Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Communication Manager R7.0, Avaya Aura® Session Manager R7.0 and Avaya Session Border Controller for Enterprise R7.0 to support Proximus SIP Trunking Service - Issue 1.0

Abstract

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between Proximus SIP Trunking service and an Avaya SIP enabled Enterprise Solution. The Avaya solution consists of Avaya Session Border Controller for Enterprise, Avaya Aura® Session Manager and Avaya Aura® Communication Manager as an Evolution Server. Proximus, previously Belgacom, is a member of the DevConnect Service Provider program.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in Section 2.1 as well as any observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program 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 Proximus SIP Trunking service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of the following: Avaya Aura® Communication Manager R7.0 (Communication Manager); Avaya Aura® Session Manager R7.0 (Session Manager); Avaya Session Border Controller for Enterprise R7.0 (Avaya SBCE). Note that the shortened names shown in brackets will be used throughout the remainder of the document. Customers using this Avaya SIP-enabled enterprise solution with the Proximus SIP Trunking service 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 connect to the Proximus SIP Trunking service.

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

2.1. Interoperability Compliance Testing

The interoperability test included the following:

- Incoming calls to the enterprise site from PSTN phones using the Proximus SIP Trunking service, calls made to SIP and H.323 telephones at the enterprise.
- Outgoing calls from the enterprise site completed via the Proximus SIP Trunking service to PSTN destinations, calls made from SIP and H.323 telephones.
- Calls using the G.711A and G.729A codecs.
- Fax calls to/from a Group 3 fax machine to a PSTN connected fax machine using T.38.
- DTMF transmission using RFC 2833 with successful Voice Mail/Vector navigation for inbound and outbound calls.
- User features such as hold and resume, transfer, conference, call forwarding, etc.
- Caller ID Presentation and Caller ID Restriction.
- Direct IP-to-IP media between the Avaya SBCE and the SIP and H.323 telephones.
- Call coverage and call forwarding for endpoints at the enterprise site.
- Transmission and response of SIP OPTIONS messages sent by the Proximus SIP Trunking service requiring Avaya response and sent by Avaya requiring Proximus response.

2.2. Test Results

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

- Outbound calls failed with Initial IP-IP Direct Media set to "y" on Communication Manager. This was because of an Avaya proprietary parameter "+Avaya-cm-keep-mpro=no" in the Contact header. This was removed using a Sigma script in Avaya SBCE.
- When there was no matching codec between Communication Manager and the network for an outgoing call, the network returned 482 "Merged Request". Though the call failed as expected and a tone was heard on the calling phone, it is listed as an observation because the commonly used response is 488 "Not Accepted Here".
- Inbound access from Toll-Free numbers was not tested as it is not part of the Proximus service.
- Outbound international access was not tested as it is not available from the Proximus Lab environment.
- Emergency calls were not tested as a test call was not booked with the Emergency Services Operator.
- Operator and Directory Enquiries were not tested as the short numbers were not available from the Proximus Lab environment
- When forwarding calls off-net, the network did not respond to the SIP INVITE message for leg 2. This was resolved by removing the "Recv-Info" header from the INVITE using a Session Manager Adaptation.
- A number of errors were observed when sending fax, this was retested with ECM turned Without ECM, the fax transmission was faster and a visual inspection showed that the fax was of satisfactory quality.
- When calling a Communication Manager extension DDI from a number configured as EC500 for another extension, no ringback was heard. Due to Lab equipment constraints, these calls were made from a VoIP user. As the signalling was correct, this was considered to be a fault in the VoIP network and not an interworking issue.
- When transferring an outbound PSTN call on a one-X® Communicator softphone to an internal extension, a failure announcement was heard from the PSTN as well as ringing from the extension. The call was successfully transferred.
- When making outbound calls using one-X® Communicator in "Other phone" mode, no ringback was heard when the "Other Phone" was a Communication Manager H.323 extension forwarded from a Belgian national number. This was considered to be specific to the test environment. During this test, the Alert-Info header was removed using an SM Adaptation.
- During testing of Consultative transfer to internal extension by Avaya one-X® Communicator in "Other Phone" Mode, audio was lost to the VoIP user being used as the "Other phone". This was a fault with the VoIP user as there was also no audio when ringing from a PSTN phone. Testing continued with signalling checked to determine whether or not a call was a success, and some tests were repeated successfully with a different type of "Other phone"

- When testing failure of the SIP Trunk, the network sent numerous re-INVITEs in response to error messages from Session Manager. This resulted in 45 seconds of silence before a failure tone was heard.
- During testing, two softphone clients were used that were registered to the Proximus IMS. These were not part of the tested solution, but were used for test cases that required outgoing calls to the PSTN. To make these clients work effectively, **Delayed SDP** was used in the Avaya SBCE configuration to prevent the sending of empty INVITE messages as described in **Section 7.4**. This resolved an issue where Communication Manager cleared calls that were put on hold. When there was no SDP in the INVITE, the 200 OK from the soft clients included an SDP that was not accepted by Communication Manager.

2.3. Support

For technical support on Proximus products please contact Proximus on 0800 55200 or visit their website at http://www.proximus.be/en/id_zwpl_s/large-companies-and-public-sector/support.html

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an Enterprise site connected to the Proximus SIP Trunking service. Located at the Enterprise site is an Avaya SBCE, Session Manager and Communication Manager. Endpoints are Avaya 96x0 series and Avaya 96x1 series IP deskphones (with SIP and H.323 firmware), Avaya 16xx series IP deskphones (with H.323 firmware), Avaya analogue telephones and an analogue fax machine. Also included in the test configuration was an Avaya one-X® Communicator soft phone and Avaya Communicator for Windows running on laptop PCs.

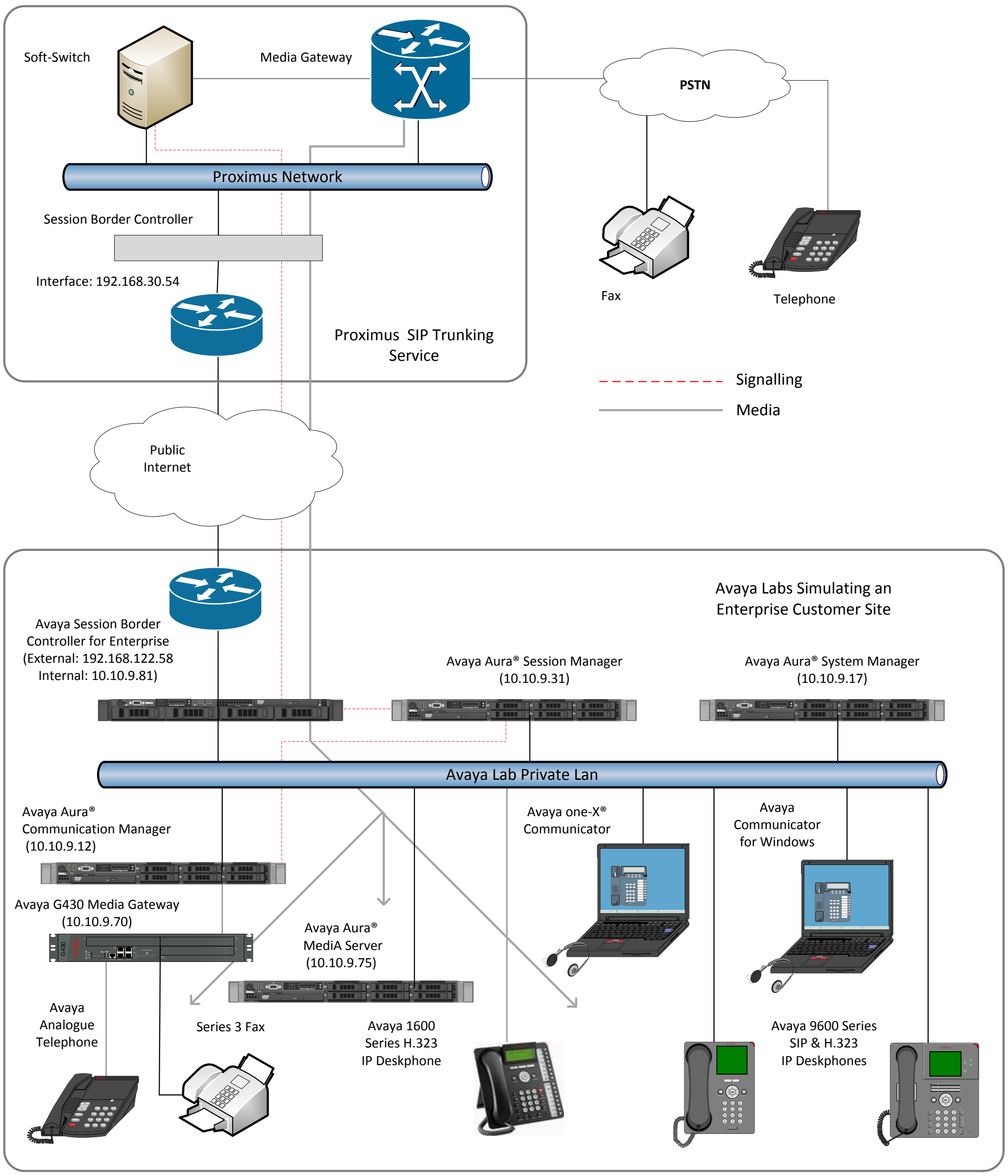


Figure 1: Test Setup Proximus SIP Trunking service to Avaya Enterprise

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya	
Avaya Aura® Session Manager	7.0.0.1.700102
Avaya Aura® System Manager	7.0.0.1.4212
Avaya Aura® Communication Manager	7.0-441 Build 0.22684
Avaya Session Border Controller for Enterprise	7.0.0-21-6602 Patch sbc700-p001-20151005-7.0.0-21.x86_64.rpm
Avaya G430 Media Gateway	37.19.0
Avaya Aura® Media Server	7.7.0.236_2015.07.24
Avaya 96x0 Deskphone (SIP)	2_6_14_5
Avaya 9608 Deskphone (SIP)	7.0.0 R39
Avaya 96x0 Deskphone (H.323)	3.230A
Avaya 9608 Deskphone (H.323)	6.3116
Avaya 1616 Deskphone (H.323)	1.380B
Avaya One-X Communicator	6.2.7.03-SP7
Avaya Communicator for Windows	2.1.2.75
Avaya 2400 Series Digital Handsets	N/A
Analogue Handset	N/A
Analogue Fax	N/A
Proximus	
Alcatel-Lucent IMS Solution	Version 10.1

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 Proximus SIP Trunking Service. For incoming calls, Session Manager receives SIP messages from the Avaya SBCE and directs the incoming SIP messages to Communication Manager. Once the message arrives at Communication Manager further incoming call treatment, such as incoming digit translations and class of service restrictions may be performed. All outgoing calls to the PSTN are processed within Communication Manager and may be first subject to outbound features such as automatic route selection, digit manipulation and class of service restrictions. Once Communication Manager selects a SIP trunk, the SIP signalling is routed to Session Manager. Session Manager directs the outbound SIP messages to the Avaya SBCE at the enterprise site that then sends the SIP messages to the Proximus 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 authorised 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 Proximus SIP Trunking service and any other SIP trunks used.

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

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

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

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

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

           IP Attendant Consoles? y
```

5.2. Administer IP Node Names

The node names defined here will be used in other configuration screens to define a SIP signalling group between Communication Manager and Session Manager. In the **IP Node Names** form, assign the node **Name** and **IP Address** for Session Manager. In this case, **Session_Manager** and **10.10.9.31** are the **Name** and **IP Address** for the Session Manager SIP interface. Also note the **procr** IP address as this is the processor interface that Communication Manager will use as the SIP signalling interface to Session Manager.

```
display node-names ip
                                  IP NODE NAMES
    Name              IP Address
AMS                 10.10.9.75
Session_Manager     10.10.9.31
default             0.0.0.0
procr               10.10.9.12
procr6              ::
```

5.3. Administer IP Network Region

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

- The **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is **avaya.com**.
- By default, **IP-IP Direct Audio** (both **Intra**- and **Inter-Region**) is enabled (**yes**) to allow audio traffic to be sent directly between endpoints without using gateway VoIP resources. When a PSTN call is shuffled, the media stream is established directly between the enterprise end-point and the internal media interface of the Avaya SBCE.
- The **Codec Set** is set to the number of the IP codec set to be used for calls within the IP network region. In this case, codec set **2** is used.
- The rest of the fields can be left at default values.

```
change ip-network-region 2                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 2
Location:          Authoritative Domain: avaya.com
    Name: Trunk                   Stub Network Region: n
MEDIA PARAMETERS                  Intra-region IP-IP Direct Audio: yes
      Codec Set: 2                Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                        IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                        RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

Note: In the test configuration, ip-network-region 1 was used within the enterprise and ip-network-region 2 was used for the SIP Trunk. In the configuration of the G430 (not shown) ip-network-region 1 was selected so that the G430 is used for calls within the enterprise and for analogue and digital endpoints. In the configuration of the Avaya Media Server (not shown), ip-network-region 2 was selected so that the Avaya Media Server (AMS) is used for the SIP Trunk.

5.4. Administer IP Codec Set

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

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

                          IP CODEC SET
    Codec Set: 2

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

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

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

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

                          IP CODEC SET

                              Allow Direct-IP Multimedia? n




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

Note: Fax was also successfully tested with G.711 fallback (**t.38-G711-fallback**); however Communication Manager always changes to G.711 on incoming faxes.

During testing, transmission of fax was unreliable due to network issues. Retesting was carried out with **ECM** set to **n** as described in **Section 2.2**. Fax transmission was faster and there were no visible quality issues.

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

5.5. Administer SIP Signaling Groups

This signalling group (and trunk group) will be used for inbound and outbound PSTN calls to the Proximus SIP Trunking service. During test, this was configured to use TCP and port 5062 though it's recommended to use TLS and port 5061 in the live environment to enhance security. Configure the **Signaling Group** using the **add signaling-group n** command as follows:

- Set **Group Type** to **sip**.
- Set **Transport Method** to **tcp**.
- Set **Peer Detection Enabled** to **y** allowing Communication Manager to automatically detect if the peer server is a Session Manager.
- Set **Near-end Node Name** to the processor interface (node name **procr** as defined in the **IP Node Names** form shown in **Section 5.2**).
- Set **Far-end Node Name** to the Session Manager (node name **Session_Manager** as defined in the **IP Node Names** form shown in **Section 5.2**).
- Set **Near-end Listen Port** and **Far-end Listen Port** as required. The standard value for TCP is **5060**, though **5062** was used in test to separate the SIP Trunk from the SIP endpoints on the Session Manager (See **Section 6.5**).
- 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 **2**).
- 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**.
- Set **Initial IP-IP Direct Media** and **H.323 Station Outgoing Direct Media** to **y**. This initiates direct media when the call is set up without the need for shuffling.
- Leave **DTMF over IP** at default value of **rtp-payload** (Enables **RFC2833** for DTMF transmission from Communication Manager).

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

 Group Number: 2              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n
     IP Video? n                                 Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                  Far-end Node Name: Session_Manager
 Near-end Listen Port: 5062                 Far-end Listen Port: 5062
                                         Far-end Network Region: 2

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

Note: The default values for the other fields may be used.

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 n** command, where **n** is an available trunk group for the SIP Trunk. On **Page 1** of this form:

- Set the **Group Type** field to **sip**.
- Choose a descriptive **Group Name**.
- Specify a trunk access code (**TAC**) consistent with the dial plan.
- The **Direction** is set to **two-way** to allow incoming and outgoing calls.
- Set the **Service Type** field to **public-ntwrk**.
- 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 2                                              Page   1 of  21
                                TRUNK GROUP

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

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

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

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 5000

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

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

On **Page 3**, set the **Numbering Format** field to **private**. This allows delivery of CLI in formats other than E.164 with leading "+". In test, CLIs were sent as Communication Manager extension numbers and were reformatted by Session Manager in an Adaptation described in **Section 6.4**. This format was successfully verified in the network.

```
add trunk-group 2                                                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** and **Support Request History** to **n** as these headers are not supported in the Proximus SIP trunk.
- Set the **Telephone Event Payload Type** to **101** to match the value preferred by Proximus (this Payload Type is not applied to calls from SIP end-points).
- Set the **Identity for Calling Party Display** to **From** to ensure that where CLI for incoming calls is withheld, it is not displayed on Communication Manager extension.

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

                                       Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                       Send Transferring Party Information? n
                                  Network Call Redirection? y
          Build Refer-To URI of REFER From Contact For NCR? n
                                    Send Diversion Header? n
                                  Support Request History? n
                             Telephone Event Payload Type: 101


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

Note: - The above screenshot shows **Network Call Redirection** set to **y**. This was temporarily set to **y** for some of the last tests that involved testing of 302 Moved Temporarily and REFER messages. When set, REFER messages are sent that are not acted on by the Proximus SIP Trunking service and so are unnecessary additional signalling.

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 numbers were sent as Communication Manager extension numbers to be modified in Session Manager. Adaptations are used in Session Manager to format the number as described in **Section 6.4**. These calling party numbers are sent in the SIP From, Contact and PAI headers. The numbers are displayed on display-equipped PSTN telephones with any reformatting performed in the network.

```
change private-numbering 0                                      Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext              Trk        Private          Total
Len Code             Grp(s)     Prefix           Len
 4  2                1-2                         4     Total Administered: 2
                                                          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 the Proximus SIP Trunking service. 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: 8
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
```

Use the **change ars analysis** command to configure the routing of dialled digits following the first digit 9. A small sample of dial patterns are shown here as an example. Further administration of ARS is beyond the scope of this document. The example entries shown will match outgoing calls to numbers beginning 0. Note that exact maximum number lengths should be used where possible to reduce post-dial delay. Calls are sent to **Route Pattern 2**.

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

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    0                     8    12     2       pubu         n
    00                    13   15     2       pubu         n
    1                     3    3      2       pubu         n
    118                   5    6      2       pubu         n
    7000                  4    4      1       pubu         n
```

Use the **change route-pattern n** command, where **n** is an available route pattern, to add the SIP trunk group to the route pattern that ARS selects. In this configuration, route pattern **2** is used to route calls to trunk group **2**. **Numbering Format** is applied to CLI and is used to set TDM signalling parameters such as type of number and numbering plan indicator. This doesn't have the same significance in SIP calls and during testing it was set to **lev0-pvt** to ensure that calling party number was not prefixed with a leading "+".

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

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

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM Sub  Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
 1: y y y y y n  n            rest                               lev0-pvt  none
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
 6: y y y y y n  n            rest                                         none
```

5.9. Administer Incoming Digit Translation

This step configures the settings necessary to map incoming DDI calls to Communication Manager extensions. The incoming digits sent in the INVITE message from Proximus can be manipulated as necessary to route calls to the desired extension. During test, the incoming DDI numbers were changed in Session Manager to Communication Manager extension numbers using an Adaptation as 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 2                     Page   1 of   3
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 public-ntwrk
 public-ntwrk
```

Note: One reason for configuring the enterprise in this way is to allow the use of the extension number as a common identifier with other network elements within the enterprise such as voice mail.

5.10. EC500 Configuration

When EC500 is enabled on a Communication Manager station, a call to that station will generate a new outbound call from Communication Manager to the configured EC500 destination, typically a mobile phone. The following screen shows an example EC500 configuration for the user with station extension 2391. Use the command **change off-pbx-telephone station-mapping x** where **x** is Communication Manager station.

- The **Station Extension** field will automatically populate with station extension.
- For **Application** enter **EC500**.
- Enter a **Dial Prefix** if required by the routing configuration.
- For the **Phone Number** enter the phone that will also be called (e.g. **02797nnnn**).
- Set the **Trunk Selection** to **ars** so that the ARS table will be used for routing.
- Set the **Config Set** to **1**.

```
change off-pbx-telephone station-mapping 2391                   Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 2391             EC500        -     02797nnnn       ars        1
```

Note: The phone number shown is for a VoIP user registered with the Proximus IMS system independently of the test environment. To use facilities for calls coming in from EC500 mobile phones, the calling party number received in Communication Manager must exactly match the number specified in the above table.

