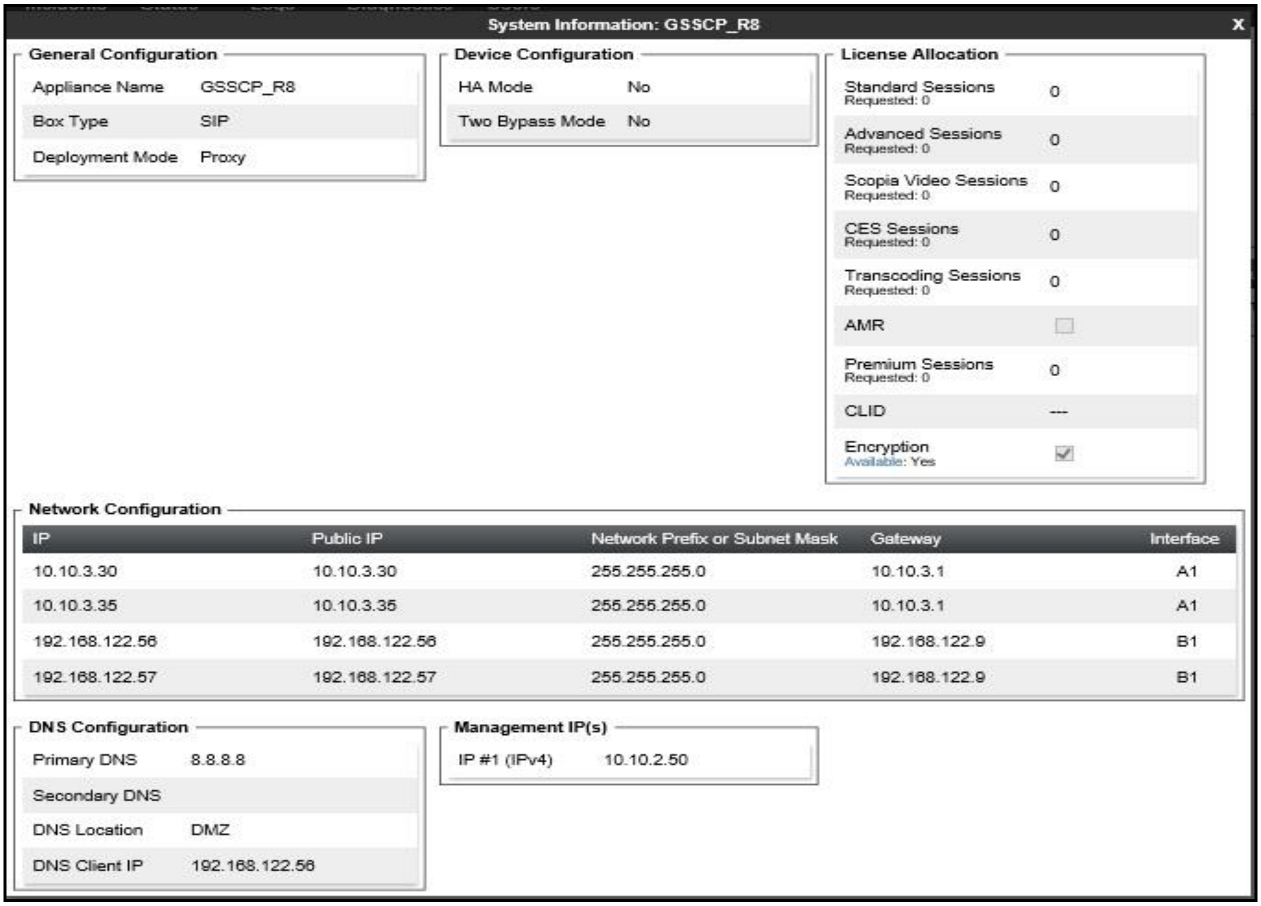
No notes found.

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_R8** is shown. To view the configuration of this device, click **View** (the third option from the right).



Device Name	Management IP	Version	Status	
GSSCP_R8	10.10.2.50	8.1.3.0-31-21052	Commissioned	Reboot Shutdown Restart Application View Edit Uninstall

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



General Configuration

Appliance Name	GSSCP_R8
Box Type	SIP
Deployment Mode	Proxy

Device Configuration

HA Mode	No
Two Bypass Mode	No

License Allocation

Standard Sessions Requested: 0	0
Advanced Sessions Requested: 0	0
Scopia Video Sessions Requested: 0	0
CES Sessions Requested: 0	0
Transcoding Sessions Requested: 0	0
AMR	□
Premium Sessions Requested: 0	0
CLID	---
Encryption Available: Yes	☑

Network Configuration

IP	Public IP	Network Prefix or Subnet Mask	Gateway	Interface
10.10.3.30	10.10.3.30	255.255.255.0	10.10.3.1	A1
10.10.3.35	10.10.3.35	255.255.255.0	10.10.3.1	A1
192.168.122.56	192.168.122.56	255.255.255.0	192.168.122.9	B1
192.168.122.57	192.168.122.57	255.255.255.0	192.168.122.9	B1

DNS Configuration

Primary DNS	8.8.8.8
Secondary DNS	
DNS Location	DMZ
DNS Client IP	192.168.122.56

Management IP(s)

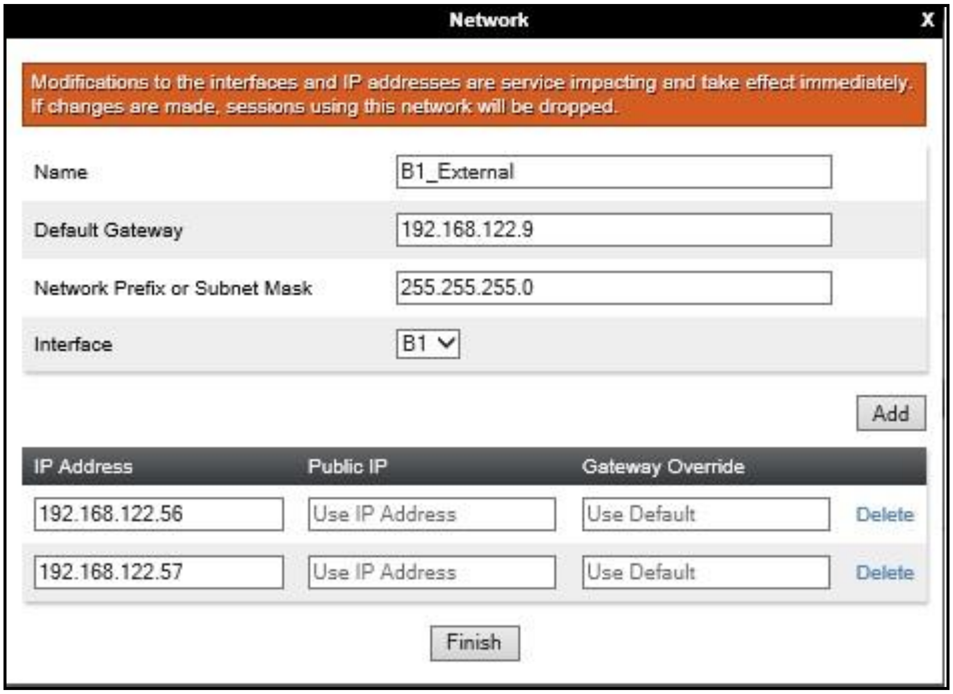
IP #1 (IPv4)	10.10.2.50

7.2. Define Network Management

Network information is required on the Avaya SBCE 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.

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 SBCE used for the DIDWW Inbound SIP trunk in the **IP Address** field and leave the **Public IP** and **Gateway Override** fields blank. In this case, IP address **192.168.122.56** was used.
- Click on **Add** again.
- Enter the external IP address of the Avaya SBCE used for the DIDWW Outbound SIP trunk in the **IP Address** field and leave the **Public IP** and **Gateway Override** fields blank. In this case, IP address **192.168.122.57** was used.
- Click on **Finish** to complete the interface definition.



Network

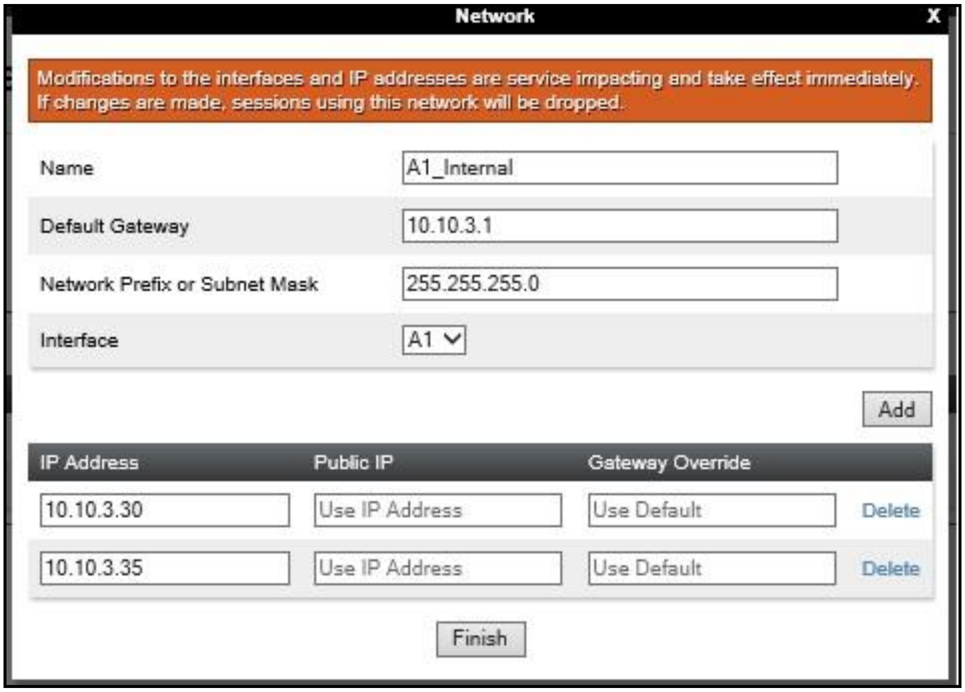
Modifications to the interfaces and IP addresses are service impacting and take effect immediately. If changes are made, sessions using this network will be dropped.

Name	B1_External
Default Gateway	192.168.122.9
Network Prefix or Subnet Mask	255.255.255.0
Interface	B1

IP Address	Public IP	Gateway Override	
192.168.122.56	Use IP Address	Use Default	Delete
192.168.122.57	Use IP Address	Use Default	Delete

Click on **Add** to define the internal interfaces or Edit if it was defined during installation of the Avaya SBCE. 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 SBCE used for the DIDWW Inbound SIP trunk in the **IP Address** field and leave the **Public IP** and **Gateway Override** fields blank. In this case, IP address **10.10.3.30** was used.
- Click on **Add** again.
- Enter the internal IP address of the Avaya SBCE used for the DIDWW Outbound SIP trunk in the **IP Address** field and leave the **Public IP** and **Gateway Override** fields blank. In this case, IP address **10.10.3.35** was used.
- Click on **Finish** to complete the interface definition.

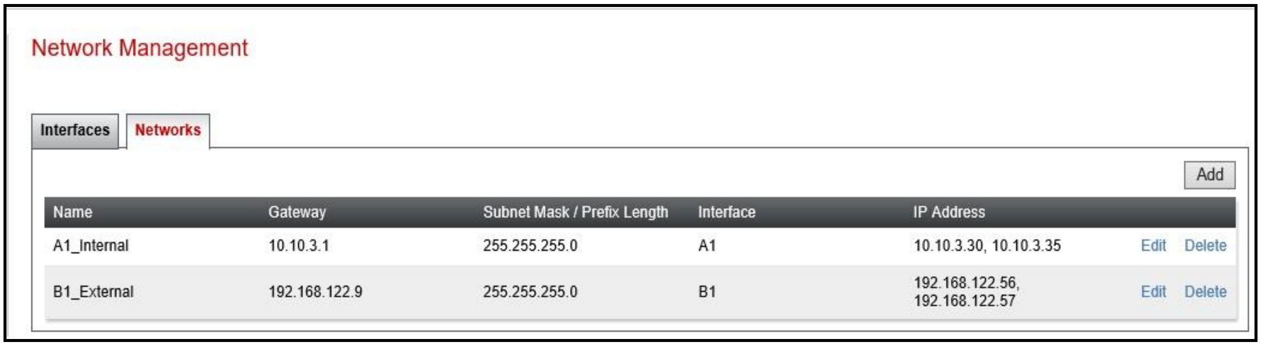


Modifications to the interfaces and IP addresses are service impacting and take effect immediately. If changes are made, sessions using this network will be dropped.

Name	A1_Internal
Default Gateway	10.10.3.1
Network Prefix or Subnet Mask	255.255.255.0
Interface	A1

IP Address	Public IP	Gateway Override	
10.10.3.30	Use IP Address	Use Default	Delete
10.10.3.35	Use IP Address	Use Default	Delete

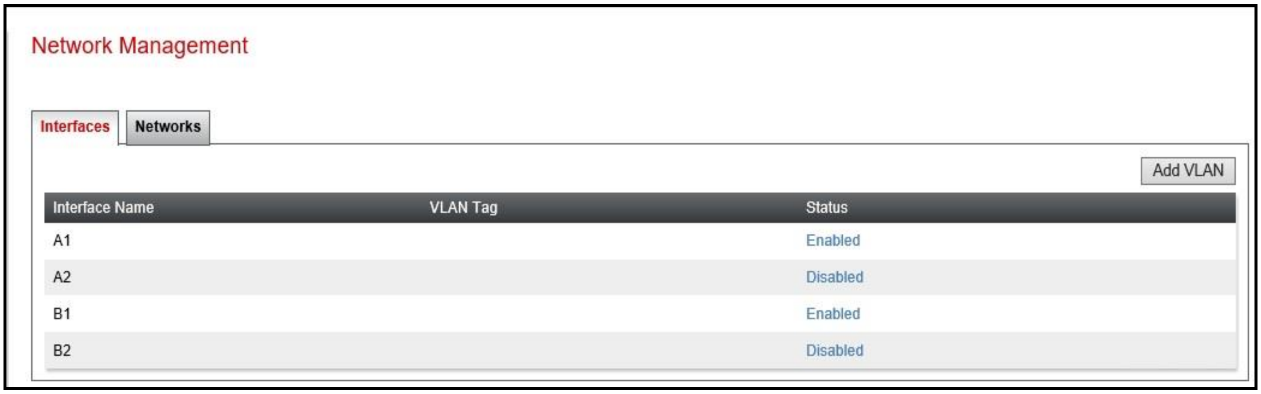
The following screenshot shows the completed Network Management configuration:



Network Management

Name	Gateway	Subnet Mask / Prefix Length	Interface	IP Address		
A1_Internal	10.10.3.1	255.255.255.0	A1	10.10.3.30, 10.10.3.35	Edit	Delete
B1_External	192.168.122.9	255.255.255.0	B1	192.168.122.56, 192.168.122.57	Edit	Delete

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



Network Management

Interface Name	VLAN Tag	Status
A1		Enabled
A2		Disabled
B1		Enabled
B2		Disabled

Note: to ensure that the Avaya SBCE 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.

7.3. Define TLS Profiles

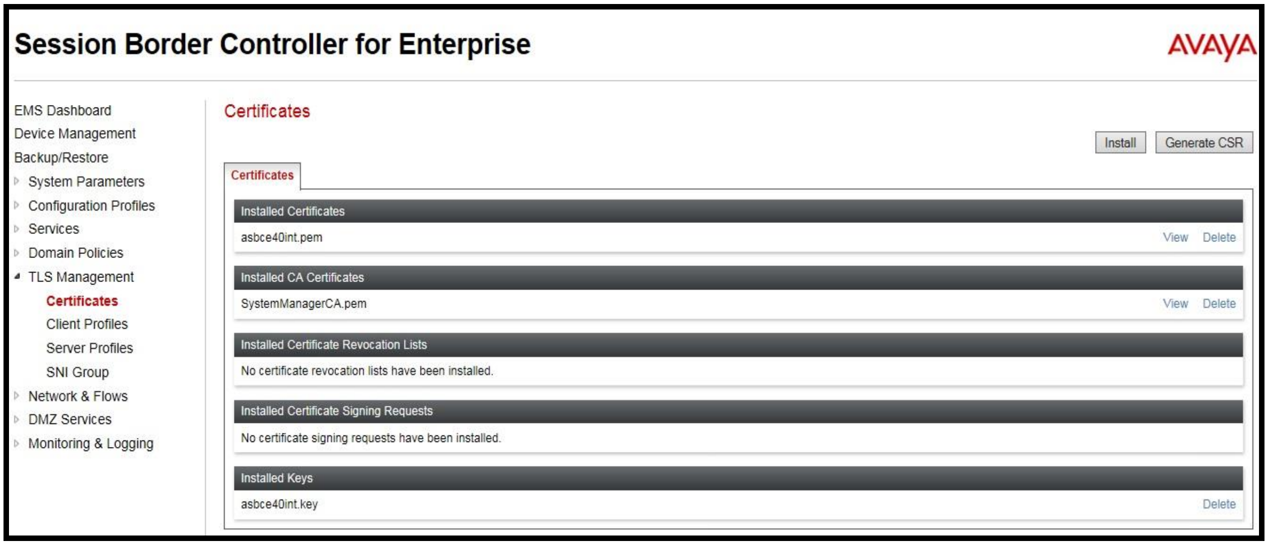
For the compliance test, TLS transport is used for signalling on the SIP trunk between Session Manager and the Avaya SBCE. Compliance testing was done using identity certificates signed by a local certificate authority. The generation and installation of these certificates are beyond the scope of these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

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

7.3.1. Certificates

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

- Verify that an Avaya SBCE identity certificate (**asbce40int.pem**) is present under **Installed Certificates**.
- Verify that certificate authority root certificate (**SystemManagerCA.pem**) is present under **Installed CA certificates**.
- Verify that private key associated with the identity certificate (**asbce40int.key**) is present under **Installed Keys**.



