Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Communication Manager R6.3, Avaya Aura® Session Manager R6.3 and Avaya Session Border Controller for Enterprise with Enterprise TLS and SRTP to support Vodafone Germany SIP Trunk Service - Issue 1.0

Abstract

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between the Vodafone Germany SIP Trunk 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. Within the Enterprise, TLS was used for transport of signalling and SRTP was used for transport of media to provide a secure solution. Vodafone Germany 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 Germany SIP Trunk Service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Session Border Controller for Enterprise (Avaya SBCE), Avaya Aura® Session Manager and Avaya Aura® Communication Manager Evolution Server. Customers using this Avaya SIP-enabled enterprise solution with Vodafone Germany SIP Trunk are able to place and receive PSTN calls via a dedicated Internet connection and the SIP protocol. 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 use the SIP Trunking service provided by Vodafone Germany.

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.

2.1. Interoperability Compliance Testing

The interoperability test included the following:

- Incoming calls to the enterprise site from PSTN phones using the SIP Trunk provided by Vodafone, calls made to analogue, SIP and H.323 endpoints at the enterprise.
- Outgoing calls from the enterprise site completed via Vodafone SIP Trunk to PSTN destinations, calls made from analogue, SIP and H.323 endpoints.
- Calls using the G.711A, G.729 and G.726-32 codecs.
- 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 (also known as "shuffling") with SIP and H.323 telephones.
- Secure transport of media within the enterprise using SRTP and transport of signalling using TLS.
- Call coverage and call forwarding for endpoints at the enterprise site.
- Transmission and response of SIP OPTIONS messages sent by Vodafone SIP Trunk requiring Avaya response and sent by Avaya requiring Vodafone response.

2.2. Test Results

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

- During testing, there was a lag of over one second in the media on some outbound calls. This was thought to be an issue with the test network.
- When a call was put on hold in the network, there was no signalling to the enterprise to indicate that the call is on hold. This meant the test was not a valid test of SIP trunk functionality.
- Placing a call on hold for longer than twenty minutes resulted in a disconnect (BYE) from the network.
- At the time of testing, T.38 fax was not supported on the Vodafone SIP trunk.
- When testing mobility, EC500 Confirmed Answer was not successful. This function is not critical for SIP certification.
- Avaya one-X® Communicator did not function correctly when connected via SIP and tests were not completed. Although outbound calls were successful, inbound calls were rejected with SIP "488 Not Acceptable Here". The fault appears to be unsuccessful negotiation of SRTP for the media. Fault report ONEXC-8777 is outstanding for this issue.

2.3. Support

For technical support on Vodafone Germany products please visit the website at www.vodafone.de or contact an authorized Vodafone representative.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an Enterprise site connected to Vodafone SIP Trunk. Located at the Enterprise site is an Avaya Session Border Controller for Enterprise, Session Manager and Communication Manager. Endpoints are Avaya 96x0 series and Avaya 96x1 series IP telephones (with SIP and H.323 firmware), Avaya 46xx series IP telephones (with H.323 firmware), Avaya 16xx series IP telephones (with H.323 firmware), Avaya A175 Desktop Video Device running Flare Experience (audio only), Avaya analogue telephones and an analogue fax machine. Also included in the test configuration was an Avaya one-X® Communicator soft phone and Avaya Flare® Experience for Windows running on a laptop PC. Within the enterprise, TLS was used for secure signalling transport and SRTP for media.

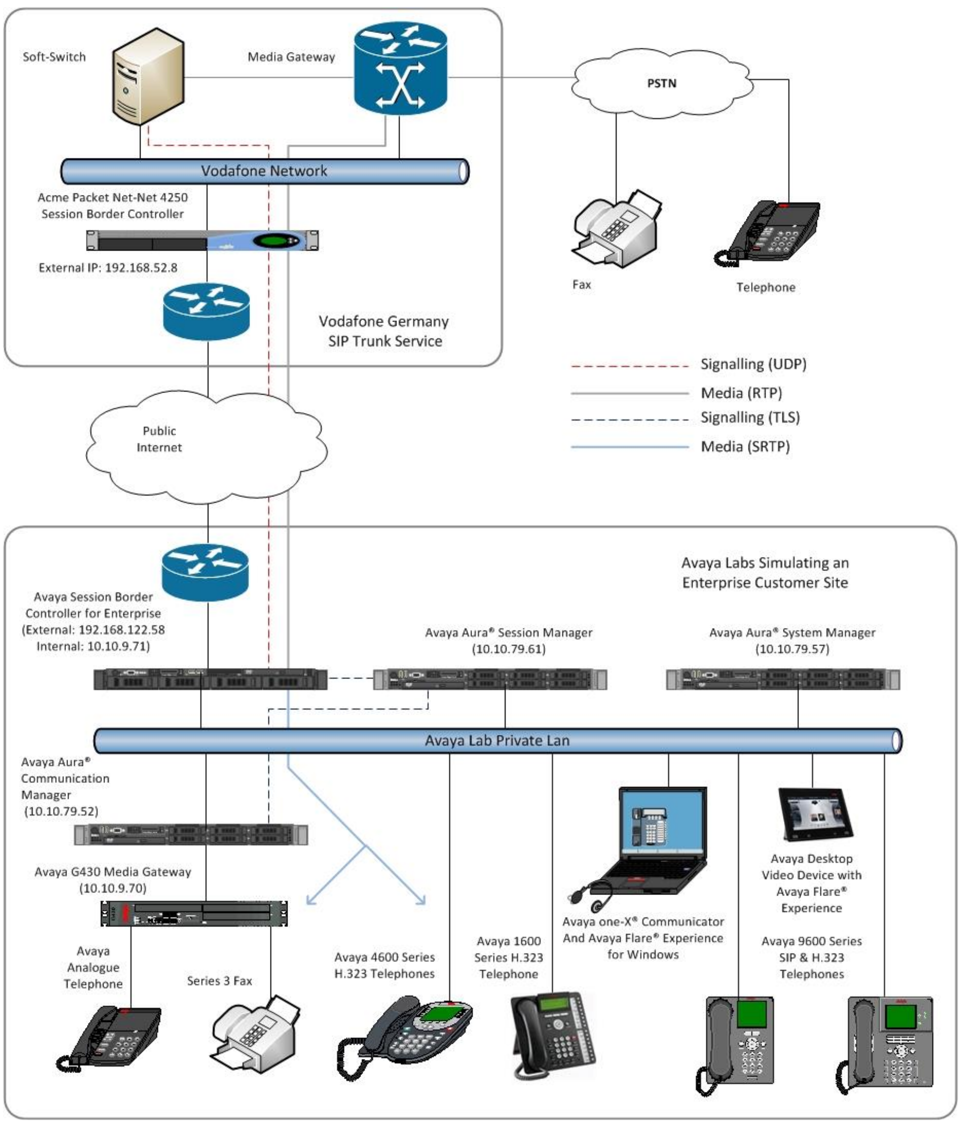


Figure 1: Test Setup Vodafone Germany SIP Trunk to Avaya Enterprise

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya	
Dell PowerEdge R620 running Session Manager on VM Version 8	6.3.4.0.634014 VMware Tools: 9.0.0.15210 (782409)
Dell PowerEdge R620 running System Manager on VM Version 8	6.3.8.0 Build No. 6.3.0.8.5682 Patch 6.3.8.2651 Build No. 6.3.4.4.1904
Dell PowerEdge R620 running Communication Manager on VM Version 8	R016x.03.0.124.0 patch 21106
Avaya Session Border Controller Advanced for Enterprise Server	6.2.0.Q48
G430 Media Gateway	FW Version/HW Vintage: 34.5.1/1
Avaya 1616 Phone (H.323)	1.3 Maintenance Release 4
Avaya 4621 Phone (H.323)	2.9 SP 2
Avaya 96x0 Phone (H.323)	3.2.1
Avaya A175 Desktop Video Device (SIP)	Flare Experience Release 1.1.2
Avaya 9630 Phone (SIP)	R2.6.9
Avaya 9608 Phone (SIP)	R6.3.0
Avaya one–X® Communicator (H.323) on Lenovo T510 Laptop PC	6.1.9.04-SP9-132
Avaya Flare experience for Windows on Lenovo T510 Laptop PC	Release 1.1.3.14
Analogue Handset	NA
Analogue Fax	NA
Vodafone	
ACME Net-Net 4250 SBC	SC6.1.0 MR-5 GA (Build 704)
Italtel iSSW Softswitch	20.50.40

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 SIP Trunk. For incoming calls, the Session Manager receives SIP messages from the Avaya SBC for Enterprise (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 the Session Manager. The Session Manager directs the outbound SIP messages to the Avaya SBCE at the enterprise site that then sends the SIP messages to the Vodafone 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 SIP Trunk network, and any other SIP trunks used.

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

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

On **Page 4**, verify that **IP Trunks** field is set to **y**. Check also that **Media Encryption Over IP** is set to **y** so that SRTP can be used within the enterprise.

```
display system-parameters customer-options                          Page   4 of  11
                                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 the Session Manager. In this case, **SMVM1** and **10.10.79.61** are the **Name** and **IP Address** for the Session Manager SIP interface. Also note the **procr** name 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
SMVM1               10.10.79.61
default             0.0.0.0
procr               10.10.79.52
procr6              ::
```

5.3. Administer IP Network Region

Use the **change ip-network-region 1** command to 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, the media stream is established directly between the enterprise end-point and the internal media interface of the Avaya SBCE.
- The **Codec Set** is set to the number of the IP codec set to be used for calls within the IP network region. In this case, codec set **1** is used.
- The rest of the fields can be left at default values.

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

5.4. Administer IP Codec Set

Open the **IP Codec Set** form for the codec set specified in the IP Network Region form in **Section 5.3.** Enter the list of audio codec's eligible to be used in order of preference. For the interoperability test the codecs supported by Vodafone were configured, namely **G.711A**, **G.729** and **G.726-32**. The **Media Encryption** fields are present if **Media Encryption Over IP** is set to **y** in the **system-parameters customer-options** form defined in **Section 5.1**. If SRTP is to be used for media within the enterprise, select the encryption. For the interoperability test, **1-srtp-aescm128-hmac80** was selected.

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

                          IP Codec Set

    Codec Set: 1

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


     Media Encryption
 1: 1-srtp-aescm128-hmac80
 2:
 3:
```

Vodafone SIP Trunk does not support T.38 for transmission of fax. To allow transmission using G.711, Navigate to **Page 2** and set the **FAX - Mode** to **t.38-G711-fallback**.

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

                          IP Codec Set

                              Allow Direct-IP Multimedia? n




                    Mode                    Redundancy
    FAX             t.38-G711-fallback       0             ECM: y
    Modem           off                      0
    TDD/TTY         US                       3
    Clear-channel   n                        0
```

Note: The fax **Mode** can be set to **off** to allow transport of fax using G.711. During test **t.38-G711-fallback** was used so that incoming fax calls set up with G729 would renegotiate to G.711. This renegotiation did not take place for outgoing fax calls and as a result they were only successful when G.711 was the first codec negotiated.

5.5. Administer SIP Signaling Groups

This signalling group (and trunk group) will be used for inbound and outbound PSTN calls to the Vodafone SIP Trunk network. During test, this was configured to use TLS and port 5061 to represent the security requirements for signalling that may be in place at the customer's site. Configure the **Signaling Group** using the **add signaling-group x** command as follows:

- Set **Group Type** to **sip**.
- Set **Transport Method** to **tls**.
- Set **Enforce SIPS URI for SRTP?** to **n** as SRTP and TLS are only used within the enterprise and are not to be used end to end.
- Set **Peer Detection Enabled** to **y** allowing the 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 the Session Manager (node name **SMVM1** as defined in the **IP Node Names** form shown in **Section 5.2**).
- Set **Near-end Listen Port** and **Far-end Listen Port** to **5061** (Common TLS port value).
- 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 network region **1**).
- Leave **Far-end Domain** blank (allows the CM to accept calls from any SIP domain on the associated trunk).
- Set **Direct IP-IP Audio Connections** to **y**.
- Leave DTMF over IP at default value of **rtp-payload** (Enables **RFC2833** for DTMF transmission from the Communication Manager).

The default values for the other fields may be used.

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

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                 Enforce SIPS URI for SRTP? n
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n

   Near-end Node Name: procr                  Far-end Node Name: SMVM1
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 1

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

5.6. Administer SIP Trunk Group

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

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

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

Group Number: 1                   Group Type: sip           CDR Reports: y
  Group Name: OUTSIDE CALL                COR: 1        TN: 1        TAC: 101
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 1
                                                     Number of Members: 10
```

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

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

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 10000

            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 test, CLI was sent as the national number with no leading zeros. This format was successfully verified in the network.

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




                     Numbering Format: private
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n
```

On **Page 4** of this form:

- Set **Send Transferring Party Information** to **y**.
- Set **Send Diversion Header** to **y**.
- Set **Support Request History** to **n** as the required information for forwarded, transferred and mobility calls will be sent in the Diversion and Transferring Party Information headers.
- Set the **Telephone Event Payload Type** to **98**.
- Set the **Identity for Calling Party Display** to **From** to ensure that where CLI for incoming calls is withheld, it is not displayed on the Communication Manager extension.

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

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

                                     Send Diversion Header? y
                                   Support Request History? n
                              Telephone Event Payload Type: 98


                        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
```

Note: The Payload Type is a dynamic value and the meaning is agreed during codec negotiation which was tested successfully. The value used is therefore not critical, 98 is shown as that is the value used during testing. The Payload Type defined on Communication Manager is not applied to calls from SIP end-points.

5.7. Administer Calling Party Number Information

Use the **change private-unknown-numbering** command to configure Communication Manager to send the calling party number in the format required. In test, calling party number was sent as the national number with leading zero as the format expected in the network for calling party number verification. This calling party number is sent in the SIP From, Contact and PAI headers as well as the Diversion header for forwarded calls. The number is 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  2000             1          069138nnnn100    13    Total Administered: 9
 4  2208             1          069138nnnn103    13       Maximum Entries: 540
 4  2316             1          069138nnnn105    13
 4  2346             1          069138nnnn102    13
 4  2396             1          069138nnnn101    13
 4  2400             1          069138nnnn106    13
 4  2401             1          069138nnnn106    13
 4  2460             1          069138nnnn107    13
 4  2611             1          069138nnnn104    13
```

Note: The private numbers in the above screenshot have been modified for security.

5.8. Administer Route Selection for Outbound Calls

In the test environment, the Automatic Route Selection (ARS) feature was used to route outbound calls via the SIP trunk to Vodafone SIP Trunk. The single digit **9** was used as the ARS access code providing a facility for telephone users to dial 9 to reach an outside line. Use the **change feature-access-codes** command to configure a digit as the **Auto Route Selection (ARS) - Access Code 1**.

```
change feature-access-codes                                        Page   1 of  10
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code: *69
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 7
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
```

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

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

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    0                     11   14    1        pubu         n
    00                    13   15    1        pubu         n
    0035391               13   13    1        pubu         n
    0800                  8    14    1        pubu         n
    118                   3    6     1        pubu         n
```

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

```
change route-pattern 1                                            Page   1 of   3
                    Pattern Number: 1      Pattern Name:
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                           DCS/ IXC
    No          Mrk Lmt List Del  Digits                             QSIG
                             Dgts                                    Intw
 1: 1    0                                                            n   user
 2:                                                                   n   user
 3:                                                                   n   user
 4:                                                                   n   user
 5:                                                                   n   user
 6:                                                                   n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                               unk-unk   none
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
 6: y y y y y n  n            rest                                         none
```

5.9. Administer Incoming Digit Translation

This step configures the settings necessary to map incoming DDI calls to the Communication Manager extensions. The incoming digits sent in the INVITE message from Vodafone can be manipulated as necessary to route calls to the desired extension. During test, the incoming DDI numbers were changed in the Session Manager to the Communication Manager Extension number using an adaptation. When done this way, there is no requirement for any incoming digit translation in the Communication Manager. If incoming digit translation is required, use the **change inc-call-handling-trmt trunk-group x** command where **x** is the Trunk Group defined in **Section 5.6**.

```
change inc-call-handling-trmt trunk-group 1                     Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
```

Note: One reason for configuring the enterprise in this way is to ensure that the message waiting indicator is successfully sent to SIP extensions when a voice mail message is available and unread.

5.10. EC500 Configuration

When EC500 is enabled on the 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 2396. Use the command **change off-pbx-telephone station-mapping x** where **x** is the 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. The normal ARS code is not required here.
- For the **Phone Number** enter the phone that will also be called (e.g. **0035389434nnnn**).
- Set the **Trunk Selection** to **1** so that Trunk Group 1 will be used for routing.
- Set the **Config Set** to **1**.