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

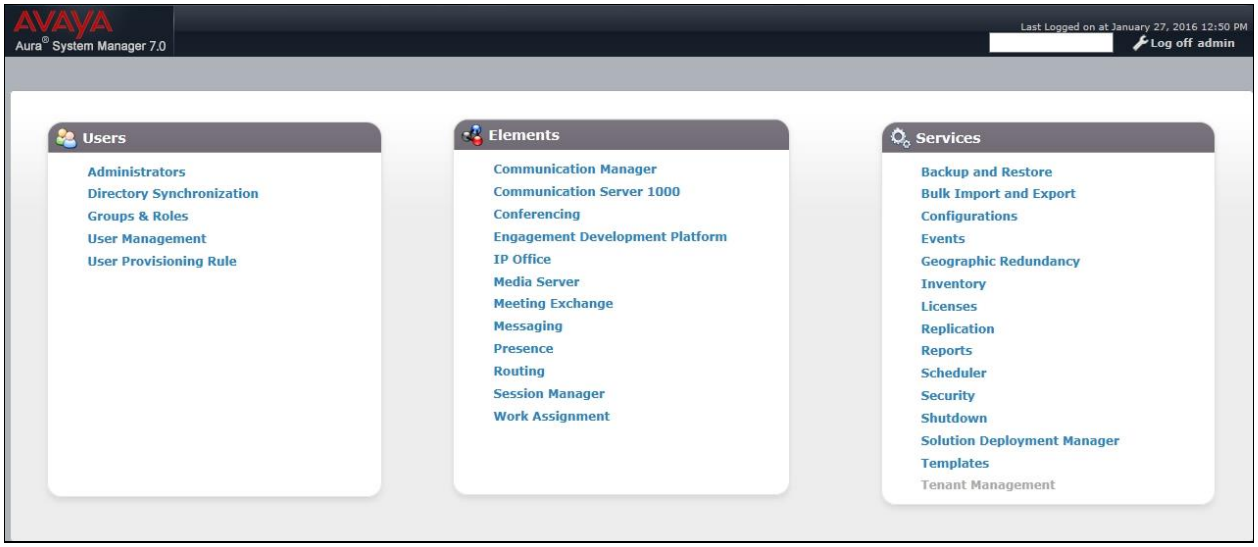
6. Configuring Avaya Aura® Session Manager

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

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

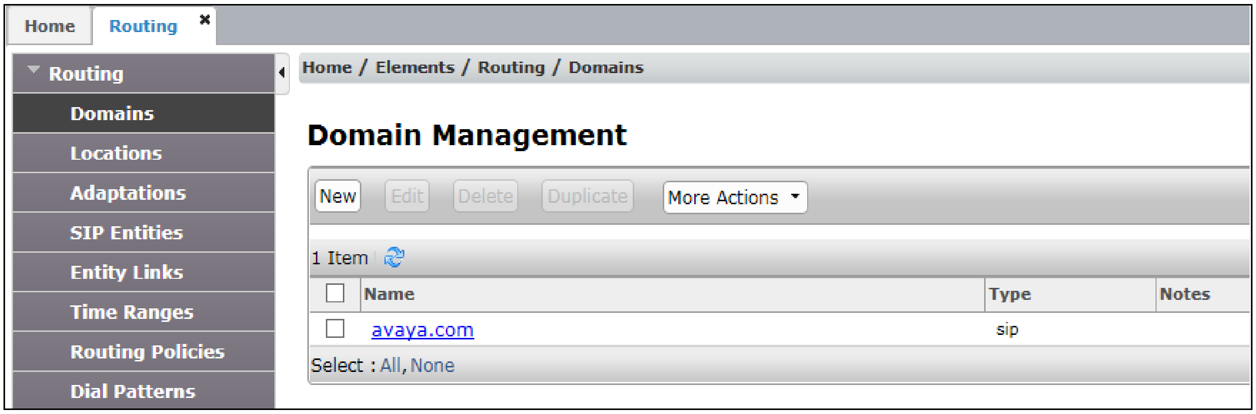
6.1. Log in to Avaya Aura® System Manager

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



6.2. Administer SIP Domain

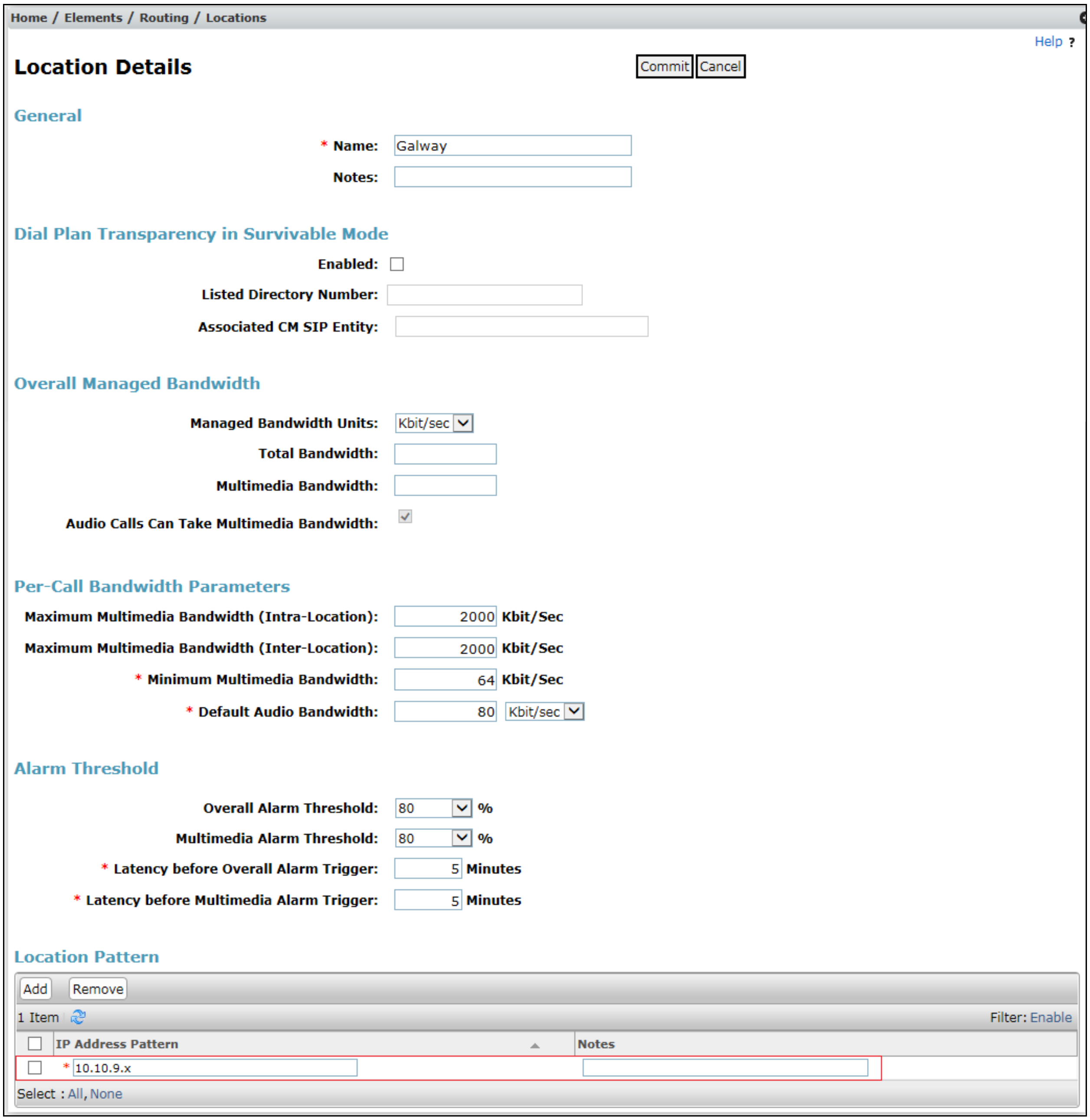
To add the SIP domain that will be used with Session Manager, select **Routing** from the **Home** tab menu and in the resulting tab select **Domains** from left hand menu. Click the **New** button to create a new SIP domain entry. In the **Name** field enter the domain name of the enterprise site or a name agreed with Proximus; 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, Topology Hiding in the Avaya SBCE can be used to change it (see **Section 7.8**).

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 (not shown). Under **General**, in the **Name** field, enter an informative name for the location. 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.

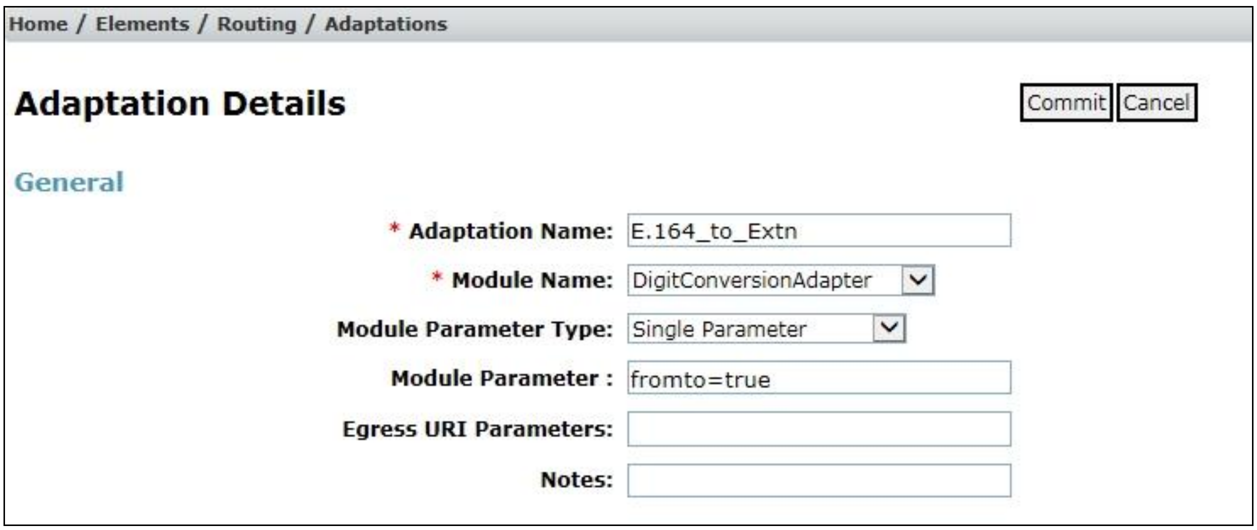


6.4. Administer Adaptations

Calls from Proximus are received at the enterprise in E.164 format with leading "+" on the Request URI. An Adaptation specific to Communication Manager is used to convert the called party number to a pre-defined extension number before onward routing to the Communication Manager SIP Entity, removing the requirement for incoming digit manipulation on Communication Manager.

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

- In the **Adaptation name** field, enter a descriptive title for the adaptation.
- In the **Module name** drop down menu, select **DigitConversionAdapter**. This is used for simple digit conversion adaptations.
- In the **Module Parameter Type** drop down menu, select **Single Parameter**.
- In the Module Parameter box (not shown), type **fromto=true.** 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 Outgoing Calls from SM**, click on **Add**. An additional row will appear (not shown). 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 E.164l format to the extension number for termination of calls on Communication Manager. In addition, the calling party number is adapted to diallable format for display on Communication Manager extensions.

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 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 Outgoing Calls from SM

Add Remove

12 Items — Filter: Enable

Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
+	11	15		1	00	origination		
+32	11	11		3	0	origination		
+32279nnnn0	11	11		11	2000	destination		
+32279nnnn1	11	11		11	2391	destination		
+32279nnnn2	11	11		11	2291	destination		
+32279nnnn3	11	11		11	2316	destination		
+32279nnnn4	11	11		11	2400	destination		
+32279nnnn5	11	11		11	2401	destination		
+32279nnnn6	11	11		11	7000	destination		
+32279nnnn7	11	11		11	6002	destination		
+32279nnnn8	11	11		11	2290	destination		
+32279nnnn9	11	11		11	2396	destination		

Select : All, None

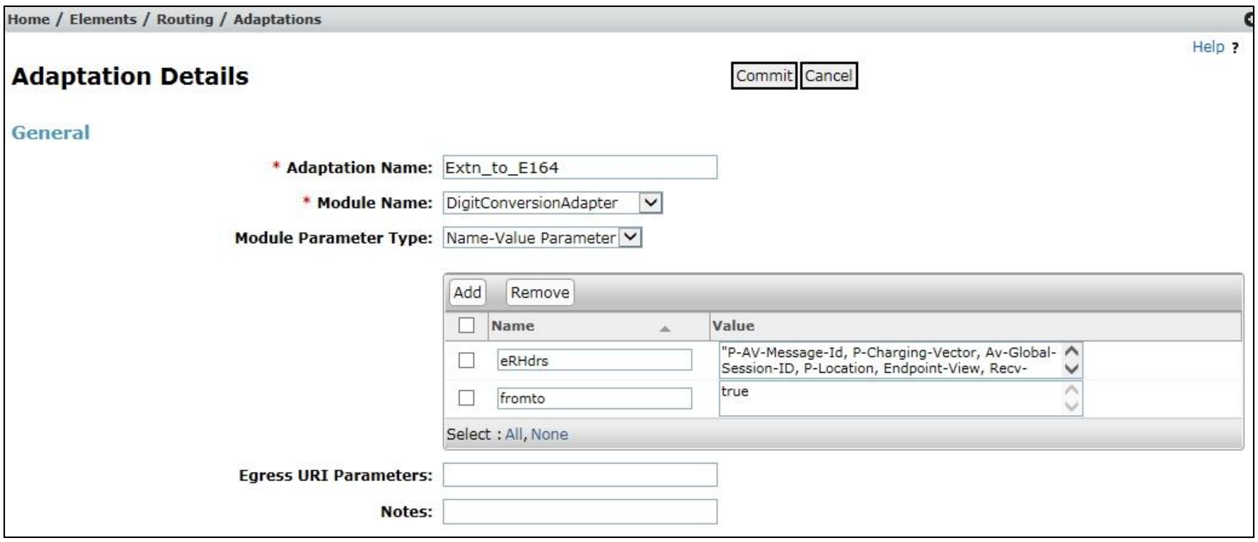
Commit Cancel

Note: In the above screenshots the DDI numbers are partially obscured. If the number is to be changed to diallable format for display on Communication Manager extensions, additional rows may be required. These would replace a leading "+" with "00" for international calling party numbers and "+32" would be replaced by "0" for national calling party numbers.

An additional Adaptation is required to convert extension numbers to national format. Calls from Communication Manager are received at Session Manager with the extension number in the From header. An Adaptation specific to Proximus is used to convert the calling party number to national format with no leading "0" before onward routing to the Proximus SIP Trunking service. This Adaptation is also used to remove unnecessary and Avaya proprietary headers from SIP messages outbound to the Proximus SIP trunk.

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

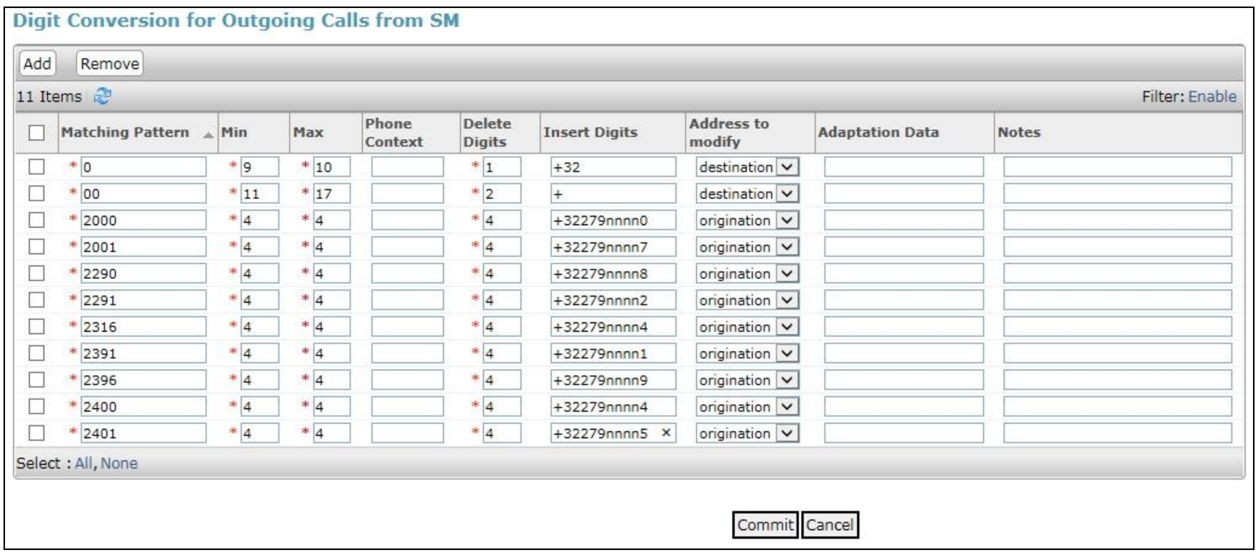
- In the **Adaptation name** field, enter a descriptive title for the adaptation.
- In the **Module name** drop down menu, select **DigitConversionAdapter**. This is used for simple digit conversion adaptations.
- In the **Module parameter Type** drop down menu, select **Name-Value Parameter**.
- In the **Name** field, type eRHdrs to remove SIP headers.
- In the **Value** field, type the names of the headers to be removed. During testing, the headers removed were as follows: **"P-AV-Message-Id, P-Charging-Vector, Av-Global-Session-ID, P-Location, Endpoint-View, Recv-Info, P-Conference, Alert-Info"**.
- Click on **Add** to specify an additional parameter.
- In the **Name** field, type **fromto**.
- In the **Value** field, type **true.** 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 Outgoing Calls from SM**, click on **Add**. An additional row will appear (not shown). This allows information to be entered for the manipulation of numbers coming from Communication Manager. This is where the calling party number is translated from the extension number to E.164 format with leading "+" for display on the terminating PSTN phones as the diallable DDI number assigned to the extension.

The screenshot below shows a translation for each calling party number. This is not normally necessary where the extension number forms part of the national number. When this is the case, a simple additional of the leading digits to build up the national format is required.

- Under **Matching Pattern** enter the extension number as received from Communication Manager.
- 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 remove any digits that will not form part of the national number, 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 E.164 number with leading "+". If the extension number forms part of the DDI number, only the necessary prefix digits will be required.
- Under **Address to Modify** choose **origination** from the drop down box to apply this rule to the From header only.



Digit Conversion for Outgoing Calls from SM

Add Remove

11 Items Filter: Enable

Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
0	9	10		1	+32	destination		
00	11	17		2	+	destination		
2000	4	4		4	+32279nnnn0	origination		
2001	4	4		4	+32279nnnn7	origination		
2290	4	4		4	+32279nnnn8	origination		
2291	4	4		4	+32279nnnn2	origination		
2316	4	4		4	+32279nnnn4	origination		
2391	4	4		4	+32279nnnn1	origination		
2396	4	4		4	+32279nnnn9	origination		
2400	4	4		4	+32279nnnn4	origination		
2401	4	4		4	+32279nnnn5	origination		

Select : All, None

Commit Cancel

Note: In the above screenshots the DDI numbers are partially obscured. Also, the called party number is converted to E.164 format with leading "+". This wasn't strictly necessary during testing as the network is able to do the conversion, but it's shown here for use if required. Add additional lines for destination numbers starting with "0" or "00". If the number starts with "0", it is a national number and the leading "0" must be replaced with "+32". If the number starts with "00", it is an international number and the leading "00" must be replaced with "+".

