Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura ® Communication Manager R7.1, Avaya Aura ® Session Manager R7.1 and Avaya Session Border Controller for Enterprise R7.2 to support Vodafone Libertel B.V. SIP Trunking Service - Issue 1.0

Abstract

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between the Vodafone Libertel B.V. SIP Trunking Service and an Avaya SIP enabled Enterprise Solution. The Avaya solution consists of Avaya Session Border Controller for Enterprise, Avaya Aura® Session Manager and Avaya Aura® Communication Manager as an Evolution Server.

The Vodafone Libertel B.V. SIP Trunk Platform provides PSTN access via a SIP trunk connected to the Vodafone Libertel B.V. 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.

Vodafone Libertel B.V. 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 Vodafone Libertel B.V. (Vodafone Libertel) SIP Trunking Service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of the following: Avaya Aura ® Communication Manager R7.1 (Communication Manager); Avaya Aura ® Session Manager R7.1 (Session Manager); Avaya Session Border Controller for Enterprise R7.2 (Avaya SBCE). Note that the shortened names shown in brackets will be used throughout the remainder of the document. Customers using this Avaya SIP-enabled enterprise solution with the Vodafone Libertel SIP Trunking Service are able to place and receive PSTN calls via a dedicated Internet connection and the SIP protocol.

The Vodafone solution incorporates routing for calls placed to and from their Mobile and Fixed networks separately and offers short dialling to and from dedicated mobile telephones. This converged network solution is an alternative to traditional PSTN trunks. 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 Vodafone Libertel SIP Trunking Service.

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

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

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

2.1. Interoperability Compliance Testing

The interoperability test included the following:

- Incoming calls to the enterprise site from PSTN phones using the Vodafone Libertel SIP Trunking Service, calls made to SIP and H.323 telephones at the enterprise.
- Outgoing calls from the enterprise site completed via the Vodafone Libertel SIP Trunking Service to PSTN destinations, calls made from SIP and H.323 telephones.
- Incoming and Outgoing PSTN calls to/from Avaya one-X® Communicator and Avaya Equinox™ for Windows soft phones.
- Calls using the G.711A and G.729A codecs.
- Fax calls to/from a group 3 fax machine to a PSTN connected fax machine using T.38.
- 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.
- Direct IP-to-IP media between the Avaya SBCE and the SIP and H.323 telephones.
- Call coverage and call forwarding for endpoints at the enterprise site.
- Transmission and response of SIP OPTIONS messages sent by the Vodafone Libertel SIP Trunking Service requiring Avaya response and sent by Avaya requiring Vodafone Libertel response.

2.2. Test Results

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

- No Inbound Toll-Free access available for test
- No Emergency Services test call booked with Operator
- When making a call from a PSTN phone with CLI Restricted (CLIR), the name of the inbound Communication Manager Trunk Group was displayed on the extension. The Trunk Group naming is described in **Section 5.6**. If the user part of the From Header is preferred, this can be displayed by removing the Privacy header from the initial INVITE message as described in **Section 6.4**.
- When putting an outbound call on hold from the PBX, it was observed that the Avaya SBCE did not maintain the conversion between SRTP within the enterprise and RTP on the SIP Trunk. To work around this, the enterprise equipment was reconfigured to use RTP as described in **Section 5.4** and a fault report was raised on the Avaya SBCE (AURORA-12076)
- When call forwarding on no answer from an H.323 phone, inbound media was lost on the calling PSTN phone. This happened because Communication manager uses SIP forking when forwarding after ringing has already been established on the forwarding phone. The To header tag is changed in the SIP 181 and subsequent 180 Ringing messages and this caused the loss of media in the Vodafone Libertel network. Initial IP-IP Direct media was disabled according to the recommendation in the Avaya support solution, document ID SOLN261152. See **Section 5.5** for details of the configuration. This fault did not occur when tested using a SIP phone as the forwarding phone.

- When initial IP-IP direct media is disabled as described in **Section 5.5**, the G430/G450 media gateway or Avaya Aura® Media Server (Media Server) is used in the call set-up. It was observed during testing that when the Media Server is used, two ringback tones are heard on the calling phone. The Vodafone Libertel network sends 183 Session Progress followed by 180 Ringing when setting up calls. Communication Manager is playing ringback when it receives the 180 Ringing despite the fact that the network is playing ringback after establishing early media with the 183 Session Progress message. This issue is currently under investigation.
- When testing consultative transfer to internal extension by one-X® Communicator in "Other Phone Mode", no ringback was heard on the calling PSTN phone. This is considered to be a local issue and not an interoperability issue with the Vodafone Libertel SIP Trunk
- When testing consultative transfer to internal extension by Avaya Equinox™ for Windows, no ringback was heard on the transferring extension. This is considered to be a local issue and not an interoperability issue with the Vodafone Libertel SIP Trunk.

2.3. Support

For technical support on Vodafone Libertel B.V. SIP Trunking Services, contact Vodafone Libertel support at http://www.vodafone.nl/midden-groot-bedrijf/oplossingen/.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an Enterprise site connected to the Vodafone Libertel SIP Trunking Service. Located at the Enterprise site is an Avaya SBCE, Session Manager and Communication Manager. Endpoints are Avaya 96x0 series and Avaya 96x1 series IP telephones (with SIP and H.323 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 Equinox™ for Windows running on laptop PCs.

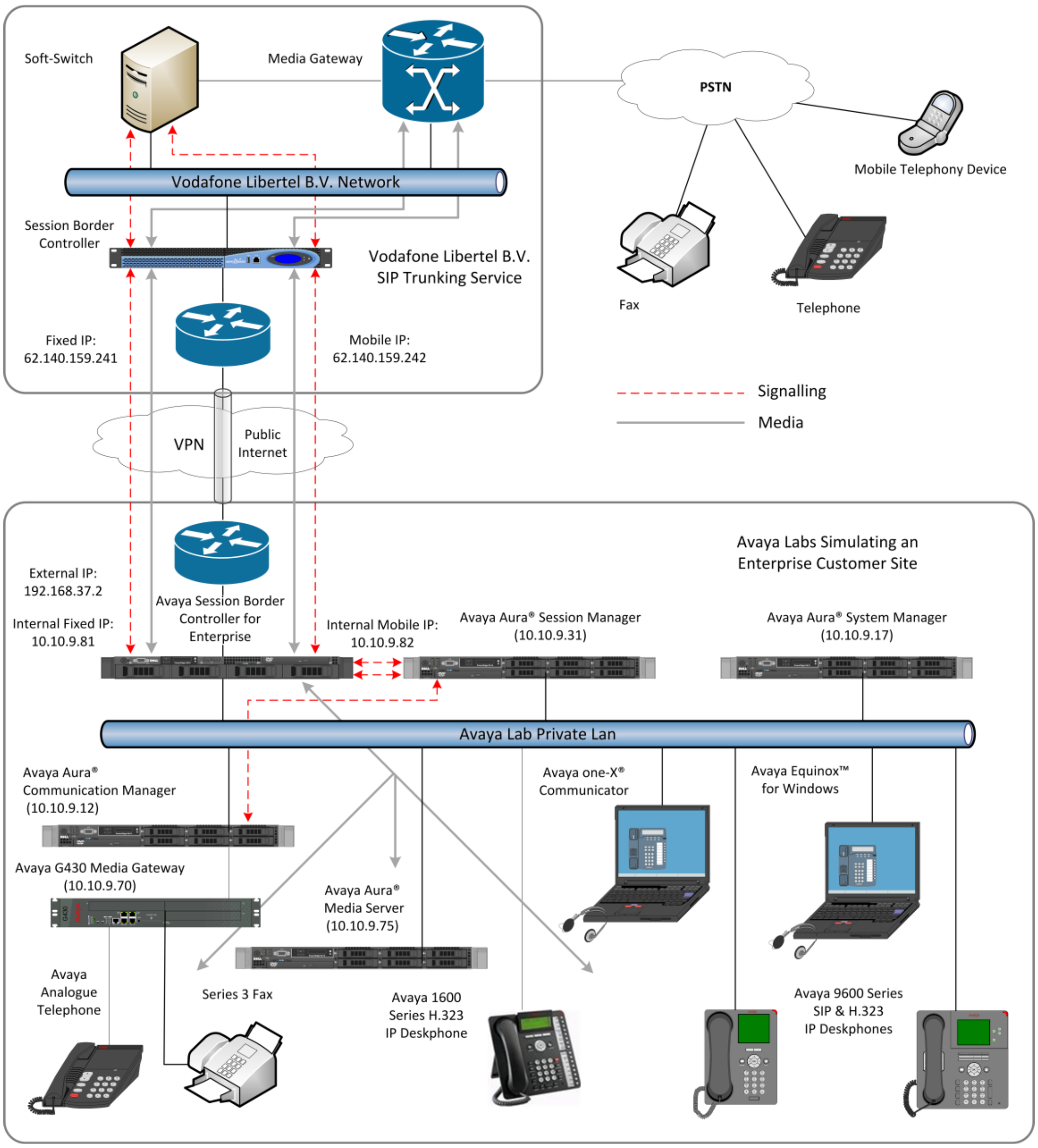


Figure 1: Test Setup Vodafone Libertel SIP Trunking Service to Avaya Enterprise

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya	
Avaya Aura® Session Manager	7.1.1.0.711008
Avaya Aura® System Manager	7.1.1.0 711006931
Avaya Aura® Communication Manager	7.1.1.0.0.532.23985
Avaya Session Border Controller for Enterprise	7.2.0.0-18-13712 patch sbce-7.2.0.0-18-13712-hotfix-08212017.tar
Avaya G430 Media Gateway	38.20.1
Avaya Aura® Media Server	7.8.0.333
Avaya 9600 series Handsets SIP 96x0 SIP 9608 H.323 96x0 H.323 9608 H.323 1616	 2.6.17 7.1.0.1 3.2.8 6.6.5 1.3.11
Avaya one-X® Communicator	6.2 SP 12
Avaya Equinox™ for Windows	3.2.1.11
Analogue Handset	N/A
Analogue Fax	N/A
Vodafone Libertel B.V.	
Acme Packet Net-Net 4500	SCX620m11p4
OneAccess One700	ONEOS11-VOIP_SIP_11N-V4.3R7C14_HC4
SIP GW CPE Cisco 2901	VF-CUBE (15.4(3)M3)

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 Vodafone Libertel 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. Session Manager directs the outbound SIP messages to the Avaya SBCE at the enterprise site that then sends the SIP messages to the Vodafone Libertel 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 Vodafone Libertel 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  30
  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 **10.10.9.31** 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                 10.10.9.75
Session_Manager     10.10.9.31
default             0.0.0.0
procr               10.10.9.12
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 **2** is used.
- The rest of the fields can be left at default values.

```
change ip-network-region 2                                     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: 2                Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                        IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                        RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

Note: In the test configuration, ip-network-region 1 was used within the enterprise and ip-network-region 2 was used for the SIP Trunk. To define a codec set for inter-region traffic, navigate to **Page 4**. In the test environment t, **codec set 2** was used.

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

 Source Region: 2     Inter Network Region Connection Management     I       M
                                                                     G  A    t
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    c
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    e
 1    2      y   NoLimit                                             n       t
 2    2                                                                 all
```

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 Vodafone Libertel were configured, namely **G.711A** and **G.729A**. In addition to the codec's, the **Media Encryption** is defined here. A typical value would be 1-srtp-aescm128-hmac80, but during testing a value of **none** was used to provide a work around for the RTP to SRTP conversion issue described in **Section 2.2**.

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

                          IP MEDIA PARAMETERS
    Codec Set: 2

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

     Media Encryption                   Encrypted SRTCP: enforce-unenc-srtcp
 1: none
```

The Vodafone Libertel SIP Trunking Service supports T.38 for transmission of fax. Navigate to **Page 2** and define T.38 fax as follows:

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

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

                          IP MEDIA PARAMETERS

                              Allow Direct-IP Multimedia? n



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

Note: **Redundancy** can be used to send multiple copies of T.38 packets which can help the successful transmission of fax over networks where packets are being dropped. This was not experienced in the test environment and **Redundancy** was left at the default value of **0**.

5.5. Administer SIP Signaling Groups

This signalling group (and trunk group) will be used for inbound and outbound PSTN calls to the Vodafone Libertel 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 **2**).
- 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 both **H.323 Station Outgoing Direct Media** and **Initial IP-IP Direct Media** to **n** so that the call is set up via the media gateway / media server, then shuffled to direct media.

The default values for the other fields may be used.

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

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

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
```

Note: The example shows **Far-end Domain** as blank. If required, this parameter can be used to treat calls from different domains differently, for example calls with CLIP could be received on one Trunk Group by specifying the enterprise domain name in one Signaling Group, and calls with CLIR could be received on a separate Trunk Group by specifying **anonymous.invalid** in another otherwise identical Signaling Group. The Trunk Group for calls with CLIR could then be given a meaningful name as described in **Section 5.6**.

In addition, **Initial IP-IP Direct Media** is shown as **n**. This setting provides a solution to the loss of media on calls forwarded on no answer from H.323 phones as described in **Section 2.2**, but introduces the issue of two ringback tones on outbound calls using the Media Server, also described in **Section 2.2**.

5.6. Administer SIP Trunk Groups

A trunk group is associated with the signaling 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** if the Diversion header is to be supported.
- Specify the **Signaling Group** defined in **Section 5.5** to be associated with this Trunk Group.
- Specify the **Number of Members** supported by this SIP Trunk Group.

```
add trunk-group 2                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 2                    Group Type: sip           CDR Reports: y
  Group Name: SIP_Trunk                   COR: 1        TN: 1        TAC: 102
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 2
                                                     Number of Members: 10
```

Note: The example shows **Group Name** as **SIP_Trunk**. If a single Trunk Group is used for incoming calls, the name may not be sufficiently meaningful for calls with CLIR as described in **Section 2.2**. In this case it may be preferable to use a Session Manager Adaptation as described in **Section 6.4** to remove the Privacy header. Removal of the Privacy header causes Communication Manager to display the user part of the From header which is "anonymous" for calls with CLIR received from Vodafone Libertel.

If a separate Trunk Group is used for calls with CLIR, a more meaningful name could be specified in the **Group Name** field such as "Number Withheld". Specify the Signaling Group defined in **Section 5.5** that has "anonymous.invalid" as the Far-end Domain. The Trunk Group would be otherwise identical to that used for calls with CLIP.

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 Vodafone Libertel to prevent unnecessary SIP messages during call setup. During testing, a value of **900** was used that sets Min-SE to 1800 in the SIP signalling.

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

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 5000

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

 Disconnect Supervision - In? y  Out? y
```

On **Page 3**, set the **Numbering Format** field to **private**. This allows delivery of CLI in formats other than E.164 with leading "+". In the test environment numbers were sent in diallable format, i.e. national numbers with a national dialling prefix of "0" and international numbers with an international dialling prefix of "00".

```
add trunk-group 2                                                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
```

Note: During testing, the **Hold/Unhold Notifications** field was set to **n** to avoid unnecessary signalling when placing a call on hold and taking it off hold.

On **Page 4** of this form:

- Set **Send Diversion Header** to **y.**
- Set the **Telephone Event Payload Type** to **101** to match the value preferred by Vodafone Libertel (this Payload Type is not applied to calls from SIP end-points).
- Set the **Identity for Calling Party Display** to **From** so that the number displayed on the Communication Manager extension is taken from the user part of the From header and not the P-Asserted-Identity header.

```
add trunk-group 2                                                 Page   4 of  21
                              PROTOCOL VARIATIONS

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

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


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

In the test environment, an additional Trunk Group was configured for on-net mobile calls so that a short CLI could be provided using the numbering tables as described in **Section 5.7**. The Trunk Group was configured with the same parameters as the previous one with the following differences:

- Choose a descriptive **Group Name** for the on-net mobile traffic.
- Specify a unique trunk access code (**TAC**) consistent with the dial plan.
- Specify the **Number of Members** supported by this SIP Trunk Group.

```
add trunk-group 3                                                 Page   1 of  21
                                TRUNK GROUP

Group Number: 3                   Group Type: sip          CDR Reports: y
  Group Name: On-net                     COR: 1        TN: 1        TAC: 103
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: public-ntwrk         Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 2
                                                     Number of Members: 10
```

Note: In the test environment the same Signaling Group was used as for the off-net fixed traffic.

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

Ext Ext             Trk        Private          Total
Len Code            Grp(s)     Prefix           Len
 4  2               1                           4     Total Administered: 11
 4  600             2          038700nnn9       10       Maximum Entries: 540
 4  2001            2          038700nnn0       10
 4  2291            2          038700nnn2       10
 4  2291            3          2092             4
 4  2316            2          038700nnn3       10
 4  2391            2          038700nnn1       10
 4  2391            3          2091             4
 4  2400            2          038700nnn4       10
 4  2401            2          038700nnn7       10
 4  7000            2          038700nnn5       10
```

Note: During testing the extension numbers were reformatted to national numbers for Trunk Group 2 only. The extension numbers were reformatted to on-net numbers for Trunk Group 3 as described in **Section 5.6**. The numbers were analysed for Trunk Group 1 but not reformatted as this Trunk Group is used for routing within the enterprise.

The public numbering table is used for numbers in E.164 format. Although this format is not used by Vodafone Libertel, the table was populated as entries are required for all extensions in both tables.

```
change public-unknown-numbering 0                                  Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                             Total
Ext Ext            Trk       CPN              CPN
Len Code           Grp(s)    Prefix           Len
                                                    Total Administered: 7
 4  2              1                          4        Maximum Entries: 240
 4  2001           2         3138700nnn0      11
 4  2291           2         3138700nnn2      11    Note: If an entry applies to
 4  2316           2         3138700nnn3      11    a SIP connection to Avaya
 4  2391           2         3138700nnn1      11    Aura(R) Session Manager,
 4  2400           2         3138700nnn4      11    the resulting number must
 4  2401           2         3138700nnn7      11    be a complete E.164 number.

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

Note: Although each extension is analysed in the example above, the single entry for Trunk Group 1 would suffice in this case.

5.8. Administer Uniform Dial Plan for On-net Mobile Calls

In the test environment, the Uniform Dial Plan (UDP) was used to analyse outbound on-net mobile calls for Automatic Route Selection (ARS). The UDP is invoked when numbers are dialled without an ARS or AAR Access Code that are not directly connected extensions. Use the **change uniform-dialplan n** command to configure the UDP where **n** is the first digit.

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

  Matching                   Insert              Node
  Pattern       Len Del      Digits     Net Conv Num
 2               4   0                  aar  n
 7000            4   0                  aar  n
 790             4   0                  ars  n
                                             n
```

The above example shows **790n** as the on-net mobile short code numbering range.

5.9. 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 Vodafone Libertel 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: 8
    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. In the example shown, off-net calls are sent to **Route Pattern 2** and on-net mobile calls are sent to **Route Pattern 3**.

```
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                      8    12    2        pubu         n
    00                     13   15    2        pubu         n
    001                    13   13    2        pubu         n
    0035391                13   13    2        pubu         n
    1                      3    4     2        pubu         n
    118                    5    6     2        pubu         n
    790                    4    4     3        pubu         n
                                                            n
```

Use the **change route-pattern n** command, where **n** is an available route pattern, to add the SIP trunk group to the route pattern that ARS selects. In this configuration, route pattern **2** is used to route calls to trunk group **2** and route pattern **3** is used to route calls to trunk group **3**. **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 **lev0-pvt** to ensure that calling party number was not prefixed with a leading "+".

```
change route-pattern 2                                           Page   1 of   3
                    Pattern Number: 2      Pattern Name: Public
    SCCAN? n    Secure SIP? n     Used for SIP stations? n

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

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM Sub  Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
 1: y y y y y n  n            rest                               lev0-pvt  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
```

The example shows **route-pattern 2**. Route pattern 3 is identical apart from the **Grp No** and **Pattern Name**

5.10. Administer Incoming Digit Translation

This step configures the settings necessary to map incoming DDI calls to Communication Manager extensions. The incoming digits sent in the INVITE message from Vodafone Libertel can be manipulated as necessary to route calls to the desired extension. Use the **change inc-call-handling-trmt trunk-group x** command where **x** is the Trunk Group defined in **Section 5.6**.

In the example shown, 10 digit numbers are received in national format for incoming off-net calls. All digits are deleted and the extension number is inserted. Note that some of the DDI digits have been obscured. In addition, 4 digit short code numbers are received for incoming on-net calls. In the test environment, all digits are deleted and the extension number is inserted. In the live network, the short codes are likely to match the extension numbers.

