Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Communication Manager R6.3 as an Evolution Server, Avaya Aura® Session Manager R6.3 and Avaya Session Border Controller for Enterprise to support SFR SIP Trunk (Collecte SIP) - Issue 1.0

Abstract

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between the SFR Collecte SIP 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. SFR 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 SFR Collecte SIP 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 SFR 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 SFR.

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 SFR, calls made to SIP and H.323 telephones at the enterprise
- Outgoing calls from the enterprise site completed via SFR SIP Trunk to PSTN destinations, calls made from SIP and H.323 telephones
- Calls using the G.729A and G.711 A Law 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 (also known as "shuffling") with 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 SFR SIP Trunk requiring Avaya response and sent by Avaya requiring SFR response

2.2. Test Results

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

- Inbound Toll Free calls were not tested as no Toll Free access was available
- There is no signalling from the network when the PSTN places the call on hold. Though this is an acceptable method of placing the call on hold, it is not a particularly informative test.
- On some of the Supplementary Features tests, there was no ringback or there was one way media. This was thought to be a network issue and Early Media was enabled to work around it. Early Media is enabled by setting Initial IP-IP Direct Media to "n".
- When a call was transferred or forwarded to the PSTN such that both parties were on the PSTN, there was a media delay in excess of half a second. This was thought to be due to multiple VoIP hops and is a characteristic of the test network.
- Incoming fax calls failed when made from a PSTN line in the Avaya Lab in Galway. Calls succeeded when made from SFR premises in Paris. There was also a fault where the network detected the fax before the call was answered and sent an UPDATE message to Communication Manager to change the media to T.38. Communication Manager was rejecting this with "488 Fax request rejected". A workaround for this is to use a separate SIP line for fax and to disable early media by setting Initial IP-IP Direct Media to "y" in the Signaling Group. Refer to **Section 5.5** for details.
- Tests using SIP one-X Communicator in Other Phone mode were not completely reliable. The application shut down during the first test of consultative transfer to the PSTN and the first conference with a PSTN endpoint filed.
- The outgoing long duration call failed on the first attempt but was successful on a subsequent attempt.

2.3. Support

Le Service Technique SFR Business Team est joignable 24H/24, 7J/7 par un numéro gratuit pour signalisation des incidents techniques sur le service Collecte SIP.

CENTRE SERVICE CLIENT SFR Business Team
0 800 950 920

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an Enterprise site connected to SFR Collecte SIP. 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 Flare® for Windows running on a laptop PC. Within the enterprise, RTP was used for transport of media.

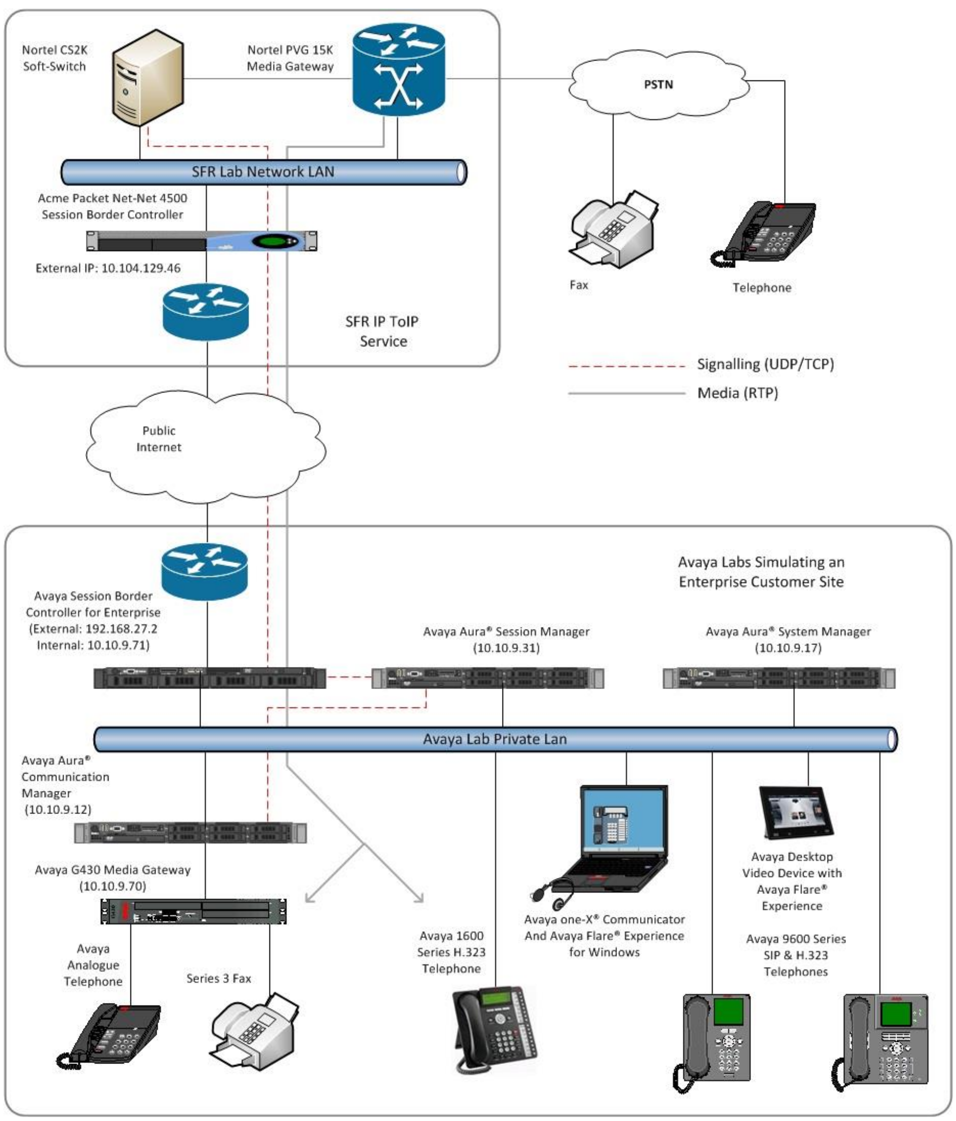


Figure 1: Test Setup SFR Collecte SIP to Avaya Enterprise

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	6.3.10.0.631008 VMware Tools: 9.0.0.15210 (782409)
Dell PowerEdge R620 running System Manager on VM Version 8	6.3.10 Build No. 6.3.0.8.5682 Patch 6.3.8.4514 Build No. 6.3.10.7.2656
Dell PowerEdge R620 running Communication Manager on VM Version 8	R016x.03.0.124.0 patch 21754
Avaya Session Border Controller Advanced for Enterprise Server	6.3.0.Q19
G430 Media Gateway	FW Version/HW Vintage: 36.9
Avaya 1616 Phone (H.323)	1.3 Maintenance Release 6
Avaya 96x0 Phone (H.323)	3.2.3
Avaya 96x1 Phone (H.323)	6.4
Avaya A175 Desktop Video Device (SIP)	Flare® Experience Release 1.1.2
Avaya 96x0 Phone (SIP)	R2.6.12
Avaya 96x1 Phone (SIP)	R6.4.1
Avaya one–X® Communicator (H.323) on Lenovo T510 Laptop PC	6.2 FP4
Avaya Flare® experience for Windows on Lenovo T510 Laptop PC	Release 1.1.4.23
Analogue Handset	NA
Analogue Fax	NA
SFR	
Nortel Media Server	Communication Server 2000 (CS2K) CVM16
Nortel PSTN gateway	PVG 15k PCR 8.2
Acme Packet Net-Net 4500 SBC	SCX6.2.0 MR-6 GA

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 SFR 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 SFR 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 SFR 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 1
             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 30
  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**.

```
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          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 **10.10.9.31** are the **Name** and **IP Address** for the Session Manager SIP interface. Also note the **procr** address as this is the processor interface that Communication Manager will use as the SIP signalling interface to Session Manager.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
SMVM1               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 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 SFR were configured, namely **G.729A** and **G.711A**.

```
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.729A           n          2        20
 2: G.711A           n          2        20
 3:
 4:
 5:
 6:
 7:


     Media Encryption
 1: none
 2:
 3:
```

SFR Collecte SIP supports T.38 for transmission of fax. To allow transmission using T.38, Navigate to **Page 2** and define as follows:

- 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 **t.38-standard** where fallback is not required. SFR also supports transmission of fax over G.711, though this did not work during testing. Refer to **Section 2.2** for details.

5.5. Administer SIP Signaling Groups

This signalling group (and trunk group) will be used for inbound and outbound PSTN calls to SFR Collecte SIP. During test, this was configured to use TCP and port 5060 to facilitate tracing and fault analysis. It is recommended however, to use TLS (Transport Layer Security) and the default TLS port of 5061 for security. Configure the **Signaling Group** using the **add signaling-group x** command as follows:

- Set **Group Type** to **sip**
- Set **Transport Method** to **tcp**
- 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 **5060** (Commonly used TCP 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 Communication Manager 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: tcp
        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: 5060                Far-end Listen Port: 5060
                                        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
```

Note: Setting **Initial IP-IP Direct Media** to **y** allows the establishment of the media directly between the Avaya SBCE and the endpoint without establishment via the Media Gateway first. This makes efficient use of Media Gateway resources as they are not required for initial set-up of the call. The disadvantage is that Early Media is not used and this was having a detrimental effect during testing with occasional one way transmission and no ringback in some call scenarios. To work around this, **Initial IP-IP Direct Media** was set to **n** for testing of voice calls.

The issues with T.38 fax testing described in **Section 2.2** were resolved by setting **Initial IP-IP Direct Media** to **y**. To have it set differently for voice and fax, two trunks are required. The two trunks would be used by splitting the voice and fax traffic at the Session Manager and differentiating by port, for example 5060 for voice and 5062 for fax.

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 **tie**
- 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: tie         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 SFR 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 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 Diversion Header** to **n**
- Set **Support Request History** to **y**
- Set the **Telephone Event Payload Type** to **101**
- 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? n
                                  Network Call Redirection? n

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


                        Convert 180 to 183 for Early Media? n
                  Always Use re-INVITE for Display Updates? 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, 101 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. Some Avaya SIP endpoints have a default value of 120.

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 unmodified as the extension number; this was adapted in the Session Manager to the national number format required by SFR Collecte SIP. See **Section 6.4** for details of the adaptation. This calling party number is sent in the SIP From, Contact and PAI headers as well as the History-Info header for forwarded calls.

```
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: 1
                                                          Maximum Entries: 540
```

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 SFR Collecte SIP. 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 with 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                    10   10    1        pubu         n
    00                   13   15    1        pubu         n
    0035391              13   13    1        pubu         n
    030                  10   10    1        pubu         n
    0800                 8    14    1        pubu         n
    0900                 8    8     1        pubu         n
    1                    4    4     1        pubu         n
    112                  3    3     1        pubu         n
    118                  3    6     1        pubu         n
    3                    4    4     1        pubu         n
    7000                 4    4     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 to map incoming Direct Dial-In (DDI) calls to the Communication Manager extensions if not already mapped using a Session Manager adaptation. The incoming digits sent in the INVITE message from the Service Provider can be manipulated as necessary to route calls to the desired extension. During test, the incoming DDI numbers were adapted in Session Manager to the Communication Manager extension numbers; this process is described in **Section 6.4**. When done this way, there is no requirement for any incoming digit translation in 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 correct routing and handling of CLI in a solution with Avaya Aura® Messaging.

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 2291. 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** (e.g., 9) if required by the routing configuration
- 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
 2291             OPS         -       2291             aar        1
 2291             EC500       -       0035389434nnnn   ars        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.

In the above screenshot the Mobile phone number is partially obscured.

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

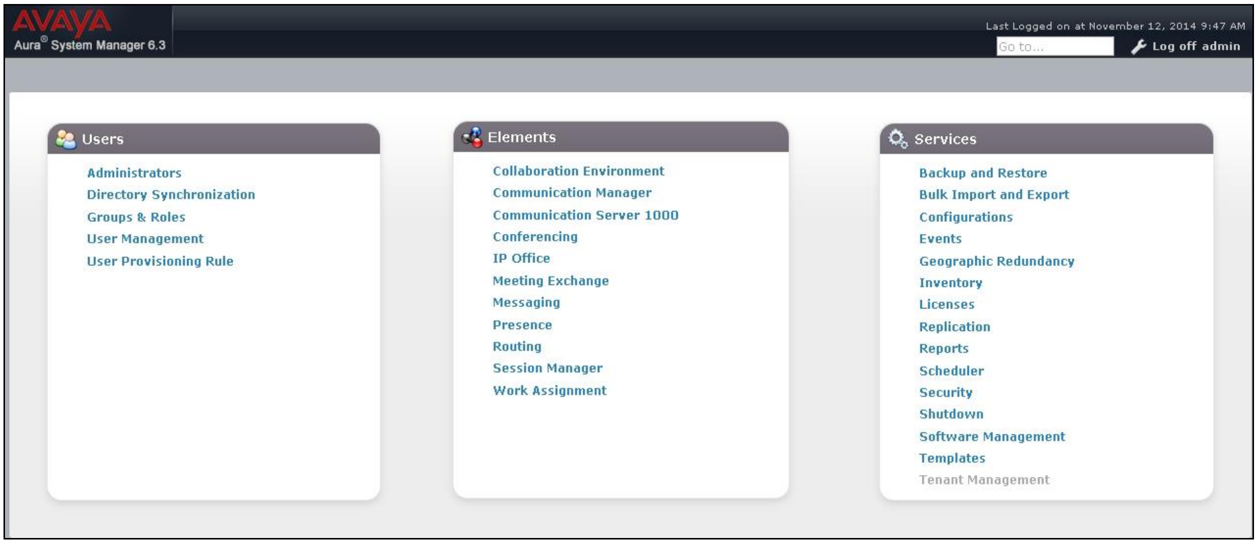
6. Configuring Avaya Aura® Session Manager

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

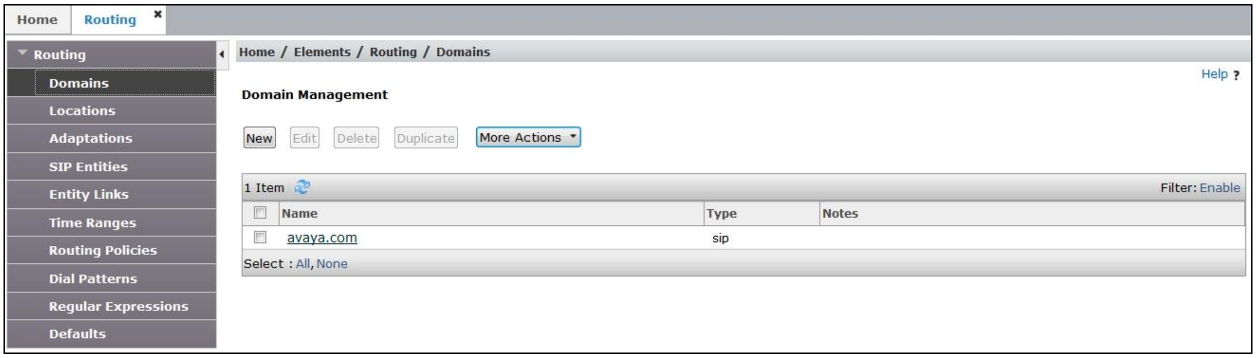
6.1. Log in to Avaya Aura® System Manager

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

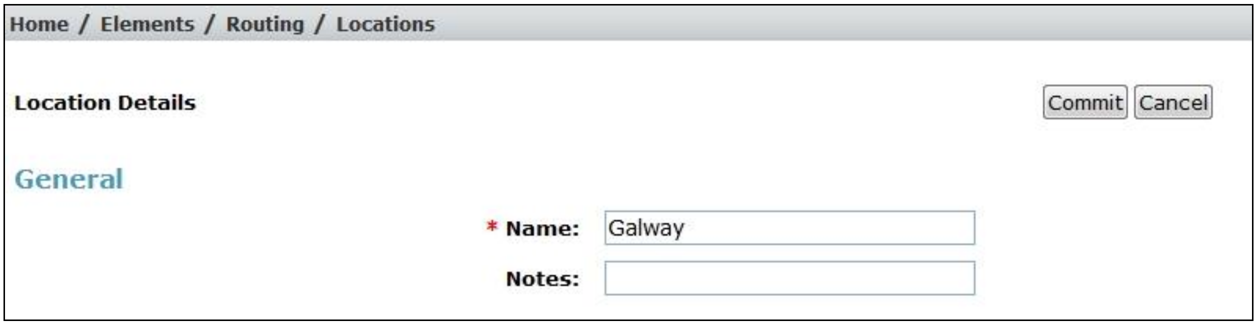
To add the SIP domain that will be used with Session Manager, select **Routing** from the **Home** tab menu and in the resulting tab select **Domains** from left hand menu. Click the **New** button to create a new SIP domain entry. In the **Name** field enter the domain name of the enterprise site or a name agreed with SFR; 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.