6.5. Administer SIP Entities

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

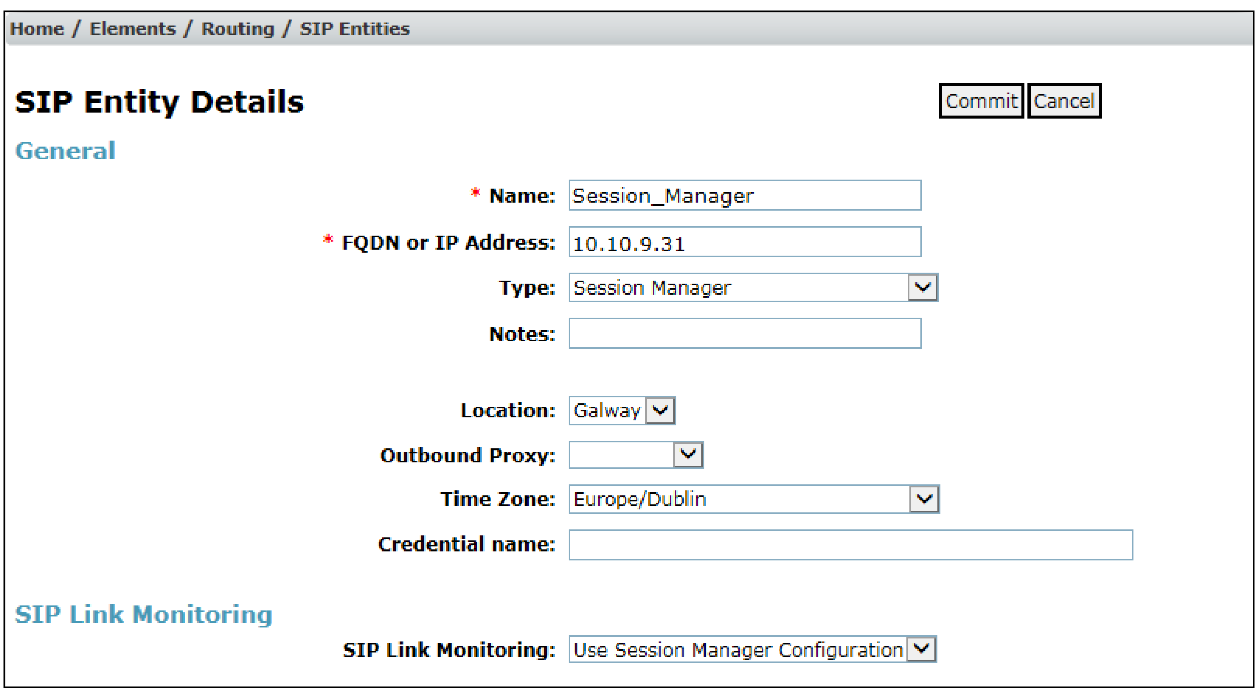
- In the **Name** field enter an informative name.
- In the **FQDN or IP Address** field enter the IP address of Session Manager or the signalling interface on the connecting system.
- In the **Type** field use **Session Manager** for a Session Manager SIP entity, **CM** for a Communication Manager SIP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

In this configuration there are four SIP Entities:

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

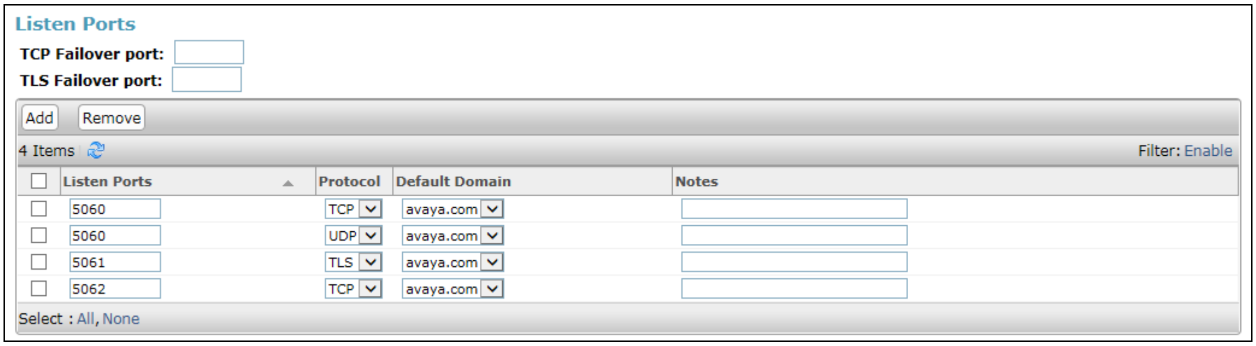
6.5.1. Avaya Aura® Session Manager SIP Entity

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



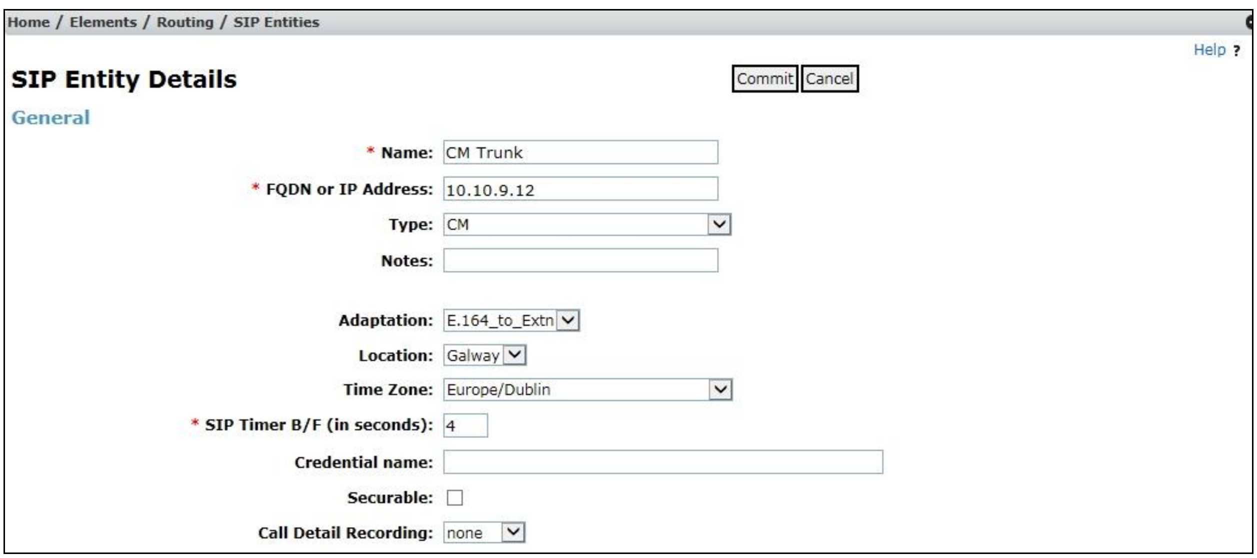
Session Manager must be configured with the port numbers on the protocols that will be used by the other SIP entities. To configure these scroll to the bottom of the page and under **Port**, click **Add**, then edit the fields in the resulting new row.

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

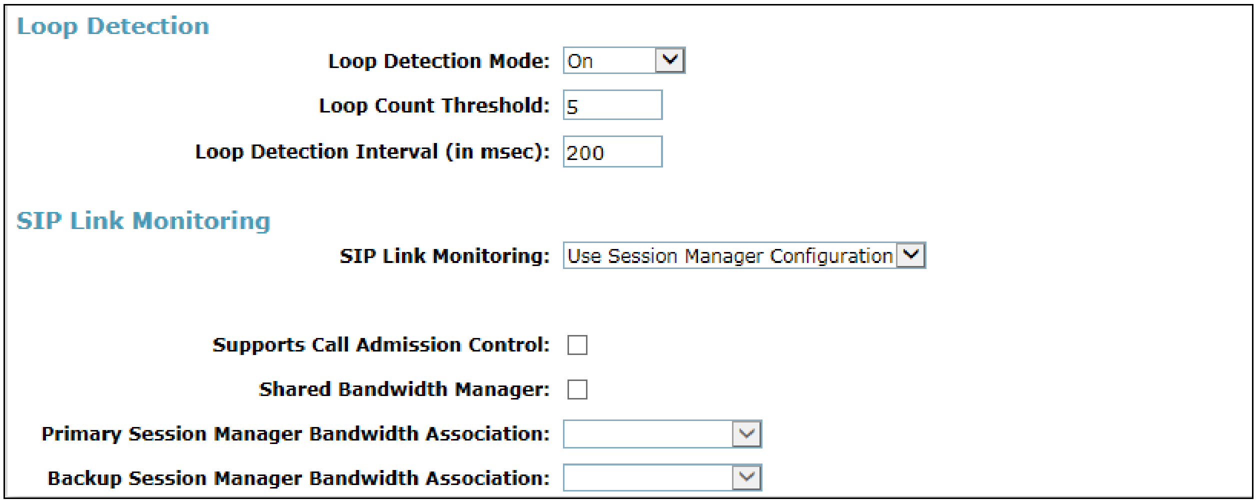


6.5.2. Avaya Aura® Communication Manager SIP Entities

The following screen shows one of the SIP entities for Communication Manager which is configured as an Evolution Server. This SIP Entity is used for the SIP Trunk. The **FQDN or IP Address** field is set to the IP address of the interface on Communication Manager that will be providing SIP signalling. Set the **Location** to that defined in **Section 6.3**.



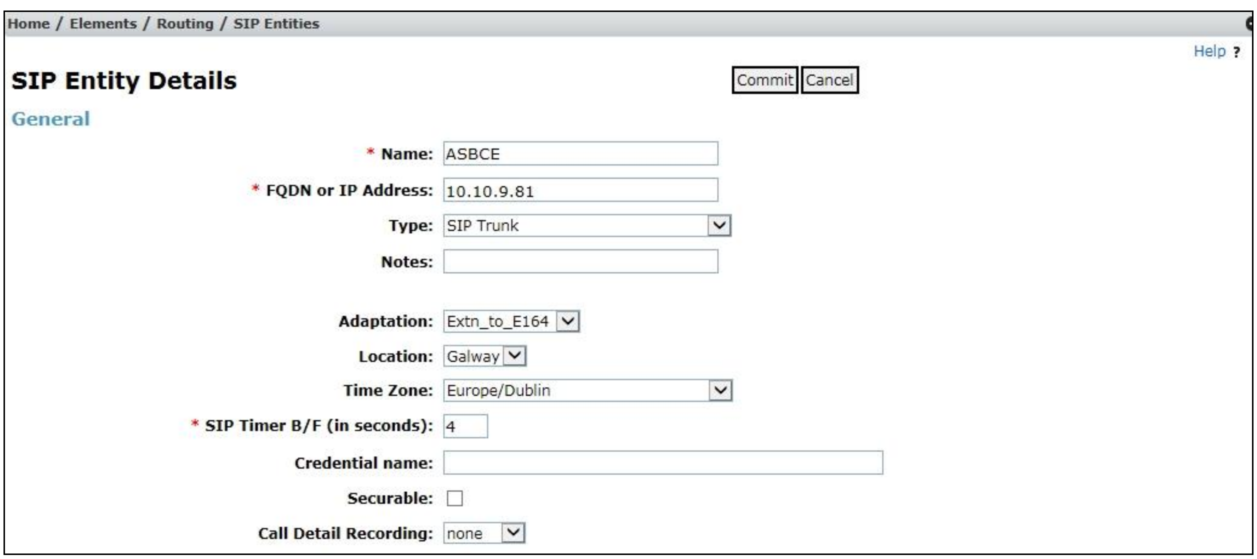
Other parameters can be set for the SIP Entity as shown in the following screenshot, but for test, these were left at default values.



Note: An identical SIP Entity for Communication Manager is required for SIP Endpoints. In the test environment, the name of this SIP Entity is **CM_SIP_Endpoints**, and it is differentiated by use of port number

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.

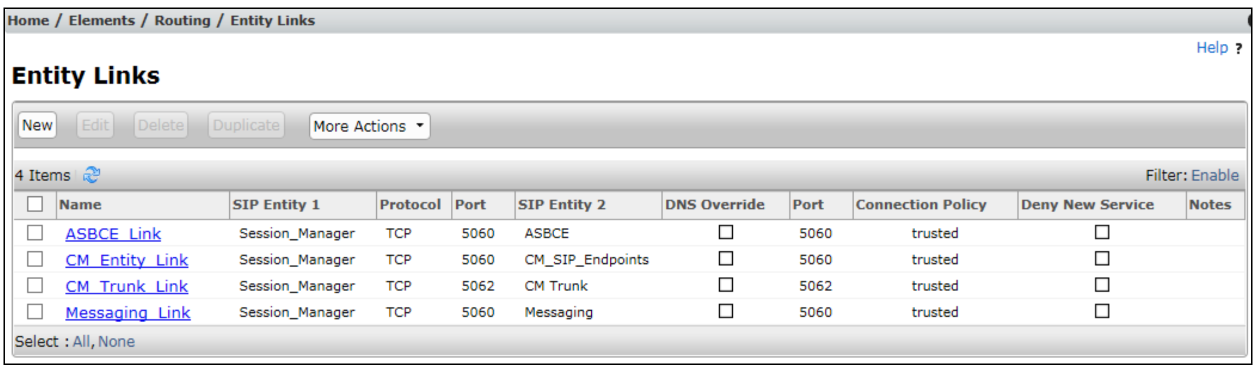


6.6. Administer Entity Links

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 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_Link	Session_Manager	TCP	5060	ASBCE	☐	5060	trusted	☐	
☐	CM_Entity_Link	Session_Manager	TCP	5060	CM_SIP_Endpoints	☐	5060	trusted	☐	
☐	CM_Trunk_Link	Session_Manager	TCP	5062	CM Trunk	☐	5062	trusted	☐	
☐	Messaging_Link	Session_Manager	TCP	5060	Messaging	☐	5060	trusted	☐	

Select : All, None

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

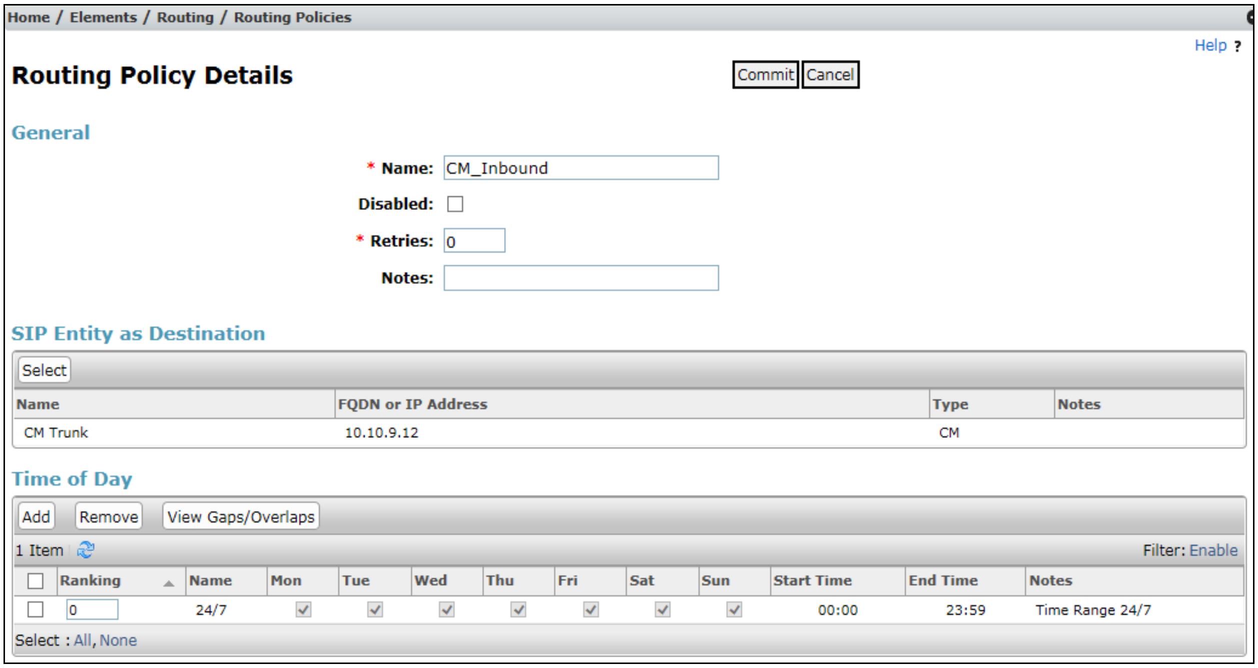
6.7. Administer Routing Policies

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

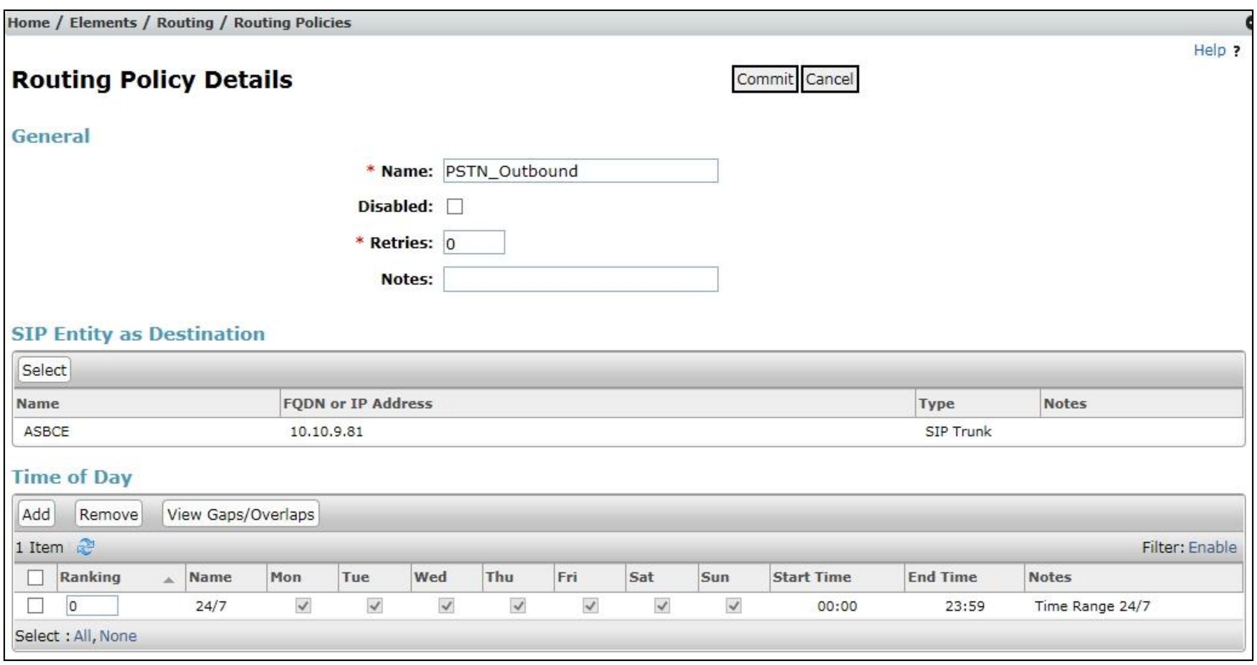
Under **General**:

