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 Motto VoIP SIP Trunk - Issue 1.0

Abstract

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between the Motto VoIP SIP Trunk 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.

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

Motto 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 Motto VoIP SIP Trunk 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) and Avaya Session Border Controller for Enterprise R7.0 (Avaya SBCE); Endpoints as described in **Section 3**. 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 Motto VoIP 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 connect to the Motto VoIP SIP Trunk.

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 Motto SIP Trunk, calls made to SIP and H.323 telephones at the enterprise.
- Outgoing calls from the enterprise site completed via Motto SIP Trunk to PSTN destinations, calls made from SIP and H.323 telephones.
- Inbound and outbound PSTN calls to/from an Avaya one-X® Communicator and Avaya Communicator for Windows soft phones.
- Calls using the G.711A, G.711MU Law and G.729A codecs.
- Fax calls to/from a group 3 fax machine to a PSTN connected fax machine using G.711.
- 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 Motto SIP Trunk requiring Avaya response and sent by Avaya requiring Motto response.

2.2. Test Results

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

- The recommended method of connecting to the Motto VoIP SIP Trunk is via DNS SRV. This is not yet fully supported on the Avaya SBCE so simple DNS was used.
- OPTIONS were not used by Motto VoIP to check the status of the SIP Trunk
- Occasional delays in signalling were observed during testing resulting in re-transmission of SIP messages. This was due to network issues that have since been resolved.
- When forwarding to a PSTN phone, there was no CLI presented on the PSTN phone. The original CLI was sent in the From header and the DDI of the Communication Manager extension was sent in the Diversion header.
- Although T.38 media attributes were sent in the SDP from the network when negotiating Fax, T.38 fax transmission did not function and Motto advised that it is not supported. G.711 fax transmission failed at first, but was successful after a configuration change in the Motto VoIP network.
- When making an EC500 call, there was no CLI presented on the mobile phone.
- When making an outbound call to a PSTN phone when connected via SIP and in "Other Phone" mode, no ringback is heard on the one-X Communicator soft phone. Ringback is heard when connected via H.323.
- When the SIP Trunk is busy and an incoming call is attempted, the Avaya equipment sends 503 Service Unavailable. The network repeatedly re-attempts the call set-up for a period of around 90 seconds before a tone is played.
- When the signalling link has failed and an incoming call is attempted, the Avaya equipment sends 408 Request Timeout then 503 Service Unavailable. The network repeatedly re-attempts the call set-up for a period of around 90 seconds before a tone is played.

Items not tested include the following:

- No Inbound Toll-Free access available for testing
- No test call was made to Emergency Services as a test call was not booked with the Emergency Services Operator.

2.3. Support

For technical support on the Avaya products described in these Application Notes visit http://support.avaya.com.

For technical support on Motto VoIP products, please contact the Motto VoIP support team:

- E-mail: support@motto.nl
- Phone: +31 454040490
- Web: http://www.motto.nl

3. Reference Configuration

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

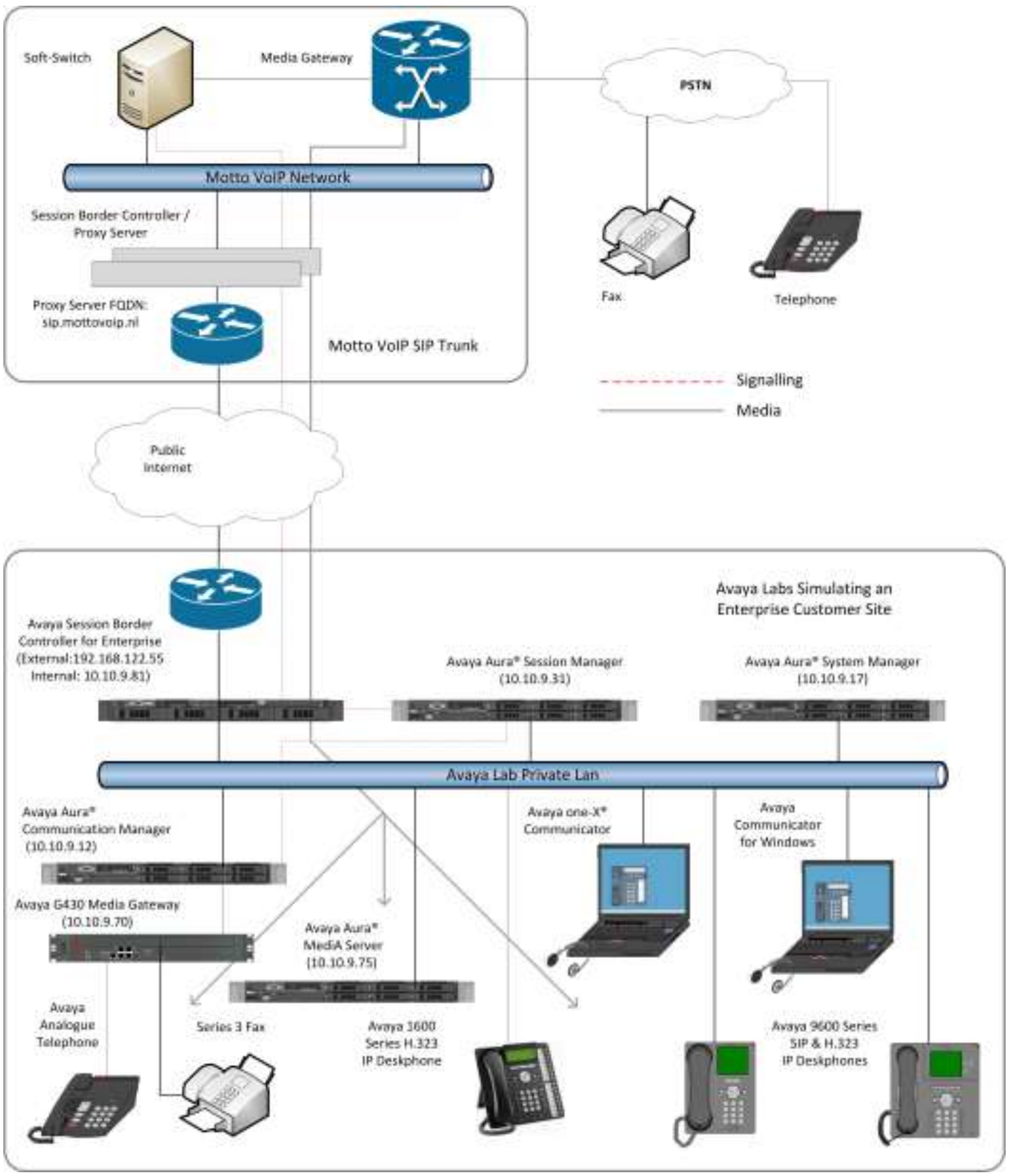


Figure 1: Test Setup Motto VoIP SIP Trunk to Avaya Enterprise

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya	
Avaya Aura® Session Manager	7.0.1.0.701007
Avaya Aura® System Manager	7.0.1.0.064859 – SP1
Avaya Aura® Communication Manager	7.0.1.0.0-23012 – FP1
Avaya Session Border Controller for Enterprise	7.0.1-03-8739
Avaya Media Server	7.7.0.334
Avaya G430 Media Gateway	37.38.0
Avaya 9600 series Handsets SIP 96x0 SIP 9608 H.323 96x0 H.323 9608 H.323 1616	 2.6.16 7.0.1 R46 3.2.6A 6.2.29 1.3.9
Avaya One-X Communicator	6.2.11.03 – SP11
Avaya Communicator for Windows	2.1.3.80
Analogue Handset	N/A
Analogue Fax	N/A
Motto	
OpenSIPS	2.1.3
Asterisk	11.14.0-motto3

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 Motto VoIP SIP Trunk. 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 Motto 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 Motto VoIP SIP Trunk 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 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                                               Page   1 of   2
                                  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 direct media is used on a PSTN call, 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 and Avaya Media Server (not shown) ip-network-region 1 was used in such a way that either one could be selected at call set-up.

5.4. Administer IP Codec Set

Open the IP Codec Set form for the codec set specified in the IP Network Region form in **Section 5.3** by typing **change ip-codec set n w**here **n** is the chosen value of the configuration for the SIP Trunk. Enter the list of audio codec's eligible to be used in order of preference. For the interoperability test the codecs supported by Motto were configured, namely **G.711A**,**G.711MU** 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.711MU          n          2        20
 3: G.729A           n          2        20
```

Note: Wideband codec G.722 is also supported but only for on-net calls. These were not tested during SIP compliance testing.

Motto SIP Trunk supports G.711 for transmission of fax. As this is in-band and requires no interaction from Communication Manager, there is no specific configuration required. Navigate to **Page 2** and set the **FAX - Mode** to **off**.

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

                          IP CODEC SET

                              Allow Direct-IP Multimedia? n



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

Note: Transmission of fax is only supported where G.711 is the codec negotiated at call set-up.

5.5. Administer SIP Signaling Groups

This signalling group (and trunk group) will be used for inbound and outbound PSTN calls to the Motto VoIP SIP Trunk. 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 Session Manager interface (node name **Session_Manager** as defined in the **IP Node Names** form shown in **Section 5.2**).
- Set **Near-end Listen Port** and **Far-end Listen Port** as required. The standard value for TCP is **5060**, though **5062** was used in test to separate the SIP Trunk from the SIP endpoints on 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 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** to avoid unnecessary use of MGW resources
- Set **Initial IP-IP Direct Media** to **n** to facilitate the use of Early Media.
- Leave **DTMF over IP** at default value of **rtp-payload** (Enables **RFC2833** for DTMF transmission from Communication Manager).

The default values for the other fields may be used.

```
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? n
H.323 Station Outgoing Direct Media? n             Alternate Route Timer(sec): 6
```

5.6. Administer SIP Trunk Group

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

- Set the **Group Type** field to **sip**.
- Choose a descriptive **Group Name**.
- Specify a trunk access code (**TAC**) consistent with the dial plan.
- The **Direction** is set to **two-way** to allow incoming and outgoing calls.
- Set the **Service Type** field to **public-netwrk** if the Diversion header is to be supported.
- Specify the 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 Motto 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 E.164 format without a leading "+" as required by Motto.

```
add trunk-group 2                                                     Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y



   Suppress # Outpulsing? n  Numbering Format: private
                                                UUI Treatment: service-provider

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

On **Page 4** of this form:

- Set **Mark Users as Phone** to **y**.
- Set **Send Transferring Party Information** to **y** to ensure that the transferring party number is sent. This information is used by the Motto VoIP network for call transfer.
- Set **Network Call Redirection** to **y** to allow the use of REFER messages for call flows such as blind call transfer.
- Set **Support Request History** to **n** as this header is not supported by Motto.
- Set the **Telephone Event Payload Type** to **101** to match the value preferred by Motto (this Payload Type is not applied to calls from SIP end-points).
- Set **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? y
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                       Send Transferring Party Information? y
                                  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: Testing was carried out with set **Network Call Redirection** to **y**, but this is only required if network call redirection is to be supported which wasn't the case during SIP compliance testing. It's shown here as it allows the use of SIP REFER messages as mentioned in the bullet points.

5.7. Administer Calling Party Number Information

Use the **change private- numbering** command to configure Communication Manager to send the calling party number in the format required. During testing, calling party numbers were sent as E.164 numbers without leading “+”. 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                             4     Total Administered: 8
 4  2000             2           311028nnnn0       11       Maximum Entries: 540
 4  2001             2           311028nnnn8       11
 4  2291             2           311028nnnn2       11
 4  2316             2           311028nnnn3       11
 4  2391             2           311028nnnn1       11
 4  2400             2           311028nnnn4       11
 4  2401             2           311028nnnn7       11
```

Note: During testing the extension numbers were reformatted to E.164 numbers for Trunk Group 2 only. The numbers were analysed for Trunk Group 1 but not reformatted.

The public numbering table was similarly populated for completeness. This table can be used in cases where AAR and ARS analysis are not used. The main difference between the two tables is that the numbers in the public numbering table are prefixed with a “+”. To change the table, use the **change public-unknown-numbering** command.

```
change public-unknown-numbering 0                                 Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                          Total
Ext Ext             Trk        CPN         CPN
Len Code            Grp(s)     Prefix      Len
                                                 Total Administered: 8
 4  2               1                      4        Maximum Entries: 240
 4  2000            2          311028nnnn0  11
 4  2001            2          311028nnnn8  11   Note: If an entry applies to
 4  2291            2          311028nnnn2  11   a SIP connection to Avaya
 4  2316            2          311028nnnn3  11   Aura(R) Session Manager,
 4  2391            2          311028nnnn1  11   the resulting number must
 4  2400            2          311028nnnn4  11   be a complete E.164 number.
 4  2401            2          311028nnnn7  11
                                                 Communication Manager
                                                 automatically inserts
                                                 a '+' digit in this case.
```

Note: If SIP endpoints are registering as third party endpoints, i.e. not as AST devices, check the above tables for completeness.

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 Motto network. 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 international numbers beginning 00 and national 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 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
    0035391              13   13    2        pubu         n
    1                    3    4     2        pubu         n
    118                  5    6     2        pubu         n
    3                    4    4     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 **unk-unk**.

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

5.9. Administer Incoming Digit Translation

This step configures the settings necessary to map incoming DDI calls to Communication Manager extensions. The incoming digits sent in the INVITE message from Motto can be manipulated as necessary to route calls to the desired extension. Use the **change inc-call-handling-trmt trunk-group x** command where **x** is the Trunk Group defined in **Section 5.6**. In the example shown, 11 digits are received in E.164 format with no prefix. All digits are deleted and the extension number is inserted. Note that some of the DDI digits have been obscured.

```
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    11 311028nnnn0      11  2000
 public-ntwrk    11 311028nnnn1      11  2391
 public-ntwrk    11 311028nnnn2      11  2291
 public-ntwrk    11 311028nnnn3      11  2316
 public-ntwrk    11 311028nnnn4      11  2400
 public-ntwrk    11 311028nnnn5      11  7000
 public-ntwrk    11 311028nnnn6      11  6002
 public-ntwrk    11 311028nnnn7      11  2401
 public-ntwrk    11 311028nnnn8      11  2001
 public-ntwrk
```

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 2291. Use the command **change off-pbx-telephone station-mapping x** where **x** is Communication Manager station.

- The **Station Extension** field will automatically populate with station extension.
- For **Application** enter **EC500**.
- Enter a **Dial Prefix** if required by the routing configuration, none was required during testing.
- For the **Phone Number** enter the phone that will also be called (e.g. **003538941nnnn7**).
- Set the **Trunk Selection** to **ars** so that the ARS table will be used for routing.
- Set the **Config Set** to **1**.