```
change off-pbx-telephone station-mapping 2396                   Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 2396             EC500        -     0035389434nnnn   1           1
-
```

Note: The phone number shown is for a mobile phone used for testing at Avaya Labs and is in international format with international dialling prefix 00. To use facilities for calls coming in from EC500 mobile phones, the number received in Communication Manager must exactly match the number specified in the above table.

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

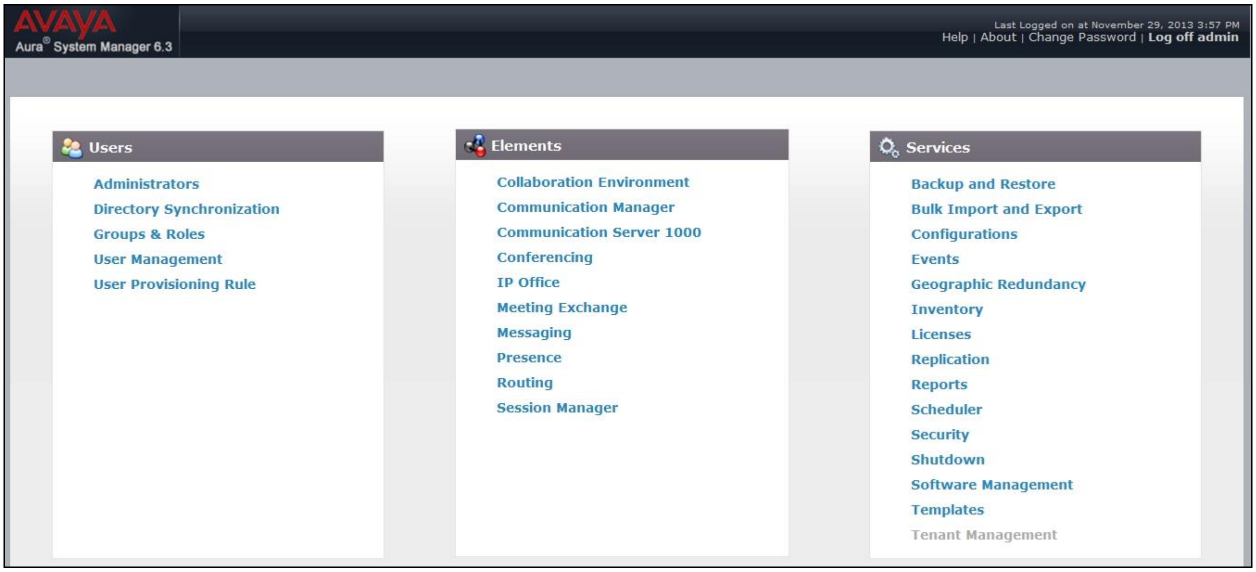
6. Configuring Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The Session Manager is configured via the 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

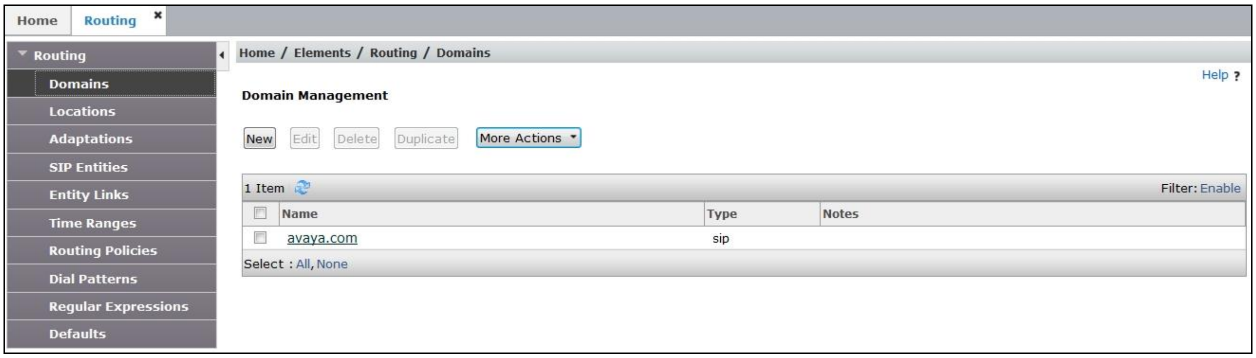
6.1. Log in to Avaya Aura® System Manager

Access the System Manager using a Web Browser by 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

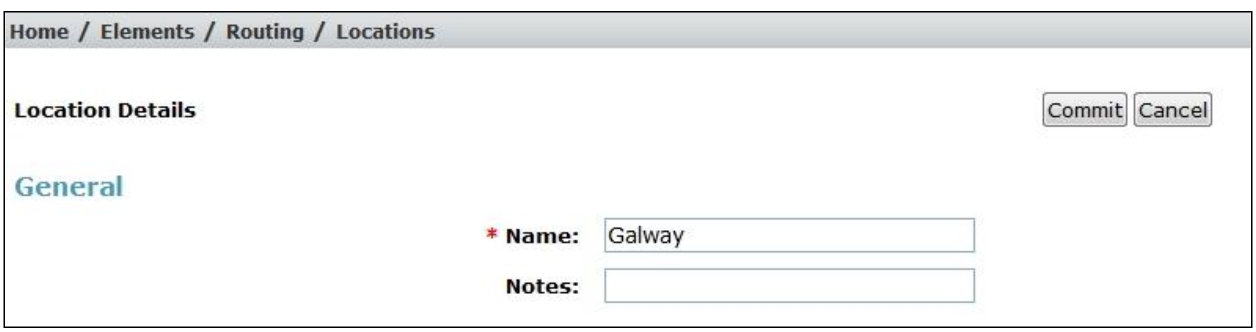
To add the SIP domain that will be used with Session Manager, select **Routing** from the **Home** tab menu and in the resulting tab select **Domains** from left hand menu. Click the **New** button to create a new SIP domain entry. In the **Name** field enter the domain name of the enterprise site or a name agreed with Vodafone; 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 test, **avaya.com** was used. Optionally, a description for the domain can be entered in the Notes field (not shown). Click **Commit** to save changes.



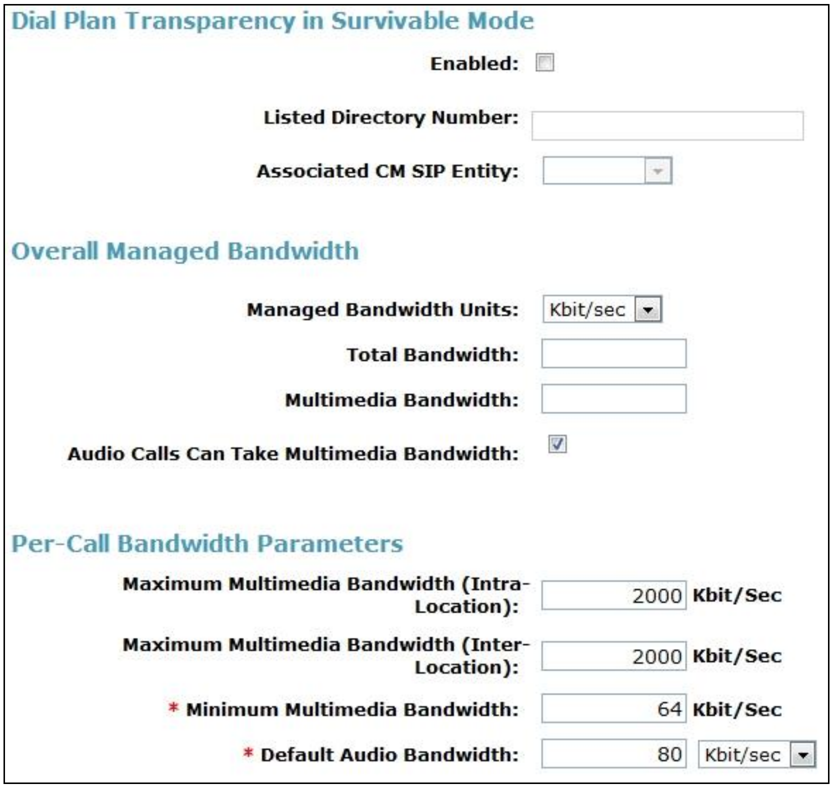
Note: If the existing domain name used in the enterprise equipment does not match that used in the network, a Session Manager adaptation can be used to change it, or it can be changed on the Avaya SBCE.

6.3. Administer Locations

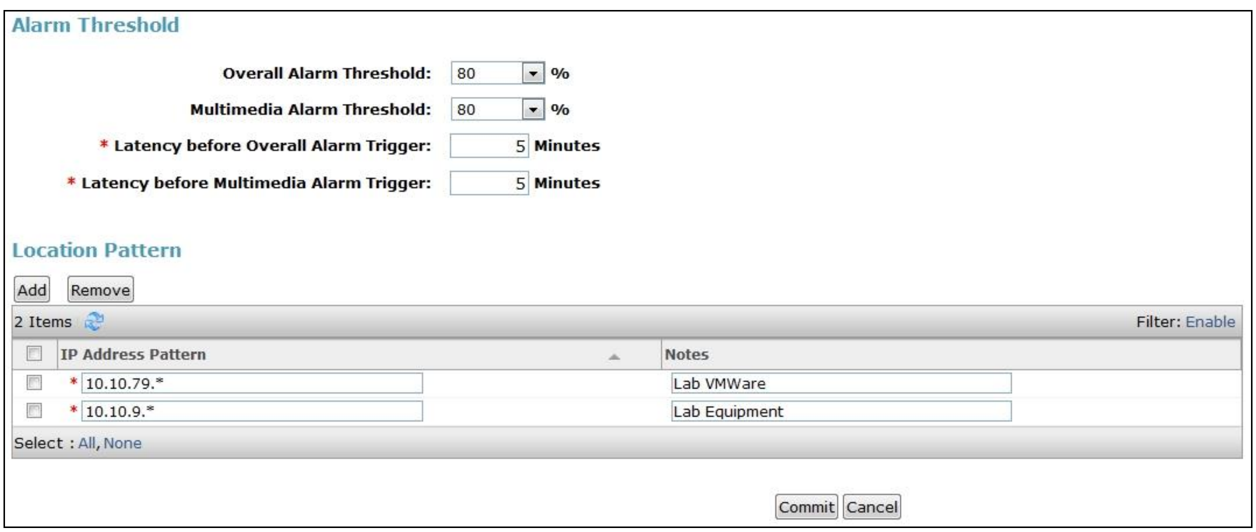
Locations can be used to identify logical and/or physical locations where SIP Entities reside for the purposes of bandwidth management. One location is added to the sample configuration for all of the enterprise SIP entities. On the **Routing** tab select **Locations** from the left hand menu. Under **General**, in the **Name** field, enter an informative name for the location.



Scroll down for bandwidth configuration. During testing, these were left at default values.



Scroll to the bottom of the page and under **Location Pattern**, click **Add**, then enter an **IP Address Pattern** in the resulting new row, * is used to specify any number of allowed characters at the end of the string. Below is the location configuration used for the test enterprise.

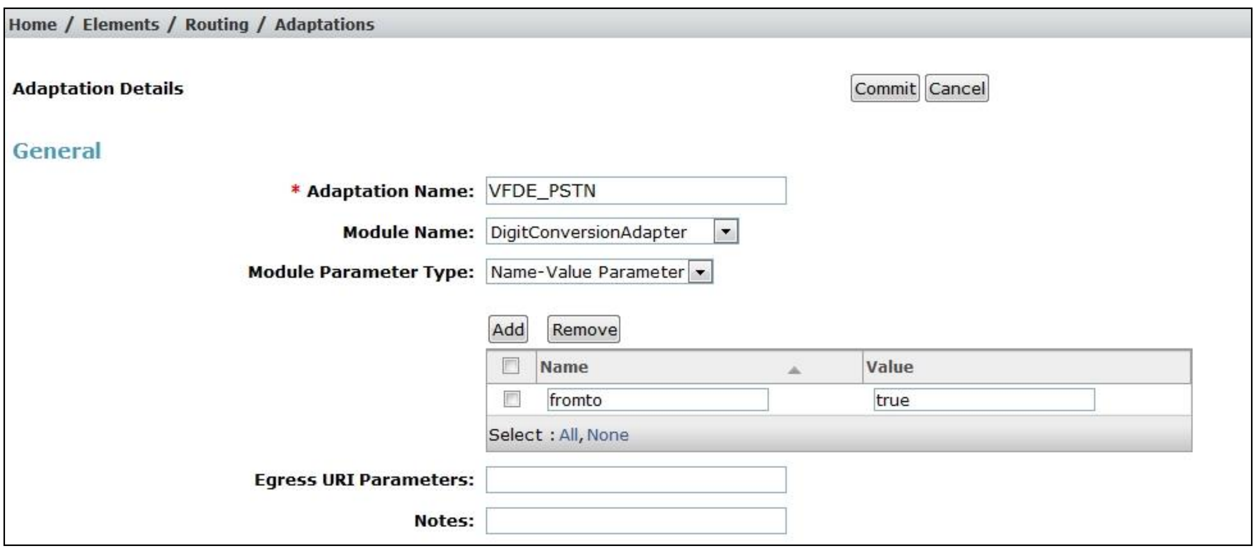


6.4. Administer Adaptations

Calls from Vodafone are received at the enterprise in national format with leading “0” on the Request URI. An Adaptation specific to Vodafone is used to convert the called number to an extension number as defined in the Communication Manager before onward routing to Communication Manager SIP Entity and removes the requirement for incoming digit manipulation on Communication Manager. It is also applied to messages coming from Communication Manager so that the SIP PUBLISH message for message waiting indicator on SIP end-points is handled correctly.

On the **Routing** tab select **Adaptations** from the left-hand menu. Click on **New** (not shown).

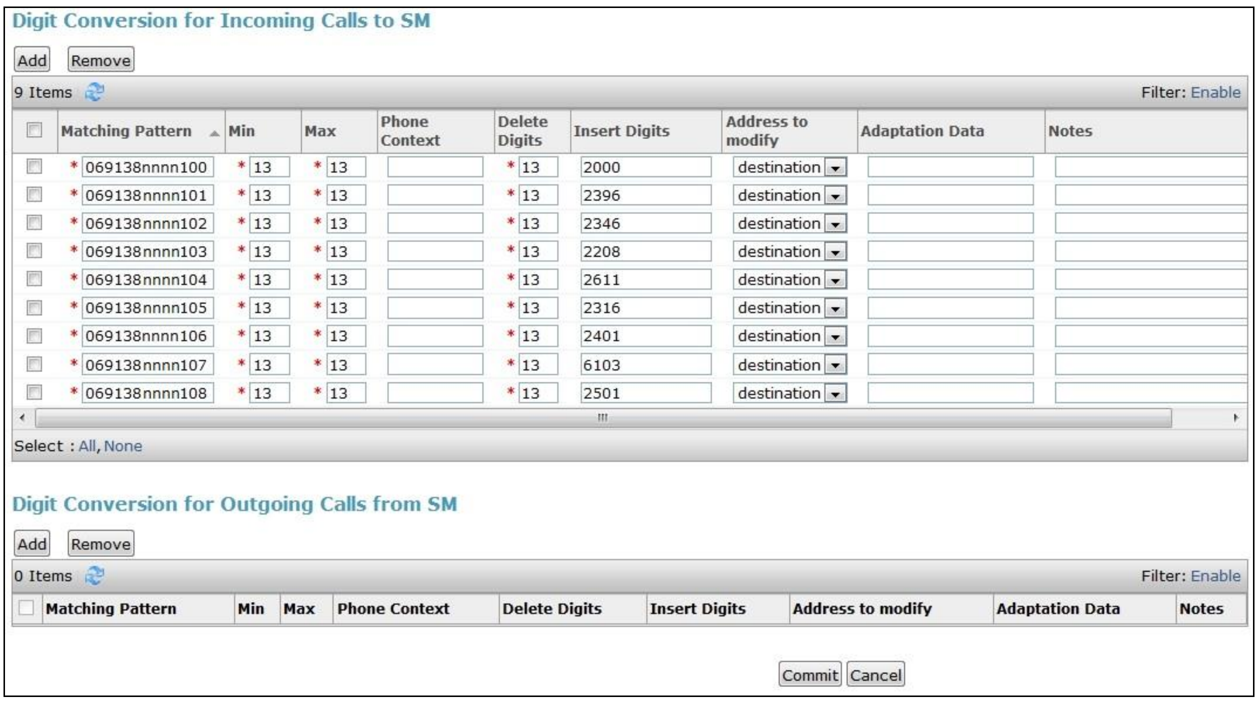
- In the **Adaptation name** field, enter a descriptive title for the adaptation.
- In the **Module name** enter **DigitConversionAdapter**. This is used for simple digit conversion adaptations.
- In the **Module parameter** field, select **Name-Value Parameter** in the **Module Parameter Type** drop down menu and enter **fromto** with a value of **true** in the resultant dialogue box. This will apply the adaptation to the From and To headers as well as the Request URI.



Scroll down and in the section **Digit Conversion for Incoming Calls to SM**, click on **Add**. An additional row will appear. This allows information to be entered for the manipulation of numbers coming from the network. This is where the called party number is translated from national format to the extension number for termination of calls on Communication Manager.

The screenshot below shows a translation for each called party number. This is not normally necessary where the extension number forms part of the national number. When this is the case, a deletion of the leading digits is required.

- Under **Matching Pattern** enter the DDI number as received from the network.
- Under **Min** and **Max** enter the Minimum and Maximum digits of the incoming DDI number.
- Under **Delete Digits** enter the number of digits to delete to leave only the extension number remaining, during test all had to be deleted as the extension number did not form part of the national number.
- Under **Insert Digits** enter digits to be inserted. During test, this was the full extension number. If the extension number forms part of the DDI number, there will be no entry required here.
- Under **Address to Modify** choose **destination** from the drop down box to apply this rule to the To and Request-Line headers only.