```
change inc-call-handling-trmt trunk-group 2                     Page   1 of   3
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 public-ntwrk    10 038700nnn0       10  2001
 public-ntwrk    10 038700nnn1       10  2391
 public-ntwrk    10 038700nnn2       10  2291
 public-ntwrk    10 038700nnn3       10  2316
 public-ntwrk    10 038700nnn4       10  2400
 public-ntwrk    10 038700nnn5       10  7000
 public-ntwrk    10 038700nnn6       10  3
 public-ntwrk    10 038700nnn7       10  2401
 public-ntwrk    10 038700nnn8       10  6002
 public-ntwrk    4  2091             4   2391
 public-ntwrk    4  2092             4   2291
 public-ntwrk
```

5.11. 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 2291. 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. **0035389434nnnn**).
- 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 2291                Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 2291             OPS          -      2291             aar        1
 2291             EC500        -      0035389434nnnn   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. Note also that the station shown is a SIP phone so is also in the off-PBX station mapping as an Off-PBX Station (OPS).

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

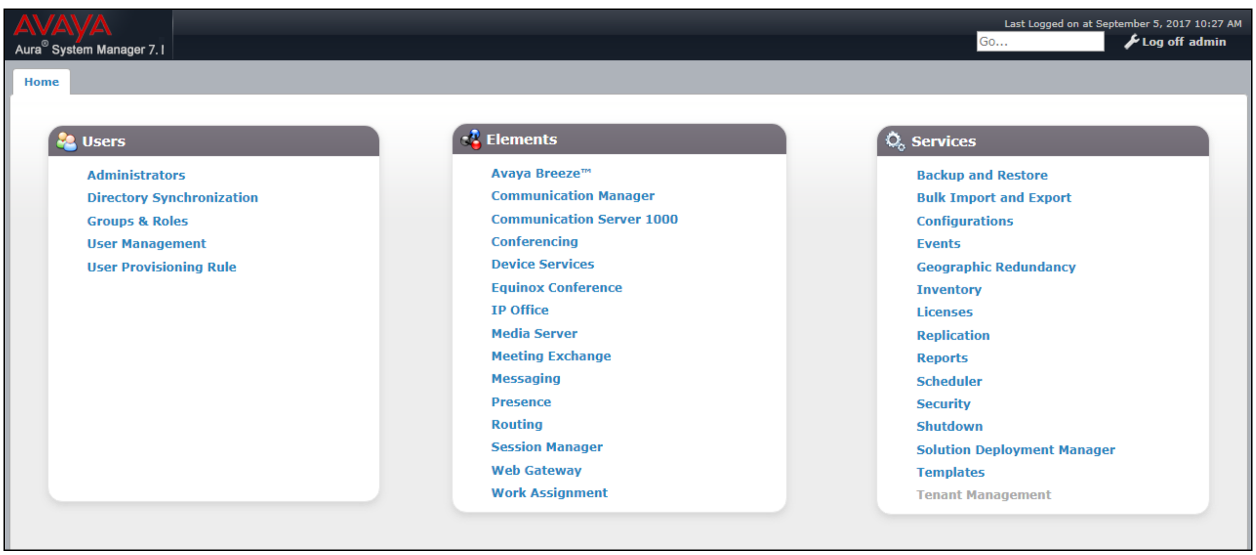
6. Configuring Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. Session Manager is configured by opening a web browser to System Manager. The procedures include the following areas:

- Log in to Avaya Aura® System Manager
- Administer SIP domain
- Administer Locations
- Administer Adaptations
- Administer SIP Entities
- Administer Entity Links
- Administer Routing Policies
- Administer Dial Patterns
- Administer Application for Avaya Aura® Communication Manager
- Administer Application Sequence for Avaya Aura® Communication Manager
- Administer SIP Extensions

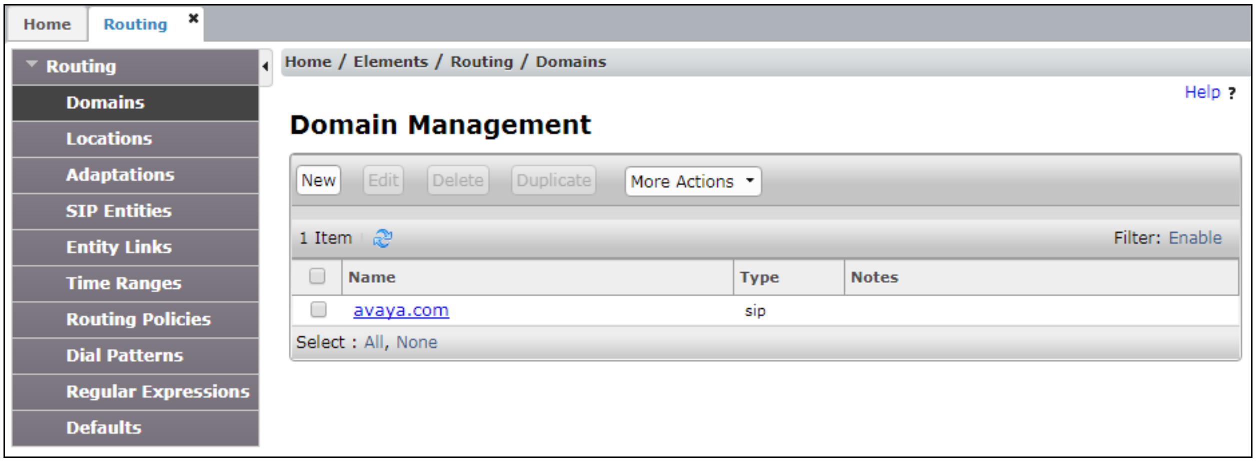
6.1. Log in to Avaya Aura® System Manager

Access 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 **Home** tab will be presented with menu options shown below.



6.2. Administer SIP Domain

To add the SIP domain that will be used with Session Manager, select **Routing** from the **Home** tab menu and in the resulting tab select **Domains** from the left hand menu. Click the **New** button to create a new SIP domain entry. In the **Name** field enter the domain name of the enterprise site or a name agreed with Vodafone Libertel; this will be the same as specified in the Authoritative Domain specified in the IP Network Region on Communication Manager. Refer to **Section 5.3** for details. In the test environment, **avaya.com** was used. Optionally, a description for the domain can be entered in the Notes field (not shown). Click **Commit** to save changes.

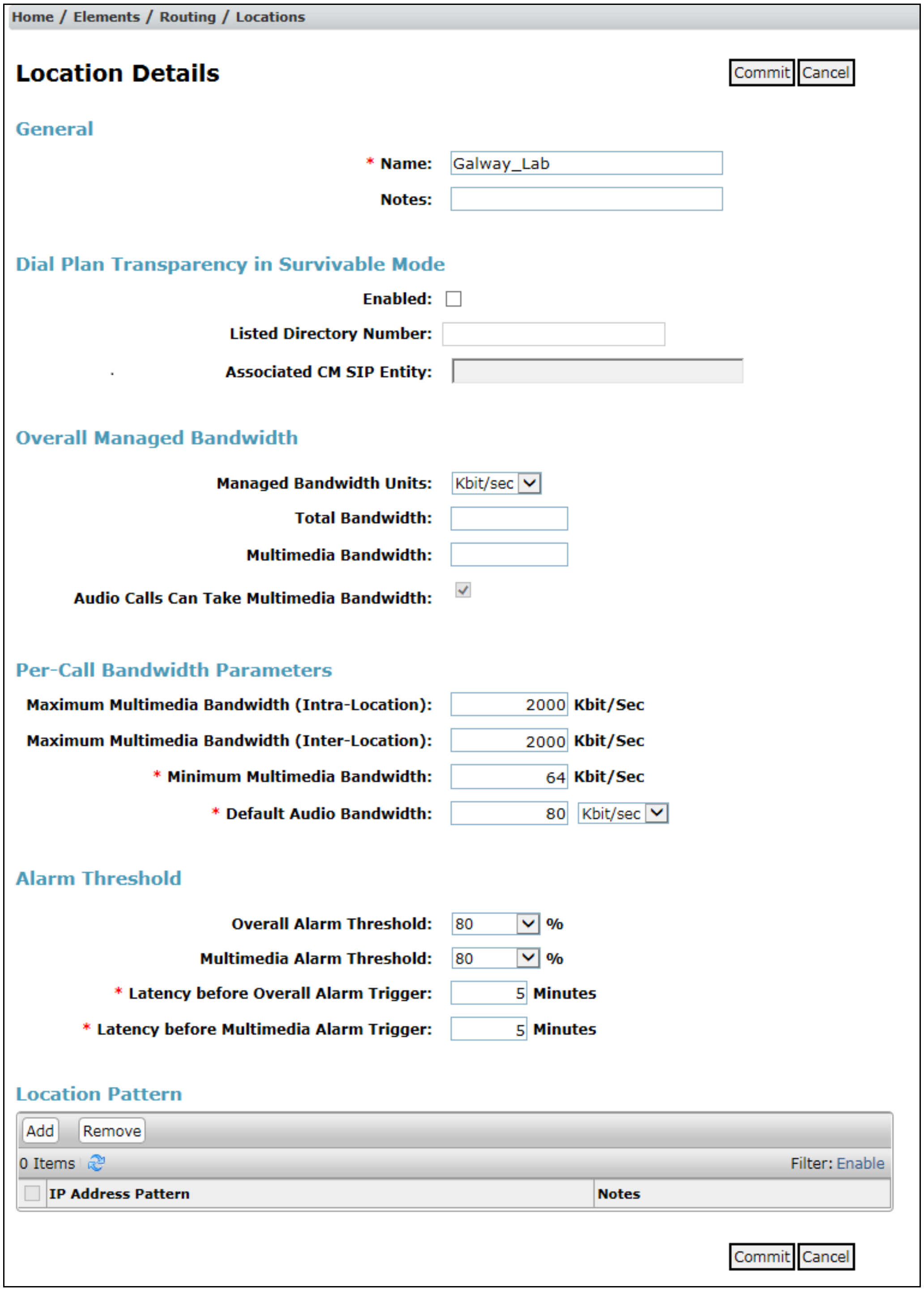


Note: If the existing domain name used in the enterprise equipment does not match that used in the network, a Session Manager Adaptation can be used to change it (see **Section 6.4**).

6.3. Administer Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for the purposes of bandwidth management. One location is added to the sample configuration for all of the enterprise SIP entities and another for the Vodafone Libertel SIP trunk. The two Locations were named Galway_Lab and Service_Provider and were identical in every other way.

On the **Routing** tab select **Locations** from the left hand menu (not shown). Under **General**, in the **Name** field, enter an informative name for the location. Below is the location configuration used for the test enterprise.



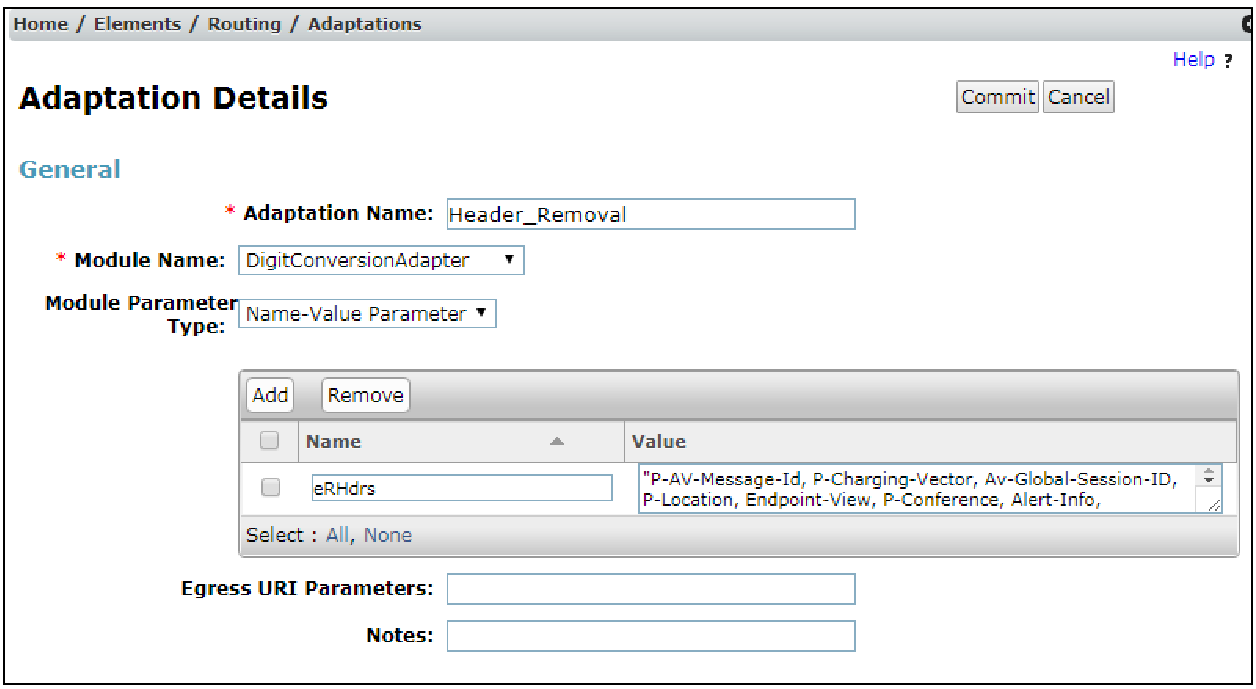
Note: Location Pattern can be used to refine the location down to specific subnets. That refinement was not required in the test environment.

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 from Session Manager. In addition, an Adaptation was tested that removed the Privacy header from calls with Calling Line Identity Restricted (CLIR).

Communication Manager and Session Manager make use of Avaya proprietary SIP headers to facilitate the full suite of Avaya functionality within the enterprise. These are not required on the SIP trunk however, and make the SIP messages unnecessarily large. A Session Manager Adaptation is used to remove proprietary headers. On the **Routing** tab select **Adaptations** from the left-hand menu. Click on **New** (not shown).

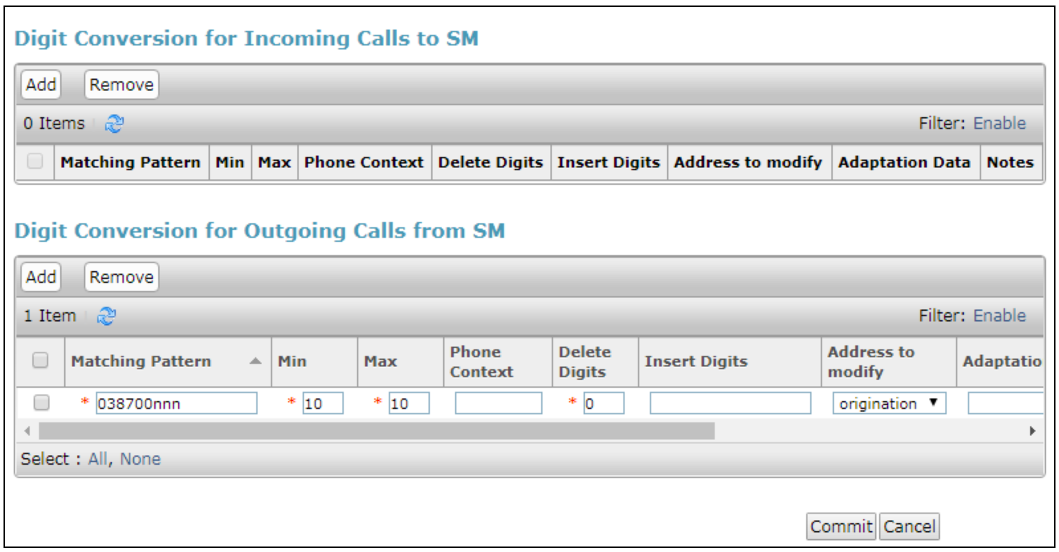
- In the **Adaptation Name** field, enter a descriptive title for the adaptation.
- In the **Module Name** drop down menu, select **DigitConversionAdapter**.
- In the **Module Parameter Type** drop down menu, select **Name-Value Parameter**.
- In the **Name** box, type **eRHdrs**.
- In the **Value** box, type the list of headers to be deleted. During testing, the following list was used: **"P-AV-Message-Id, P-Charging-Vector, Av-Global-Session-ID, P-Location, Endpoint-View, P-Conference, Alert-Info, Correlation-ID, Accept-Language"**.



This Adaptation is intended for traffic from Session Manager to the Avaya SBCE and was applied to the Avaya SBCE SIP Entities as described in **Section 6.5**. To apply the module, it is matched to the calling party number of calls from Session Manager, this is the DDI assigned to the Communication Manager extension.

Scroll down and in the section **Digit Conversion for Outgoing Calls from SM**, click on **Add**. An additional row will appear for digit manipulation.

- Enter a **Matching Pattern** to identify calls for which the Adaptation is required. In the test environment, this was the calling party number of the Communication Manager extensions.
- Enter the **Min** and **Max** values, the example is for **10** digit numbers only.
- Enter **Delete Digits**, a value of **0** ensures that the number is analysed but not modified.
- Select **origination** from the **Address to modify** drop down menu as matching is only to be done on the calling party number.

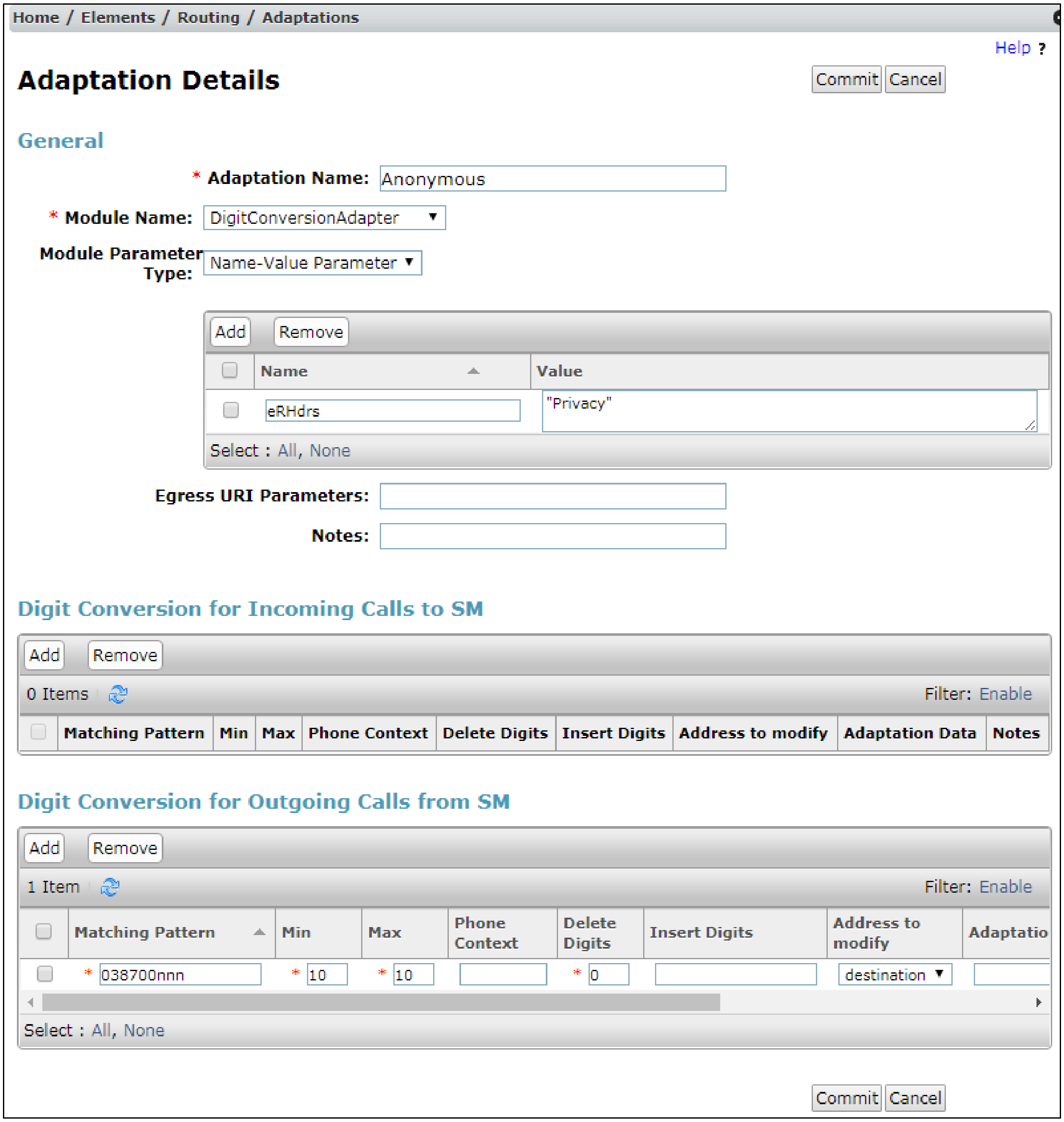


An additional Adaptation was tested that removed the Privacy header from calls with CLIR so that Communication Manager extensions displayed the user part of the From header. This was one method of ensuring a meaningful display for these calls as described in **Section 2.2**.

This Adaptation is intended for traffic from Session Manager to Communication Manager and was applied to the Communication Manager SIP Entity as described in **Section 6.5**. To apply the module, it is matched to the called party number of calls from Session Manager, this is the DDI assigned to the Communication Manager extension. The following parameters are set differently:

- In the **Adaptation Name** field, enter a descriptive title for the adaptation. In the test environment this was **Anonymous**.
- In the **Value** box, type **"Privacy"**.
- Scroll down and in the section **Digit Conversion for Outgoing Calls from SM**, click on **Add**. An additional row will appear for digit manipulation.
- Define the same matching pattern as previously, but select **destination** in the **Address to modify** drop down menu.

The screenshot over the page shows the complete Adaptation.


Home / Elements / Routing / Adaptations
Help ?
Adaptation Details
Commit Cancel
General
* Adaptation Name: Anonymous
* Module Name: DigitConversionAdapter
Module Parameter Type: Name-Value Parameter
Add Remove
Name Value
eRHdrs "Privacy"
Select : All, None
Egress URI Parameters:
Notes:
Digit Conversion for Incoming Calls to SM
Add Remove
0 Items
Filter: Enable