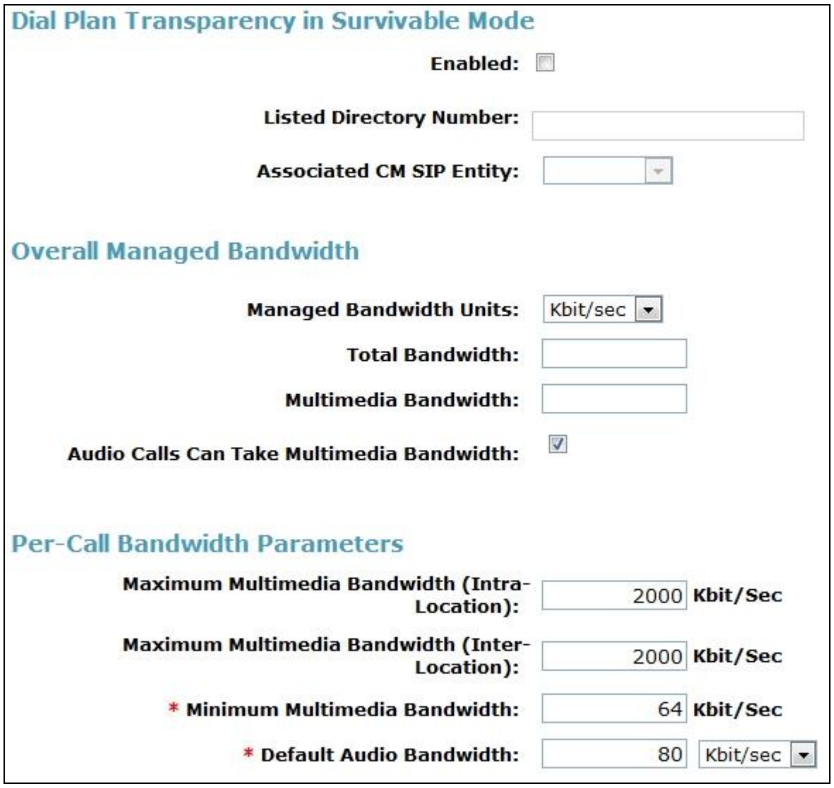
Note: If the existing domain name used in the enterprise equipment does not match that used in the network, a Session Manager adaptation can be used to change it.

6.3. Administer Locations

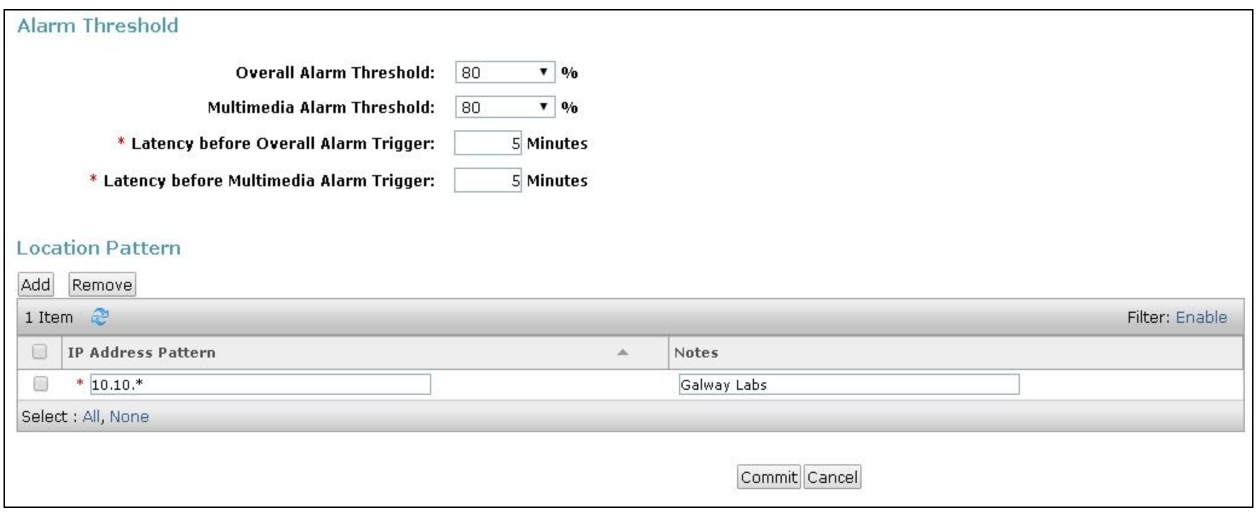
Locations can be used to identify logical and/or physical locations where SIP Entities reside for the purposes of bandwidth management. One location is added to the sample configuration for all of the enterprise SIP entities. On the **Routing** tab select **Locations** from the left hand menu. Under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

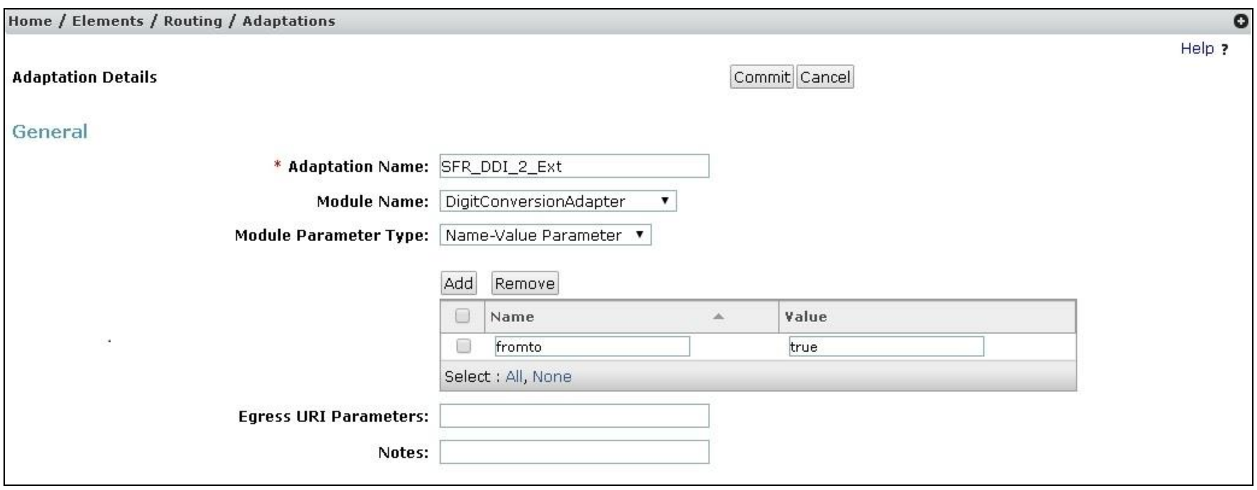


6.4. Administer Adaptations

Calls from SFR are received at the enterprise in national format with leading “0” on the Request URI. An Adaptation specific to SFR 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).

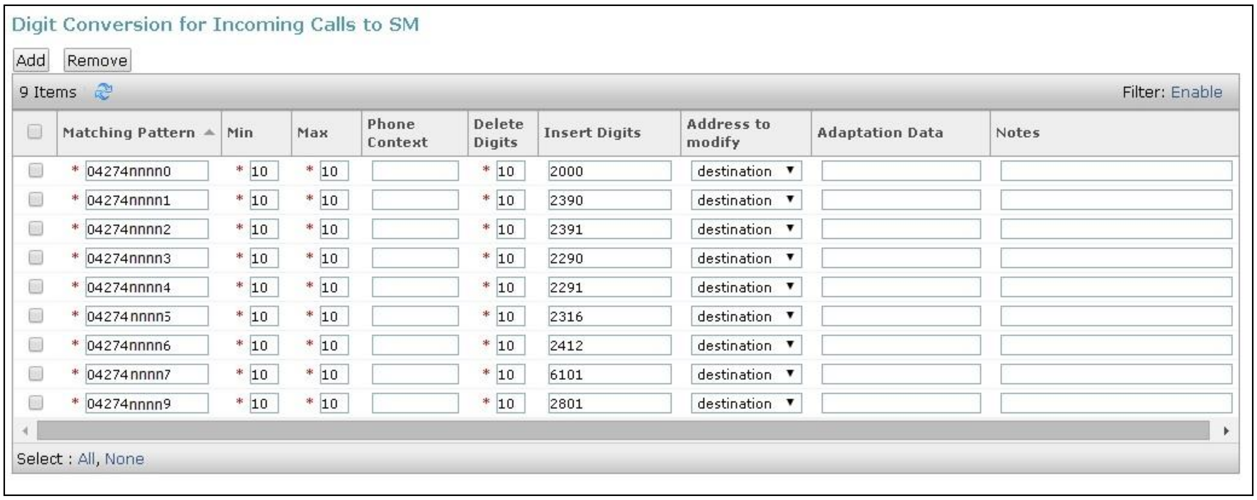
- In the **Adaptation name** field, enter a descriptive title for the adaptation.
- In the **Module name** enter **DigitConversionAdapter**. This is used for simple digit conversion adaptations.
- In the **Module parameter** field, select **Name-Value Parameter** in the **Module Parameter Type** drop 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 simple prefix 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

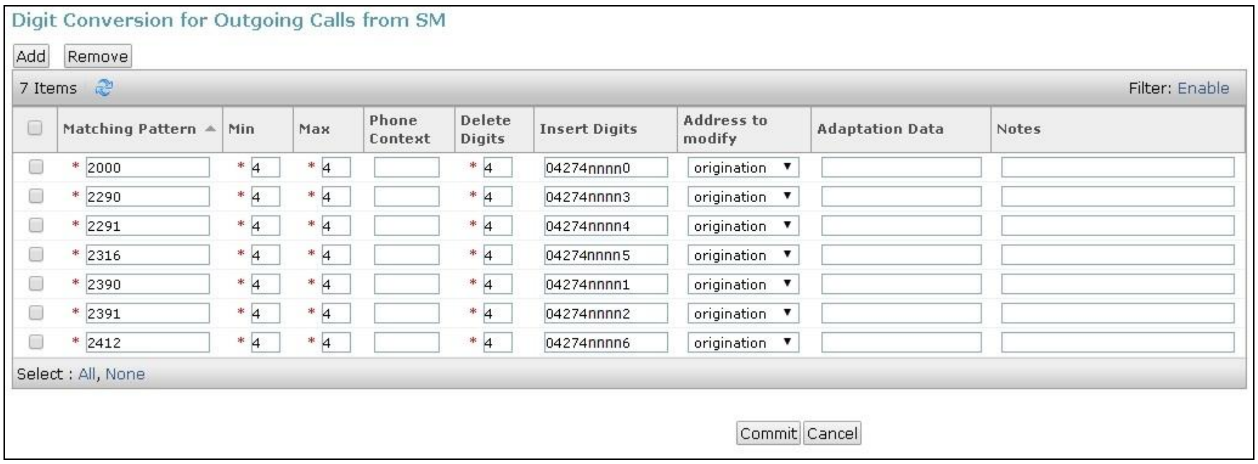
	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
	* 04274nnnn0	* 10	* 10		* 10	2000	destination		
	* 04274nnnn1	* 10	* 10		* 10	2390	destination		
	* 04274nnnn2	* 10	* 10		* 10	2391	destination		
	* 04274nnnn3	* 10	* 10		* 10	2290	destination		
	* 04274nnnn4	* 10	* 10		* 10	2291	destination		
	* 04274nnnn5	* 10	* 10		* 10	2316	destination		
	* 04274nnnn6	* 10	* 10		* 10	2412	destination		
	* 04274nnnn7	* 10	* 10		* 10	6101	destination		
	* 04274nnnn9	* 10	* 10		* 10	2801	destination		

Select : All, None

Note: In the above screenshot the DDI numbers are partially obscured

The screenshot below shows a translation for each calling party number. Again, this is not normally necessary where the extension number forms part of the national number.

- Under **Matching Pattern** enter the extension number as received from the CM.
- Under **Min** and **Max** enter the Minimum and Maximum digits of the extension number, during testing this was **4**.
- Under **Delete Digits** enter the number of digits to delete to remove all digits that don't form part of the national number, during testing this was all of them..
- Under **Insert Digits** enter digits to be inserted. During test, this was the full national number. If the extension number forms part of the DDI number, only the most significant digits are entered here.
- Under **Address to Modify** choose **origination** from the drop down box to apply this rule to the From and P-Asserted-Identity headers only.


Digit Conversion for Outgoing Calls from SM

Add Remove

7 Items Filter: Enable

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
	* 2000	* 4	* 4		* 4	04274nnnn0	origination		
	* 2290	* 4	* 4		* 4	04274nnnn3	origination		
	* 2291	* 4	* 4		* 4	04274nnnn4	origination		
	* 2316	* 4	* 4		* 4	04274nnnn5	origination		
	* 2390	* 4	* 4		* 4	04274nnnn1	origination		
	* 2391	* 4	* 4		* 4	04274nnnn2	origination		
	* 2412	* 4	* 4		* 4	04274nnnn6	origination		

Select : All, None

Commit Cancel

Note: In the above screenshot 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**:

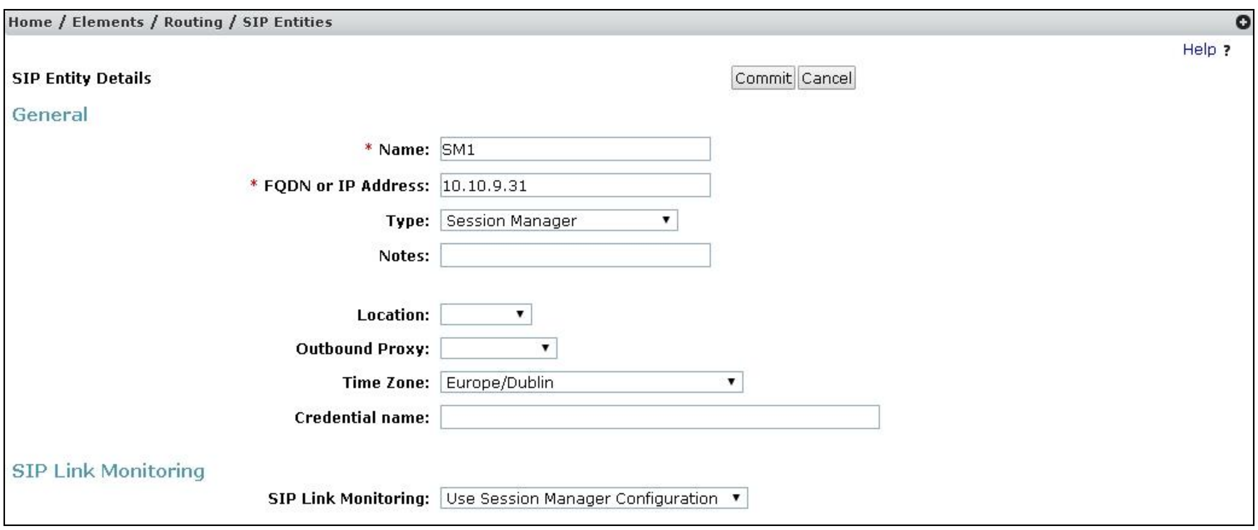
- In the **Name** field enter an informative name
- In the **FQDN or IP Address** field enter the IP address of the Session Manager or the signalling interface on the connecting system
- In the **Type** field use **Session Manager** for a Session Manager SIP entity, **CM** for a Communication Manager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

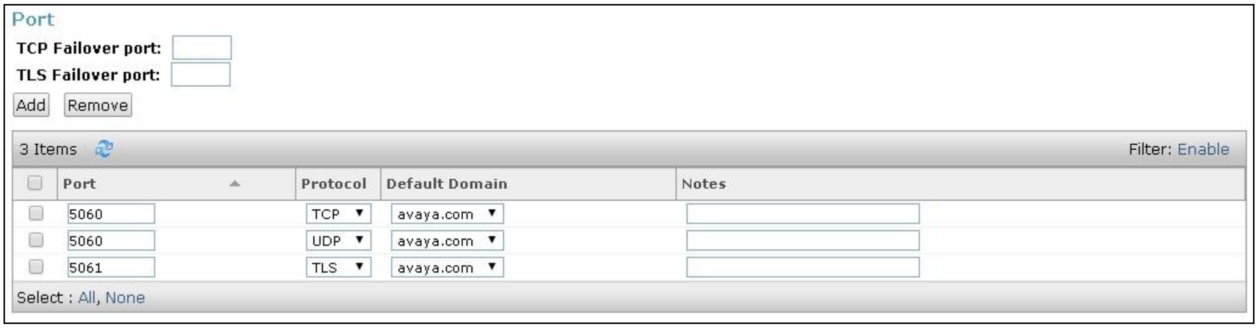
6.5.1. Avaya Aura® Session Manager SIP Entity

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.