```
change off-pbx-telephone station-mapping 2291                    Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 2291             OPS          -     2291              aar        1
 2291             EC500        -     003538941nnnn7    ars        1
```

Note: The phone number shown is for a mobile phone in the Avaya Lab. To use facilities such as Feature Name Extension (FNE) 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.

The additional line in the previous screenshot with **Application** of **OPS** is standard on SIP endpoints where the phone is registered to the Session Manager and is essentially “Off PBX”.

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

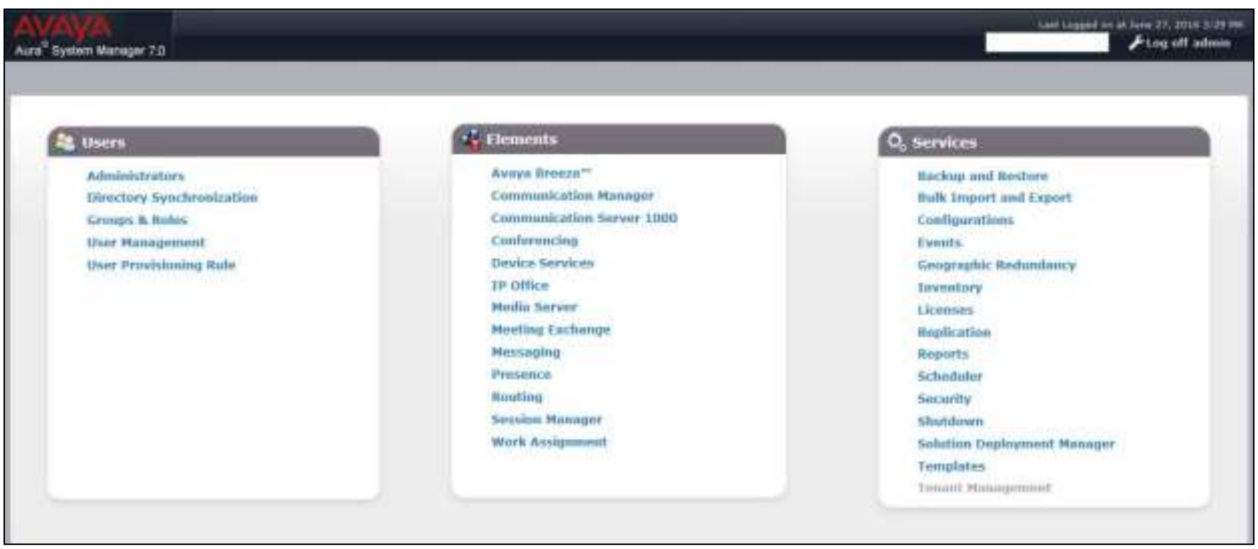
6. Configuring Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The Session Manager is configured by opening a web browser to 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 and enter **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.



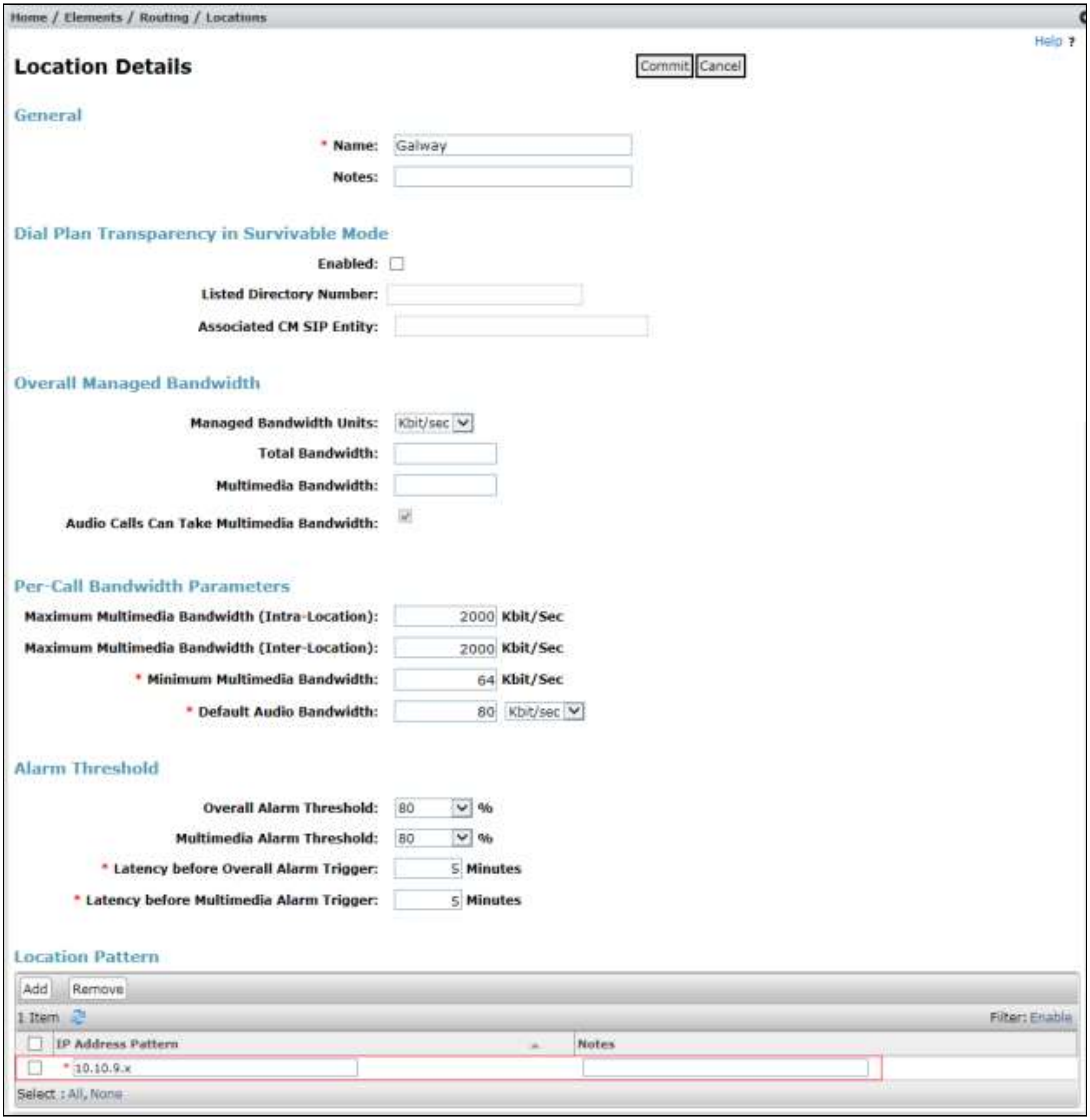
6.2. Administer SIP Domain

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 Motto; 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 (see **Section 6.4**).

6.3. Administer Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for the purposes of bandwidth management. One location is added to the sample configuration for all of the enterprise SIP entities. 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.

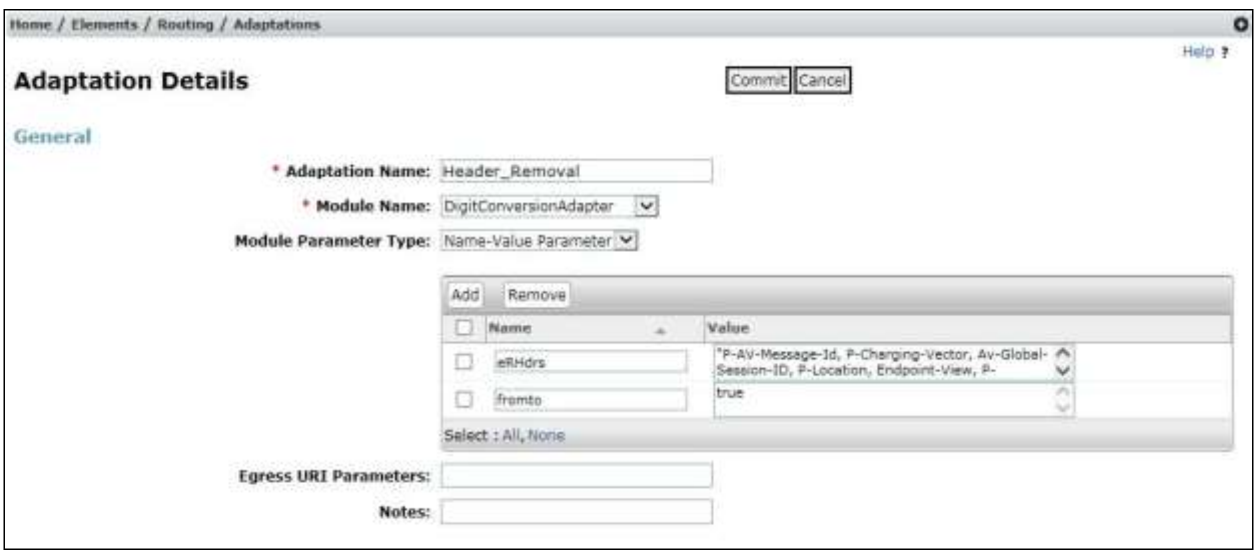


6.4. Administer Adaptations

Communication Manager and Session Manager make use of Avaya proprietary SIP headers to facilitate the full suite of Avaya functionality within the enterprise. These are not required on the SIP trunk however, and are not recognized by the Motto network. In addition, the called and calling party number formats passed between the Enterprise and the Motto VoIP network are in E.164 format without any prefix. A Session Manager Adaptation is used both to remove proprietary headers and to convert numbers to and from diallable format.

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** box, type **eRHdrs**.
- In the **Value** box, type the list of headers to be deleted. During testing, the following list was used: **"P-AV-Message-Id, P-Charging-Vector, Av-Global-Session-ID, P-Location, Endpoint-View, P-Conference"**.
- Click on Add.
- In the **Name** box, type **fromto**.
- In the **Value** box, type **true**. This will apply the number conversion rules to the From and To headers in the SIP messages.



Number analysis is used to apply the above Module Parameter rule and to convert the called and calling party numbers between E.164 and diallable format. Scroll down and in the section **Digit Conversion for Incoming Calls to 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.

The screenshot below shows analysis of called and calling party numbers for incoming calls. The called party number is the DDI number associated with the Communication manager extensions.

- 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, in this case the DDI number length is fixed at **11**.
- Under **Delete Digits** enter 0 as the number is not to be modified.
- Leave the **Insert Digits** field blank as the number is not to be modified.