Matching Pattern Min Max Phone Context Delete Digits Insert Digits Address to modify Adaptation Data Notes
Digit Conversion for Outgoing Calls from SM
Add Remove
1 Item
Filter: Enable
Matching Pattern Min Max Phone Context Delete Digits Insert Digits Address to modify Adaptatio
038700nnn 10 10 0 destination
Select : All, None
Commit Cancel

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**:

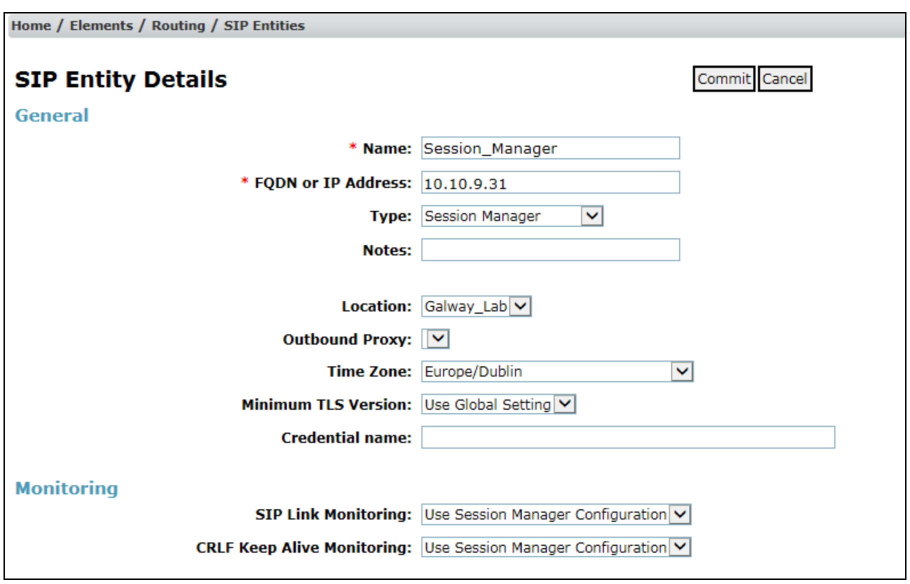
- In the **Name** field enter an informative name.
- In the **FQDN or IP Address** field enter the IP address of Session Manager or the signalling interface on the connecting system.
- In the **Type** field use **Session Manager** for a Session Manager SIP Entity, **CM** for a Communication Manager SIP Entity and **SIP Trunk** for the Avaya SBCE SIP Entity.
- In the **Adaptation** field (not available for the Session Manager SIP Entity), select the appropriate Adaptation from the drop down menu.
- 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 five SIP Entities:

- Avaya Aura® Session Manager SIP Entity.
- Avaya Aura® Communication Manager SIP Entity for the SIP Endpoints.
- Avaya Aura® Communication Manager SIP Entity for the SIP Trunk.
- Avaya Session Border Controller for Enterprise (Avaya SBCE) SIP Entity for off-net PSTN destinations.
- Avaya Session Border Controller for Enterprise (Avaya SBCE) SIP Entity for on-net mobile destinations.

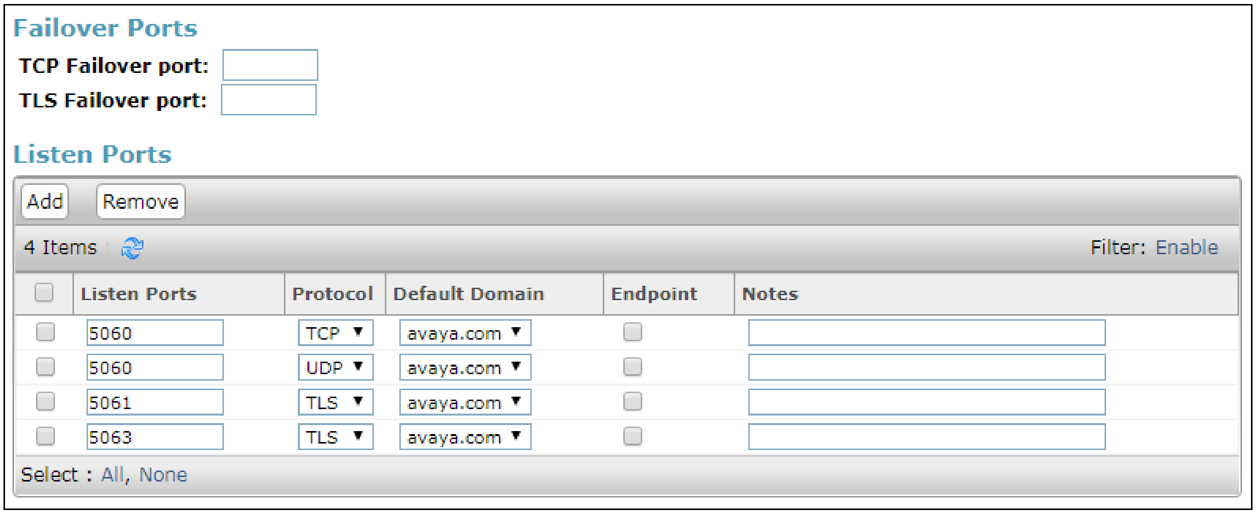
6.5.1. Avaya Aura® Session Manager SIP Entity

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.



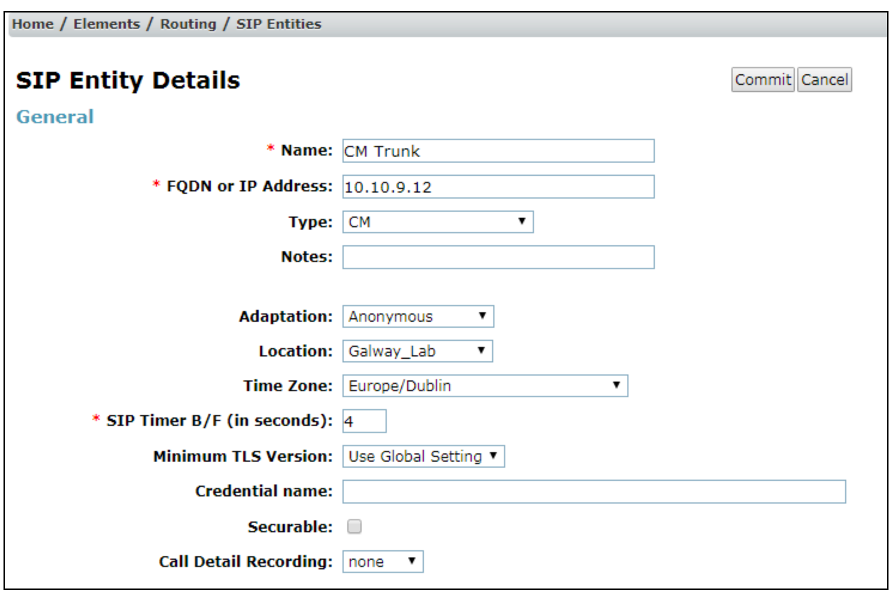
Session Manager must be configured with the port numbers on the protocols that will be used by the other SIP entities. To configure these scroll to the bottom of the page and under **Port**, click **Add**, then edit the fields in the resulting new row.

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

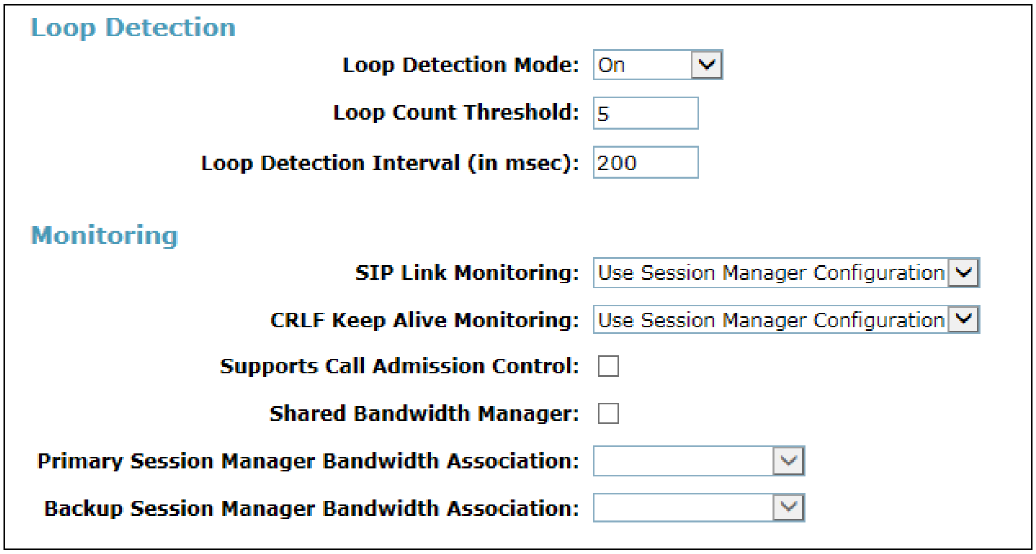


6.5.2. Avaya Aura® Communication Manager SIP Entities

The following screen shows one of the SIP entities 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**.



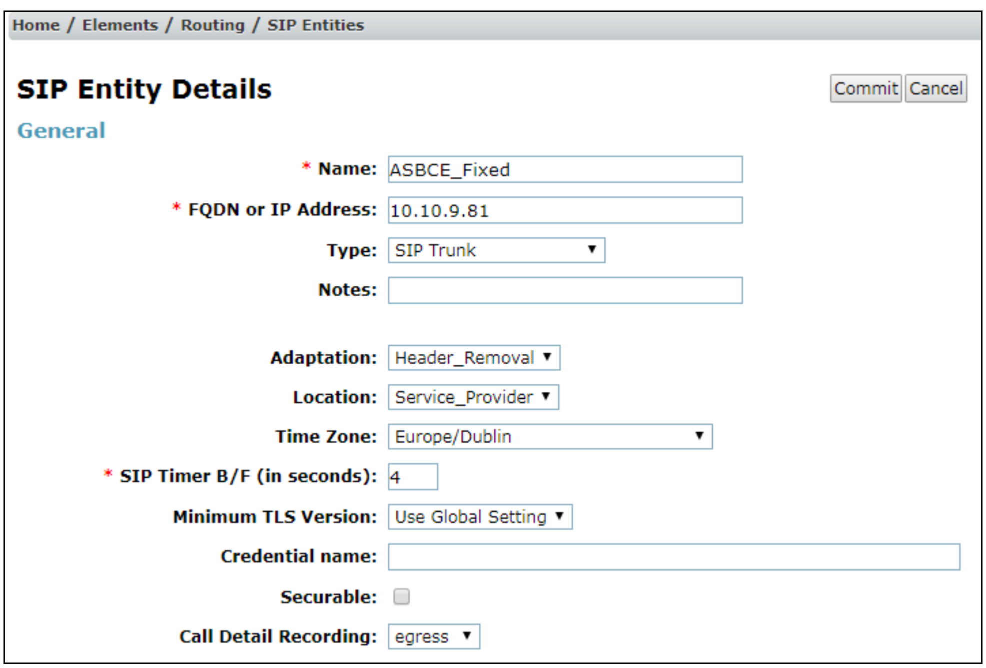
Other parameters can be set for the SIP Entity as shown in the following screenshot, but for test, these were left at default values.



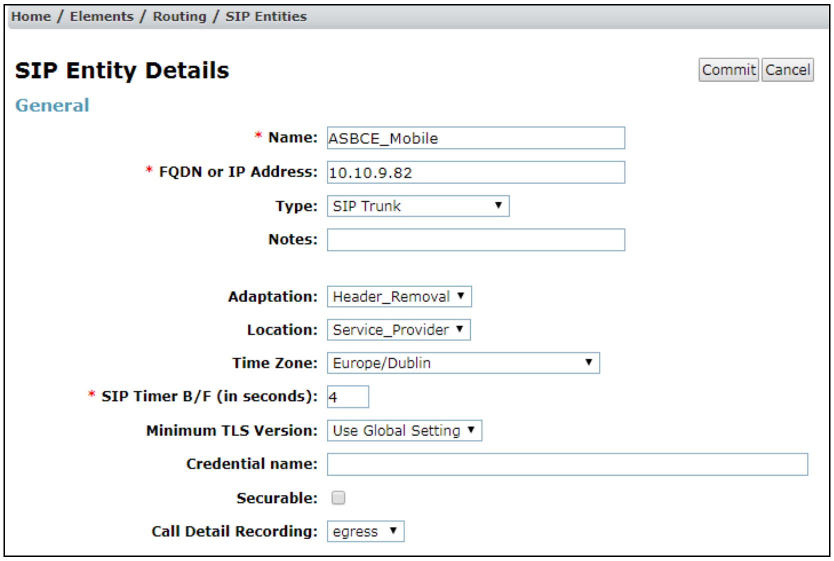
Note: A second SIP Entity for Communication Manager is required for SIP Endpoints. In the test environment this is named "CM_SIP_Endpoints".

6.5.3. Avaya Session Border Controller for Enterprise SIP Entity

The following screen shows the SIP Entity for the Avaya SBCE used for PSTN destinations. The **FQDN or IP Address** field is set to the IP address of the Avaya SBCE private network interface used for PSTN fixed calls (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.



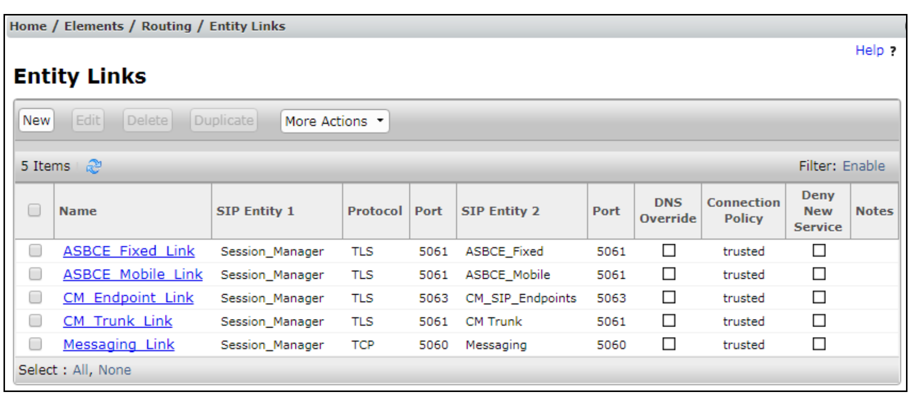
The next screen shows the SIP Entity for the Avaya SBCE used for mobile destinations. The **FQDN or IP Address** field is set to the IP address of the Avaya SBCE private network interface used for mobile destinations (See **Section 7.4.1**). Set the **Adaptation**, **Location** and **Time Zone** as for the Avaya SBCE SIP Entity used for PSTN fixed destinations.



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 **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. For the Avaya SBCE, this matches the port defined in **Section 7.4.1**.
- Select the **Trusted** tick box to make the other system trusted.
- In the **Protocol** field enter the transport protocol to be used to send SIP requests.



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

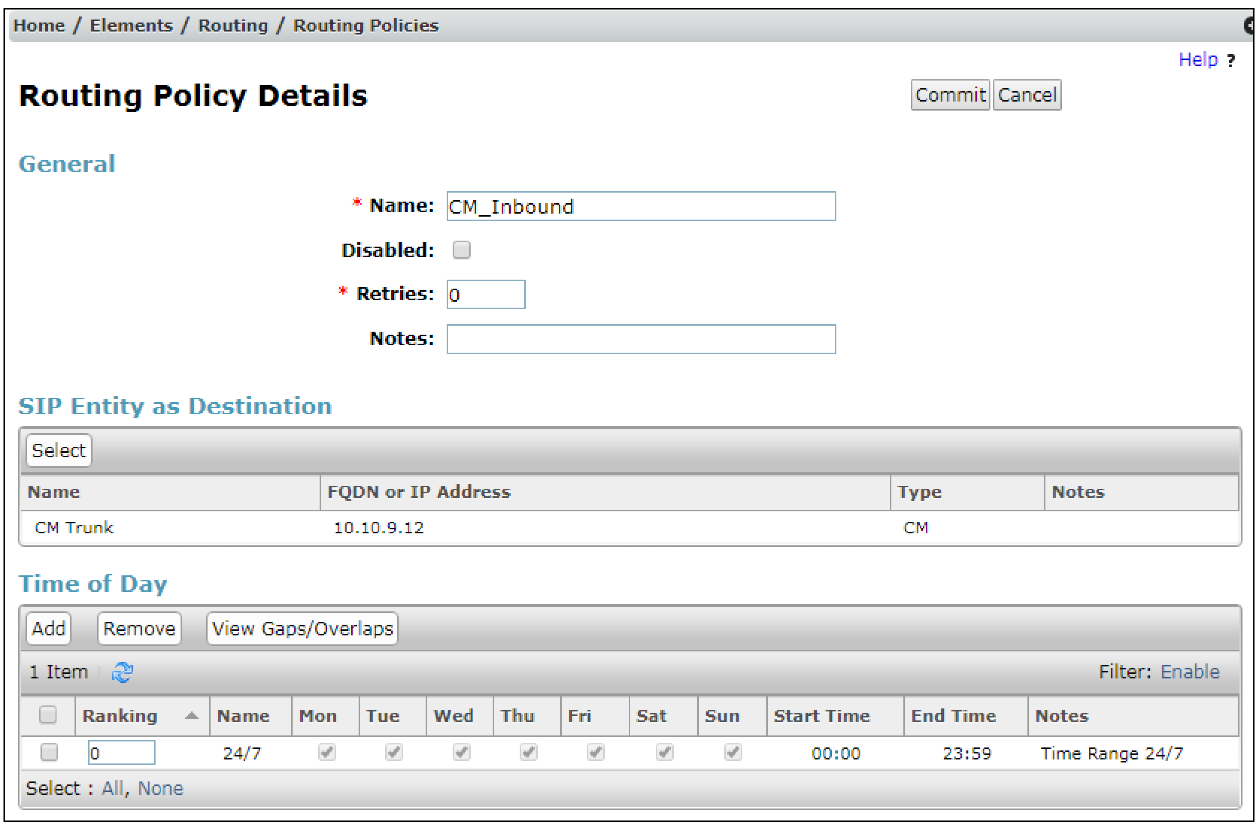
Note: There are two Entity Links for Communication Manager, one for the SIP Endpoints and the other for the SIP Trunk. These are differentiated by port number. There are also two Entity Links for the Avaya SBCE, one for PSTN destinations and the other for mobile destinations. The **Messaging_Link** Entity Link is used for the Avaya Aura ® Messaging system and is not described in this document.

6.7. Administer Routing Policies

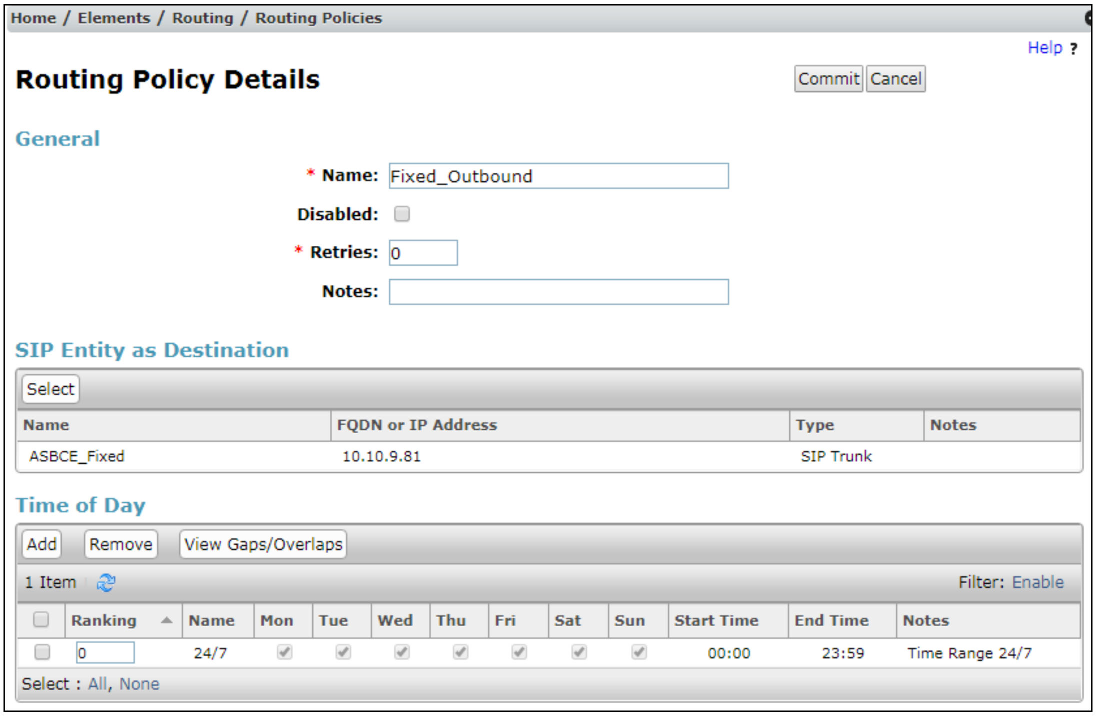
Routing policies must be created to direct how calls will be routed to a system. To add a routing policy, select **Routing Policies** on the left panel menu and then click on the **New** button (not shown). Under **General**:

- 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

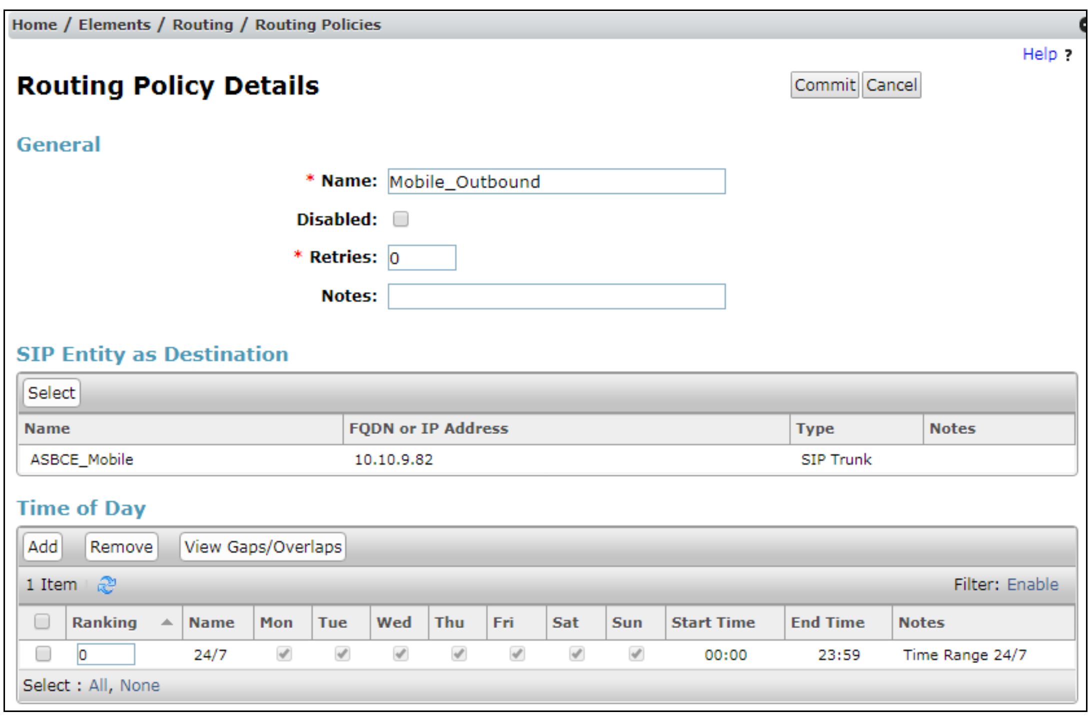
The following screen shows the routing policy for calls inbound from the SIP Trunk to Communication Manager.



The following screen shows the Routing Policy for the Avaya SBCE interface that will be routed off-net to PSTN fixed destinations via the Vodafone Libertel SIP Trunking Service.



The next screen shows the Routing Policy for the Avaya SBCE interface that will be routed on-net to mobile destinations via the Vodafone Libertel SIP Trunking Service.



6.8. Administer Dial Patterns

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

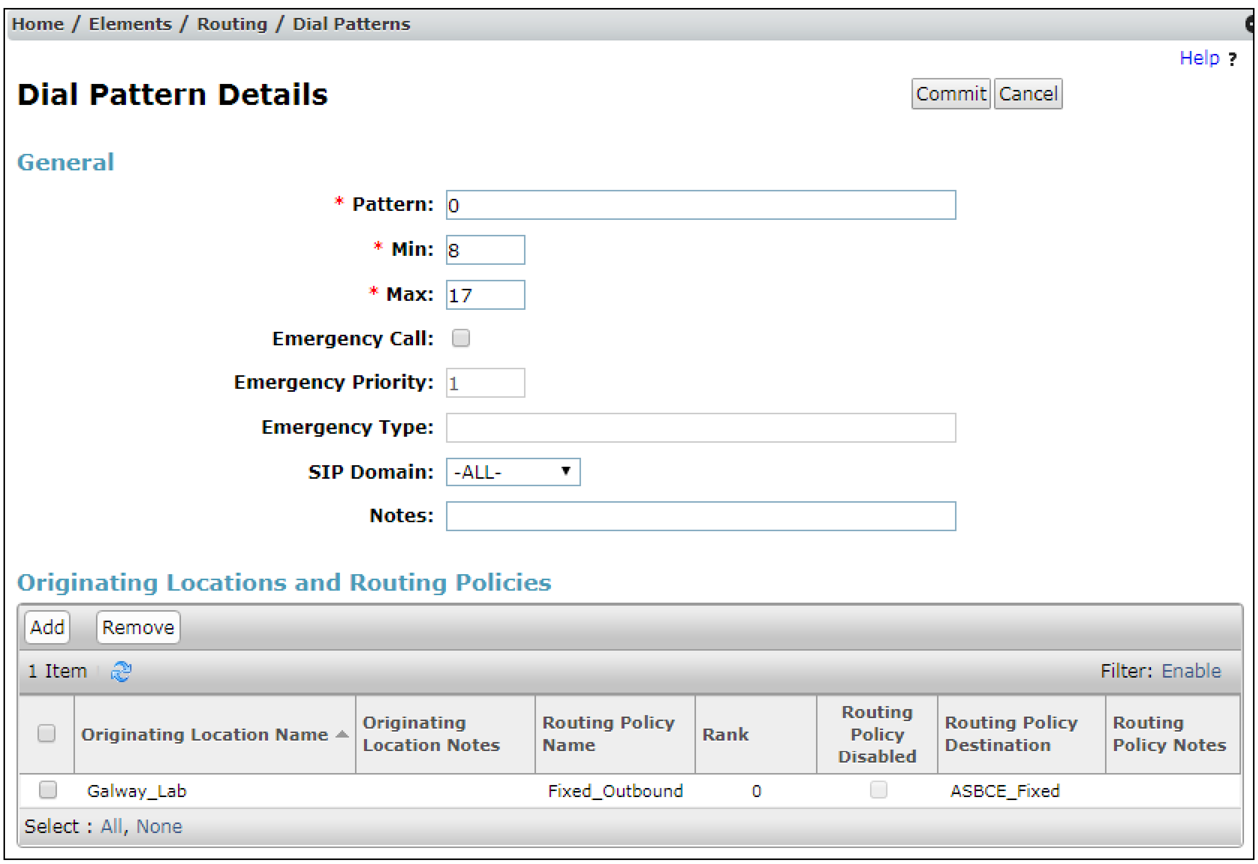
Under **General**:

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

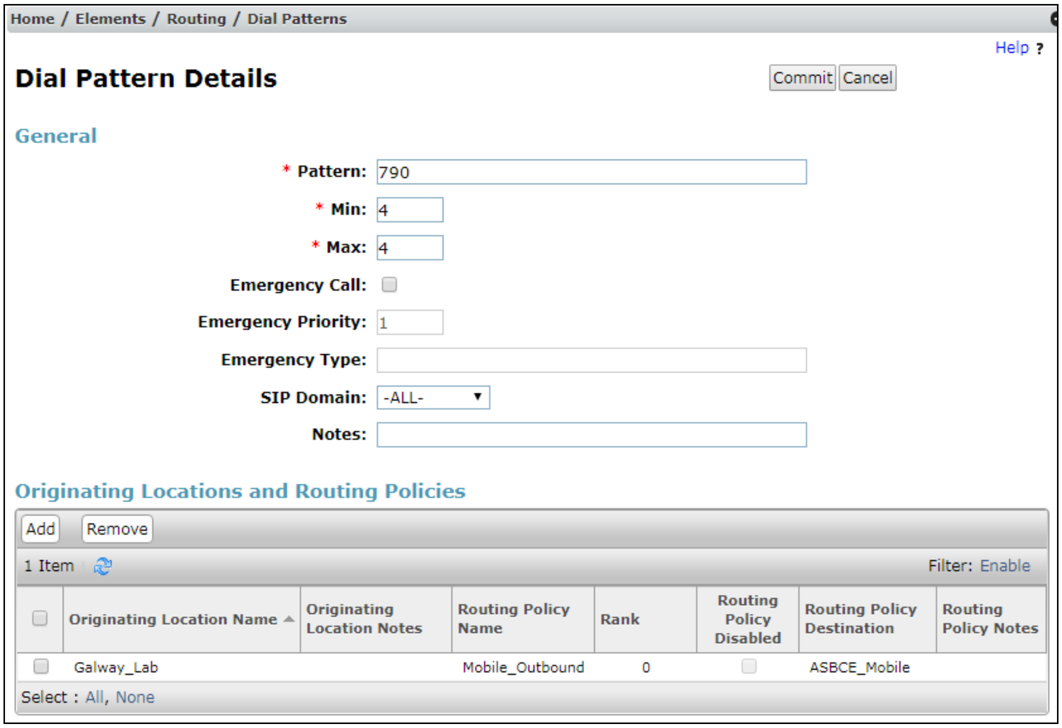
Under **Originating Locations and Routing Policies**:

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

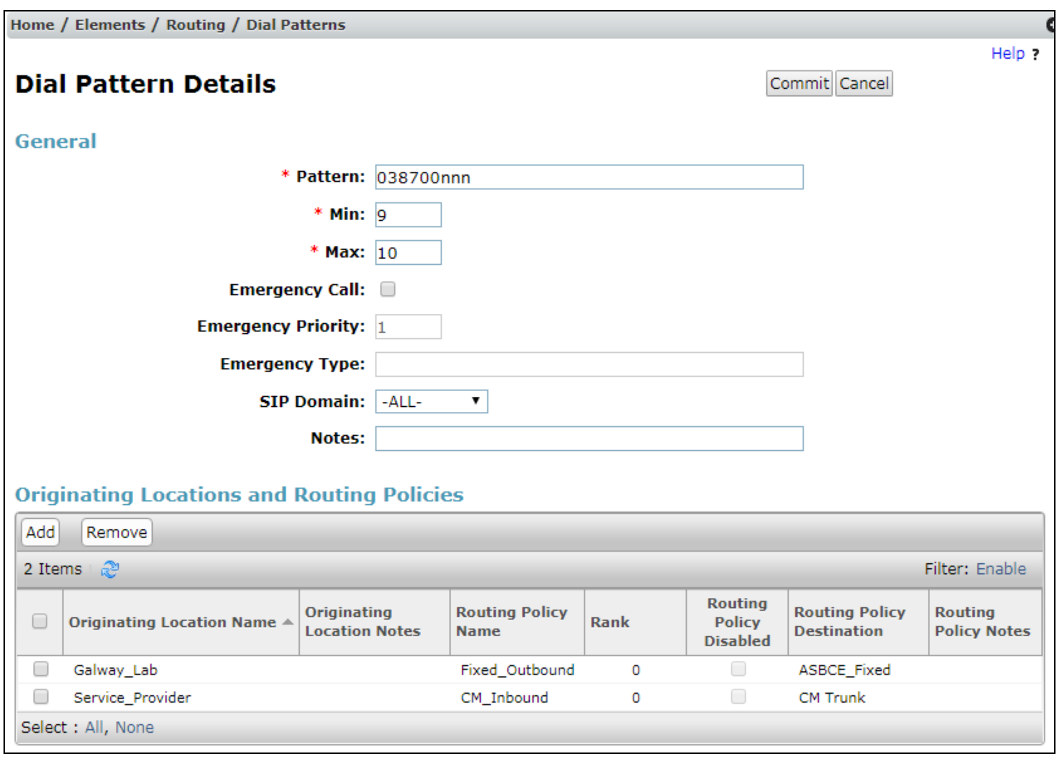
The following screen shows an example dial pattern configured for the Avaya SBCE which will route the calls off-net to PSTN destinations via the Vodafone Libertel SIP Trunking Service.



The next screen shows an example dial pattern configured for the Avaya SBCE which will route the calls on-net to mobile destinations via the Vodafone Libertel SIP Trunking Service.

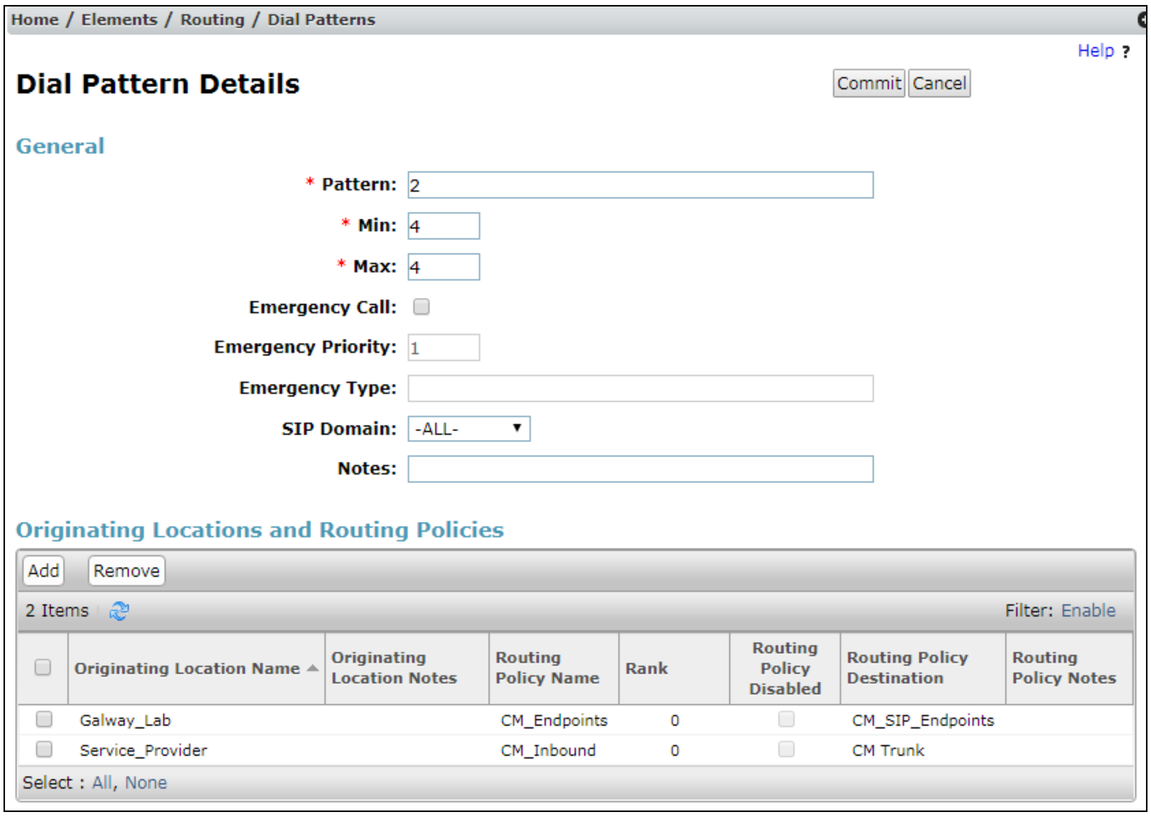


The following screen shows the test dial pattern configured for assigned DDI numbers.



Note: The configuration on the previous page was used to analyse the DDI numbers assigned to the extensions on Communication Manager. It was required for testing that a DDI number could be dialled from a Communication Manager extension and the call would route via the network and back to the enterprise. To allow this, locations were used so that if the call originated in the network, it would route to the enterprise. If the call originated in the enterprise, it would route to the network. Some of the digits of the pattern to be matched have been obscured.

The following screen shows the test dial pattern configured for Communication Manager extension numbers and on-net numbers.

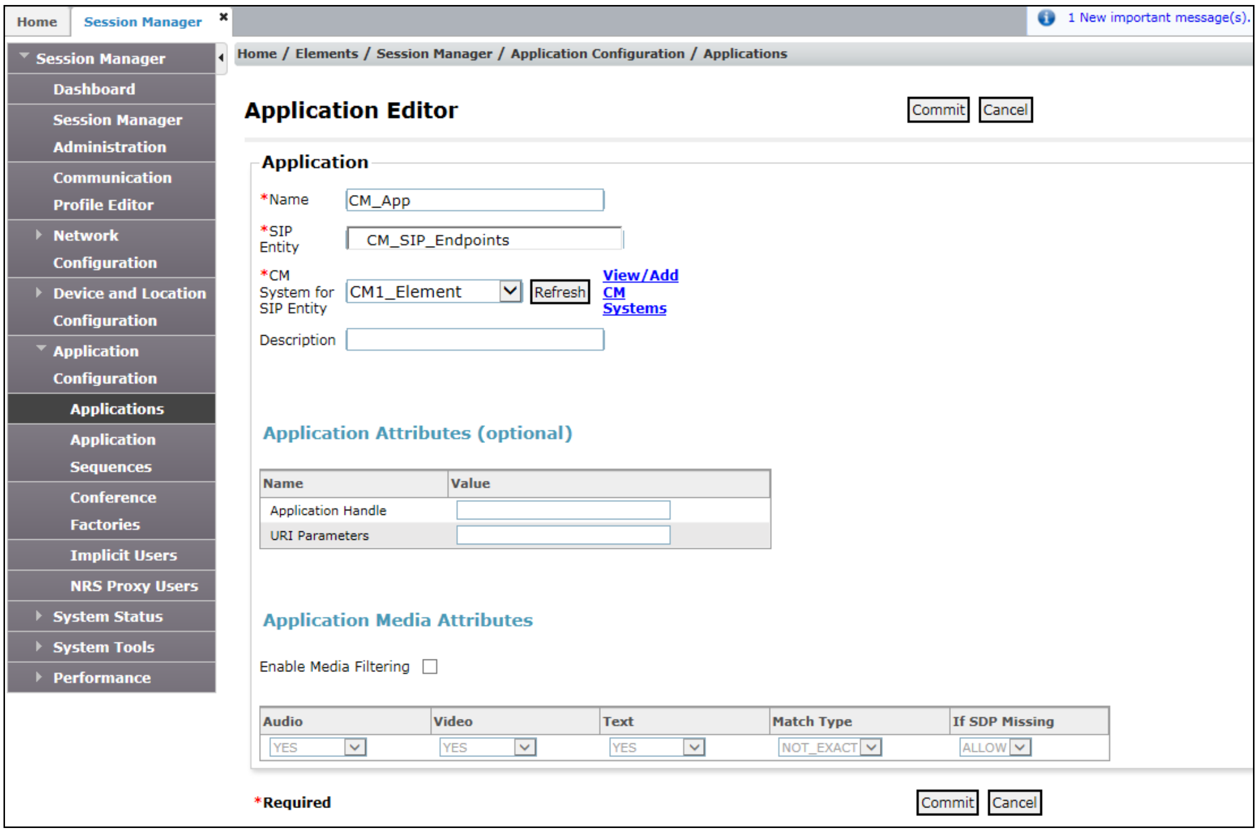


Note: The configuration above was used to analyse the extension numbers on Communication Manager and the on-net numbers which may be the same in the live network. In the test environment, they were different but both were **4** digit numbers starting with **2**. Locations were specified to use different Entity Links depending on whether the call was within the enterprise or an incoming on-net call. Examples of calls within the enterprise are voicemail and off-PBX endpoints such as SIP phones.

6.9. Administer Application for Avaya Aura® Communication Manager

From the **Home** tab select **Session Manager** from the menu. In the resulting tab from the left panel menu select **Application Configuration → Applications** and click **New** (not shown).

- In the **Name** field enter a name for the application.
- In the **SIP Entity** field select the SIP entity for Communication Manager.
- In the **CM System for SIP Entity** field select the SIP entity for Communication Manager and select **Commit** to save the configuration.