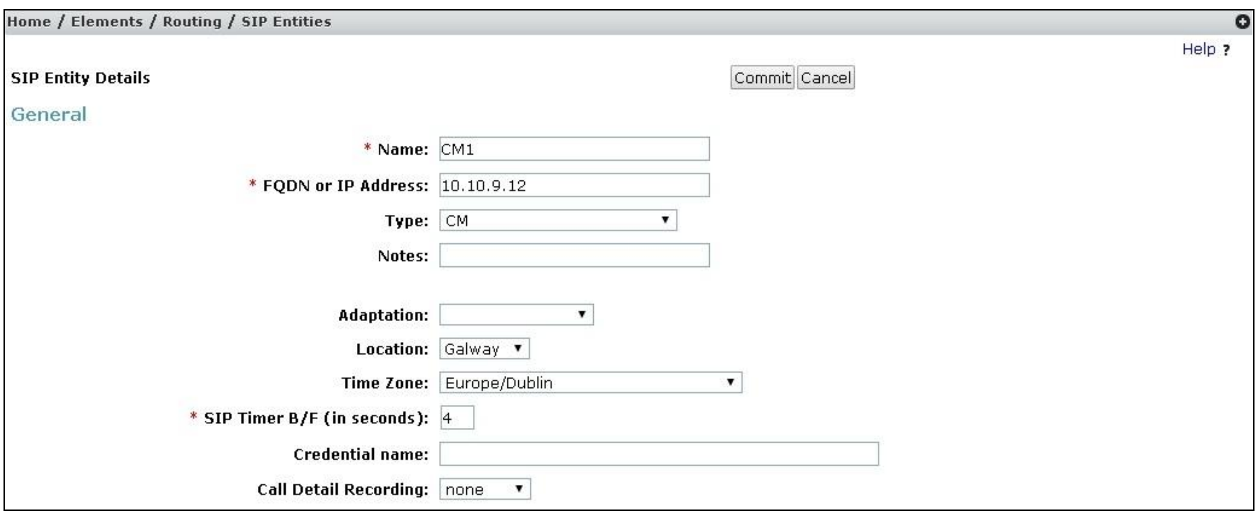
The Session Manager must be configured with the port numbers on the protocols that will be used by the other SIP entities. To configure these scroll to the bottom of the page and under **Port**, click **Add**, then edit the fields in the resulting new row.

- 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 **Location** to that defined in **Section 6.3** and the **Time Zone** to the appropriate time zone.

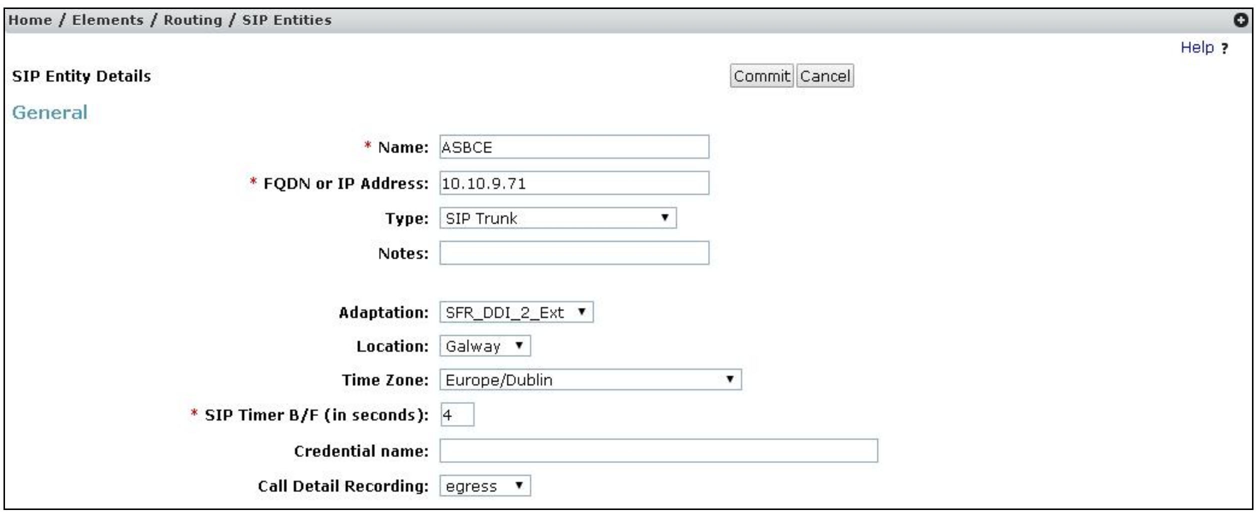


Note: No adaptation is required for Communication Manager as all required number modifications are performed by the adaptation applied to the Avaya SBCE.

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.

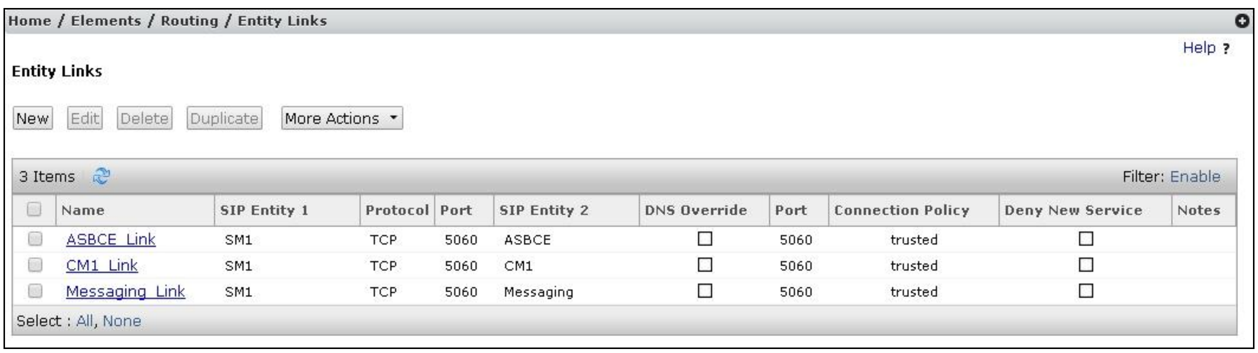


6.6. Administer Entity Links

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 **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 **Trusted** from the **Connection Policy** drop down menu 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.



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

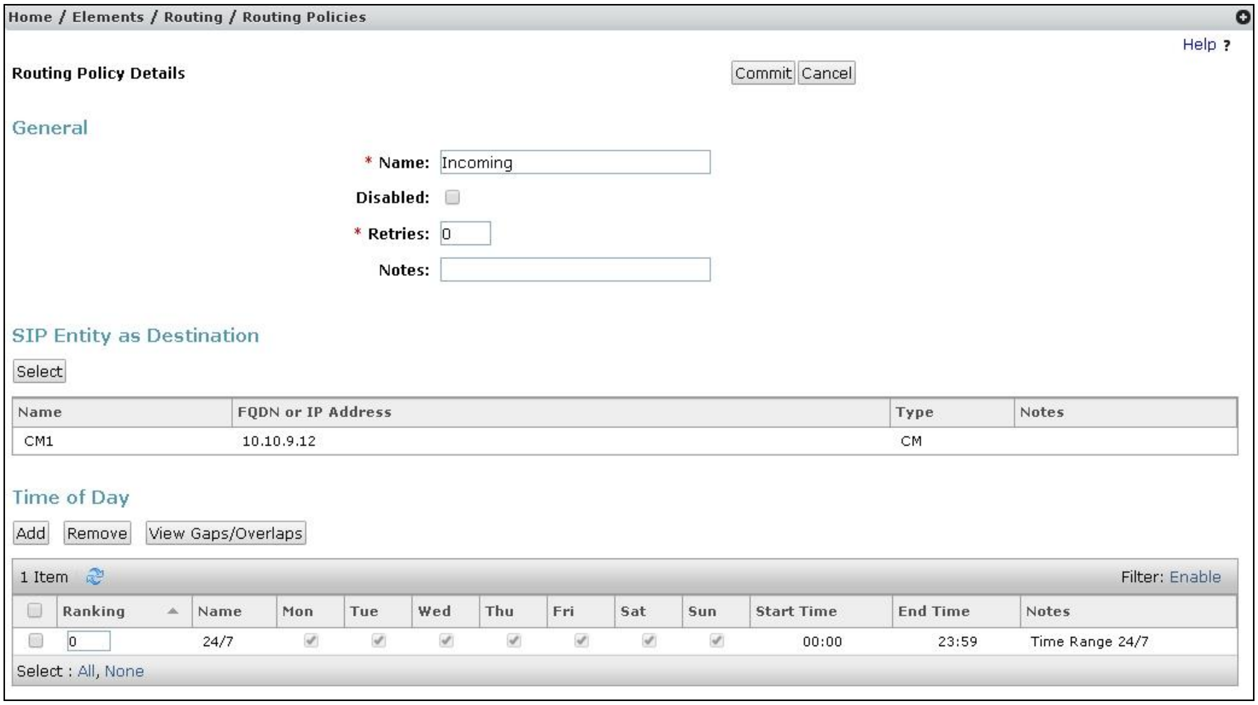
6.7. Administer Routing Policies

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).

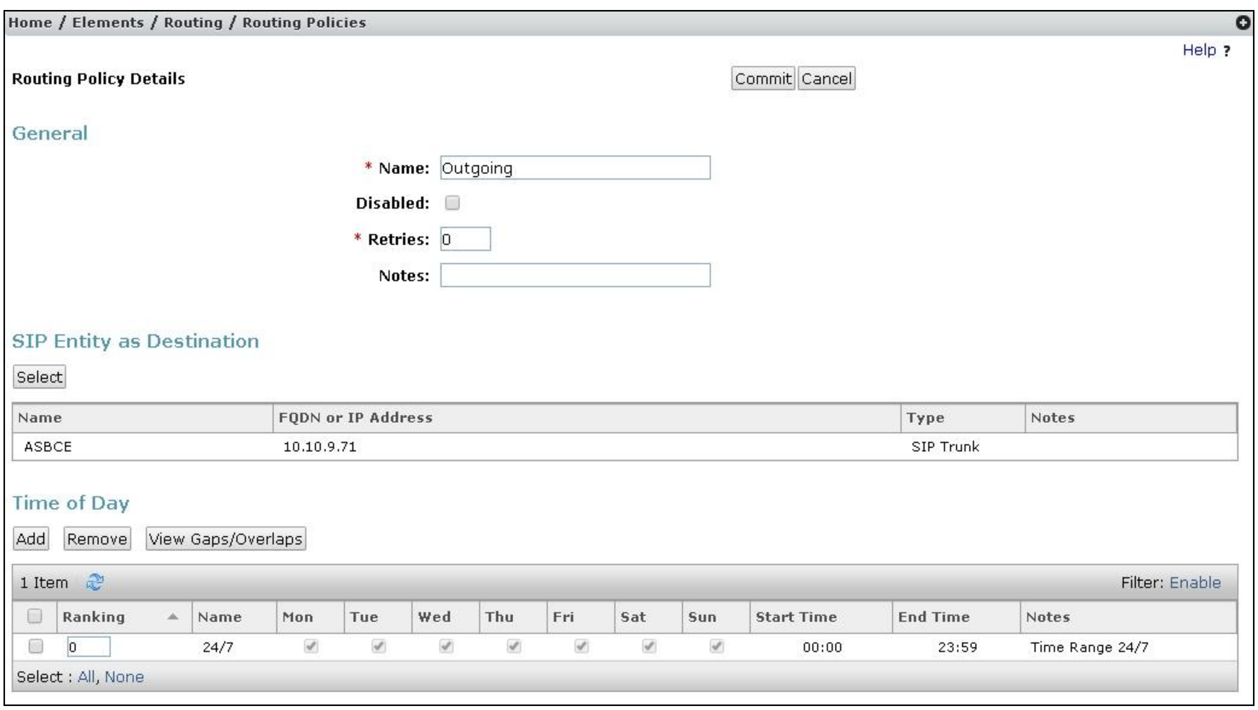
Under **General**:

- 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 SFR Collecte SIP.



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).