- Under **Address to Modify** choose **destination** from the drop down box to apply this rule to the called party number.


Digit Conversion for Incoming Calls to SM

Add Remove

2 Items Filter: Enable

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
	* 311028nnnn	* 11	* 11		* 0		destination		

Select : All, None

Note: In the above screenshot the DDI number is partially obscured. If the calling party number is to be modified for display on Communication Manager extensions in diallable format, it should be done here. For international numbers, prefix with 00. For national numbers, analyse country code 31 and replace with 0. **Address to modify** would be **origination**.

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 going out to the network

The screenshot below shows analysis of called party numbers for outgoing calls. The called party number is the dialled public number.

- Under **Matching Pattern** enter the first dialled digits. For international calls, these will be **00**. For national calls, these will be **0**.
- Under **Min** and **Max** enter the Minimum and Maximum digits of the dialled number.
- Under **Delete Digits** enter **2** for international numbers and **1** for national.
- Under **Insert Digits**, enter the Netherlands country code for national numbers.
- Under **Address to Modify** choose **destination** from the drop down box to apply this rule to the called party number.


Digit Conversion for Outgoing Calls from SM

Add Remove

2 Items Filter: Enable

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
	* 0	* 8	* 15		* 1	31	destination		
	* 00	* 10	* 17		* 2		destination		

Select : All, None

Commit Cancel

Click **Commit** to save changes.

Note: For international calls, the maximum number length would be that specified by E.164, i.e. 15 without the international dialling prefix, **17** in total. The maximum number length for national calls was set to **15** during testing. This value is not critical as long as it is the same or higher than the maximum according to the national numbering plan.

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**:

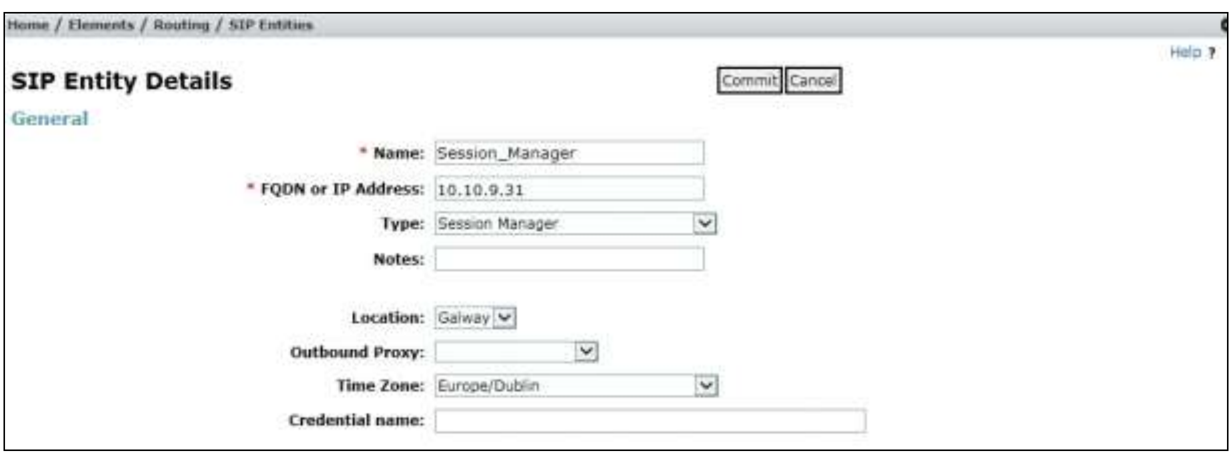
- In the **Name** field enter an informative name.
- In the **FQDN or IP Address** field enter the IP address of Session Manager or the signalling interface on the connecting system.
- In the **Type** field use **Session Manager** for a Session Manager SIP Entity, **CM** for a Communication Manager SIP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

6.5.1. Avaya Aura® Session Manager SIP Entity

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

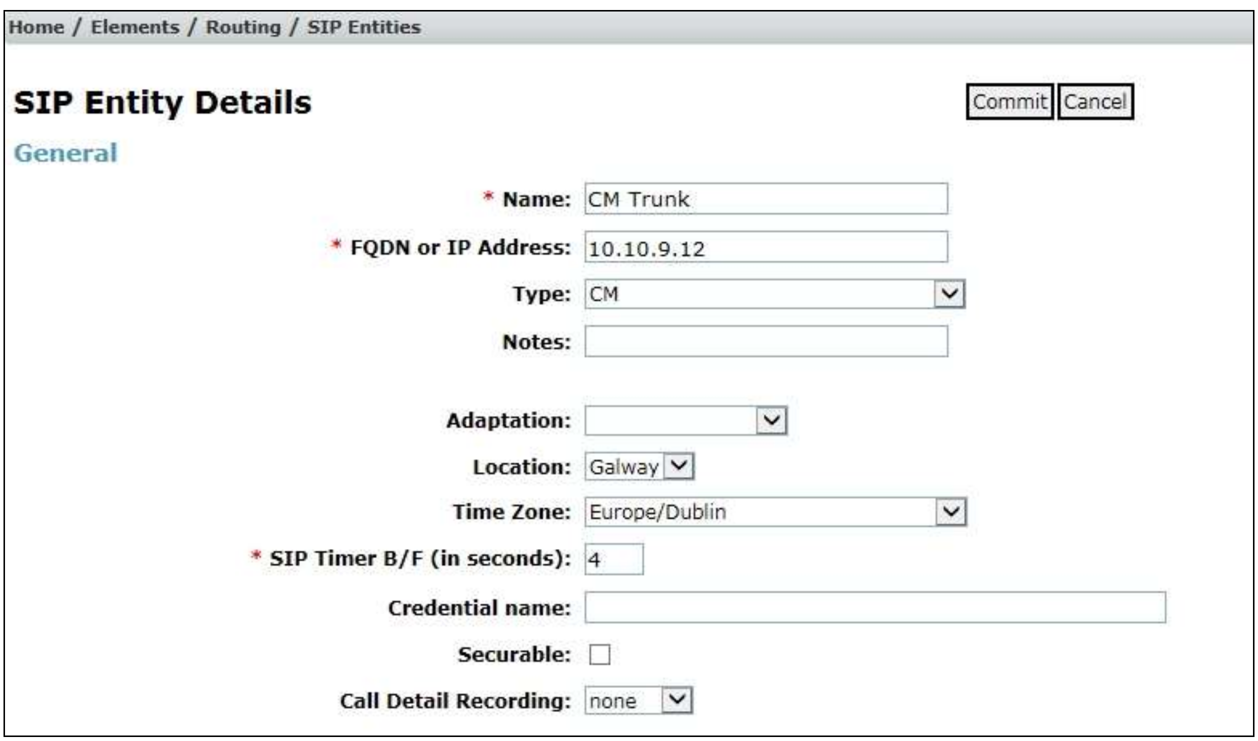


The Session Manager must be configured with the port numbers on the protocols that will be used by the other SIP entities. To configure these scroll to the bottom of the page and under **Port**, click **Add**, then edit the fields in the resulting new row.

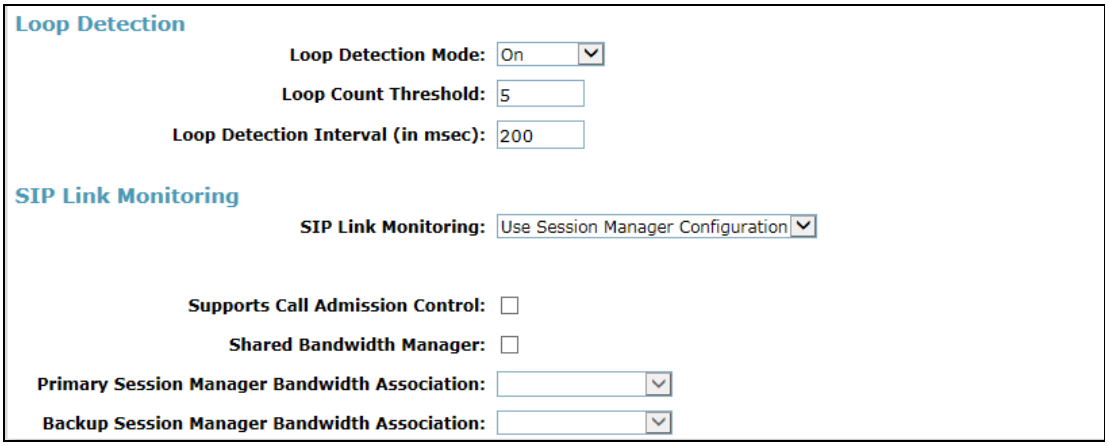
- In the **Port** field enter the port number on which the system listens for SIP requests.
- In the **Protocol** field enter the transport protocol to be used for SIP requests.
- In the **Default Domain** field, from the drop down menu select the domain added in **Section 6.2** as the default domain.