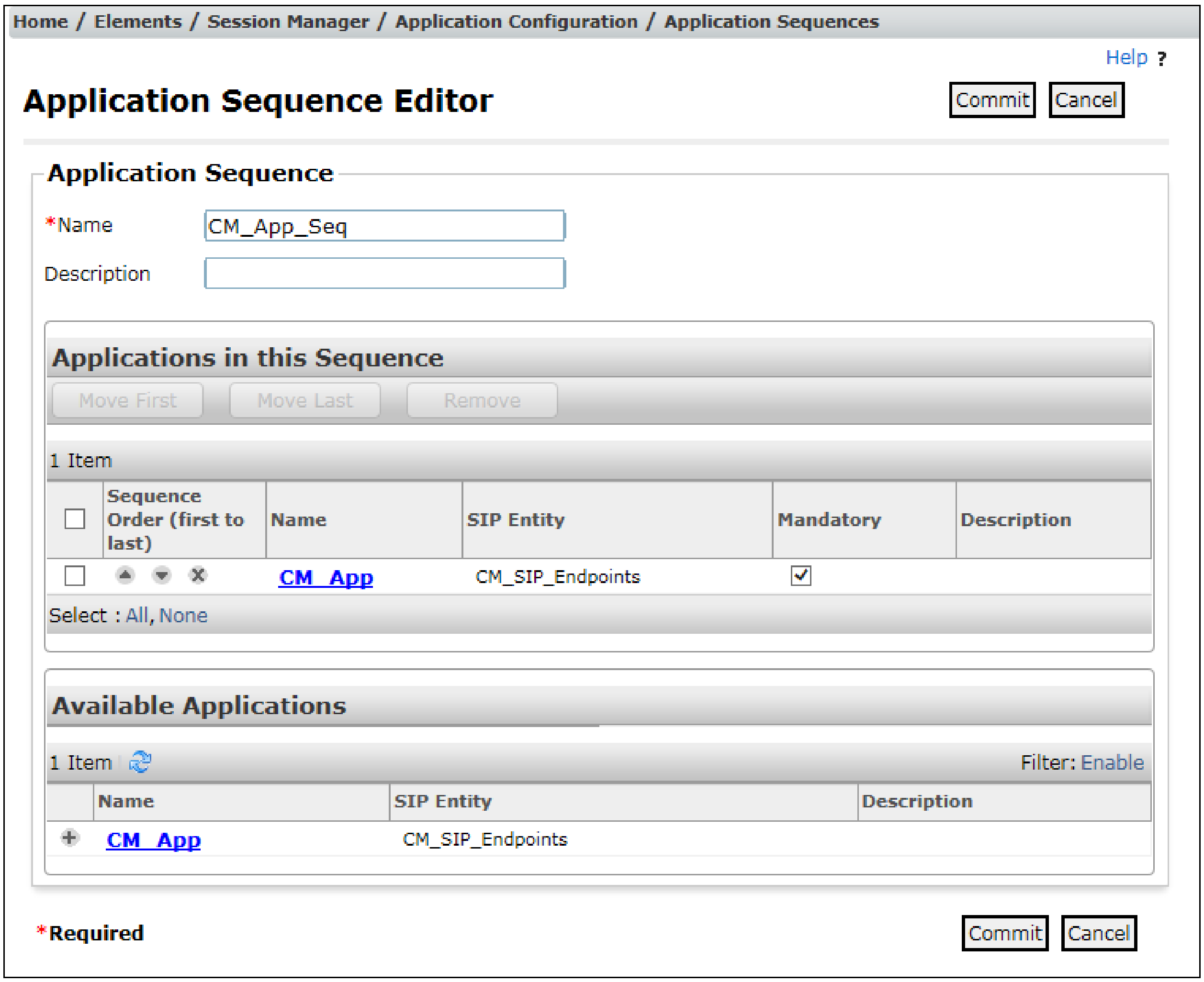


Note: The Application described here and the Application Sequence described in the next section are likely to have been defined during installation. The configuration is shown here for reference. Note also that the Communication Manager SIP Entity selected is that set up specifically for SIP endpoints. In the test environment there is also a Communication Manager SIP Entity that is used specifically for the SIP Trunk and is not to be used in this case.

6.10. Administer Application Sequence for Avaya Aura® Communication Manager

From the left panel navigate to **Session Manager → Application Configuration → Application Sequences** and click on **New** (not shown).

- In the **Name** field enter a descriptive name.
- Under **Available Applications**, click the + sign in front of the appropriate application instance. When the screen refreshes the application should be displayed under the **Applications in this Sequence** heading. Select **Commit**.

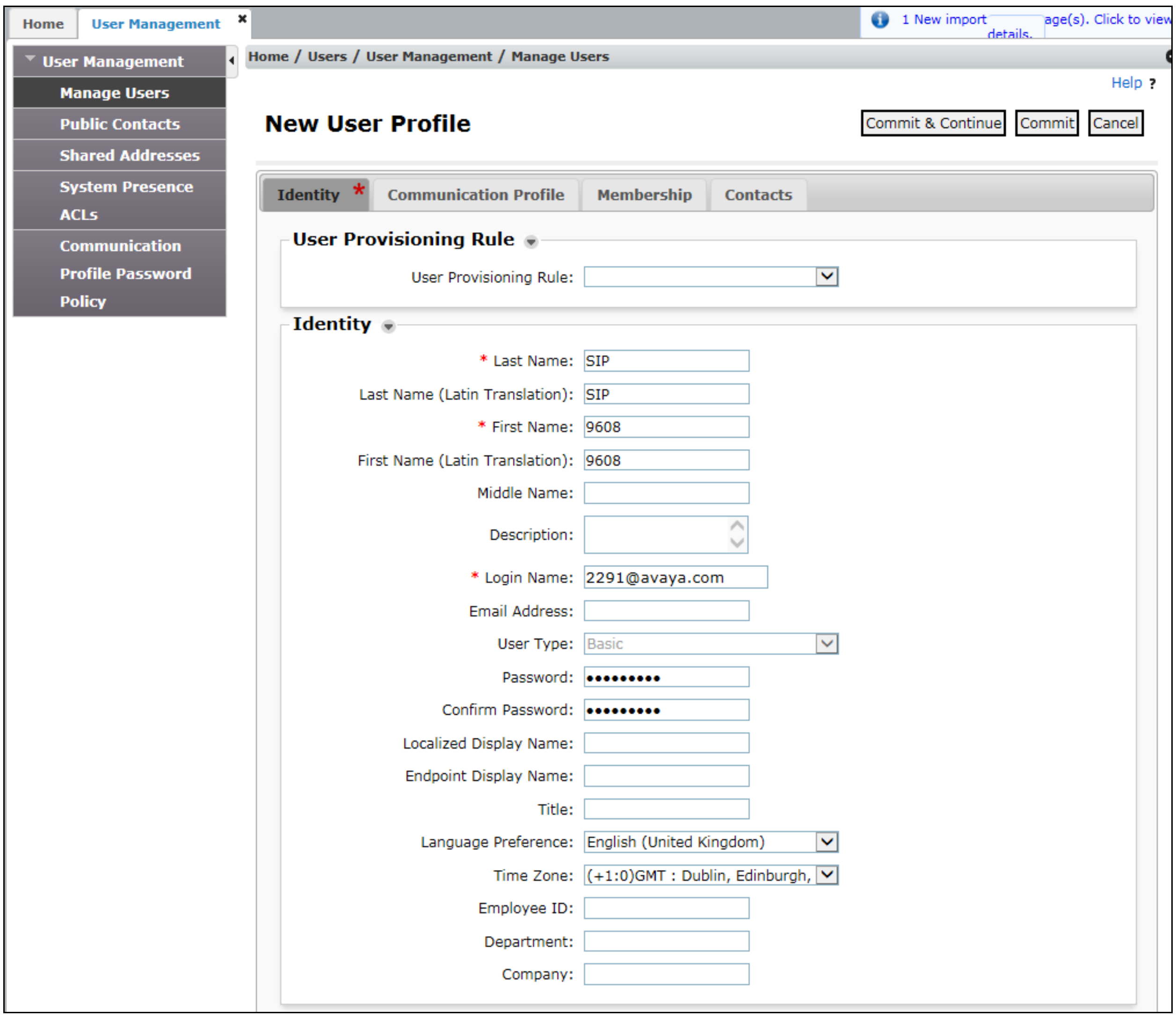


6.11. Administer SIP Extensions

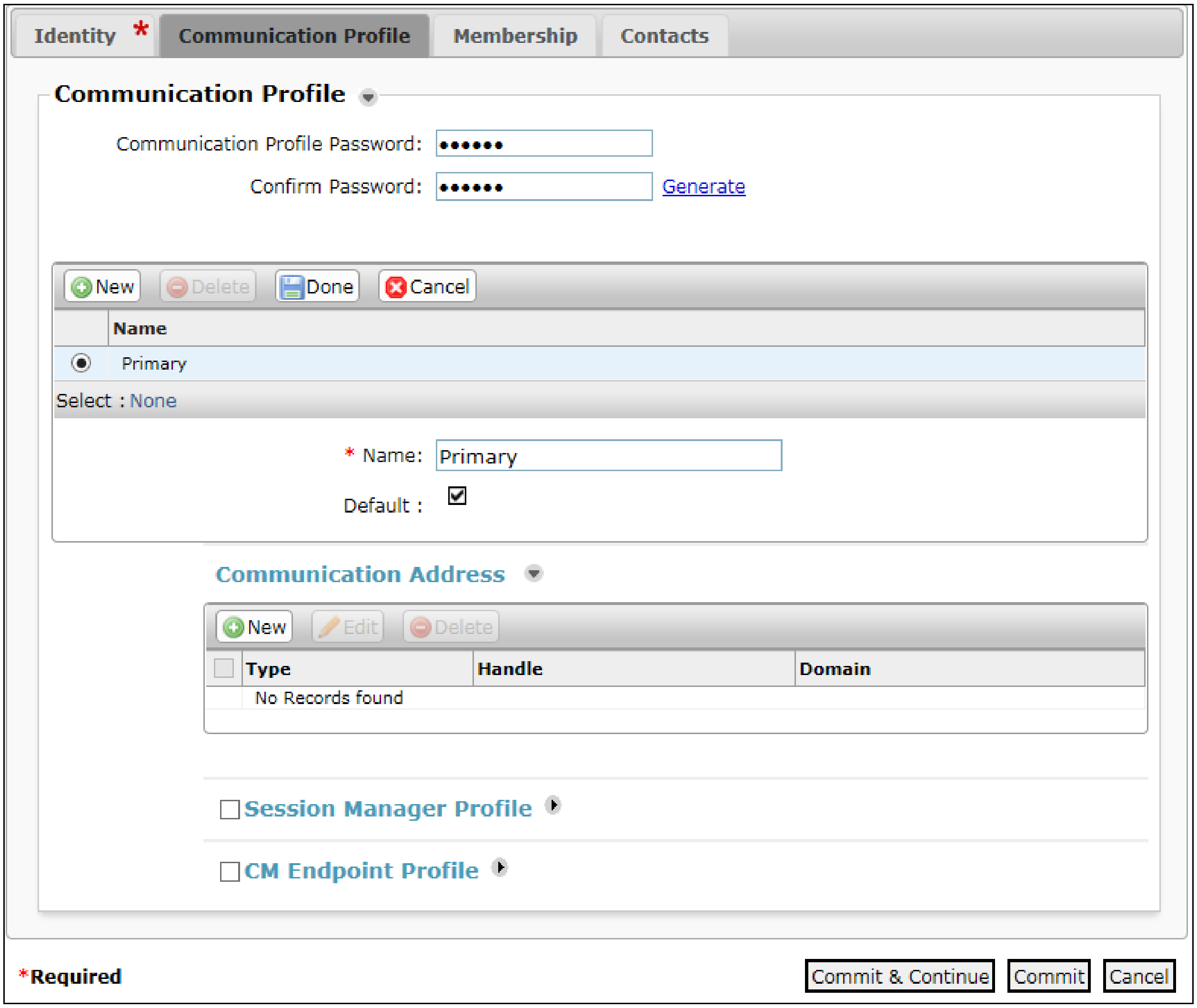
The SIP extensions are likely to have been defined during installation. The configuration shown in this section is for reference. SIP extensions are registered with Session Manager and use Communication Manager for their feature and configuration settings. From the **Home** tab select **User Management** from the menu. Then select **Manage Users** and click **New** (not shown).

On the **Identity** tab:

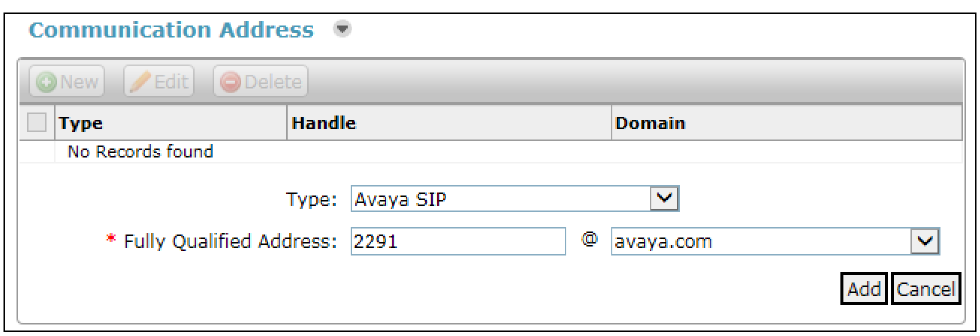
- Enter the user's name in the **Last Name** and **First Name** fields.
- In the **Login Name** field enter a unique system login name in the form of user@domain e.g. **2291@avaya.com** which is used to create the user's primary handle.
- The **Authentication Type** should be **Basic**.
- In the **Password/Confirm Password** fields enter an alphanumeric password.
- Set the **Language Preference** and **Time Zone** as required.



In the **Communication Profile** tab, enter a numeric **Communication Profile Password** and confirm it.

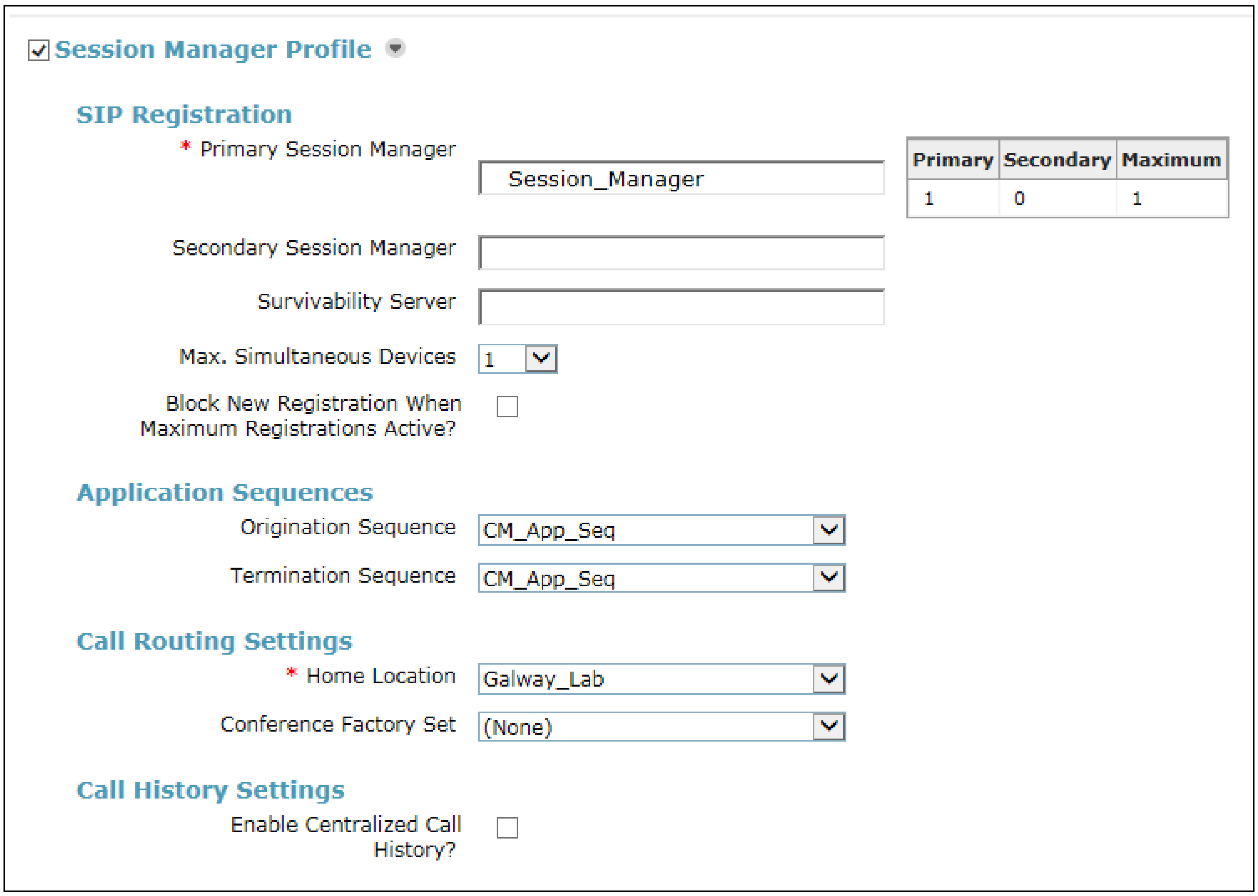


Expand the **Communication Address** section and click **New**. For the **Type** field select **Avaya SIP** from the drop-down menu. In the **Fully Qualified Address** field, enter an extension number and select the relevant domain from the drop-down menu. Click the **Add** button.



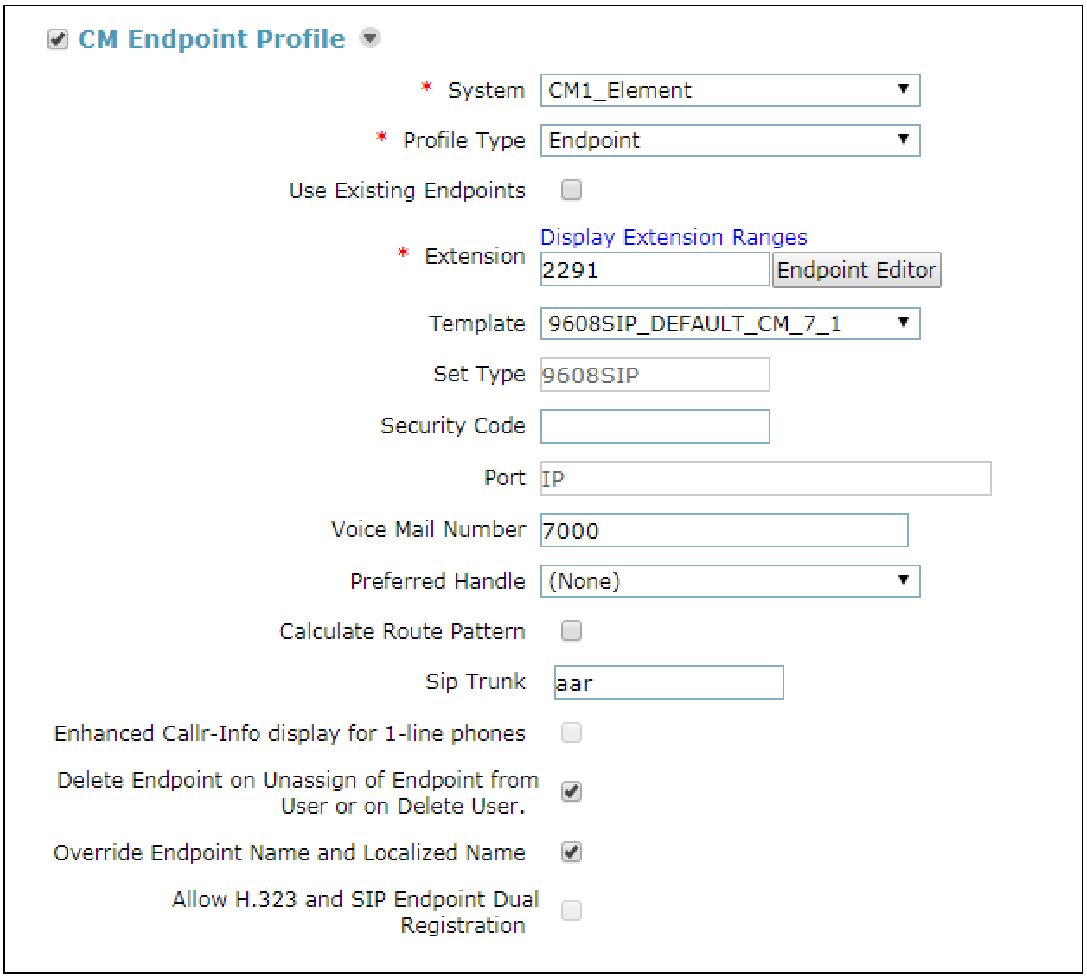
Expand the **Session Manager Profile** section.

- Make sure the **Session Manager Profile** check box is checked.
- Select the appropriate Session Manager instance from the drop-down menu in the **Primary Session Manager** field.
- Select the appropriate application sequence from the drop-down menu in the **Origination Sequence** field configured in **Section 6.10**.
- Select the appropriate application sequence from the drop-down menu in the **Termination Sequence** field configured in **Section 6.10**.
- Select the appropriate location from the drop-down menu in the **Home Location** field.



Expand the **Endpoint Profile** section.

- Select Communication Manager Element from the **System** drop-down menu.
- Select **Endpoint** from the drop-down menu for **Profile Type**.
- Enter the extension in the **Extension** field.
- Select the desired template from the **Template** drop-down menu.
- In the **Port** field **IP** is automatically inserted.
- Select the **Delete Endpoint on Unassign of Endpoint from User or on Delete User** check box.
- Select **Commit** (Not Shown) to save changes and the System Manager will add Communication Manager user configuration automatically.



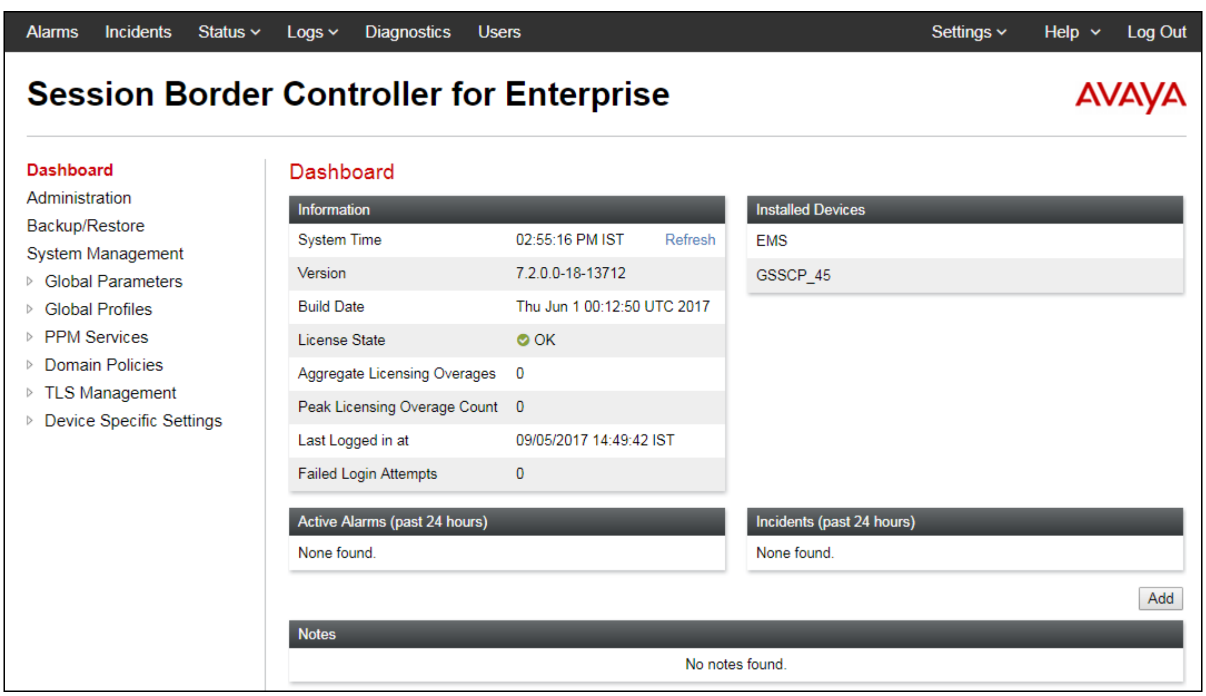
7. Configure Avaya Session Border Controller for Enterprise

This section describes the configuration of the Avaya 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 Session Border Controller using a web browser by entering the URL **https://<ip-address>**, where **<ip-address>** is the private IP address configured at installation. A log in screen is presented. Log in using the appropriate username and password.

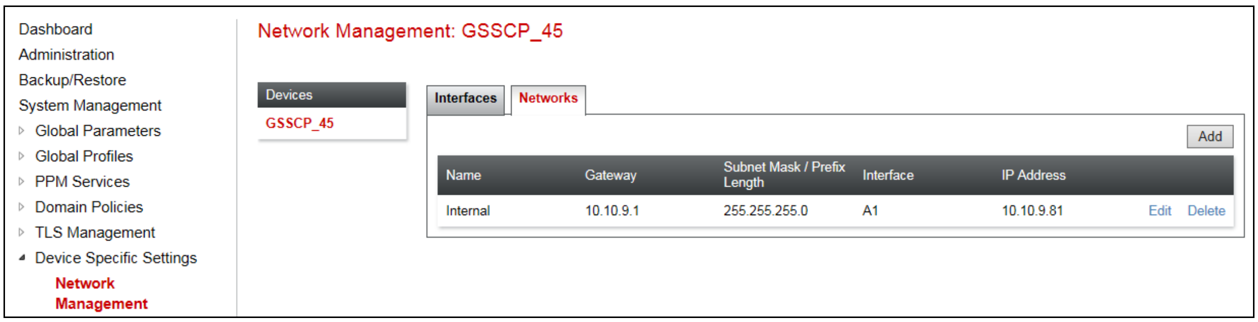
Once logged in, a dashboard is presented with a menu on the left-hand side. The menu is used as a starting point for all configuration of the Avaya SBCE.



7.2. Define Network Management

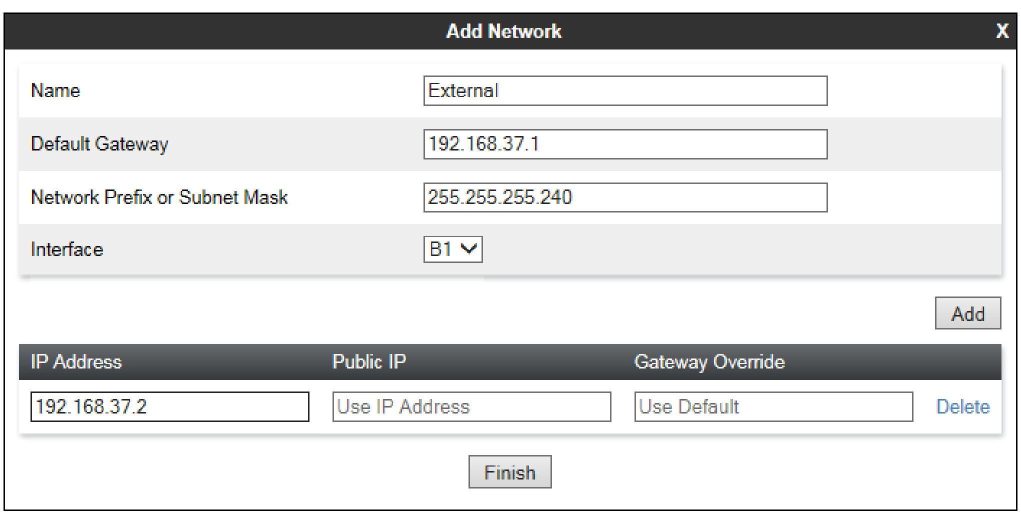
Network information is required on the Avaya SBCE to allocate IP addresses and masks to the interfaces. Note that only the **A1** and **B1** interfaces are used, typically the **A1** interface is used for the internal side and **B1** is used for external. Each side of the Avaya SBCE can have only one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

To define the network information, navigate to **Device Specific Settings → Network Management** in the main menu on the left hand side and click on **Add**.



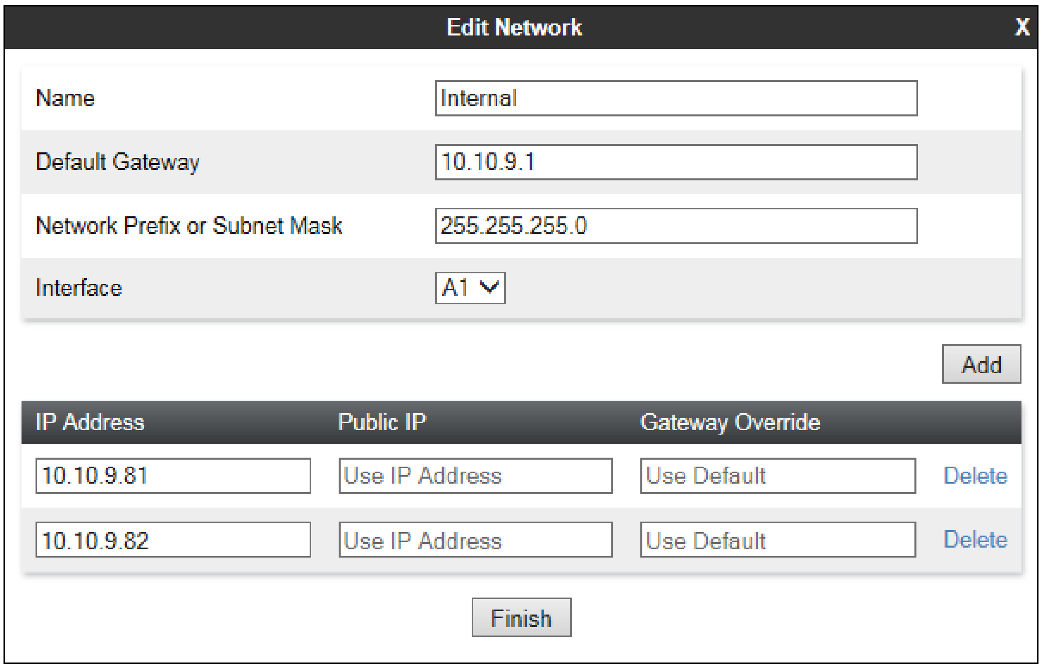
Enter details for the external interfaces in the dialogue box:

- 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 **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 on the SIP trunk in the **IP Address** field and leave the **Public IP** and **Gateway Override** fields blank.
- Click on **Finish** to complete the interface definition.

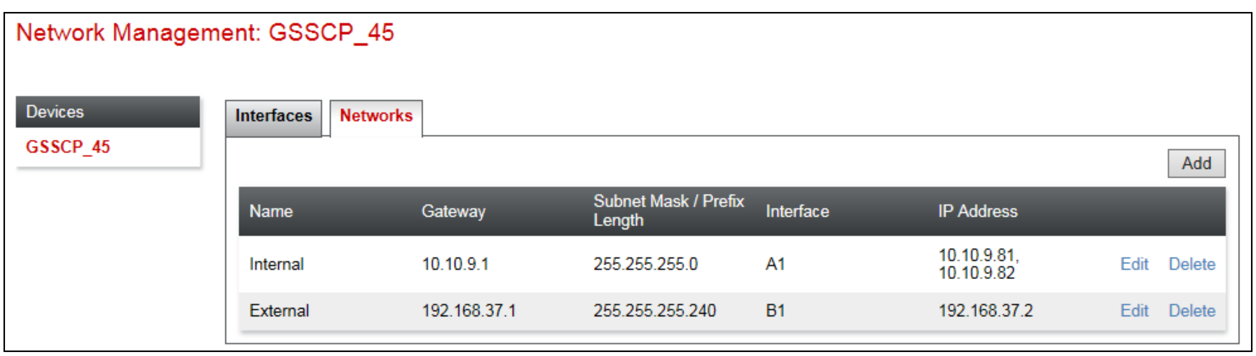


Click on **Add** to define the internal interfaces or Edit if it was defined during installation of the Avaya SBCE. Enter details in the dialogue box:

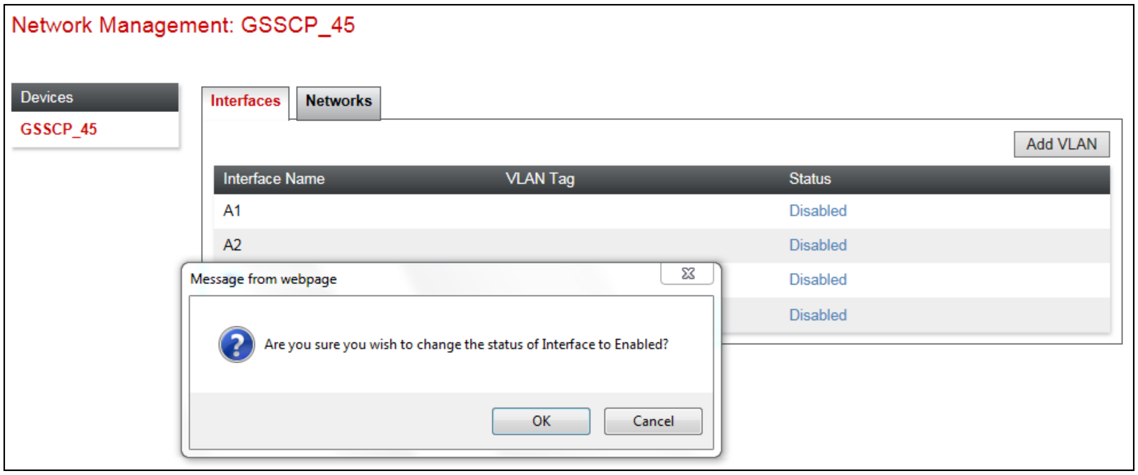
- Enter a descriptive name in the **Name** field.
- Enter the default gateway IP address for the internal interfaces in the **Default Gateway** field.
- Enter the subnet mask in the **Subnet Mask** field.
- Select the internal physical interface to be used from the **Interface** drop down menu. In the test environment, this was **A1.**
- Click on **Add** twice and additional rows will appear allowing IP addresses to be entered.
- Enter the internal IP addresses for the fixed and mobile trunks in the **IP Address** fields and leave the **Public IP** and **Gateway Override** fields blank.
- Click on **Finish** to complete the interface definition.



The following screenshot shows the completed Network Management configuration:



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



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

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

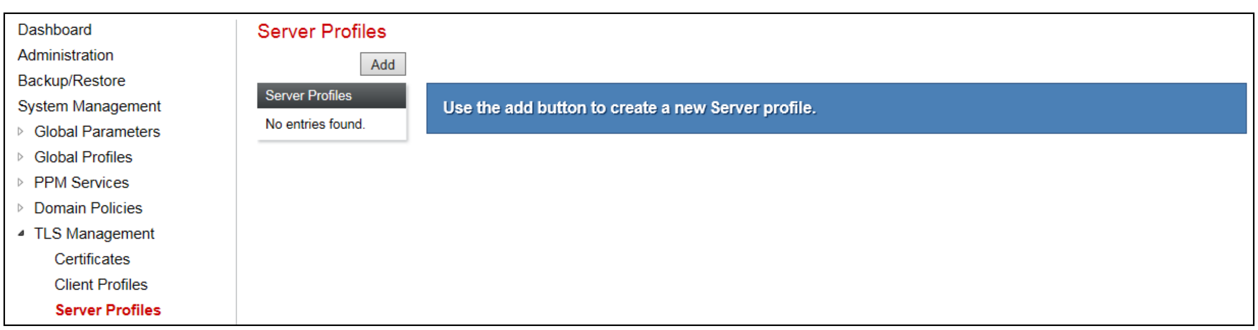
A status box will appear that will indicate when the restart is complete.

7.3. Define TLS Profiles

TLS profiles are required to support TLS on the interfaces. The implementation of certificates is beyond the scope of this document and is assumed to be already in place. The signalling interfaces require a TLS server profile and the server configuration requires a TLS client profile.

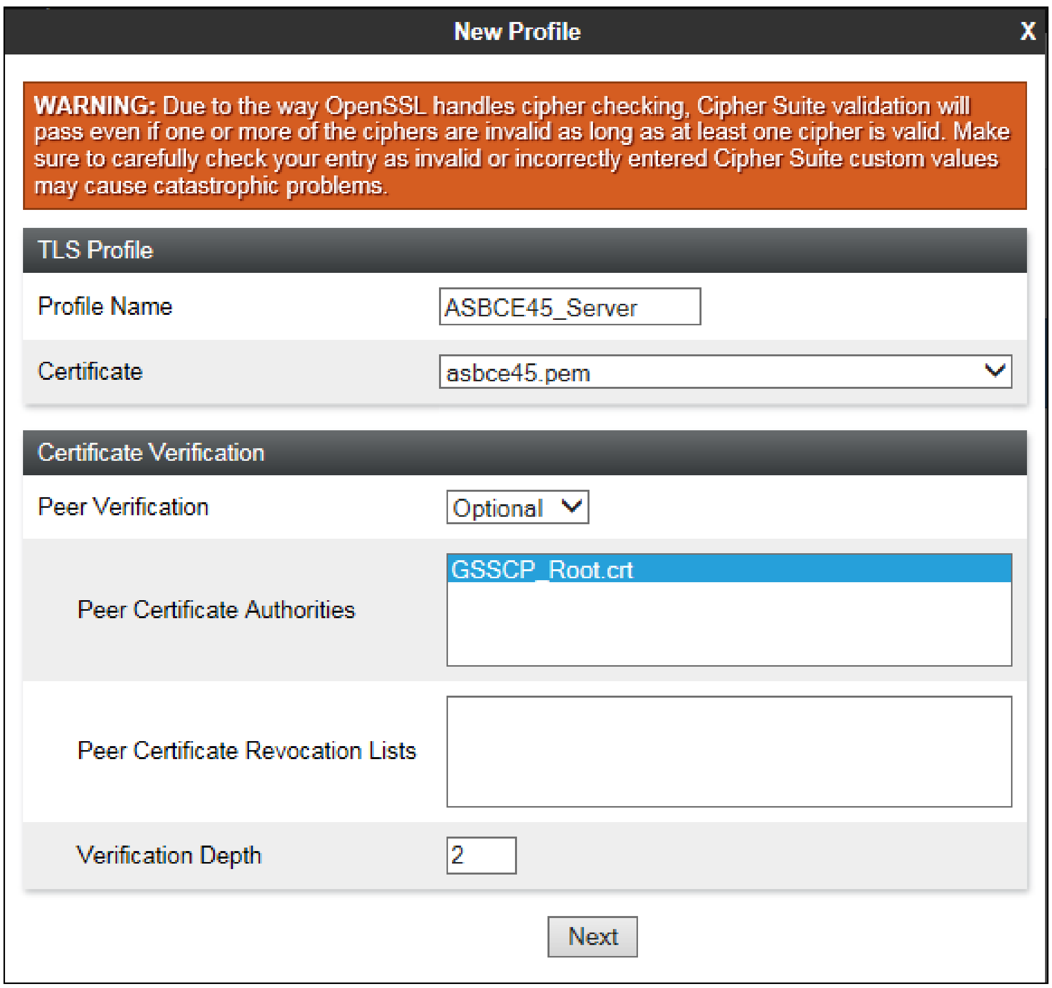
7.3.1. Server Profile

To define a TLS server profile on the Avaya SBCE, navigate to **TLS Management → Server Profiles** in the main menu on the left hand side. Click on **Add**.

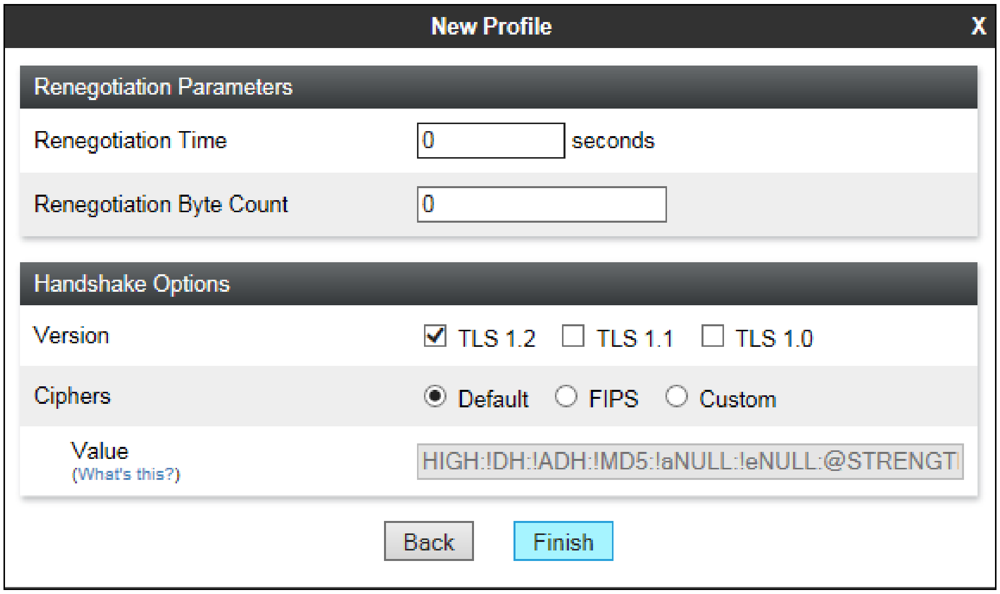


Details of the TLS server profile for the signalling interfaces are entered here.

- In the **Name** field enter a descriptive name for the server profile.
- In the **Certificate** drop down menu, select the Avaya SBCE identity certificate to be used for this profile.
- Select **Peer Verification** as required. In the test environment peer verification was made optional by selecting **Optional** in the drop down menu.