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

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



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



6.8. Administer Dial Patterns

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

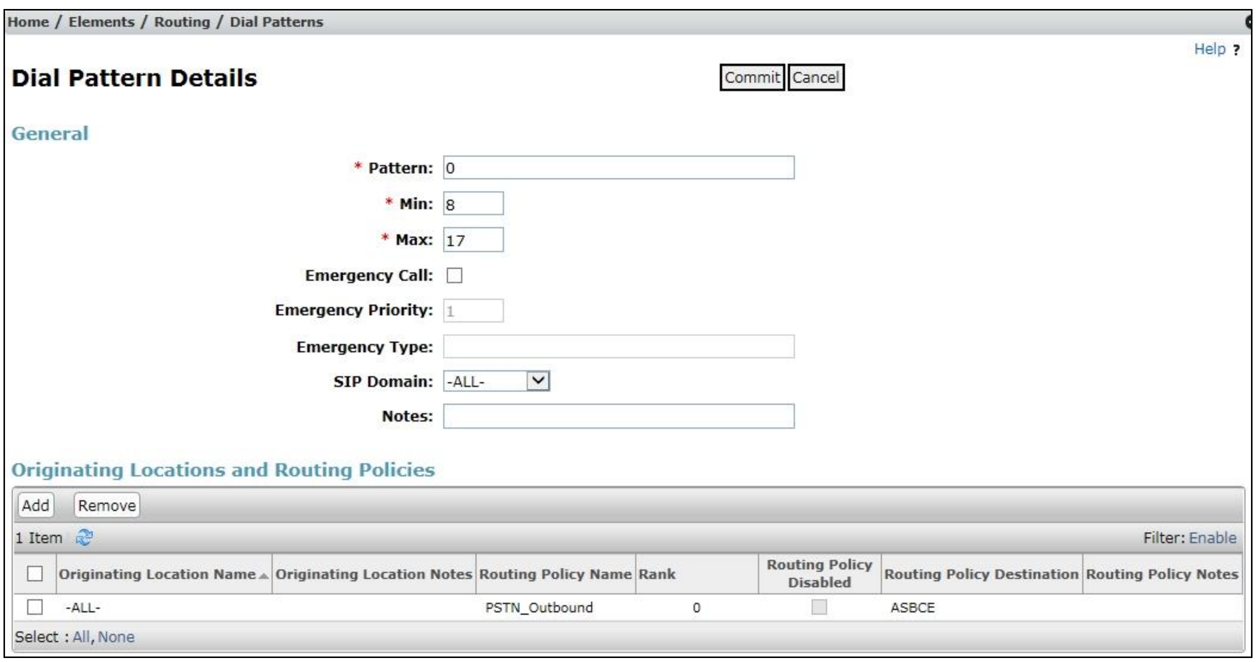
Under **General**:

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

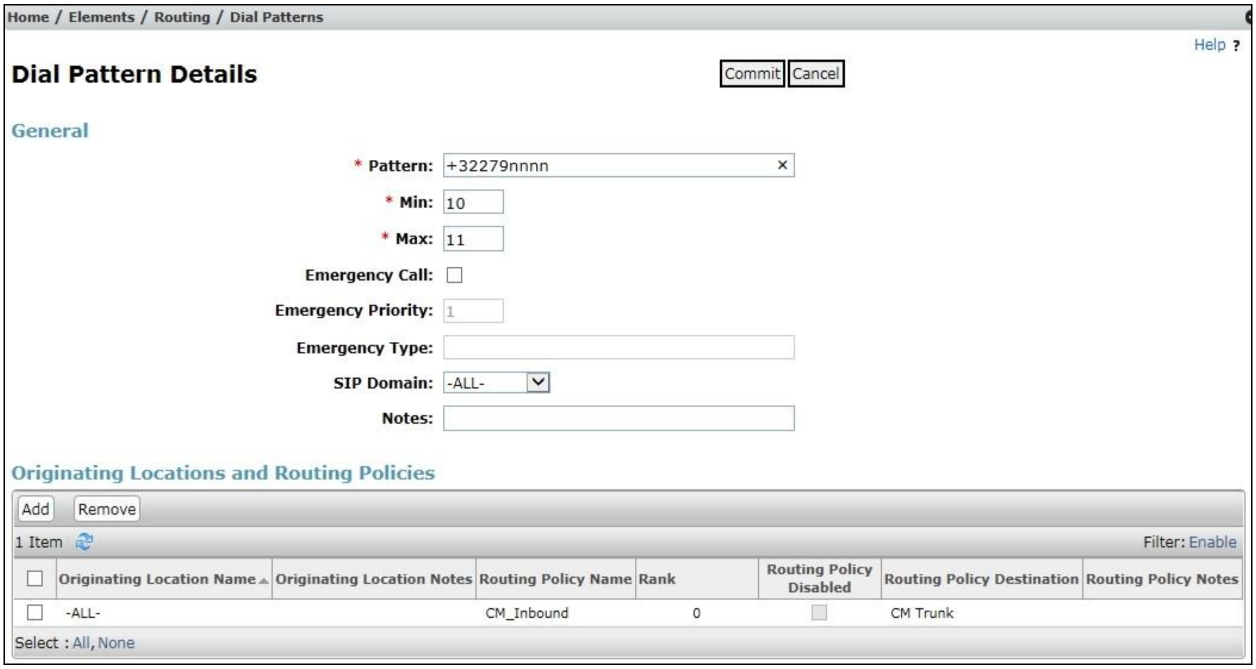
Under **Originating Locations and Routing Policies**:

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

The following screen shows an example dial pattern configured for the Avaya SBCE which will route the calls out to the PSTN via the Proximus SIP Trunking service.



The following screen shows the test dial pattern configured for Communication Manager.

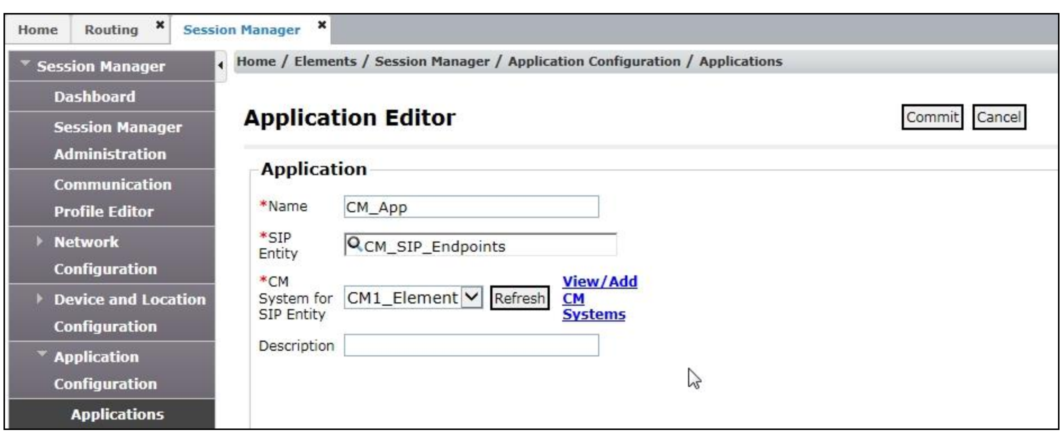


Note: The above configuration is used to analyse the DDI numbers assigned to the extensions on Communication Manager. Some of the digits of the pattern to be matched have been obscured.

6.9. Administer Application for Avaya Aura® Communication Manager

The Application for Communication Manager would normally be defined at system installation, but is shown here for reference. From the **Home** tab select **Session Manager** from the menu. In the resulting tab from the left panel menu select **Application Configuration → Applications** and click **New** (not shown).

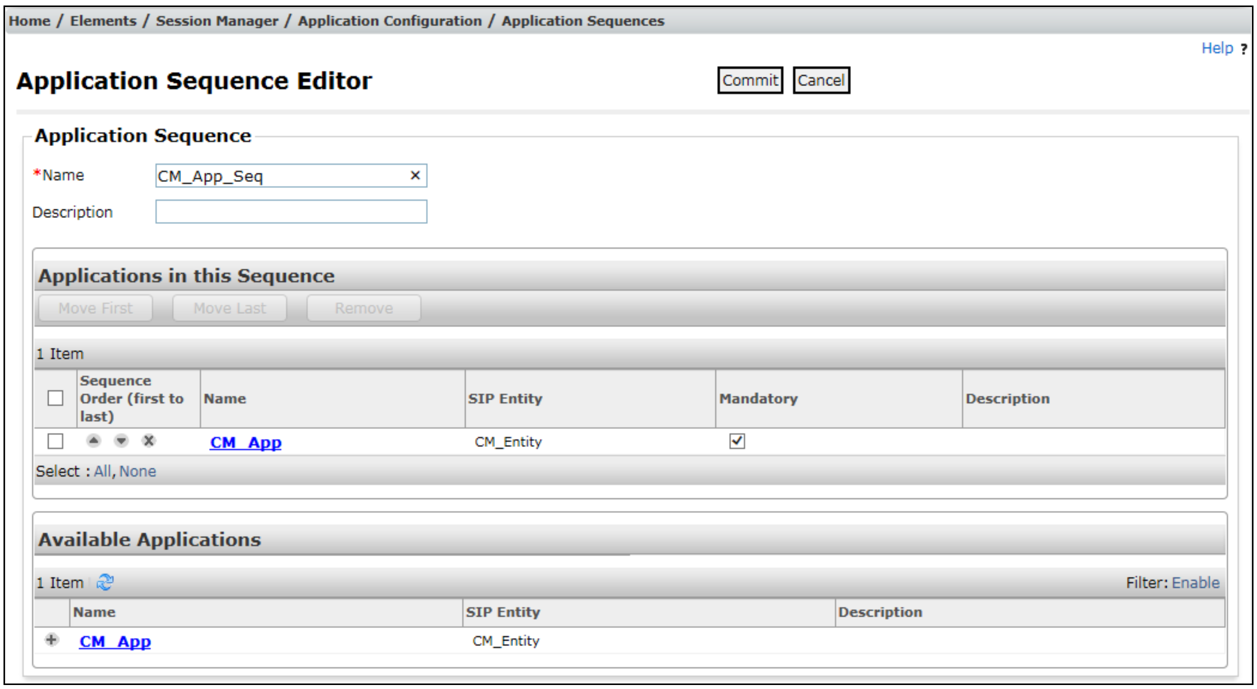
- In the **Name** field enter a name for the application.
- In the **SIP Entity** field select the SIP entity for Communication Manager.
- In the **CM System for SIP Entity** field select the SIP entity for Communication Manager SIP Endpoints and select **Commit** to save the configuration.



6.10. Administer Application Sequence for Avaya Aura® Communication Manager

The Application Sequence for Communication Manager would normally be defined at system installation, but is shown here for reference. From the left panel navigate to **Session Manager → Application Configuration → Application Sequences** and click on **New** (not shown).

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

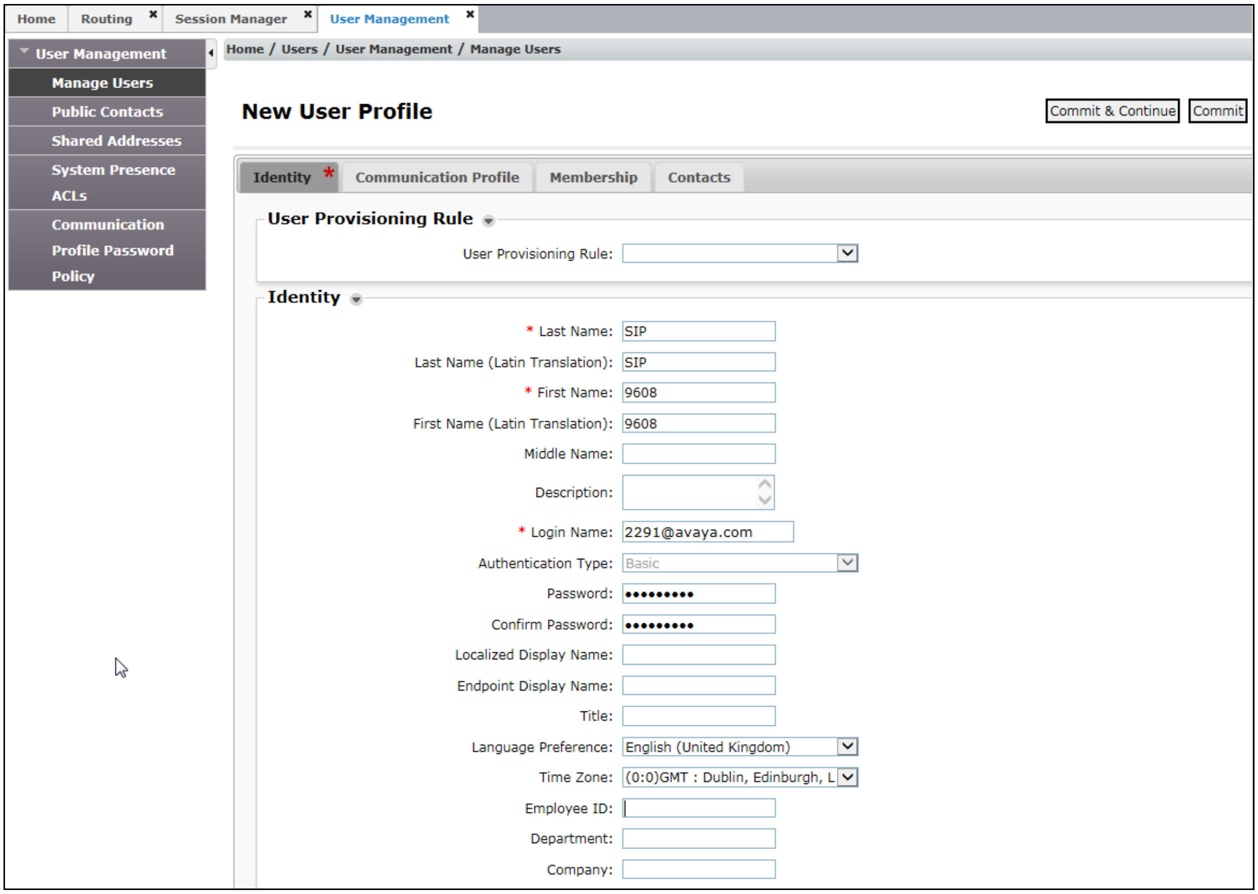


6.11. Administer SIP Extensions

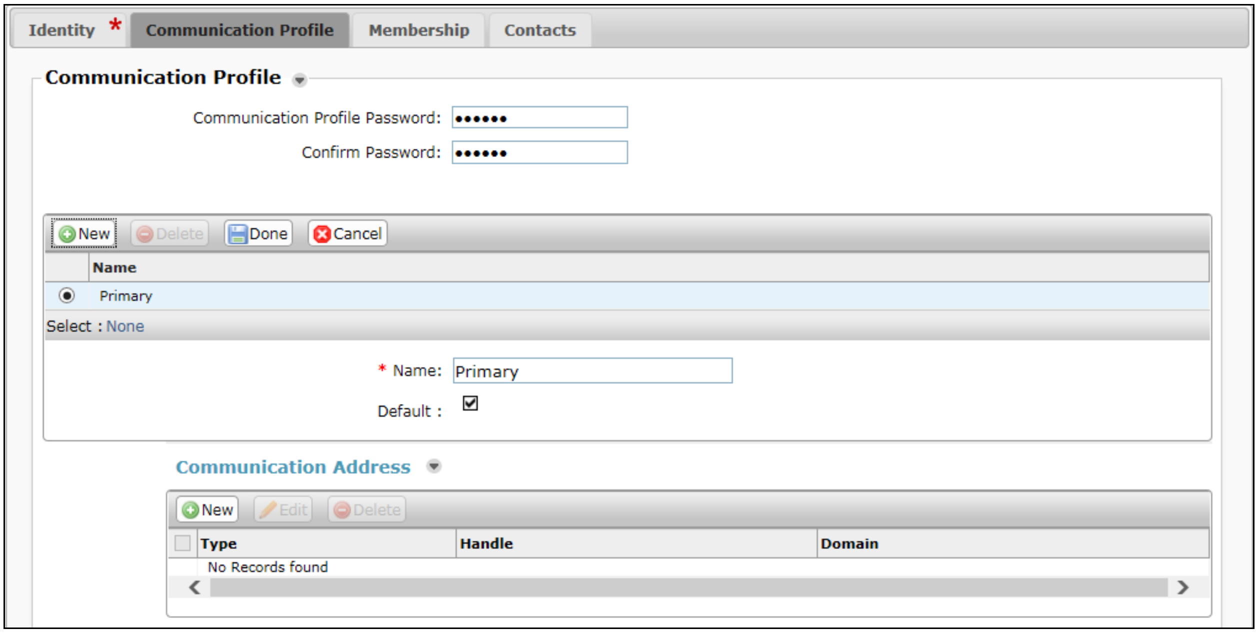
SIP extensions are registered with the Session Manager and use Communication Manager for their feature and configuration settings. From the **Home** tab select **User Management** from the menu. Then select **Manage Users** and click **New** (not shown).

On the **Identity** tab:

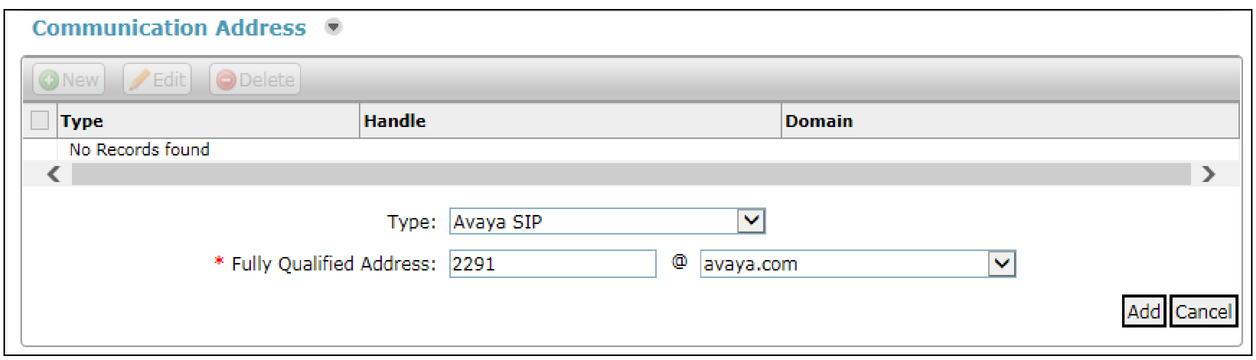
- Enter the user's name in the **Last Name** and **First Name** fields.
- In the **Login Name** field enter a unique system login name in the form of user@domain e.g. 2291@avaya.com which is used to create the user's primary handle.
- The **Authentication Type** should be **Basic**.
- In the **Password/Confirm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



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

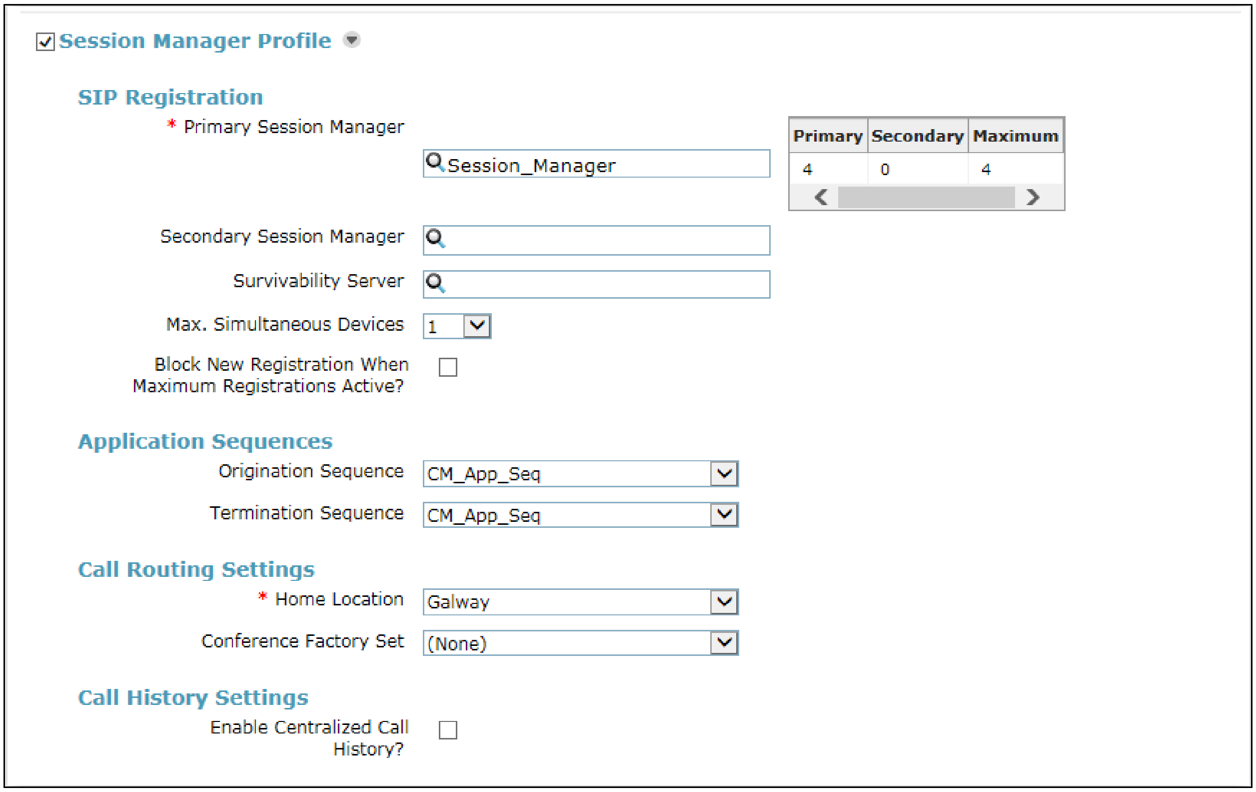


Expand the **Communication Address** section and click **New**. For the **Type** field select **Avaya SIP** from the drop-down menu. In the **Fully Qualified Address** field, enter an extension number and select the relevant domain from the drop-down menu. Click the **Add** button.