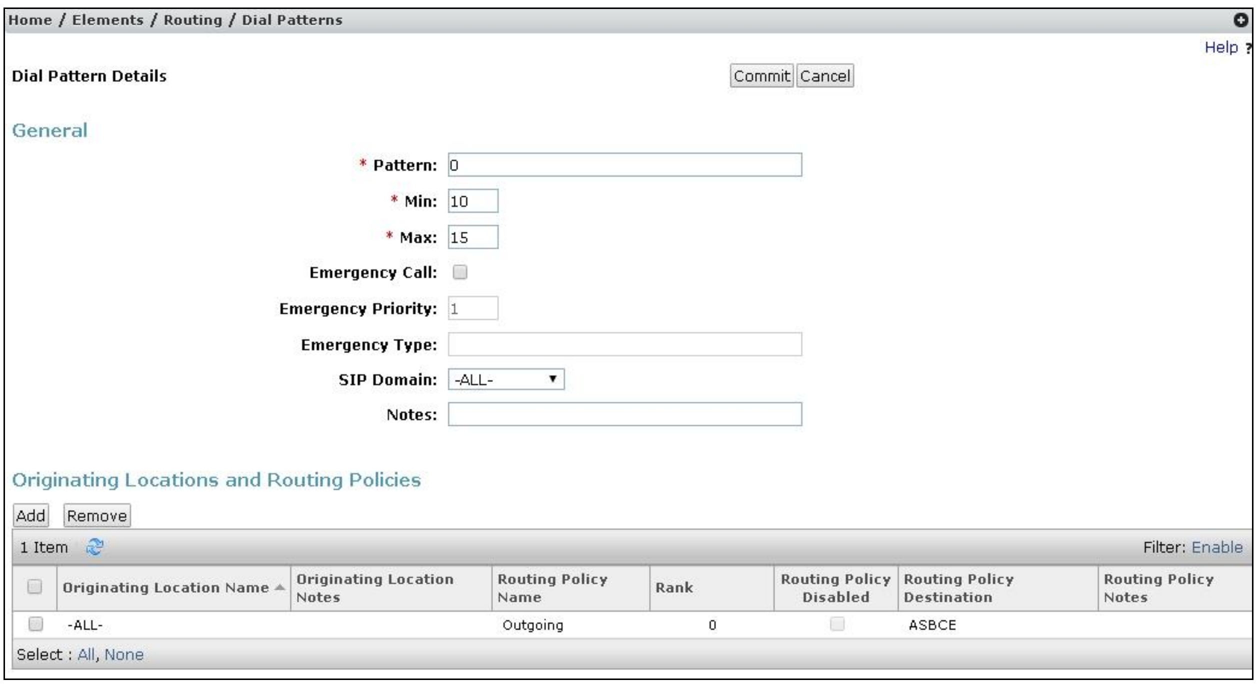
Under **General**:

- 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**

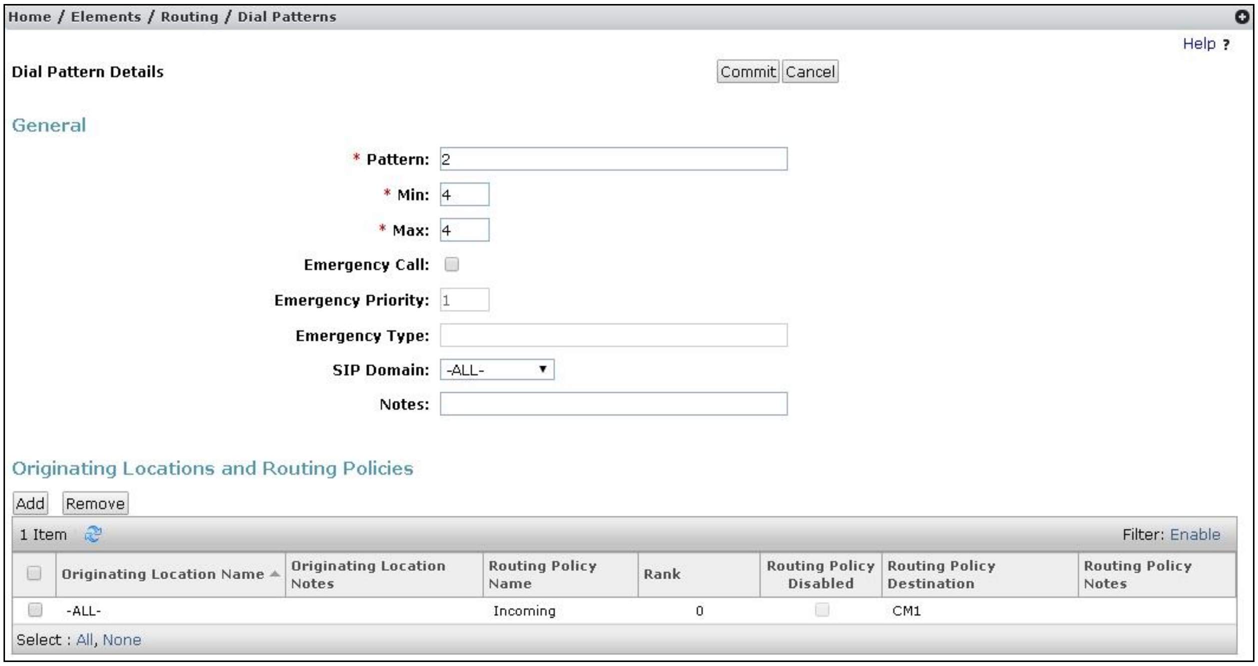
Under **Originating Locations and Routing Policies**:

- 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 SFR Collecte SIP.



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.

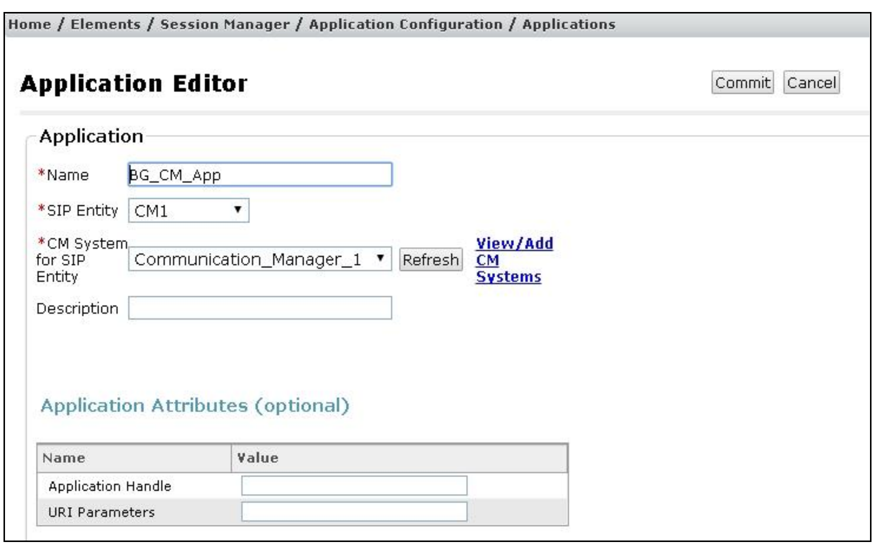


Note: The above configuration is used where the called party number has been converted to an extension number on Session Manager using an adaptation. If an adaptation is not used, a dial pattern will be required for the incoming DDI number.

6.9. Administer Application for Avaya Aura® Communication Manager

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**.

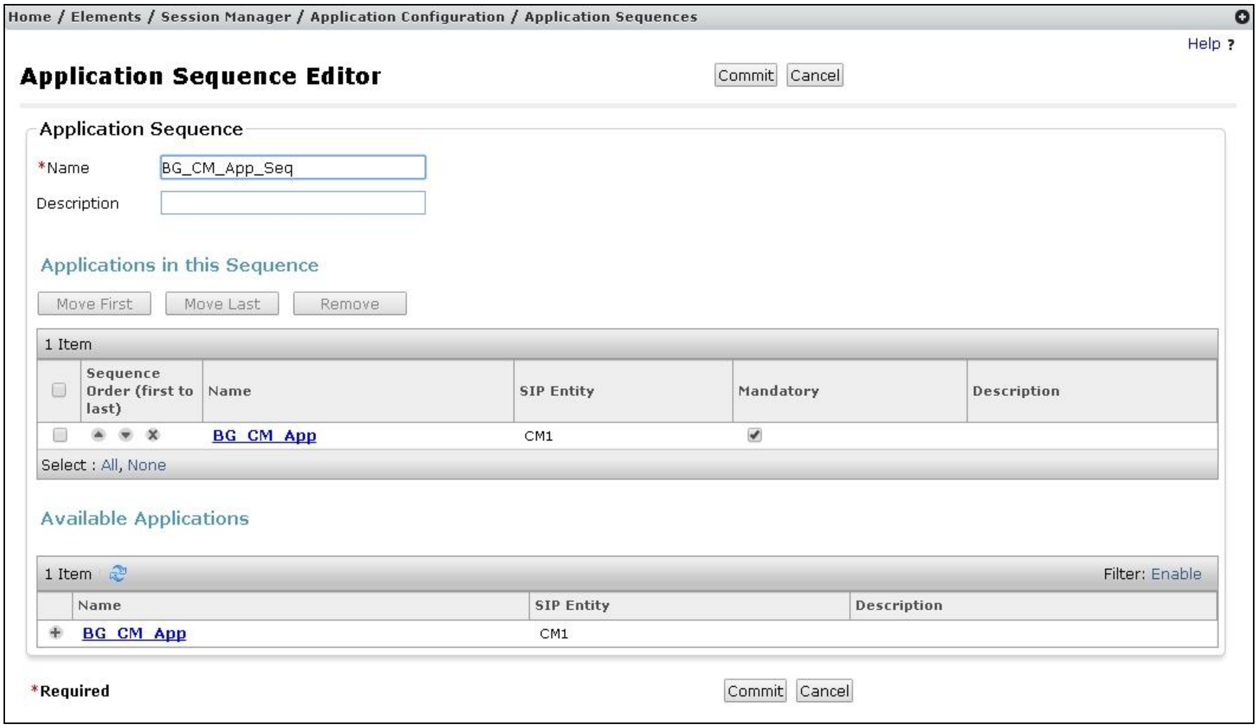
- In the **Name** field enter a name for the application
- In the **SIP Entity** field select the SIP entity for the Communication Manager
- In the **CM System for SIP Entity** field select the SIP entity for the Communication Manager and select **Commit** to save the configuration



6.10. Administer Application Sequence for Avaya Aura® Communication Manager

From the left panel navigate to **Session Manager → Application Configuration → Application Sequences** and click on **New**.

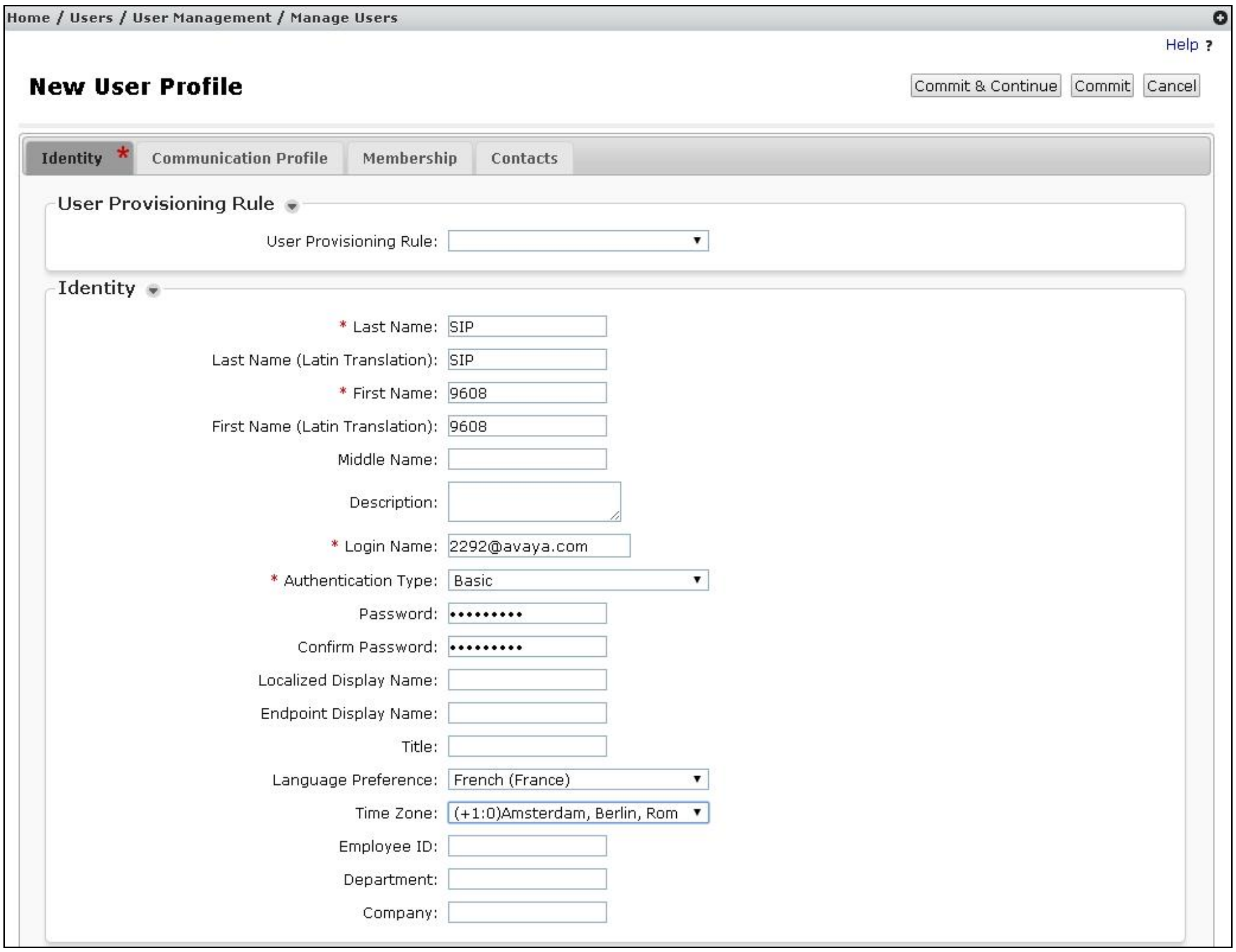
- In the **Name** field enter a descriptive name
- Under **Available Applications**, click the + sign in front of the appropriate application instance. When the screen refreshes the application should be displayed under the **Applications in this Sequence** heading. Select **Commit**.



6.11. Administer SIP Extensions

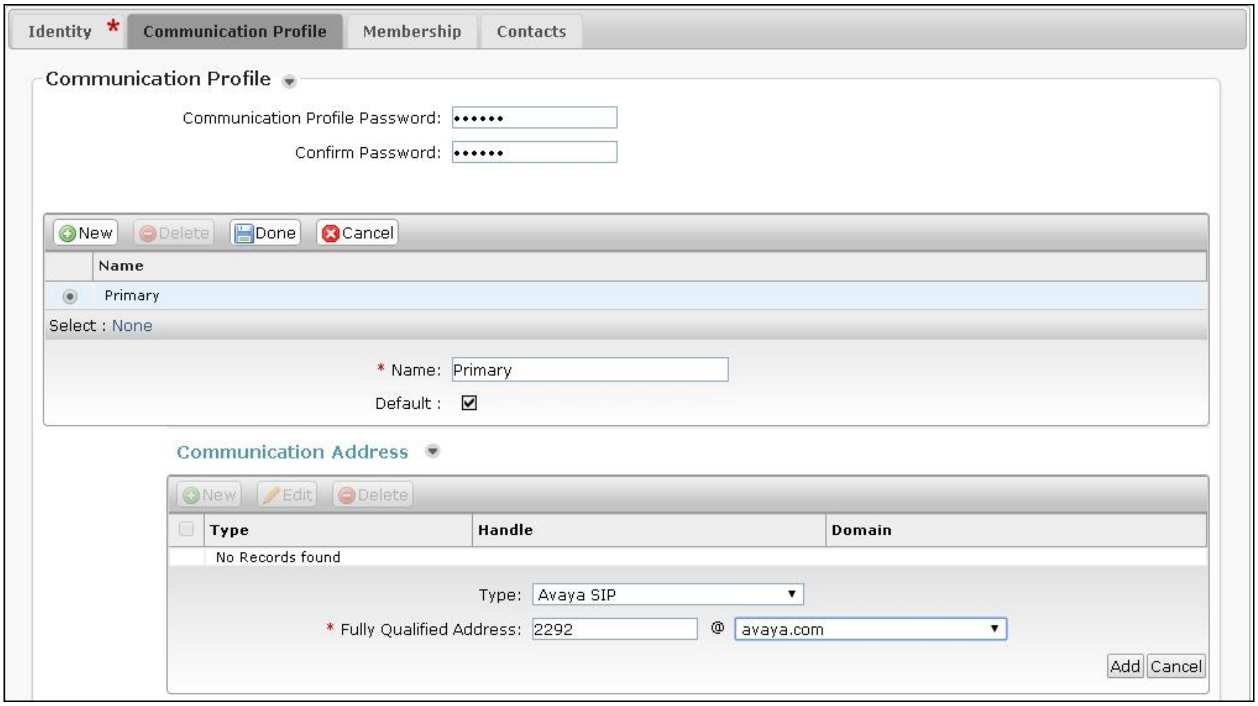
SIP extensions are registered with the Session Manager and use Communication Manager for their feature and configuration settings. From the **Home** tab select **User Management** from the menu. Then select **Manage Users** and click **New** (not shown). On the **Identity** tab:

- 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. **2292@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