- Highlight the trusted root certificate in the **Peer Certificate Authorities** field.
- Set the **Verification Depth** as required. The example shown is for the link with Session Manager which has an identity certificate provided by a System Manager implemented as a sub-CA. This means that the Session Manager identity certificate is signed by an intermediate certificate which is in turn signed by a root certificate. This gives it a verification depth of **2**.



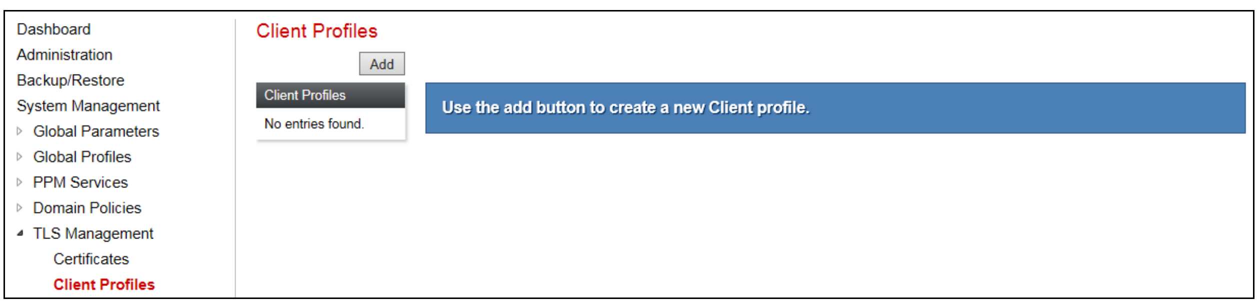
Click on **Next** to complete the server profile configuration. In the test environment, these parameters were left at default values.



Click on **Finish**.

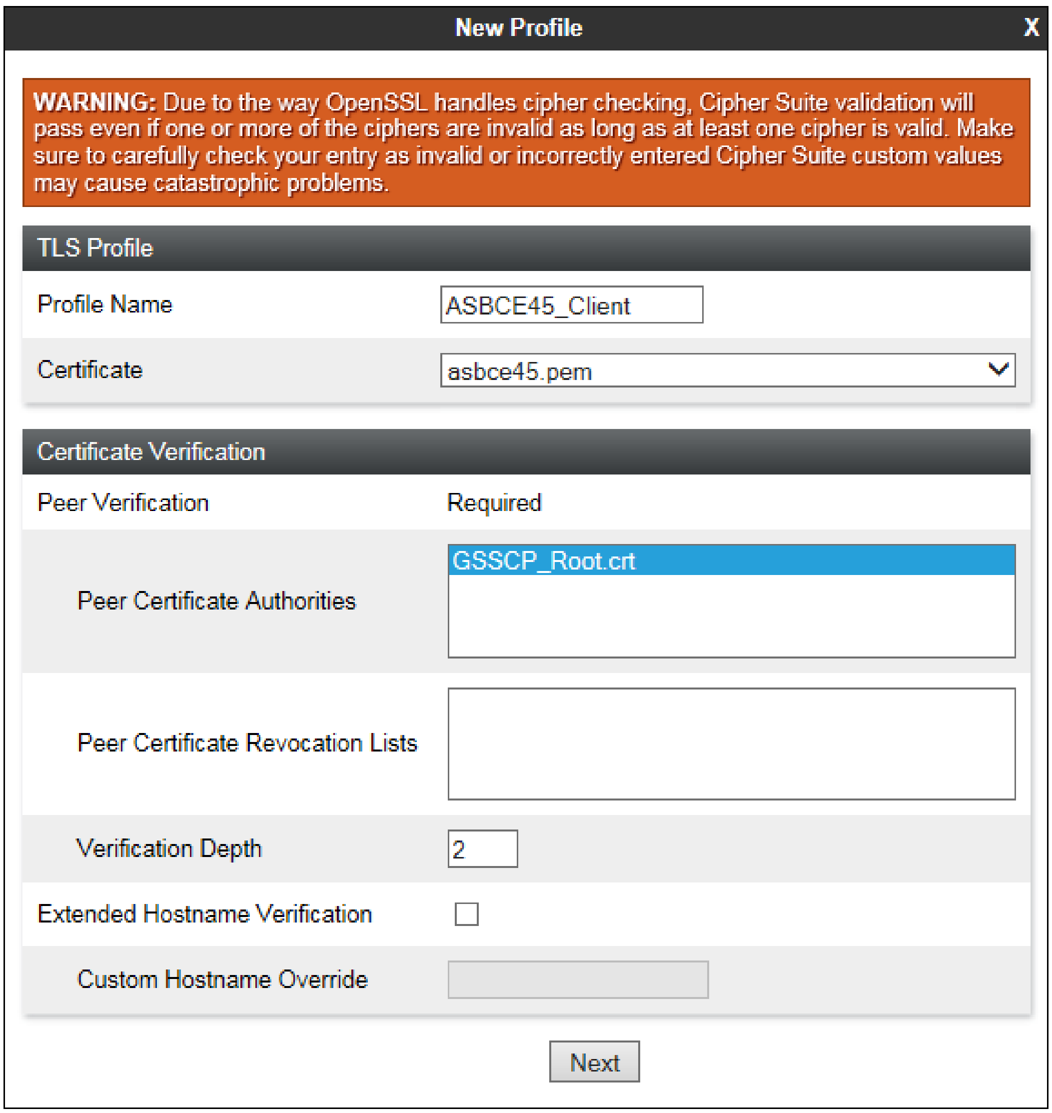
7.3.2. Client Profile

To define a TLS client profile on the Avaya SBCE, navigate to **TLS Management → Client Profiles** in the main menu on the left hand side. Click on **Add**.

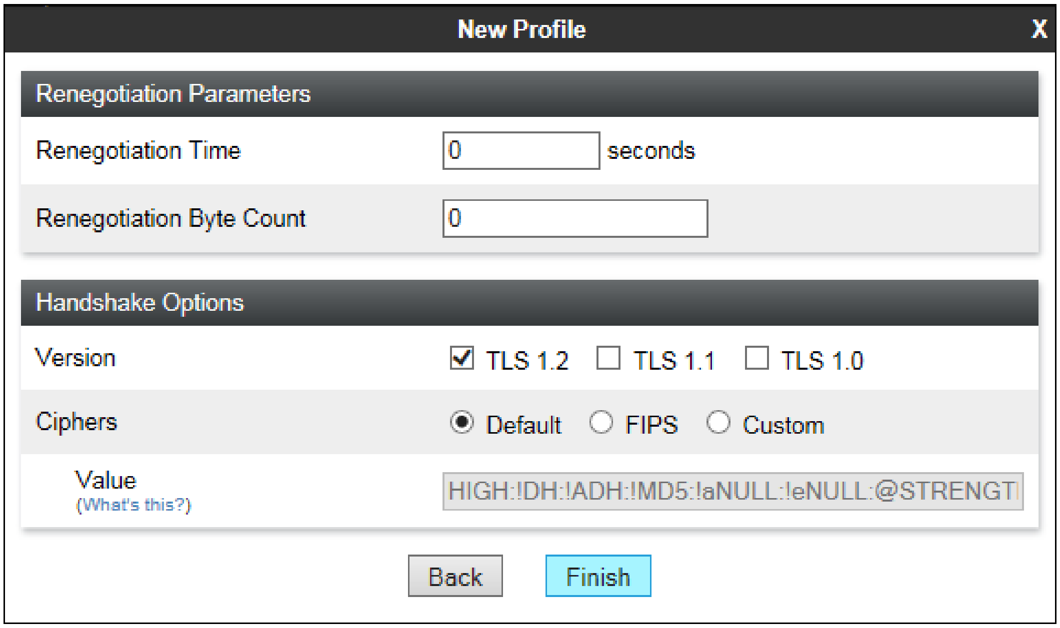


Details of the TLS client profile for the signalling interfaces are entered here.

- In the **Name** field enter a descriptive name for the server profile.
- In the **Certificate** drop down menu, select the Avaya SBCE identity certificate to be used for this profile.
- Note that **Peer Verification** is always **Required** for the client profile.
- Highlight the trusted root certificate in the **Peer Certificate Authorities** field.
- Set the **Verification Depth** as required. The example shown is for the link with Session Manager which has an identity certificate provided by a System Manager implemented as a sub-CA. This means that the Session Manager identity certificate is signed by an intermediate certificate which is in turn signed by a root certificate. This gives it a verification depth of **2**.



Click on **Next** to complete the client profile configuration. In the test environment, these parameters were left at default values.



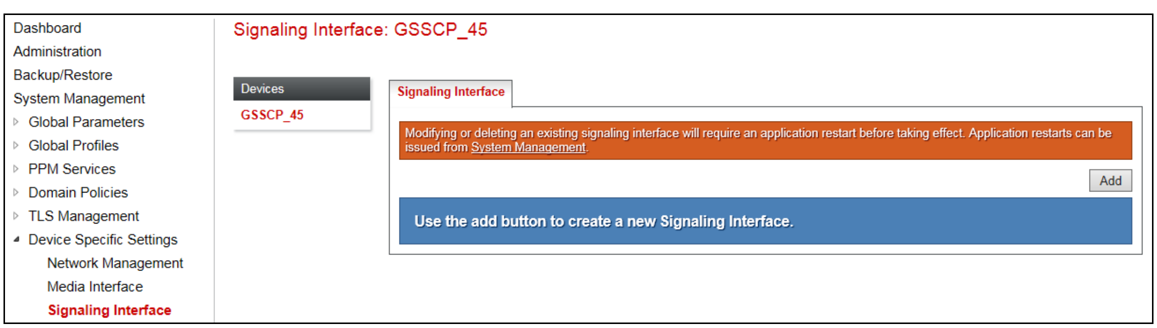
Click on **Finish**.

7.4. Define Interfaces

When the IP addresses and masks are assigned to the interfaces, these are then configured as signalling and media interfaces. Testing was carried out with TLS used for transport of signalling between Session Manager and the Avaya SBCE, and UDP for transport of signalling between the Avaya SBCE and the Vodafone Libertel SIP Trunking Service. Two signalling and two media interfaces were required on both the internal and external sides of the Avaya SBCE to handle on-net and off-net traffic.

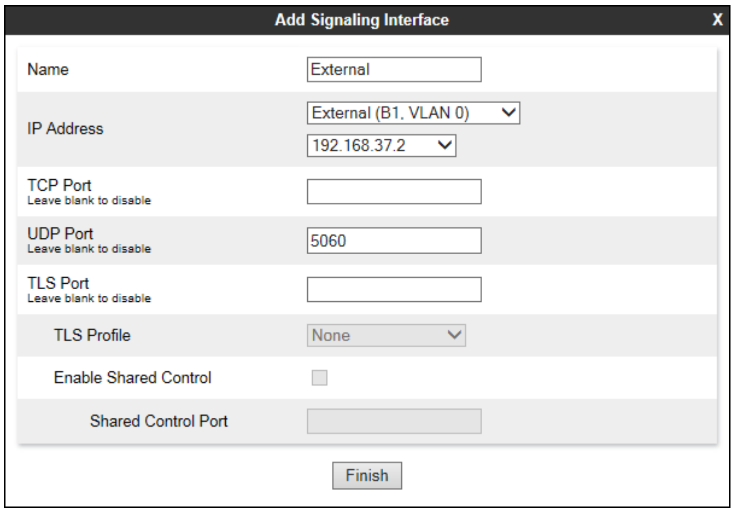
7.4.1. Signalling Interfaces

To define the signalling interfaces on the Avaya SBCE, navigate to **Device Specific Settings → Signaling Interface** (not shown) in the main menu on the left hand side. Click on **Add**.



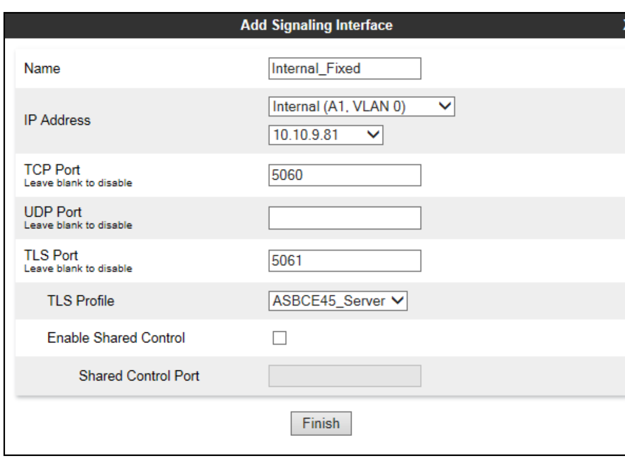
Details of transport protocol and ports for the external and internal SIP signalling are entered here.

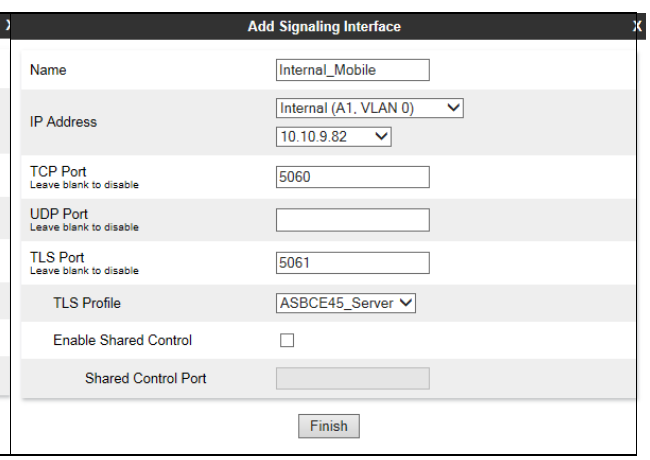
- In the **Name** field enter a descriptive name for the external signalling interface.
- In the **IP Address** drop down menus, select the external network interface and IP address. Note that when the external network interface is selected, the bottom drop down menu is populated with the available IP addresses as defined in **Section 7.2**. In the test environment, this was IP address **192.168.37.2**.
- Enter the UDP port number in the **UDP Port** field, **5060** is used for the Vodafone Libertel SIP Trunking Service.



The internal signalling interfaces are defined in the same way. One is for on-net and the other for off-net mobile signalling between the Avaya SBCE and Session Manager.

- Select **Add** and enter details of the internal signalling interface in the pop-up menu.
- In the **Name** field enter a descriptive name for the internal signalling interface.
- In the **IP Address** drop down menus, select the internal network interface and IP address.
- Select **TLS** port number, **5061** is used for Session Manager.
- Select the **TLS Profile** defined in **Section 7.3** from the drop down menu.

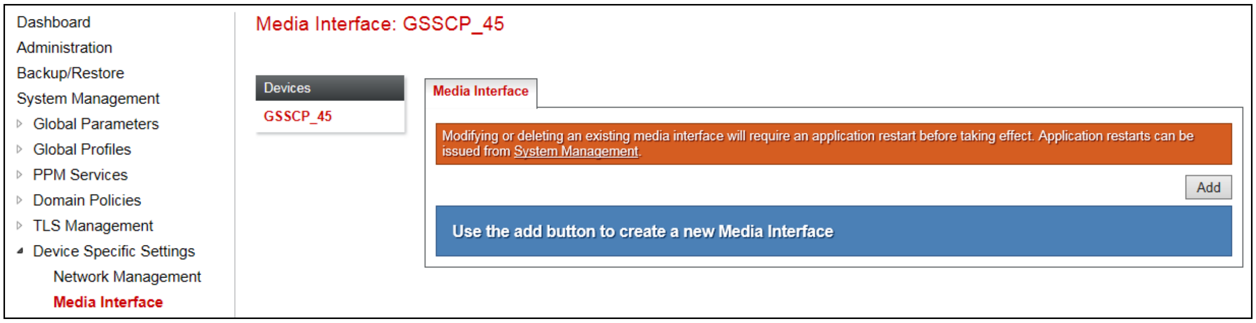




Note: In the test environment, the internal IP addresses were **10.10.9.81** for off-net PSTN signalling and **10.10.9.82** for on-net mobile signalling.

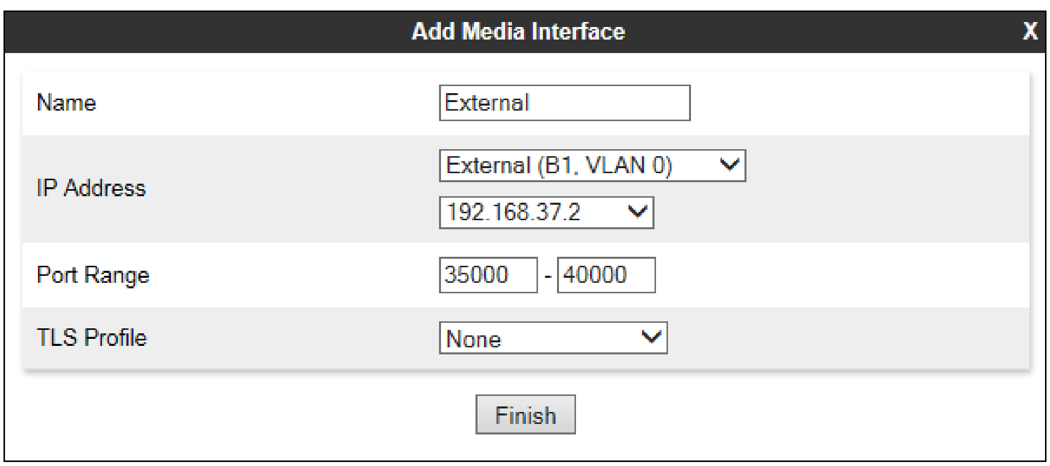
7.4.2. Media Interfaces

To define the media interfaces on the Avaya SBCE, navigate to **Device Specific Settings → Media Interface** in the main menu on the left hand side. Click on **Add**.



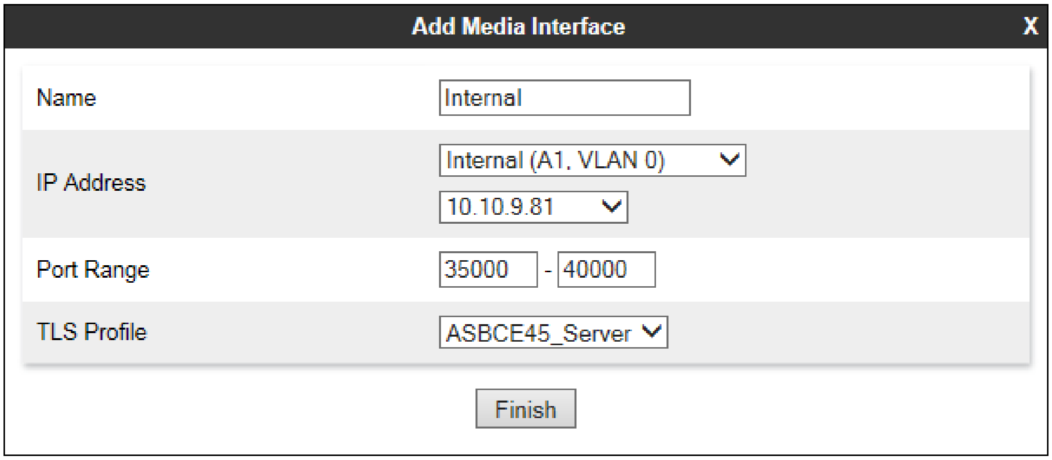
Details of the RTP port ranges for the internal and external media streams are entered here. The IP addresses for media can be the same as those used for signalling.

- In the **Name** field enter a descriptive name for the external media interface.
- In the **IP Address** drop down menus, select the external network interface and IP address. Note that when the external network interface is selected, the bottom drop down menu is populated with the available IP addresses as defined in **Section 7.2**. In the test environment, this was IP address **192.168.37.2**.
- Define the RTP **Port Range** for the media path with the Vodafone Libertel SIP Trunking Service, during testing this was left at the default values.



The internal media interface is defined in the same way:

- Select **Add** and enter details of the internal media interface in the pop-up menu.
- In the **Name** field enter a descriptive name for the internal media interface.
- In the **IP Address** drop down menus, select the internal network interface and IP address.
- Define the RTP **Port Range** for the media path with the enterprise endpoints, during testing this was left at the default values.
- Select the TLS server profile defined in **Section 7.3**.

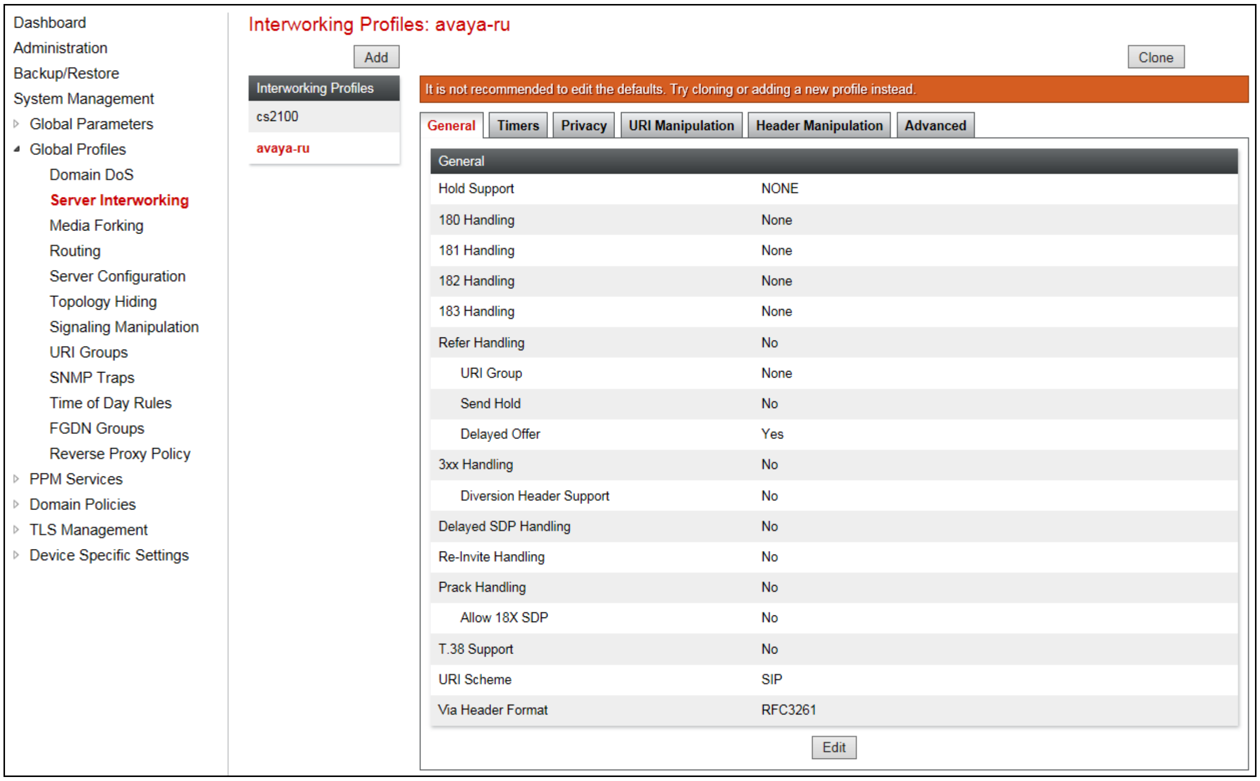


Note: In the test environment, only one internal media interface was defined as separate interfaces for off-net and on-net mobile were not required.

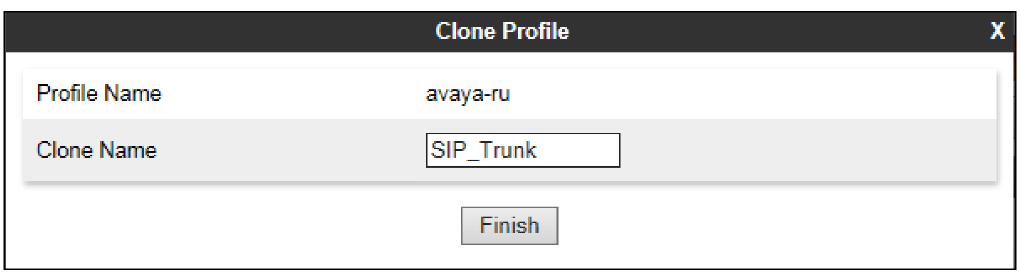
7.5. Define Server Interworking

Server interworking is defined for each server connected to the Avaya SBCE. In this case, Vodafone Libertel SIP Trunking is connected as the Trunk Server and Session Manager is connected as the Call Server. Configuration of interworking includes Hold support, T.38 fax support and SIP extensions. Note that only one interworking profile is required for Vodafone Libertel even though there are separate signalling links for off-net PSTN and on-net mobile

To define server interworking on the Avaya SBCE, navigate to **Global Profiles → Server Interworking** in the main menu on the left hand side. To define Server Interworking for Vodafone Libertel SIP Trunking, highlight the **avaya-ru** profile and click on **Clone**.

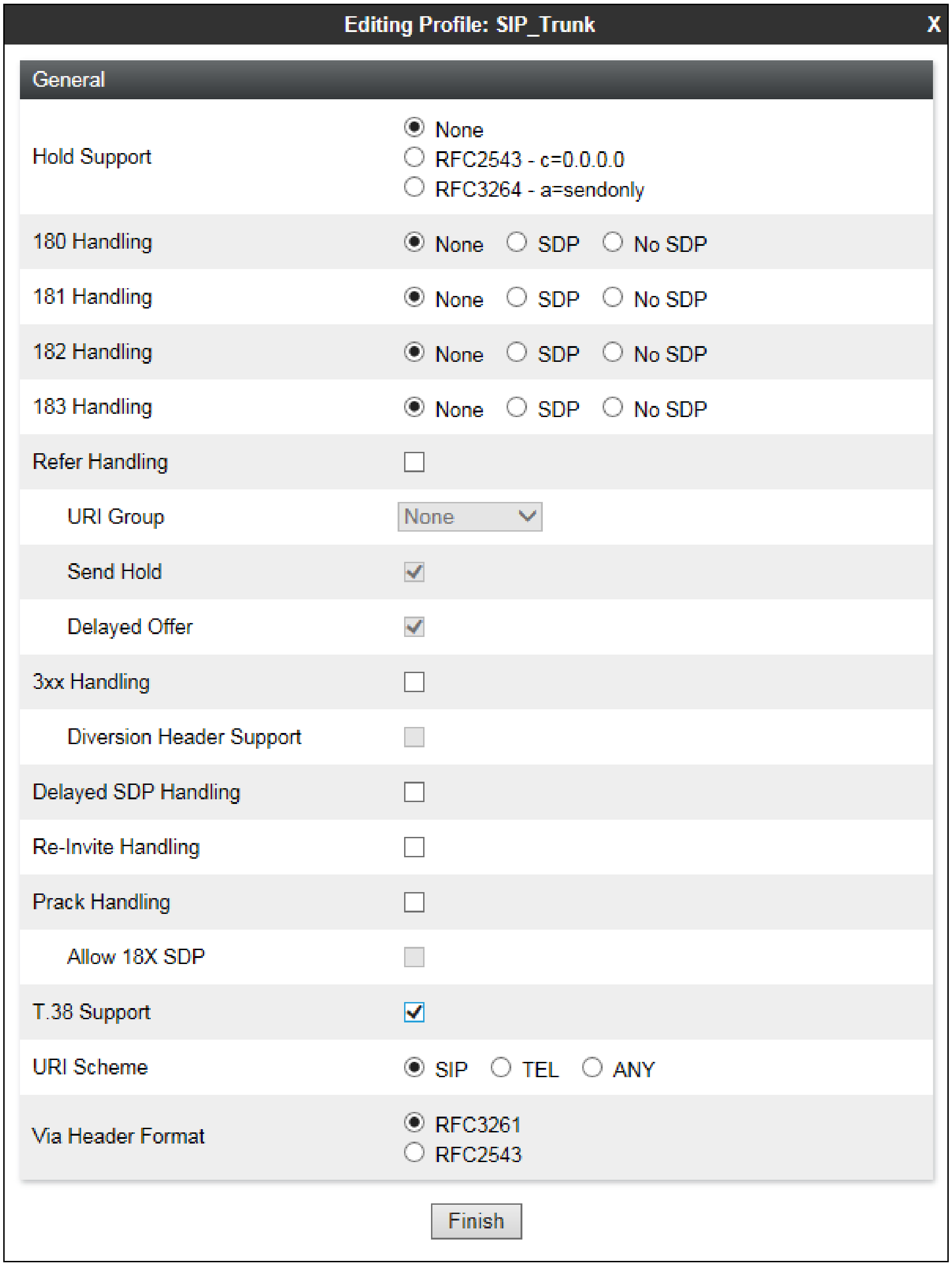


A pop-up menu is generated. In the **Name** field enter a descriptive name for the Vodafone Libertel SIP Trunking network and click **Finish**.



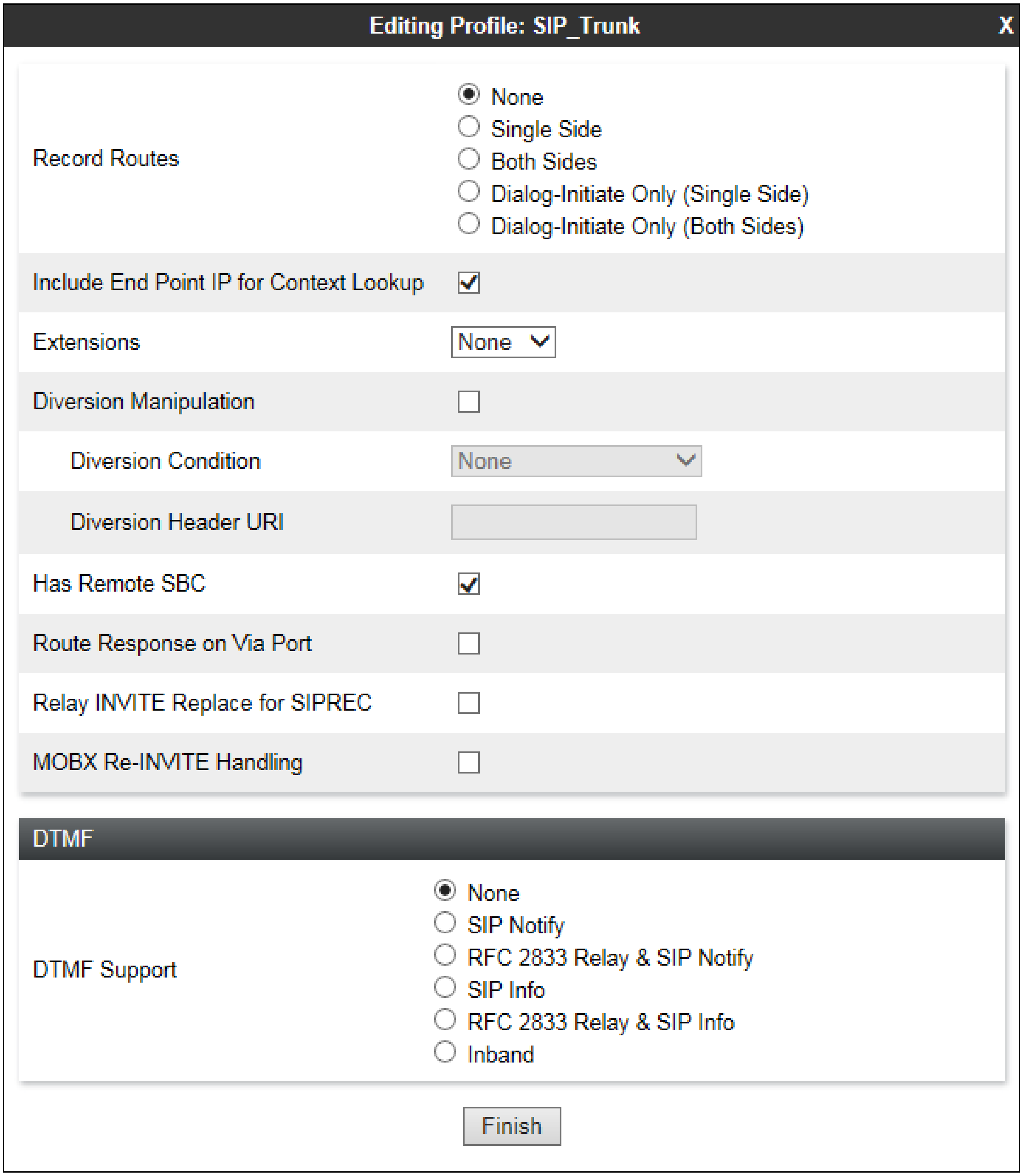
Select the General tab of the resulting Interworking Profile and click on Edit (not shown).

Select the General tab of the resulting Interworking Profile and click on Edit (not shown). The screenshot shows the cloned profile. Check the **T.38 Support** box and leave the rest of the parameters at their original settings. Click on **Finish**.

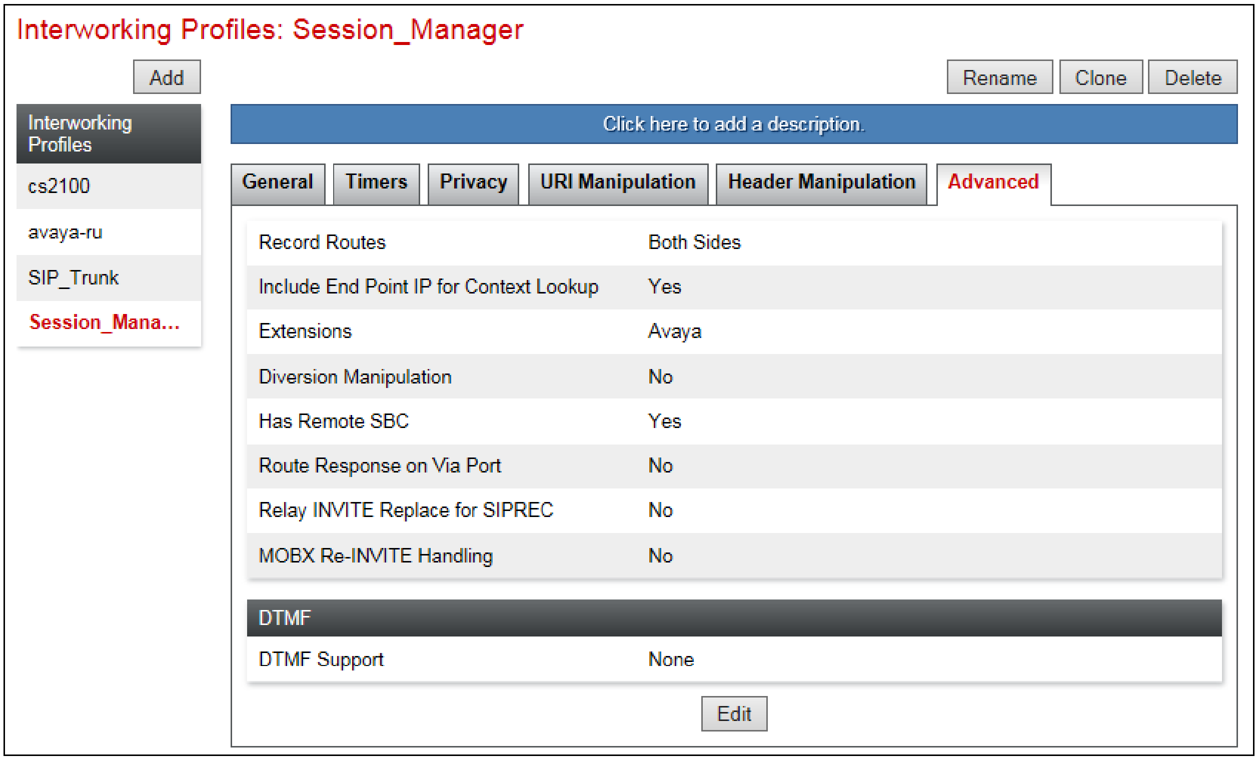


Select the **Advanced** tab (not shown) and click on **Edit**.

Set **Record Routes** to **None** as this header is not used by the network and select **None** in the **Extensions** drop down menu. Ensure that the **Has Remote SBC** box is checked. Click on **Finish**.