Certificates

Certificates

Installed Certificates

asbce40int.pem

Installed CA Certificates

SystemManagerCA.pem

Installed Certificate Revocation Lists

No certificate revocation lists have been installed.

Installed Certificate Signing Requests

No certificate signing requests have been installed.

Installed Keys

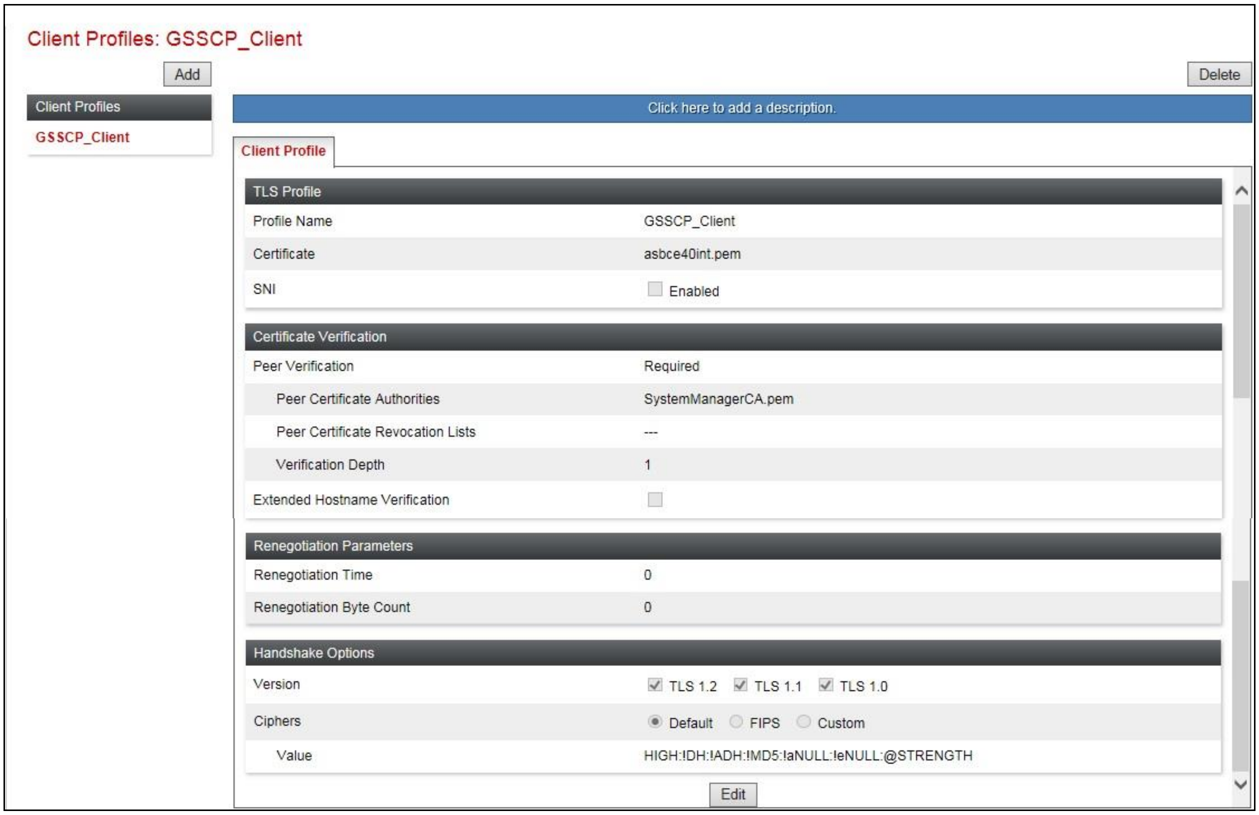
asbce40int.key

7.3.2. 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 **asbce40int.pem** 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).



Client Profiles: GSSCP_Client

TLS Profile	
Profile Name	GSSCP_Client
Certificate	asbce40int.pem
SNI	□ Enabled

Certificate Verification	
Peer Verification	Required
Peer Certificate Authorities	SystemManagerCA.pem
Peer Certificate Revocation Lists	---
Verification Depth	1
Extended Hostname Verification	□

Renegotiation Parameters	
Renegotiation Time	0
Renegotiation Byte Count	0

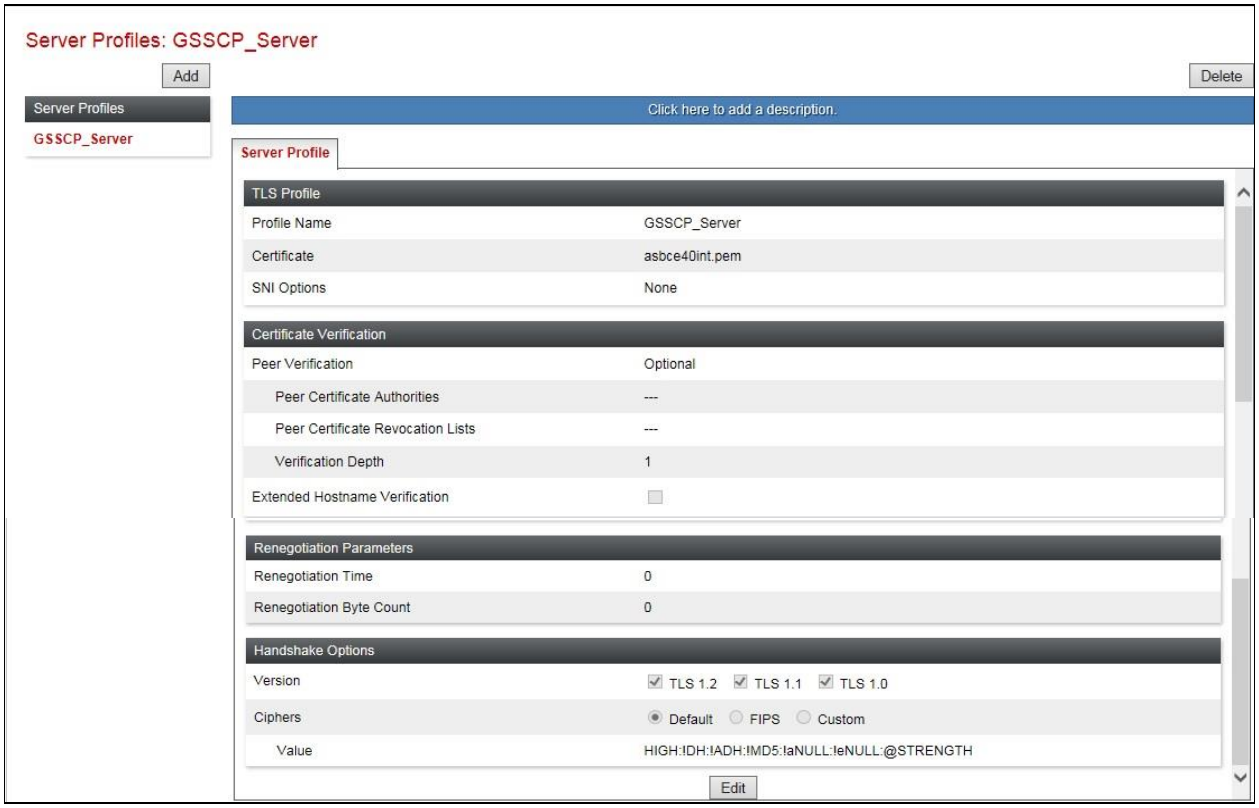
Handshake Options	
Version	☑ TLS 1.2 ☑ TLS 1.1 ☑ TLS 1.0
Ciphers	● Default ○ FIPS ○ Custom
Value	HIGH:!DH:!ADH:!MD5:!aNULL:!eNULL:@STRENGTH

7.3.3. 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 **asbce40int.pem** 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).



Server Profiles: GSSCP_Server

TLS Profile	
Profile Name	GSSCP_Server
Certificate	asbce40int.pem
SNI Options	None

Certificate Verification	
Peer Verification	Optional
Peer Certificate Authorities	---
Peer Certificate Revocation Lists	---
Verification Depth	1
Extended Hostname Verification	□

Renegotiation Parameters	
Renegotiation Time	0
Renegotiation Byte Count	0

Handshake Options	
Version	☑ TLS 1.2 ☑ TLS 1.1 ☑ TLS 1.0
Ciphers	● Default ○ FIPS ○ Custom
Value	HIGH:!DH:!ADH:!MD5:!aNULL:!eNULL:@STRENGTH

7.4. Define Interfaces

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

7.4.1. Signalling Interfaces

To define the signalling interfaces on the Avaya SBCE, 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 to be used in the server flow for DIDWW Inbound SIP trunk:

- Select **Add** and enter details of the first 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 one of the **A1_Internal** signalling interface IP addresses defined in **Section 7.2**. IP address **10.10.3.30** was used in the test configuration.
- Select **TLS** port number, **5061** is used for Session Manager.
- Select a **TLS Profile** defined in **Section 7.3.3** from the drop-down menu.

To enter details of transport protocol and ports for the SIP signalling on internal interface to be used in the server flow for DIDWW Outbound SIP trunk:

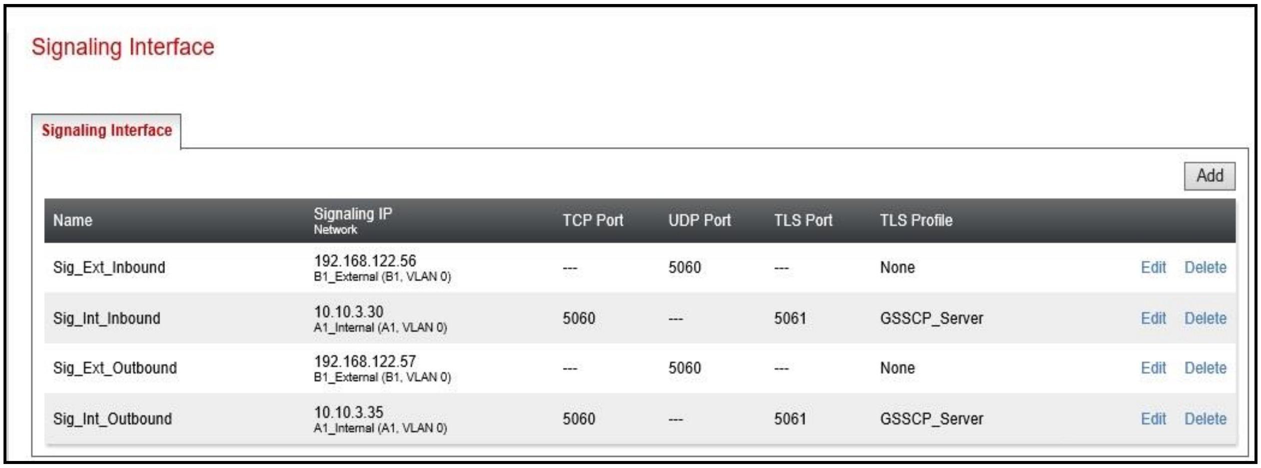
- Select **Add** and enter details of the internal signalling interface in the pop-up menu (not shown).
- In the **Name** field enter a descriptive name for interface.
- For **Signaling IP**, select the other **A1_internal** signalling interface IP address defined in **Section 7.2**. IP address **10.10.3.35** was used in the test configuration.
- Select **TLS** port number, **5061** is used for Session Manager.
- Select a **TLS Profile** defined in **Section 7.3.3** from the drop-down menu.

To enter details of transport protocol and ports for the SIP signalling on the external interface to be used in the server flow for DIDWW Inbound SIP trunk:

- Select **Add** and enter details of the first external 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 **B1_external** signalling interface IP address defined in **Section 7.2**. IP address **192.168.122.56** was used in the test configuration.
- Select **UDP** port number, **5060** is used for the DIDWW Inbound SIP trunk.
- Click **Finish**.

To enter details of transport protocol and ports for the SIP signalling on external interface to be used in the server flow for DIDWW Outbound SIP trunk:

- Select **Add** and enter details of the first external 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 **B1_external** signalling interface IP address defined in **Section 7.2**. IP address **192.168.122.57** was used in the test configuration.
- Select **UDP** port number, **5060** is used for the DIDWW Outbound SIP trunk.
- Click **Finish**.



Name	Signaling IP Network	TCP Port	UDP Port	TLS Port	TLS Profile		
Sig_Ext_Inbound	192.168.122.56 B1_External (B1, VLAN 0)	---	5060	---	None	Edit	Delete
Sig_Int_Inbound	10.10.3.30 A1_Internal (A1, VLAN 0)	5060	---	5061	GSSCP_Server	Edit	Delete
Sig_Ext_Outbound	192.168.122.57 B1_External (B1, VLAN 0)	---	5060	---	None	Edit	Delete
Sig_Int_Outbound	10.10.3.35 A1_Internal (A1, VLAN 0)	5060	---	5061	GSSCP_Server	Edit	Delete

7.4.2. Media Interfaces

To define the media interfaces on the Avaya SBCE, 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 on the internal interface to be used in the server flow for DIDWW Inbound SIP trunk:

- Select **Add** 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 for DIDWW Inbound SIP trunk defined in **Section 7.2**. IP address **10.10.3.30** was used in the test configuration.
- Select **Port Range**, enter **35000-40000**.
- Click **Finish**.

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

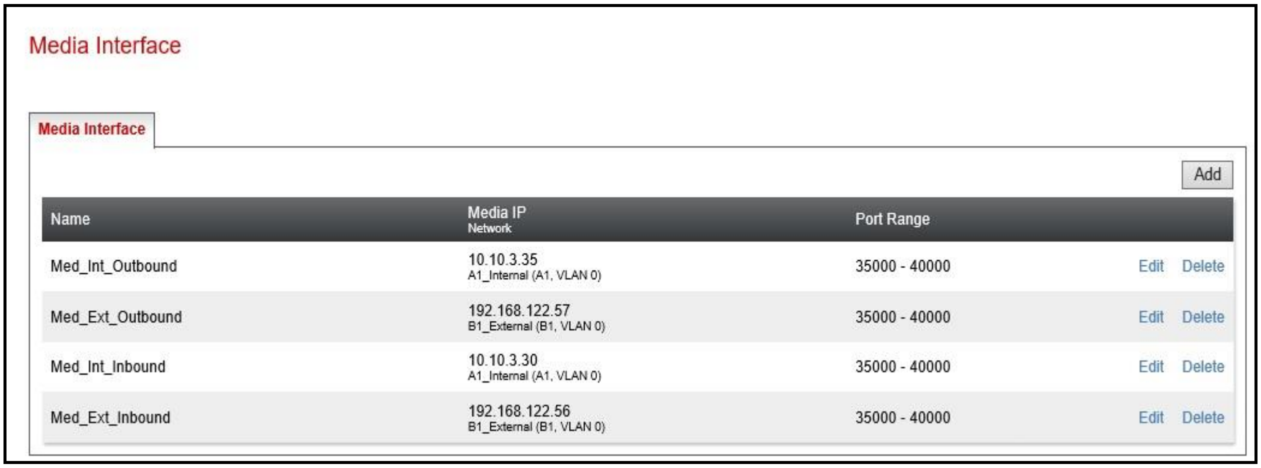
- Select **Add** and enter details in the pop-up menu.
- In the **Name** field enter a descriptive name for the internal media interface.
- For **Media IP**, select the **A1_Internal** media interface IP address for DIDWW Outbound SIP trunk defined in **Section 7.2**. IP address **10.10.3.35** was used in the test configuration.
- Select **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 for DIDWW Inbound SIP trunk:

- Select **Add** 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 for DIDWW Inbound SIP trunk defined in **Section 7.2**. IP address **192.168.122.56** was used in the test configuration.
- Select **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 for DIDWW Outbound SIP trunk:

- Select **Add** 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 for DIDWW Outbound SIP trunk defined in **Section 7.2**. IP address **192.168.122.57** was used in the test configuration.
- Select **Port Range**, enter **35000-40000**.
- Click **Finish**.