6.5.2. Avaya Aura® Communication Manager SIP Entities

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



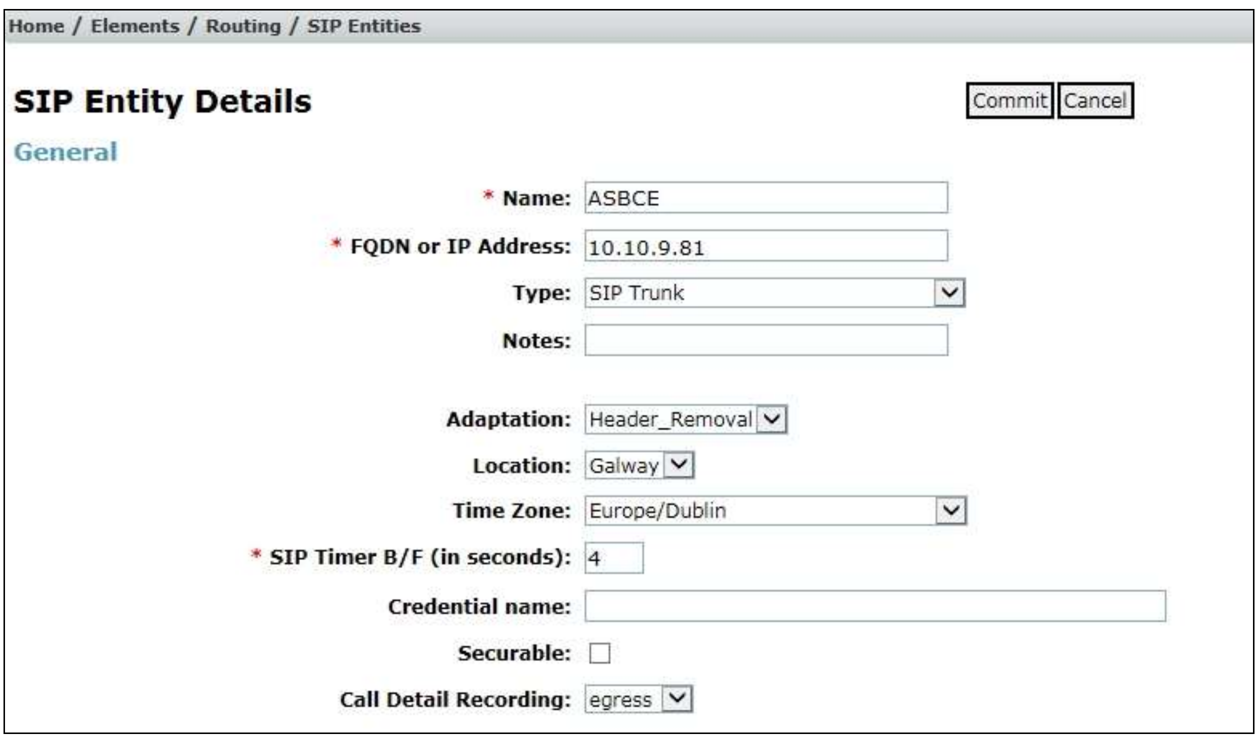
Other parameters can be set for the SIP Entity as shown in the following screenshot, but for test, these were left at default values.



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

6.5.3. Avaya Session Border Controller for Enterprise SIP Entity

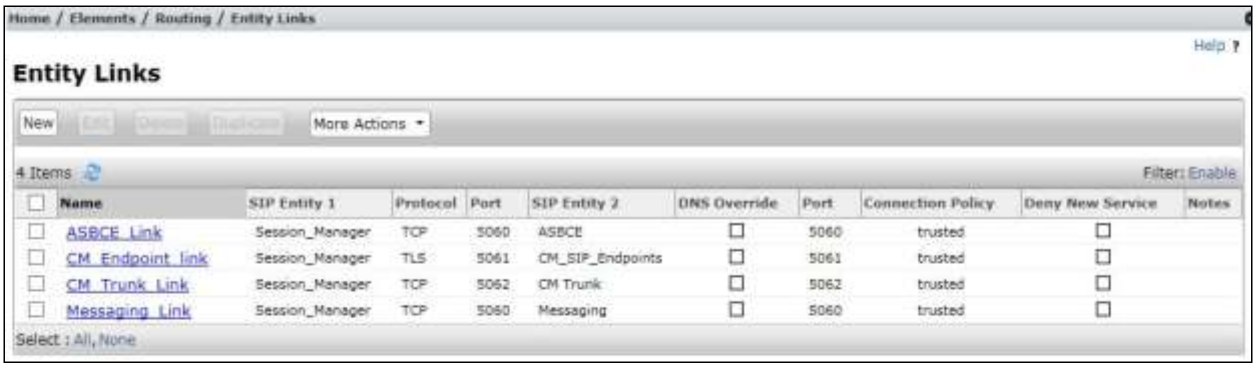
The following screen shows the SIP Entity for the Avaya SBCE used for PSTN destinations. The **FQDN or IP Address** field is set to the IP address of the Avaya SBCE private network interface used for PSTN fixed calls (see **Figure 1**). Set the **Adaptation** to that defined in **Section 6.4**, the **Location** 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 previous screen shows the Entity Links used in this configuration.

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

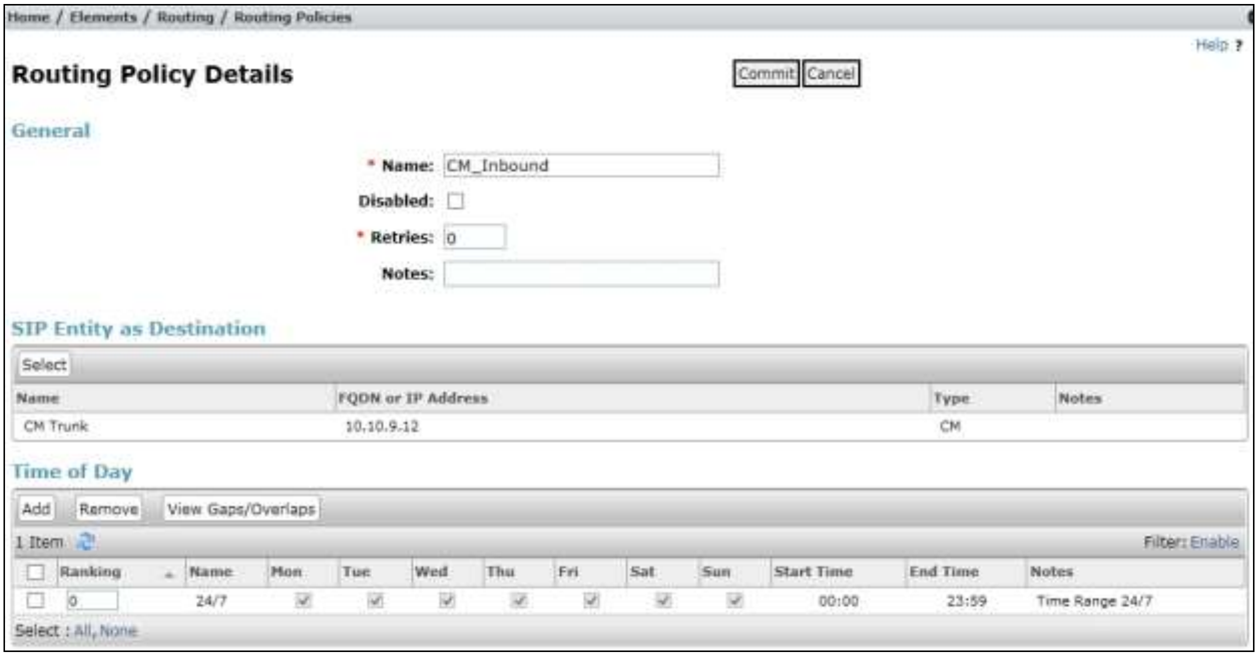
6.7. Administer Routing Policies

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).

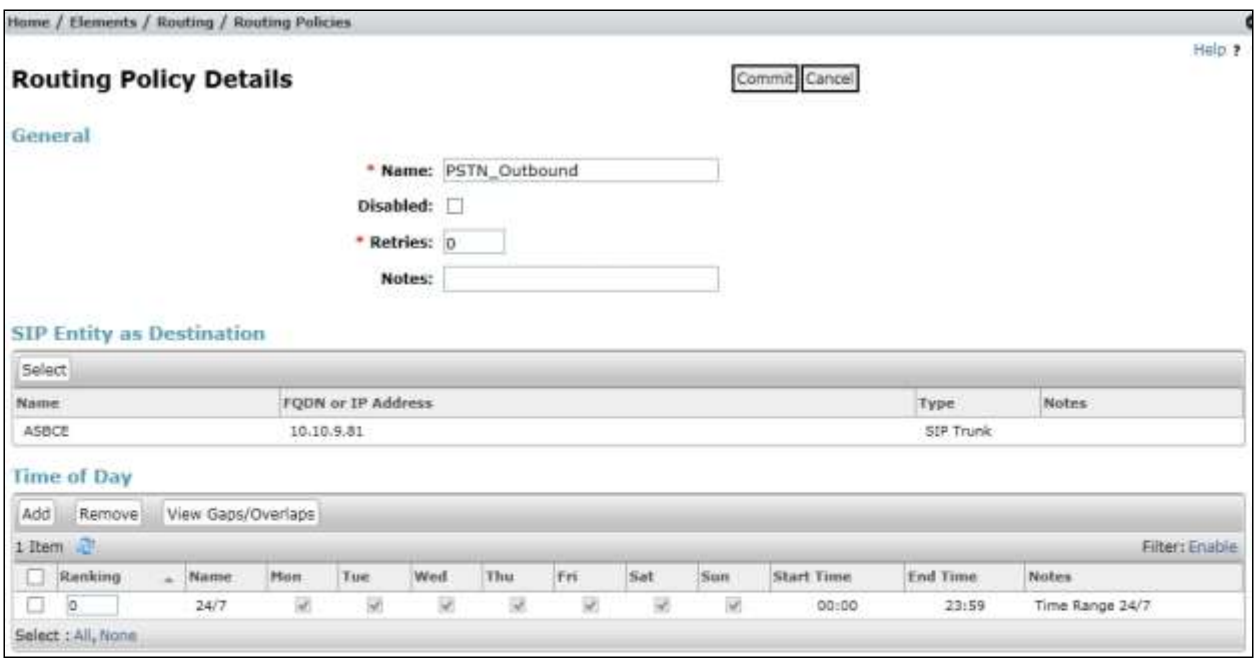
Under **General**:

- 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 PSTN destinations via the Motto VoIP SIP Trunk.



6.8. Administer Dial Patterns

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

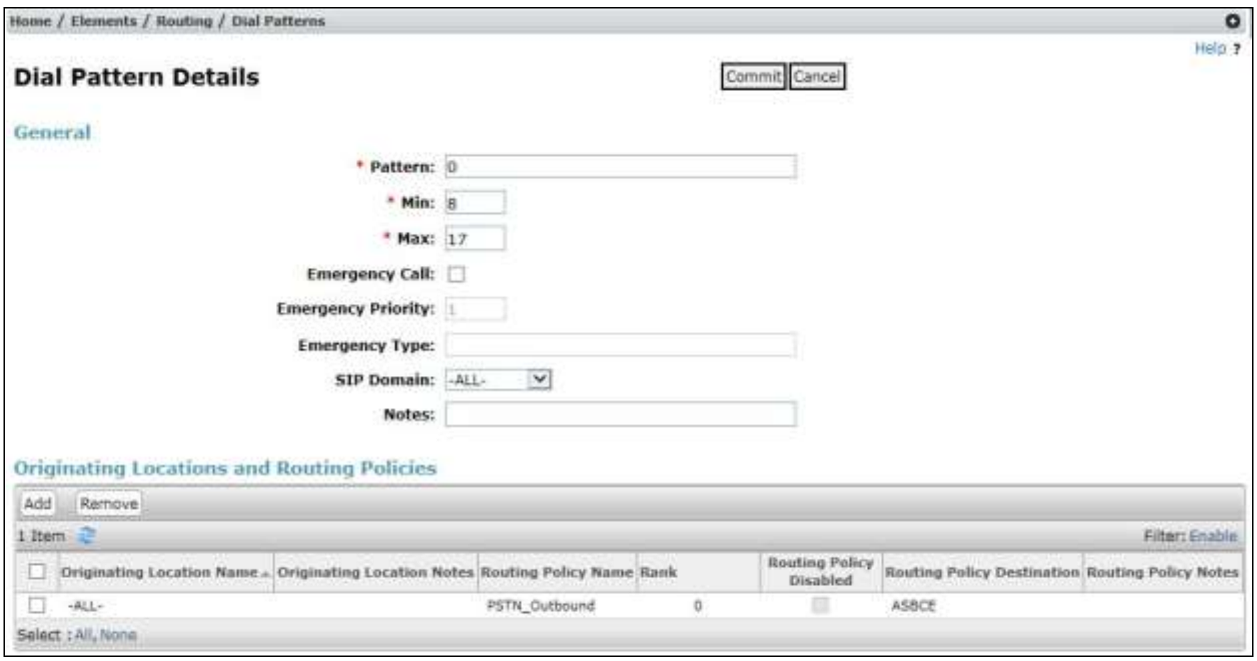
Under **General**:

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

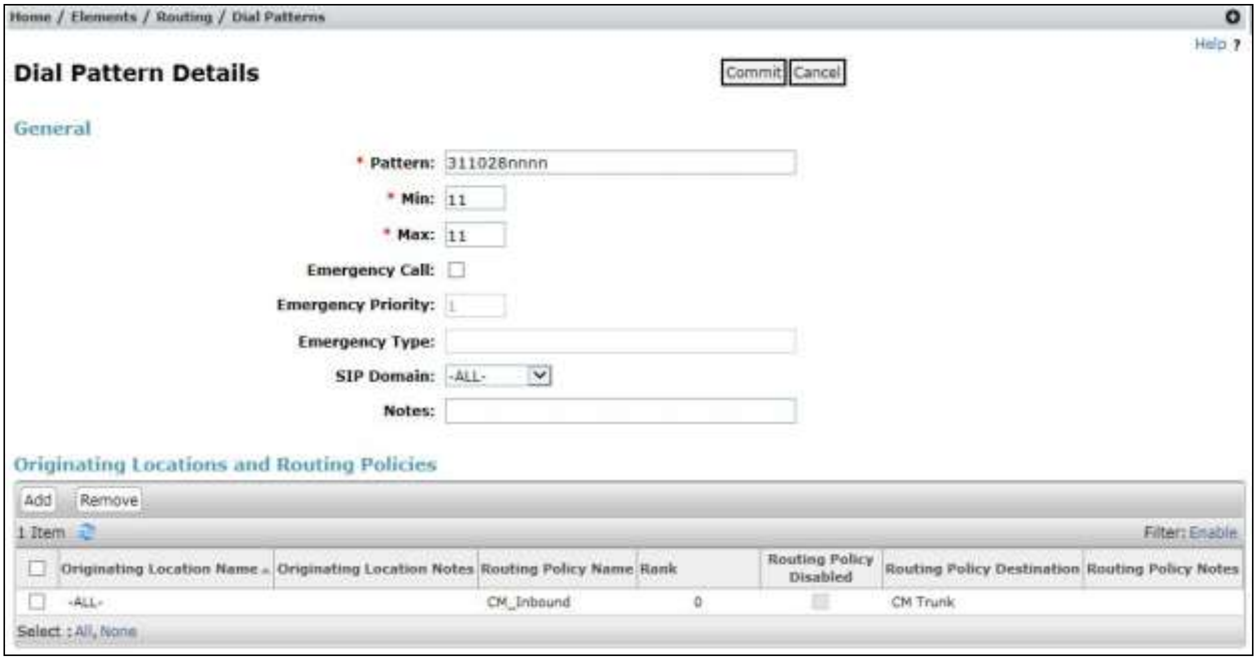
Under **Originating Locations and Routing Policies**:

- Click **Add**, 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 to PSTN destinations via the Motto VoIP SIP Trunk.



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

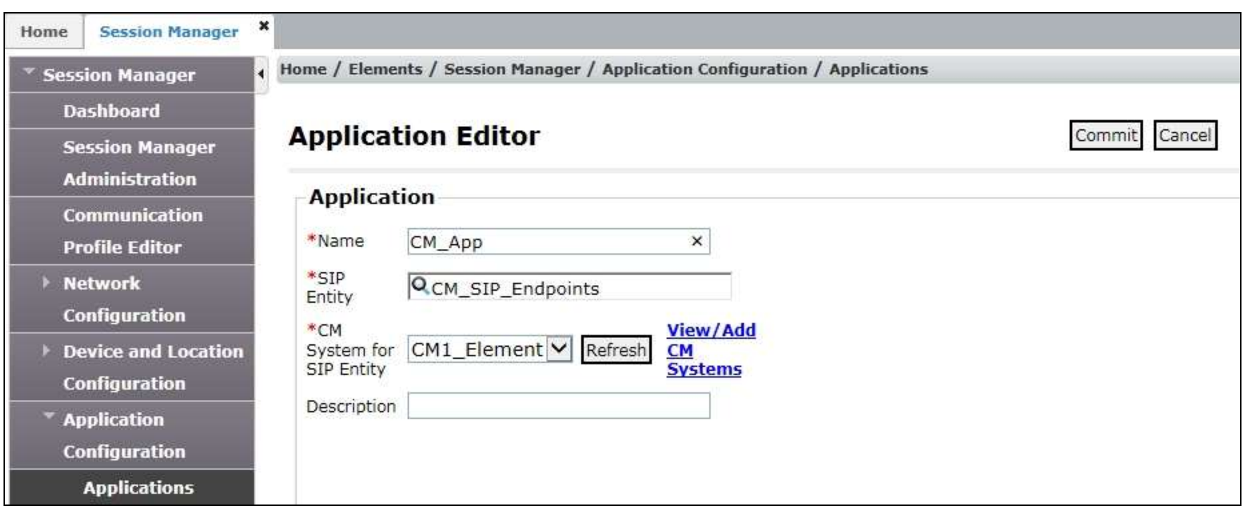


Note: The above configuration is used to analyze the DDI numbers assigned to the extensions on Communication Manager. Some of the digits of the pattern to be matched have been obscured.

6.9. Administer Application for Avaya Aura® Communication Manager

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

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

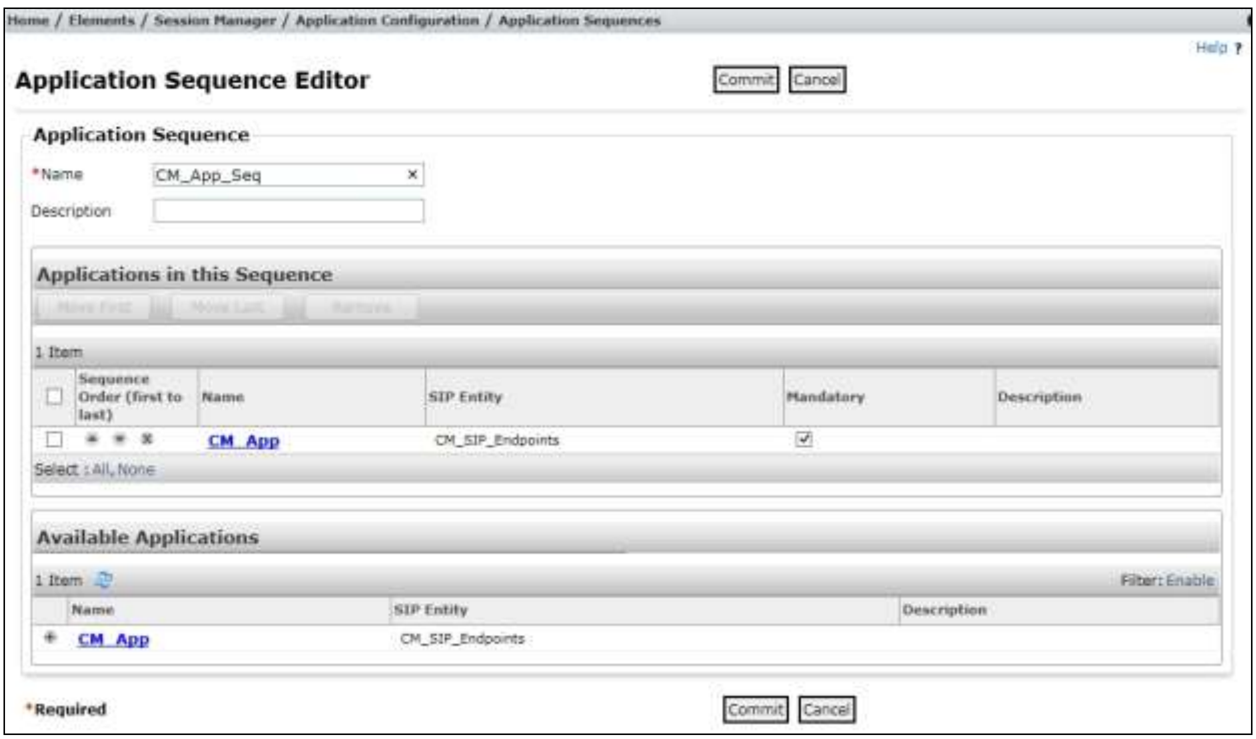


Note: The Application described here and the Application Sequence described in the next section are likely to have been defined during installation. The configuration is shown here for reference. Note also that the Communication Manager SIP Entity selected is that set up specifically for SIP endpoints.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

6.10. Administer Application Sequence for Avaya Aura® Communication Manager

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

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

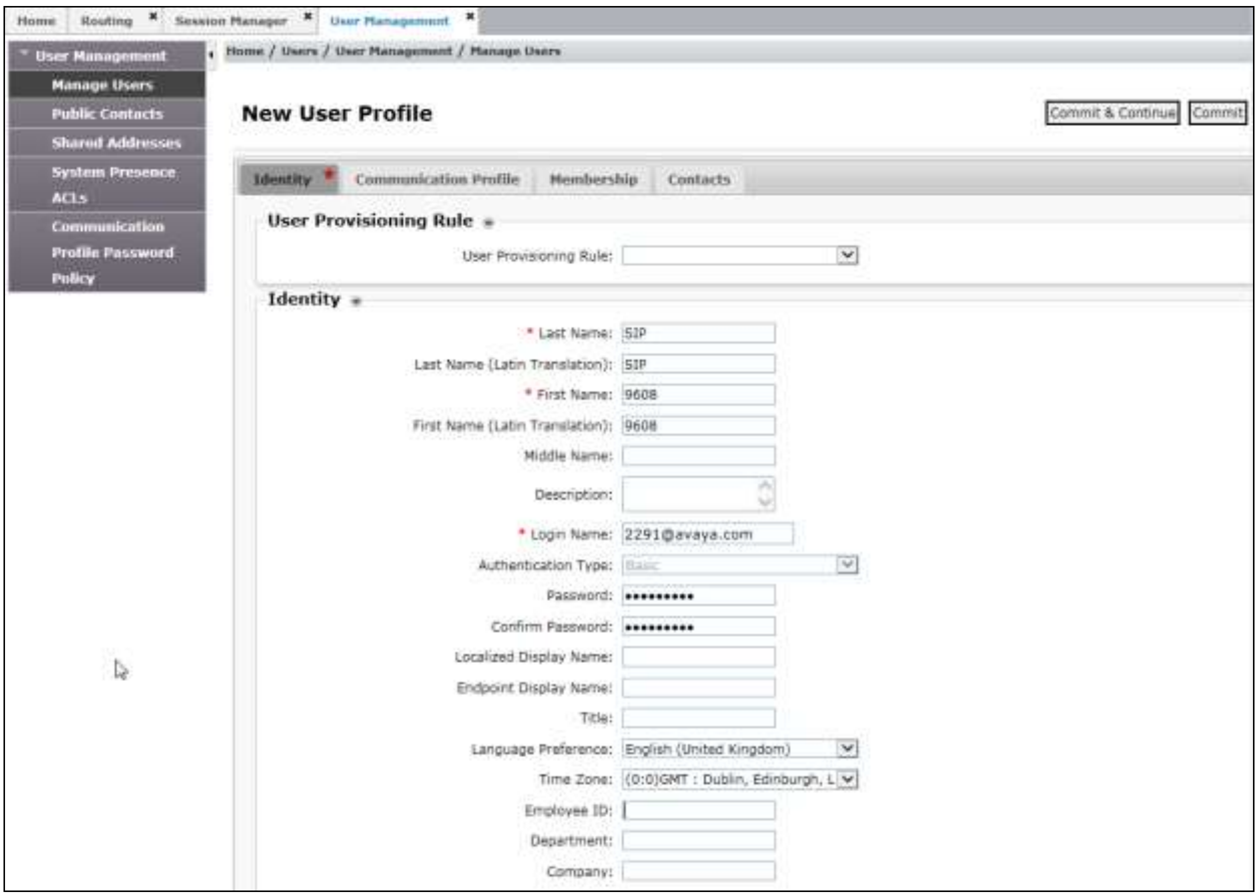


6.11. Administer SIP Extensions

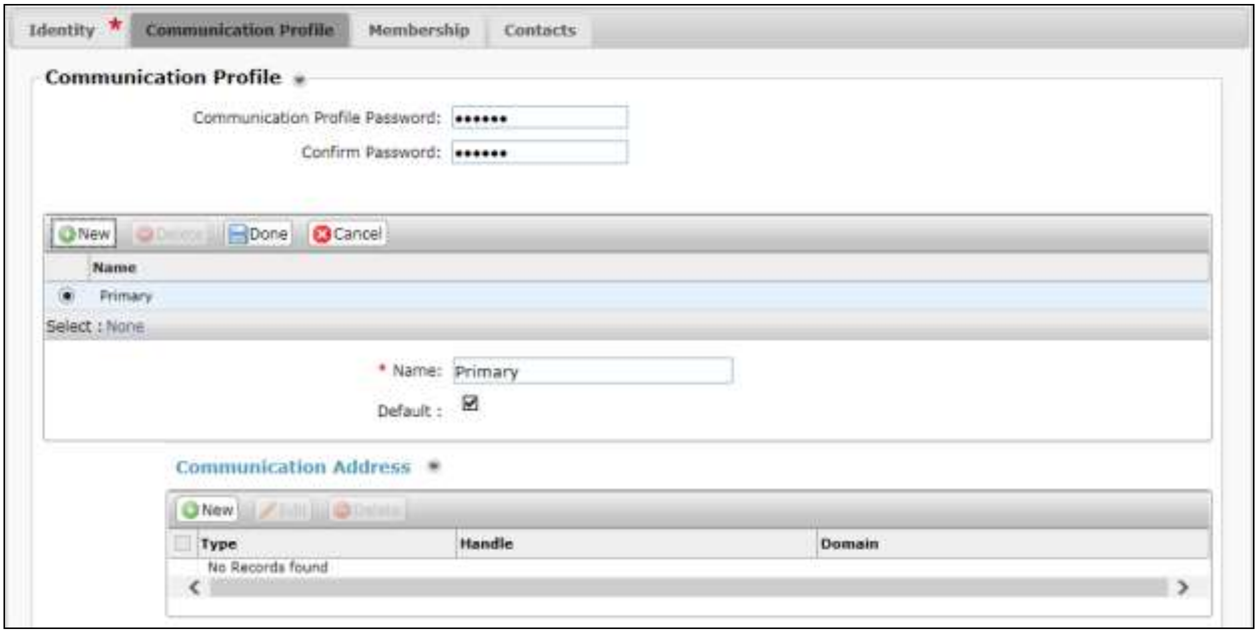
The SIP extensions are likely to have been defined during installation. The configuration shown in this section is for reference. SIP extensions are registered with Session Manager and use Communication Manager for their feature and configuration settings. From the **Home** tab select **User Management**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

On the **Identity** tab:

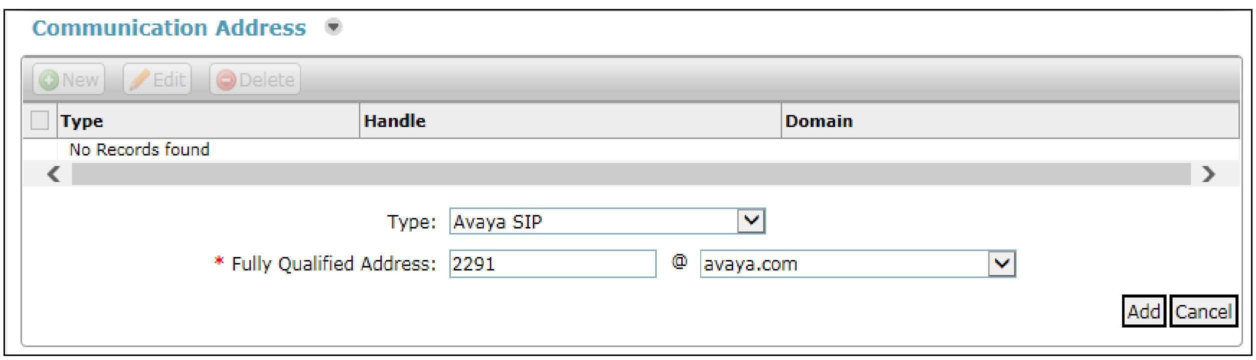