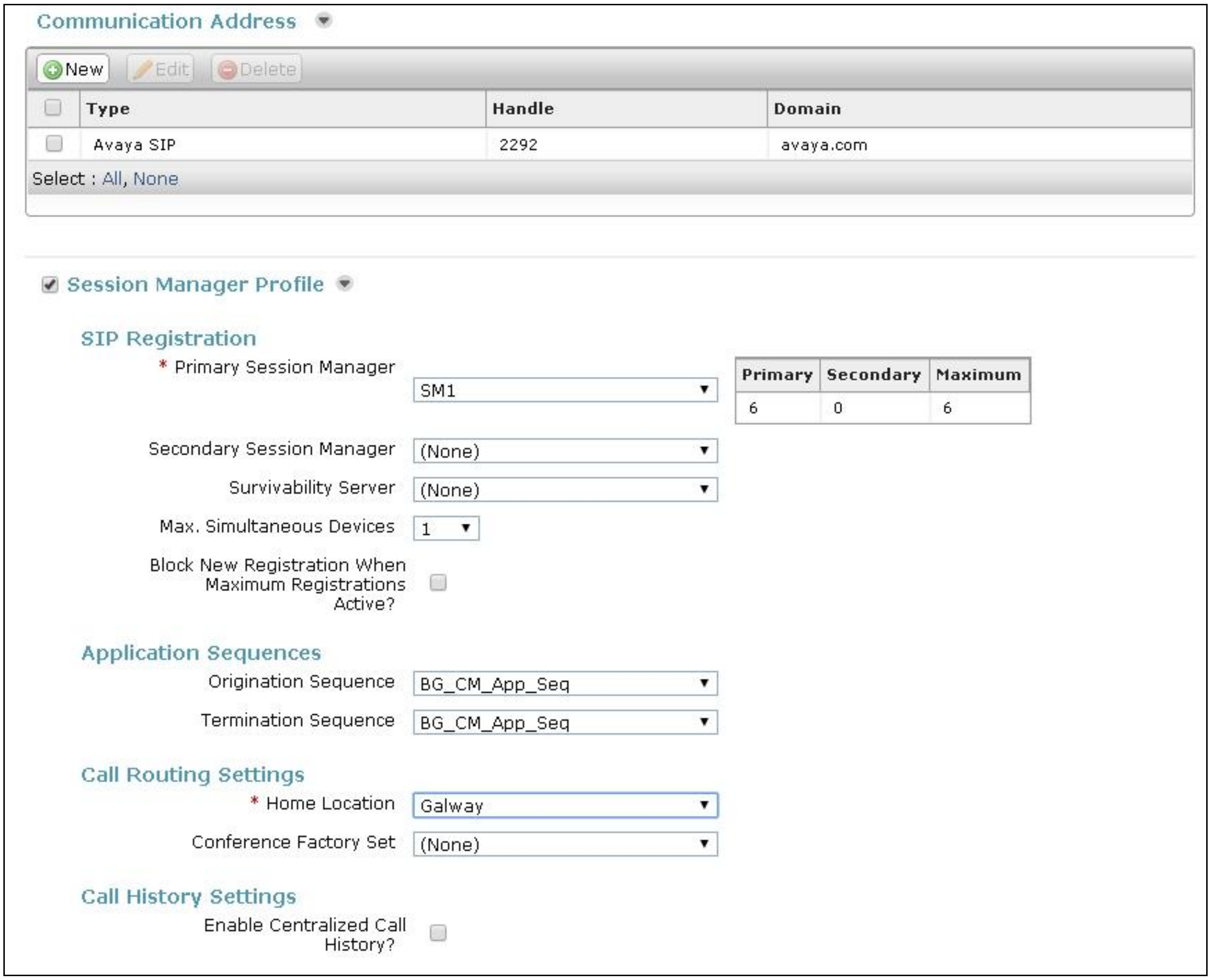
On the **Communication Profile** tab, enter a numeric **Communication Profile Password** and confirm it.

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.



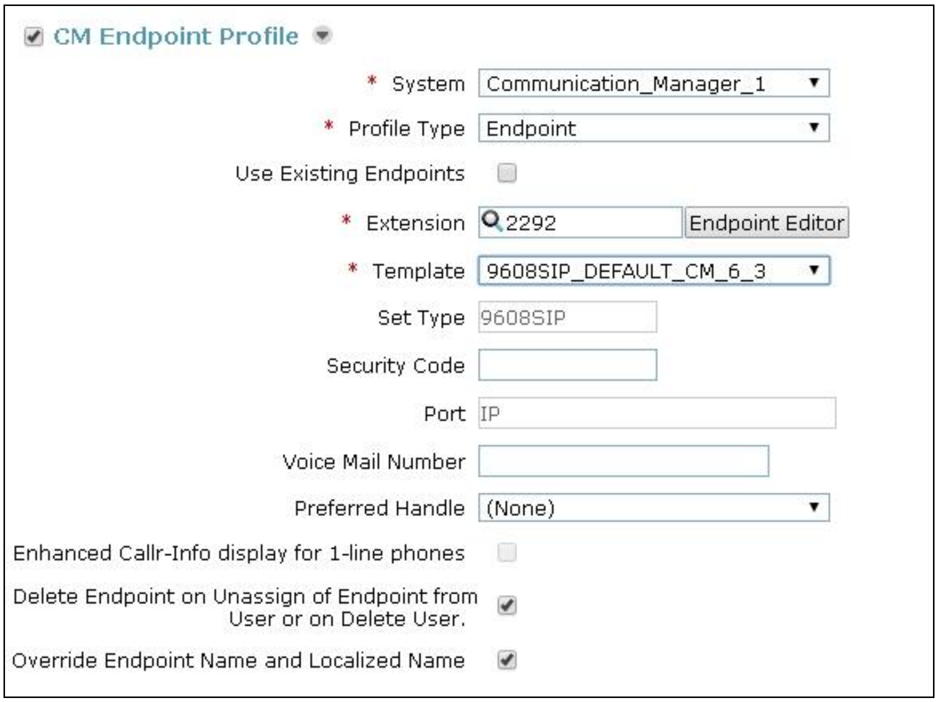
Expand the **Session Manager Profile** section.

- 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

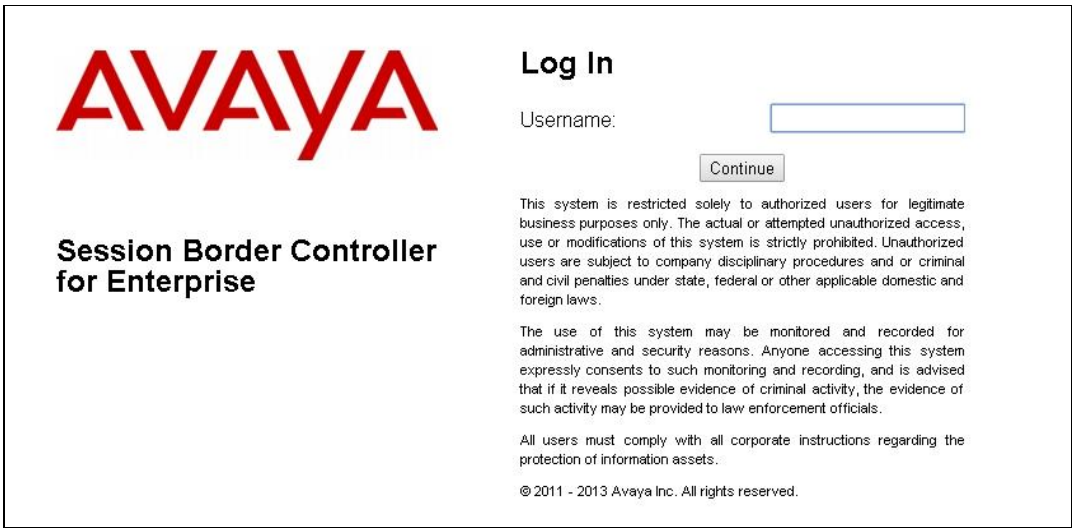


7. Configure Avaya Session Border Controller for Enterprise

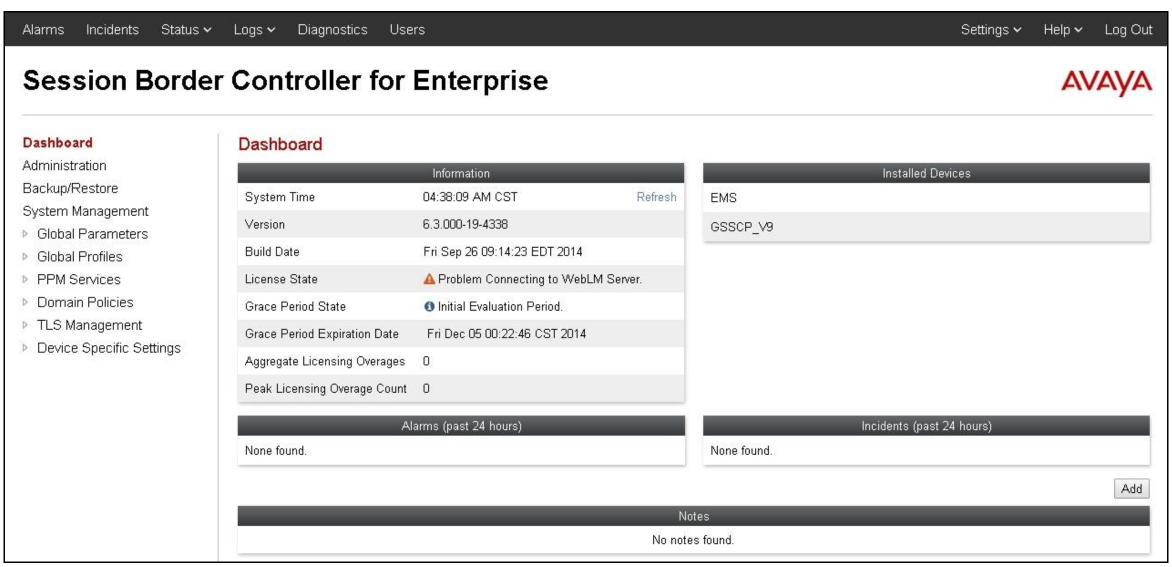
This section describes the configuration of the Session Border Controller for Enterprise (Avaya SBCE). The Avaya SBCE provides security and manipulation of signalling to provide an interface to the Service Provider's SIP Trunk that is standard where possible and adapted to the Service Provider's SIP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



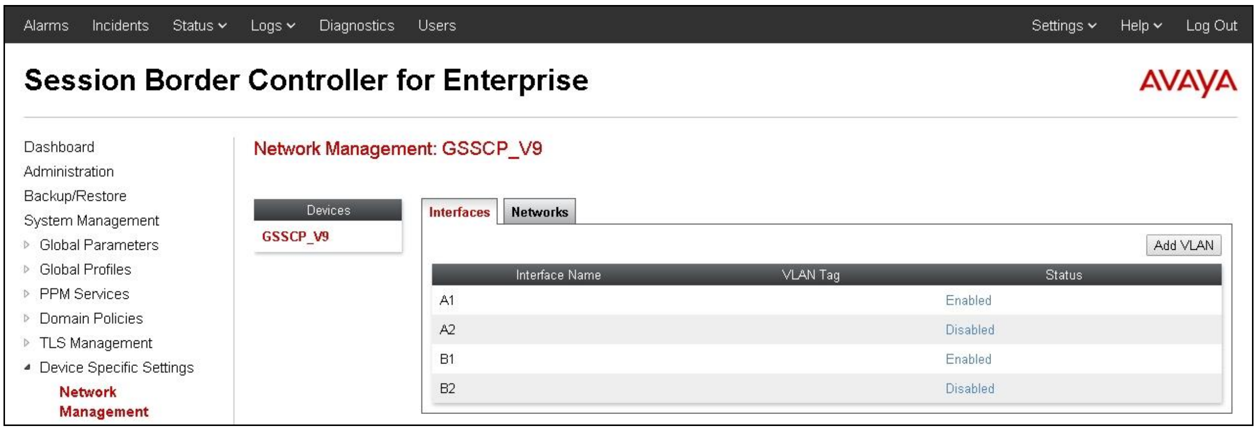
Once logged in, a dashboard is presented with a menu on the left-hand side. The menu is used as a starting point for all configuration of the Avaya SBCE.



7.2. Define Network Information

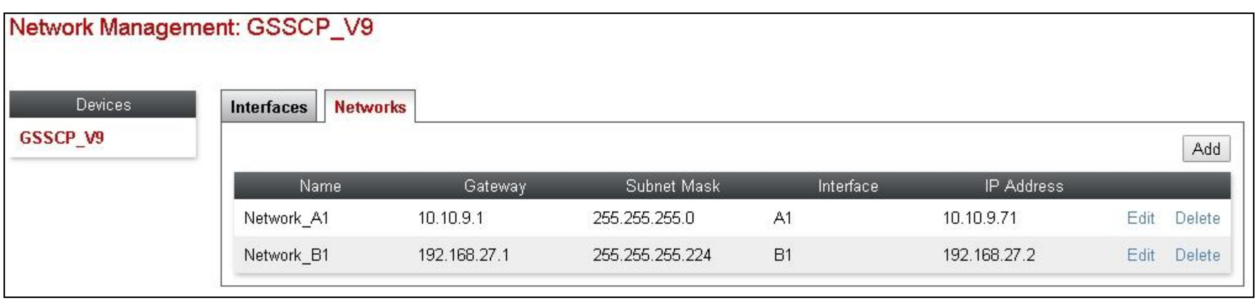
Network information is required on the Avaya SBCE to allocate IP addresses and masks to the interfaces. Note that only the **A1** and **B1** interfaces are used, typically the **A1** interface is used for the internal side and **B1** is used for external. Each side of the Avaya SBCE can have only one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

To define the network information, navigate to **Device Specific Settings → Network Management** in the main menu on the left hand side. The **Interface** tab appears first, click on the status of the required interfaces to change the state.



Select the **Networks** tab and click on **Add**. Enter details in the blank box that appears at the end of the list.

- Define the internal IP address with screening mask and assign to interface **A1**
- Select **Save** to save the information
- Click on **Add**
- Define the external IP address with screening mask 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)



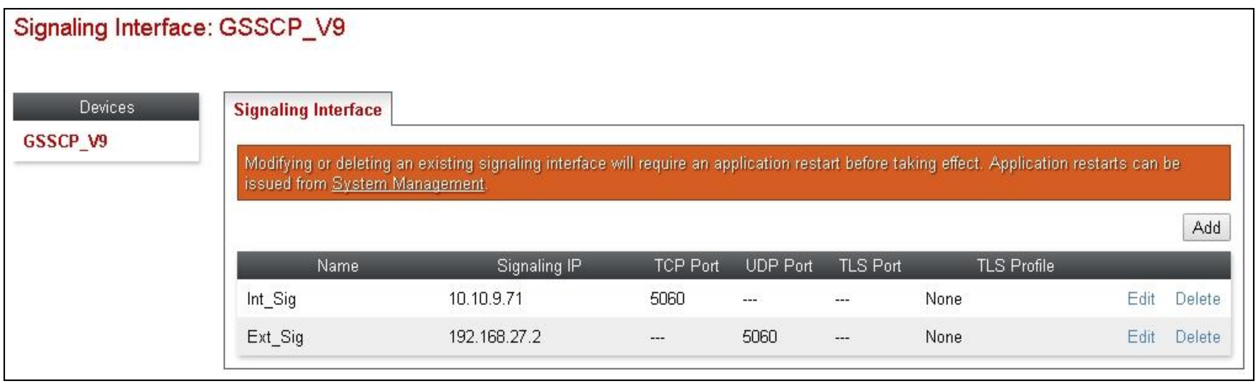
7.3. Define Interfaces

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

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 **TCP** port number, **5060** is used for the Session Manager
- 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 SFR Collecte SIP