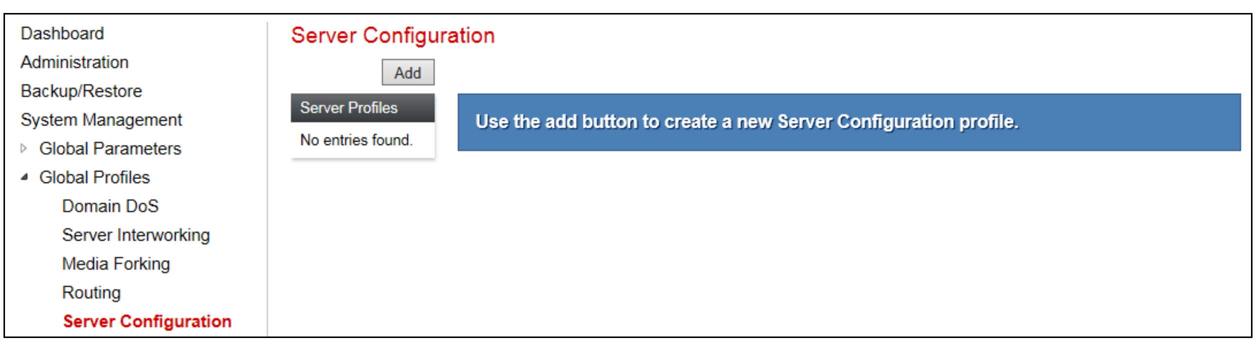
Repeat the process to define Server Interworking for Session Manager. In the **Advanced** tab, leave the settings at the original values cloned from the avaya-ru profile. **Record Routes** is set to **Both Sides** as Session Manager uses the Record-Route header and **Avaya** is selected in the **Extensions** drop down menu:



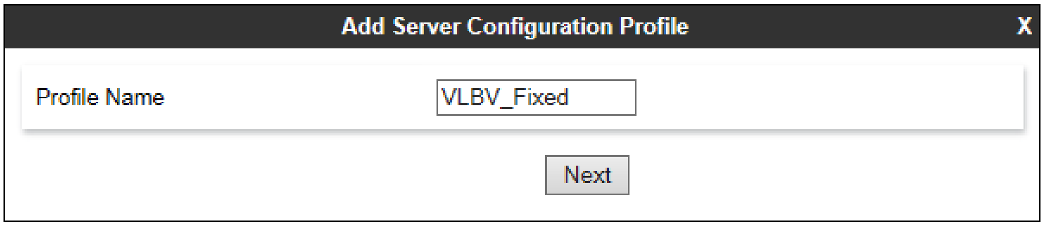
7.6. Define Servers

A server definition is required for each server connected to the Avaya SBCE. The Vodafone Libertel SIP Trunking Service has two separate interfaces for off-net PSTN traffic and on-net mobile traffic. Each of these is connected as a separate Trunk Server. Session Manager has a single signalling interface and is connected as a Call Server.

To define the Vodafone Libertel SIP Trunk Server for off-net PSTN traffic, navigate to **Global Profiles → Server Configuration** in the main menu on the left hand side. Click on **Add**.

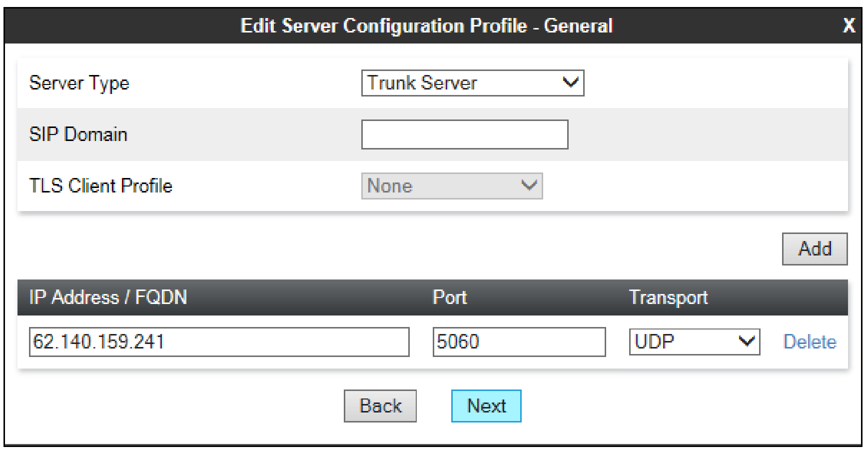


Enter an appropriate name in the pop-up menu.

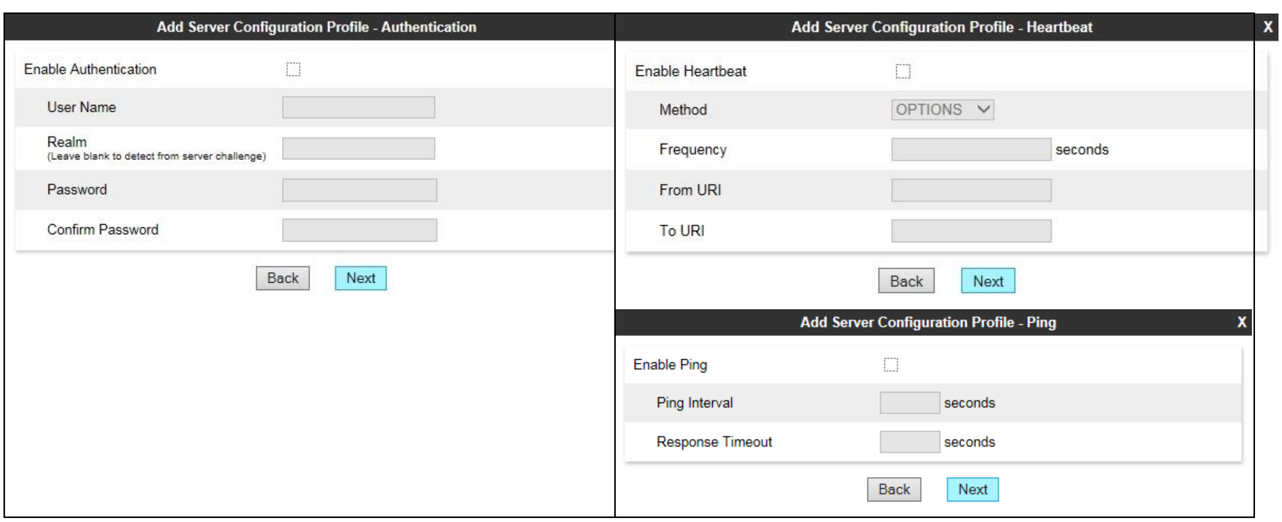


Click on **Next** and enter details in the dialogue box.

- In the **Server Type** drop down menu, select **Trunk Server**.
- Click on **Add** to enter an IP address
- In the **IP Addresses / FQDN** box, type the Vodafone Libertel IP address for the off-net PSTN trunk.
- In the **Port** box, enter the port to be used for the SIP Trunk.
- In the **Transport** drop down menu, select **UDP**.

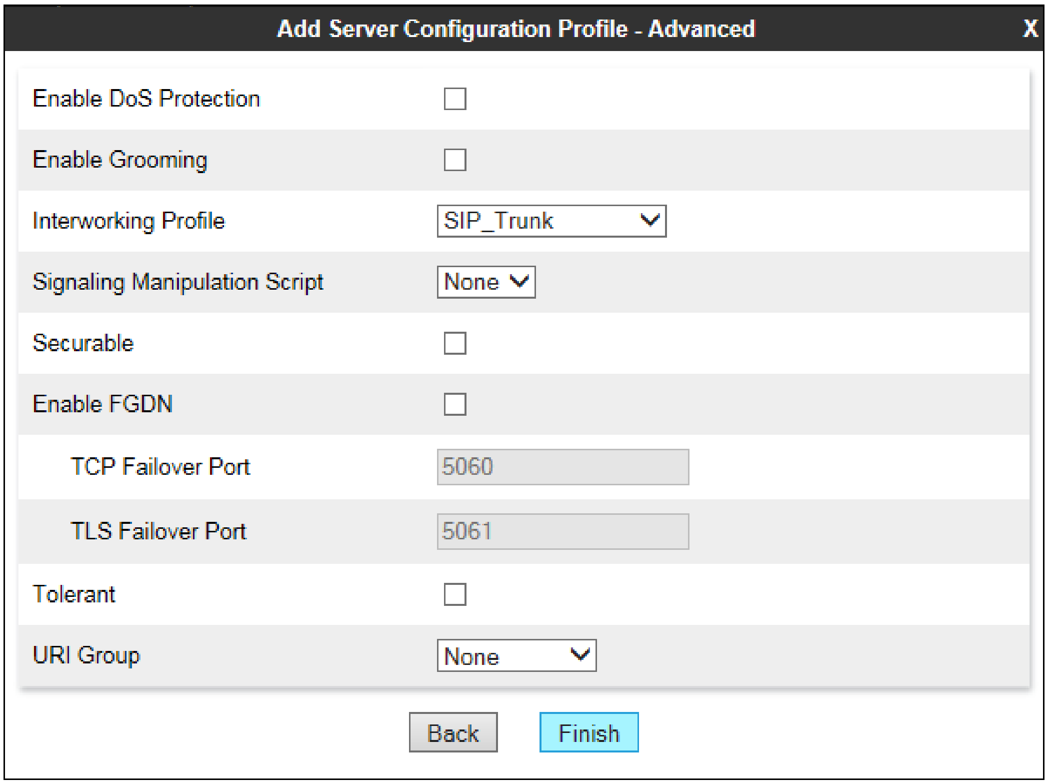


Click on **Next** three times for the Authentication, Heartbeat and Ping dialogue boxes:

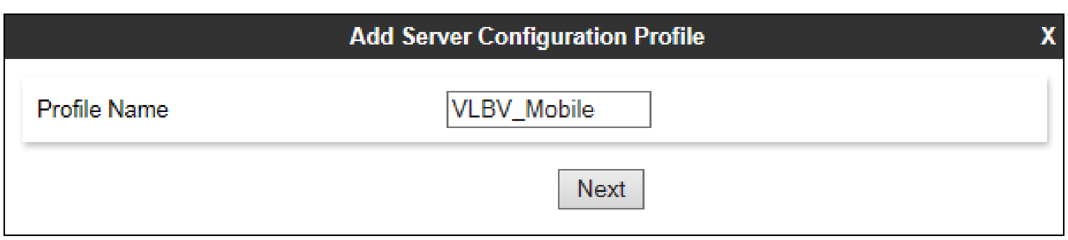


In the test environment, the Authentication, Heartbeat and Ping dialogue boxes were left at default values. Click on **Next** again to get to the final dialogue box. This contains the **Advanced** settings:

- In the **Interworking Profile** drop down menu, select the **Interworking Profile** for the Vodafone Libertel SIP Trunking Service defined in **Section 7.5**.
- Leave the other fields at default settings.
- Click **Finish**.

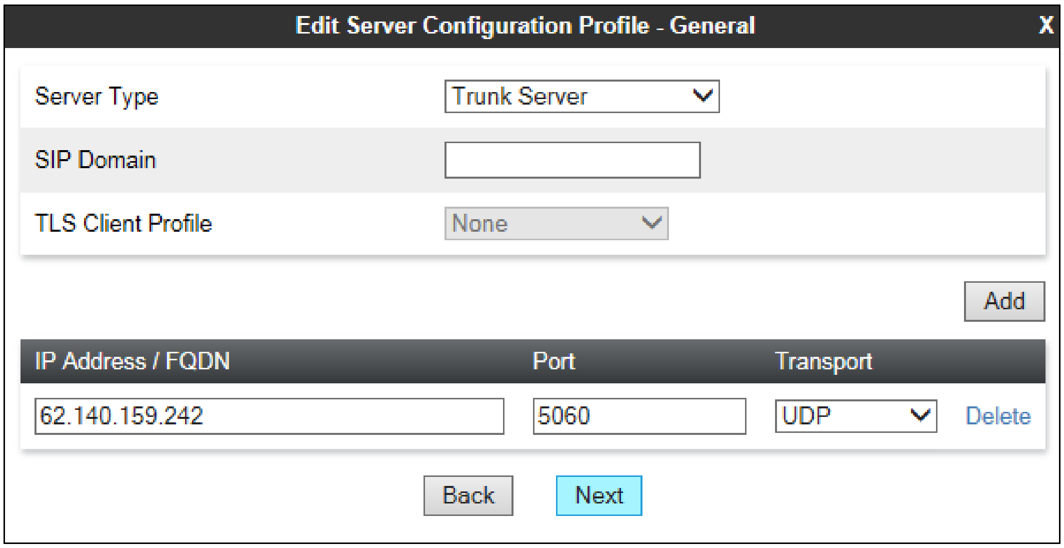


To define the Vodafone Libertel SIP Trunk Server for on-net mobile traffic, return to **Global Profiles → Server Configuration** in the main menu on the left hand side (not shown). Click on **Add**. Enter an appropriate name in the pop-up menu.

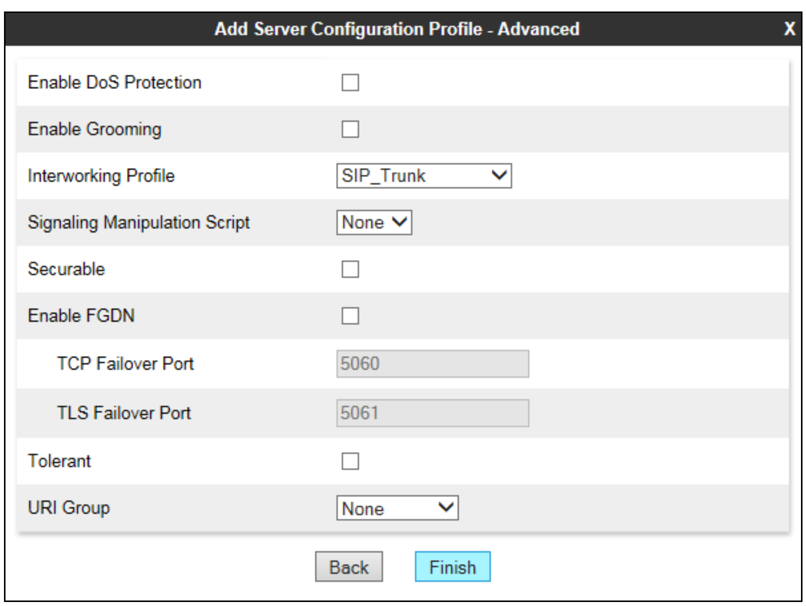


Click on **Next** and enter details in the dialogue box.

- In the **Server Type** drop down menu, select **Trunk Server**.
- Click on **Add** to enter an IP address
- In the **IP Addresses / FQDN** box, type the Vodafone Libertel IP address for the on-net mobile trunk.
- In the **Port** box, enter the port to be used for the SIP Trunk.
- In the **Transport** drop down menu, select **UDP**.



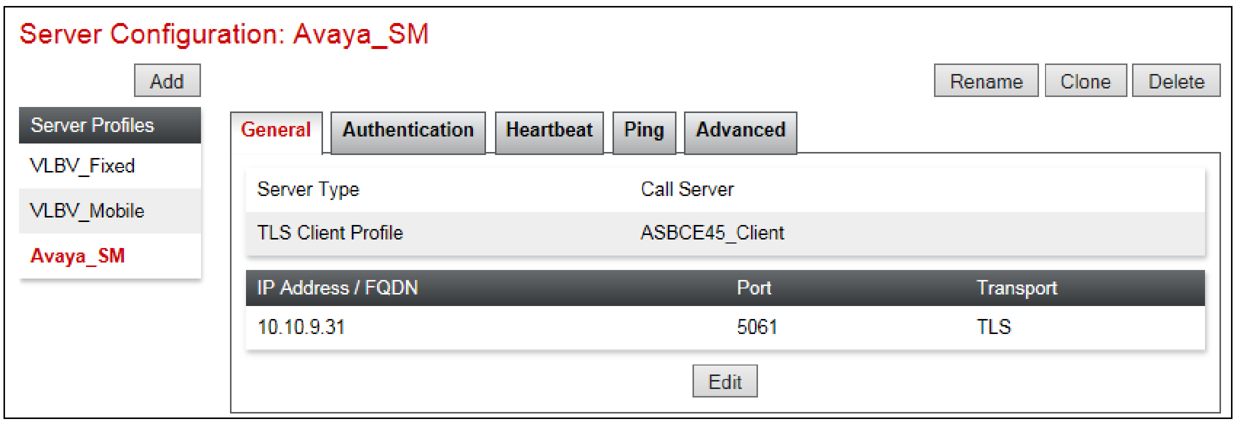
Click on **Next** three times for the Authentication, Heartbeat and Ping dialogue boxes (not shown) and click on **Next** again to get to the final dialogue box. This contains the **Advanced** settings which are the same as those for the off-net PSTN trunk:



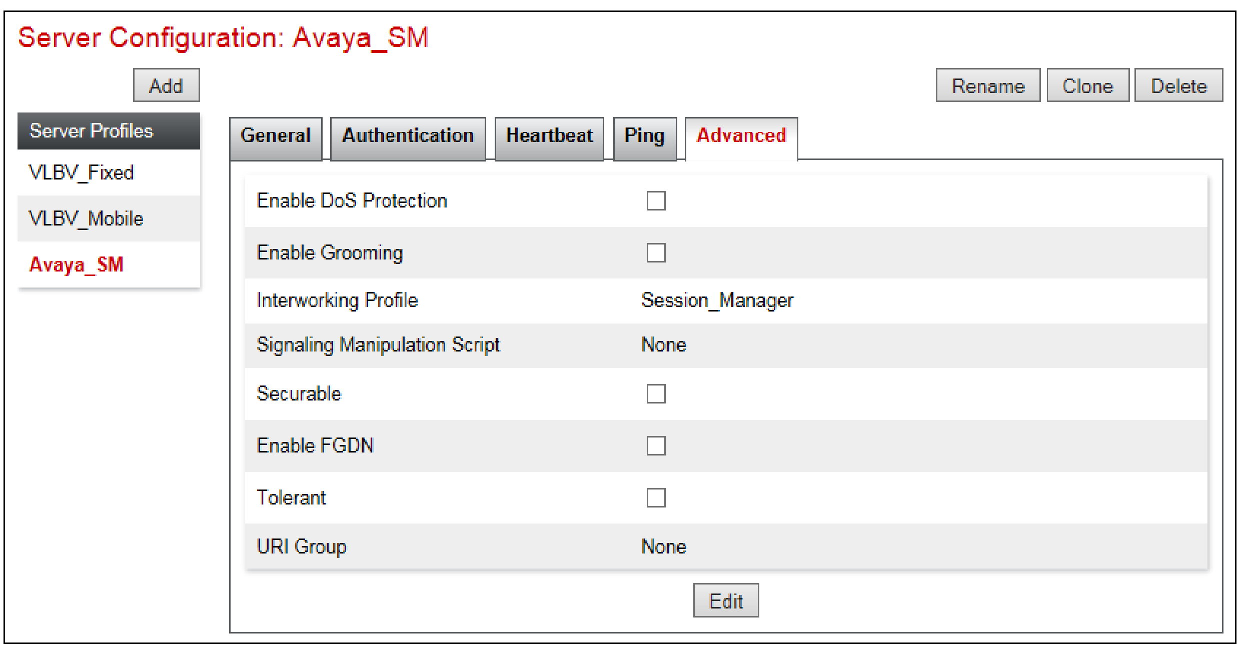
Use the process above to define the Call Server configuration for Session Manager if not already defined.

- Ensure that **Call Server** is selected in the **Server Type** drop down menu in the **General** dialogue box.
- If TLS is used between the Avaya SBCE and Session manager, ensure that the TLS client profile created in **Section 7.3** is selected in the **TLS Client Profile** drop down menu in the **General** dialogue box.
- Ensure that the Interworking Profile defined for Session Manager in **Section 7.5** is selected in the **Interworking Profile** drop down menu in the Advanced dialogue box

The following screenshot shows the **General** tab of the completed Server Configuration:

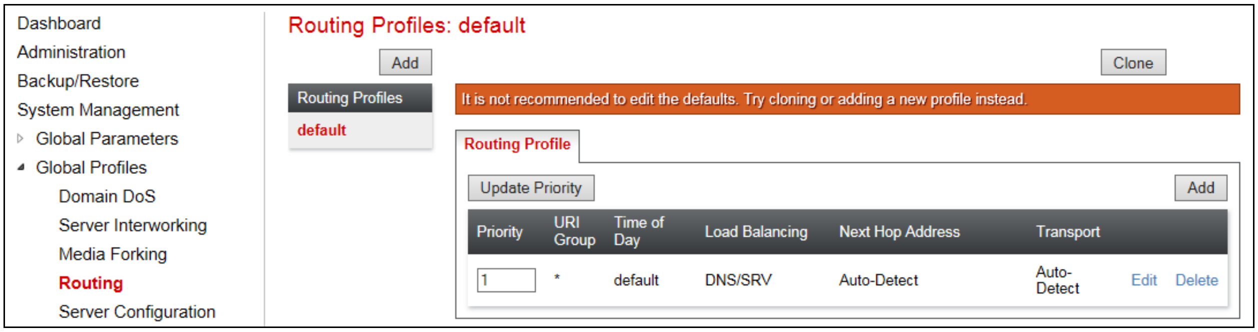


Note: The IP Address matches the SIP Entity for Session Manager described in **Section 6.5** and the Port and Transport matches the Entity Link described in **Section 6.6**. The next screenshot shows the **Advanced** tab.

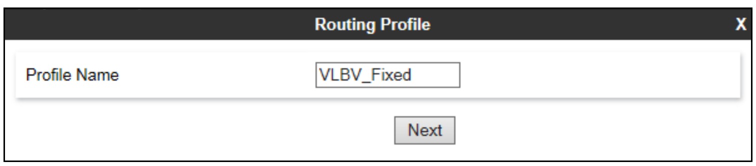


7.7. Define Routing

Routing information is required for routing to the Vodafone Libertel off-net PSTN and on-net mobile services on the external side and Session Manager on the internal side. The IP addresses and ports defined here will be used as the destination addresses for signalling. To define routing to the Vodafone Libertel off-net PSTN service, navigate to **Global Profiles → Routing** in the main menu on the left hand side. Click on **Add.**

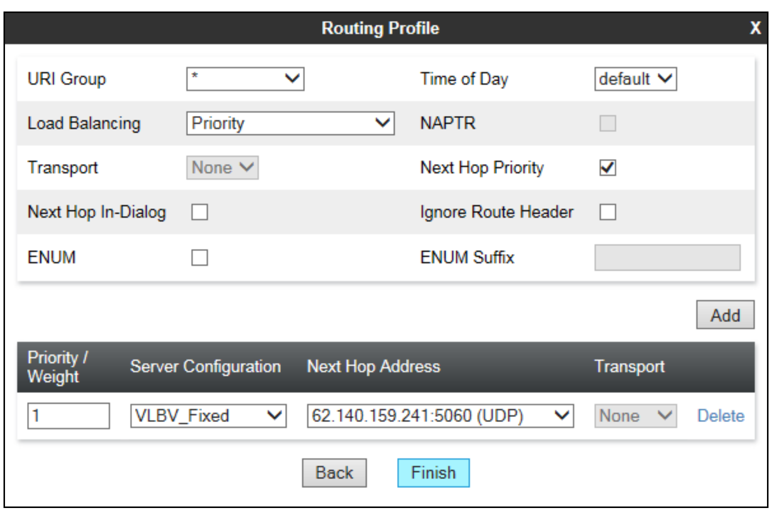


Enter an appropriate name in the dialogue box.

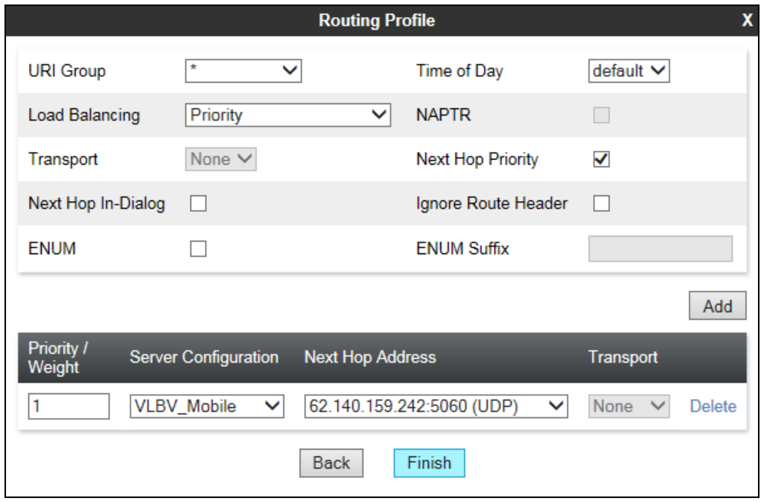


Click on **Next** and enter details for the Routing Profile for the off-net PSTN trunk:

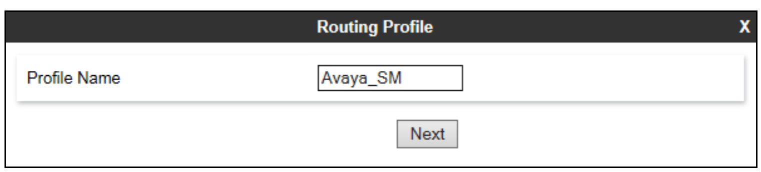
- During testing, **Load Balancing** was left at the default value of **Priority**.
- Click on **Add** to specify an IP address for the off-net PSTN trunk.
- Assign a priority in the **Priority / Weight** field, during testing **1** was used.
- Select the Server Configuration defined in **Section 7.5** in the **Server Configuration** drop down menu. This automatically populates the **Next Hop Address** field



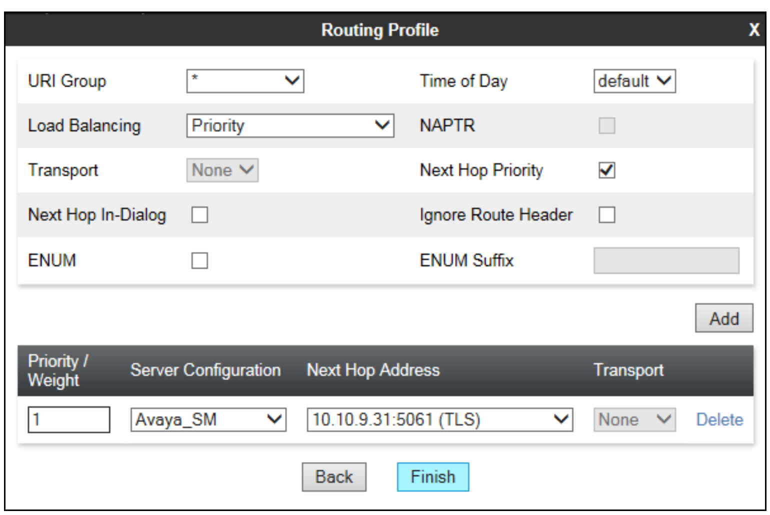
Click **Finish** and repeat the above process for the Routing Profile for the on-net mobile trunk. return to **Global Profiles → Routing** in the main menu on the left hand side. Click on **Add** (not shown). Enter an appropriate name in the dialogue box (not shown), in the test environment, **VLBV_Mobile** was used. Click on **Next** and enter details for the Routing Profile for the off-net PSTN trunk:



Repeat the process for the Routing Profile for Session Manager: return to **Global Profiles → Routing** in the main menu on the left hand side. Click on **Add** (not shown). Enter an appropriate name in the dialogue box:



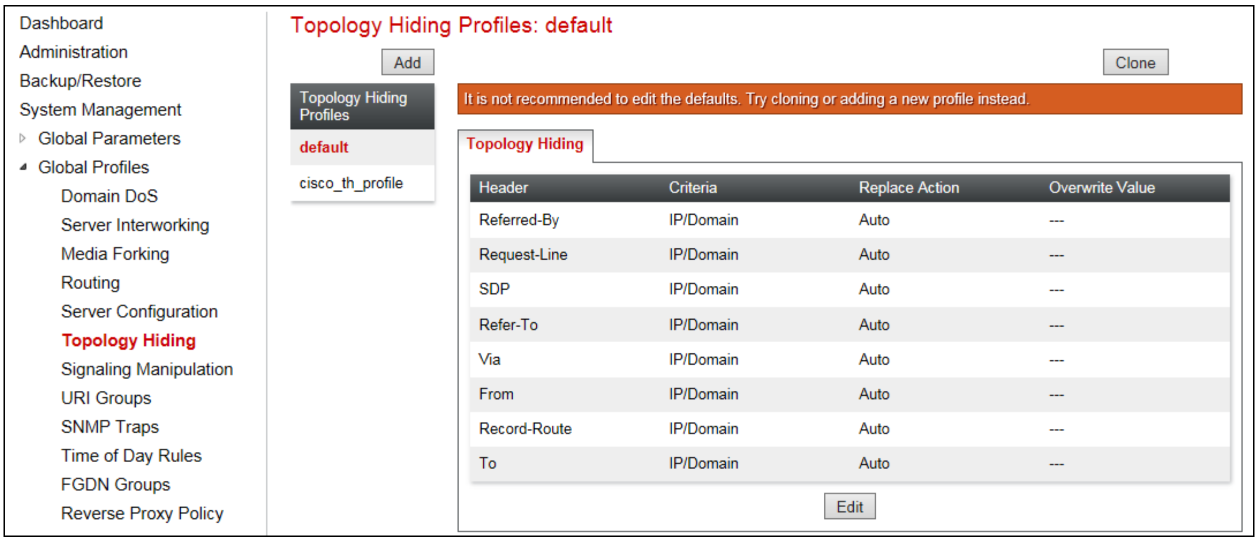
Click on **Next** and enter details for the Routing Profile for the off-net PSTN trunk:



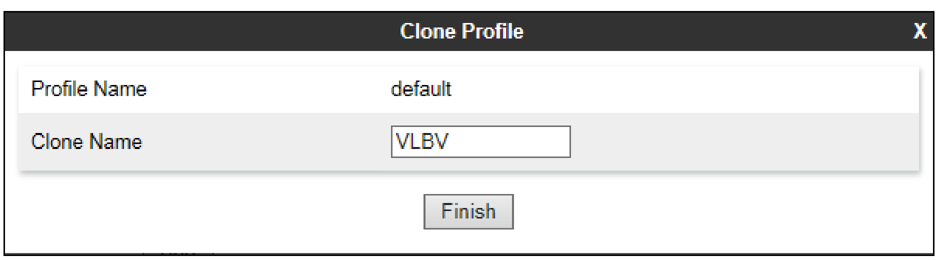
7.8. Topology Hiding

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 for termination information and the external interfaces for origination information.

To define Topology Hiding for Vodafone Libertel SIP Trunking, navigate to **Global Profiles → Topology Hiding** in the main menu on the left hand side. Select the default profile and click on **Clone**.



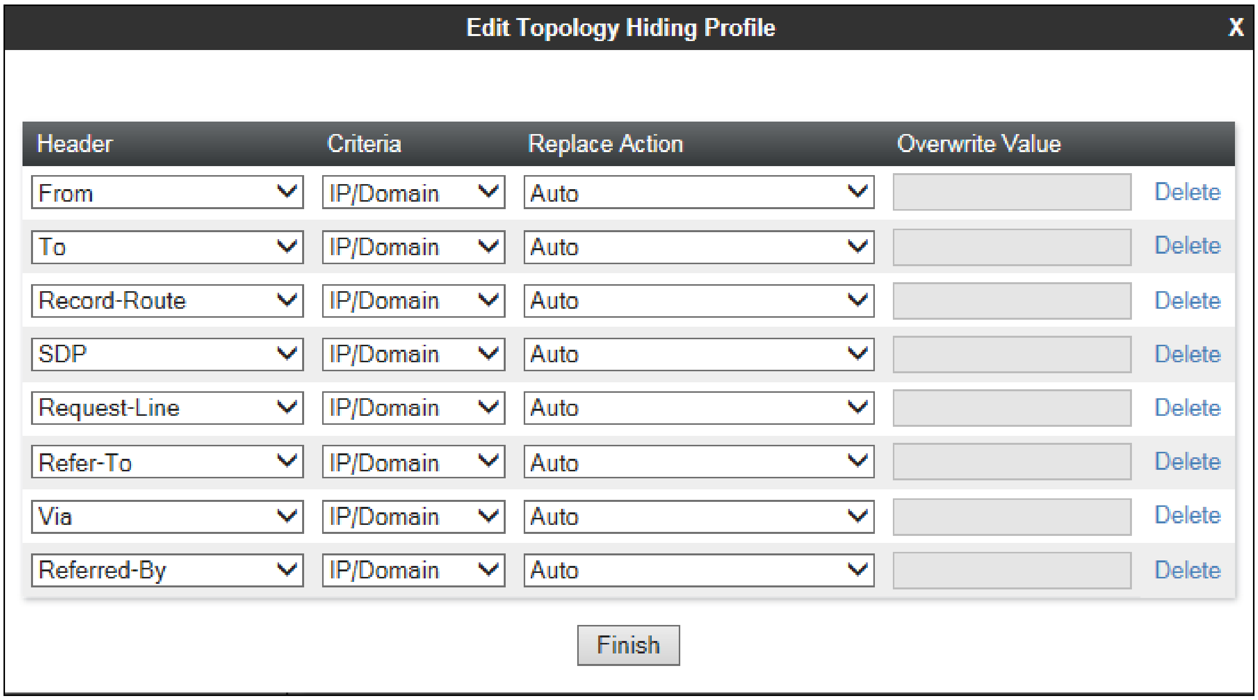
Assign an appropriate name in the dialogue box and click on **Finish**:



Highlight the new Topology Hiding profile (not shown) and click on **Edit**. Make changes if required.

During testing, fields were left at default values. If changes are required:

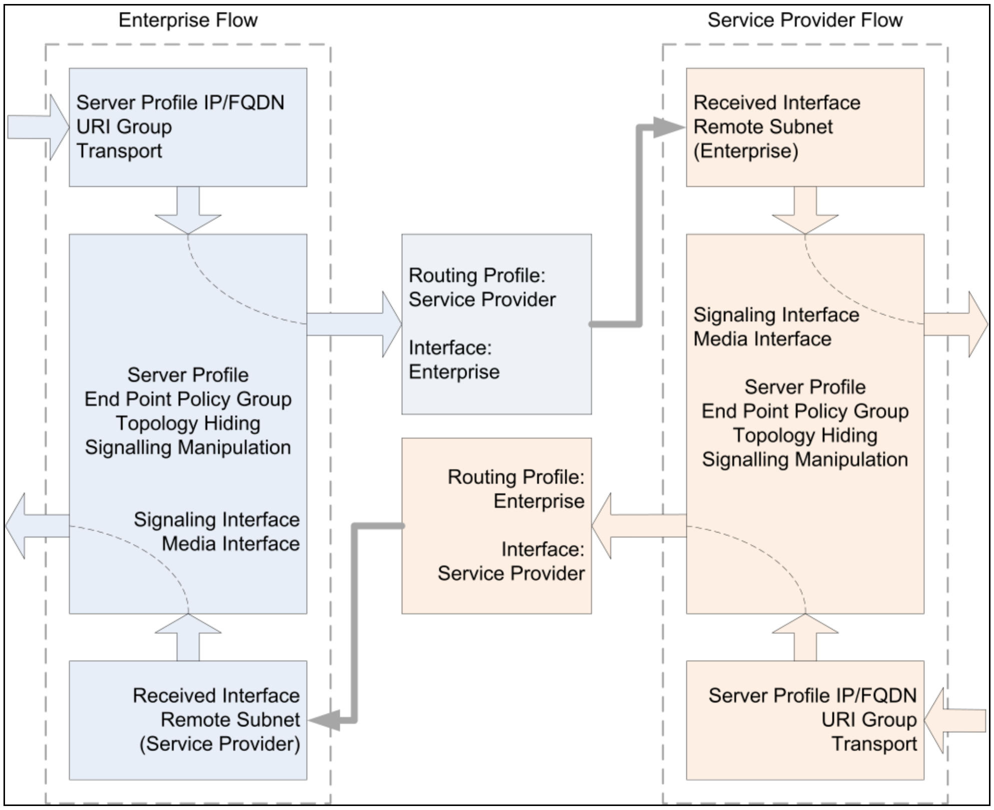
- Select **IP** or **IP/Domain** from the **Criteria** drop down menu. The default setting **IP/Domain** hides both domain names and IP addresses.
- Default action **Auto** in the **Replace Action** drop down menu replaces internal IP addresses or domain names with external IP addresses.
- If **Overwrite** is selected as the action, define the required domain name in the **Overwrite Value** field. This was not used during testing.
- Click on **Finish**.



To define Topology Hiding for Session Manager, follow the same process. During testing, the default profile was used so an additional profile was not required.

7.9. Server Flows

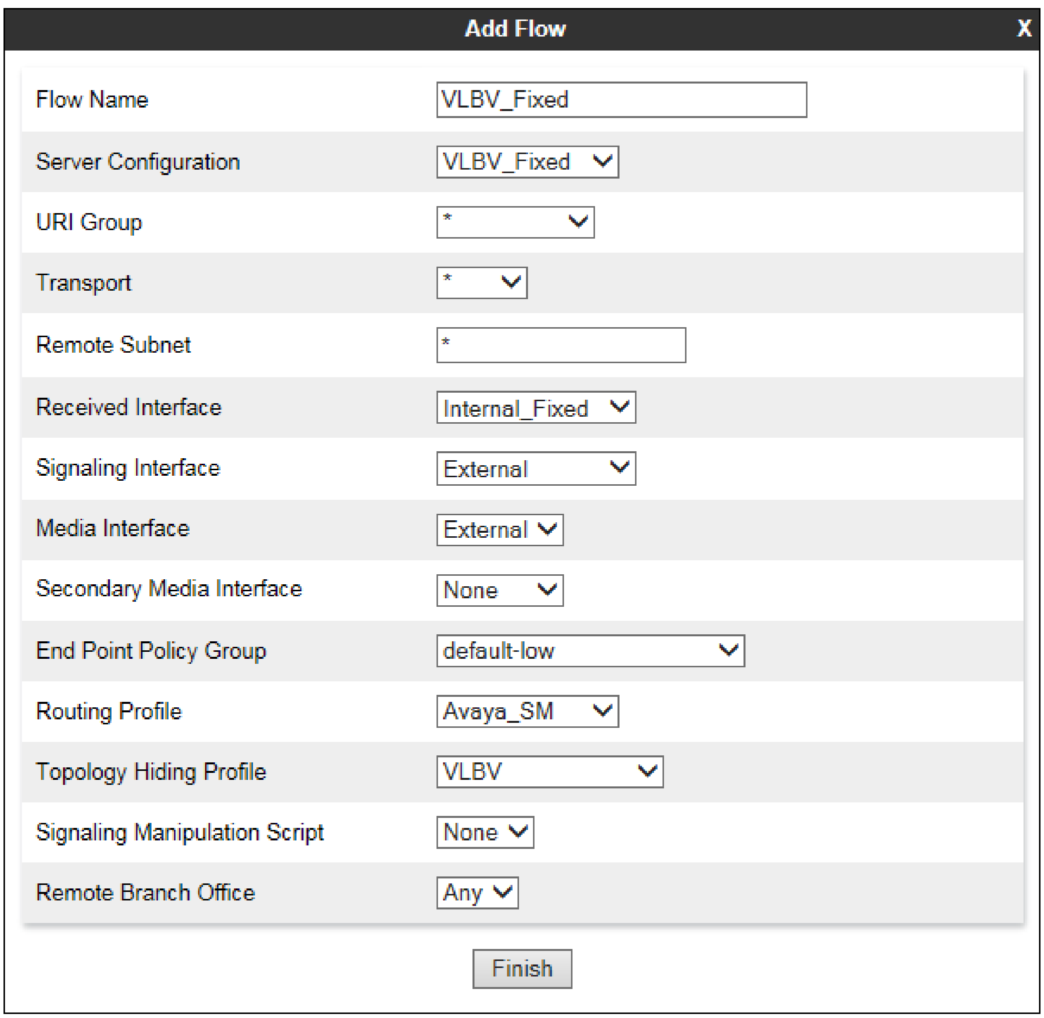