Digit Conversion for Incoming Calls to SM

Add Remove

9 Items Filter: Enable

Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
* 069138nnnn100	* 13	* 13		* 13	2000	destination		
* 069138nnnn101	* 13	* 13		* 13	2396	destination		
* 069138nnnn102	* 13	* 13		* 13	2346	destination		
* 069138nnnn103	* 13	* 13		* 13	2208	destination		
* 069138nnnn104	* 13	* 13		* 13	2611	destination		
* 069138nnnn105	* 13	* 13		* 13	2316	destination		
* 069138nnnn106	* 13	* 13		* 13	2401	destination		
* 069138nnnn107	* 13	* 13		* 13	6103	destination		
* 069138nnnn108	* 13	* 13		* 13	2501	destination		

Select : All, None

Digit Conversion for Outgoing Calls from SM

Add Remove

0 Items Filter: Enable

Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes

Commit Cancel

Note: In the above screenshots the DDI numbers are partially obscured.

6.5. Administer SIP Entities

A SIP Entity must be added for each SIP-based telephony system supported by a SIP connection to the 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**:

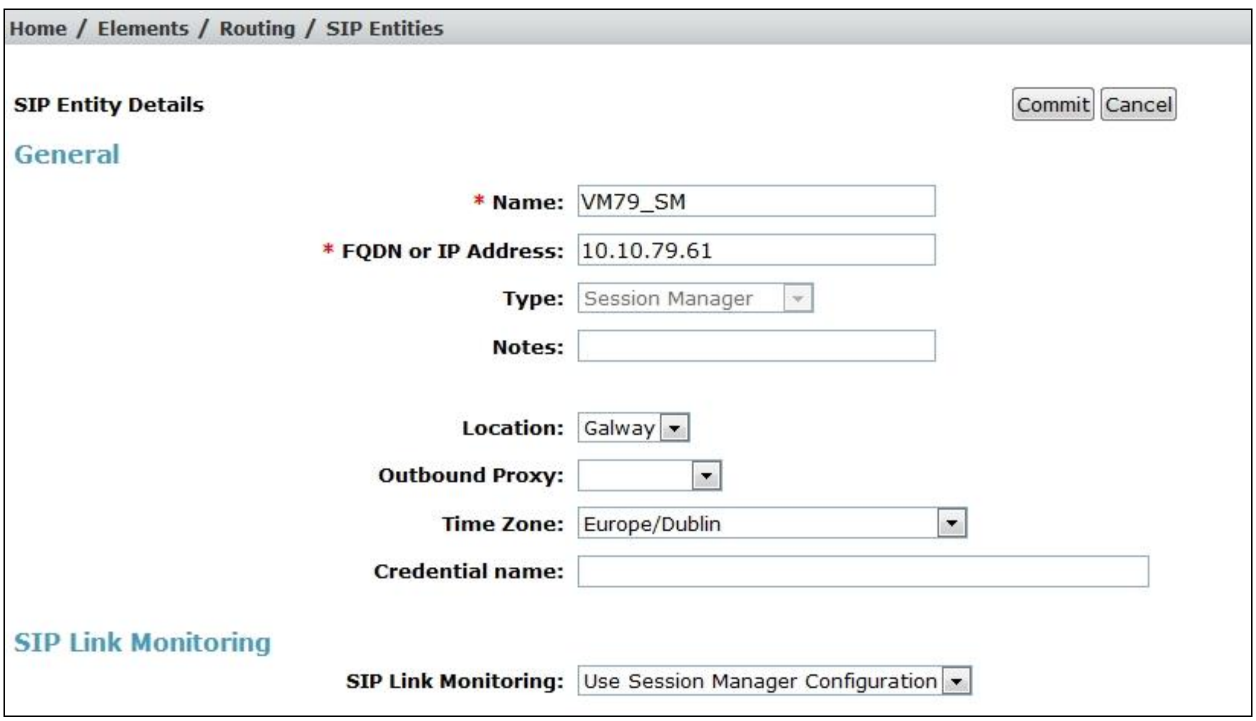
- In the **Name** field enter an informative name.
- In the **FQDN or IP Address** field enter the IP address of the Session Manager or the signalling interface on the connecting system.
- In the **Type** field use **Session Manager** for a Session Manager SIP entity, **CM** for a Communication Manager 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 three SIP Entities:

- Avaya Aura® Session Manager SIP Entity.
- Avaya Aura® Communication Manager SIP Entity.
- Avaya Session Border Controller for Enterprise (Avaya SBCE) SIP Entity.

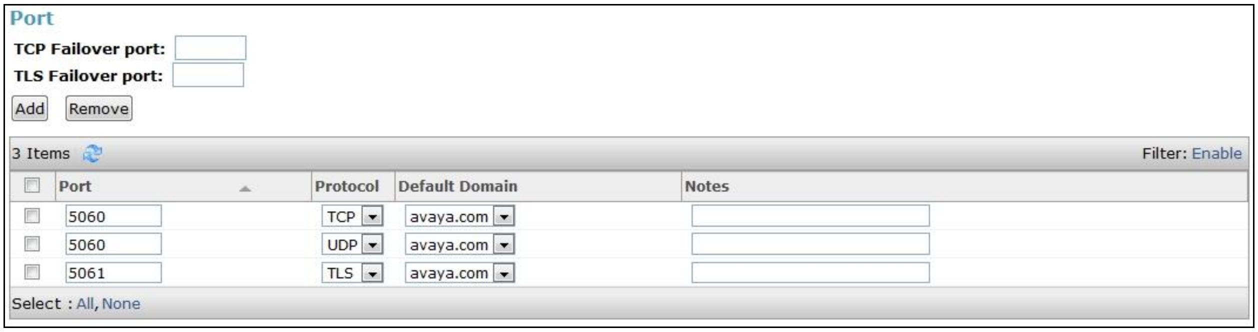
6.5.1. Avaya Aura® Session Manager SIP Entity

The following screens show the SIP entity for Session Manager. The **FQDN or IP Address** field is set to the IP address of the Session Manager SIP signalling interface.



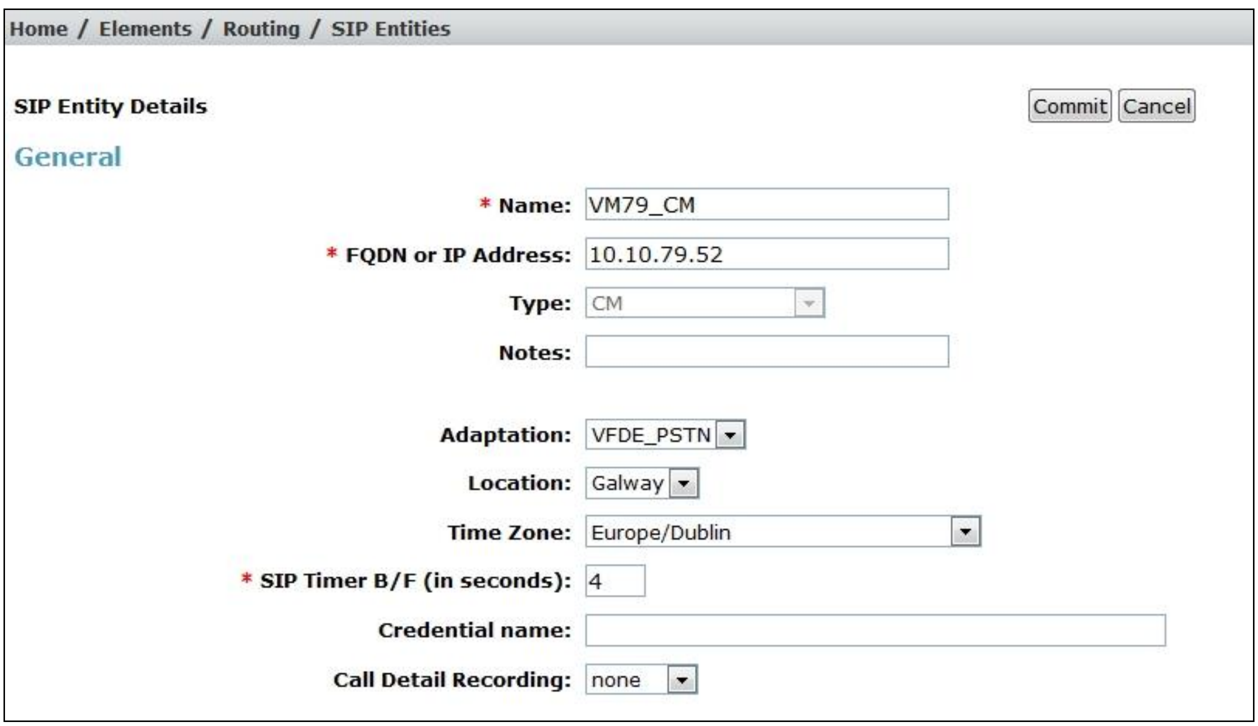
The Session Manager must be configured with the port numbers on the protocols that will be used by the other SIP entities. To configure these scroll to the bottom of the page and under **Port**, click **Add**, then edit the fields in the resulting new row.

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



6.5.2. Avaya Aura® Communication Manager SIP Entity

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

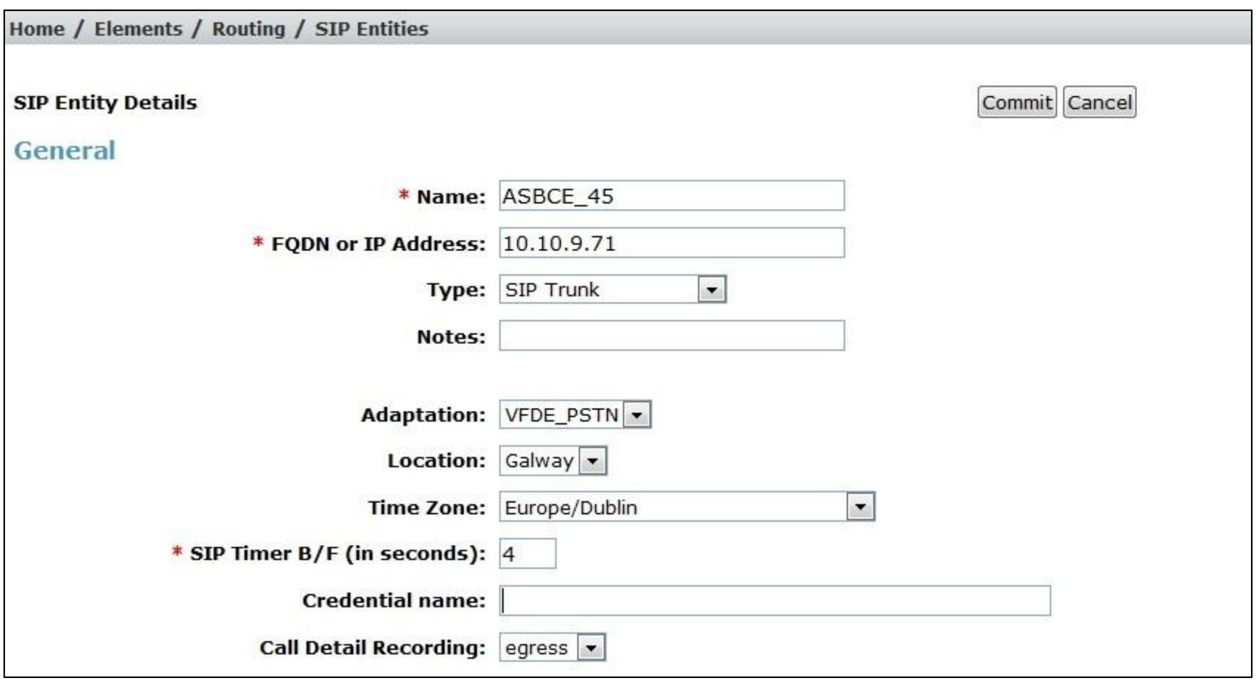


Note: The adaptation selected for Communication Manager modifies the called party number when it corresponds to the DDI number for the enterprise. This is an unusual case as the majority of calls are going out to the PSTN. It is useful to apply it to Communication Manager, however, as the message waiting indicator for SIP endpoints is sent to the address in the contact header, i.e. the DDI number of the extension. The adaptation ensures the message waiting indicator is sent correctly to the SIP endpoint.

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

6.5.3. Avaya Session Border Controller for Enterprise SIP Entity

The following screen shows the SIP Entity for the Avaya SBCE. The **FQDN or IP Address** field is set to the IP address of the Avaya SBCE private network interface (see **Figure 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.

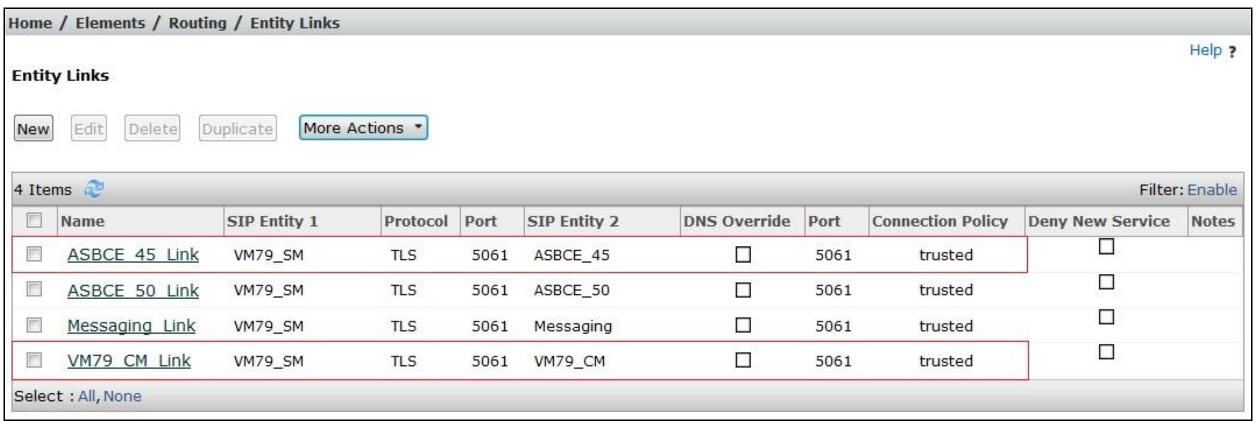


6.6. Administer Entity Links

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 the SIP entity for 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.
- 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 following screen shows the Entity Links used in this configuration.


Home / Elements / Routing / Entity Links

Help ?

Entity Links

New Edit Delete Duplicate More Actions

4 Items Filter: Enable

	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DNS Override	Port	Connection Policy	Deny New Service	Notes
☐	ASBCE_45_Link	VM79_SM	TLS	5061	ASBCE_45	☐	5061	trusted	☐	
☐	ASBCE_50_Link	VM79_SM	TLS	5061	ASBCE_50	☐	5061	trusted	☐	
☐	Messaging_Link	VM79_SM	TLS	5061	Messaging	☐	5061	trusted	☐	
☐	VM79_CM_Link	VM79_SM	TLS	5061	VM79_CM	☐	5061	trusted	☐	

Select : All, None

Note: The **Messaging_Link** Entity Link is used for the Avaya Aura ® Messaging system and is not described in this document.

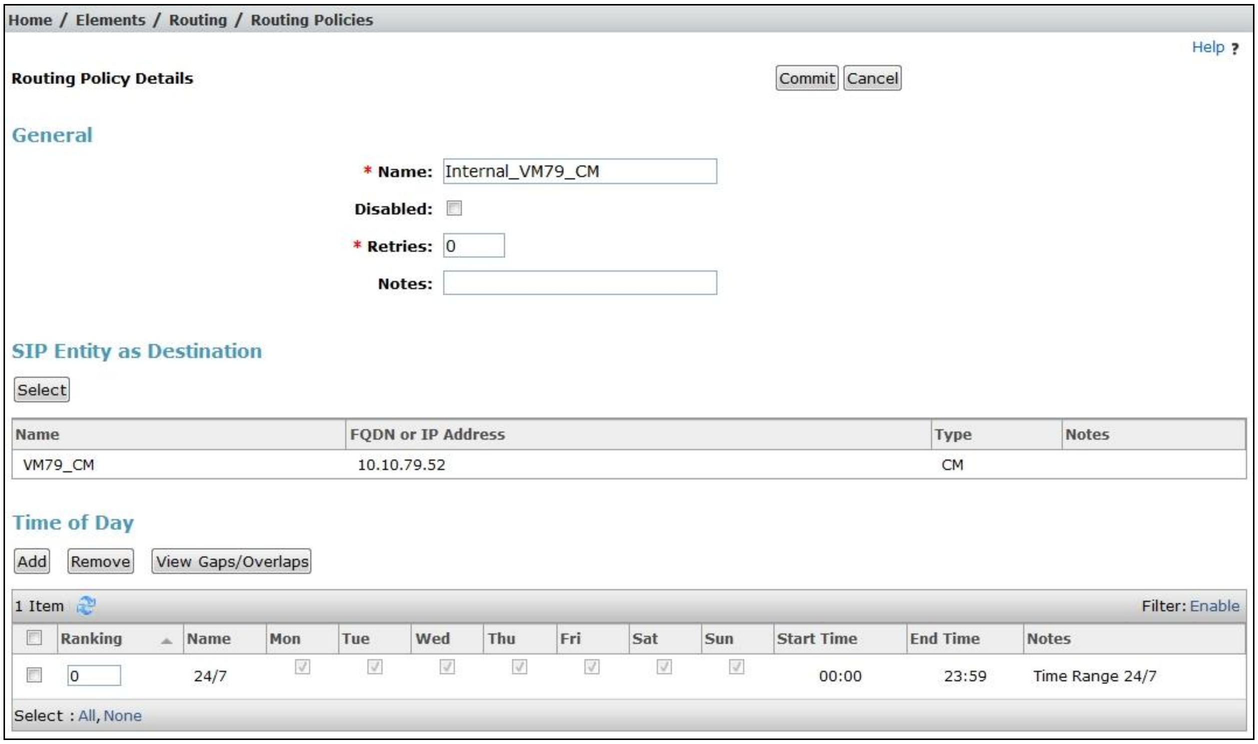
6.7. Administer Routing Policies

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).

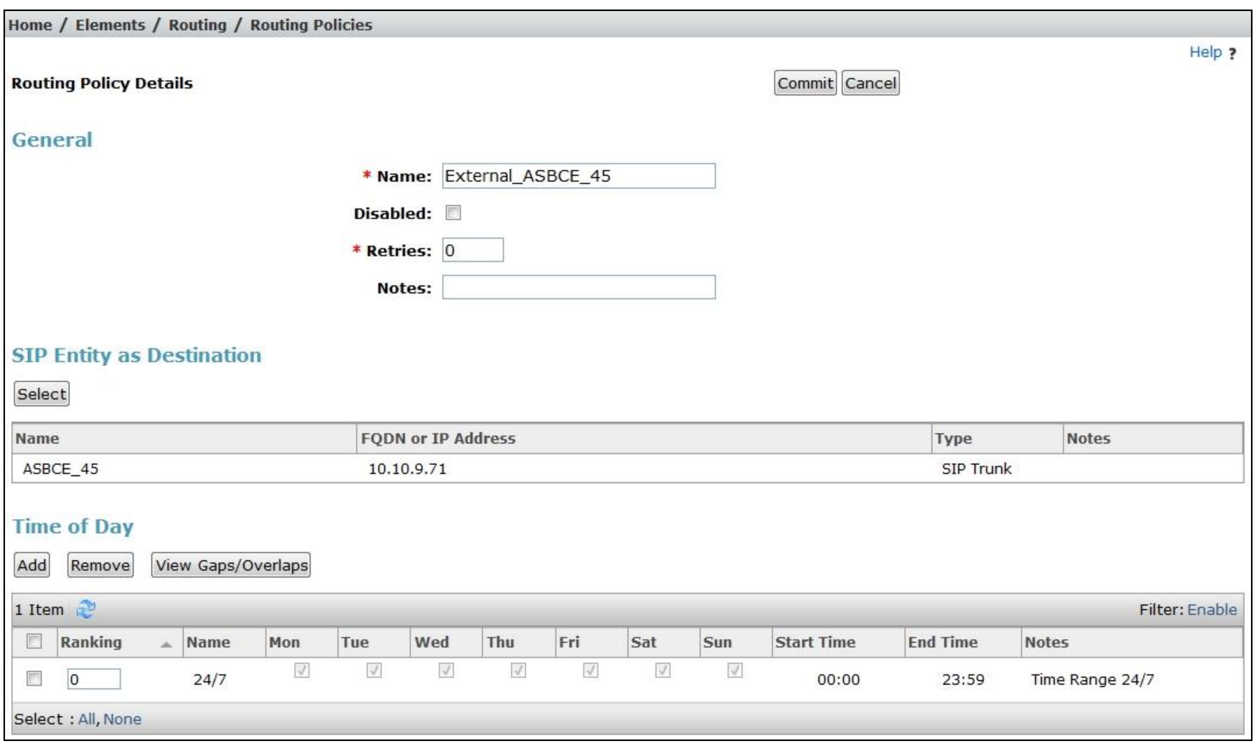
Under **General**:

- 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 Communication Manager.



The following screen shows the Routing Policy for the Avaya SBCE interface that will be routed to the PSTN via the Vodafone SIP Trunk.



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).

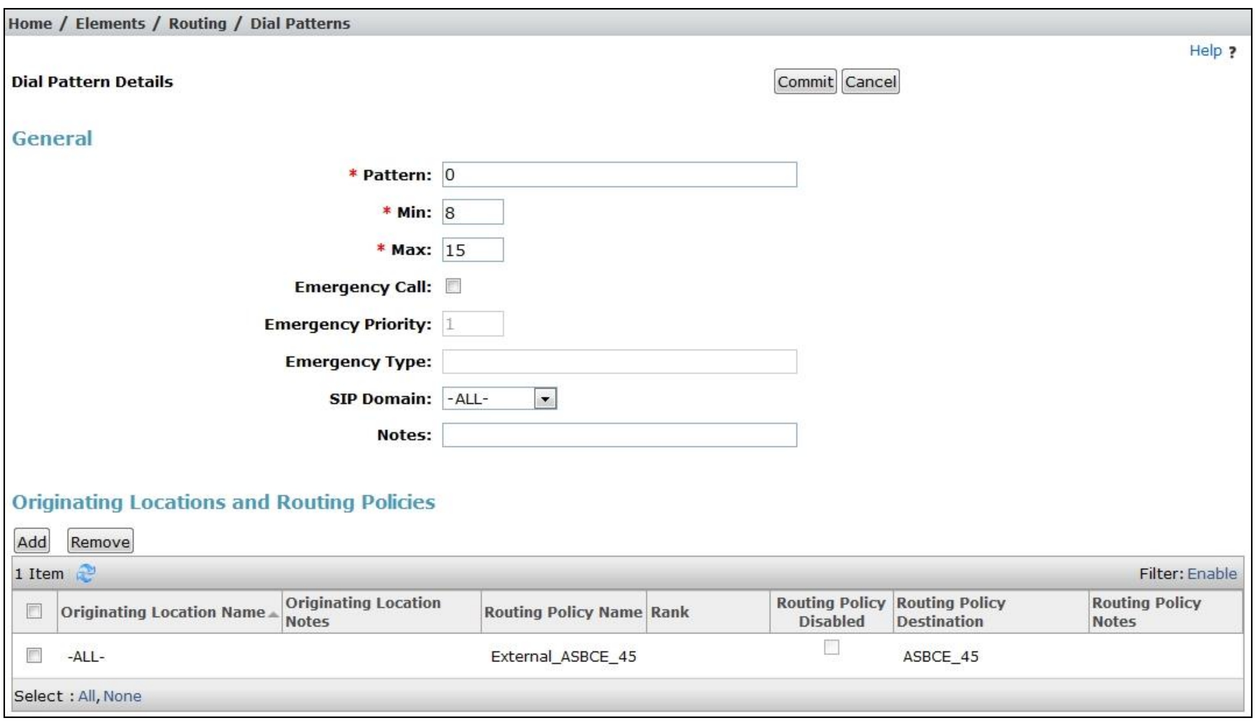
Under **General**:

- 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**.

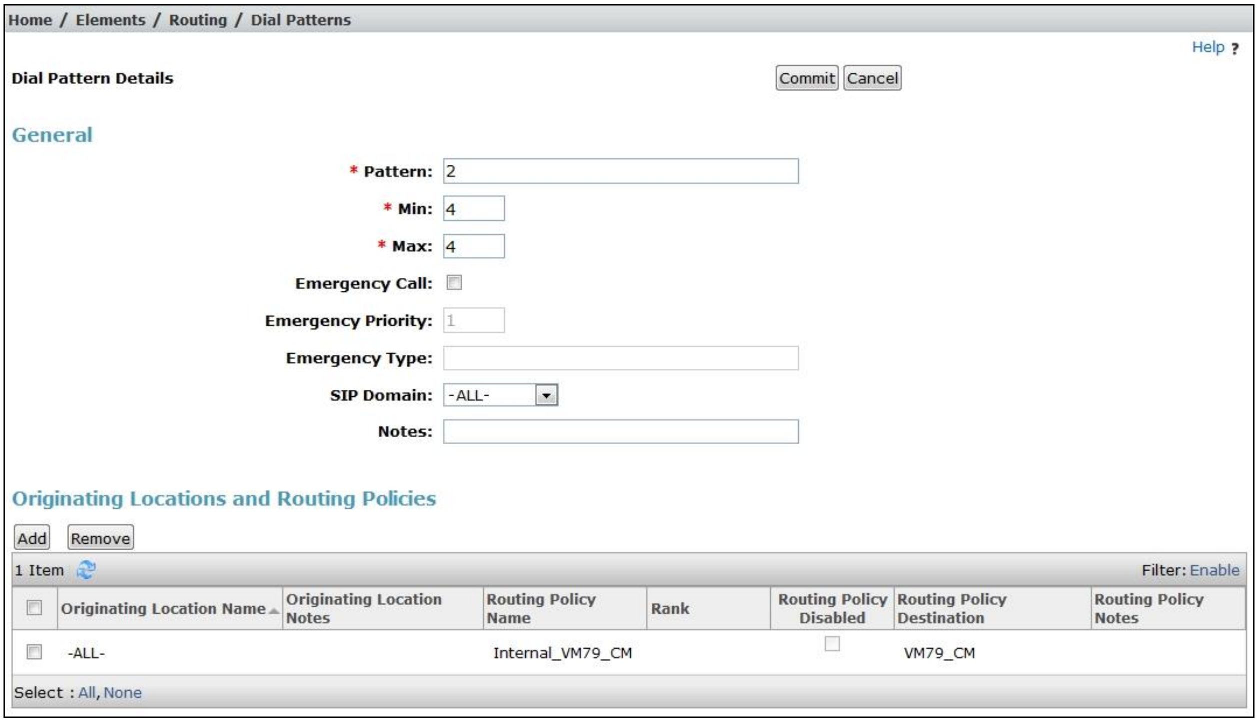
Under **Originating Locations and Routing Policies**:

- Click **Add** and enter details 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 out to the PSTN via the Vodafone SIP Trunk.



The following screen shows the test dial pattern configured for Communication Manager which identifies the extension number. All extension numbers used during testing were four digit numbers starting with 2.