- Enter the user's name in the **Last Name** and **First Name** fields.
- In the **Login Name** field enter a unique system login name in the form of user@domain e.g. 2291@avaya.com which is used to create the user's primary handle.
- The **Authentication Type** should be **Basic**.
- In the **Password/Confirm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



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

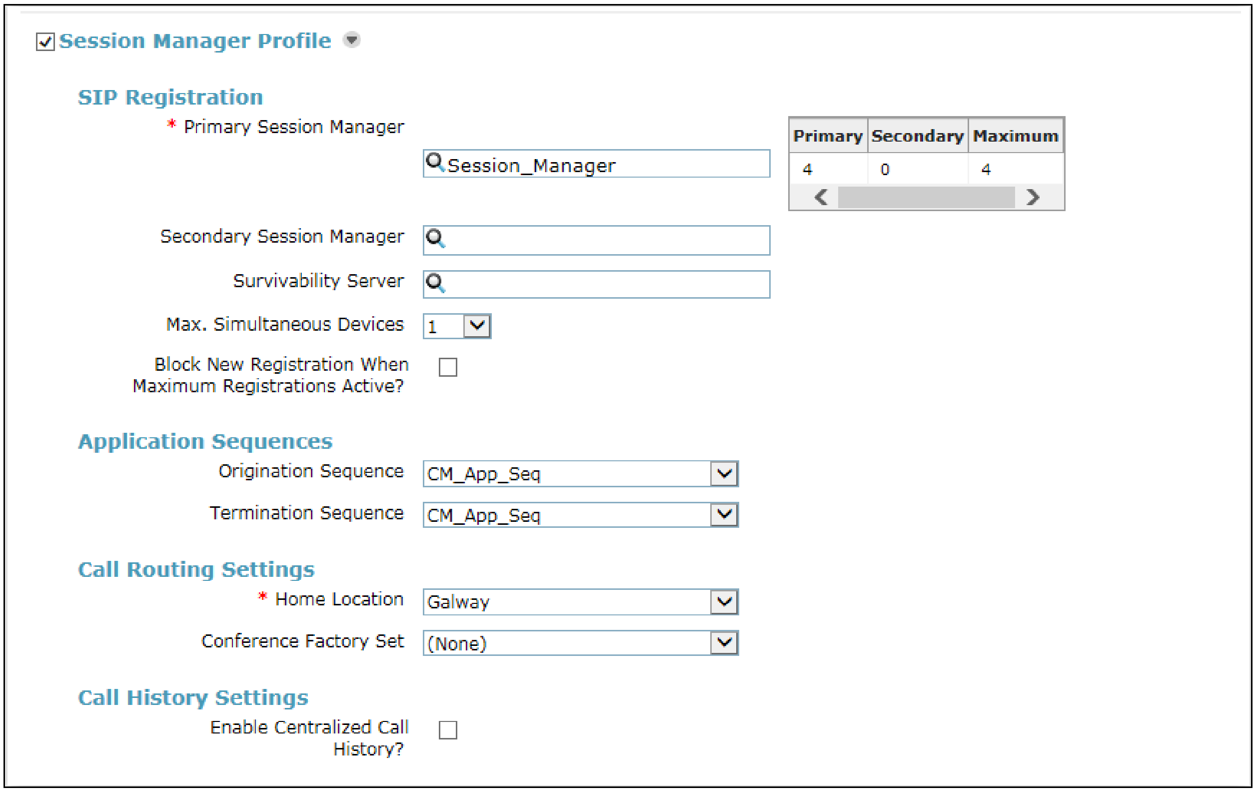


Expand the **Communication Address** section and click **New**. For the **Type** field select **Avaya SIP** from the drop-down menu. In the **Fully Qualified Address** field, enter an extension number and select the relevant domain from the drop-down menu. Click the **Add** button.



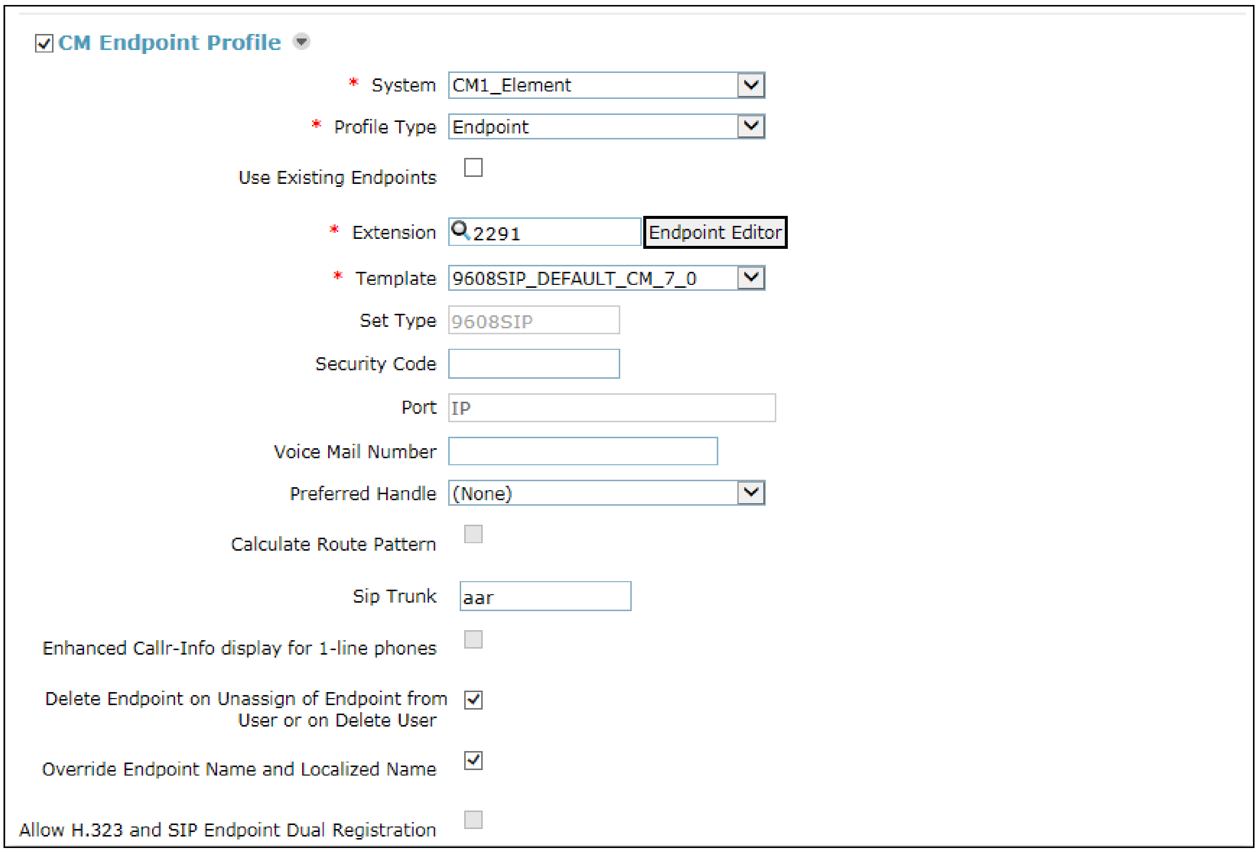
Expand the **Session Manager Profile** section.

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



Expand the **Endpoint Profile** section.

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



7. Configure Avaya Session Border Controller for Enterprise

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

7.1. Access Avaya Session Border Controller for Enterprise

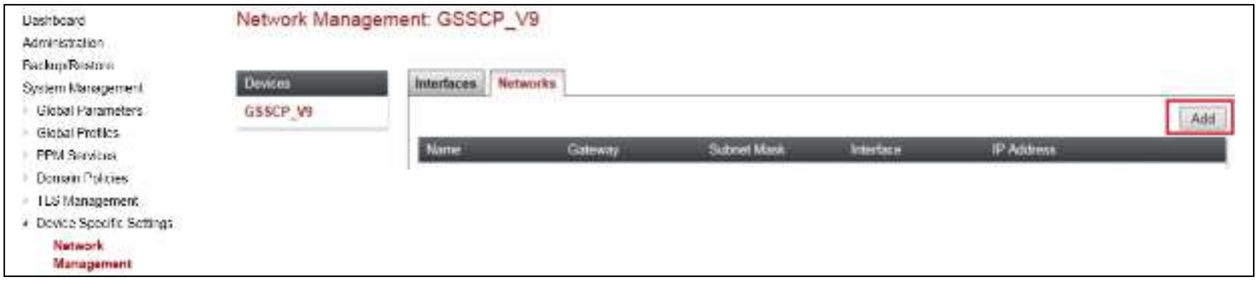
Access the Session Border Controller using a web browser by entering the URL **https://<ip-address>**, where **<ip-address>** is the private IP address configured at installation. A log in screen is presented. Log in using the appropriate username and password.

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

7.2. Define Network Management

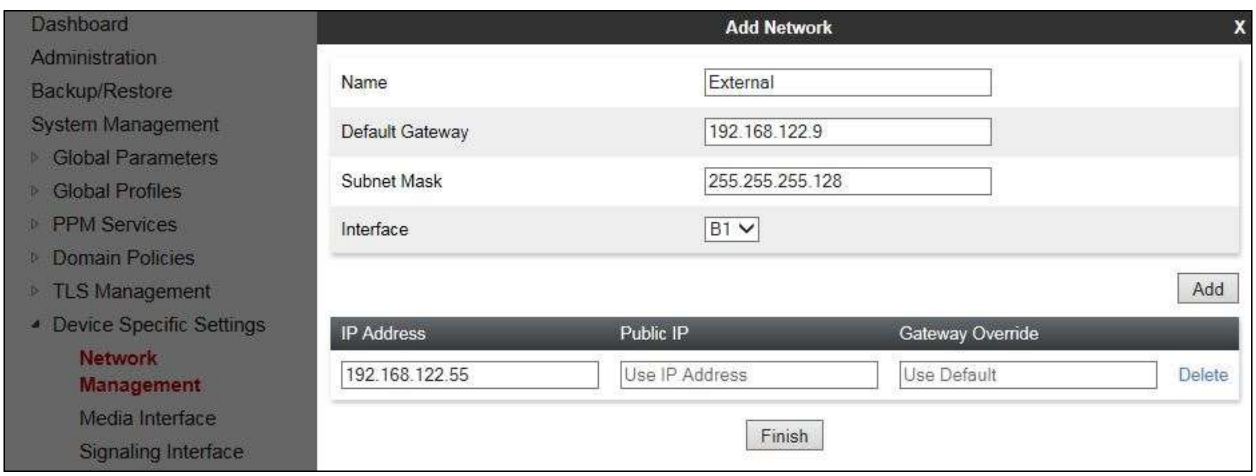
Network information is required on the Avaya SBCE to allocate IP addresses and masks to the interfaces. Note that only the **A1** and **B1** interfaces are used, typically the **A1** interface is used for the internal side and **B1** is used for external. Each side of the Avaya SBCE can have only one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

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



Enter details for the external interfaces in the dialogue box:

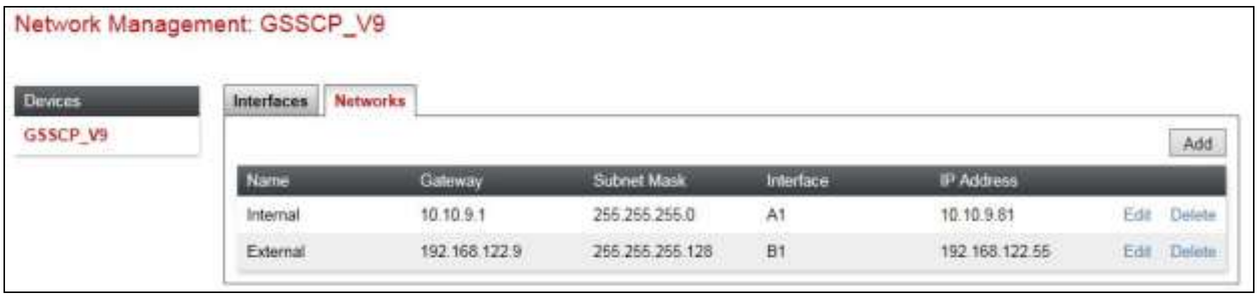
- Enter a descriptive name in the **Name** field.
- Enter the default gateway IP address for the external interfaces in the **Default Gateway** field.
- Enter the subnet mask in the **Subnet Mask** field.
- Select the external physical interface to be used from the **Interface** drop down menu. In the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



Click on **Add** to define the internal 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 interfaces in the **Default Gateway** field.
- Enter the subnet mask in the **Subnet Mask** field.
- Select the internal physical interface to be used from the **Interface** drop down menu. In the test environment, this was **A1.**
- Click on **Add** and an additional row will appear allowing an IP address to be entered.
- Enter the internal IP address for the Avaya SBCE 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 the state. Change the state to **Enabled** where required.

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

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

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

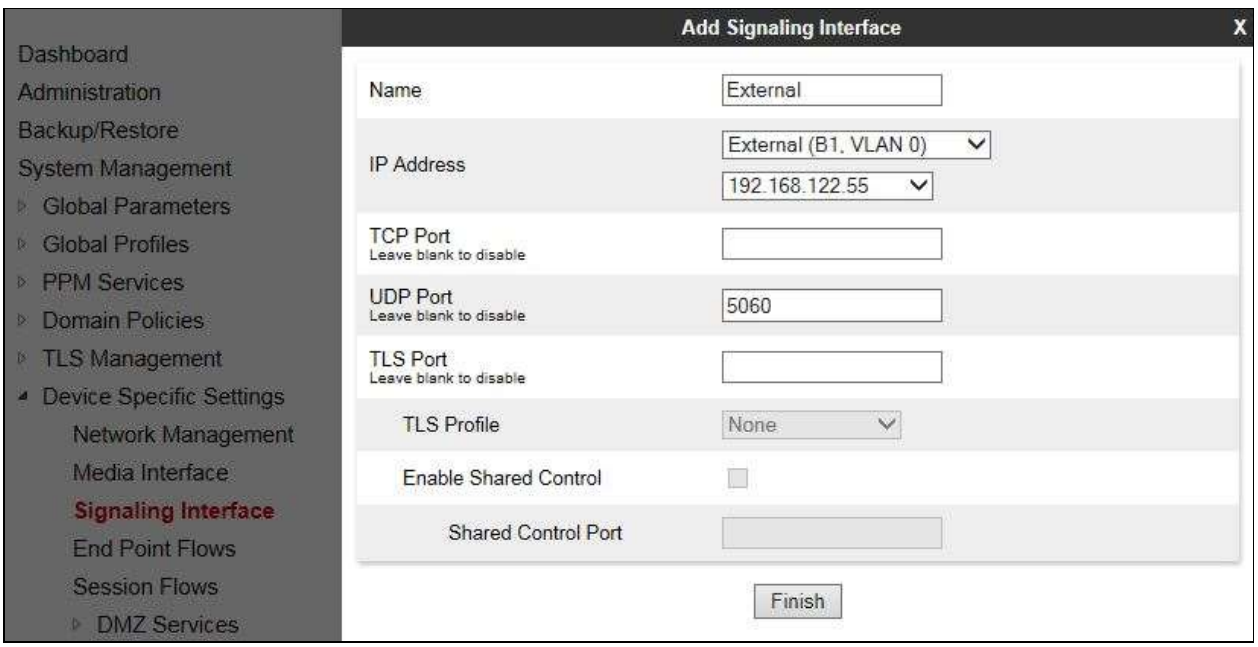
7.3. Define Interfaces