Server Flows combine the previously defined profiles into End Point Server Flows. The following diagram shows the inputs and outputs of the server flows:



Three End Point Server Flows are defined for the Vodafone Libertel SIP Trunking Service, one for off-net PSTN, one for on-net mobile and one for Session Manager. These End Point Server Flows allow calls to be routed from Session Manager to the Vodafone Libertel SIP Trunks and vice versa. To define a Server Flow for the Vodafone Libertel off-net PSTN trunk, navigate to **Device Specific Settings → End Point Flows**. Click on the **Server Flows** tab and click on **Add**.

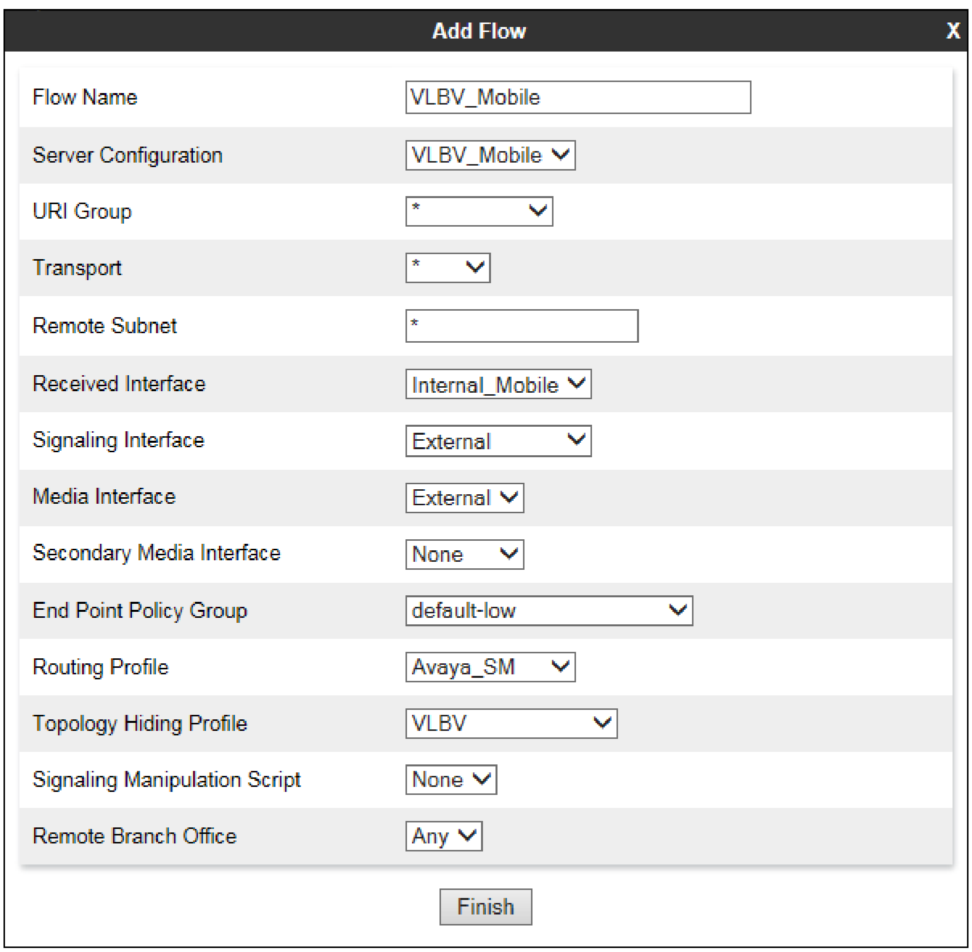
Enter details in the pop-up menu.

- In the **Flow Name** field enter a descriptive name for the server flow for the Vodafone Libertel off-net PSTN trunk, in the test environment **VLBV_Fixed** was used.
- In the **Server Configuration** drop-down menu, select the server configuration for the off-net PSTN trunk defined in **Section 7.6**.
- In the **Received Interface** drop-down menu, select the internal SIP signalling interface defined in **Section 7.4**. This is the interface that signalling bound for the off-net PSTN trunk is received on.
- In the **Signaling Interface** drop-down menu, select the external SIP signalling interface defined in **Section 7.4**. This is the interface that signalling bound for the off-net PSTN trunk is sent on.
- In the **Media Interface** drop-down menu, select the external media interface defined in **Section 7.4**. This is the interface that media bound for the off-net PSTN trunk is sent on.
- In the **Routing Profile** drop-down menu, select the routing profile of Session Manager defined in **Section 7.7**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of the Vodafone Libertel SIP Trunking Service defined in **Section 7.8** and click **Finish**.



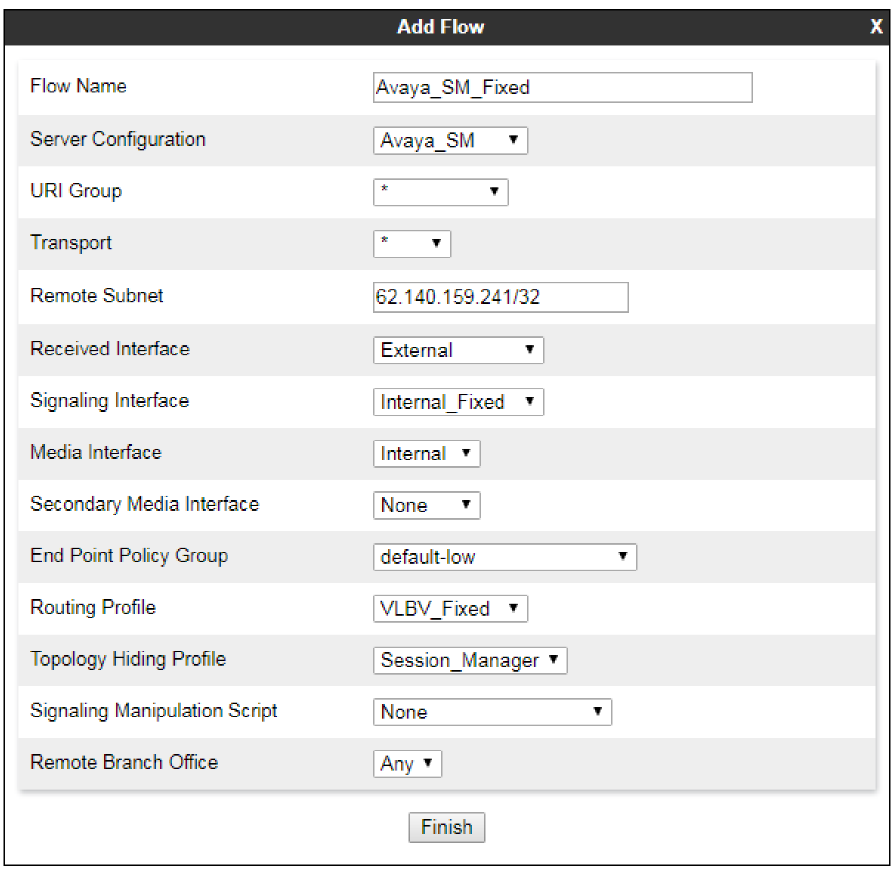
To define a Server Flow for the Vodafone Libertel on-net mobile trunk, return to **Device Specific Settings → End Point Flows** and click on the **Server Flows** tab.

- Select **Add Flow** and enter details in the pop-up menu.
- In the **Flow Name** field enter a descriptive name for the server flow for the Vodafone Libertel on-net mobile trunk, in the test environment **VLBV_Mobile** was used.
- In the **Server Configuration** drop-down menu, select the server configuration for the on-net mobile trunk defined in **Section 7.6**.
- In the **Received Interface** drop-down menu, select the internal SIP signalling interface defined in **Section 7.4**. This is the interface that signalling bound for the on-net mobile trunk is received on.
- In the **Signaling Interface** drop-down menu, select the external SIP signalling interface defined in **Section 7.4**. This is the interface that signalling bound for the on-net mobile trunk is sent on.
- In the **Media Interface** drop-down menu, select the external media interface defined in **Section 7.4**. This is the interface that media bound for the on-net mobile trunk is sent on.
- In the **Routing Profile** drop-down menu, select the routing profile of Session Manager defined in **Section 7.7**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of the Vodafone Libertel SIP Trunking Service defined in **Section 7.8** and click **Finish**.



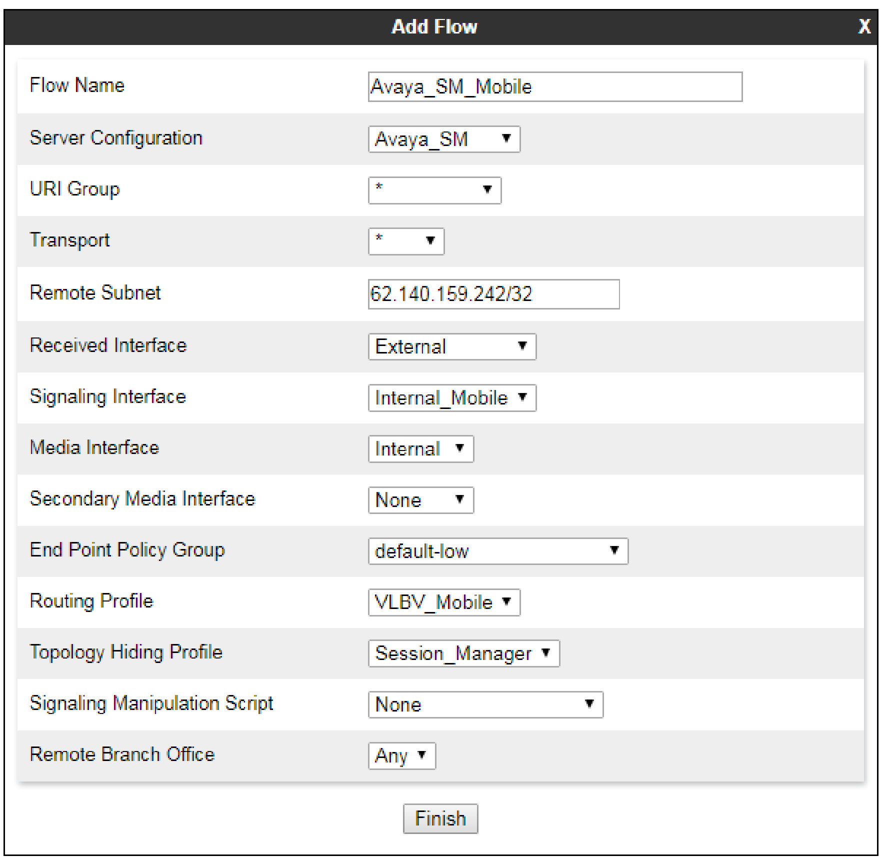
To define a Server Flow for Session Manager for the off-net PSTN traffic, return to **Device Specific Settings → End Point Flows** and click on the **Server Flows** tab.

- Select **Add Flow** and enter details in the pop-up menu.
- In the **Flow Name** field enter a descriptive name for the off-net PSTN server flow for Session Manager, in the test environment **Avaya_SM_Fixed** was used.
- In the **Server Configuration** drop-down menu, select the server configuration for Session Manager defined in **Section 7.6**.
- In the **Remote Subnet** field, enter the IP address of the off-net PSTN trunk as /.32
- In the **Received Interface** drop-down menu, select the external SIP signalling interface defined in **Section 7.4**. This is the interface that off-net PSTN signalling bound for Session Manager is received on.
- In the **Signaling Interface** drop-down menu, select the internal SIP signalling interface defined in **Section 7.4**. This is the interface that off-net PSTN signalling bound for Session Manager is sent on.
- In the **Media Interface** drop-down menu, select the internal media interface defined in **Section 7.4**. This is the interface that off-net PSTN media bound for Session Manager is sent on.
- In the **Routing Profile** drop-down menu, select the routing profile of the Vodafone Libertel SIP Trunking Service defined in **Section 7.7**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of Session Manager defined in **Section 7.8** and click **Finish**.

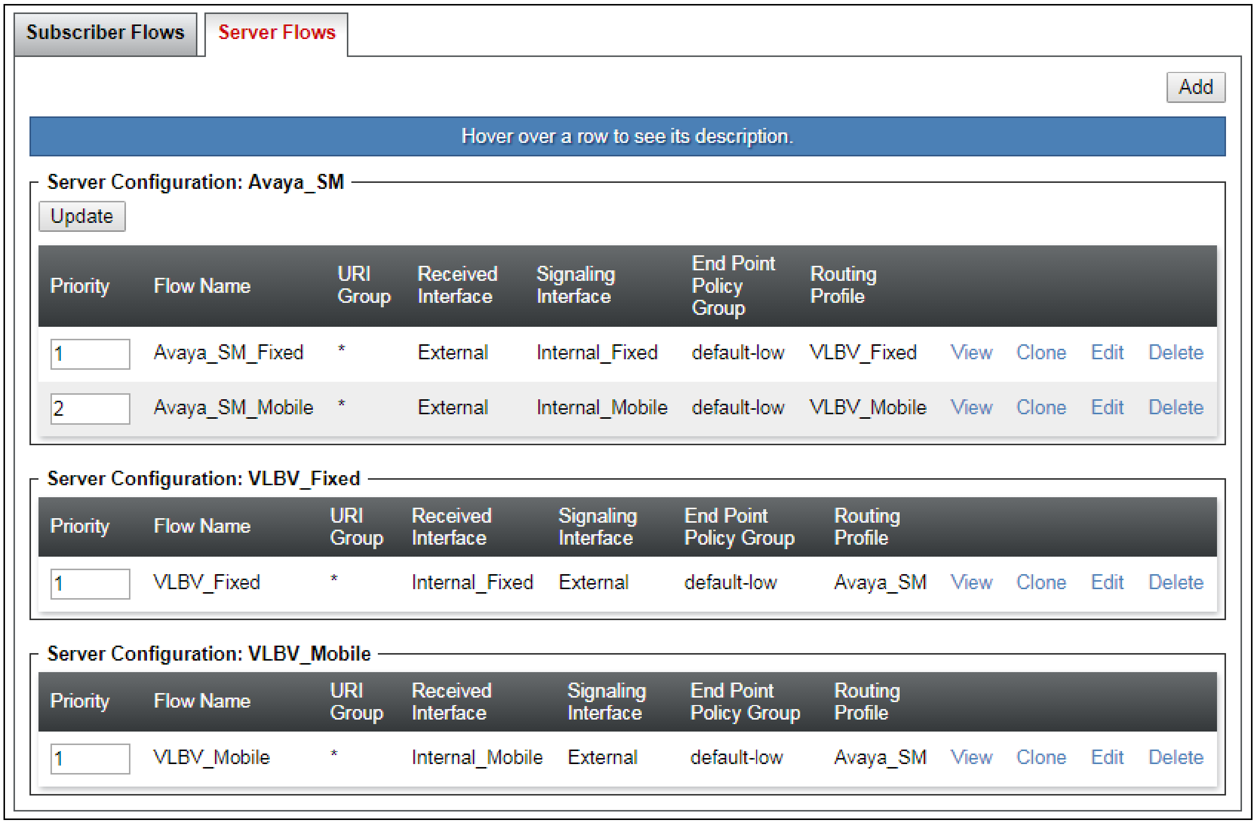


To define a Server Flow for Session Manager for the on-net mobile traffic, return to **Device Specific Settings → End Point Flows** and click on the **Server Flows** tab.

- Select **Add Flow** and enter details in the pop-up menu.
- In the **Flow Name** field enter a descriptive name for the o on-net mobile server flow for Session Manager, in the test environment **Avaya_SM_Mobile** was used.
- In the **Server Configuration** drop-down menu, select the server configuration for Session Manager defined in **Section 7.6**.
- In the **Remote Subnet** field, enter the IP address of the on-net mobile trunk as /.32
- In the **Received Interface** drop-down menu, select the external SIP signalling interface defined in **Section 7.4**. This is the interface that on-net mobile signalling bound for Session Manager is received on.
- In the **Signaling Interface** drop-down menu, select the internal SIP signalling interface defined in **Section 7.4**. This is the interface that on-net mobile signalling bound for Session Manager is sent on.
- In the **Media Interface** drop-down menu, select the internal media interface defined in **Section 7.4**. This is the interface that on-net mobile media bound for Session Manager is sent on.
- In the **Routing Profile** drop-down menu, select the routing profile of the Vodafone Libertel SIP Trunking Service defined in **Section 7.7**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of Session Manager defined in **Section 7.8** and click **Finish**.



The information for all Server Flows is shown on a single screen on the Avaya SBCE.



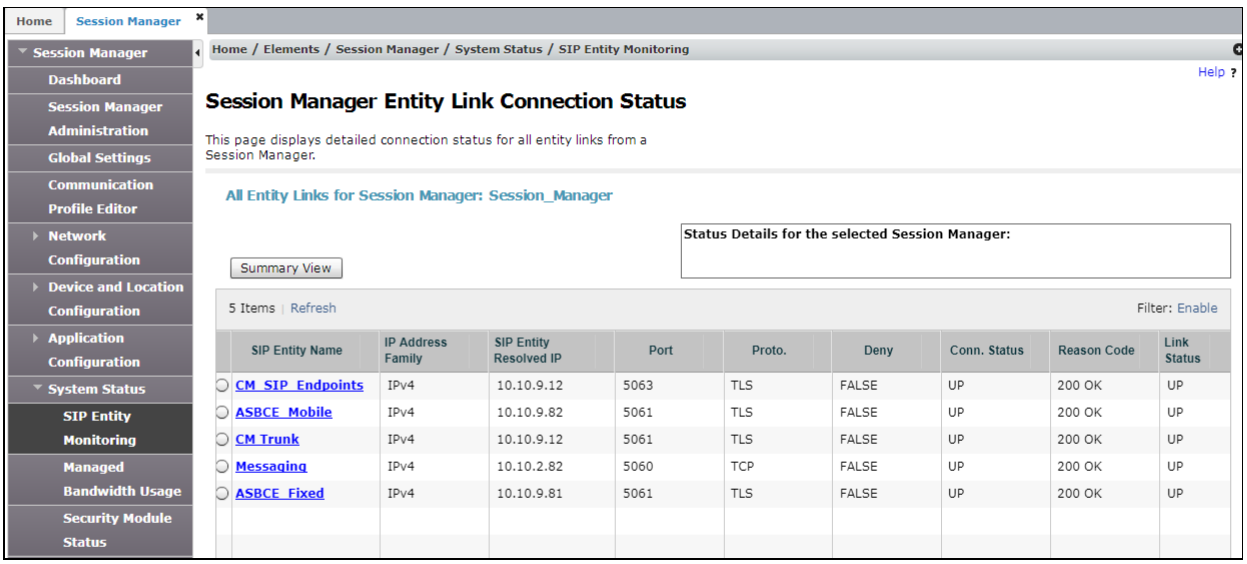
8. Configure the Vodafone Libertel SIP Trunking Service Equipment

The configuration of the Vodafone Libertel B.V. equipment used to support the SIP Trunk is outside the scope of these Application Notes and will not be covered. To obtain further information on Vodafone Libertel equipment and system configuration please contact an authorised Vodafone Libertel representative.

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**.



SIP Entity Name	IP Address Family	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
CM_SIP_Endpoints	IPv4	10.10.9.12	5063	TLS	FALSE	UP	200 OK	UP
ASBCE_Mobile	IPv4	10.10.9.82	5061	TLS	FALSE	UP	200 OK	UP
CM Trunk	IPv4	10.10.9.12	5061	TLS	FALSE	UP	200 OK	UP
Messaging	IPv4	10.10.2.82	5060	TCP	FALSE	UP	200 OK	UP
ASBCE_Fixed	IPv4	10.10.9.81	5061	TLS	FALSE	UP	200 OK	UP

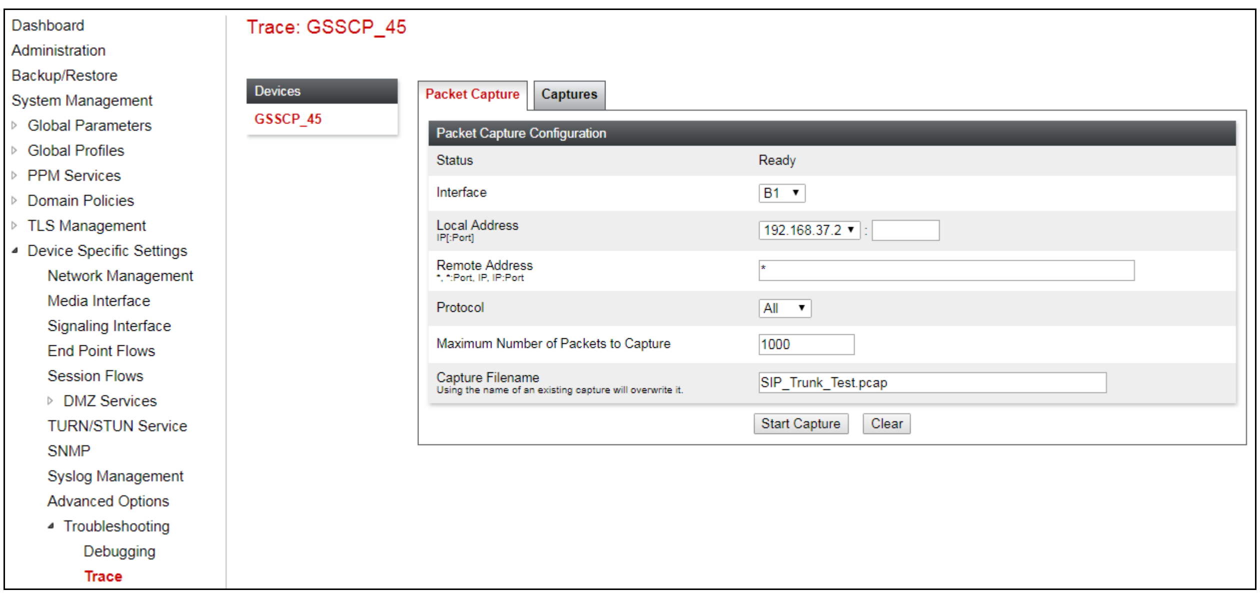
2. From Communication Manager SAT interface run the command **status trunk n** where **n** is a previously configured SIP trunk. Observe if all channels on the trunk group display **in-service/idle**.