Name	Media IP Network	Port Range		
Med_Int_Outbound	10.10.3.35 A1_Internal (A1, VLAN 0)	35000 - 40000	Edit	Delete
Med_Ext_Outbound	192.168.122.57 B1_External (B1, VLAN 0)	35000 - 40000	Edit	Delete
Med_Int_Inbound	10.10.3.30 A1_Internal (A1, VLAN 0)	35000 - 40000	Edit	Delete
Med_Ext_Inbound	192.168.122.56 B1_External (B1, VLAN 0)	35000 - 40000	Edit	Delete

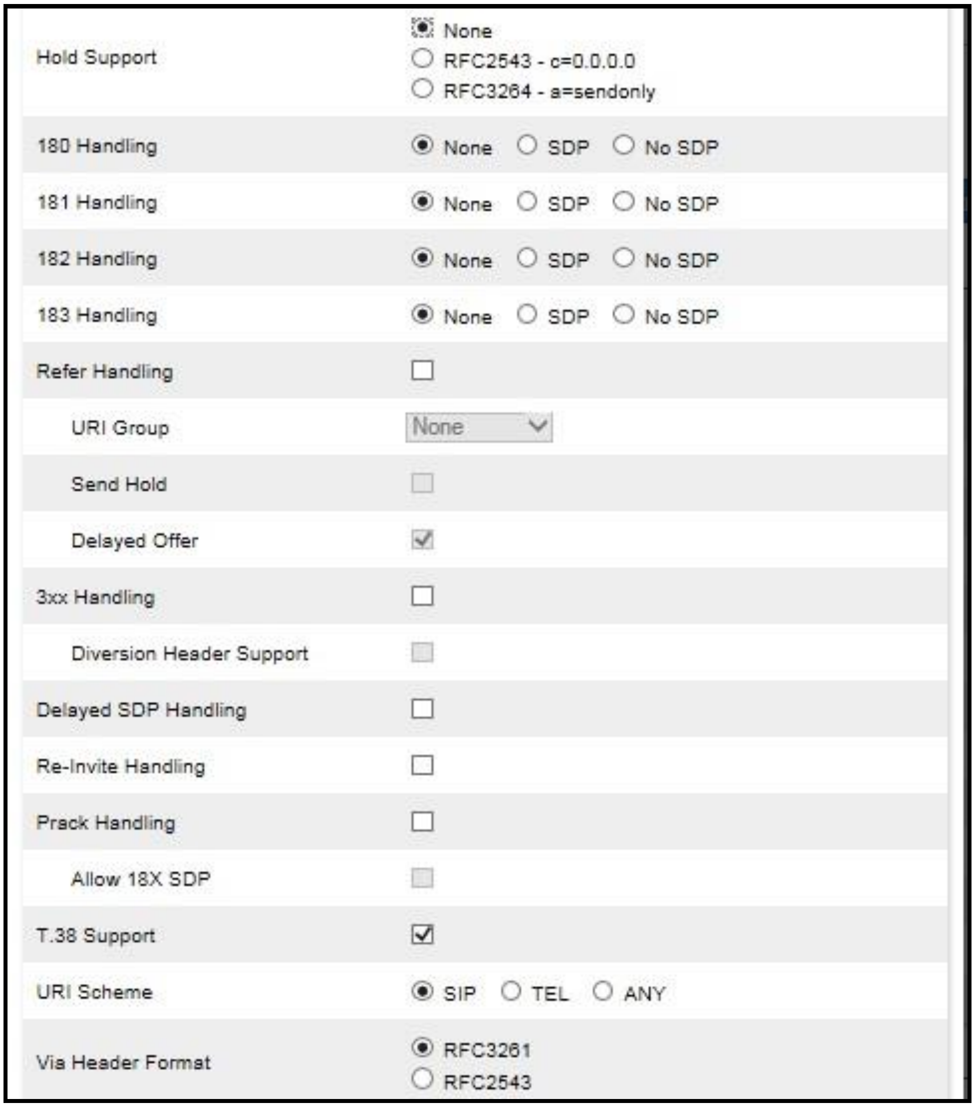
7.5. Define Server Interworking

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

7.5.1. 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**.
- All other options on the **General** Tab can be left at default.

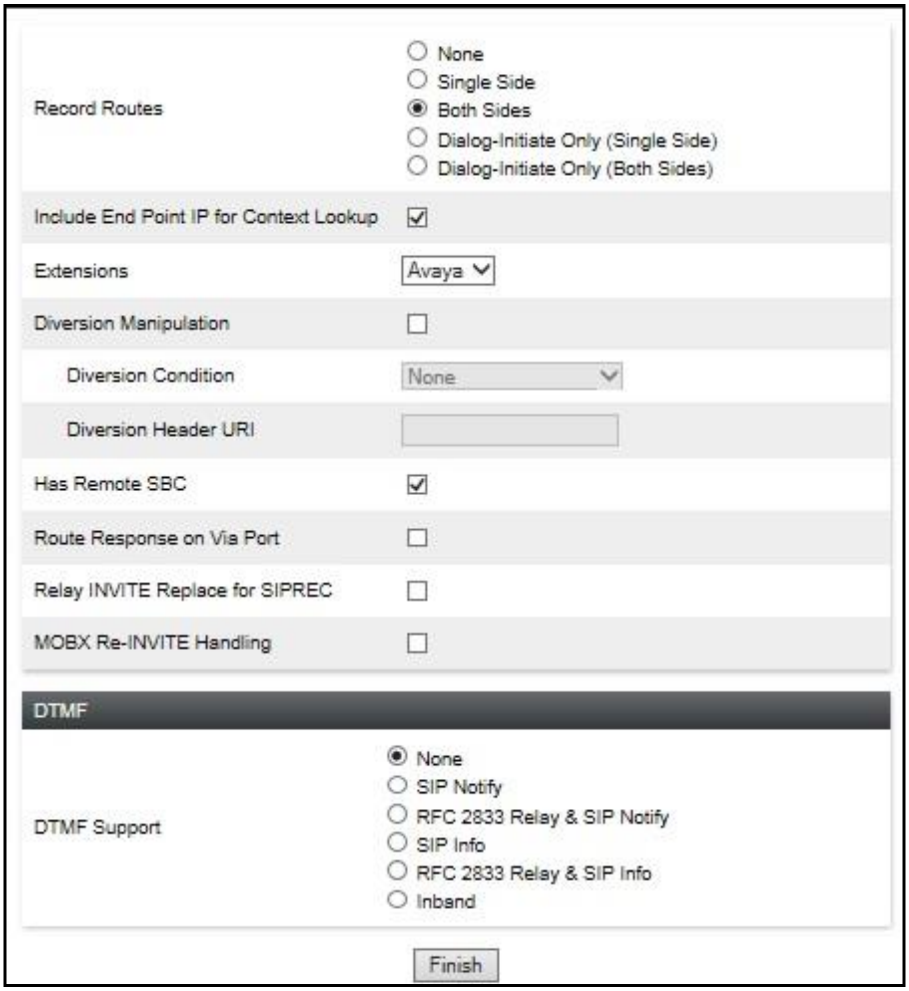


Hold Support	● None ○ RFC2543 - c=0.0.0.0 ○ RFC3264 - a=sendonly
180 Handling	● None ○ SDP ○ No SDP
181 Handling	● None ○ SDP ○ No SDP
182 Handling	● None ○ SDP ○ No SDP
183 Handling	● None ○ SDP ○ No SDP
Refer Handling	□
URI Group	None
Send Hold	□
Delayed Offer	☑
3xx Handling	□
Diversion Header Support	□
Delayed SDP Handling	□
Re-Invite Handling	□
Prack Handling	□
Allow 18X SDP	□
T.38 Support	☑
URI Scheme	● SIP ○ TEL ○ ANY
Via Header Format	● RFC3261 ○ RFC2543

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**.



Record Routes	○ None ○ Single Side ● Both Sides ○ Dialog-Initiate Only (Single Side) ○ Dialog-Initiate Only (Both Sides)
Include End Point IP for Context Lookup	☑
Extensions	Avaya
Diversion Manipulation	□
Diversion Condition	None
Diversion Header URI	
Has Remote SBC	☑
Route Response on Via Port	□
Relay INVITE Replace for SIPREC	□
MOBX Re-INVITE Handling	□

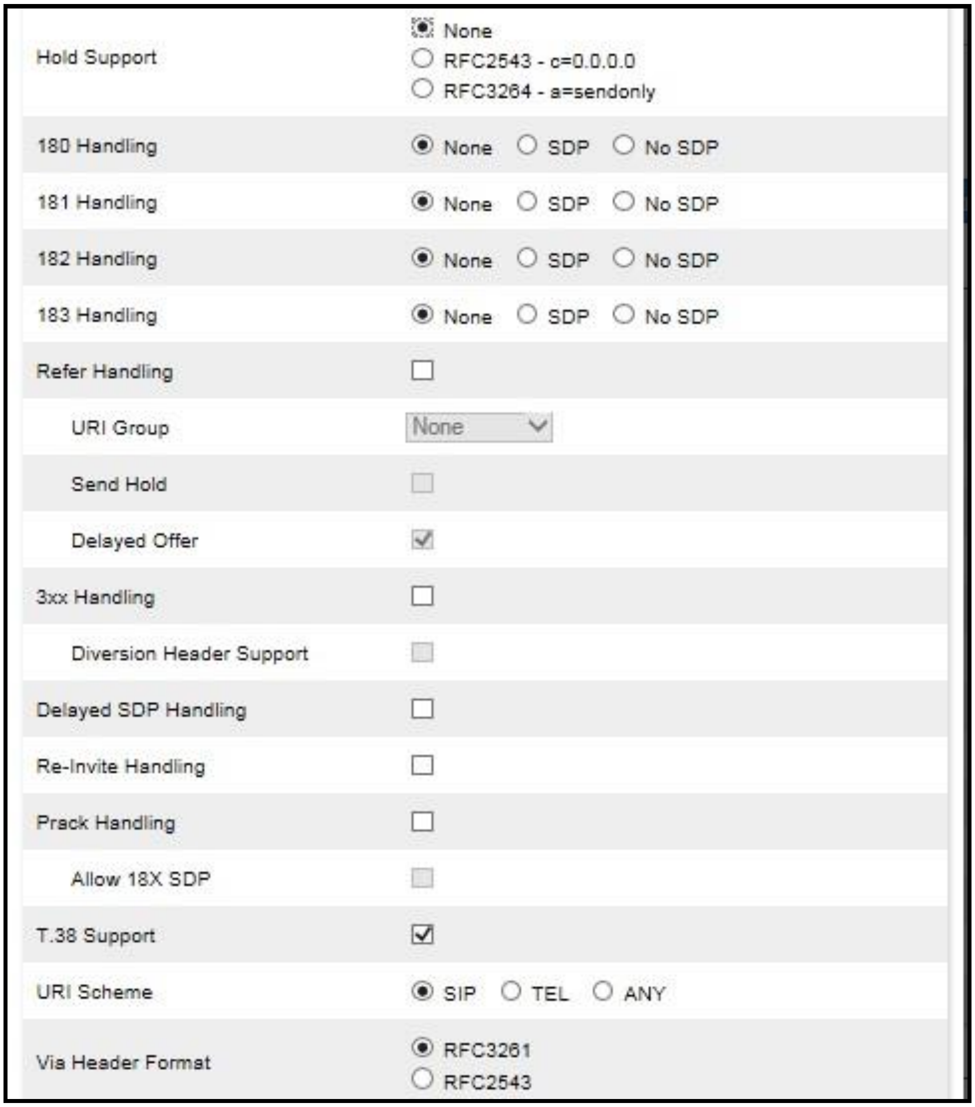
DTMF

DTMF Support	● None ○ SIP Notify ○ RFC 2833 Relay & SIP Notify ○ SIP Info ○ RFC 2833 Relay & SIP Info ○ Inband

7.5.2. Server Interworking – DIDWW

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 **DIDWW** and click **Next** (Not Shown).
- Check **Hold Support = None**.
- Check **T.38 Support**.
- All other options on the **General** Tab can be left at default.

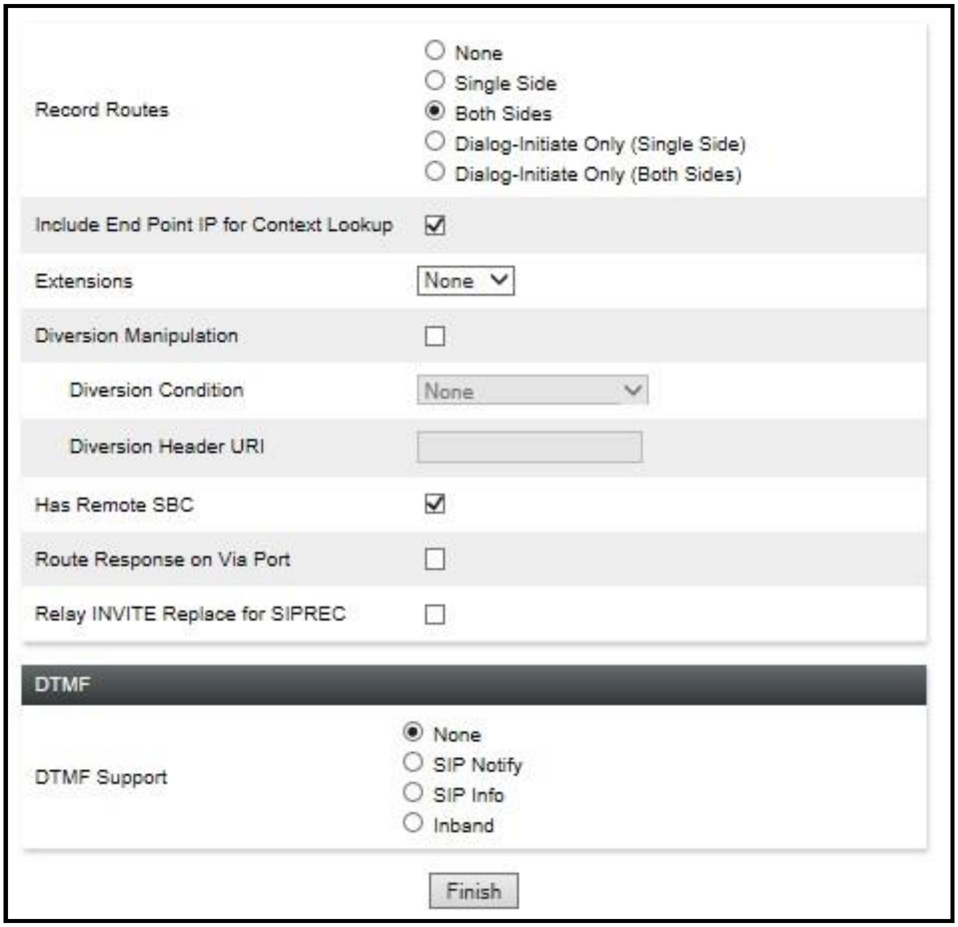


Hold Support	● None ○ RFC2543 - c=0.0.0.0 ○ RFC3264 - a=sendonly
180 Handling	● None ○ SDP ○ No SDP
181 Handling	● None ○ SDP ○ No SDP
182 Handling	● None ○ SDP ○ No SDP
183 Handling	● None ○ SDP ○ No SDP
Refer Handling	□
URI Group	None
Send Hold	□
Delayed Offer	☑
3xx Handling	□
Diversion Header Support	□
Delayed SDP Handling	□
Re-Invite Handling	□
Prack Handling	□
Allow 18X SDP	□
T.38 Support	☑
URI Scheme	● SIP ○ TEL ○ ANY
Via Header Format	● RFC3261 ○ RFC2543

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**.