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 Motto VoIP SIP Trunk. 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 IP address **192.168.122.55**.
- Enter the UDP port number in the **UDP Port** field, **5060** is used for the Motto SIP Trunk.



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.

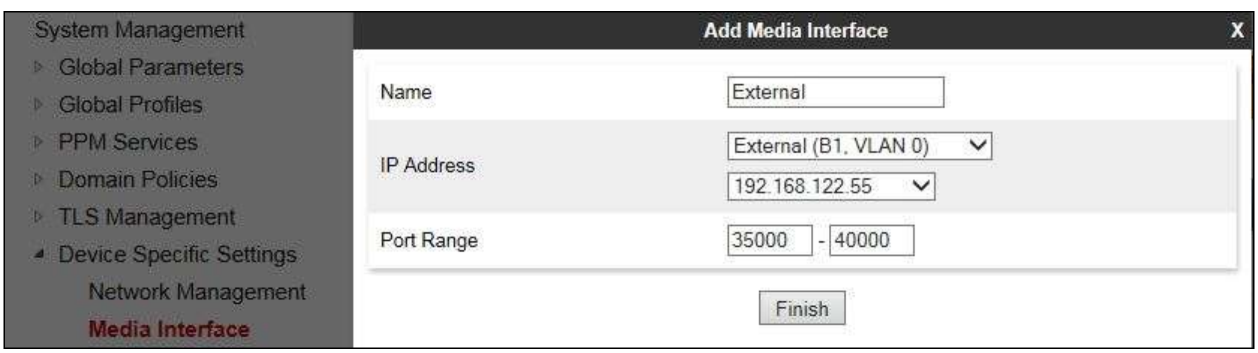
The following screenshot shows details of the signalling interfaces:

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 IP address **192.168.122.55**.
- Define the RTP **Port Range** for the media path with the Motto SIP Trunk, during testing this was left at default values of **35000** to **40000**.



The internal media interfaces are 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:

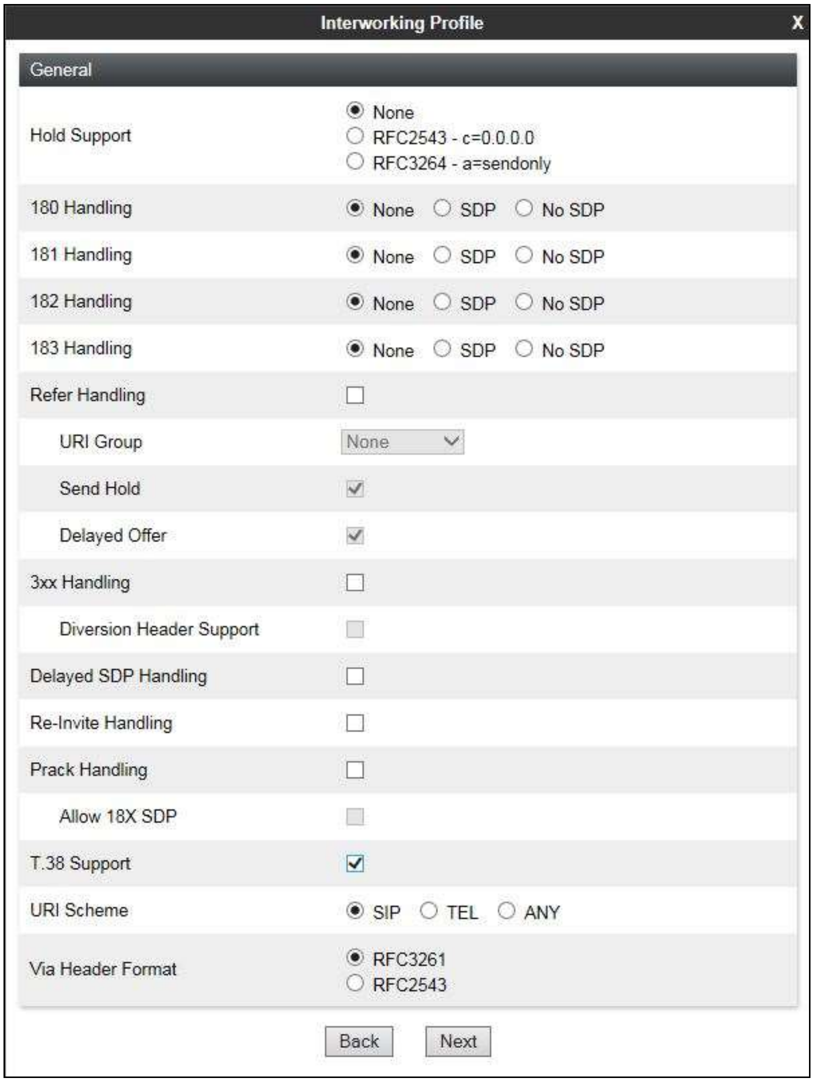
Note: In the test environment, the internal IP address was **10.10.9.81** and the port range was left at default values.

7.4. Define Server Interworking

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

To define server interworking on the Avaya SBCE, navigate to **Global Profiles → Server Interworking** in the main menu on the left hand side. To define Server Interworking for the Motto VoIP SIP Trunk, click on **Add** (not shown). A pop-up menu is generated. In the **Name** field enter a descriptive name for the Motto VoIP network and click **Next**.

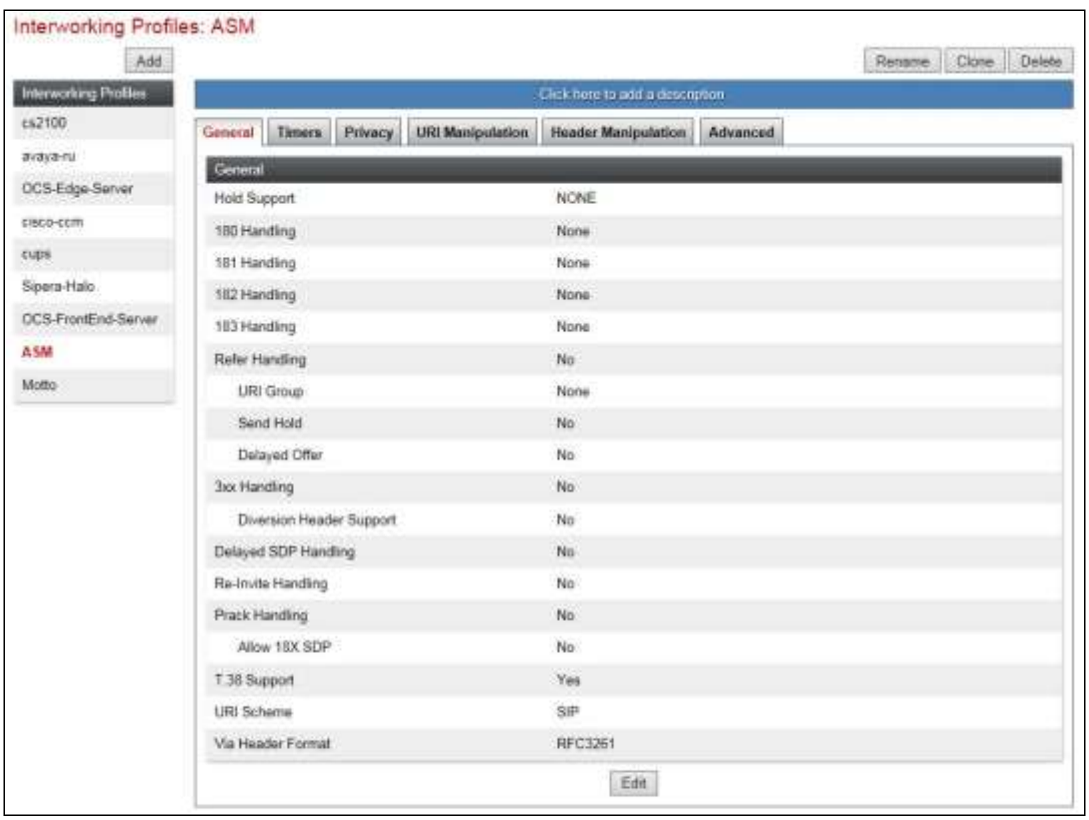
Check the **T.38 Support** box and click on **Next**.



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, leave the **Record Routes** at the default setting of **None** and ensure that the **Has Remote SBC** box is checked. Note that Avaya extensions are not supported for the SIP Trunk. Click on **Finish**

Repeat the process to define Server Interworking for Session Manager using the same parameter settings apart from **Record Routes**. The following screenshot shows the **General** tab.



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

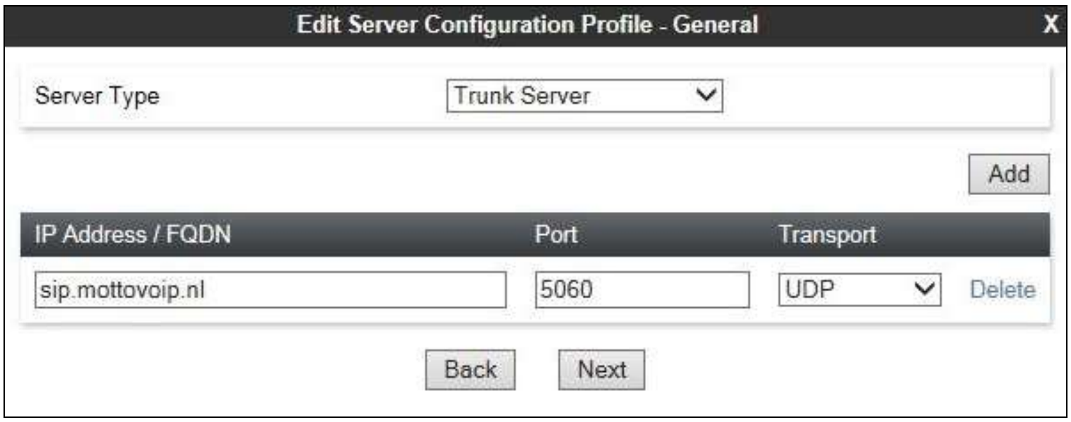
7.5. Define Servers

A server definition is required for each server connected to the Avaya SBCE. The Motto SIP Trunk is connected as a Trunk Server. Session Manager is connected as a Call Server.

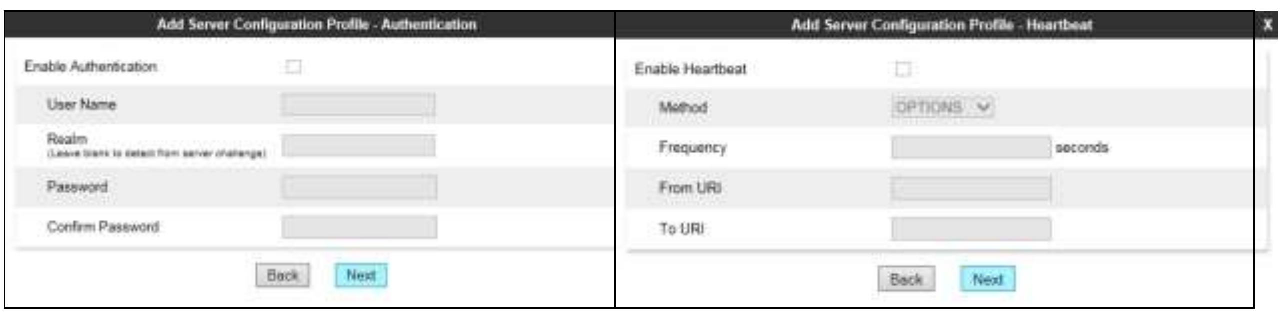
To define the Motto SIP Trunk Server, navigate to **Global Profiles → Server Configuration** in the main menu on the left hand side. Click on **Add** and enter an appropriate name in the pop-up menu.

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 Motto SIP Trunk FQDN.
- 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**.



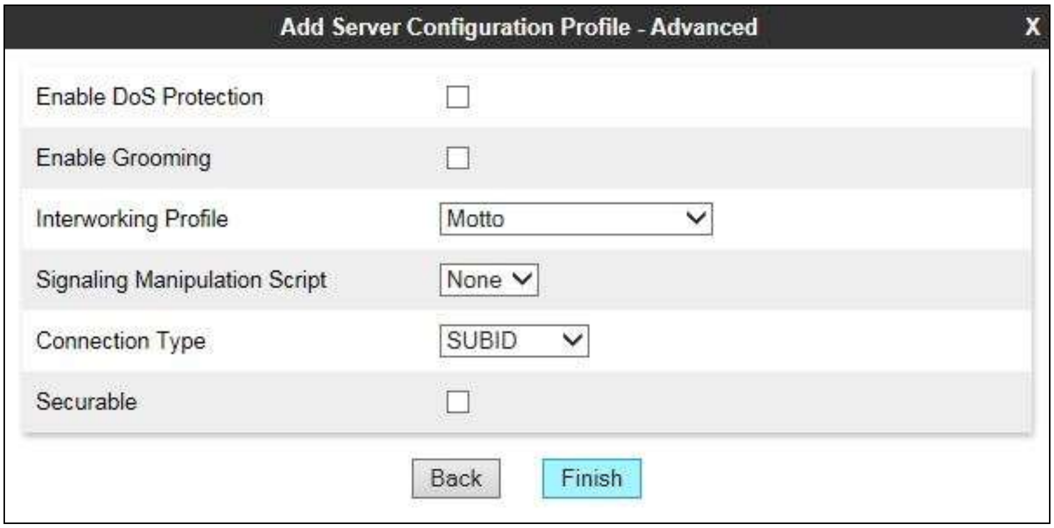
Click on **Next** and **Next** again. Leave the fields in the dialogue boxes at default values.



Click on **Next** again to get to the final dialogue box.

The final dialogue box contains the **Advanced** settings:

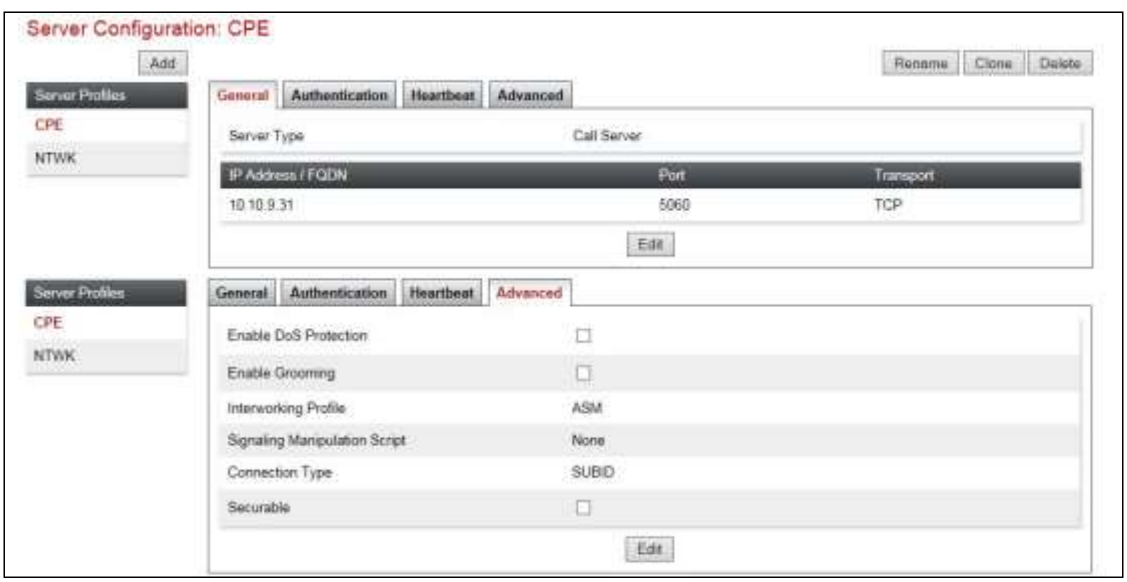
- In the **Interworking Profile** drop down menu, select the **Interworking Profile** for Motto SIP Trunk defined in **Section 7.4**.
- Leave the other fields at default settings.
- Click **Finish**.



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

- Ensure that **Call Server** is selected in the **Server Type** drop down menu in the **General** dialogue box.