```
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
```

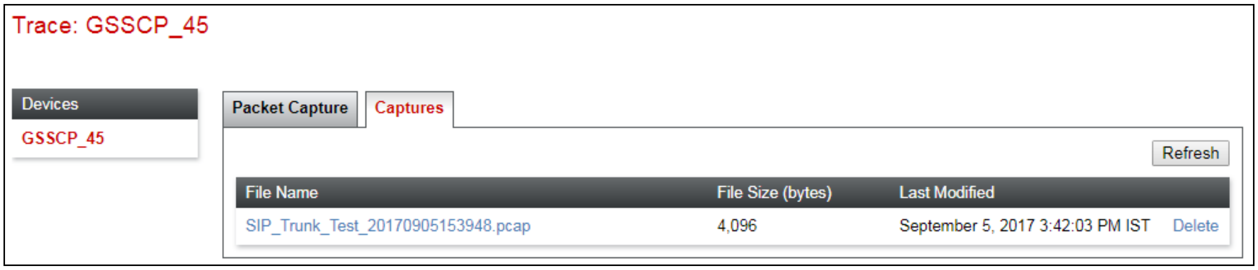
3. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active.
4. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active.
5. Verify that the user on the PSTN can end an active call by hanging up.
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**, **1000** is shown as an example.
- Specify the filename of the resultant pcap file in the **Capture Filename** field.
- Click on **Start Capture**.



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



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

10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura ® Communication Manager R7.1, Avaya Aura ® Session Manager 7.1 and Avaya Session Border Controller for Enterprise R7.2 to the Vodafone Libertel SIP Trunking Service. The Vodafone Libertel SIP Trunking 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] *Migrating and Installing Avaya Appliance Virtualization Platform*, Release 7.1, May 2017.

[2] *Upgrading and Migrating Avaya Aura® applications to Release 7.1.1 from System Manager*, Aug 2017.

[3] *Deploying Avaya Aura® applications from System Manager*, Release 7.1.1, Aug 2017

[4] *Deploying Avaya Aura® Communication Manager*, Release 7.1.1, Aug 2017

[5] *Administering Avaya Aura® Communication Manager*, Release 7.1.1, Aug 2017.

[6] *Upgrading Avaya Aura® Communication Manager*, Release 7.1.1, Aug 2017

[7] *Deploying Avaya Aura® System Manager Release 7.1.1*, Aug 2017

[8] *Upgrading Avaya Aura® System Manager to Release 7.1.1*, Aug 2017.

[9] *Administering Avaya Aura® System Manager for Release 7.1.1*, Aug 2017

[10] *Deploying Avaya Aura® Session Manager*, Release 7.1 May 2017

[11] *Upgrading Avaya Aura® Session Manager* Release 7.1.1, Aug 2017

[12] *Administering Avaya Aura® Session Manager* Release 7.1.1, Aug 2017,

[13] *Deploying Avaya Session Border Controller for Enterprise Release 7.2*, Sep 2017

[14] *Upgrading Avaya Session Border Controller for Enterprise Release 7.2*, Aug 2017

[15] *Administering Avaya Session Border Controller for Enterprise Release 7.2*, Sep 2017

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

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.