Record Routes	○ None ○ Single Side ● Both Sides ○ Dialog-Initiate Only (Single Side) ○ Dialog-Initiate Only (Both Sides)
Include End Point IP for Context Lookup	☑
Extensions	None
Diversion Manipulation	□
Diversion Condition	None
Diversion Header URI	
Has Remote SBC	☑
Route Response on Via Port	□
Relay INVITE Replace for SIPREC	□

DTMF

DTMF Support	● None ○ SIP Notify ○ SIP Info ○ Inband

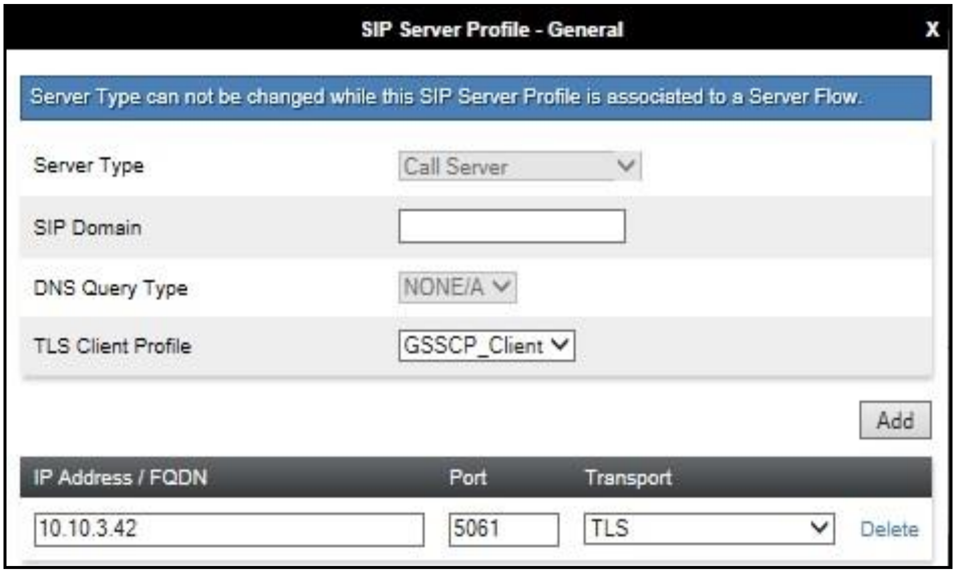
7.6. Define Servers

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

7.6.1. 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 7.3.2**.
- 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.



SIP Server Profile - General

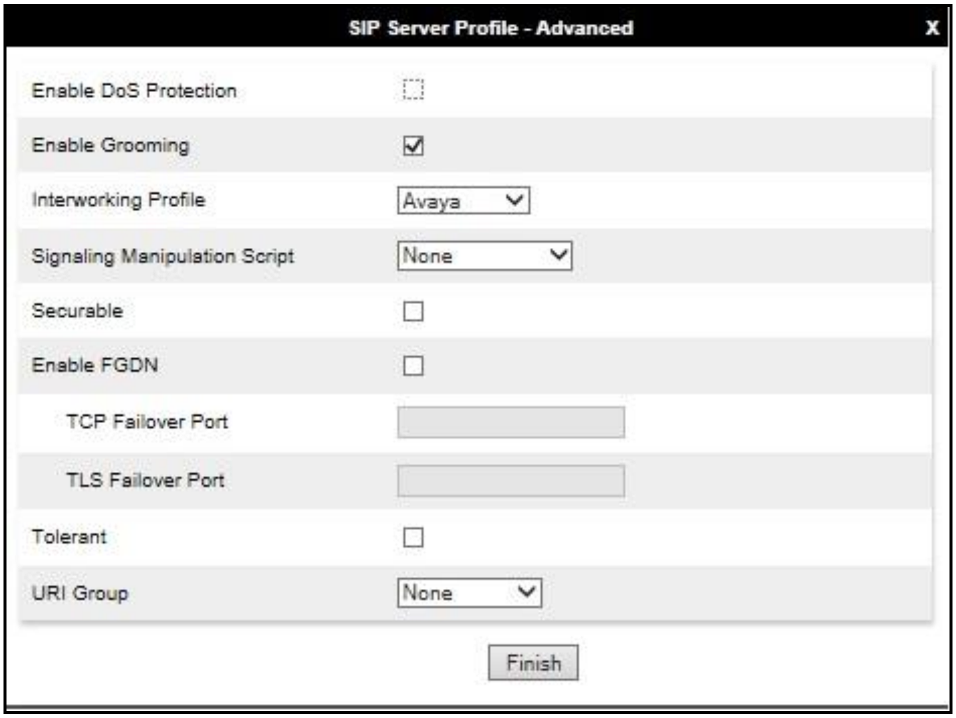
Server Type can not be changed while this SIP Server Profile is associated to a Server Flow.

Server Type	Call Server
SIP Domain	
DNS Query Type	NONE/A
TLS Client Profile	GSSCP_Client

IP Address / FQDN	Port	Transport	
10.10.3.42	5061	TLS	Delete

On the **Advanced** tab:

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



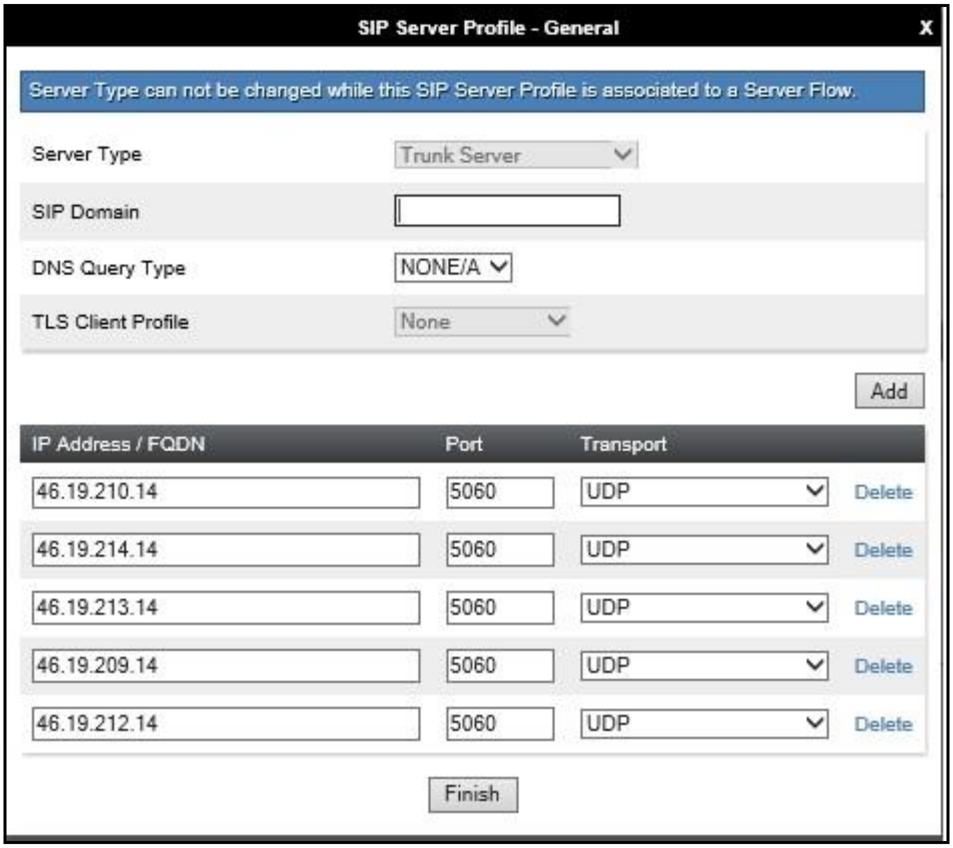
SIP Server Profile - Advanced

Enable DoS Protection	□
Enable Grooming	☑
Interworking Profile	Avaya
Signaling Manipulation Script	None
Securable	□
Enable FGDN	□
TCP Failover Port	
TLS Failover Port	
Tolerant	□
URI Group	None

7.6.2. Server Configuration – DIDWW

To define the DIDWW Inbound 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 **46.19.210.14**.
- For **Port**, enter **5060**.
- For **Transport,** select **UDP**.
- Click **Add** and repeat the process for the next four IP address entries in the screenshot below.
- Click on **Next** (not shown) to use default entries on the **Authentication** and **Heartbeat** tabs.



SIP Server Profile - General

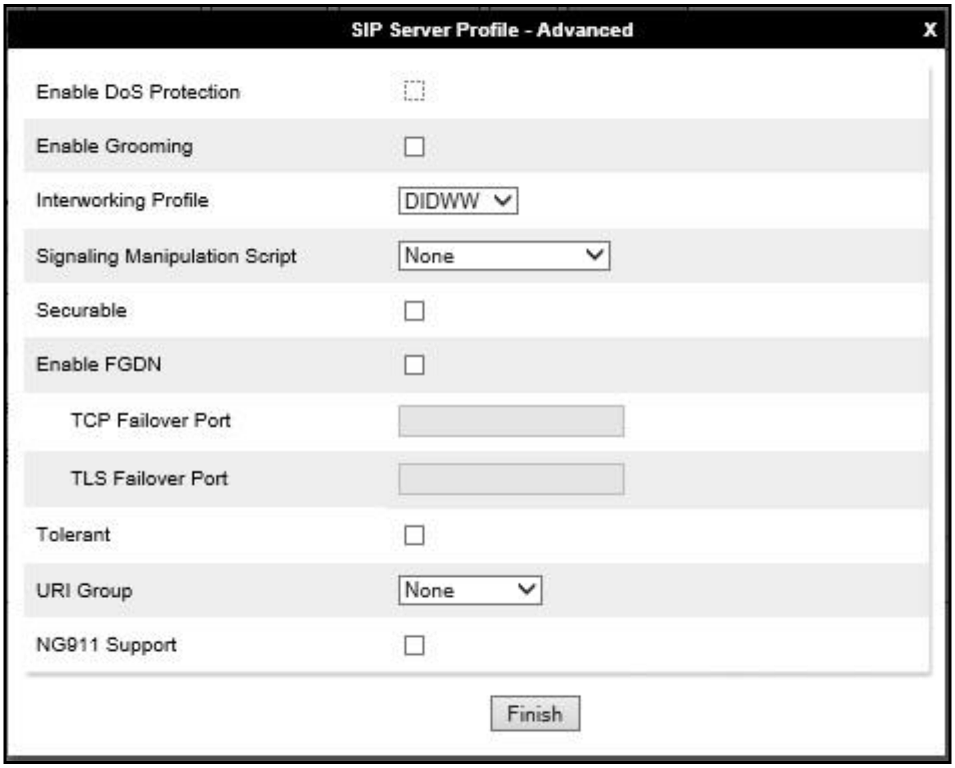
Server Type can not be changed while this SIP Server Profile is associated to a Server Flow.

Server Type	Trunk Server
SIP Domain	
DNS Query Type	NONE/A
TLS Client Profile	None

IP Address / FQDN	Port	Transport	
46.19.210.14	5060	UDP	Delete
46.19.214.14	5060	UDP	Delete
46.19.213.14	5060	UDP	Delete
46.19.209.14	5060	UDP	Delete
46.19.212.14	5060	UDP	Delete

On the Advanced tab:

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

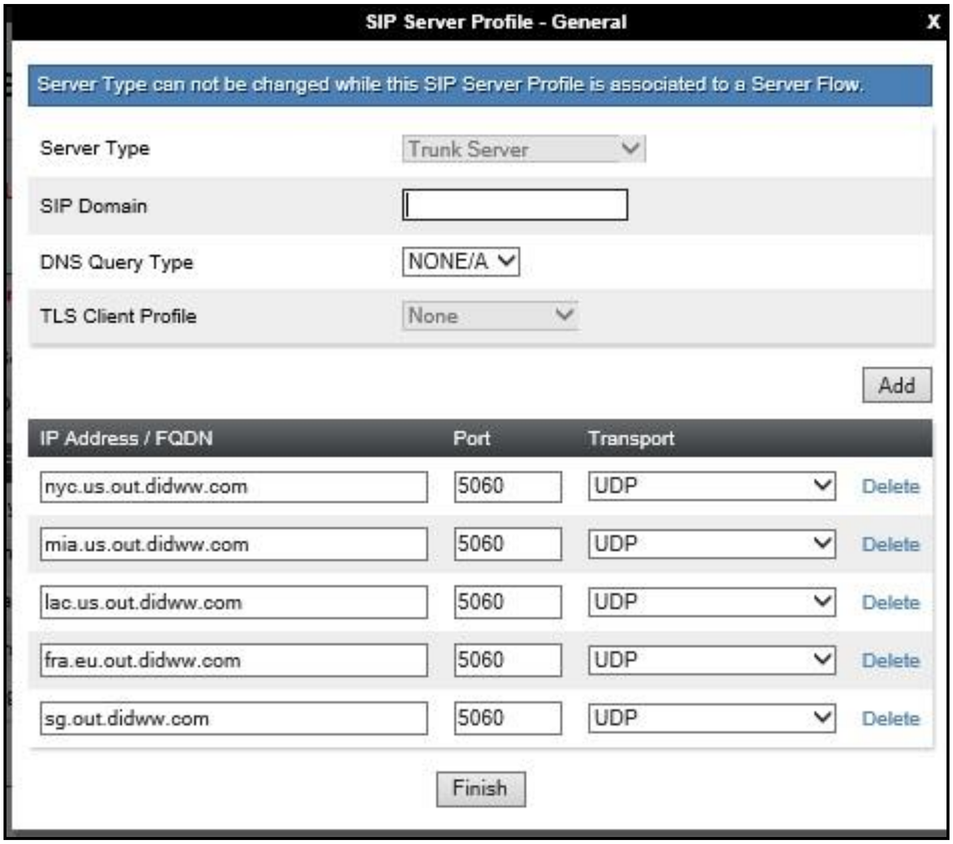


SIP Server Profile - Advanced

Enable DoS Protection	□
Enable Grooming	□
Interworking Profile	DIDWW
Signaling Manipulation Script	None
Securable	□
Enable FGDN	□
TCP Failover Port	
TLS Failover Port	
Tolerant	□
URI Group	None
NG911 Support	□

To define the DIDWW Outbound 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 **nyc.us.out.didww.com**.
- For **Port**, enter **5060**.
- For **Transport,** select **UDP**.
- Click **Add** and repeat the process for the next four FQDN address entries in the screenshot below.
- Click on **Next** (not shown).



SIP Server Profile - General

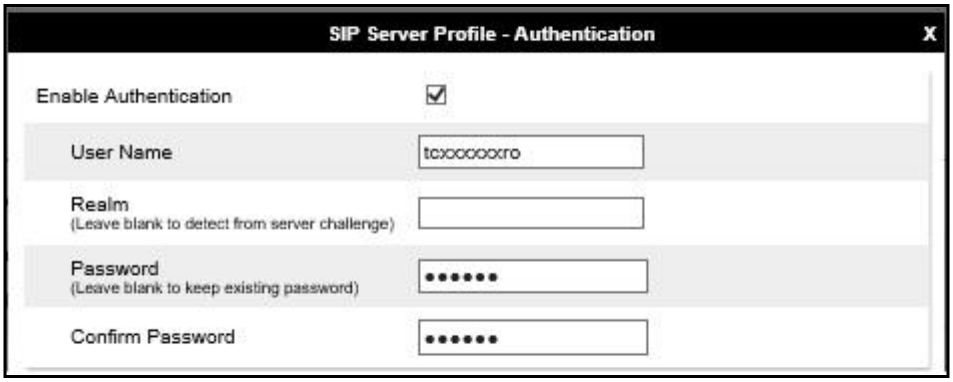
Server Type can not be changed while this SIP Server Profile is associated to a Server Flow.

Server Type	Trunk Server
SIP Domain	
DNS Query Type	NONE/A
TLS Client Profile	None

IP Address / FQDN	Port	Transport	
nyc.us.out.didww.com	5060	UDP	Delete
mia.us.out.didww.com	5060	UDP	Delete
lac.us.out.didww.com	5060	UDP	Delete
fra.eu.out.didww.com	5060	UDP	Delete
sg.out.didww.com	5060	UDP	Delete

In the new window that appears, enter the following values as DIDWW require authentication to connect to the Outbound SIP trunk:

- **Enabled Authentication:** Checked.
- **User Name:** Enter username provided by the Service Provider.
- **Realm:** Enter realm details provided by the Service Provider.
- **Password** Enter password provided by the Service Provider.
- **Confirm Password** Re-enter password provided by the Service Provider.



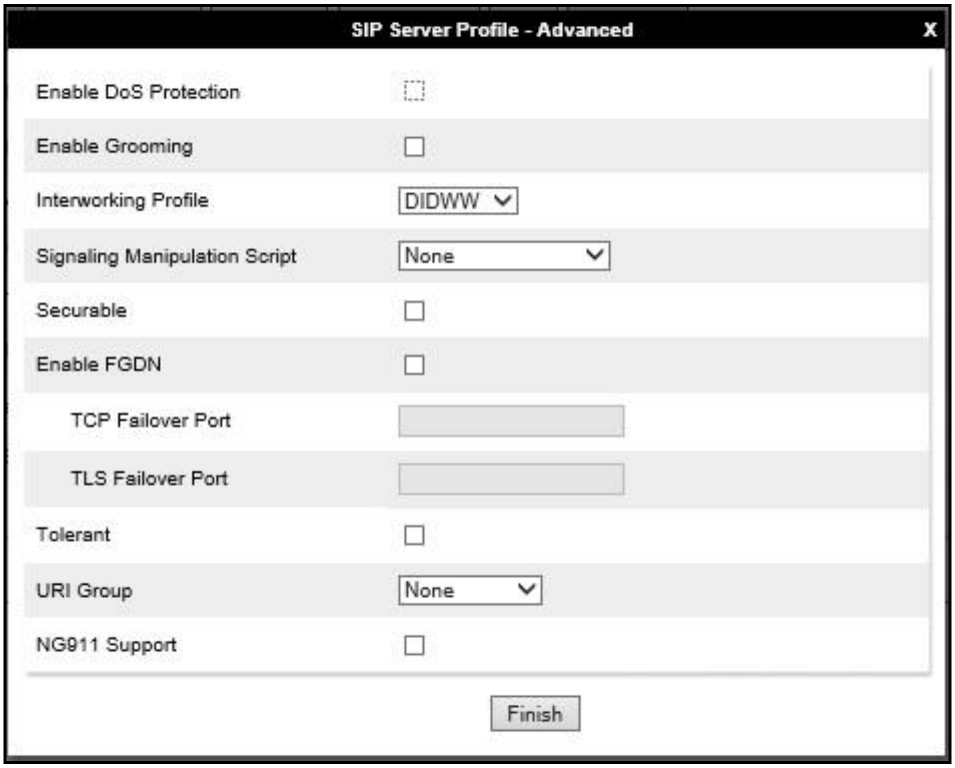
SIP Server Profile - Authentication

Enable Authentication	☑
User Name	tcxxxxxxro
Realm (Leave blank to detect from server challenge)	
Password (Leave blank to keep existing password)	••••••
Confirm Password	••••••

Click on **Next** (not shown) to use default entries on the **Heartbeat**, **Registration** and **Ping** tabs as registration to the DIDWW Outbound SIP trunk was not required during testing.