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 SFR Collecte SIP

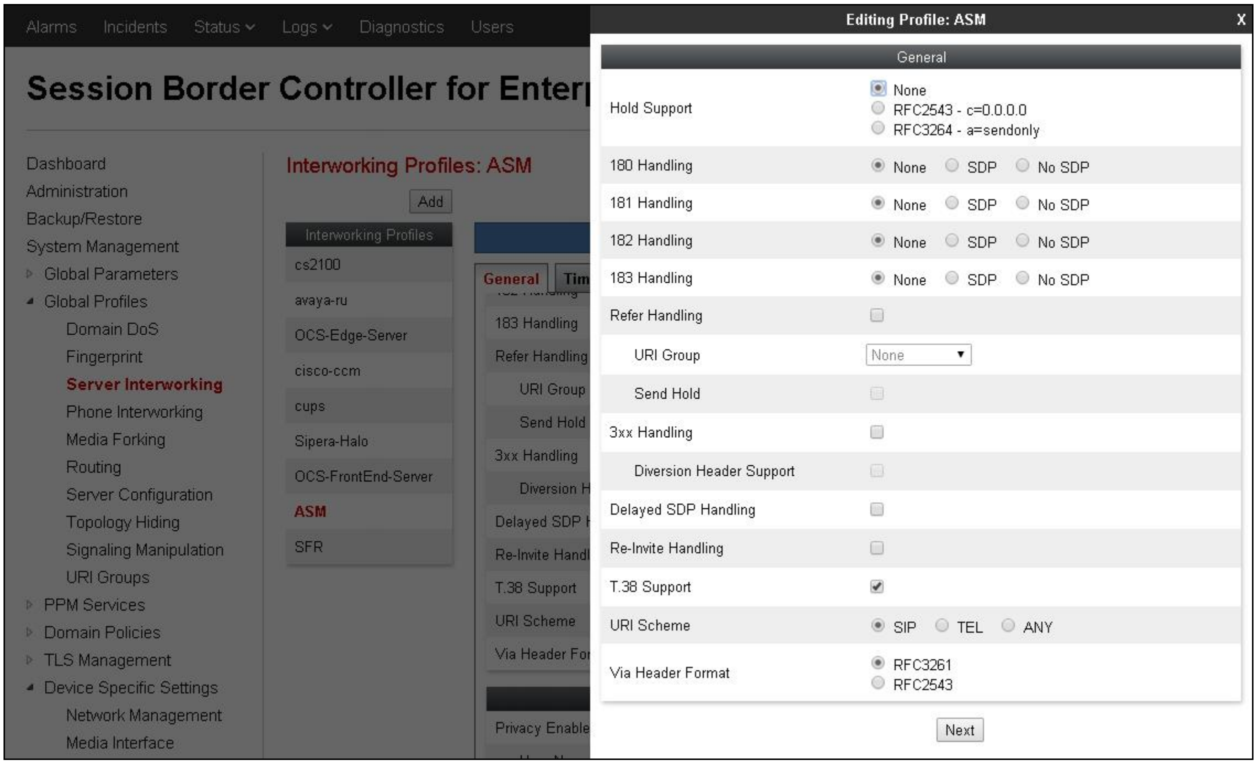
Note: During test, the port ranges for the internal and external media interfaces were set to the default values used on Communication Manager.

7.4. Define Server Interworking

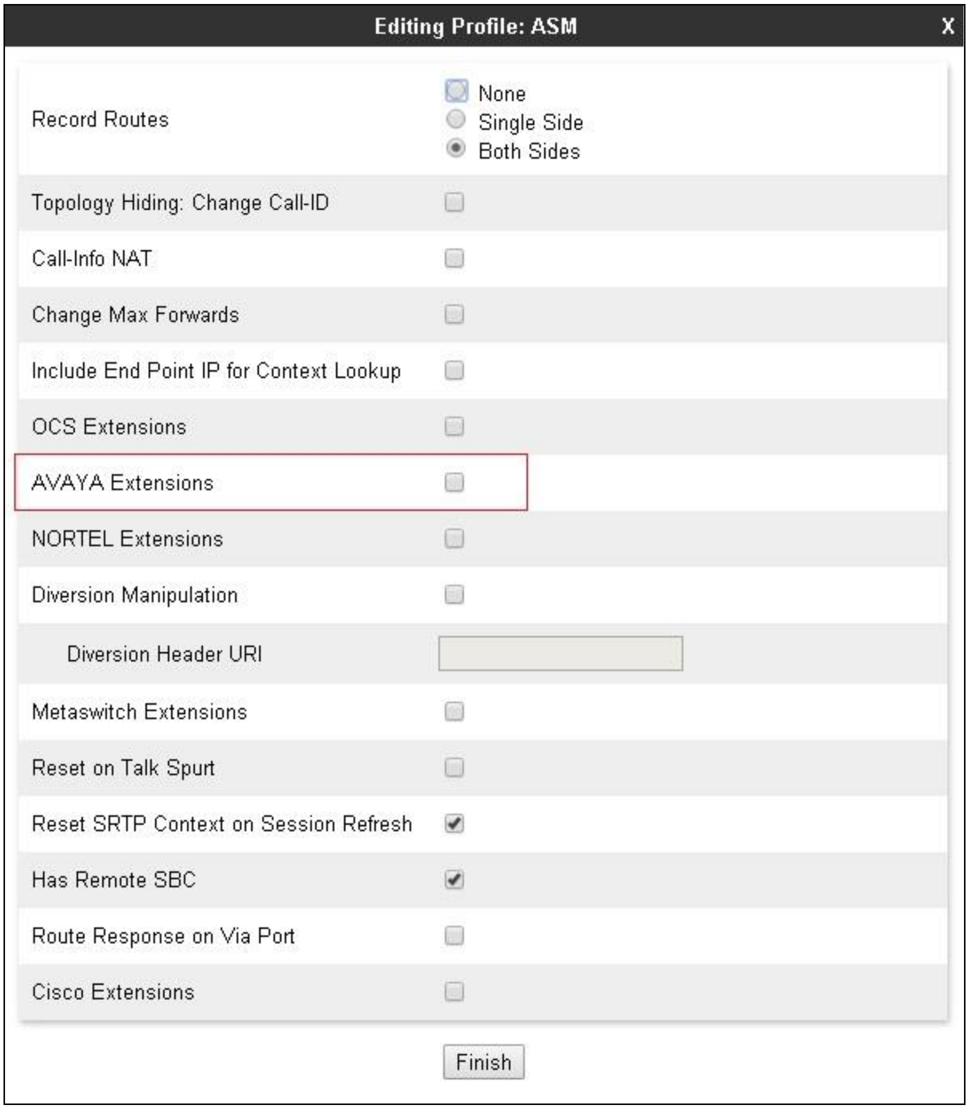
Server interworking is defined for each server connected to the Avaya SBCE. In this case, SFR Collecte SIP is connected as the Trunk Server and the Session Manager is connected as the Call Server. Configuration of interworking includes Hold support, T.38 fax support and SIP extensions.

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 **ASM** was used
- In the **General** tab (not shown) Select **Edit** and enter details in the pop-up menu
- Check the **T.38 Support** 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



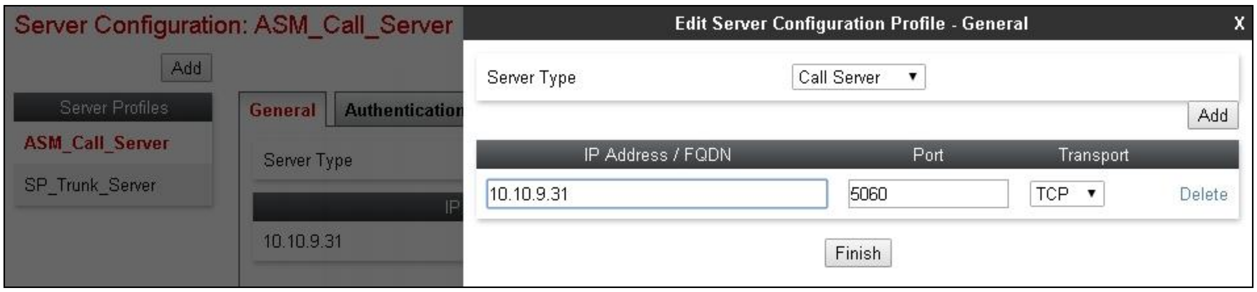
To define Server Interworking for SFR Collecte SIP, highlight the previously defined profile for the Session Manager and select **Clone Profile**. A pop-up menu is generated headed **Clone Profile** (not shown).

- In the **Clone Name** field enter a descriptive name for server interworking profile for SFR Collecte SIP and click **Finish** – in test **SFR** was used
- Select **Edit** and enter details in the pop-up menu
- Ensure the **T.38 Support** box is checked
- Select **Next** three times and **Finish**

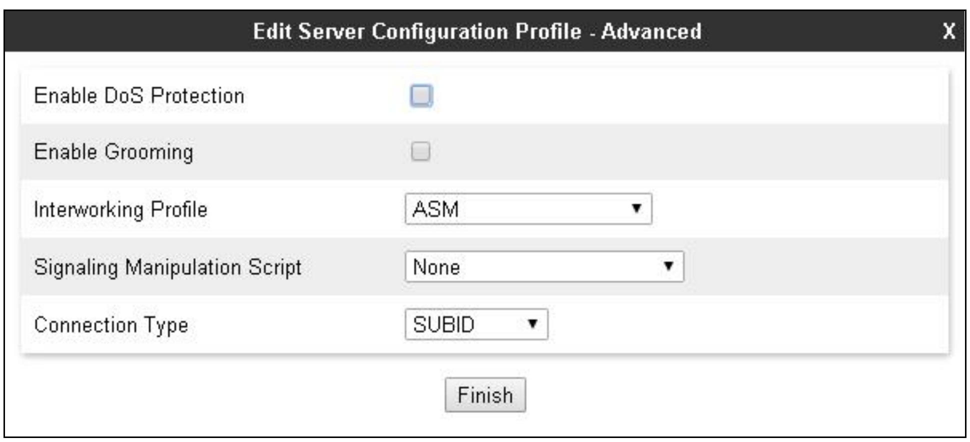
7.5. Define Servers

A server definition is required for each server connected to the Avaya SBCE. In this case, SFR Collecte SIP 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 on the Communication Manager in **Section 5.2**
- Check **TCP** in **Supported Transports**
- Define the **TCP** port for SIP signalling, **5060** is used for the Session Manager and click **Finish**

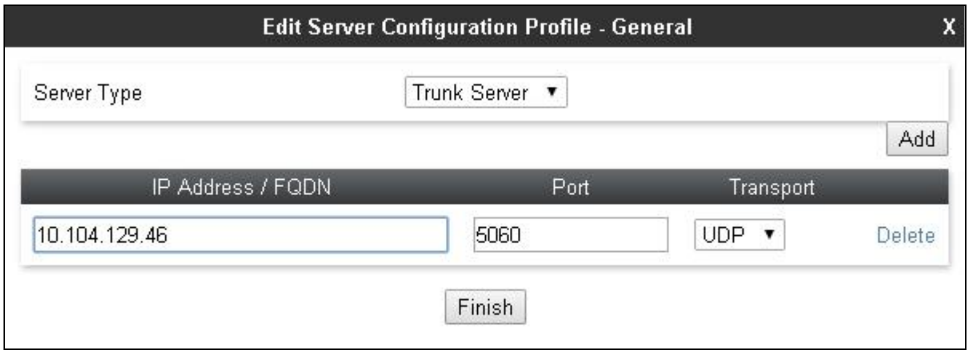


- Select the **Advanced** tab (not shown)
- In the **Interworking Profile** drop down menu, select the **Interworking Profile** for the Session Manager defined in **Section 7.4**
- Click **Finish**

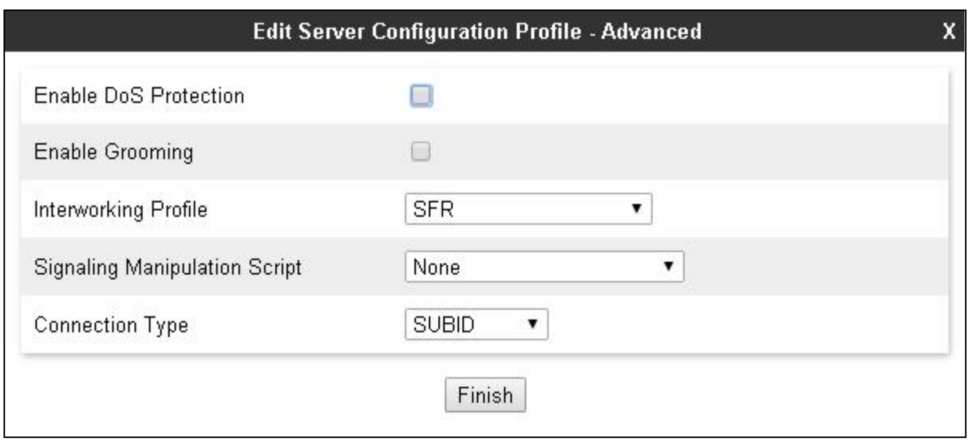


To define SFR Collecte SIP as a Trunk Server, navigate to **Global Profiles → Server Configuration** in the main menu on the left hand side. Click on **Add** and enter details in the pop-up menu.

- In the **Profile Name** field enter a descriptive name for SFR Collecte SIP 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 SFR Collecte SIP
- Check **UDP** in **Supported Transports**
- Define the **UDP** port for SIP signaling, **5060** is used for SFR
- Click **Finish**



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

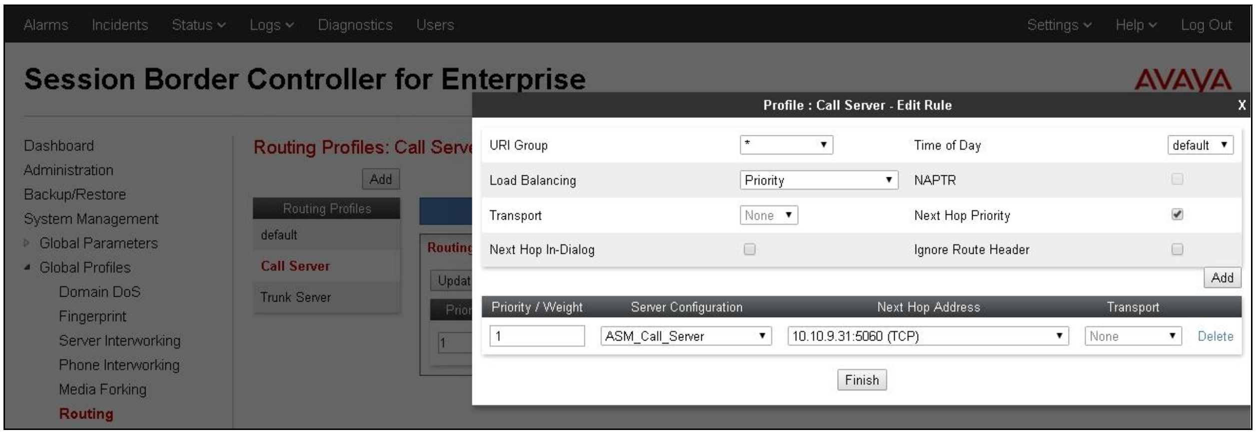


7.6. Define Routing

Routing information is required for routing to the Session Manager on the internal side and SFR Collecte SIP 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 Address**, default 5060 is used for TCP and UDP, and 5061 for TLS.

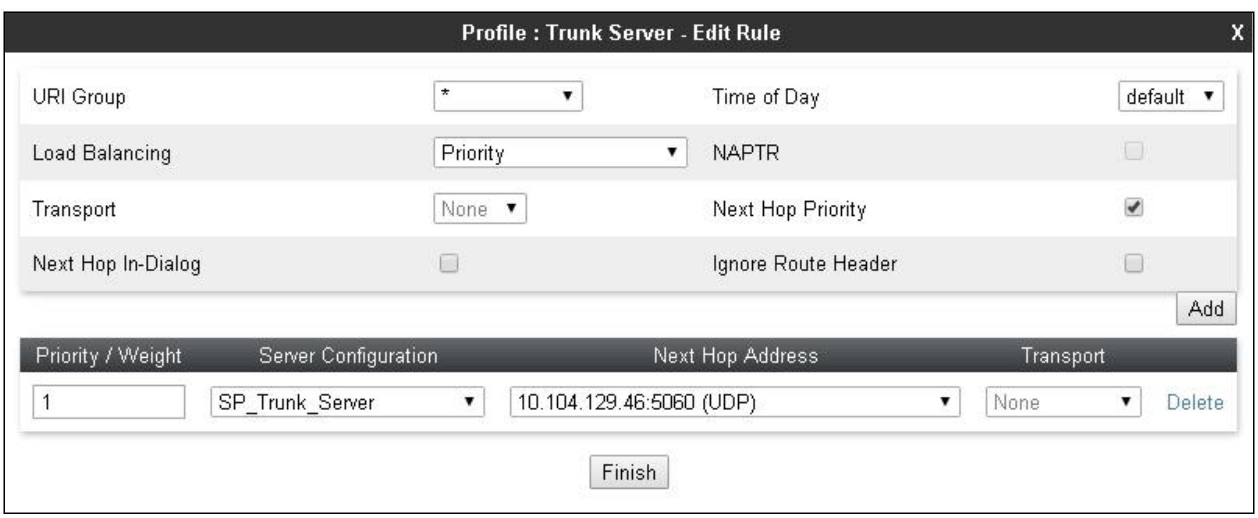
To define routing to the Session Manager, navigate to **Global Profiles → Routing** in the main menu on the left hand side. Click on **Add** and enter details in the **Routing Profile** pop-up menu.