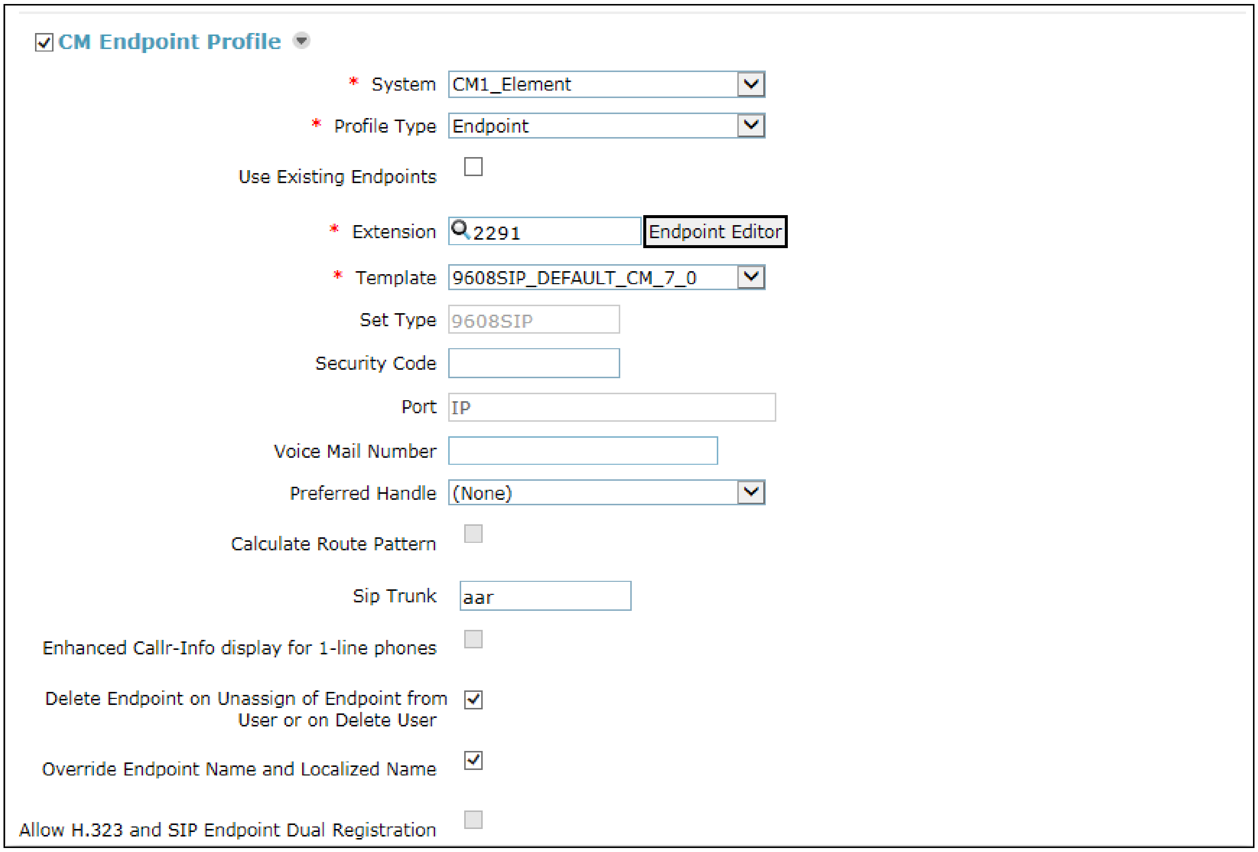
Expand the **Session Manager Profile** section.

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

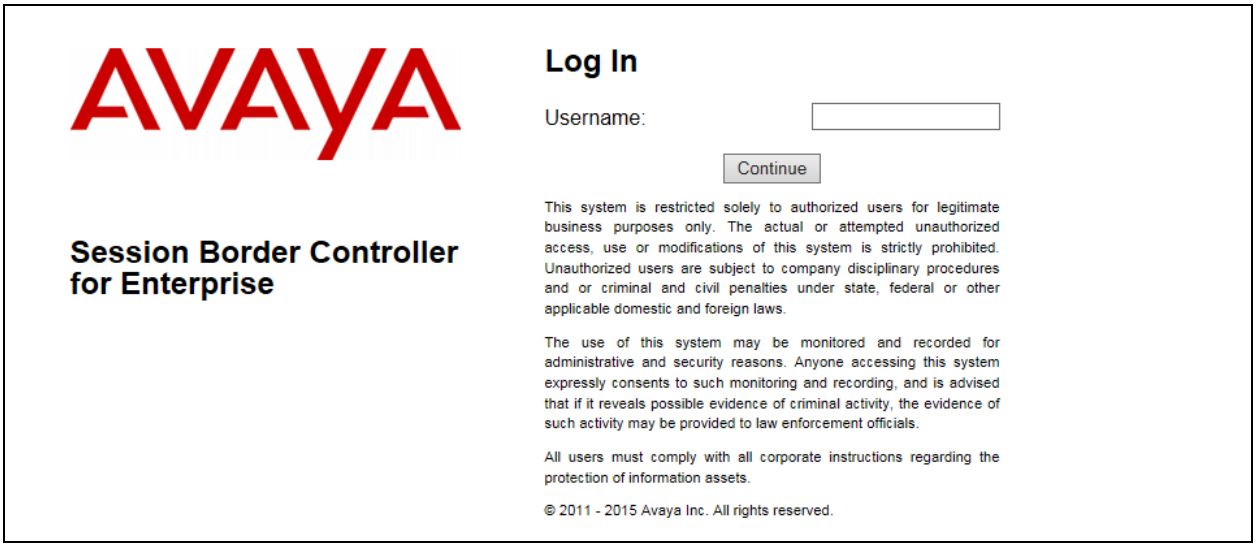


7. Configure Avaya Session Border Controller for Enterprise

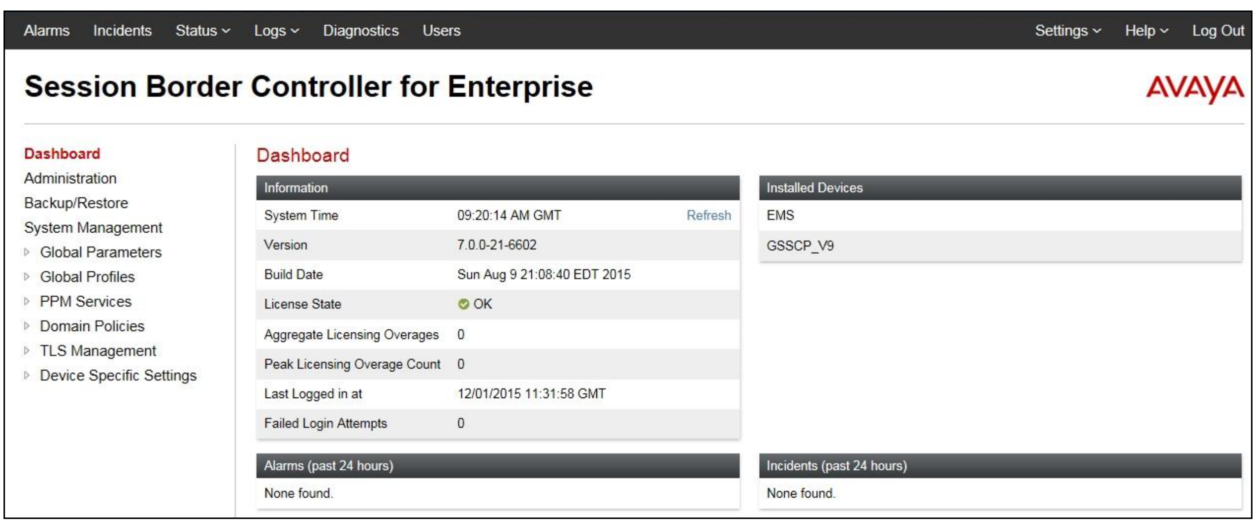
This section describes the configuration of the Avaya Session Border Controller for Enterprise (Avaya SBCE). The Avaya SBCE provides security and manipulation of signalling to provide an interface to the Service Provider's SIP Trunk that is standard where possible and adapted to the Service Provider's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

7.1. Access Avaya Session Border Controller for Enterprise

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



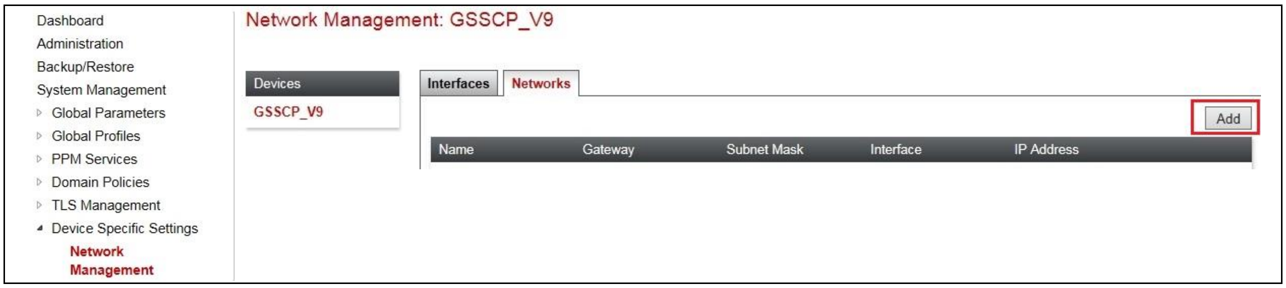
Once logged in, a dashboard is presented with a menu on the left-hand side. The menu is used as a starting point for all configuration of the Avaya SBCE.



7.2. Define Network Management

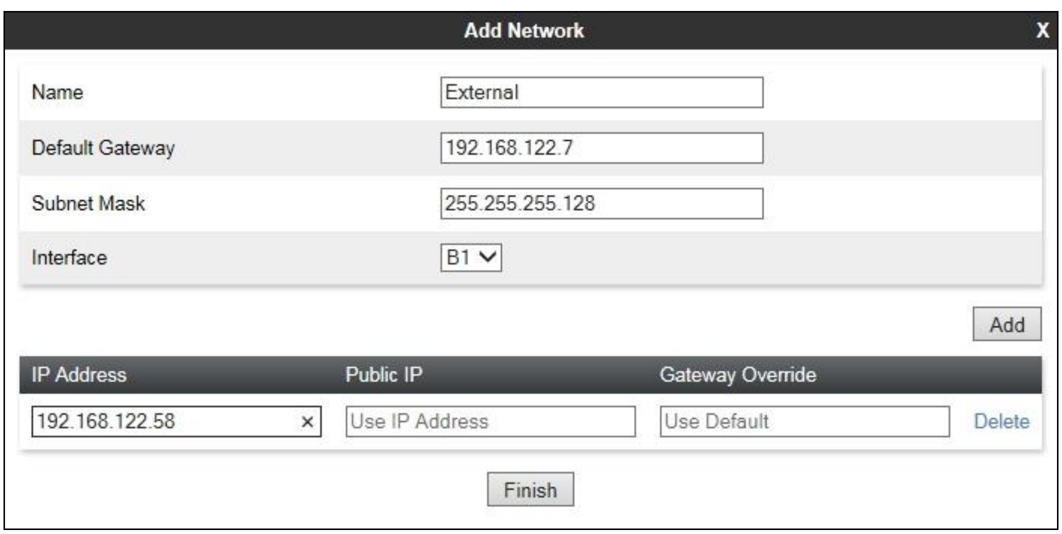
Network information is required on the Avaya SBCE to allocate IP addresses and subnet masks to the interfaces. Note that only the **A1** and **B1** interfaces are used, typically the **A1** interface is used for the internal side and **B1** is used for external. Each side of the Avaya SBCE can have only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

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



Enter details for the external interface in the dialogue box:

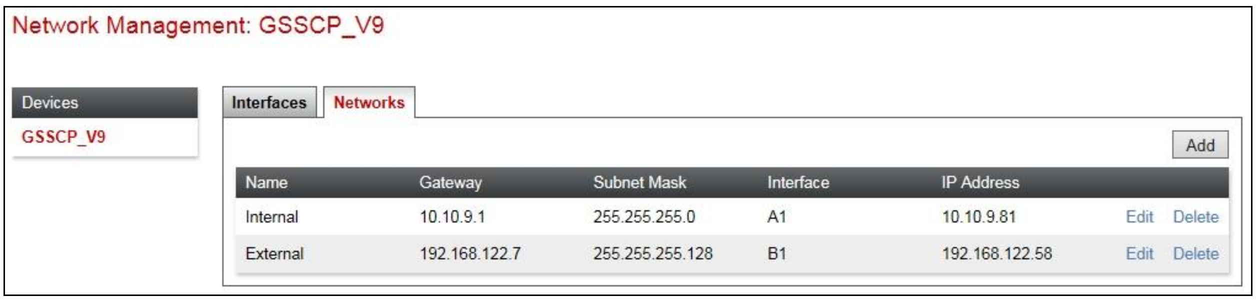
- Enter a descriptive name in the **Name** field.
- Enter the default gateway IP address for the external interface in the **Default Gateway** field.
- Enter the subnet mask in the **Subnet Mask** field.
- Select the external interface to be used from the **Interface** drop down menu. In the test environment,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



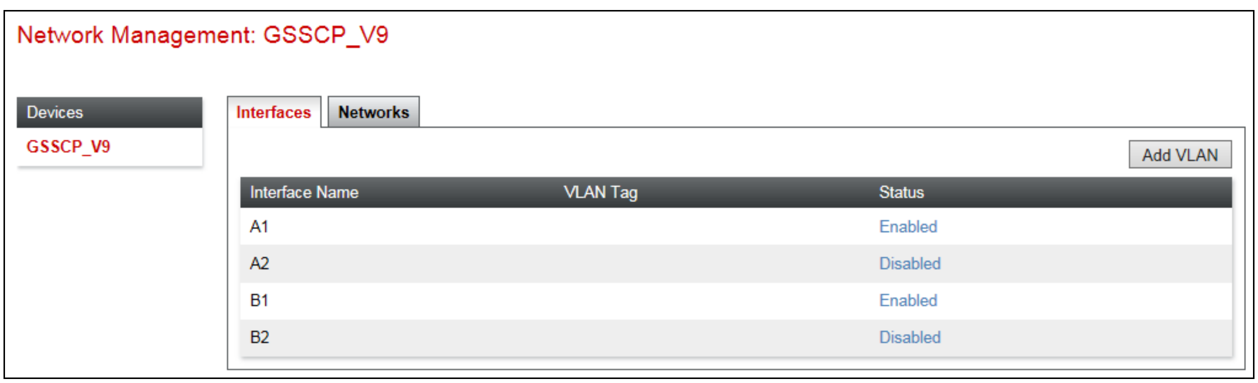
Click on **Add** to define the internal interface. Enter details in the dialogue box (not shown):

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

The following screenshot shows the completed Network Management configuration:



Select the **Interface Configuration** tab and click on the **Status** of the physical interface to toggle it. A status of **Disabled** will be changed to **Enabled**.



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

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

A status box will appear (not shown) that will indicate when the application has restarted.

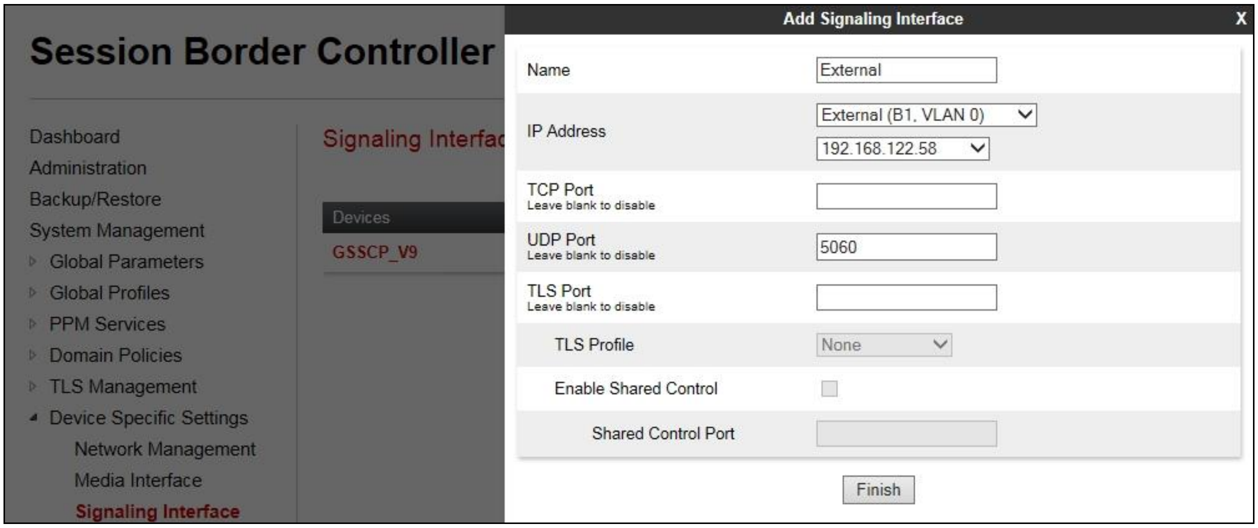
7.3. Define Interfaces

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 TCP used for transport of signalling between Session Manager and the Avaya SBCE, and UDP for transport of signalling between the Avaya SBCE and the Proximus SIP Trunking service. This document shows the configuration for TCP and UDP, if additional security is required, it's recommended to use TLS and port 5061.

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 external and internal SIP signalling are entered here.

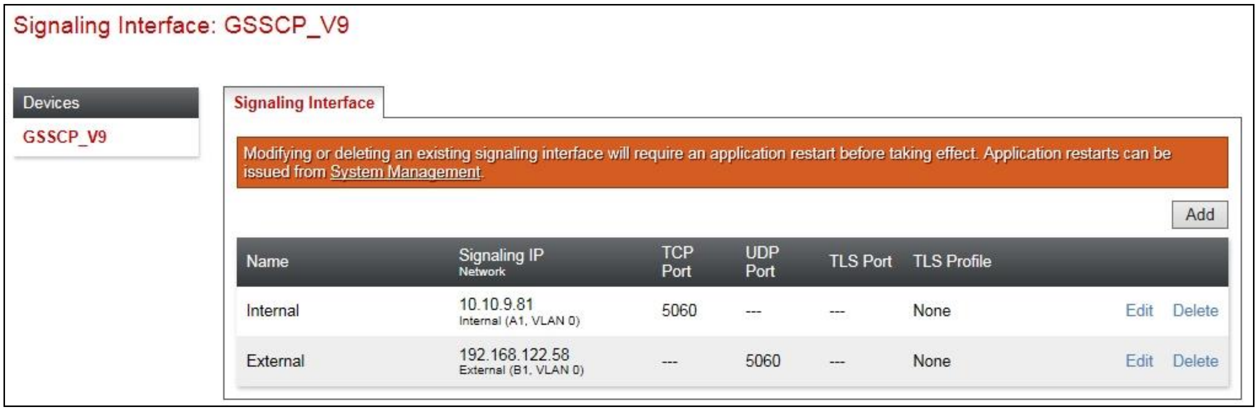
- Select **Add** and enter details of the external signalling interface in the pop-up menu.
- In the **Name** field enter a descriptive name for the external signalling interface.
- In the **IP Address** drop down menus, select the external network interface and IP address. Note that when the external network interface is selected, the bottom drop down menu is populated with the available IP addresses as defined in **Section 7.2**. In the test environment, this was a single IP address **192.168.122.58**.
- Enter the UDP port number in the **UDP Port** field, **5060** is used for the Proximus SIP Trunking service.