On the Advanced tab:

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



SIP Server Profile - Advanced

Enable DoS Protection	□
Enable Grooming	□
Interworking Profile	DIDWW
Signaling Manipulation Script	None
Securable	□
Enable FGDN	□
TCP Failover Port	
TLS Failover Port	
Tolerant	□
URI Group	None
NG911 Support	□

7.7. Routing

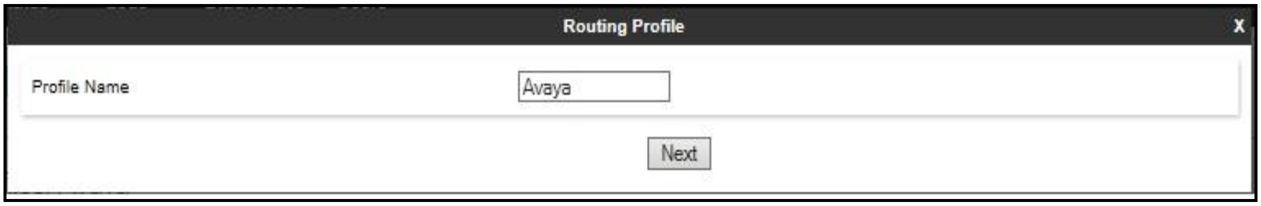
Routing profiles define a specific set of packet routing criteria that are used in conjunction with other types of domain policies to identify a particular call flow and thereby ascertain which security features will be applied to those packets. Parameters defined by Routing Profiles include packet transport 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 DIDWW 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.

7.7.1. Routing – Avaya

Create a Routing Profile for Session Manager.

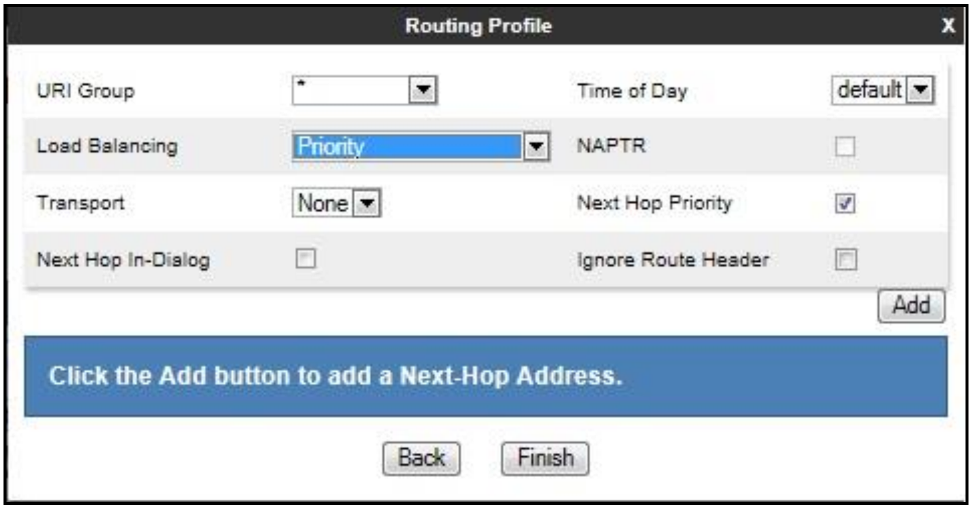
- Navigate to **Configuration Profiles → Routing** and select **Add Profile**.
- Enter a **Profile Name** and click **Next**.



Routing Profile

Profile Name	Avaya

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



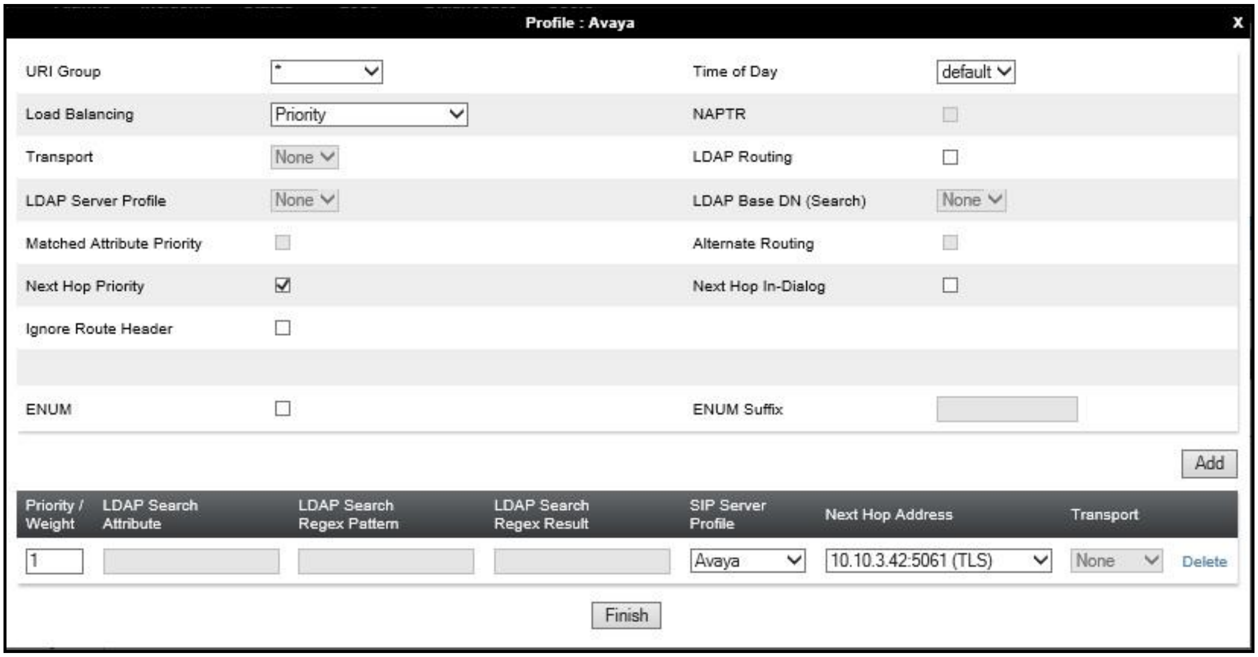
Routing Profile

URI Group	*	Time of Day	default
Load Balancing	Priority	NAPTR	□
Transport	None	Next Hop Priority	☑
Next Hop In-Dialog	□	Ignore Route Header	□

Click the Add button to add a Next-Hop Address.

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

- **Priority/Weight** = **1**.
- **SIP Server Profile** = **Avaya** (**Section 7.6.1**) from drop down menu.
- **Next Hop Address** = Select **10.10.3.42:5061(TLS)** from drop down menu.
- Click **Finish**.



Profile : Avaya

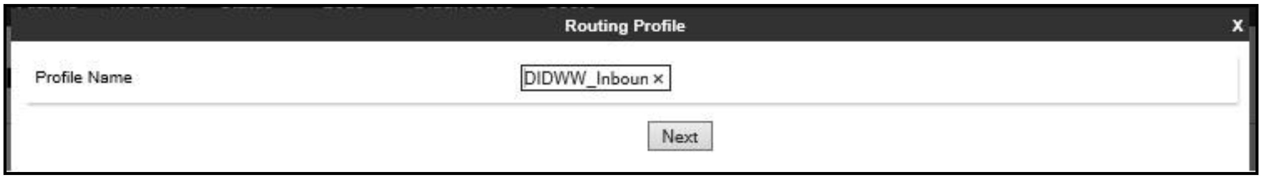
URI Group	*	Time of Day	default
Load Balancing	Priority	NAPTR	□
Transport	None	LDAP Routing	□
LDAP Server Profile	None	LDAP Base DN (Search)	None
Matched Attribute Priority	□	Alternate Routing	□
Next Hop Priority	☑	Next Hop In-Dialog	□
Ignore Route Header	□		
ENUM	□	ENUM Suffix	

Priority / Weight	LDAP Search Attribute	LDAP Search Regex Pattern	LDAP Search Regex Result	SIP Server Profile	Next Hop Address	Transport	
1				Avaya	10.10.3.42:5061 (TLS)	None	Delete

7.7.2. Routing – DIDWW

Create a Routing Profile for DIDWW Inbound SIP trunk.

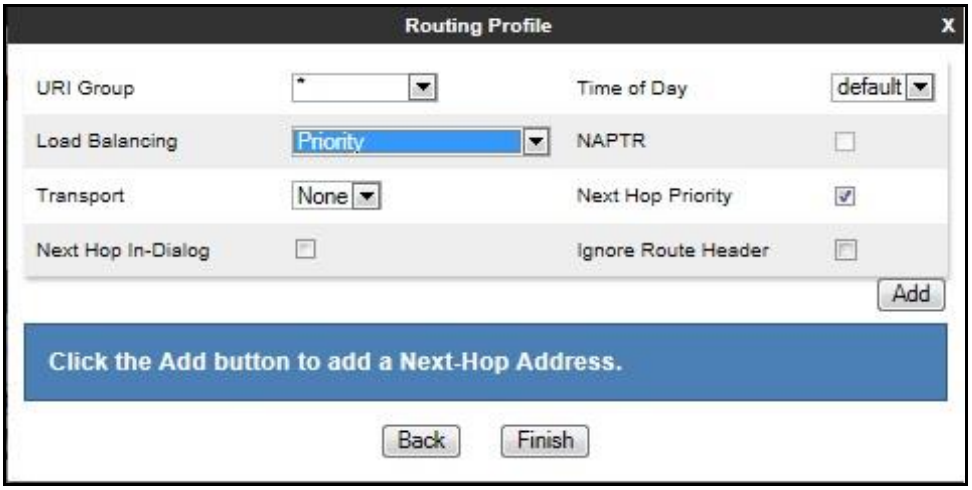
- Navigate to **Configuration Profiles** → **Routing** and select **Add Profile**.
- Enter a **Profile Name** such as **DIDWW_Inbound** and click **Next**.



Routing Profile

Profile Name: DIDWW_Inboun

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

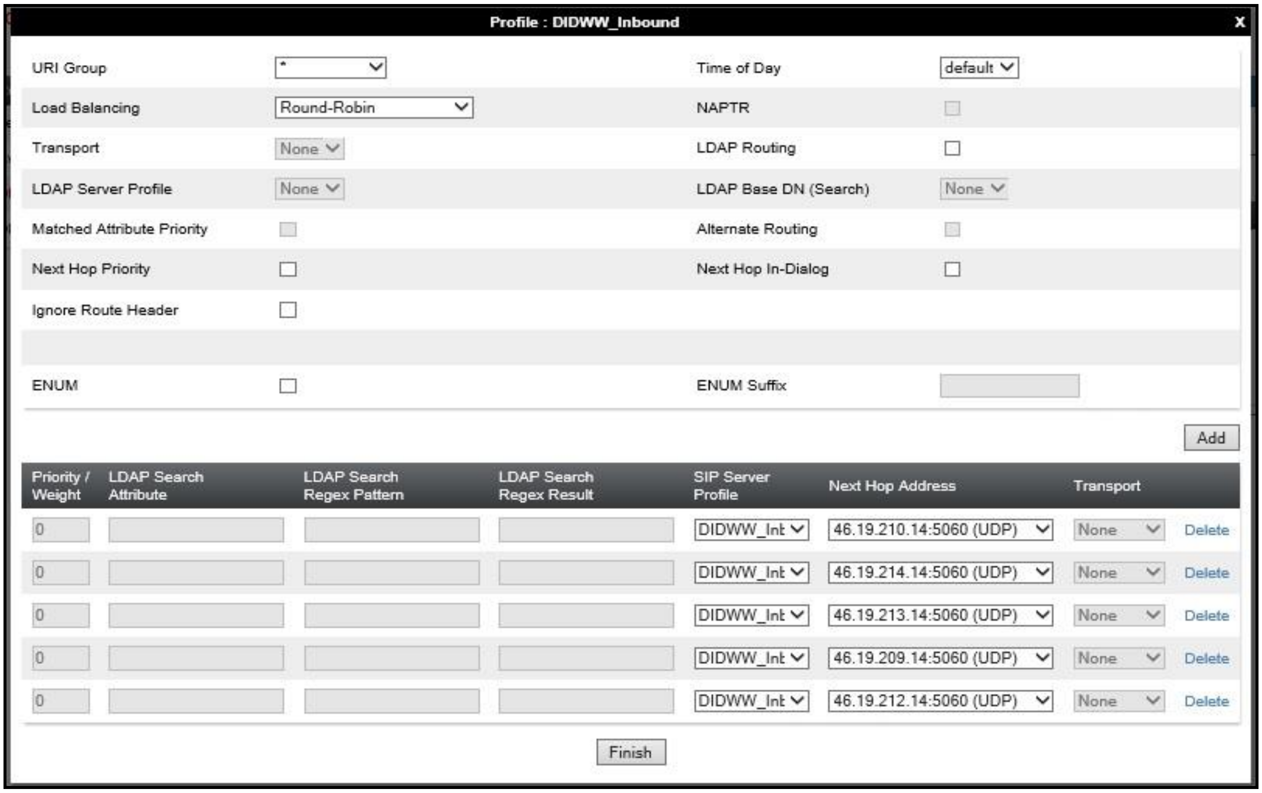


Routing Profile

URI Group	*	Time of Day	default
Load Balancing	Priority	NAPTR	□
Transport	None	Next Hop Priority	☑
Next Hop In-Dialog	□	Ignore Route Header	□

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

- **Load Balancing = Round-Robin**.
- **SIP Server Profile** = **DIDWW_Inbound** (**Section 7.6.2**) from drop down menu.
- **Next Hop Address** = Select **46.19.210.14:5060 (UDP)** from drop down menu.
- Click **Add** and repeat the process for the next four IP address entries in the screenshot below.
- Click **Finish**.



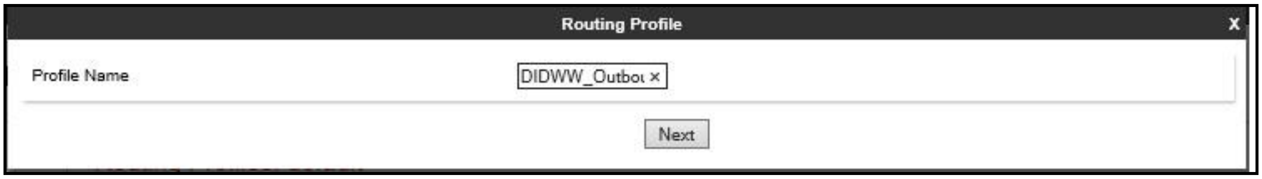
Profile : DIDWW_Inbound

URI Group	*	Time of Day	default
Load Balancing	Round-Robin	NAPTR	□
Transport	None	LDAP Routing	□
LDAP Server Profile	None	LDAP Base DN (Search)	None
Matched Attribute Priority	□	Alternate Routing	□
Next Hop Priority	□	Next Hop In-Dialog	□
Ignore Route Header	□		
ENUM	□	ENUM Suffix	

Priority / Weight	LDAP Search Attribute	LDAP Search Regex Pattern	LDAP Search Regex Result	SIP Server Profile	Next Hop Address	Transport	
0				DIDWW_Int	46.19.210.14:5060 (UDP)	None	Delete
0				DIDWW_Int	46.19.214.14:5060 (UDP)	None	Delete
0				DIDWW_Int	46.19.213.14:5060 (UDP)	None	Delete
0				DIDWW_Int	46.19.209.14:5060 (UDP)	None	Delete
0				DIDWW_Int	46.19.212.14:5060 (UDP)	None	Delete

Create a Routing Profile for DIDWW Outbound SIP trunk.