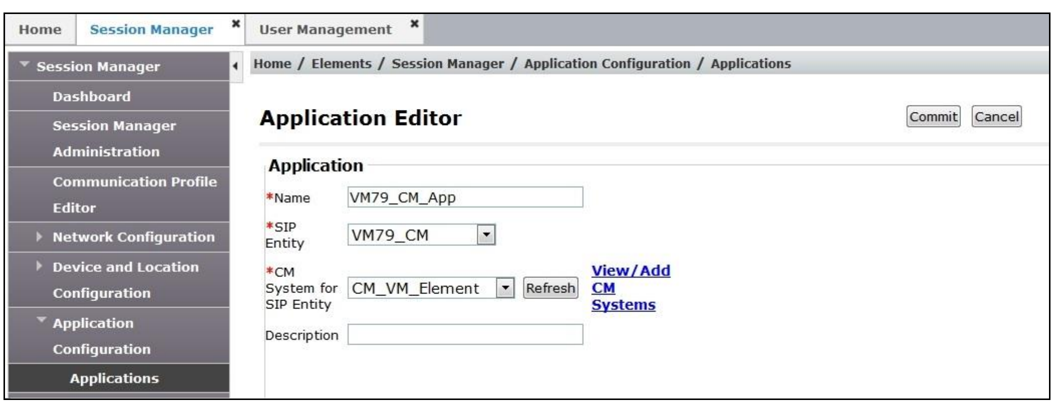


Note: The above configuration is used where the called party number has been converted to an extension number on Communication Manager using an adaptation. If an adaptation is not used, a dial pattern will be required for the incoming DDI number.

6.9. Administer Application for Avaya Aura® Communication Manager

The Application for SIP endpoints should already be defined, the following is shown for information. 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**.

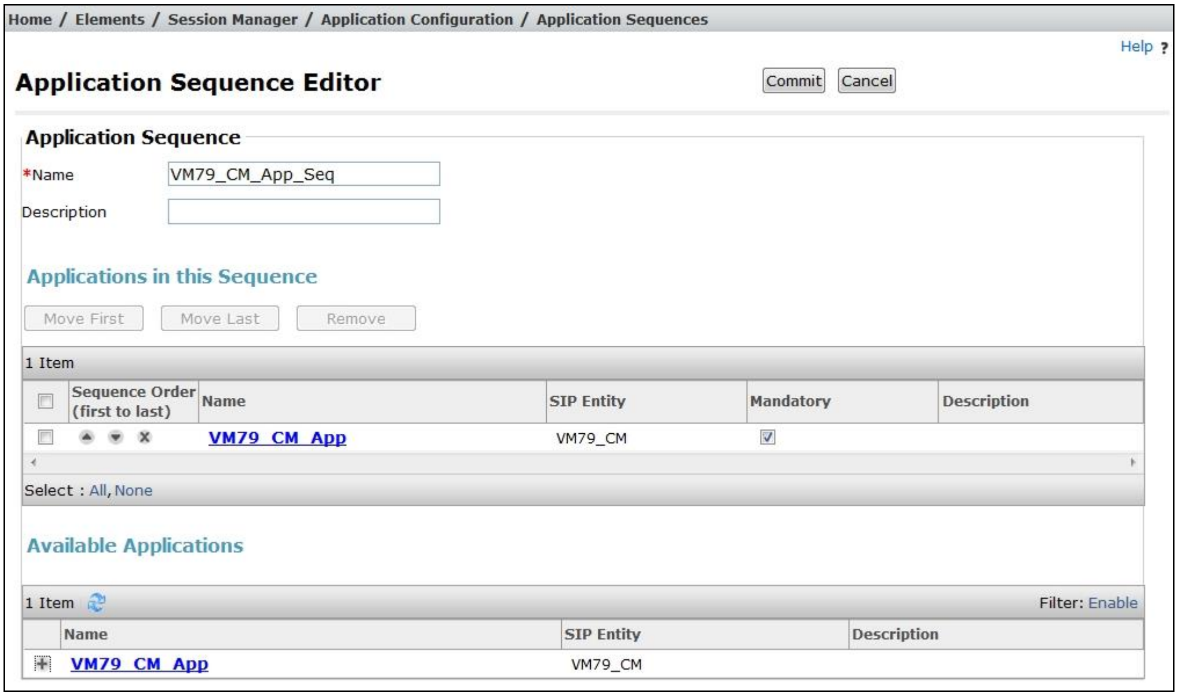
- In the **Name** field enter a name for the application.
- In the **SIP Entity** field select the SIP entity for the Communication Manager.
- In the **CM System for SIP Entity** field select the SIP entity for the Communication Manager and select **Commit** to save the configuration.



6.10. Administer Application Sequence for Avaya Aura® Communication Manager

The Application Sequence for SIP endpoints should already be defined, the following is shown for information. From the left panel navigate to **Session Manager → Application Configuration → Application Sequences** and click on **New**.

- 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**.

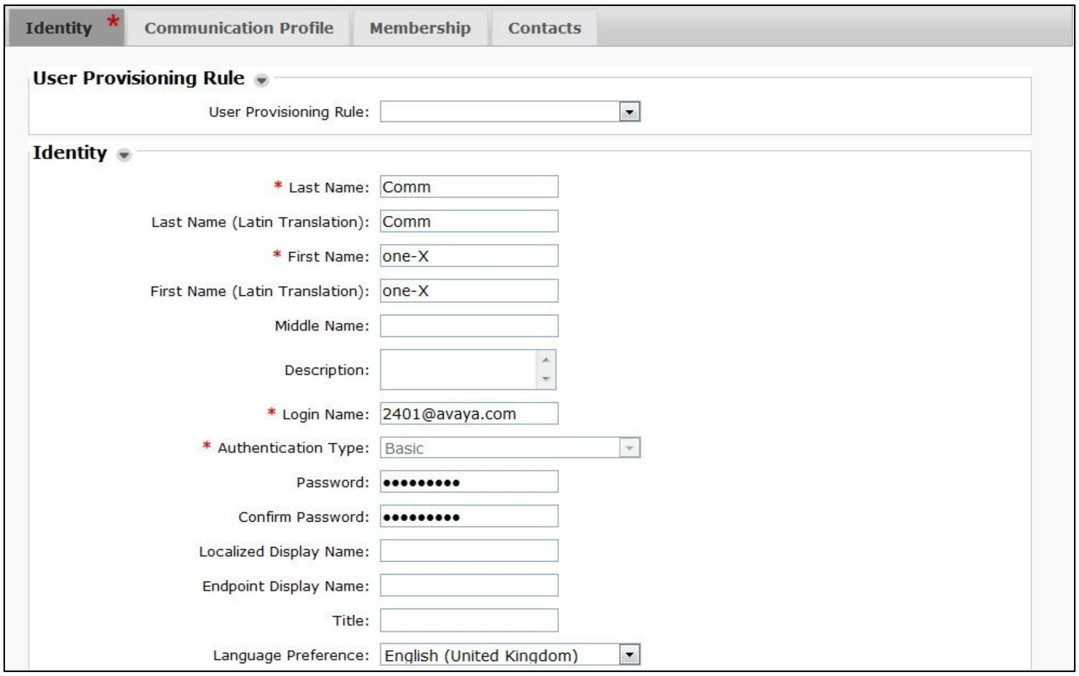


6.11. Administer SIP Extensions

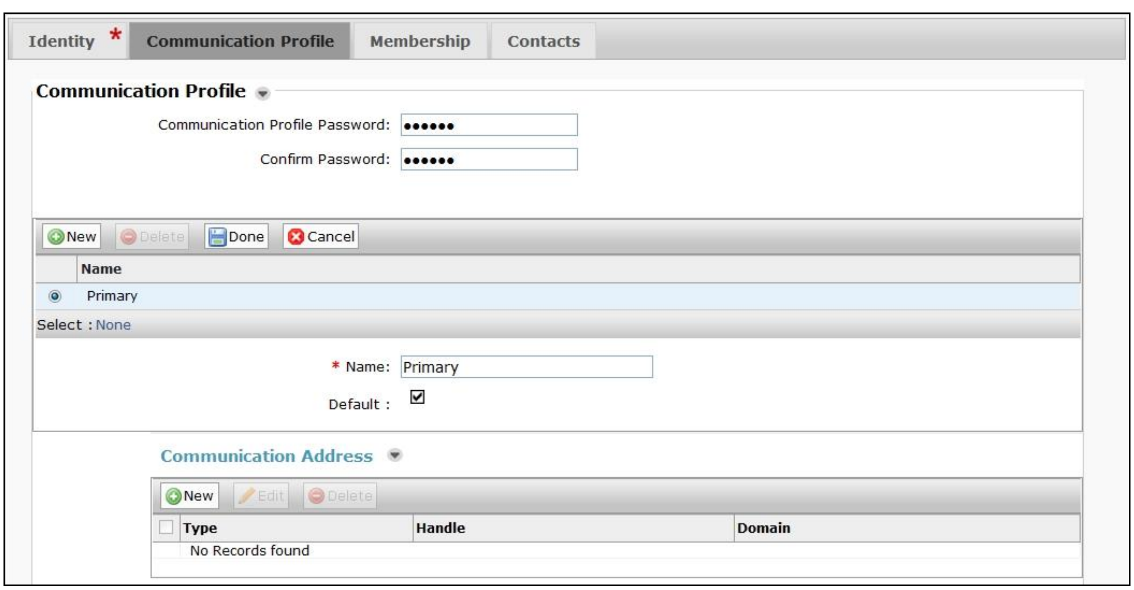
SIP extensions are registered with the Session Manager and use Communication Manager for their feature and configuration settings. From the **Home** tab select **User Management** from the menu. Then select **Manage Users** and click **New** (not shown).

On the **Identity** tab:

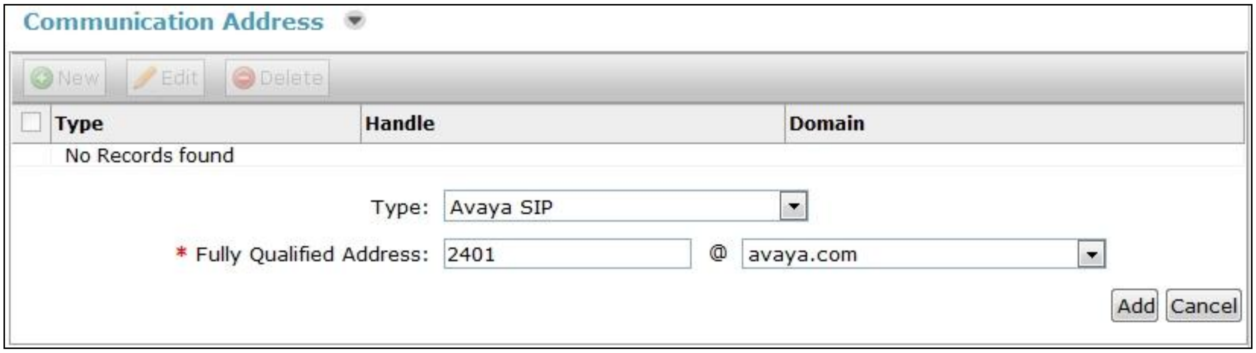
- Enter the user's name in the **Last Name** and **First Name** fields.
- In the **Login Name** field enter a unique system login name in the form of user@domain e.g. **2401@avaya.com** which is used to create the user's primary handle.
- The **Authentication Type** should be **Basic**.
- In the **Password/Confirm Password** fields enter an alphanumeric password.
- Set the **Language Preference** and **Time Zone** (not shown) as required.



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

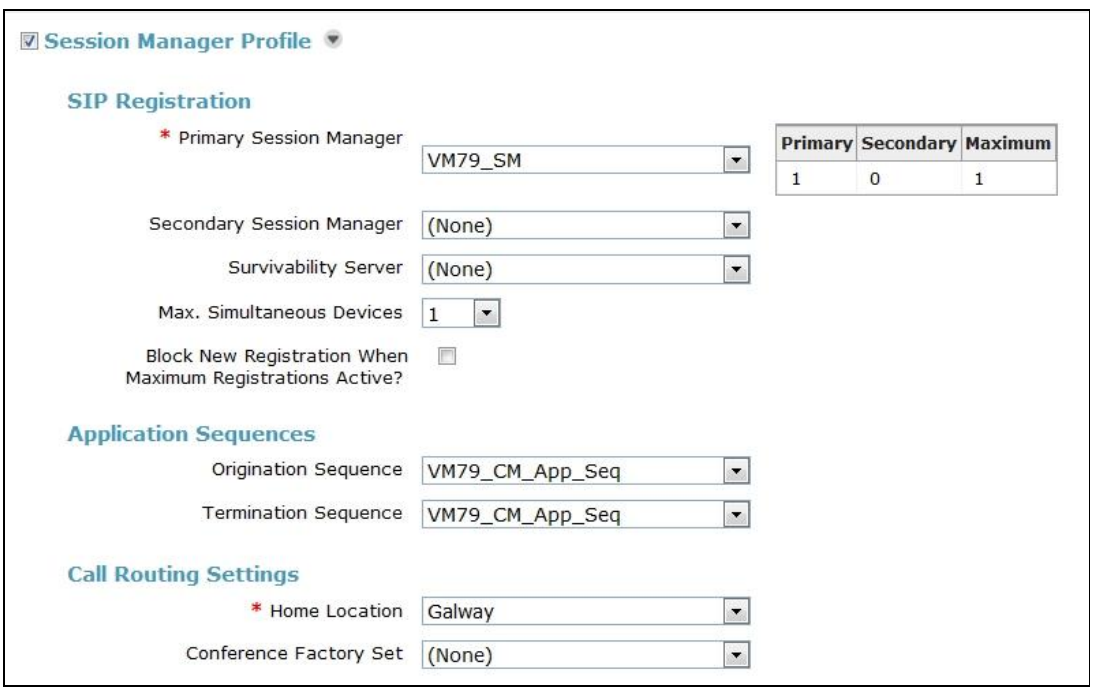


Expand the **Communication Address** section and click **New**. For the **Type** field select **Avaya SIP** from the drop-down menu. In the **Fully Qualified Address** field, enter an extension number and select the relevant domain from the drop-down menu. Click the **Add** button.



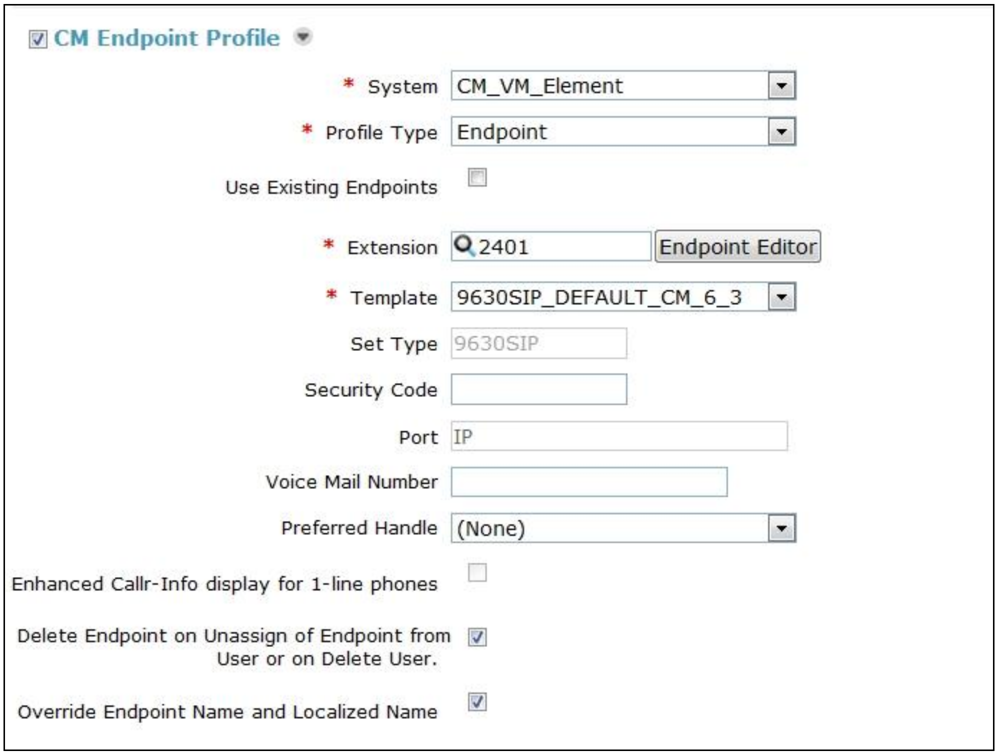
Expand the **Session Manager Profile** section.

- 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 the Communication Manager SIP Entity 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 the Communication Manager user configuration automatically.



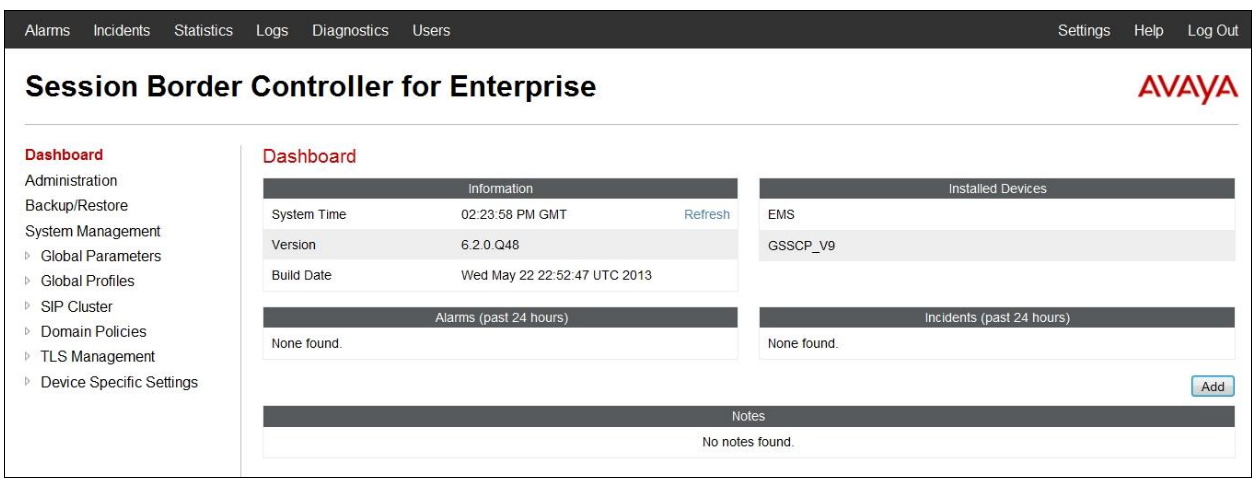
7. Configure Avaya Session Border Controller for Enterprise

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

7.1. Access Avaya Session Border Controller for Enterprise

Access the 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 username ucsec and the appropriate 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.

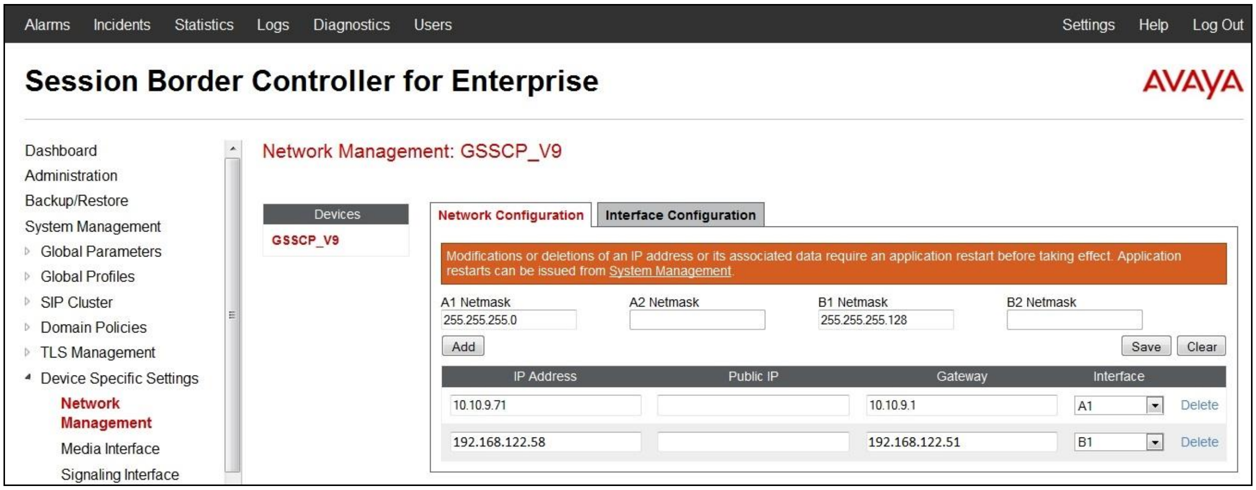


7.2. Define Network Information

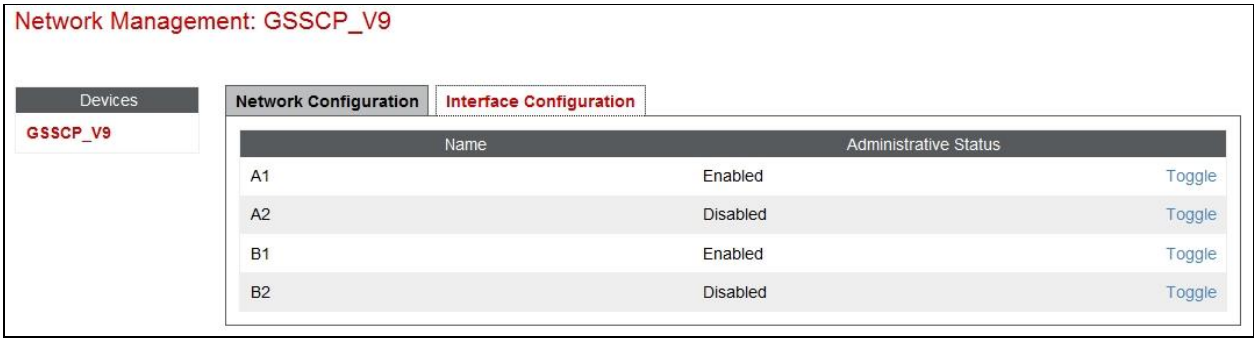
Network information is required on the Avaya SBCE to allocate IP addresses and masks to the interfaces. Note that only the **A1** and **B1** interfaces are used, typically the **A1** interface is used for the internal side and **B1** is used for external. Each side of the Avaya SBCE can have only one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

- Define the **Netmask** for interfaces A1 and B1.
- Define the internal IP address and Gateway and assign to interface **A1**.
- Click on **Add**.
- Define the external IP address and Gateway and assign to interface **B1**.
- Select **Save** to save the information.
- Click on **System Management** in the main menu.
- Select **Restart Application** indicated by an icon in the status bar (not shown).



Select the **Interface Configuration** tab and click on **Toggle State** to enable the interfaces.



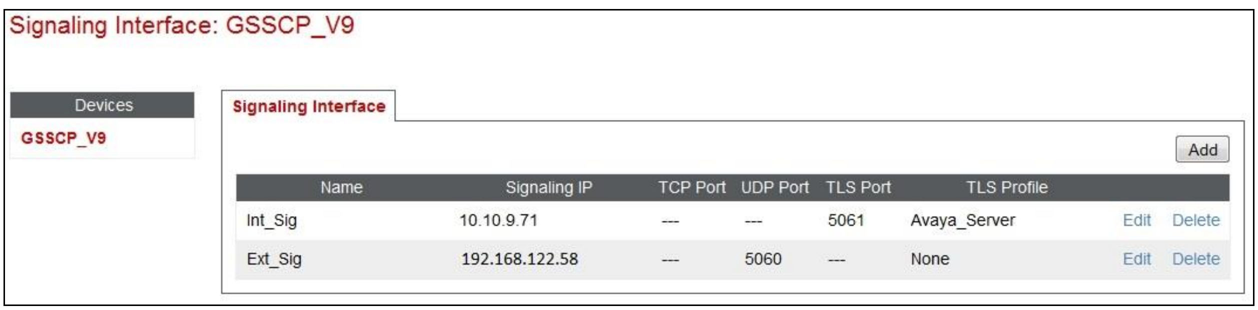
7.3. Define Interfaces

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 the Session Manager and the Avaya SBCE. This document shows the configuration for TLS, if another transport protocol is required, substitute it where TLS is specified.

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

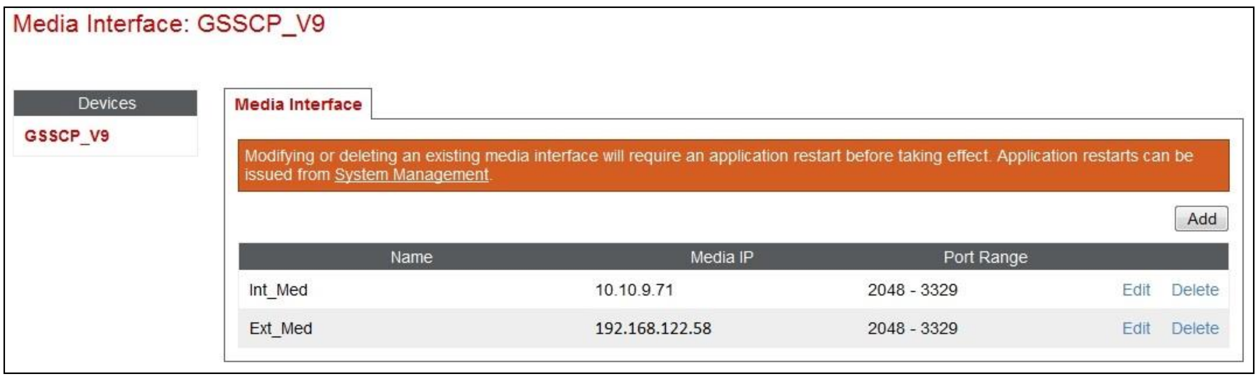
- Select **Add** and enter details of the internal signalling interface in the pop-up menu (not shown).
- In the **Name** field enter a descriptive name for the internal signalling interface.
- For **Signaling IP**, select an **internal** signalling interface IP address defined in **Section 7.2**
- Select **TLS** port number, **5061** is used for the Session Manager.
- When the TLS port number is defined, an additional field (not shown) becomes available for **TLS Profile**, select the predefined Avaya profile **Avaya_Server**.
- Select **Add** and enter details of the external signalling interface in the pop-up menu (not shown).