The internal signalling interface is defined in the same way; the dialogue box is not shown:

- Select **Add** and enter details of the internal signalling interface in the pop-up menu.
- In the **Name** field enter a descriptive name for the internal signalling interface.
- In the **IP Address** drop down menus, select the internal network interface and IP address.
- Select **TCP** port number, **5060** is used for Session Manager.

The following screenshot shows details of the signalling interfaces:



Note. In the test environment, the internal IP address was **10.10.9.81**.

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 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 external media interface in the pop-up menu.
- In the **Name** field enter a descriptive name for the external media interface.
- In the **IP Address** drop down menus, select the external network interface and IP address. Note that when the external network interface is selected, the bottom drop down menu is populated with the available IP addresses as defined in **Section 7.2**. In the test environment, this was a single IP address **192.168.122.58**.
- Define the RTP **Port Range** for the media path with the Proximus SIP Trunking service, during testing this was set to **1000** – **10019** which were the port values opened up in the Proximus firewall.

The internal media interface is defined in the same way; the dialogue box is not shown:

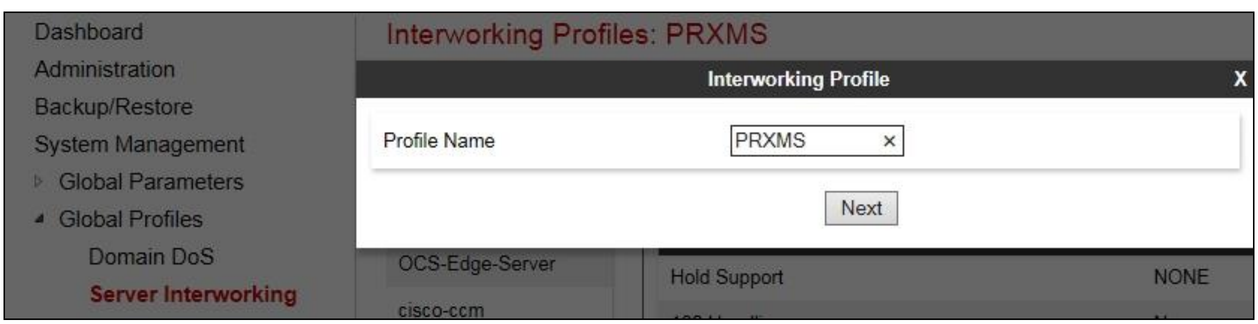
- Select **Add** and enter details of the internal media interface in the pop-up menu.
- In the **Name** field enter a descriptive name for the internal media interface.
- In the **IP Address** drop down menus, select the internal network interface and IP address.

The following screenshot shows details of the media interfaces:

7.4. Define Server Interworking

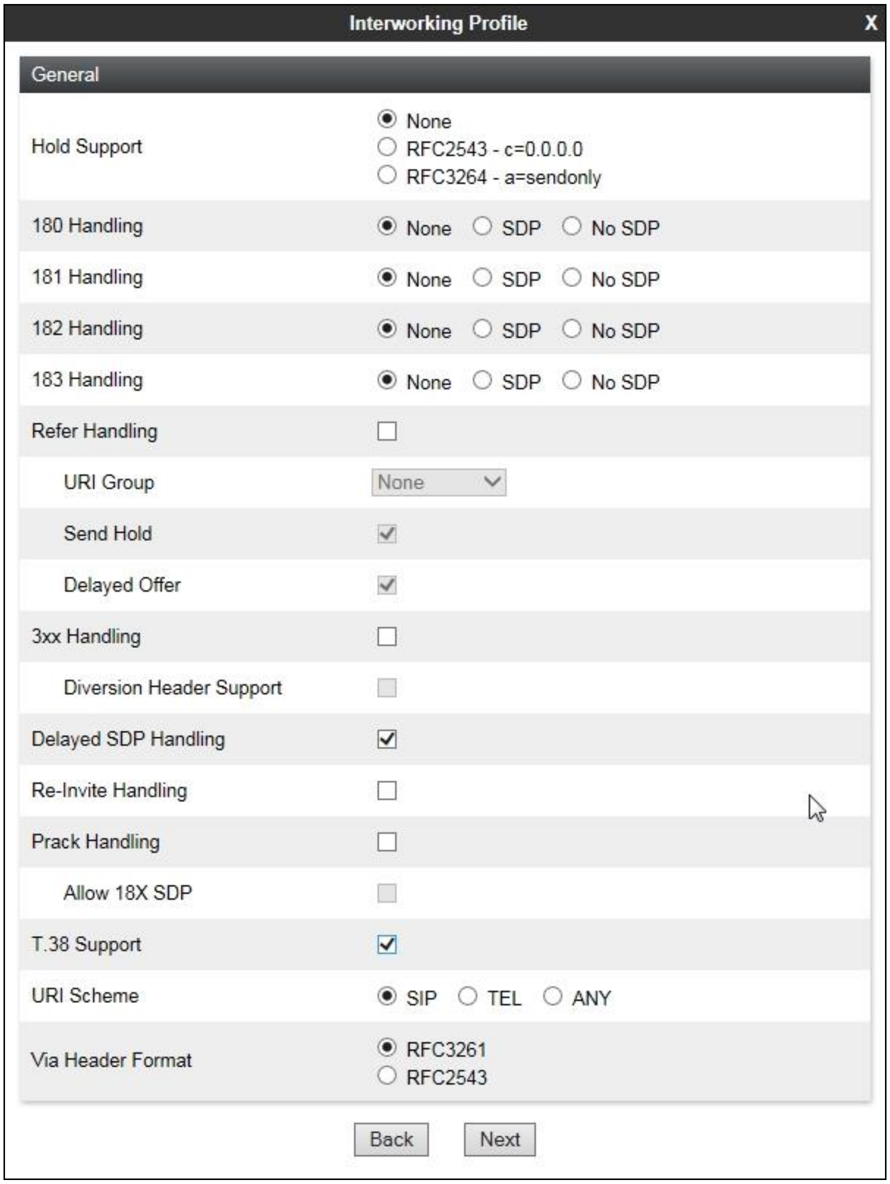
Server interworking is defined for each server connected to the Avaya SBCE. In this case, the Proximus SIP Trunking service is connected as the Trunk Server and Session Manager is connected as the Call Server.

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 Proximus SIP trunk, click on **Add** (not shown). A pop-up menu is generated. In the **Name** field enter a descriptive name for Proximus and click **Next**.



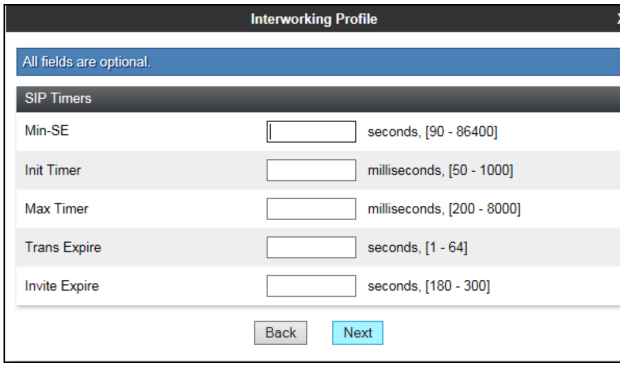
Configuration of interworking includes Hold support, T.38 fax support and SIP extensions.

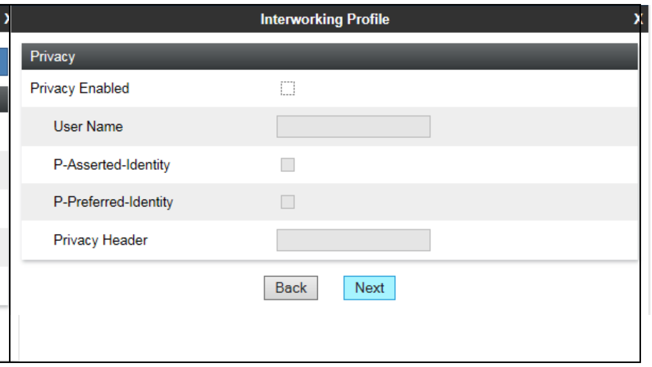
- In the General dialogue box, check the **Delayed SDP Handling** which ensures that an SDP is always included in the INVITE messages. This resolves an issue described in **Section 2.2** that was specific to the test environment, but could possibly be encountered in the live network.
- Check the **T.38 Support** box.



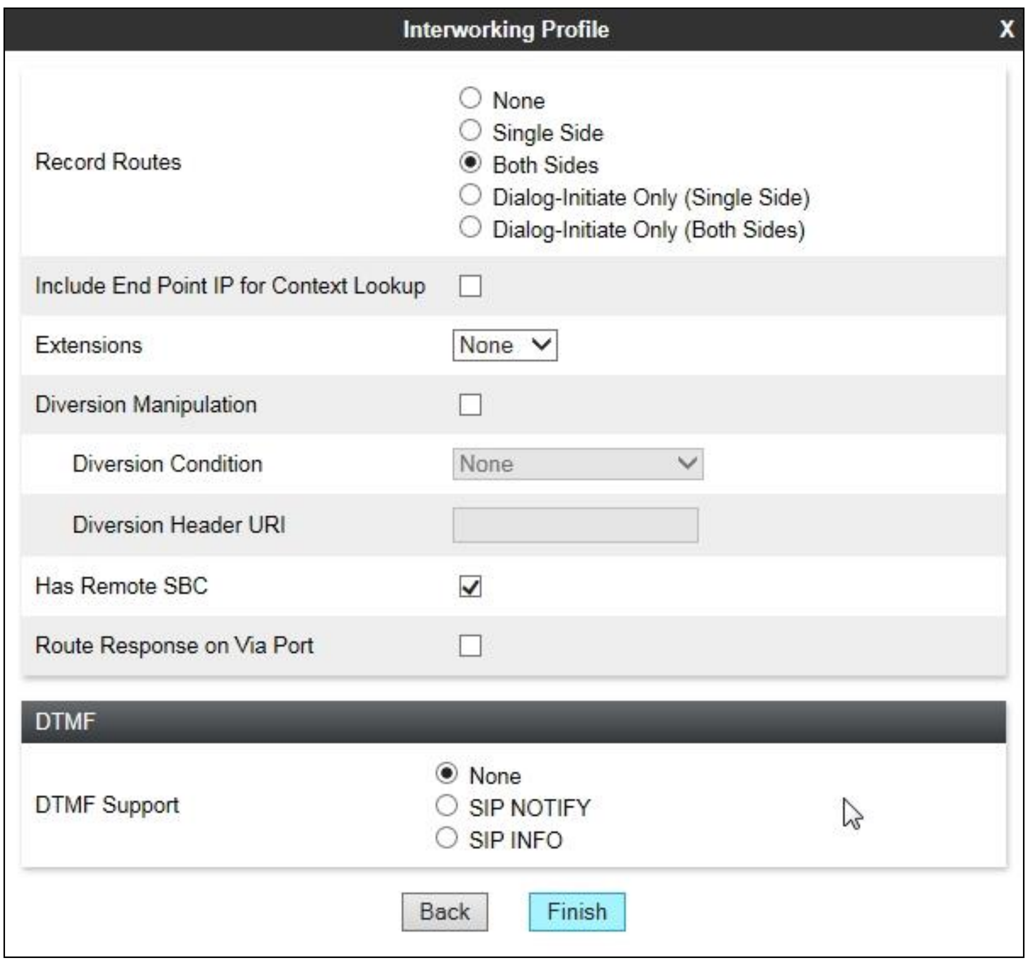
Note: During testing, the rest of the parameters were left at default values.

Click on **Next** and **Next** again to go through the next two dialogue boxes. During testing, these were left at default values.

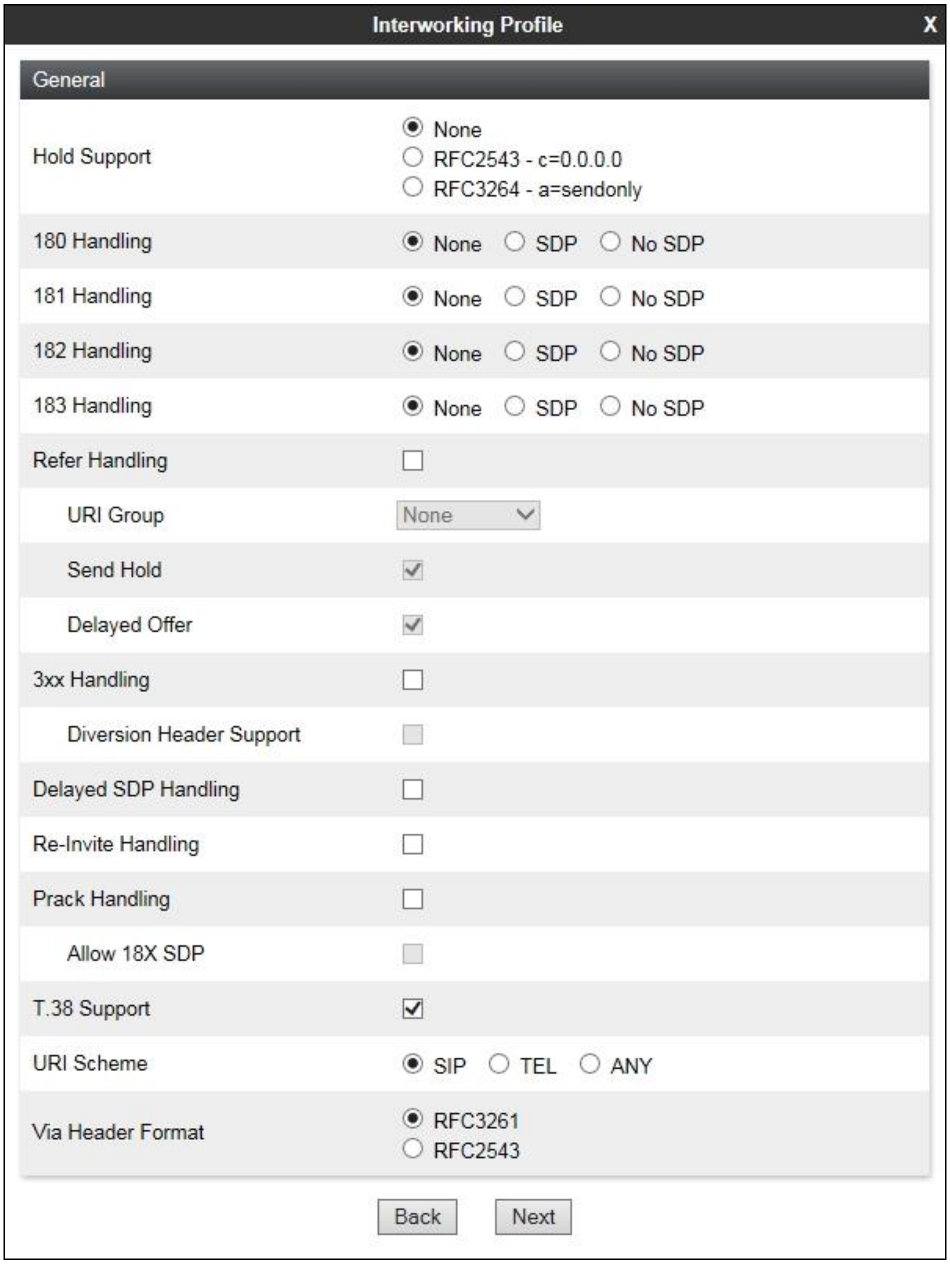




In the final dialogue box, ensure that the **Both Sides** button is checked in the **Record Routes** field and the **Has Remote SBC** box is checked. Click on **Finish**



To define Server Interworking for the Session Manager, click on **Add** (not shown). A pop-up menu (not shown) is generated. In the **Name** field enter a descriptive name for the Session Manager and click **Next**. Check the **T.38** box.



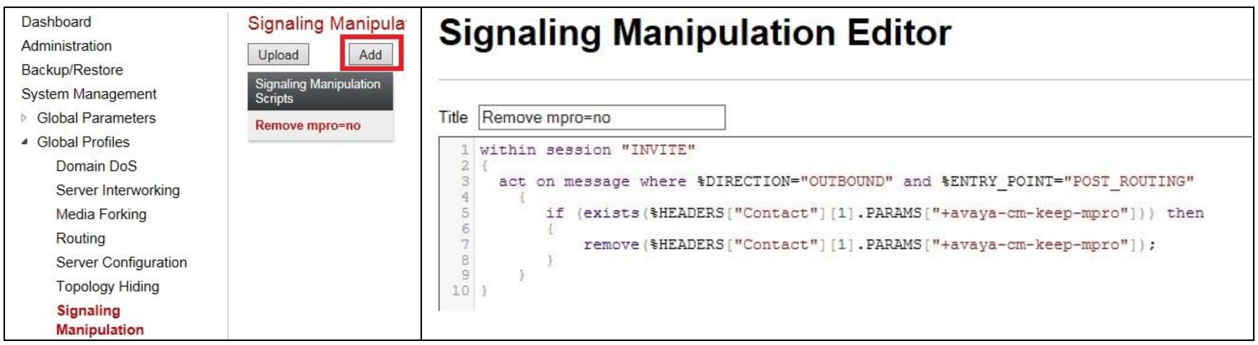
- Click on **Next** and **Next** again to go through the next two dialogue boxes (not shown). During testing, these were left at default values.
- Ensure the settings in the final dialogue box (not shown) are the same as those used for the Proximus SIP Trunking service.
- Click on **Finish**.

7.5. Define Signalling Manipulation

Signalling manipulation is required in cases where there is non-standard signalling between the Call Server and Trunk Server that can't be resolved by the Server Interworking described in the previous section. During testing, an issue was found with the handling of an Avaya specific parameter in the Contact header.

The Avaya proprietary parameter is "+avaya-cm-keep-mpro" and is present with a value of "no" when the Media Gateway is not used for call set-up i.e., when Initial IP-IP Direct Media is used on Communication Manager SIP Trunk. This can't be removed using the Header Manipulation tab in the Server Interworking profile described in the previous section, and a fault report AURORA-7477 has been raised to address this. During testing, Signalling Manipulation was used to remove the parameter

To define the signalling manipulation to remove the Avaya proprietary parameter, navigate to **Global Profiles → Signaling Manipulation** in the main menu on the left hand side. Click on **Add** which will open a script editor. Enter a title and the script.