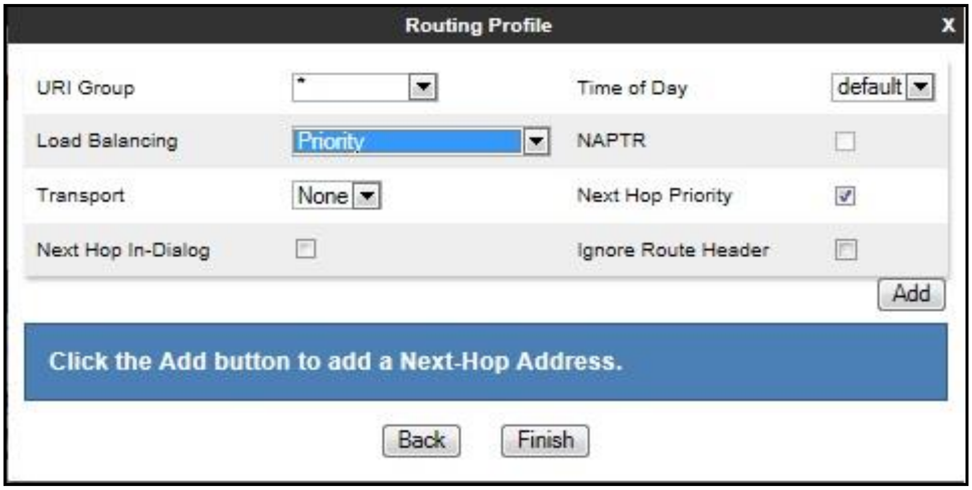
- Navigate to **Configuration Profiles → Routing** and select **Add Profile**.
- Enter a **Profile Name** such as **DIDWW_Outbound** and click **Next**.



Routing Profile

Profile Name: DIDWW_Outbo

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



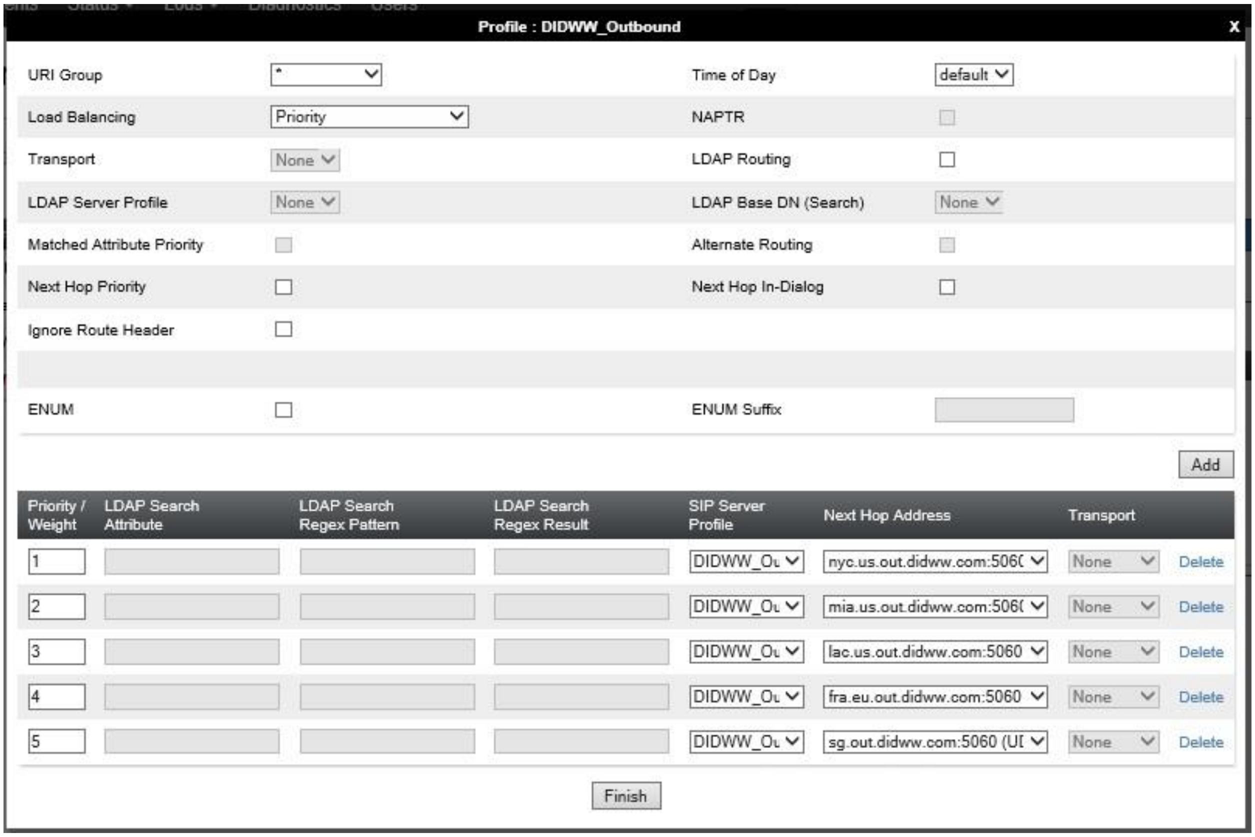
Routing Profile

URI Group	*	Time of Day	default
Load Balancing	Priority	NAPTR	□
Transport	None	Next Hop Priority	☑
Next Hop In-Dialog	□	Ignore Route Header	□

Click the Add button to add a Next-Hop Address.

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

- **Load Balancing = Priority**.
- **Priority/Weight** = **1**.
- **SIP Server Profile** = **DIDWW_Outbound** (**Section 7.6.2**) from drop down menu.
- **Next Hop Address** = Select **nyc.us.out.didww.com:5060 (UDP)** from drop down menu.
- Click **Add** and repeat the process for the next four FQDN address entries as per the screenshot below.
- Click **Finish**.



Profile : DIDWW_Outbound

URI Group	*	Time of Day	default
Load Balancing	Priority	NAPTR	□
Transport	None	LDAP Routing	□
LDAP Server Profile	None	LDAP Base DN (Search)	None
Matched Attribute Priority	□	Alternate Routing	□
Next Hop Priority	□	Next Hop In-Dialog	□
Ignore Route Header	□		
ENUM	□	ENUM Suffix	

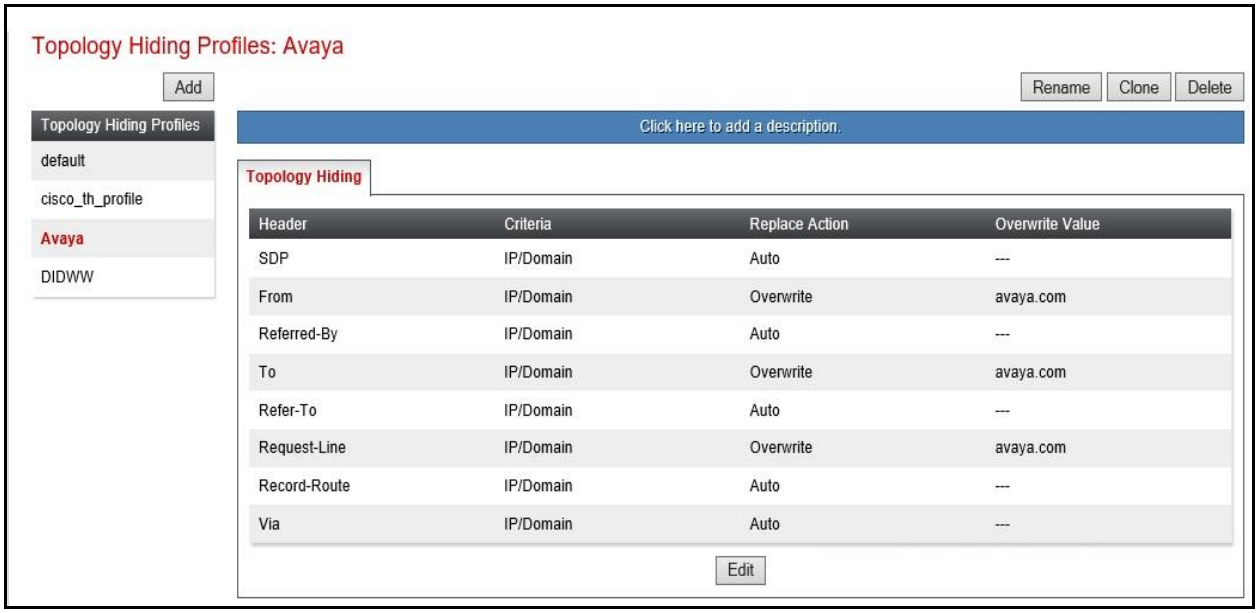
Priority / Weight	LDAP Search Attribute	LDAP Search Regex Pattern	LDAP Search Regex Result	SIP Server Profile	Next Hop Address	Transport	
1				DIDWW_Ou	nyc.us.out.didww.com:506(	None	Delete
2				DIDWW_Ou	mia.us.out.didww.com:506(	None	Delete
3				DIDWW_Ou	lac.us.out.didww.com:5060	None	Delete
4				DIDWW_Ou	fra.eu.out.didww.com:5060	None	Delete
5				DIDWW_Ou	sg.out.didww.com:5060 (UI	None	Delete

7.8. 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 can't be applied, in particular the Contact header, IP addresses are translated to the Avaya SBCE 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).

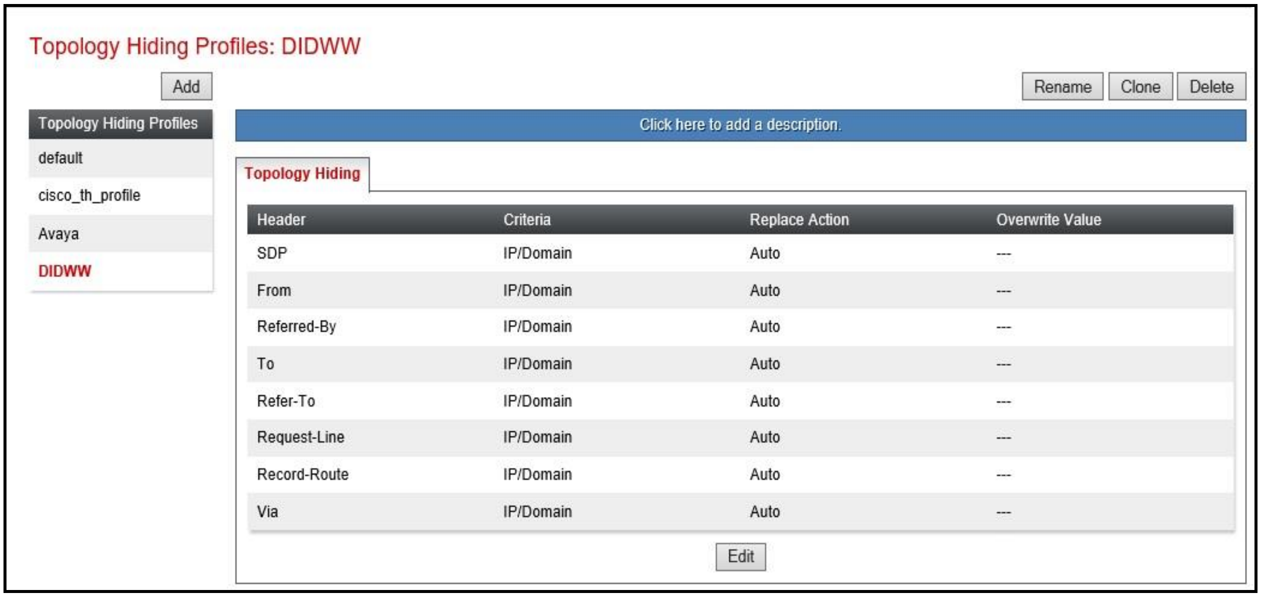
- Enter a descriptive Profile Name such as **Avaya**.
- If the required Header is not shown, click on **Add Header**.
- Under the **Header** field for **To**, **From** and **Request Line**, select **IP/Domain** under **Criteria** and **Overwrite** under **Replace Action**. For Overwrite value, insert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



Header	Criteria	Replace Action	Overwrite Value
SDP	IP/Domain	Auto	---
From	IP/Domain	Overwrite	avaya.com
Referred-By	IP/Domain	Auto	---
To	IP/Domain	Overwrite	avaya.com
Refer-To	IP/Domain	Auto	---
Request-Line	IP/Domain	Overwrite	avaya.com
Record-Route	IP/Domain	Auto	---
Via	IP/Domain	Auto	---

To define Topology Hiding for DIDWW, 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 DIDWW and click **Next**.
- 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 **Auto** under **Replace Action**.
- Click **Finish** (not shown).



Header	Criteria	Replace Action	Overwrite Value
SDP	IP/Domain	Auto	---
From	IP/Domain	Auto	---
Referred-By	IP/Domain	Auto	---
To	IP/Domain	Auto	---
Refer-To	IP/Domain	Auto	---
Request-Line	IP/Domain	Auto	---
Record-Route	IP/Domain	Auto	---
Via	IP/Domain	Auto	---

7.9. 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 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.

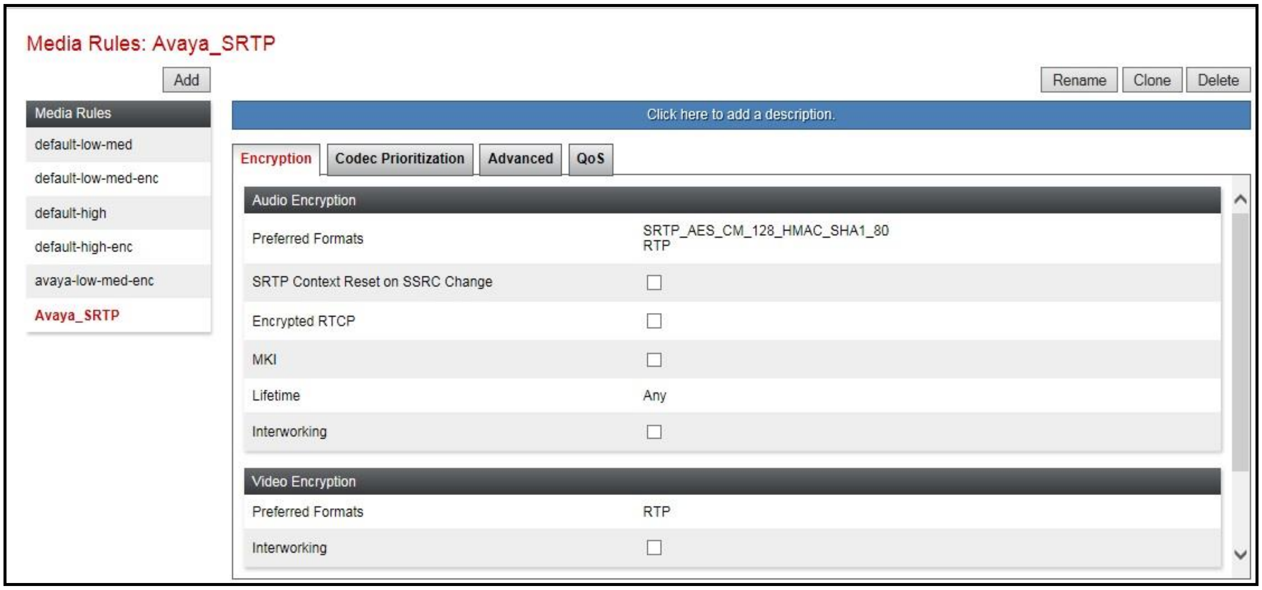
7.9.1. Media Rules

A media rule defines the processing to be applied to the selected media. For the compliance test, a media rule was created for Session Manager to use SRTP, while the predefined **default-low-med** media rule was used for the DIDWW SIP trunk.

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_SHDIDWW_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).



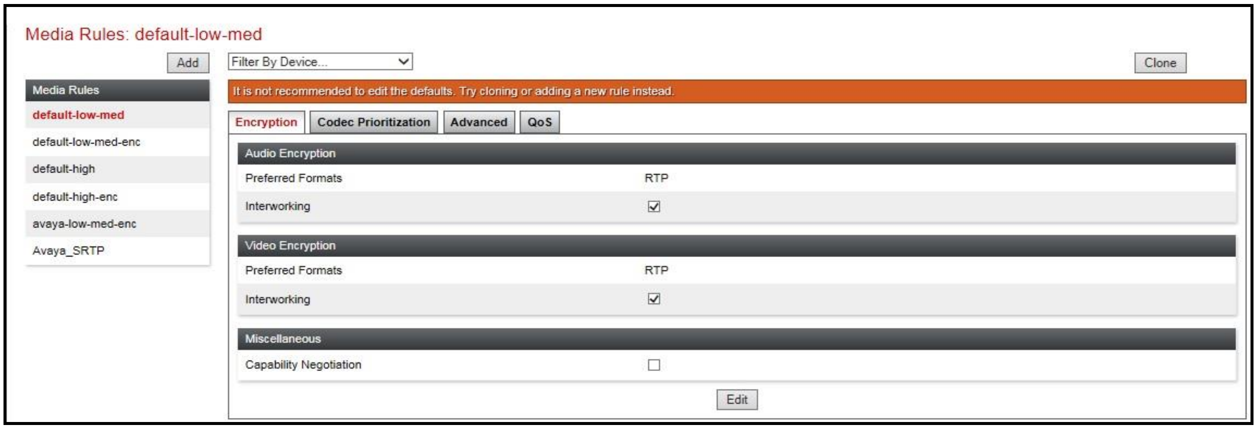
Media Rules: Avaya_SRTP

Media Rules
- default-low-med
- default-low-med-enc
- default-high
- default-high-enc
- avaya-low-med-enc
- Avaya_SRTP

Audio Encryption	
Preferred Formats	SRTP_AES_CM_128_HMAC_SHA1_80 RTP
SRTP Context Reset on SSRC Change	□
Encrypted RTCP	□
MKI	□
Lifetime	Any
Interworking	□

Video Encryption	
Preferred Formats	RTP
Interworking	□

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



Media Rules: default-low-med

It is not recommended to edit the defaults. Try cloning or adding a new rule instead.