- In the **Name** field enter a descriptive name for the external signalling interface.
- For **Signaling IP**, select an **external** signalling interface IP address defined in **Section 7.2**.
- Select **UDP** port number, **5060** is used for the Vodafone SIP Trunk.



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

- 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.
- For **Media IP**, select an **internal** media interface IP address defined in **Section 7.2**.
- Select **RTP port** ranges for the media path with the enterprise end-points.
- Select **Add** and enter details of the external media interface in the pop-up menu.
- In the **Name** field enter a descriptive name for the external media interface.
- For **Media IP**, select an **external** media interface IP address defined in **Section 7.2**.
- Select **RTP port** ranges for the media path with Vodafone SIP Trunk.



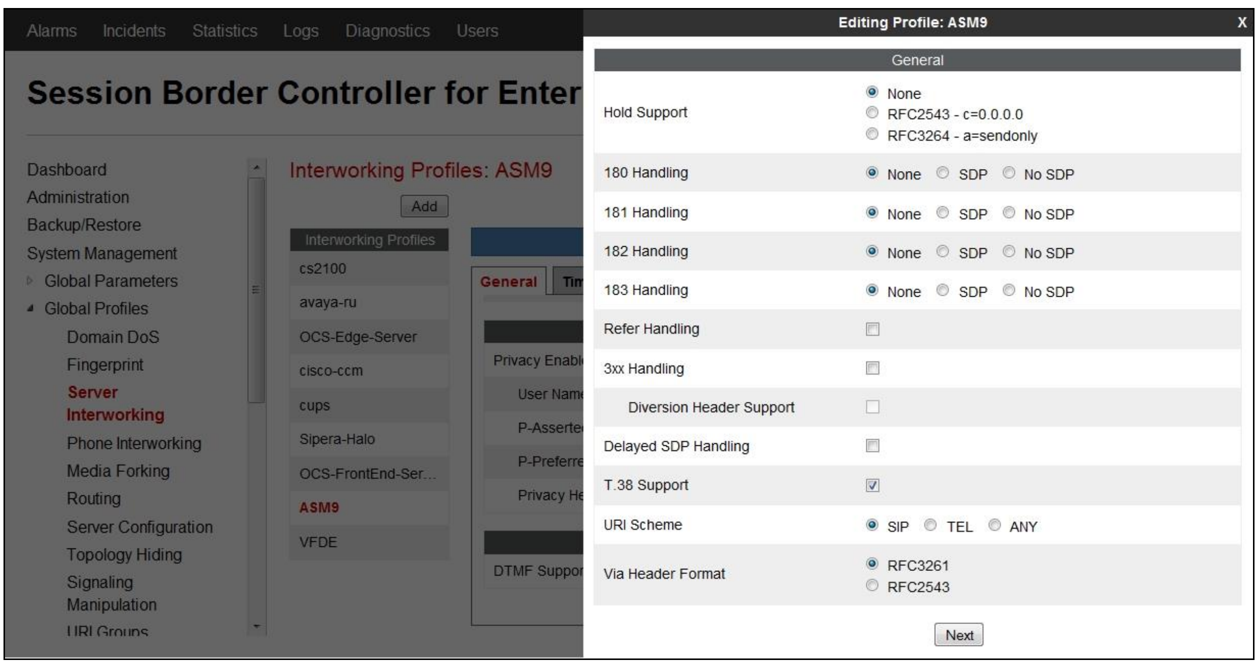
Note: During test the port ranges for the internal and external media interfaces were set to the default values used on Communication Manager.

7.4. Define Server Interworking

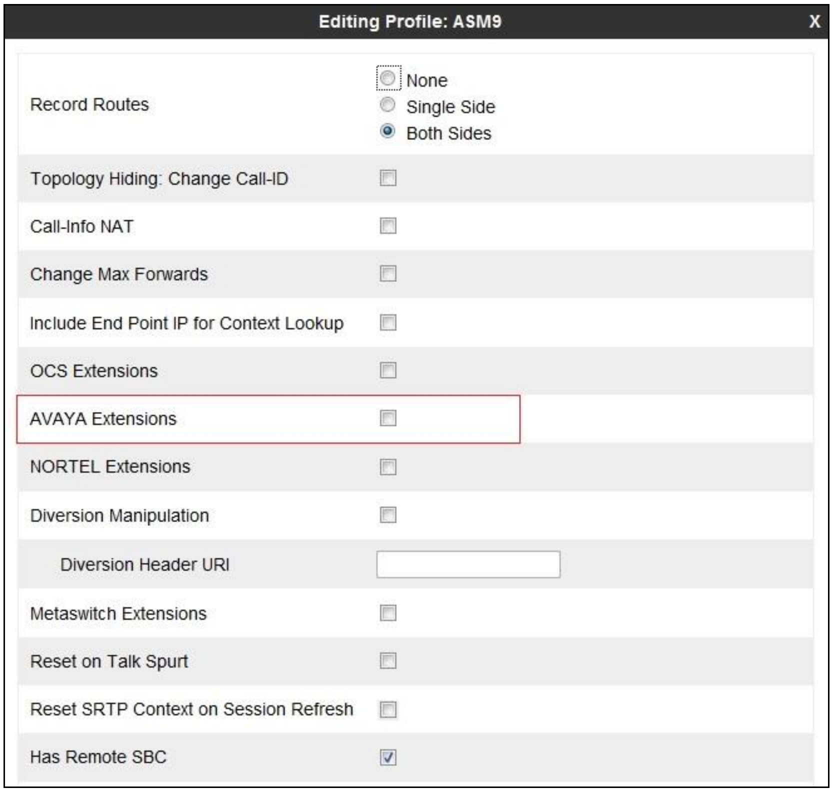
Server interworking is defined for each server connected to the Avaya SBCE. In this case, Vodafone SIP Trunk is connected as the Trunk Server and the Session Manager is connected as the Call Server. Configuration of interworking includes Hold support, T.38 fax support and SIP extensions.

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 the Session Manager, highlight the **avaya-ru** profile which is a factory setting appropriate for Avaya equipment and select **Clone Profile**. A pop-up menu is generated headed **Clone Profile** (not shown).

- In the **Clone Name** field enter a descriptive name for the Session Manager and click **Finish** – in test **ASM9** was used.
- In the **General** tab (not shown) Select **Edit** and enter details in the pop-up menu.
- Check the **T.38** box then click **Next** and **Finish** (not shown).



- In the **Advanced** tab (not shown) Select **Edit** and enter details in the pop-up menu
- Uncheck the **AVAYA Extensions** box



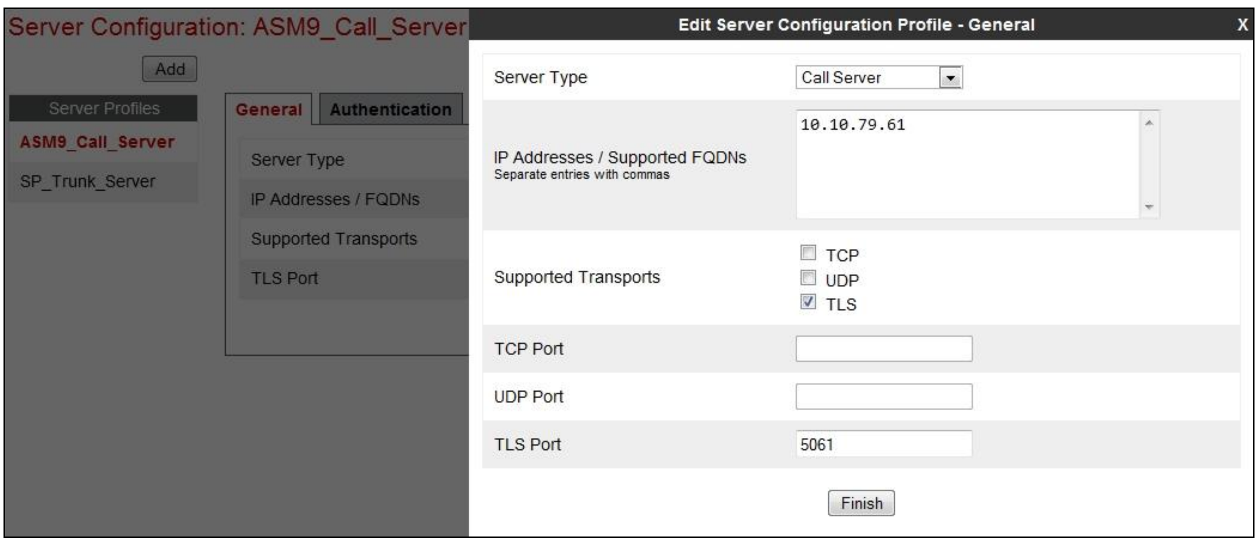
To define Server Interworking for Vodafone SIP Trunk, highlight the previously defined profile for the Session Manager and select **Clone Profile**. A pop-up menu is generated headed **Clone Profile** (not shown).

- In the **Clone Name** field enter a descriptive name for server interworking profile for Vodafone SIP Trunk and click **Finish** – in test **VFDE** was used.
- Select **Edit** and enter details in the pop-up menu.
- Check the **T.38** box.
- Select **Next** three times and **Finish**.

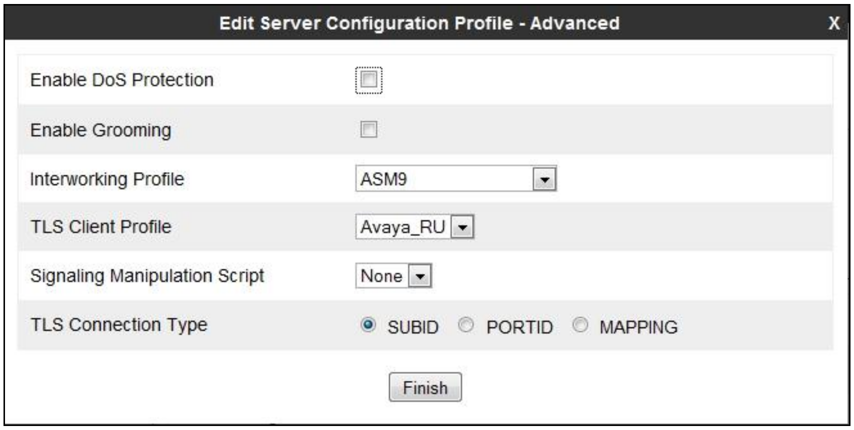
7.5. Define Servers

A server definition is required for each server connected to the Avaya SBCE. In this case, Vodafone SIP Trunk is connected as the Trunk Server and the Session Manager is connected as the Call Server. To define the Session Manager, navigate to **Global Profiles → Server Configuration** in the main menu on the left hand side. Click on **Add** and enter details in the pop-up menu.

- In the **Profile Name** field enter a descriptive name for the Session Manager and click **Next** (not shown).
- In the **Server Type** drop down menu, select **Call Server**.
- In the **IP Addresses / Supported FQDNs** box, type the Session Manager SIP interface address which is the same as that defined for the Session Manager SIP Entity in Section 6.5.1.
- If TLS is to be used for the signalling transport between the Session Manager and the Avaya SBCE, check **TLS** in **Supported Transports**.
- Define the **TLS** port for SIP signalling, **5061** is used for the Session Manager and click **Finish**.

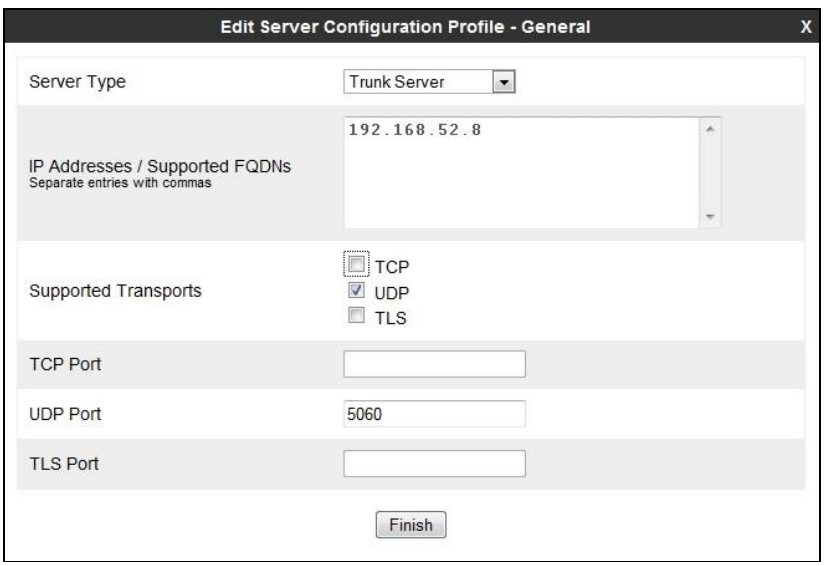


- Select the **Advanced** tab (not shown).
- In the **Interworking Profile** drop down menu, select the **Interworking Profile** for the Session Manager defined in **Section 7.4**.
- If TLS is to be used between the Session Manager and the Avaya SBCE, select the predefined Avaya TLS client in the **TLS Client Profile drop** down menu. The predefined TLS client on the Avaya SBC used in test was **Avaya_RU**, on other systems this may be **AvayaSBCClient**.
- Click **Finish**.

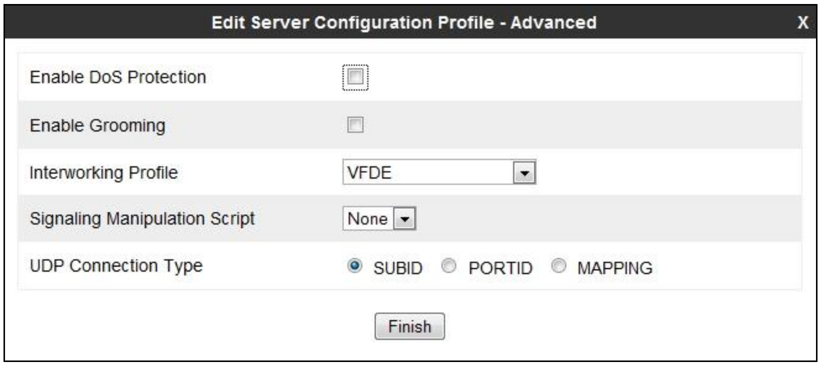


To define Vodafone SIP Trunk as a Trunk Server, navigate to **Global Profiles → Server Configuration** in the main menu on the left hand side. Click on **Add** and enter details in the pop-up menu.

- In the **Profile Name** field enter a descriptive name for Vodafone SIP Trunk and click **Next** (not shown).
- In the **Server Type** drop down menu, select **Trunk Server**.
- In the **IP Addresses / Supported FQDNs** box, type the IP address of Vodafone SIP Trunk.
- Check **UDP** in **Supported Transports**.
- Define the **UDP** port for SIP signaling, **5060** is used for Vodafone.
- Click **Finish**.



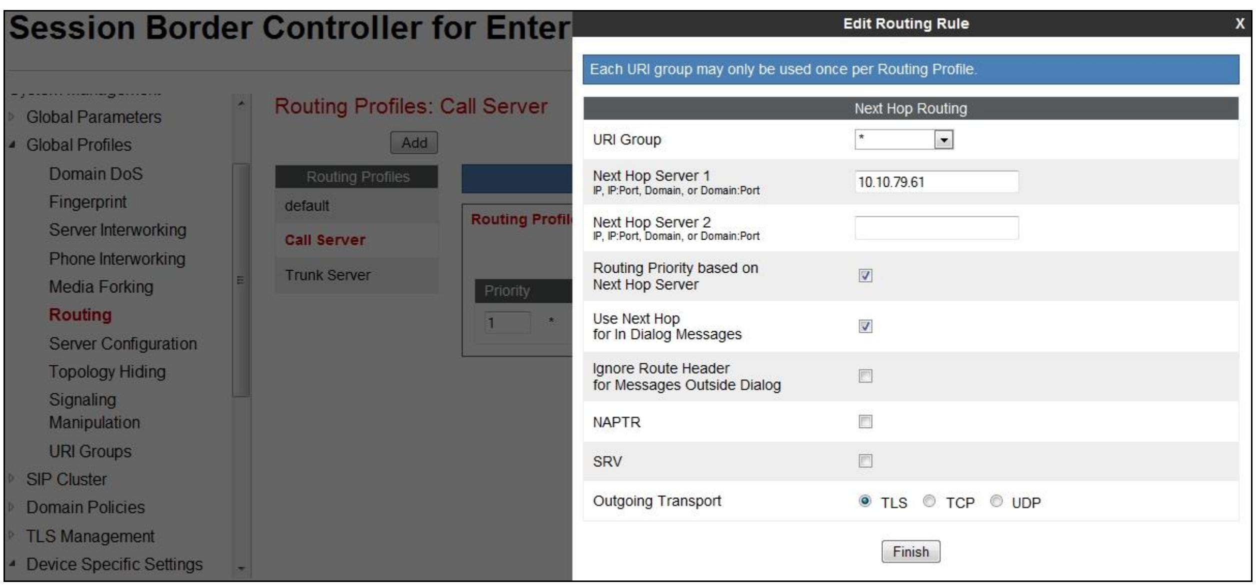
- Select the **Advanced** tab (not shown).
- Select the **Interworking Profile** for the Vodafone SIP Trunk defined in **Section 7.4** from the drop down menu.



7.6. Define Routing