- Ensure that the Interworking Profile defined for 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 completed Server Configuration:



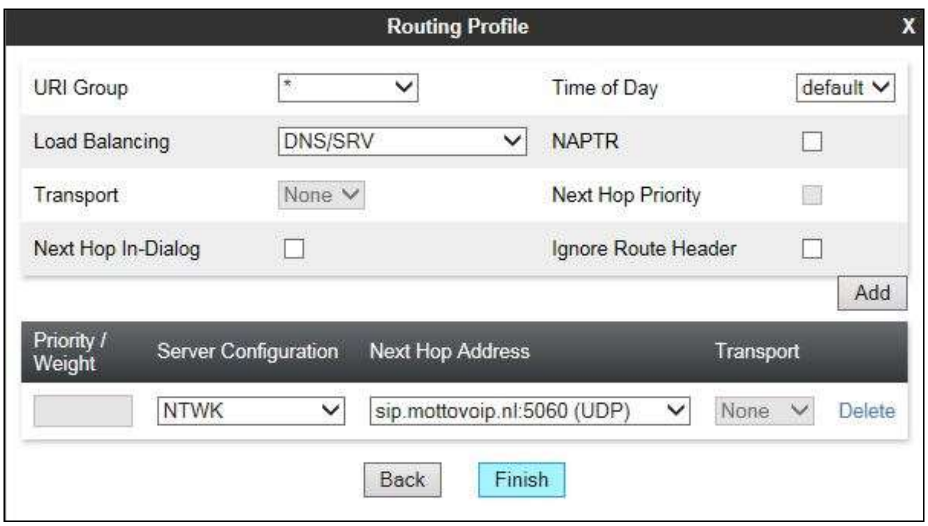
7.6. Define Routing

Routing information is required for routing to the Motto VoIP SIP Trunk 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 Motto 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 for the SIP Trunk:

- During testing, DNS was used to connect to Motto VoIP. To use DNS, select **DNS/SRV** from the **Load Balancing** drop down menu.
- Click on **Add** to specify an FQDN for the SIP Trunk.
- Select the Server Configuration defined in **Section 7.5** in the **Server Configuration** drop down menu. This automatically populates the **Next Hop Address** field
- Click **Finish**.



Note: While DNS SRV can be selected here, the Avaya SBCE does not resolve SRV records. It will only handle simple DNS where an IP address is returned for the DNS query.

Repeat the process for the Routing Profile for Session Manager: The screenshot over the page shows the completed configuration. The **Next Hop Address** in this case is the private IP address of the Session Manager:

7.7. Topology Hiding

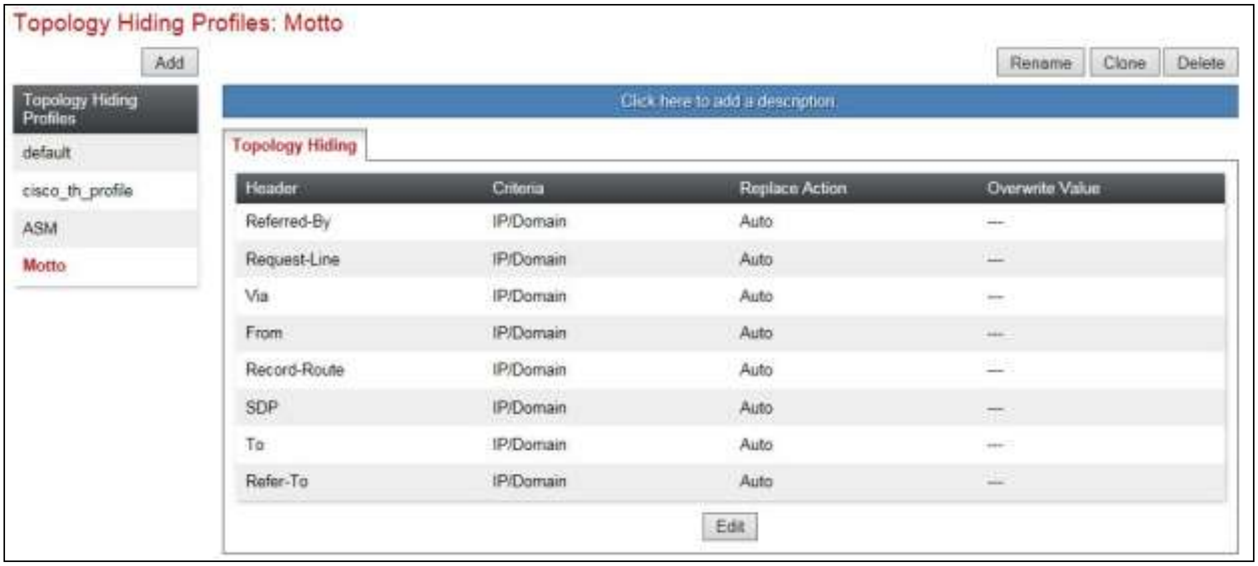
Topology hiding is used to hide local information such as private IP addresses and local domain names. The local information can be overwritten with a domain name or IP address. The default **Replace Action** is **Auto**, this replaces local information with IP addresses, generally the next hop or external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

To define Topology Hiding for the Motto VoIP SIP Trunk, 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** to configure Topology Hiding for each header as required:

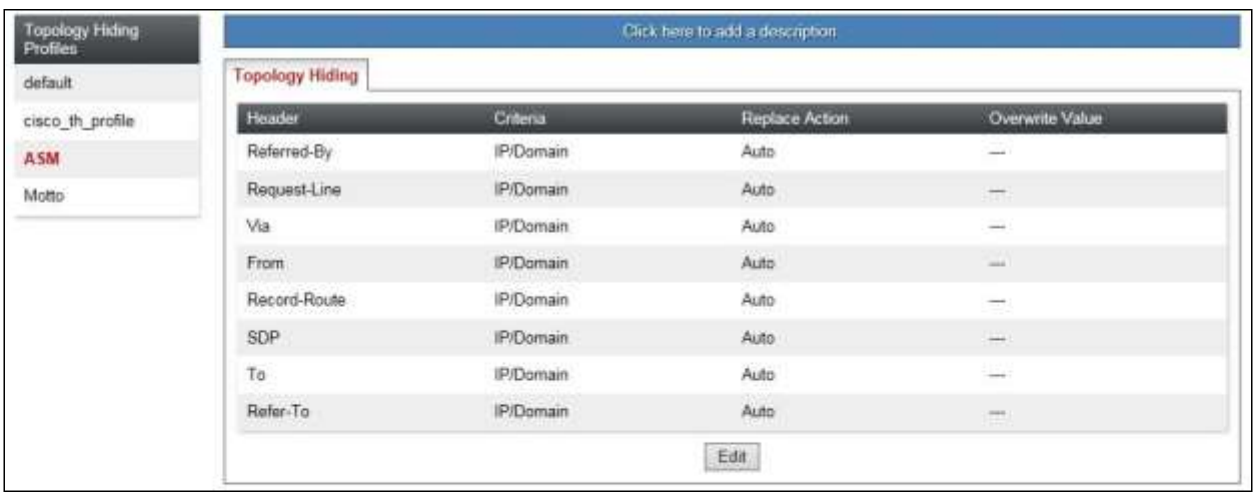
Enter details in the **Topology Hiding Profile** pop-up menu.

- 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. During testing the default **IP/Domain** was used for all headers that hides both domain names and IP addresses.
- Leave the **Replace Action** at the default value of **Auto** unless a specific domain name is required. In this case, select **Overwrite** and define a domain name in the **Overwrite Value** field.
- Topology hiding was defined for all headers where the function is available.

The following screenshot shows the completed **Topology** Hiding configuration for the Motto SIP Trunk.



To define Topology hiding for Session Manager, follow the same process. This can be simplified by cloning the profile defined for Motto SIP Trunk. Do this by highlighting the profile defined for Motto and clicking on **Clone**. Enter an appropriate name for Session Manager and click on **Next** (not shown). Make any changes where required, none were made in the test environment.

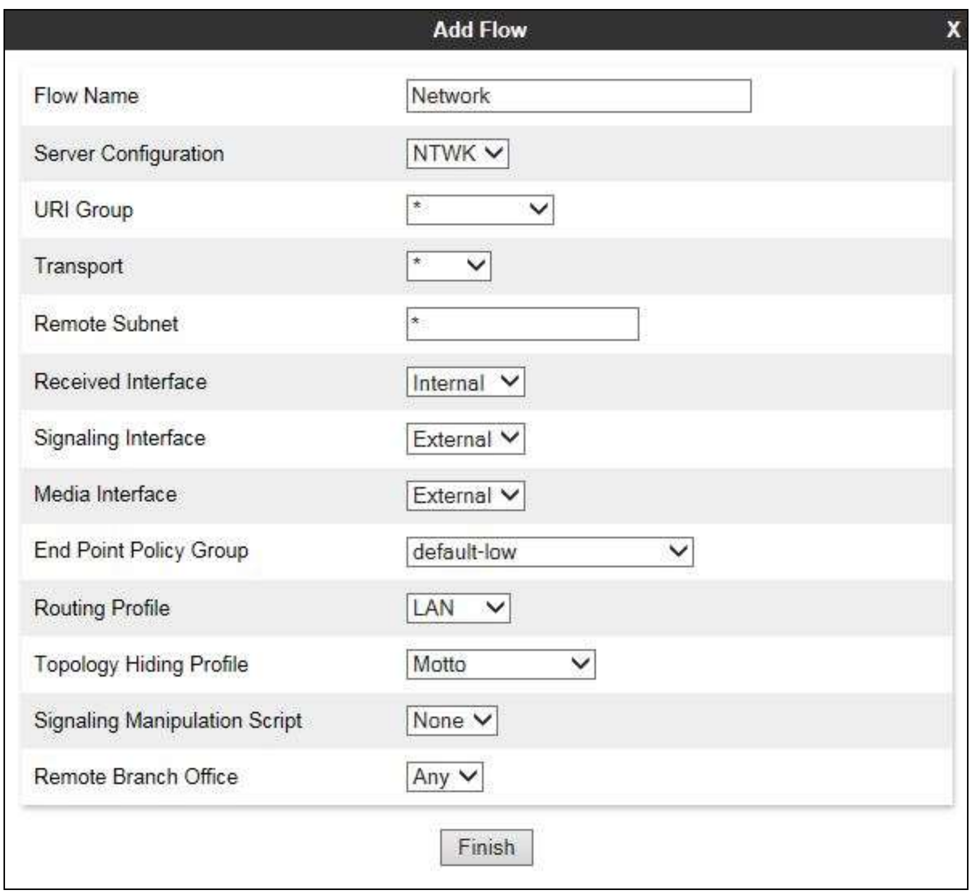


7.8. Server Flows

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

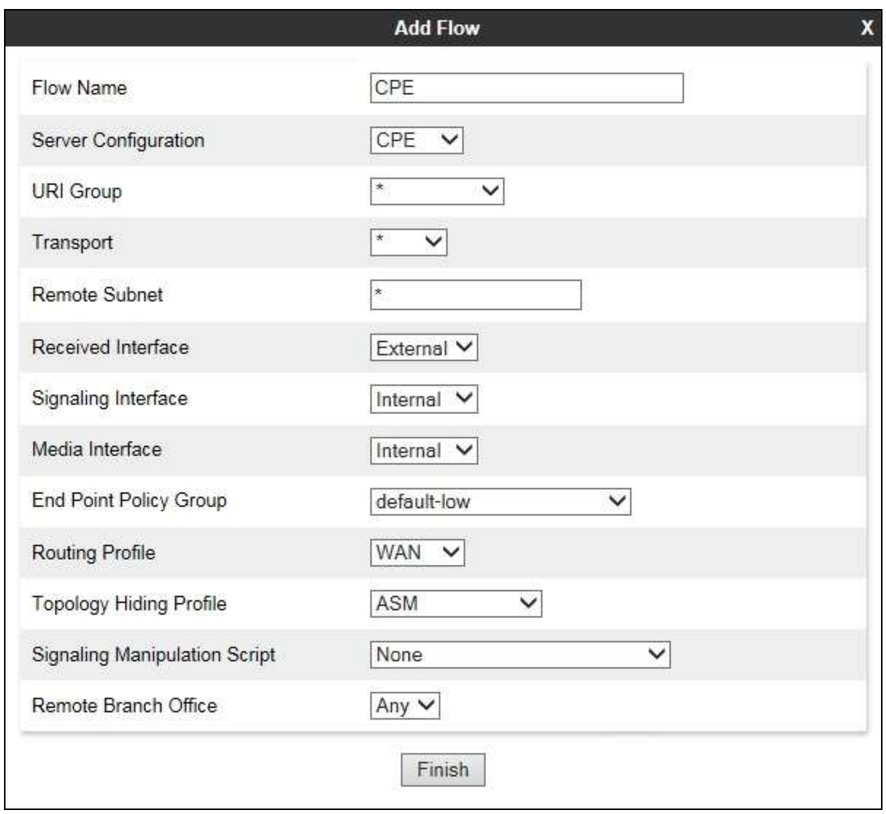
To define a Server Flow for the Motto VoIP SIP Trunk, navigate to **Device Specific Settings → End Point Flows**.

- 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 the Motto SIP Trunk, in the test environment **Network** was used.
- In the **Server Configuration** drop-down menu, select the server configuration for the Motto VoIP SIP Trunk 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 the SIP Trunk is received on.