Audio Encryption	
Preferred Formats	RTP
Interworking	☑
Video Encryption	
Preferred Formats	RTP
Interworking	☑
Miscellaneous	
Capability Negotiation	□

7.10. End Point Policy Groups

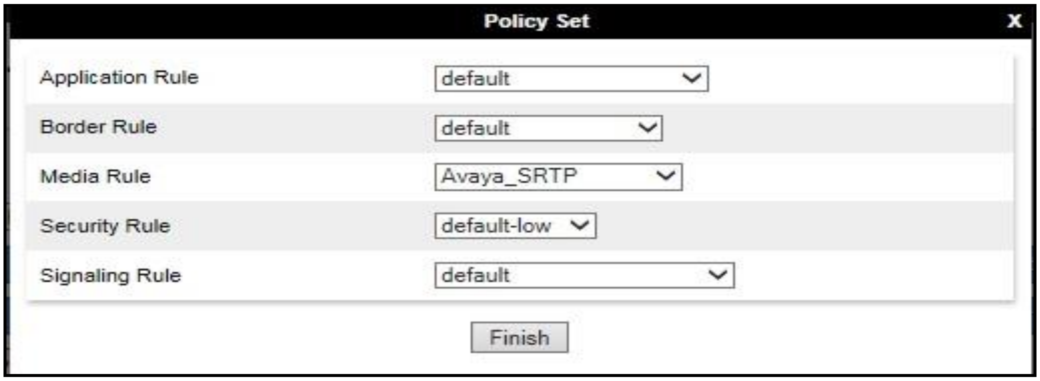
An end point policy group is a set of policies that will be applied to traffic between the Avaya SBCE and a signaling endpoint (connected server). Thus, one end point policy group must be created for Session Manager and another for the DIDWW SIP trunk. The end point policy group is applied to the traffic as part of the end point flow defined in **Section 7.11**.

7.10.1. 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**.

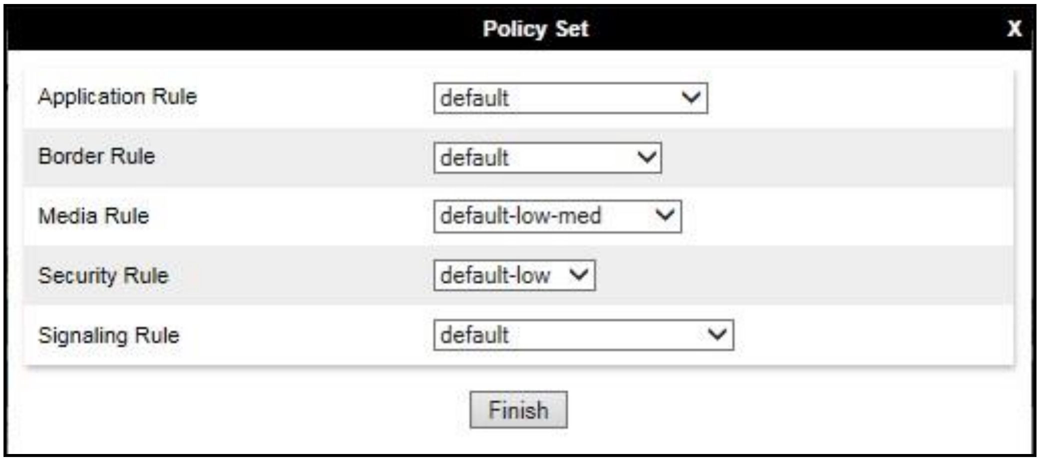
Click **Finish**.



Application Rule	default
Border Rule	default
Media Rule	Avaya_SRTP
Security Rule	default-low
Signaling Rule	default

7.10.2. End Point Policy Group – DIDWW

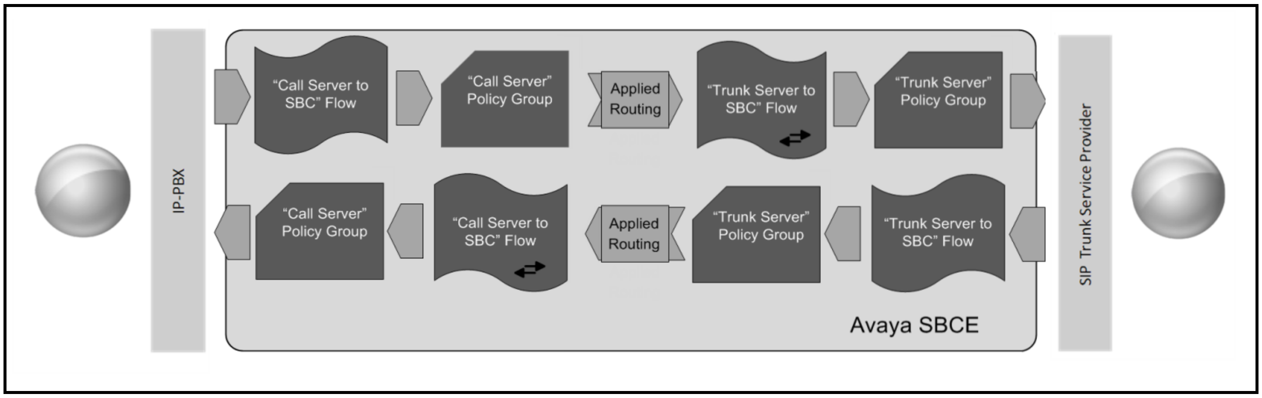
For the compliance test, the predefined End Point Policy **default-low** was used for the DIDWW End Point Policy Group.



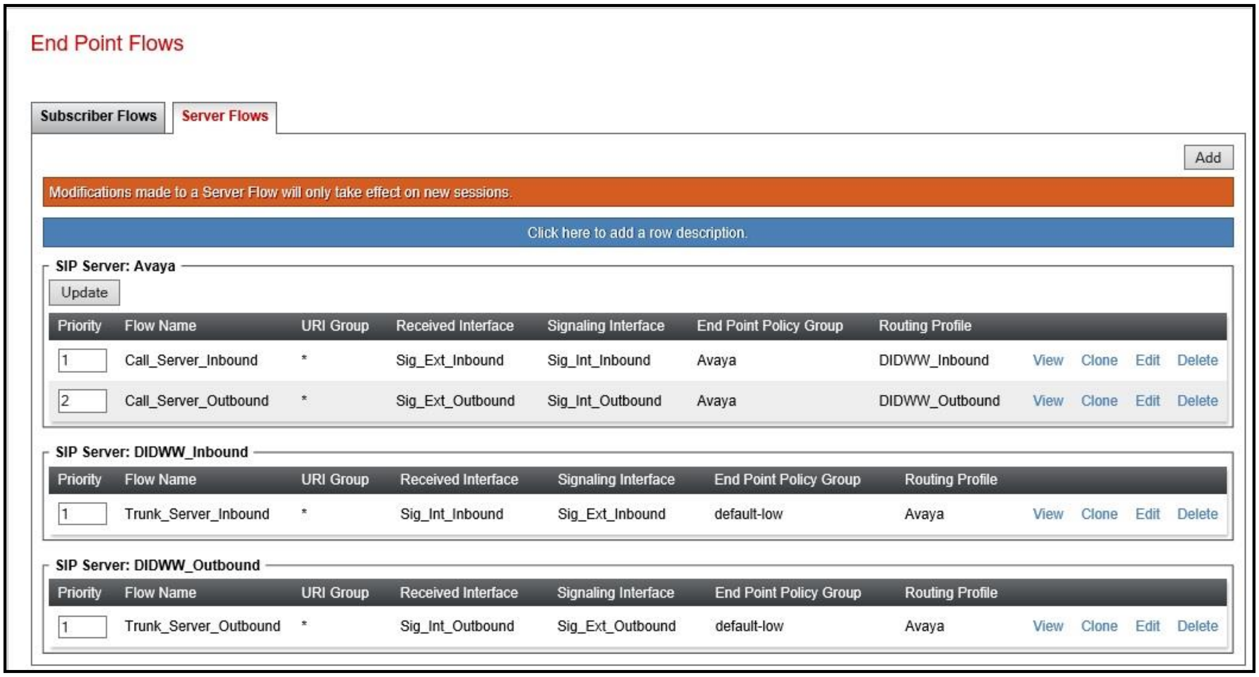
Application Rule	default
Border Rule	default
Media Rule	default-low-med
Security Rule	default-low
Signaling Rule	default

7.11. Server Flows

Server Flows combine the previously defined profiles into outgoing flows from Session Manager to DIDWW’s SIP trunks and incoming flows from DIDWW s SIP trunks to Session Manager. The following screen illustrates the flow through the Avaya SBCE to secure a SIP trunk call.


IP-PBX
“Call Server to SBC” Flow
“Call Server” Policy Group
Applied Routing
“Trunk Server to SBC” Flow
“Trunk Server” Policy Group
“Call Server” Policy Group
“Call Server to SBC” Flow
Applied Routing
“Trunk Server” Policy Group
“Trunk Server to SBC” Flow
Avaya SBCE
SIP Trunk Service Provider

Two server flows are required for outgoing traffic and two are required for incoming. This is so that traffic can be routed to both the DIDWW Inbound and Outbound SIP trunks and can also be received from both the DIDWW Inbound and Outbound SIP trunks. This configuration ties all the previously entered information together so that calls can be routed from Session Manager to the DIDWW SIP trunks and vice versa. The following screenshot shows all configured flows.



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

Click here to add a row description.

SIP Server: Avaya

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile				
1	Call_Server_Inbound	*	Sig_Ext_Inbound	Sig_Int_Inbound	Avaya	DIDWW_Inbound	View	Clone	Edit	Delete
2	Call_Server_Outbound	*	Sig_Ext_Outbound	Sig_Int_Outbound	Avaya	DIDWW_Outbound	View	Clone	Edit	Delete

SIP Server: DIDWW_Inbound

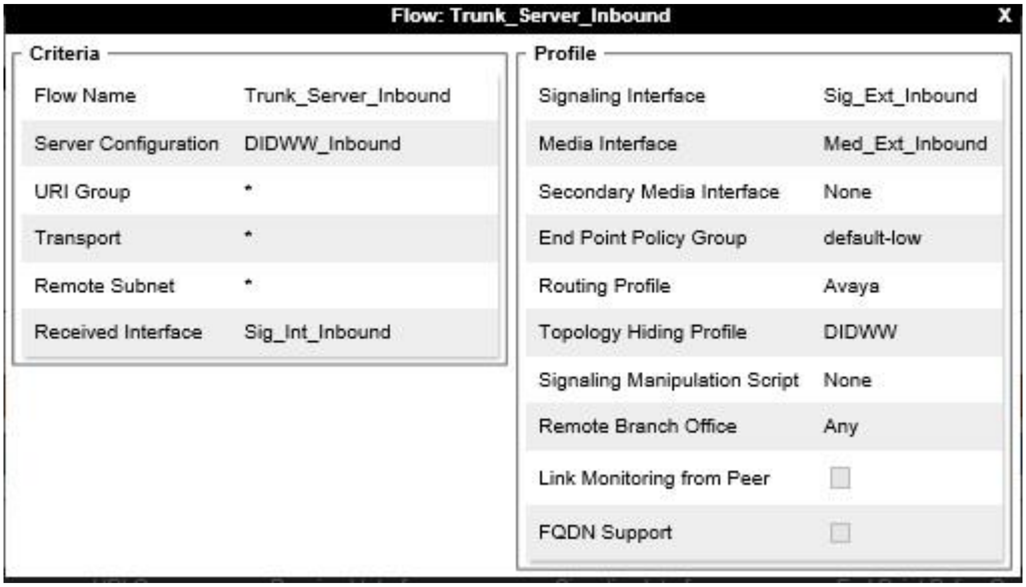
Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile				
1	Trunk_Server_Inbound	*	Sig_Int_Inbound	Sig_Ext_Inbound	default-low	Avaya	View	Clone	Edit	Delete

SIP Server: DIDWW_Outbound

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile				
1	Trunk_Server_Outbound	*	Sig_Int_Outbound	Sig_Ext_Outbound	default-low	Avaya	View	Clone	Edit	Delete

To define the inbound Trunk Server Flow for the DIDWW Inbound 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 DIDWW Inbound SIP trunk, in the test environment **Trunk_Server_Inbound** was used.
- In the **Server Configuration** drop-down menu, select the **DIDWW_Inbound** server configuration defined in **Section 7.6.2**.
- In the **Received Interface** drop-down menu, select the internal SIP signalling interface **Sig_Int_Inbound** defined in **Section 7.4.1**.
- In the **Signaling Interface** drop-down menu, select the external SIP signalling interface **Sig_Ext_Inbound** defined in **Section 7.4.1**.
- In the **Media Interface** drop-down menu, select the external media interface **Med_Ext_Inbound** defined in **Section 7.4.2**.
- Set the **End Point Policy Group** to the endpoint policy group **default-low**.
- In the **Routing Profile** drop-down menu, select the routing profile **Avaya** for Session Manager defined in **Section 7.7.1**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile **DIDWW** defined in **Section 7.8** and click **Finish** (not shown).



Flow: Trunk_Server_Inbound

Criteria

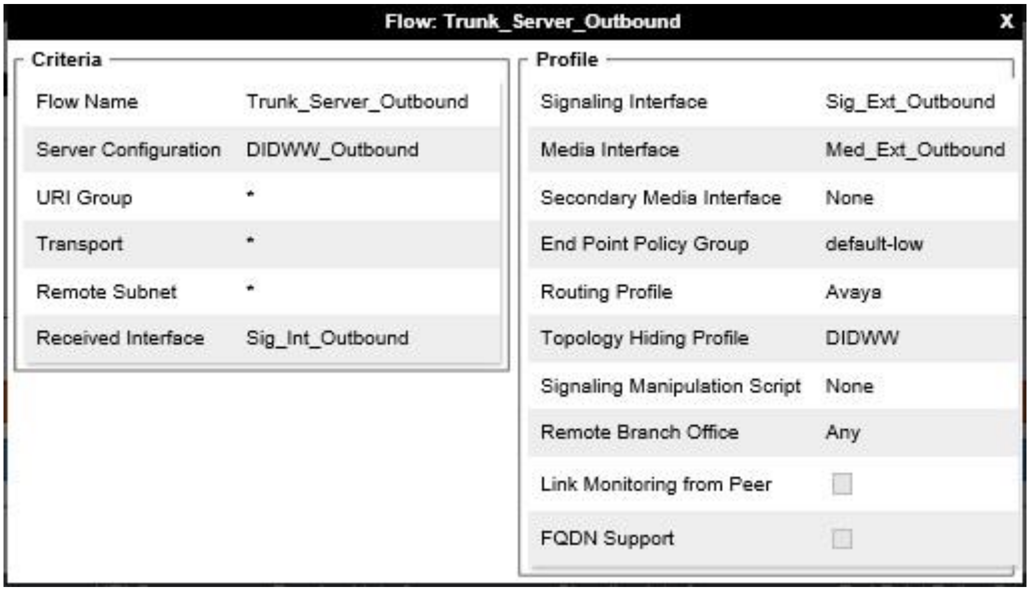
Flow Name	Trunk_Server_Inbound
Server Configuration	DIDWW_Inbound
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Sig_Int_Inbound

Profile

Signaling Interface	Sig_Ext_Inbound
Media Interface	Med_Ext_Inbound
Secondary Media Interface	None
End Point Policy Group	default-low
Routing Profile	Avaya
Topology Hiding Profile	DIDWW
Signaling Manipulation Script	None
Remote Branch Office	Any
Link Monitoring from Peer	□
FQDN Support	□

To define the outbound Trunk Server Flow for the DIDWW Outbound 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 DIDWW Outbound SIP trunk, in the test environment **Trunk_Server_Outbound** was used.
- In the **Server Configuration** drop-down menu, select the **DIDWW_Outbound** server configuration defined in **Section 7.6.2**.
- In the **Received Interface** drop-down menu, select the internal SIP signalling interface **Sig_Int_Outbound** defined in **Section 7.4.1**.
- In the **Signaling Interface** drop-down menu, select the external SIP signalling interface **Sig_Ext_Outbound** defined in **Section 7.4.1**.
- In the **Media Interface** drop-down menu, select the external media interface **Med_Ext_Outbound** defined in **Section 7.4.2**.
- Set the **End Point Policy Group** to the endpoint policy group **default-low**.
- In the **Routing Profile** drop-down menu, select the routing profile **Avaya** for Session Manager defined in **Section 7.7.1**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile **DIDWW** defined in **Section 7.8** and click **Finish** (not shown).