Routing information is required for routing to the Session Manager on the internal side and Vodafone SIP Trunk on the external side. The IP addresses and ports defined here will be used as the destination addresses for signalling. If no port is specified in the **Next Hop IP Address**, default 5060 is used for TCP and UDP, and 5061 for TLS. To define routing to the Session Manager, navigate to **Global Profiles → Routing** in the main menu on the left hand side. Click on **Add** and enter details in the **Routing Profile** pop-up menu.

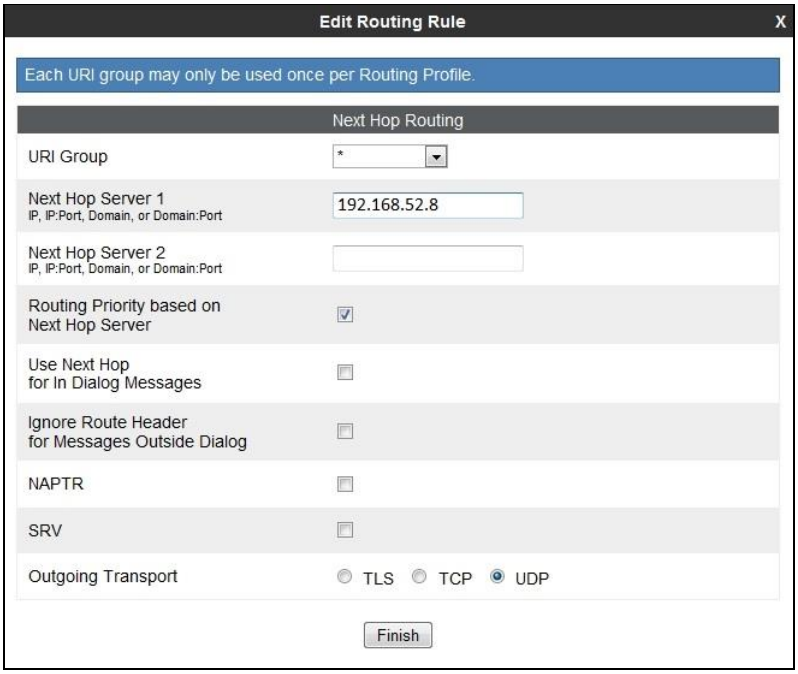
- In the **Profile Name** field enter a descriptive name for the Session Manager, in this case **Call Server**, and click **Next**.
- Enter the Session Manager SIP interface address and port in the **Next Hop Server 1** field.
- Select **TLS** for the **Outgoing Transport**.
- Click **Finish**.



To define routing to Vodafone SIP Trunk, navigate to **Global Profiles → Routing** in the main menu on the left hand side. Click on **Add** and enter details in the **Routing Profile** pop-up menu.

- In the **Profile Name** field enter a descriptive name for Vodafone SIP Trunk, in this case a generic name of **Trunk Server** was used, and click **Next**.

- Enter the Vodafone SIP Trunk IP address and port in the **Next Hop Server 1** field.
- Select **UDP** for the **Outgoing Transport**.
- Click **Finish**.

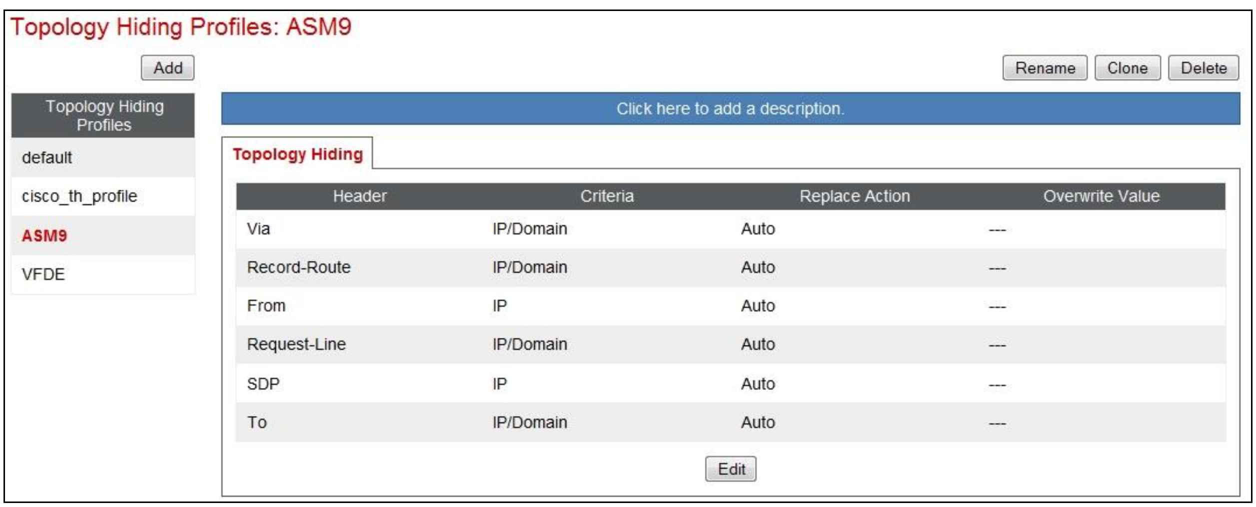


7.7. Topology Hiding

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 destination headers and local IP for source headers. Topology hiding has the advantage of presenting single Via and Record-Route headers externally where multiple headers may be received from the enterprise, particularly from the Session Manager. In some cases where Topology Hiding can't be applied, in particular the Contact header, IP addresses are translated to the Avaya SBCE external addresses using NAT.

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

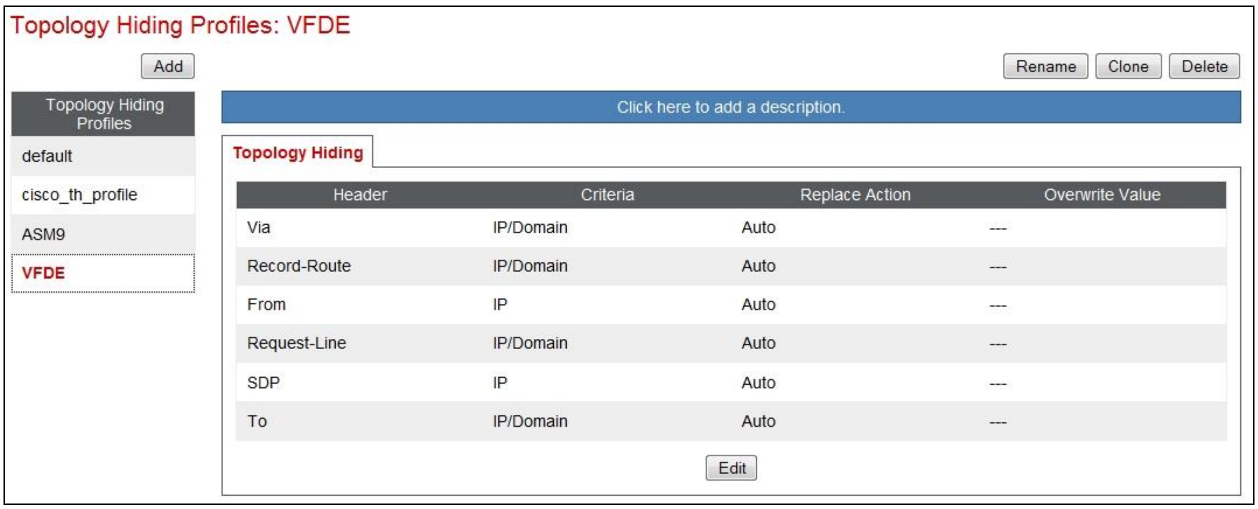
- In the **Profile Name** field enter a descriptive name for the Session Manager and click **Next**.
- If the **Request-Line**, **Record-Route, Via** and **To** Headers aren't shown, click on **Add Header** and **s**elect from the **Header** drop down menu.
- For each of the above headers, leave the **Replace Action** at the default value of **Auto**.
- If the **From** and **SDP** Headers aren't shown, click on **Add Header** and **s**elect from the **Header** drop down menu.
- For each of the above headers, select **IP** from the **Criteria** drop down menu (important for the **From** header so that the "anonymous.invalid" domain name for restricted CLI is not overwritten).
- For each of the headers leave the **Replace Action** at the default value of **Auto**.



Note: The use of **Auto** results in an IP address being inserted in the host portion of the header URI as opposed to a domain name. If a domain name is required, the action **Overwrite** must be used where appropriate, and the required domain names entered in the **Overwrite Value** field. Different domain names can be used for the enterprise and Vodafone SIP Trunk.

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

- In the **Profile Name** field enter a descriptive name for Vodafone SIP Trunk and click **Next**.
- If the **Request-Line**, **Record-Route**, **Via** and **To** Headers aren't shown, click on **Add Header** and select from the **Header** drop down menu.
- For each of the above headers, leave the **Replace Action** at the default value of **Auto**.
- If the **From** and **SDP** Headers aren't shown, click on **Add Header** and select from the **Header** drop down menu.
- For each of the above headers, select **IP** from the **Criteria** drop down menu (important for the **From** header so that the "anonymous.invalid" domain name for restricted CLI is not overwritten).



7.8. End Point Policy Groups

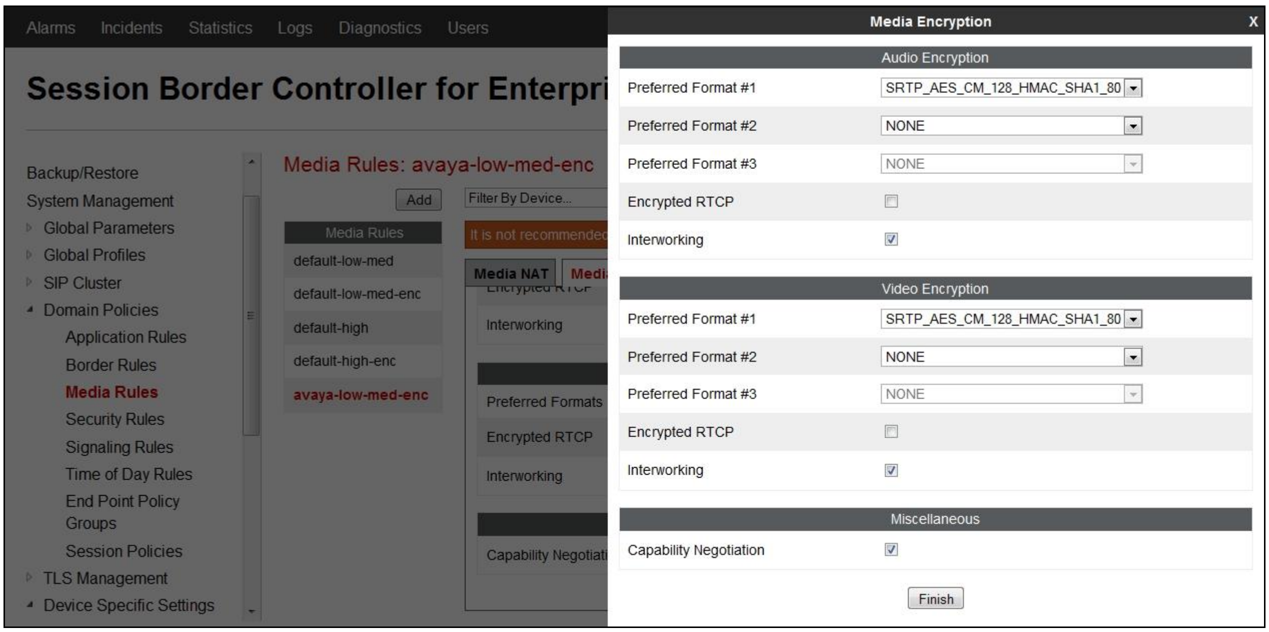
End Point Policy Groups are used to bring together a number of different rules for use in a server flow described in **Section 7.10**. The Vodafone SIP Trunk was tested with SRTP in the enterprise, and a Media Rule was required for conversion between SRTP and RTP.

7.8.1. Media Rules

Media rules are a mechanism on the Avaya SBCE to handle any unusual media handling scenarios that may be encountered for a particular Service Provider. In the case of Vodafone SIP Trunk, this was the conversion between SRTP used within the enterprise and RTP used on the SIP Trunk.

To define the media rule, navigate to **Domain Policies →Media Rules** in the main menu on the left hand side. Click on **Add** and enter details in the Media Rule pop-up box (not shown). In the **Rule Name** field enter a descriptive name for the Vodafone SIP Trunk media rule and click **Next** and **Next** again, then **Finish**

- Click on the **Media Encryption** tab (not shown) and then click on **Edit**.
- Select the **Preferred Format #1** from the drop down menu, in test **SRTP_AES_CM_128_HMAC_SHA1_80** was used.
- Video is not currently offered as part of the solution, but if required select the preferred format for video.