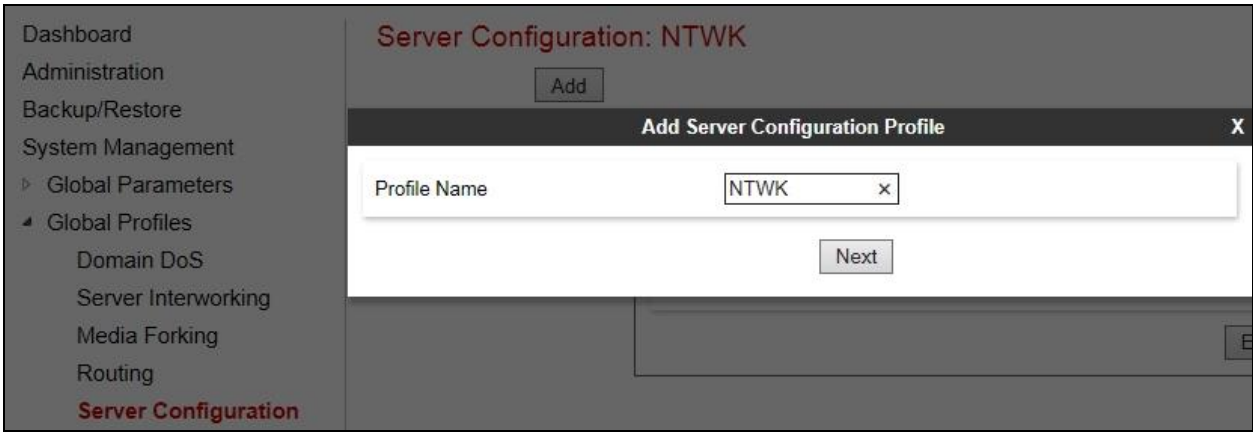
Click on **Save** (not shown). The script text is shown for clarity:

```
within session "INVITE"
{
  act on message where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"
    {
       if (exists(%HEADERS["Contact"][1].PARAMS["+avaya-cm-keep-mpro"])) then
       {
           remove(%HEADERS["Contact"][1].PARAMS["+avaya-cm-keep-mpro"]);
       }
    }
}
```

7.6. Define Servers

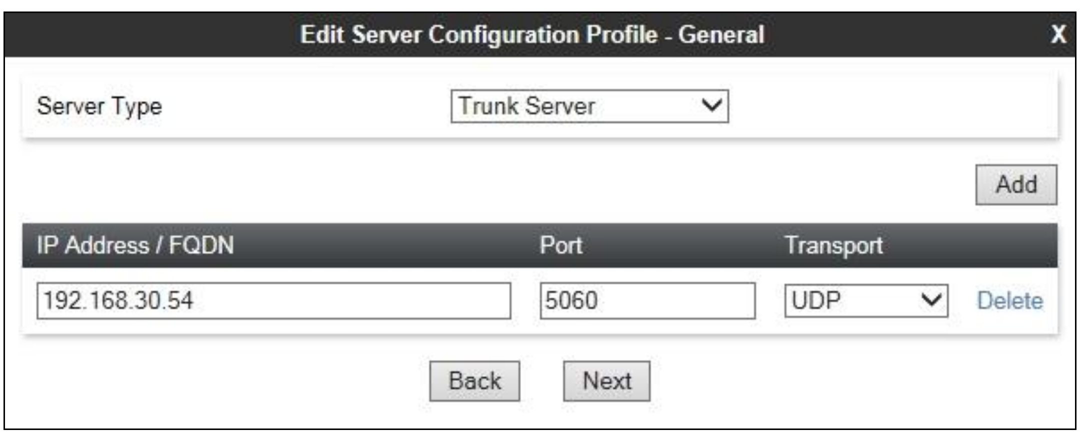
A server definition is required for each server connected to the Avaya SBCE. In this case, the Proximus SIP Trunking service is connected as the Trunk Server and Session Manager is connected as the Call Server.

To define the Proximus SIP Trunk Server, navigate to **Global Profiles → Server Configuration** in the main menu on the left hand side. Click on **Add** and enter an appropriate name in the pop-up menu.

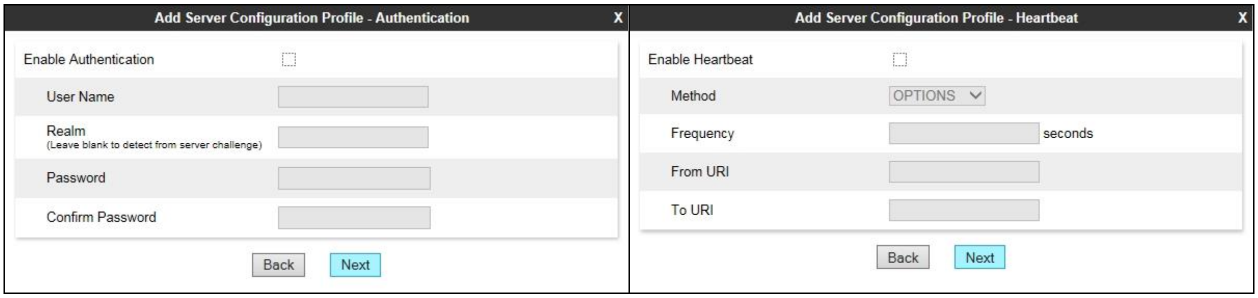


Click on **Next** and enter details in the dialogue box.

- In the **Server Type** drop down menu, select **Trunk Server**.
- Click on **Add** to enter an IP address
- In the **IP Addresses / FQDN** box, type the Proximus SIP trunk interface address.
- In the **Port** box, enter the port to be used for the SIP Trunk. This was left blank during testing which defaults to 5060 when UDP is used for transport.
- In the **Transport** drop down menu, select **UDP**.
- Click on **Next**.

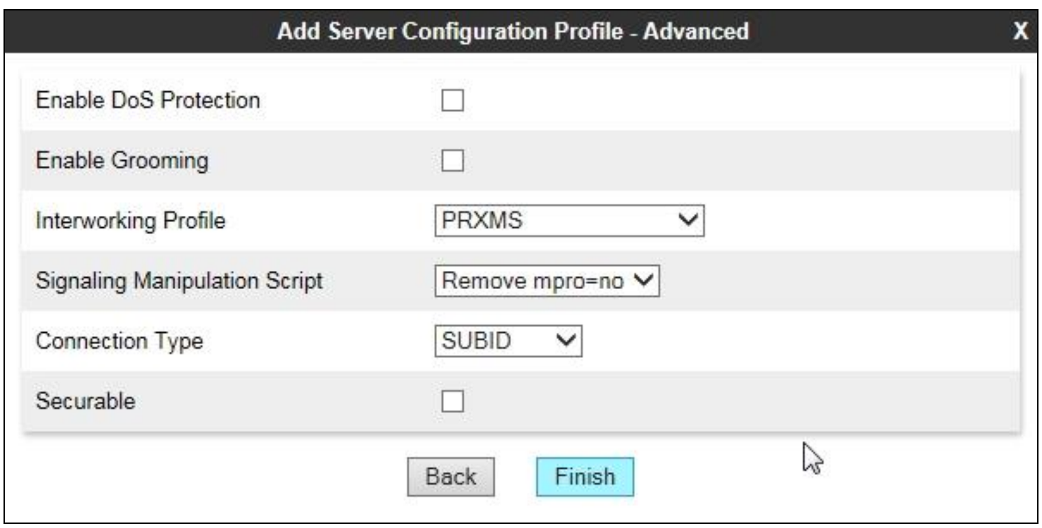


Click on **Next** and **Next** again. Leave the fields in the dialogue boxes at their default values.



Configure the Advanced Server Configuration Profile which is the final dialogue box.

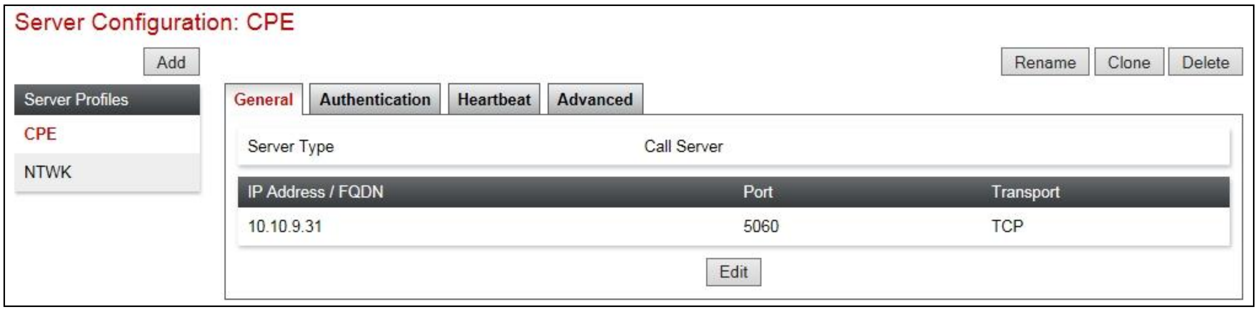
- In the **Interworking Profile** drop down menu, select the **Interworking Profile** for the Proximus SIP Trunking service defined in **Section 7.4**.
- In the **Signaling Manipulation Script** dialogue box, select the signalling manipulation script defined in **Section 7.5** to remove the Avaya proprietary parameter from the Contact header.
- Click **Finish**.



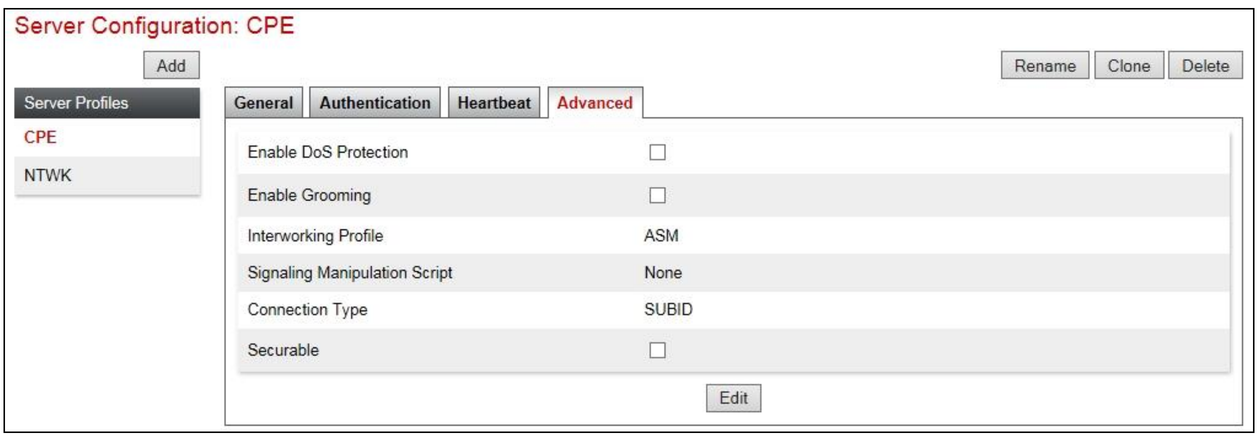
Use the same process to define the Call Server configuration for the Session Manager if not already defined.

- Ensure that **Call Server** is selected in the **Server Type** drop down menu in the **General** dialogue box.
- Ensure that the Interworking Profile defined for the Session Manager in **Section 7.4** is selected in the **Interworking Profile** drop down menu in the Advanced dialogue box

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



The next screenshot shows the **Advanced** tab.



Note that there is no **Signaling Manipulation Script** required for the Session manager server configuration.

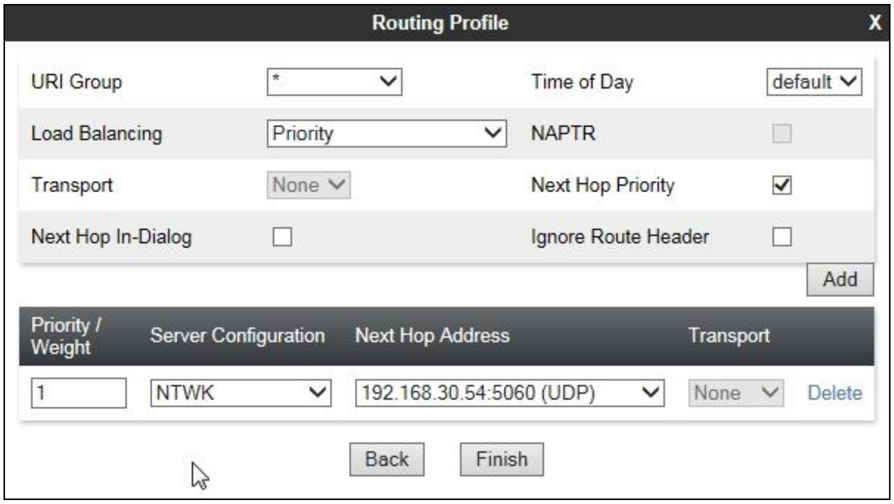
7.7. Define Routing

Routing information is required for routing to the Proximus SIP Trunking service on the external side and Session Manager on the internal side. The IP addresses and ports defined here will be used as the destination addresses for signalling.

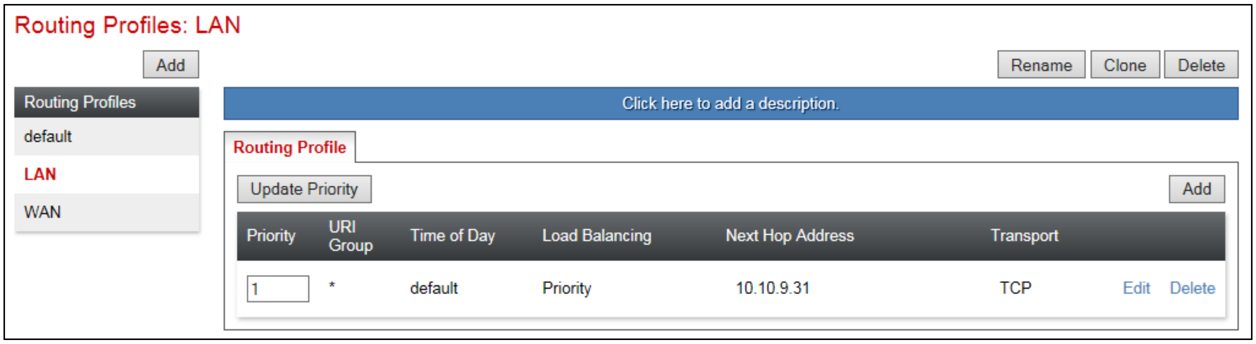
To define routing to the Proximus SIP Trunk, navigate to **Global Profiles → Routing** in the main menu on the left hand side. Click on **Add** and enter an appropriate name in the dialogue box.

Click on **Next** and enter details for the Routing Profile:

- In the **Load Balancing** drop down menu, select the method of load balancing required. During testing there was only one network interface provided for the Proximus SIP trunk so there was no load balancing required. This field was left at default value.
- Click on **Add** to specify the IP address for the Proximus SIP trunk.
- Assign a priority in the **Priority / Weight** field, during testing a value of **1** was used.
- Select the Server Configuration defined in **Section 7.6** in the **Server Configuration** drop down menu. This automatically populates the **Next Hop Address** field
- Click **Finish**.



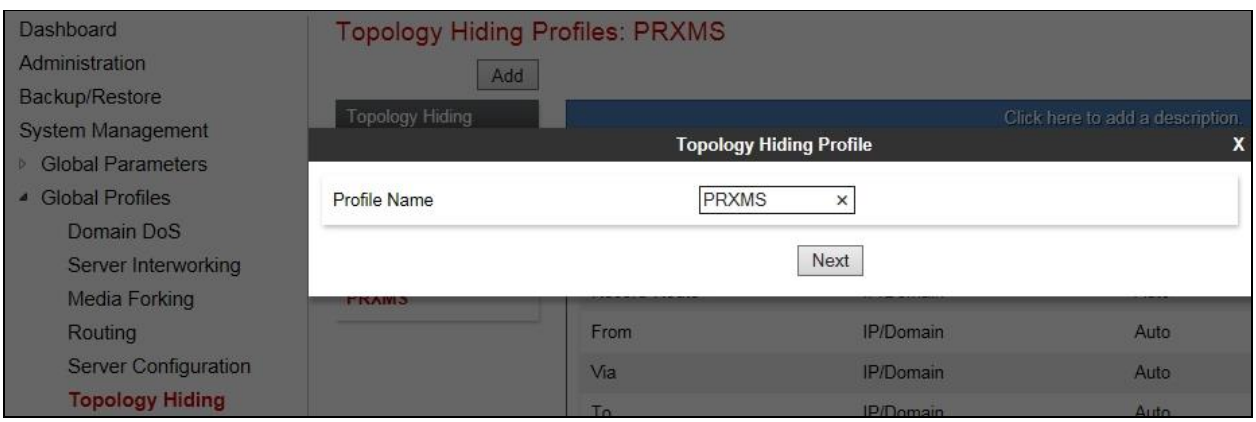
Repeat the process for the Routing Profile for Session Manager: The following screenshot shows the completed configuration:



7.8. Topology Hiding

Topology Hiding is used to hide local information such as private IP addresses and local domain names. The local information can be overwritten with a domain name or IP addresses. The default **Replace Action** is **Auto**, this replaces local information with IP addresses, generally the next hop or external interfaces. 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 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.

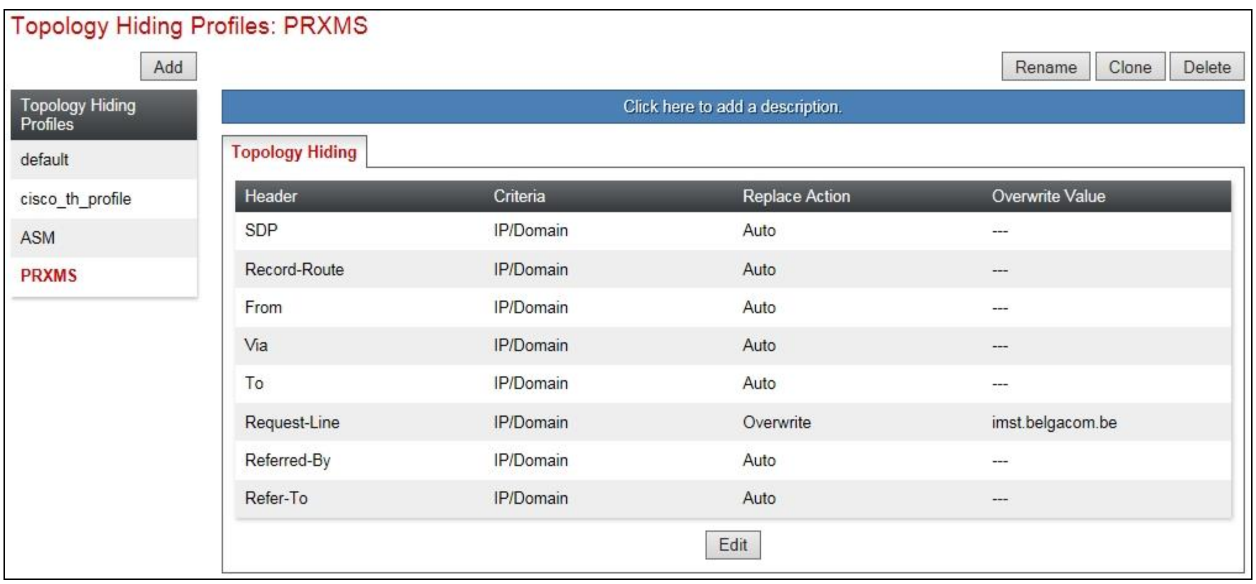
To define Topology Hiding for the Proximus SIP Trunking service, navigate to **Global Profiles → Topology Hiding** in the main menu on the left hand side. Click on **Add** to bring up a dialogue box, assign an appropriate name and click on **Next**.