Flow: Trunk_Server_Outbound

Criteria

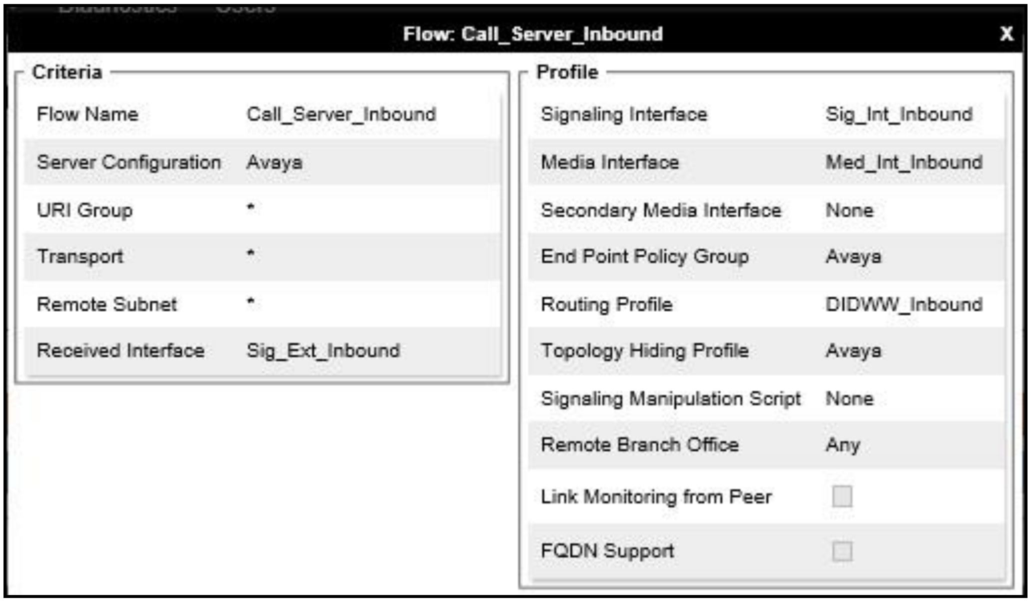
Flow Name	Trunk_Server_Outbound
Server Configuration	DIDWW_Outbound
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Sig_Int_Outbound

Profile

Signaling Interface	Sig_Ext_Outbound
Media Interface	Med_Ext_Outbound
Secondary Media Interface	None
End Point Policy Group	default-low
Routing Profile	Avaya
Topology Hiding Profile	DIDWW
Signaling Manipulation Script	None
Remote Branch Office	Any
Link Monitoring from Peer	☐
FQDN Support	☐

To define the inbound Call Server Flow for the DIDWW Inbound 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 Session Manager, in the test environment **Call_Server_Inbound** was used.
- In the **Server Configuration** drop-down menu, select the server configuration **Avaya** for Session Manager defined in **Section 7.6.1**.
- In the **Received Interface** drop-down menu, select the internal SIP signalling interface **Sig_Ext_Inbound** defined in **Section 7.4.1**.
- In the **Signaling Interface** drop-down menu, select the external SIP signalling interface **Sig_Int_Inbound** defined in **Section 7.4.1**.
- In the **Media Interface** drop-down menu, select the external media interface **Med_Int_Inbound** defined in **Section 7.4.2**.
- Set the **End Point Policy Group** to the endpoint policy group **Avaya**.
- In the **Routing Profile** drop-down menu, select the routing profile **DIDWW_Inbound** defined in **Section 7.7.2**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile **Avaya** defined in **Section 7.8** and click **Finish** (not shown).



Flow: Call_Server_Inbound

Criteria

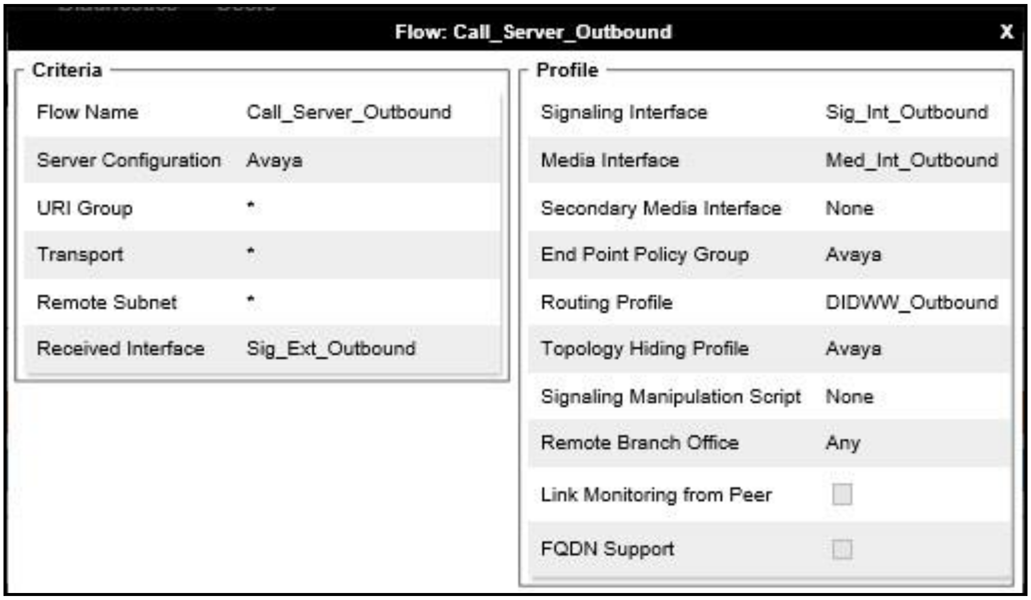
Flow Name	Call_Server_Inbound
Server Configuration	Avaya
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Sig_Ext_Inbound

Profile

Signaling Interface	Sig_Int_Inbound
Media Interface	Med_Int_Inbound
Secondary Media Interface	None
End Point Policy Group	Avaya
Routing Profile	DIDWW_Inbound
Topology Hiding Profile	Avaya
Signaling Manipulation Script	None
Remote Branch Office	Any
Link Monitoring from Peer	□
FQDN Support	□

To define the outbound Call Server Flow for the DIDWW Outbound 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 Session Manager, in the test environment **Call_Server_Outbound** was used.
- In the **Server Configuration** drop-down menu, select the server configuration **Avaya** for Session Manager defined in **Section 7.6.1**.
- In the **Received Interface** drop-down menu, select the internal SIP signalling interface **Sig_Ext_Outbound** defined in **Section 7.4.1**.
- In the **Signaling Interface** drop-down menu, select the external SIP signalling interface **Sig_Int_Outbound** defined in **Section 7.4.1**.
- In the **Media Interface** drop-down menu, select the external media interface **Med_Int_Outbound** defined in **Section 7.4.2**.
- 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 **DIDWW_Outbound** defined in **Section 7.7.2**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile **Avaya** defined in **Section 7.8** and click **Finish** (not shown).



Criteria

Flow Name	Call_Server_Outbound
Server Configuration	Avaya
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Sig_Ext_Outbound

Profile

Signaling Interface	Sig_Int_Outbound
Media Interface	Med_Int_Outbound
Secondary Media Interface	None
End Point Policy Group	Avaya
Routing Profile	DIDWW_Outbound
Topology Hiding Profile	Avaya
Signaling Manipulation Script	None
Remote Branch Office	Any
Link Monitoring from Peer	□
FQDN Support	□

8. DIDWW SIP Trunk Configuration

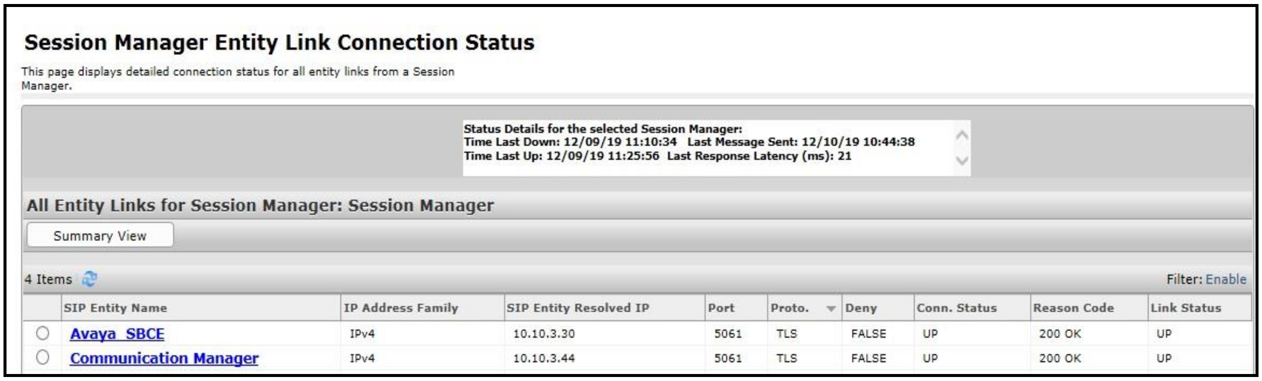
The configuration of the DIDWW equipment used to support the DIDWW SIP trunks is outside of the scope of these Application Notes and will not be covered. To obtain further information on DIDWW equipment and system configuration please visit or contact the details listed below.

- Website: https://www.didww.com/
- Email: sales@didww.com
- Contact phone numbers: US 1-212-6600065, UK: 44-20-80995011.

9. 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**.



Session Manager Entity Link Connection Status

This page displays detailed connection status for all entity links from a Session Manager.

Status Details for the selected Session Manager:
Time Last Down: 12/09/19 11:10:34 Last Message Sent: 12/10/19 10:44:38
Time Last Up: 12/09/19 11:25:56 Last Response Latency (ms): 21

All Entity Links for Session Manager: Session Manager

Summary View

4 Items Filter: Enable

	SIP Entity Name	IP Address Family	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
○	Avaya_SBCE	IPv4	10.10.3.30	5061	TLS	FALSE	UP	200 OK	UP
○	Communication Manager	IPv4	10.10.3.44	5061	TLS	FALSE	UP	200 OK	UP

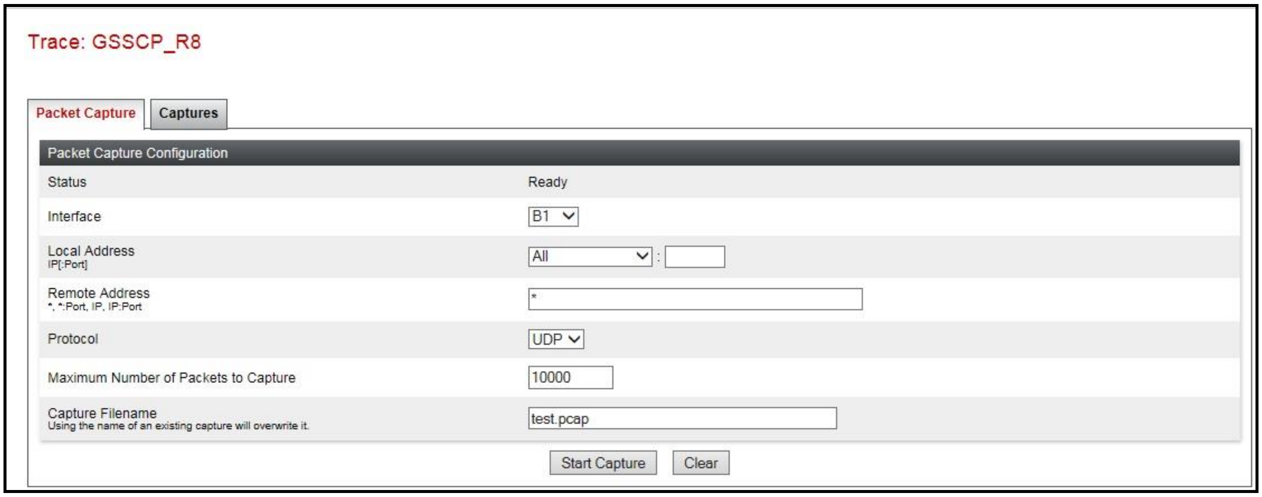
2. From Communication Manager SAT interface run the command **status trunk n** where **n** is a previously configured SIP trunk. Observe if all channels on the trunk group display **in-service/idle**.

```
status trunk 2

                             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 SBCE trace facility to check that the OPTIONS requests sent from Session Manager via the Avaya SBCE to the network SBCs are receiving a response.

To define the trace, navigate to **Device Specific Settings → Advanced Options → Troubleshooting → Trace** in the main menu on the left-hand side and select the **Packet Capture** tab.

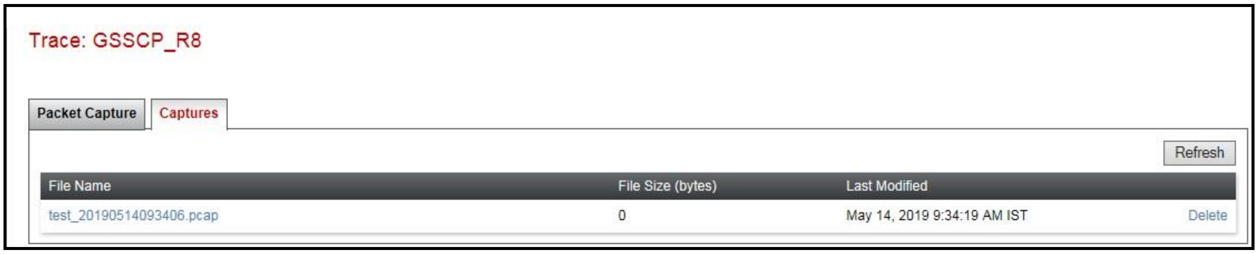
- Select the SIP trunk interface from the **Interface** drop down menu.
- Select the signalling interface IP address or from the **Local Address** drop down menu.
- Enter the IP address of the network SBC in the **Remote Address** field or enter a ***** to capture all traffic.
- Specify the **Maximum Number of Packets to Capture**, **10000** is shown as an example.
- Specify the filename of the resultant pcap file in the **Capture Filename** field.
- Click on **Start Capture**.



Packet Capture Configuration

Status	Ready
Interface	B1
Local Address IP[:Port]	All :
Remote Address *, *:Port, IP, IP:Port	*
Protocol	UDP
Maximum Number of Packets to Capture	10000
Capture Filename Using the name of an existing capture will overwrite it.	test.pcap

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


Trace: GSSCP_R8

File Name	File Size (bytes)	Last Modified	
test_20190514093406.pcap	0	May 14, 2019 9:34:19 AM IST	Delete

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 DIDWW network.

10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura ® Communication Manager R8.1, Avaya Aura ® Session Manager 8.1 and Avaya Session Border Controller for Enterprise R8.1 to the DIDWW SIP platform. The DIDWW SIP Trunk 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**.

11. 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 8.1, Jun 2021

[2] *Upgrading Avaya Aura® applications*, Release 8.1, Jun 2021

[3] *Deploying Avaya Aura® applications from System Manager*, Release 8.1, Jun 2021

[4] *Deploying Avaya Aura® Communication Manager*, Release 8.1, Jul 2021

[5] *Administering Avaya Aura® Communication Manager*, Release 8.1, Jul 2021

[6] *Upgrading Avaya Aura® Communication Manager*, Release 8.1, Jun 2021

[7] *Deploying Avaya Aura® System Manager*, Release 8.1, May 2021

[8] *Upgrading Avaya Aura® System Manager*, Release 8.1, Jul 2021

[9] *Administering Avaya Aura® System Manager*, Release 8.1, Jul 2021

[10] *Deploying Avaya Aura® Session Manager*, Release 8.1 Mar 2021

[11] *Upgrading Avaya Aura® Session Manager*, Release 8.1, Mar 2021

[12] *Administering Avaya Aura® Session Manager*, Release 8.1, Mar 2021

[13] *Deploying Avaya Session Border Controller for Enterprise*, Release 8.1, Dec 2020

[14] *Upgrading Avaya Session Border Controller for Enterprise*, Release 8.1 Dec 2020

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

*[16]Administering DIDWW Inbound SIP Trunks :*https://doc.didww.com/services/did/sip.html

[17]*Administering DIDWW Outbound SIP Trunks:* https://doc.didww.com/services/termination/sip-details.html

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.