- Ensure **Interworking** is checked so that the conversion to RTP can take place.
- Leave **Capability Negotiation** unchecked as it is not required in this solution.
- Click **Finish**.



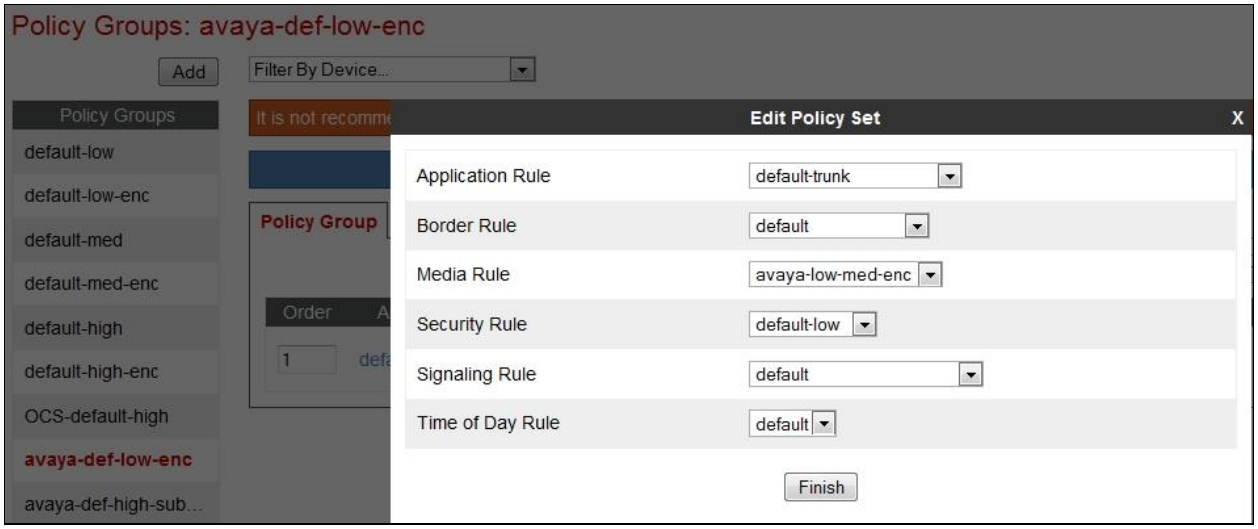
Note: Existing media rule **avaya-low-med-enc** was used for testing. To verify an existing rule, click on the **Media Encryption** tab, then check that the settings are consistent with those described above.

7.8.2. End Point Policy Group

An End Point Policy Group is required to implement the media rule. To define one for use in the Session Manager server flow, navigate to **Domain Policies → End Point Policy Groups** in the main menu on the left hand side. Click on **Add** and enter details in the Policy Group pop-up box (not shown).

- In the **Group Name** field enter a descriptive name for the Session Manager Policy Group, in this case **avaya-def-low-enc**, and click **Next.**
- In the **Application Rule** field, select **default-trunk**.
- Leave the **Application Rule** and **Border Rule** at their default values.

- Select the **Media Rule** created in the previous section in the drop down menu.
- Leave the **Security Rule**, **Signalling Rule** and **Time of Day Rule** at their default values.



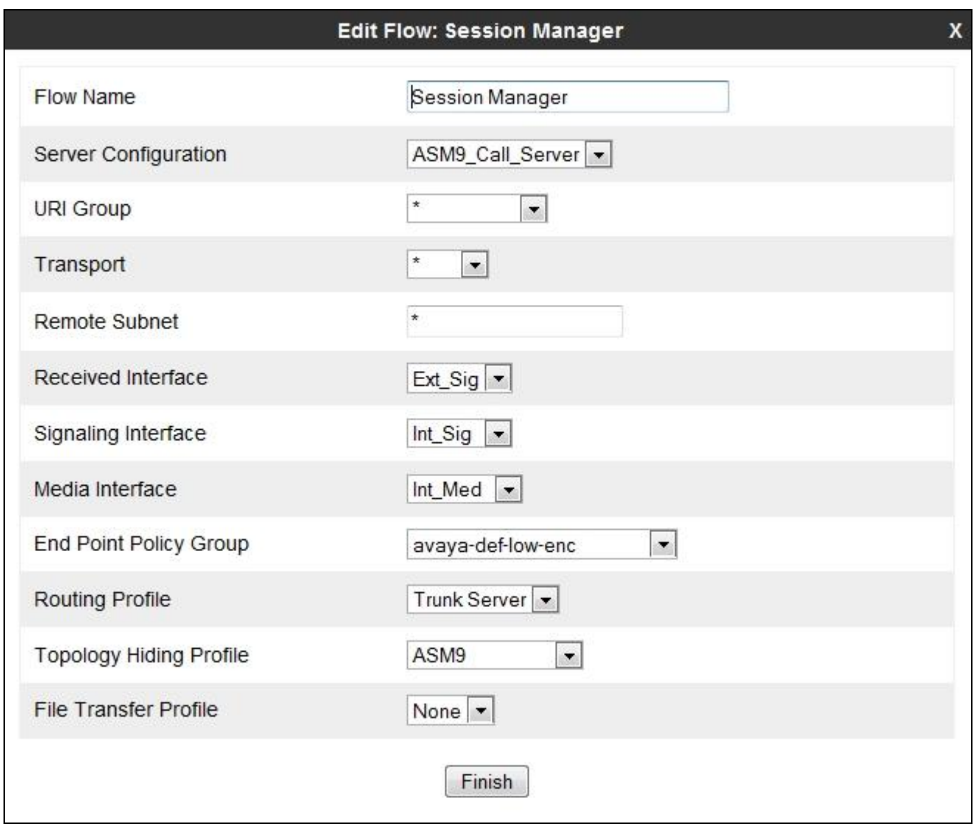
Note: Existing end point policy group **avaya-def-low-enc** was used for testing. To verify an existing end point policy group, check that the settings are consistent with those described above.

7.9. Server Flows

Server Flows combine the previously defined profiles into two End Point Server Flows, one for the Session Manager and another for the Vodafone SIP Trunk. This configuration ties all the previously entered information together so that calls can be routed from the Session Manager to Vodafone SIP Trunk and vice versa.

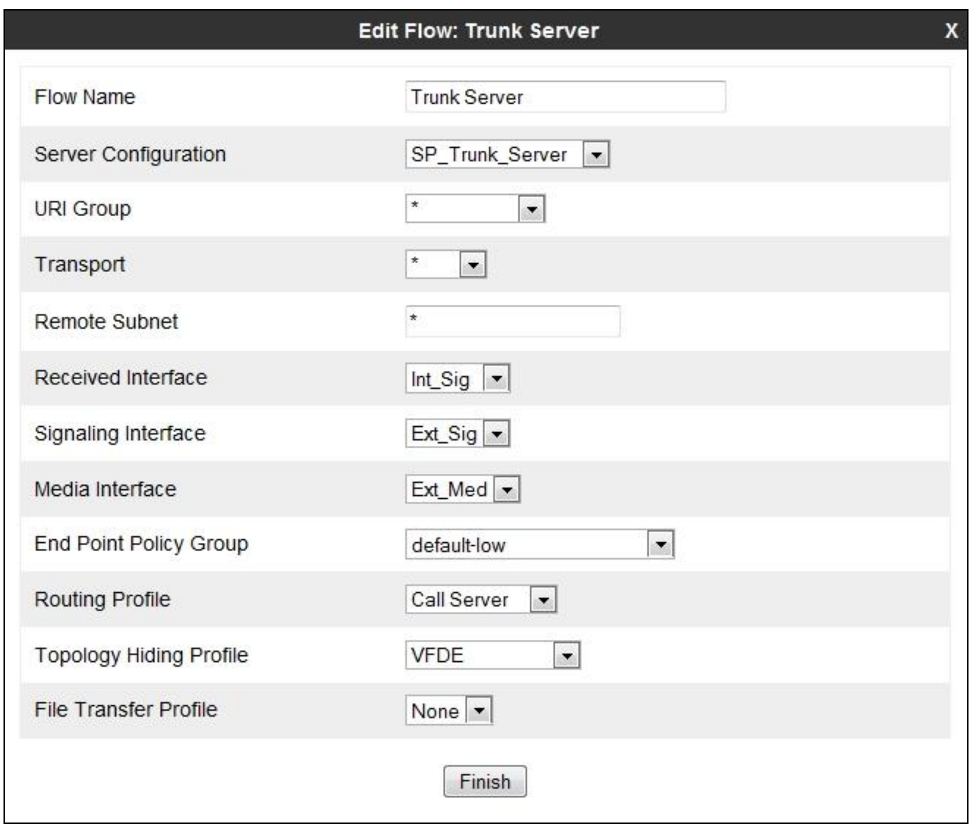
To define a Server Flow for the Session Manager, navigate to **Device Specific Settings → End Point Flows**.

- Click on the **Server Flows** tab (not shown).
- Select **Add Flow** and enter details in the pop-up menu (not shown).
- In the **Name** field (not shown) enter a descriptive name for the server flow for the Session Manager, in this case **ASM Call Server** was used.
- In the **Server Configuration** drop-down menu, select the server configuration for the Session Manager defined in **Section 7.5**.
- In the **Received Interface** drop-down menu, select the external SIP signalling interface defined in **Section 7.3**. This is the interface that signalling bound for the Session Manager is received on.
- In the **Signaling Interface** drop-down menu, select the internal SIP signalling interface defined in **Section 7.3**. This is the interface that signalling bound for the Session Manager is sent on.
- In the **Media Interface** drop-down menu, select the internal media interface defined in **Section 7.3**. This is the interface that media bound for the Session Manager is sent on.
- In the **Routing Profile** drop-down menu, select the routing profile of Vodafone SIP Trunk defined in **Section 7.7**.
- In the **End Point Policy Group** drop down menu, select the policy group defined in **Section 7.9**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of the Session Manager defined in **Section 7.8** and click **Finish**.

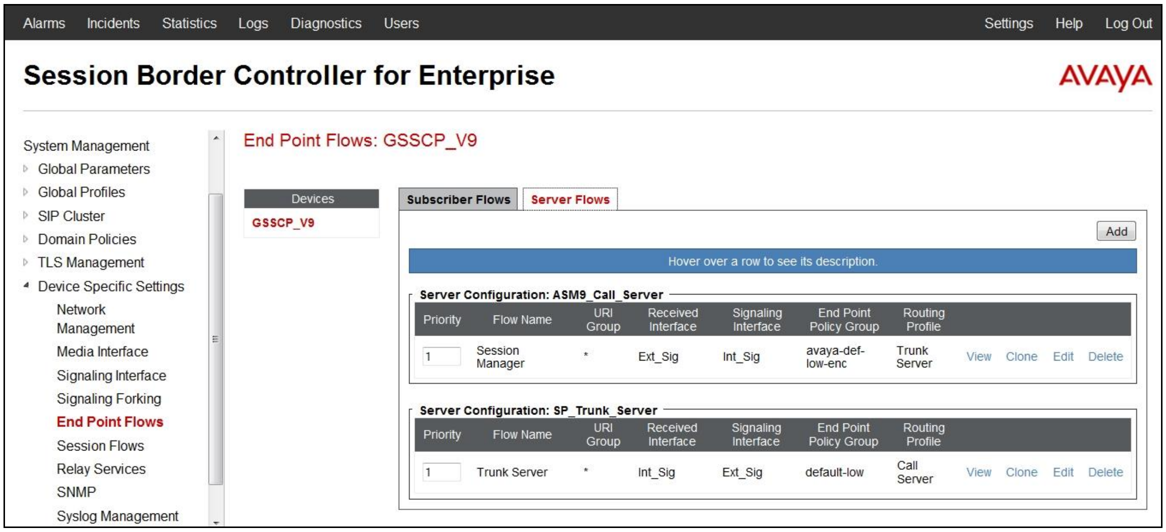


To define a Server Flow for Vodafone SIP Trunk, navigate to **Device Specific Settings → End Point Flows**.

- Click on the **Server Flows** tab (not shown).
- Select **Add Flow** and enter details in the pop-up menu (not shown).
- In the **Flow Name** field (not shown) enter a descriptive name for the server flow for Vodafone SIP Trunk, in this case a generic name of **Trunk Server** was used.
- In the **Server Configuration** drop-down menu, select the server configuration for the Trunk Server defined in **Section 7.5**.
- In the **Received Interface** drop-down menu, select the internal SIP signalling interface defined in **Section 7.3**. This is the interface that signalling bound for Vodafone SIP Trunk is received on.
- In the **Signaling Interface** drop-down menu, select the external SIP signalling interface defined in **Section 7.3**. This is the interface that signalling bound for Vodafone SIP Trunk is sent on.
- In the **Media Interface** drop-down menu, select the external media interface defined in **Section 7.3**. This is the interface that media bound for Vodafone SIP Trunk is sent on.
- In the **Routing Profile** drop-down menu, select the routing profile of the Session Manager defined in **Section 7.7**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of the Vodafone SIP Trunk defined in **Section 7.8** and click **Finish**.



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



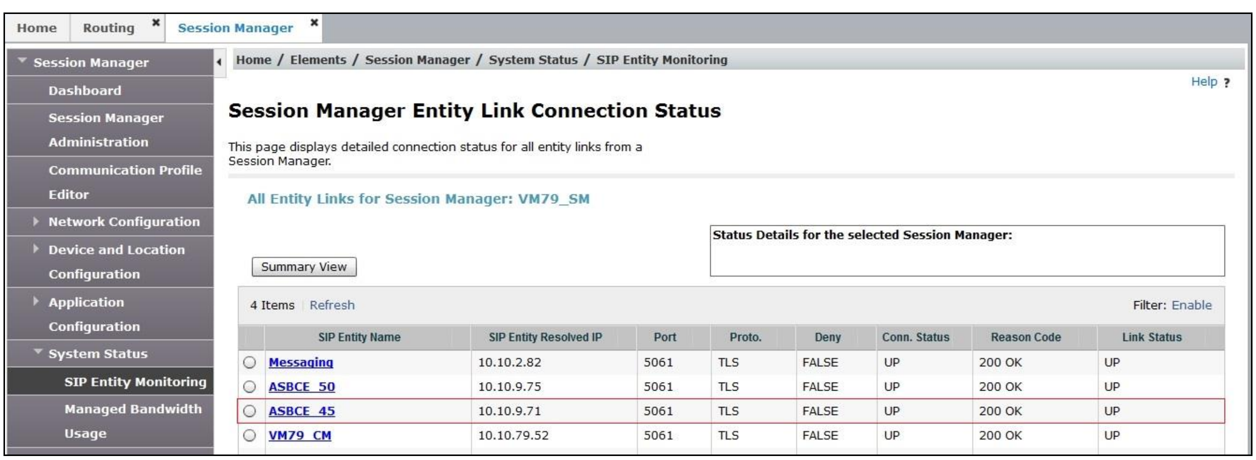
8. Configure Vodafone Germany SIP Trunk Equipment

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



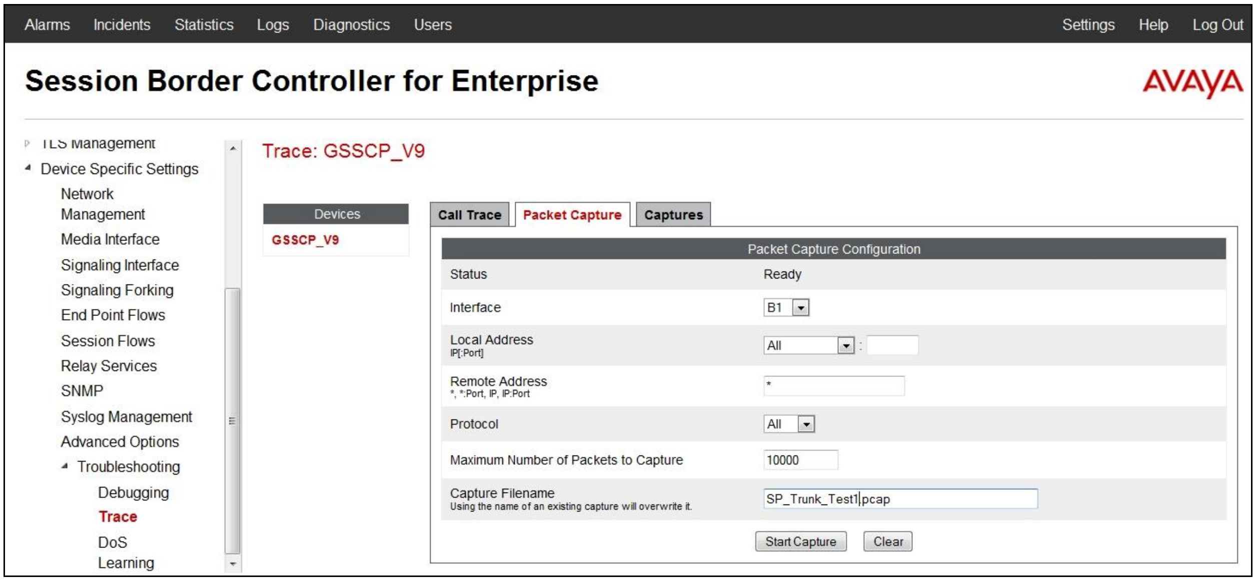
2. From the Communication Manager SAT interface run the command **status trunk n** where **n** is a previously configured SIP trunk. Observe if all channels on the trunk group display **in-service/idle**.