- In the **Signaling Interface** drop-down menu, select the external SIP signalling interface defined in **Section 7.3**. This is the interface that signalling bound for the SIP Trunk is sent on.
- In the **Media Interface** drop-down menu, select the external media interface defined in **Section 7.3**. This is the interface that media bound for the SIP Trunk is sent on.
- In the **Routing Profile** drop-down menu, select the routing profile of Session Manager defined in **Section 7.6**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of the Motto VoIP SIP Trunk defined in **Section 7.7** and click **Finish**.

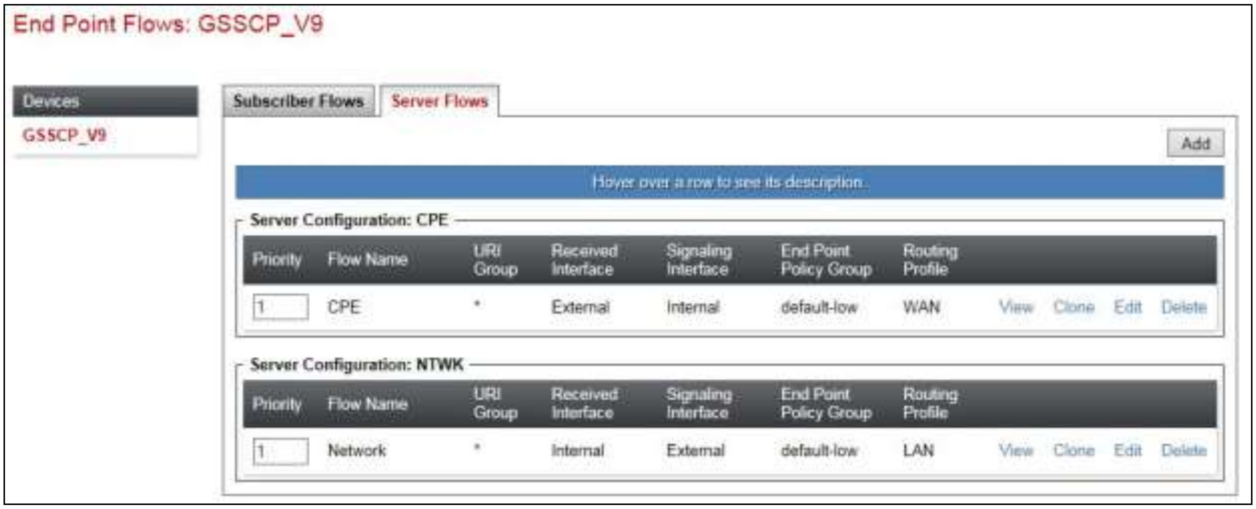


To define a Server Flow for Session Manager, navigate to **Device Specific Settings → End Point Flows**.

- Click on the **Server Flows** tab.
- Select **Add Flow** and enter details in the pop-up menu.
- In the **Flow Name** field enter a descriptive name for the server flow for Session Manager, in the test environment **CPE** was used.
- In the **Server Configuration** drop-down menu, select the server configuration for 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 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 Motto VoIP SIP Trunk defined in **Section 7.6**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of Session Manager defined in **Section 7.7** and click **Finish**.



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



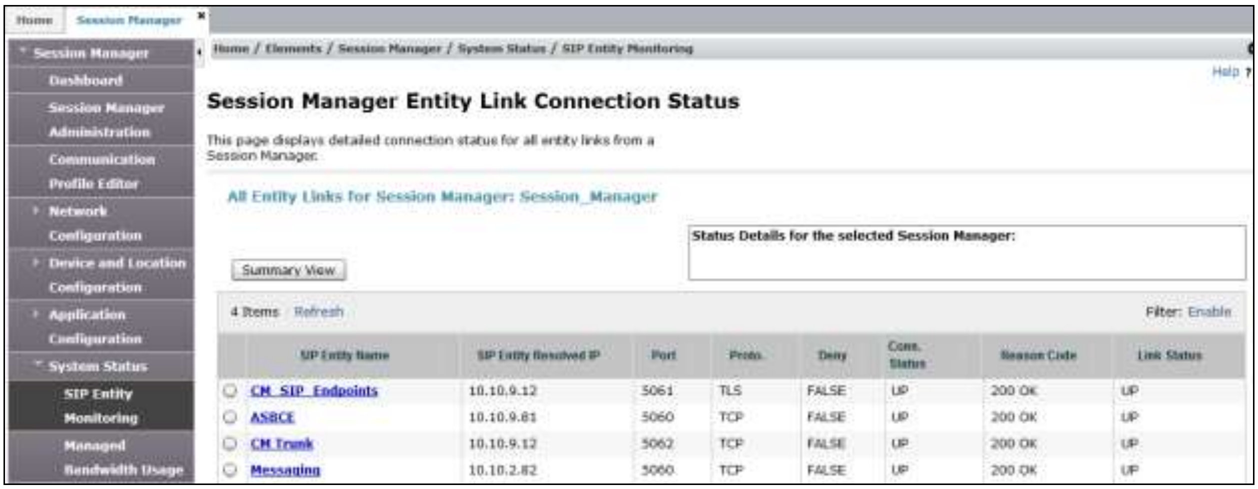
8. Configure the Motto SIP Trunk Equipment

The configuration of the Motto 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 Motto VoIP equipment and system configuration please contact an authorized Motto 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 the 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 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 Motto VoIP 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 Motto VoIP SIP Trunk. The Motto VoIP SIP Trunk 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. The Motto SIP Trunk is described by the *Motto SIP Specification – MBN* document provided by Motto.

Additional Avaya product documentation is available at http://support.avaya.com.

[1] *Migrating and Installing Avaya Appliance Virtualization Platform*, Release 7.0.1, 06 May 2016.

[2] *Upgrading and Migrating Avaya Aura® applications to 7.0*, Release 7.0.1, 06 May 2016.

[3] *Deploying Avaya Aura® 7.0.1 applications*, Release 7.0.1, 09 May 2016

[4] *Deploying Avaya Aura® Communication Manager*, Release 7.0.1, 08 May 2016

[5] *Administering Avaya Aura® Communication Manager* Release 7.0.1, 09 May 2016.

[6] *Upgrading Avaya Aura® Communication Manager*, Release 7.0.1, 08 May 2016

[7] *Deploying Avaya Aura® System Manager*, Release 7.0.1, 09 May 2016

[8] *Upgrading Avaya Aura® System Manager to 7.0.1*, 06 May 2016.

[9] *Administering Avaya Aura® System Manager for 7.0.1*, 25 Jun 2016

[10] *Deploying Avaya Aura® Session Manager*, Release 7.0.1, 09 May 2016

[11] *Upgrading Avaya Aura® Session Manager* Release 7.0.1, 09 May 2016

[12] *Administering Avaya Aura® Session Manager* Release 7.0.1, 09 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.