- In the **Profile Name** field (not shown) enter a descriptive name for the Session Manager, in this case **Call Server**, and click **Next**
- Select the Session Manager Server Configuration in the **Server Configuration** field
- Select the Session Manager SIP interface address and port in the **Next Hop Address** field
- Select **TCP** for the **Transport**
- Click **Finish**



To define routing to SFR Collecte SIP, 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 (not shown) enter a descriptive name for SFR Collecte SIP, in this case a generic name of **Trunk Server** was used, and click **Next**
- Select the SFR Collecte SIP Server Configuration in the **Server Configuration** field
- Select the SFR Collecte SIP IP address and port in the **Next Hop Address** field
- Select **UDP** for the **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. 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**, **Via**, **Refer-To**, **To** and **Record-Route** 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 **Referred-By**, **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)
- For each of the headers leave the **Replace Action** at the default value of **Auto**

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

Note: The use of **Auto** results in an IP address being inserted in the host portion of the Request-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 SFR Collecte SIP.

To define Topology Hiding for SFR Collecte SIP, 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).

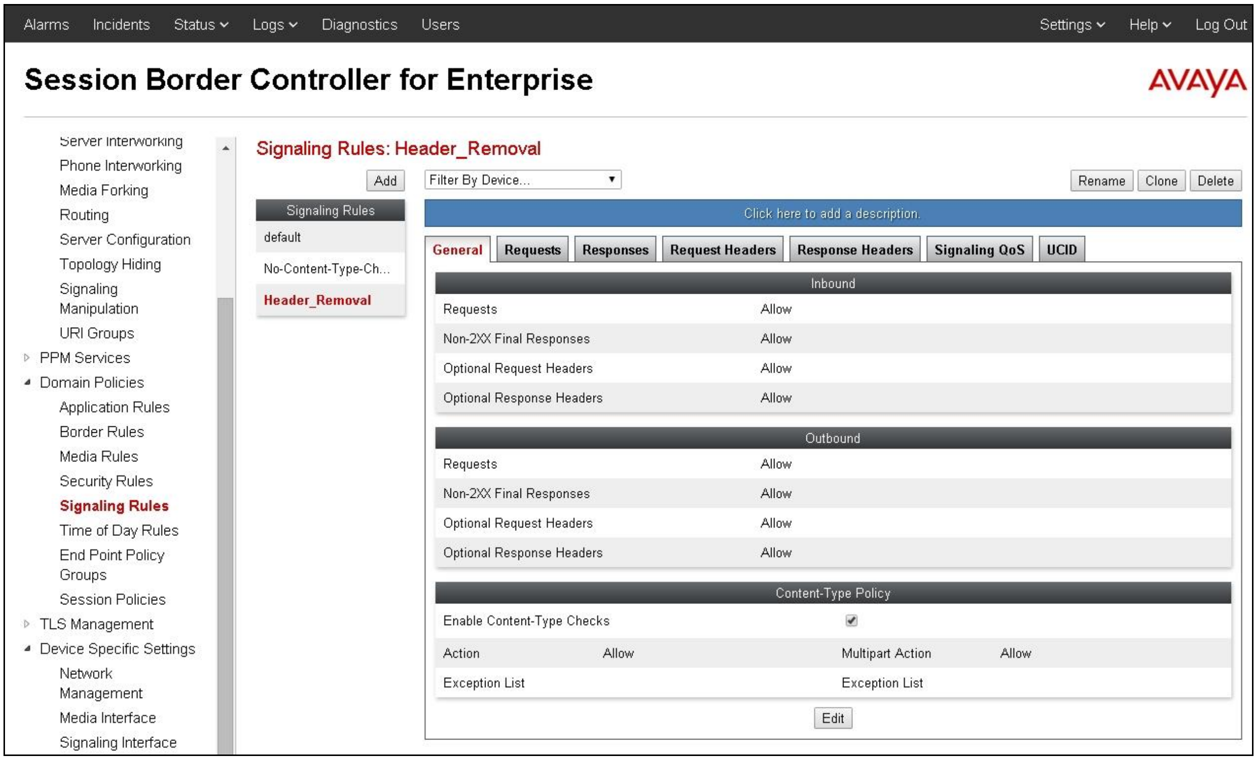
- In the **Profile Name** field enter a descriptive name for SFR Collecte SIP and click **Next**
- If the **Request-Line**, **Via**, **Refer-To**, **To** and **Record-Route** Headers aren't shown, click on **Add Header** and **s**elect from the **Header** drop down menu
- For each of the above headers, leave the **Replace Action** at the default value of **Auto**
- If the **Referred-By**, **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**

7.8. Signalling Rules

Signalling rules are a mechanism on the Avaya SBCE to manipulate the signalling beyond simple header manipulation. In the case of SFR, the SIP messages are manipulated to avoid the overhead of re-assembling fragmented UDP packets. This is achieved by removing Avaya proprietary and unnecessary headers to reduce the SIP messages to below the Maximum Transmission Unit (MTU) so that fragmentation does not occur.

To define the signalling rule, navigate to **Domain Policies →Signalling Rules** in the main menu on the left hand side. Click on **Add** and enter details in the Signalling Rule pop-up box

- In the **Rule Name** field enter a descriptive name for the signalling rule to remove Avaya proprietary and unnecessary headers and click **Next** and **Next** again, then **Finish** (not shown).



Select the **Request Headers** tab and define the rules to remove Avaya proprietary headers as follows:

- Click on **Add In Header Control** (not shown)
- Check the **Proprietary Request Header** box
- Enter the name of the header to be removed in the **Header Name** field

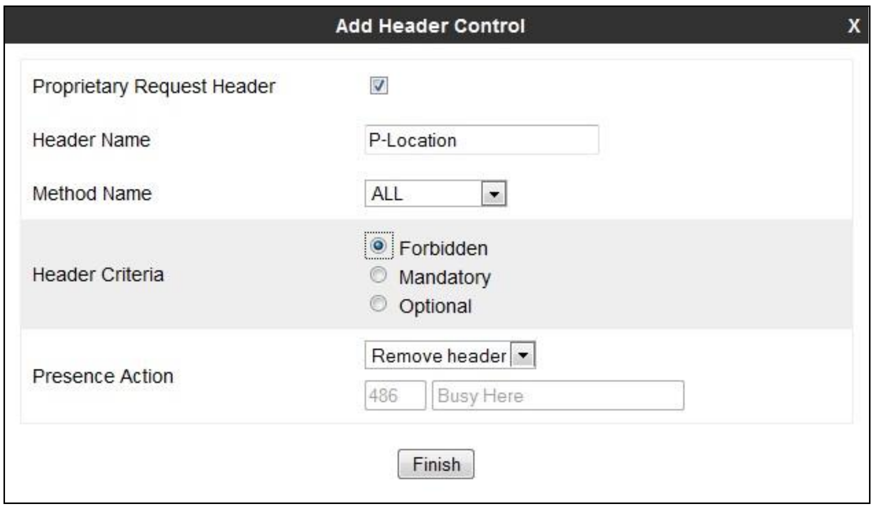
Rules to remove unnecessary headers are slightly different

- Click on **Add In Header Control** (not shown)
- Select the name of the header to be removed in the **Header Name** field

In both cases, the following steps are required:

- Select **ALL** in the Method Name field
- Check **Forbidden** in the Header Criteria options
- In the **Presence Action** drop down menu, select **Remove Header**
- Click **Finish**

The following example shows configuration for removal of **P-Location** headers from request messages.



Note: The above is an example of a proprietary header. During test, the same was done for Accept, Alert-Info, AV-Global-Session-ID, Endpoint-View, P-AV-Message-ID and P-Charging-Vector.

When finished, all the Request Headers defined will be shown under the Request Headers tab as shown in the screenshot.

The same is required for Response headers. Select the **Response Headers** tab and define the rules to remove proprietary headers as follows:

- Click on **Add In Header Control** (not shown)
- Check the **Proprietary Request Header** box
- Enter the name of the header to be removed in the **Header Name** field

As described for request headers, the process to remove unnecessary headers is slightly different:

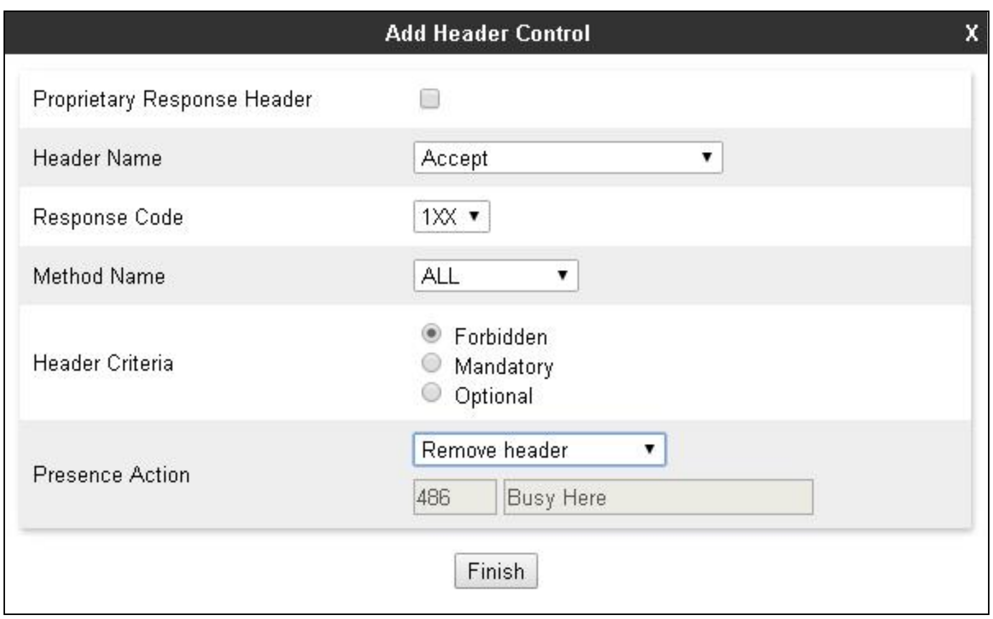
- Click on **Add In Header Control** (not shown)
- Select the name of the header to be removed from the **Header Name** drop down menu

The following steps are required in both cases:

- Select **1XX** in the **Response Code** drop down menu, this will remove the header from 183 Session Progress and 180 Ringing messages.
- Select **ALL** in the Method Name field
- Check **Forbidden** in the Header Criteria options
- In the **Presence Action** drop down menu, select **Remove Header**
- Click **Finish**

Repeat above process and select **2XX** in the **Response Code** so that the header is removed from 200 OK messages.

The following example shows configuration for removal of **Accept** headers from 1XX responses.



Note: The previous screenshot shows an example of an unnecessary header. During test, the same was done for Alert-Info, AV-Global-Session-ID, Endpoint-View, P-AV-Message-ID, P-Charging-Vector and P-Location.

When finished, all the Response Headers defined will be shown under the Response Headers tab as shown in the screenshot.

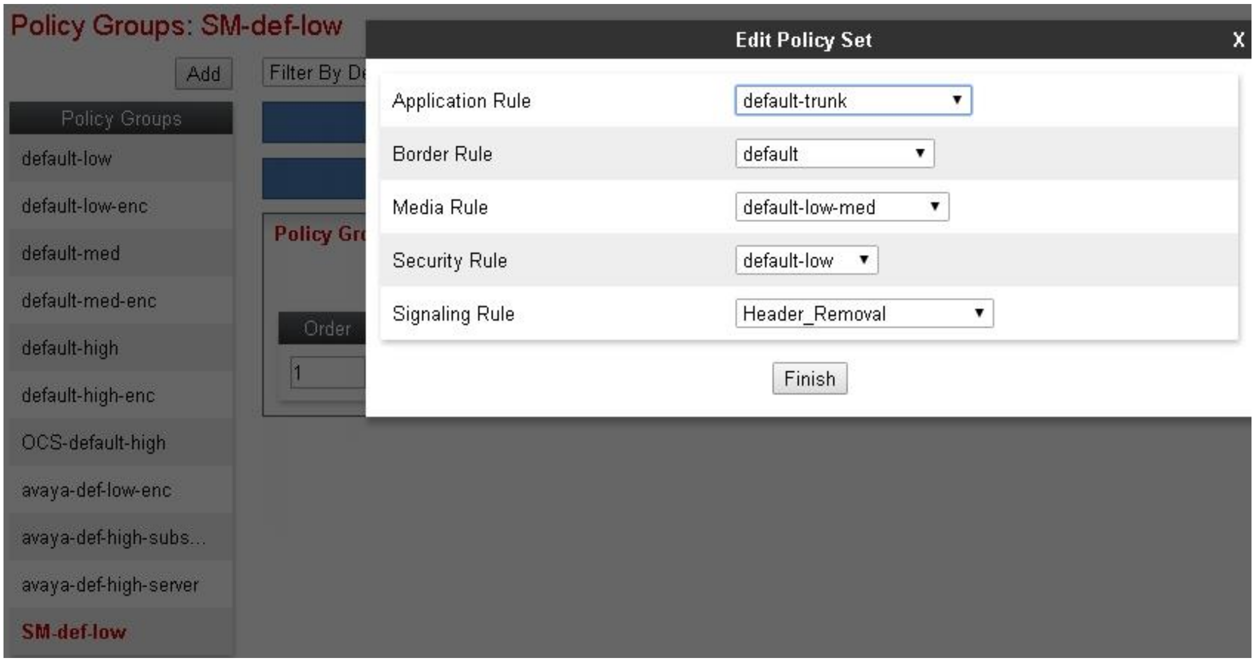
General | Requests | Responses | Request Headers | **Response Headers** | Signaling QoS | UCID

Add In Header Control | Add Out Header Control

Row	Header Name	Response Code	Method Name	Header Criteria	Action	Proprietary	Direction		
1	Accept-Language	1XX	ALL	Forbidden	Remove Header	No	IN	Edit	Delete
2	Accept-Language	2XX	ALL	Forbidden	Remove Header	No	IN	Edit	Delete
3	Alert-Info	1XX	ALL	Forbidden	Remove Header	No	IN	Edit	Delete
4	Alert-Info	2XX	ALL	Forbidden	Remove Header	No	IN	Edit	Delete
5	Av-Global-Session-ID	1XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
6	Av-Global-Session-ID	2XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
7	Endpoint-View	1XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
8	Endpoint-View	2XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
9	P-AV-Message-Id	1XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
10	P-AV-Message-Id	2XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
11	P-Charging-Vector	1XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
12	P-Charging-Vector	2XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
13	P-Location	1XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
14	P-Location	2XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete

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

- In the **Group Name** field enter a descriptive name for the Session Manager Policy Group, in this case **SM-def-low**, and click **Next**
- Leave the **Application Rule**, **Border Rule**, **Media Rule** and **Security Rule** fields at their default values
- In the **Signaling** drop down menu, select the recently added signalling rule for the Session Manager (**Header_Removal**)