```
status trunk 1

                              TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0001/001 T00001   in-service/idle    no
0001/002 T00002   in-service/idle    no
0001/003 T00003   in-service/idle    no
0001/004 T00004   in-service/idle    no
0001/005 T00005   in-service/idle    no
0001/006 T00006   in-service/idle    no
0001/007 T00007   in-service/idle    no
0001/008 T00008   in-service/idle    no
0001/009 T00009   in-service/idle    no
0001/010 T00010   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 the 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 **All** from the **Local Address** drop down menu.
   - Enter the IP address of the network SBC in the **Remote Address** field or enter a ***** to capture all traffic.
   - Specify the **Maximum Number of Packets to Capture**, 10000 is shown as an example.
   - Specify the filename of the resultant pcap file in the **Capture Filename** field
   - Click on **Start Capture**.



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 in the form of a 200 OK will be seen from the Vodafone network.

10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Communication Manager R6.3 as an Evolution Server, Avaya Aura® Session Manager R6.3 and Avaya Session Border Controller for Enterprise to Vodafone Germany SIP Trunk Service. Vodafone SIP Trunk Service is a SIP-based Voice over IP solution providing businesses a flexible, cost-saving alternative to traditional hardwired telephony trunks. The service was successfully tested with a number of observations listed in **Section 2.2**. Testing was carried out on Avaya SBC build 6.2.0.Q48. Because of issues observed with SRTP handling during testing with another Service provider, it is recommended to use the latest GA build. At the time of writing, this was 6.2.1 Q7 (FP1)

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] *Installing and Configuring Avaya Aura® System Platform*, Release 6.3, May 2013.
[2] *Administering Avaya Aura® System Platform*, Release 6.3, May 2013.
[3] *Avaya Aura® Communication Manager using VMware® in the Virtualized Environment Deployment Guide*, May 2013
[4] *Avaya Aura® Communication Manager 6.3 Documentation library*, August 2013.
[5] *Avaya Aura® System Manager using VMware® in the Virtualized Environment Deployment Guide* Release 6.3 May 2013
[6] *Implementing Avaya Aura® System Manager* Release 6.3, May 2013
[7] *Upgrading Avaya Aura® System Manager to 6.3.2*, May 2013.
[8] *Administering Avaya Aura® System Manager* Release 6.3, May 2013
[9] *Avaya Aura® Session Manager using VMware® in the Virtualized Environment Deployment Guide* Release 6.3 May 2013
[10] *Implementing Avaya Aura® Session Manager* Release 6.3, May 2013
[11] *Upgrading Avaya Aura® Session Manager* Release 6.3, May 2013
[12] *Administering Avaya Aura® Session Manager* Release 6.3, June 2013,
[13] *Installing Avaya Session Border Controller for Enterprise*, Release 6.2 June 2013
[14] *Upgrading Avaya Session Border Controller for Enterprise* Release 6.2 July 2013
[15] *Administering Avaya Session Border Controller for Enterprise* Release 6.2 March 2013
[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.