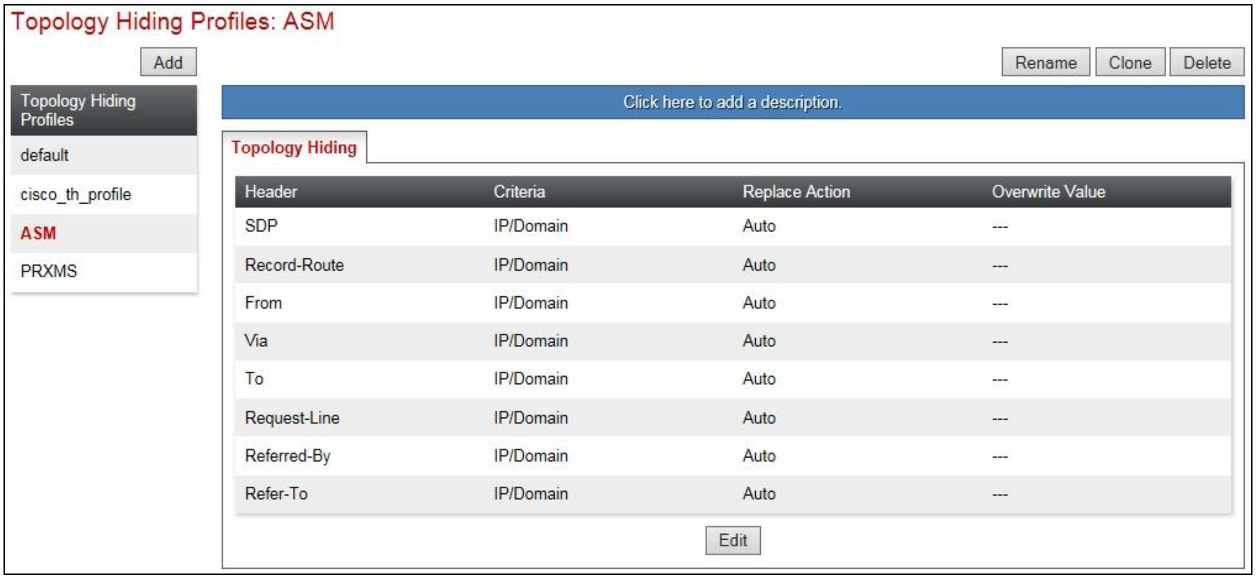
Enter details in the **Topology Hiding Profile** pop-up menu.

- In the **Profile Name** field enter a descriptive name for the Proximus SIP Trunking service and click **Next**.
- Click on **Add Header** and select from the **Header** drop down menu.
- Select **IP** or **IP/Domain** from the **Criteria** drop down menu depending on requirements.
- Leave the **Replace Action** at the default value of **Auto** unless a specific domain name is required. During testing, the domain name in the **Request-Line** was overwritten to that used by Proximus.
- Topology Hiding was defined for all headers where the function is available.

The screenshot shows the completed Topology Hiding definition for the Proximus SIP trunk:



To define Topology Hiding for Session Manager, follow the same process. This can be simplified by cloning the profile defined for the Proximus SIP Trunking service. Do this by highlighting the profile defined for the Proximus and clicking on **Clone**. Enter an appropriate name for the Session Manager profile and click on **Next**. Make any changes where required, in the test environment the **Replace Action** for **Request-Line** was left at the default value of **Auto**.



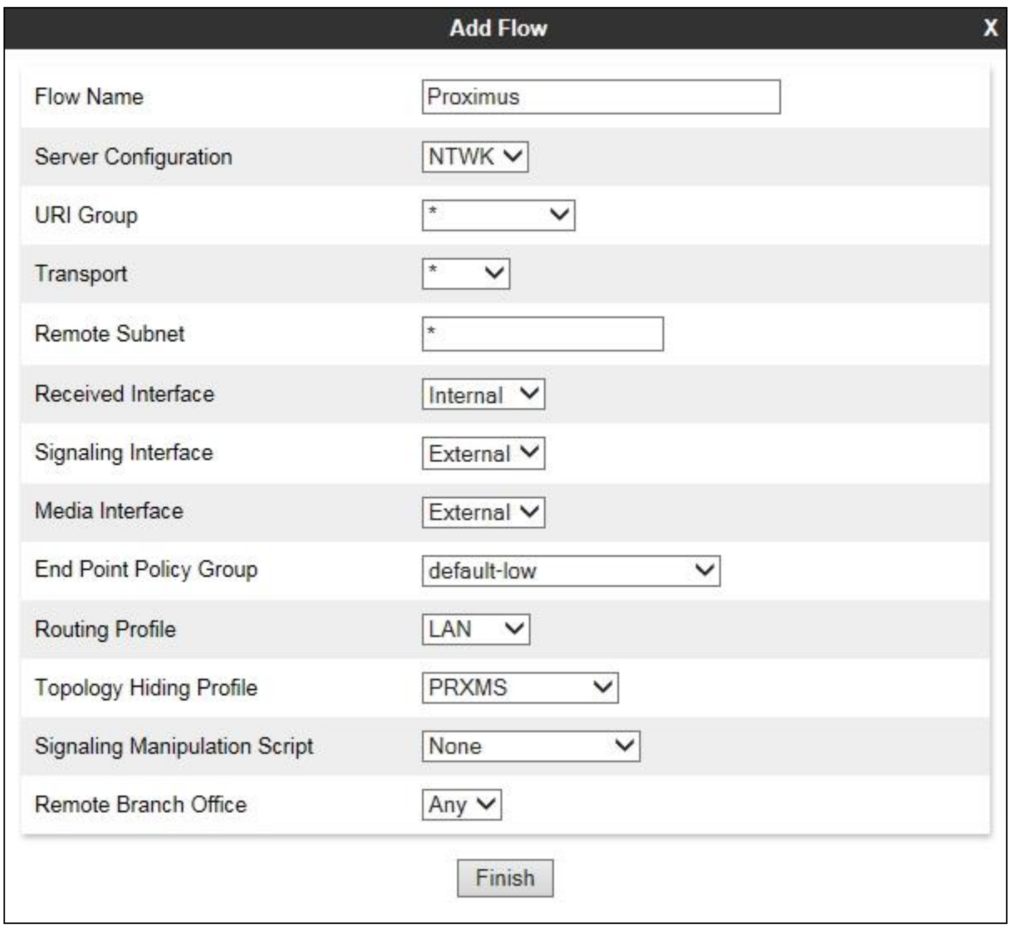
Header	Criteria	Replace Action	Overwrite Value
SDP	IP/Domain	Auto	---
Record-Route	IP/Domain	Auto	---
From	IP/Domain	Auto	---
Via	IP/Domain	Auto	---
To	IP/Domain	Auto	---
Request-Line	IP/Domain	Auto	---
Referred-By	IP/Domain	Auto	---
Refer-To	IP/Domain	Auto	---

7.9. Server Flows

Server Flows combine the previously defined profiles into two End Point Server Flows, one for the Proximus SIP Trunking service and another for Session Manager. These End Point Server Flows allow calls to be routed from Session Manager to the Proximus SIP Trunk and vice versa.

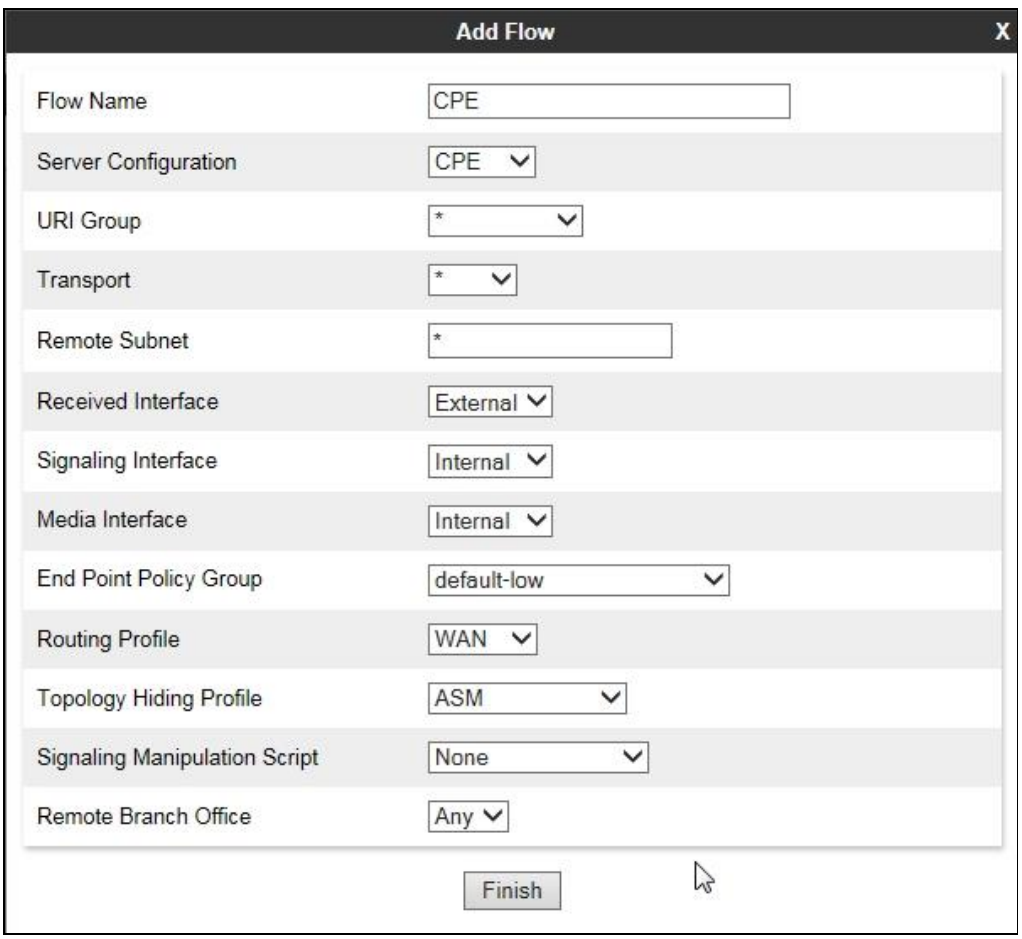
To define a Server Flow for the Proximus SIP Trunking service, navigate to **Device Specific Settings → End Point Flows**.

- 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 Proximus SIP Trunking service, in the test environment **Proximus** was used.
- 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 the Proximus SIP Trunking service 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 the Proximus SIP Trunking service 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 the Proximus SIP Trunking service 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 Proximus SIP Trunking service defined in **Section 7.8** and click **Finish**.

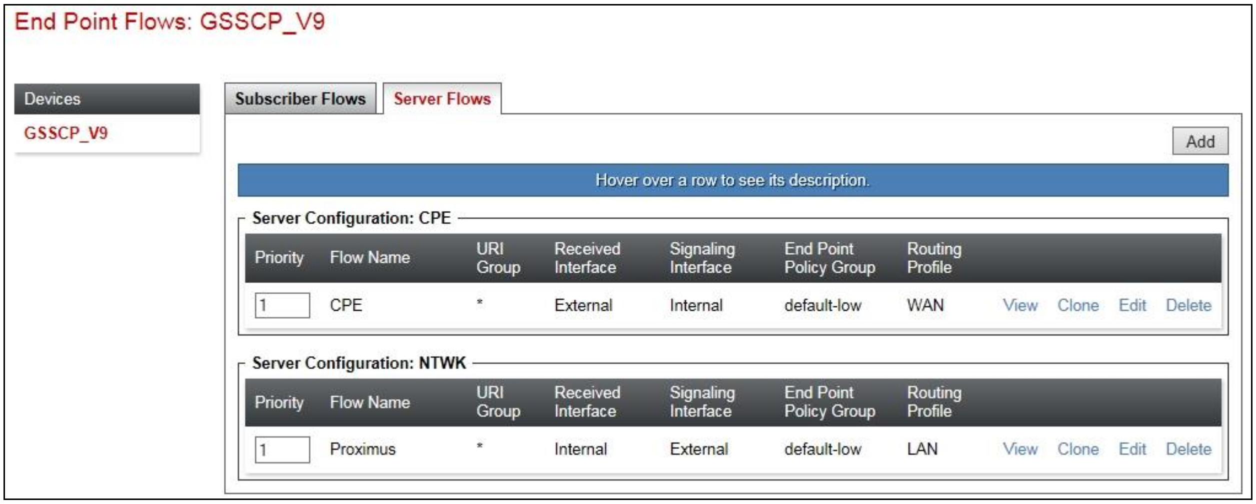


To define a Server Flow for Session Manager, navigate to **Device Specific Settings → End Point Flows**.

- 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 Session Manager, in the test environment **CPE** was used.
- 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 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 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 Session Manager is sent on.
- In the **Routing Profile** drop-down menu, select the routing profile of the Proximus SIP Trunking service defined in **Section 7.7**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of Session Manager defined in **Section 7.8** and click **Finish**.



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



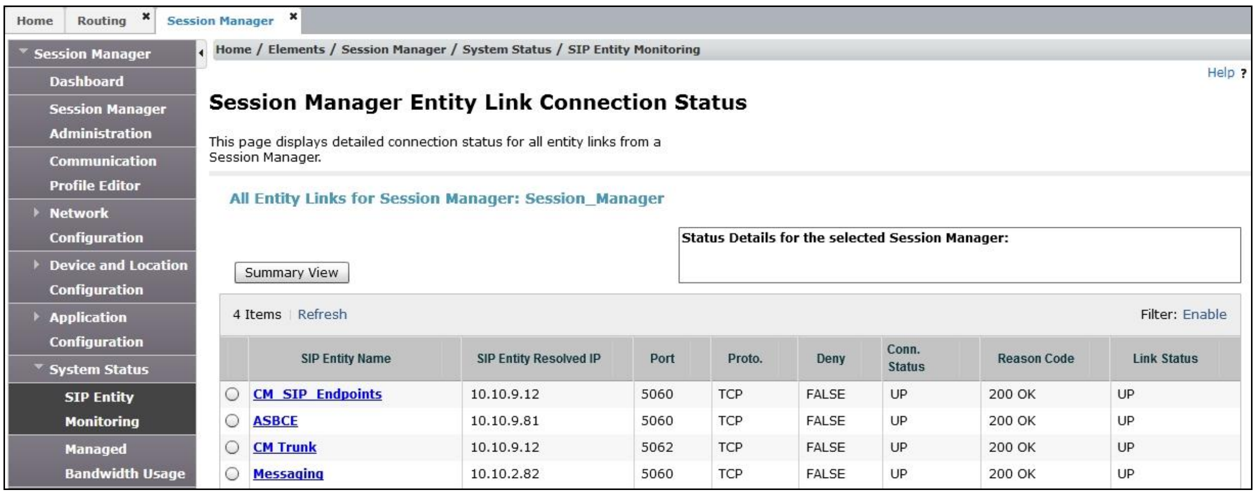
8. Configure the Proximus SIP Trunking service Equipment

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



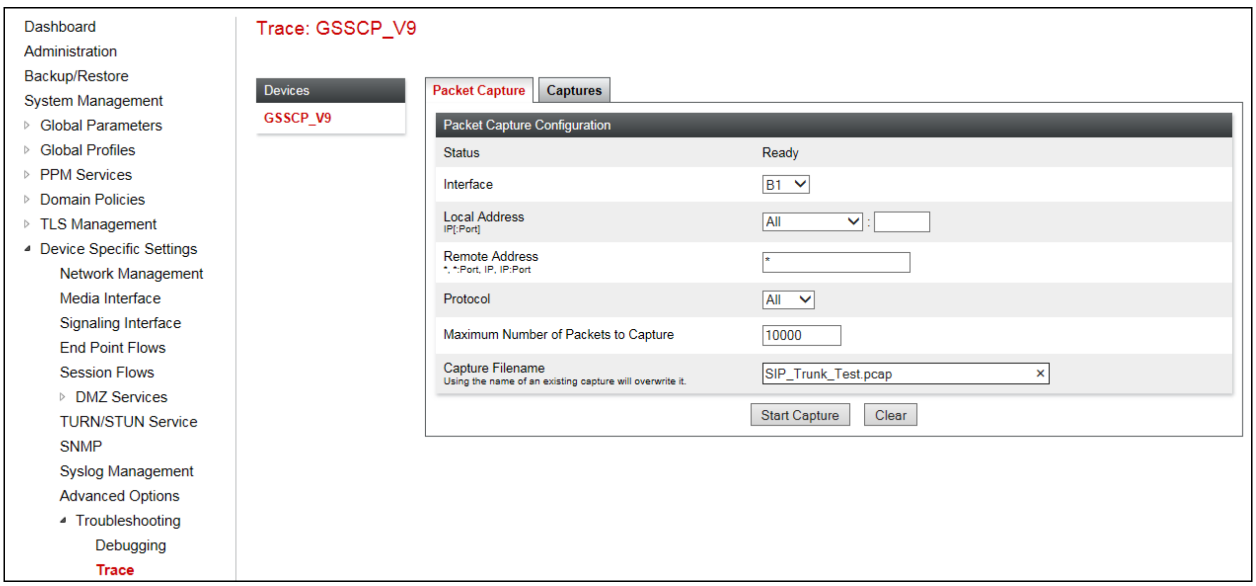
2. From Communication Manager SAT interface run the command **status trunk n** where **n** is a previously configured SIP trunk. Observe if all channels on the trunk group display **in-service/idle**.

```
status trunk 2

                              TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0002/001 T00011   in-service/idle    no
0002/002 T00012   in-service/idle    no
0002/003 T00013   in-service/idle    no
0002/004 T00014   in-service/idle    no
0002/005 T00015   in-service/idle    no
0002/006 T00016   in-service/idle    no
0002/007 T00017   in-service/idle    no
0002/008 T00018   in-service/idle    no
0002/009 T00019   in-service/idle    no
0002/010 T00020   in-service/idle    no
```

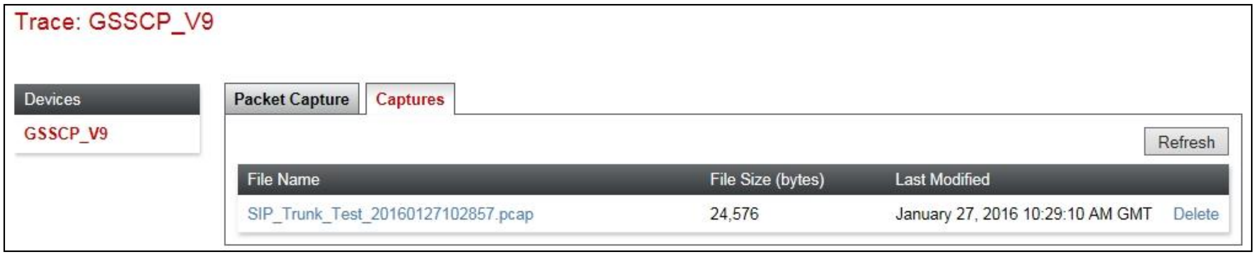
3. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active.
4. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active.
5. Verify that the user on the PSTN can end an active call by hanging up.
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 to OPTIONS in the form of a 200 OK will be seen from the Proximus network.

10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura ® Communication Manager R7.0, Avaya Aura ® Session Manager 7.0 and Avaya Session Border Controller for Enterprise R7.0 to the Proximus SIP Trunking service. The Proximus SIP Trunking service is a SIP-based Voice over IP solution providing businesses a flexible, cost-saving alternative to traditional hardwired telephony trunks. The service was successfully tested with a number of observations listed in **Section 2.2**.

11. Additional References

This section references the documentation relevant to these Application Notes. Additional Avaya product documentation is available at http://support.avaya.com.

[1] *Migrating and Installing Avaya Appliance Virtualization Platform*, Release 7.0.1, Aug 2016.

[2] *Upgrading and Migrating Avaya Aura® applications to 7.0.1 from System Manager*, Release 7.0.1, Aug2016.

[3] *Deploying Avaya Aura® applications*, Release 7.0, Dec 2015

[4] *Deploying Avaya Aura® Communication Manager*, Oct 2016

[5] *Administering Avaya Aura® Communication Manager*, Release 7.0.1, May 2016.

[6] *Deploying Avaya Aura® System Manager*, Release 7.0.1 Aug 2016

[7] *Upgrading Avaya Aura® Communication Manager*, Release 7.0.1, Oct 2016

[8] *Upgrading Avaya Aura® System Manager to Release 7.0.1*, Aug 2016.

[9] *Administering Avaya Aura® System Manager for Release 7.0.1*, Nov 2016

[10] *Deploying Avaya Aura® Session Manager*, Release 7.0.1 Nov 2016

[11] *Upgrading Avaya Aura® Session Manager* Release 7.0.1, Nov 2016

[12] *Administering Avaya Aura® Session Manager* Release 7.0.1, May 2016,

[13] *Deploying Avaya Session Border Controller for Enterprise*, Release 7.0, August 2015

[14] *Upgrading Avaya Session Border Controller for Enterprise*, Release 7.0, August 2015

[15] *Administering Avaya Session Border Controller for Enterprise*, Release 7.0, Jan 2016

[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.