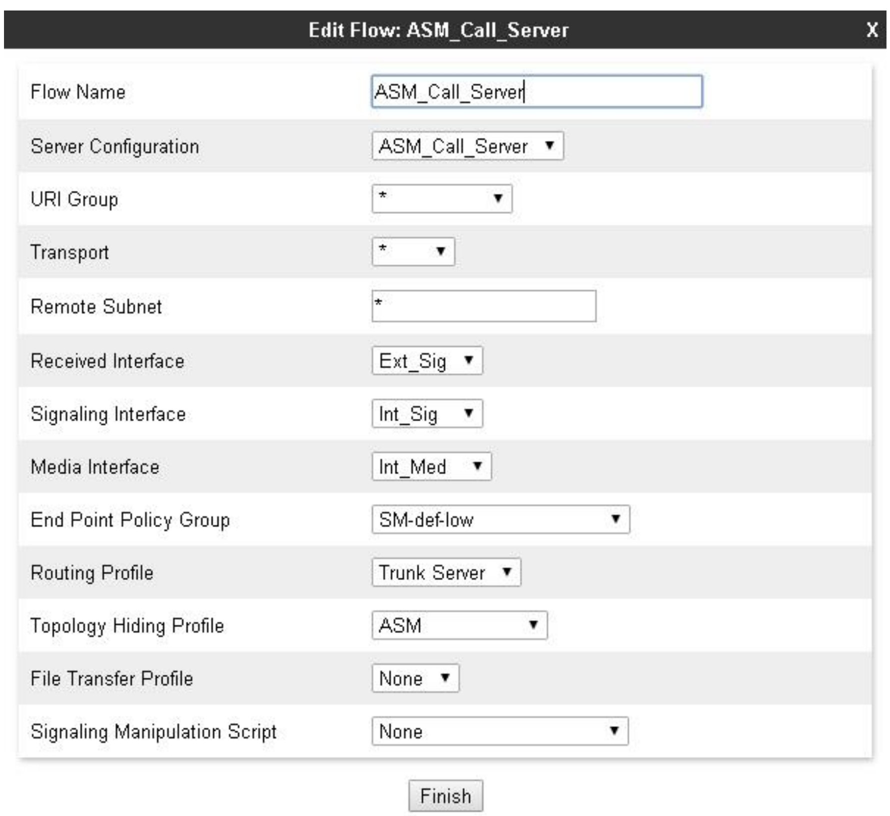


7.9. Server Flows

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

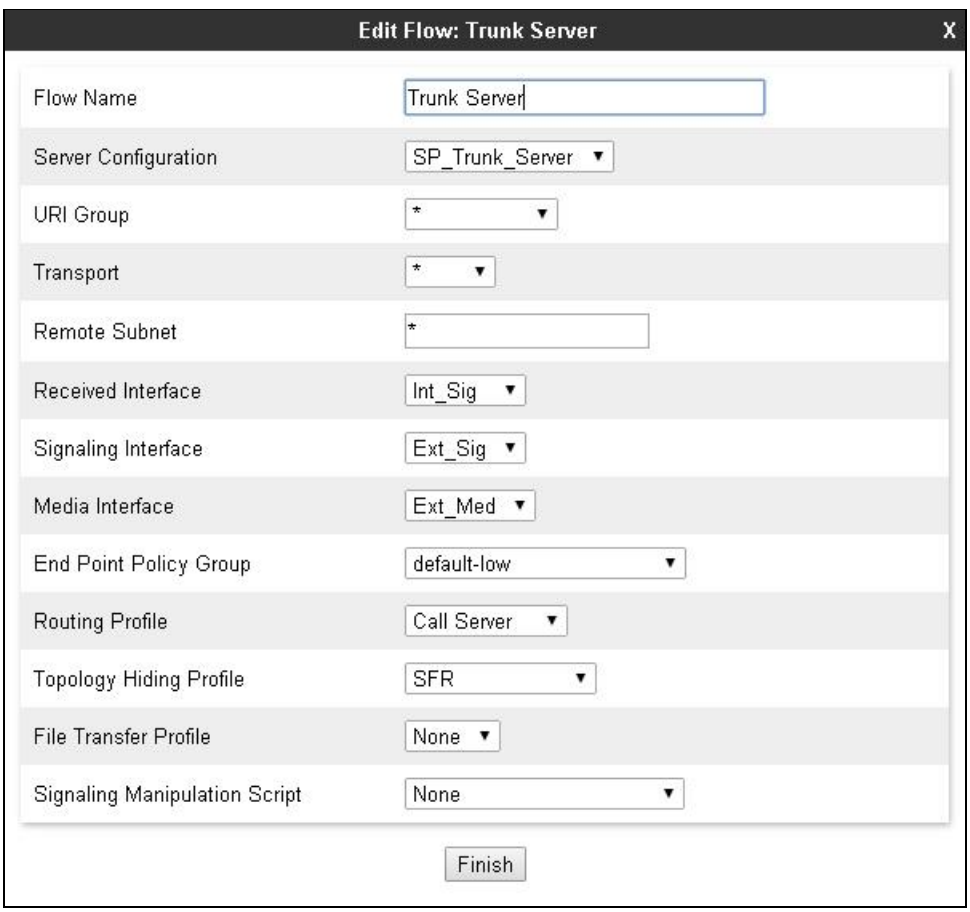
To define a Server Flow for the Session Manager, navigate to **Device Specific Settings → End Point Flows**.

- Click on the **Server Flows** tab.
- Select **Add Flow** and enter details in the pop-up menu.
- In the **Name** field 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 the SFR SIP Trunk defined in **Section 7.6**.
- In the **End Point Policy Group** drop down menu, select the End Point Policy Group that contains the Signalling Rules for the Session Manager defined in **Section 7.8**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of the Session Manager defined in **Section 7.7** and click **Finish**.

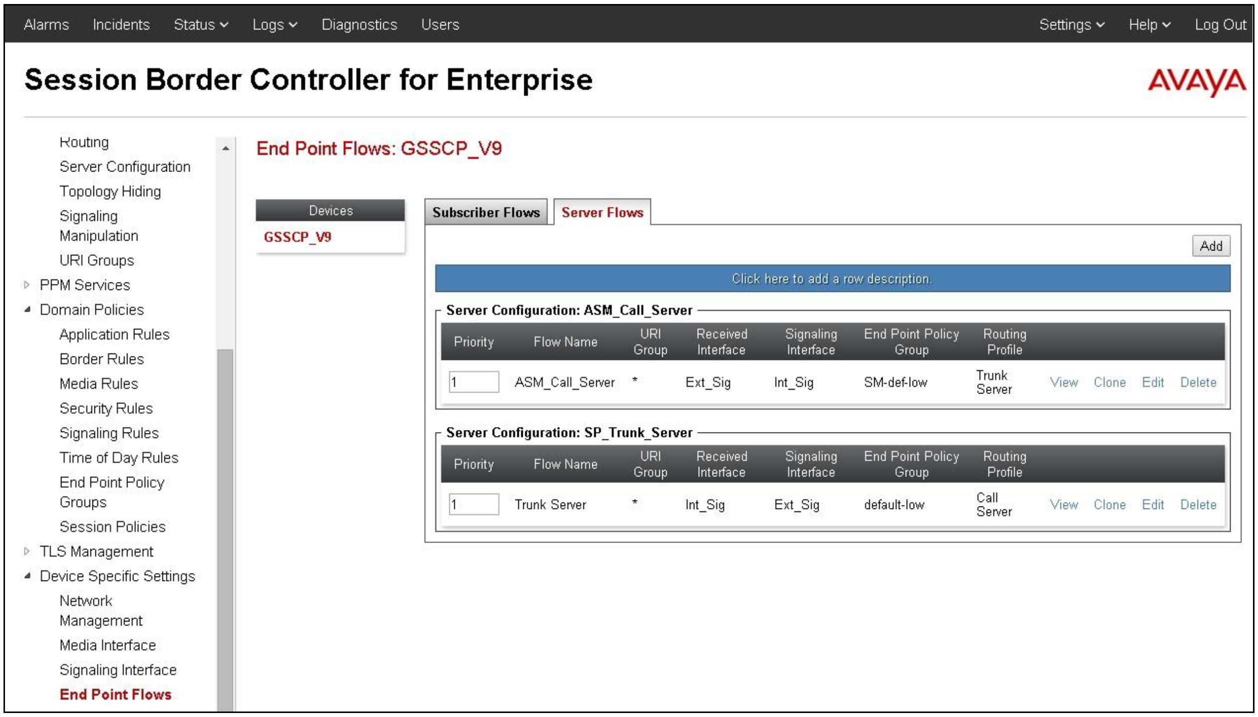


To define a Server Flow for SFR Collecte SIP, navigate to **Device Specific Settings → End Point Flows**.

- 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 SFR Collecte SIP, 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 SFR Collecte SIP 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 SFR Collecte SIP 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 SFR Collecte SIP is sent on.
- In the **Routing Profile** drop-down menu, select the routing profile of the Session Manager defined in **Section 7.6**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of the SFR Collecte SIP defined in **Section 7.7** and click **Finish**.



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



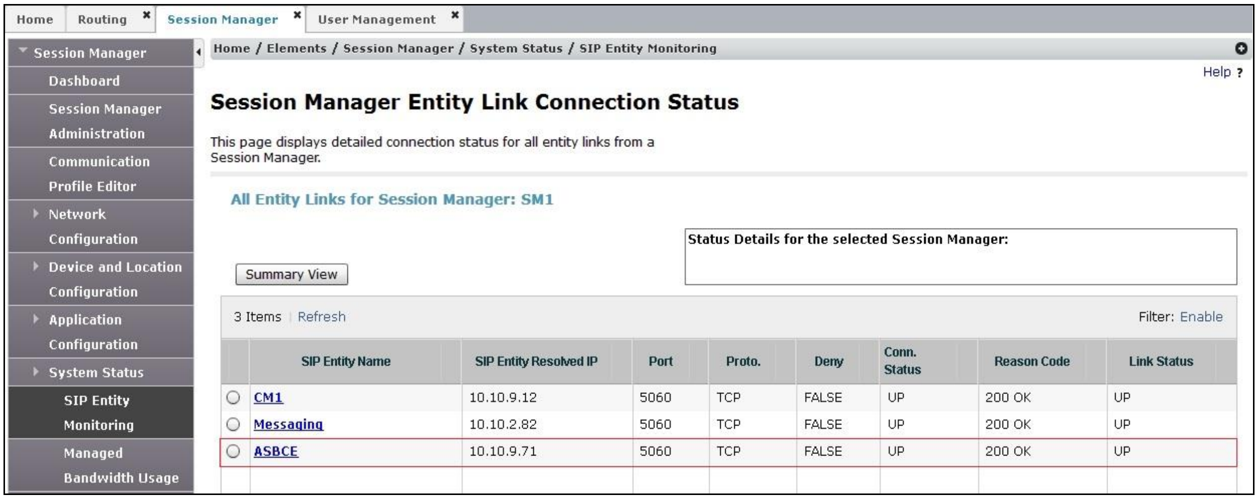
8. Configure SFR Collecte SIP Equipment

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



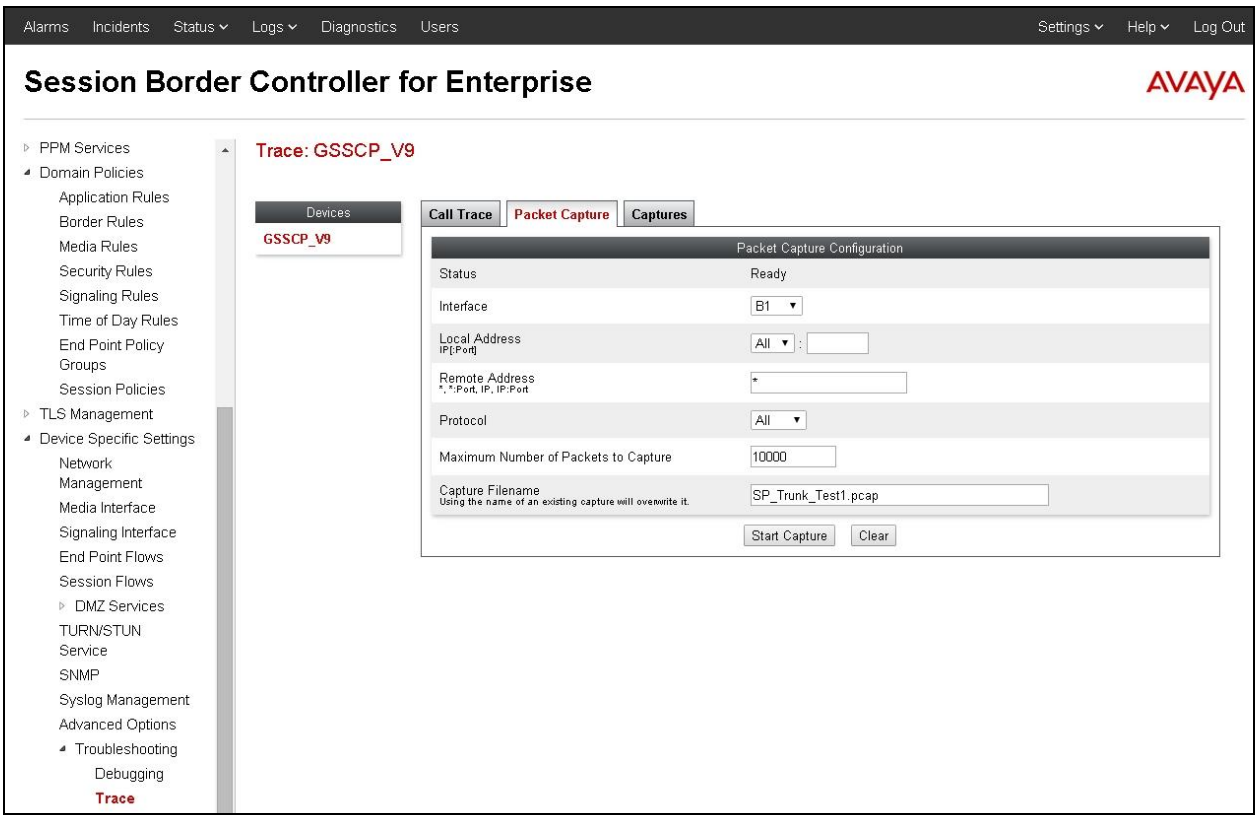
2. From the Communication Manager SAT interface run the command **status trunk n** where **n** is a previously configured SIP trunk. Observe if all channels on the trunk group display **in-service/idle**.

```
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
```

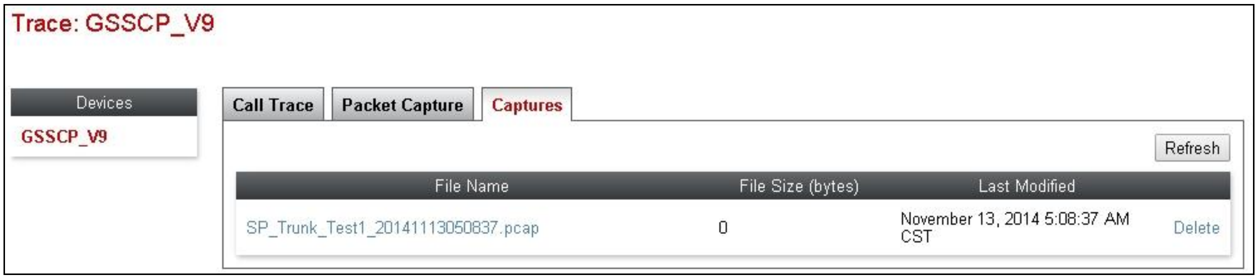
3. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active.
4. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active.
5. Verify that the user on the PSTN can end an active call by hanging up.
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 SFR 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 SFR Collecte SIP service. SFR Collecte SIP Service is a SIP-based Voice over IP solution providing businesses a flexible, cost-saving alternative to traditional hardwired telephony trunks. The service was successfully tested with a number of observations listed in **Section 2.2**.

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.