Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Communication Manager R7.0.1, Avaya Aura® Session Manager R7.0.1 and Avaya Session Border Controller for Enterprise R7.1 to support Telenet SIP Trunking Service - Issue 1.0

Abstract

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

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.

Telenet 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 Telenet SIP Trunking Service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of the following: Avaya Aura ® Communication Manager R7.0.1; Avaya Aura ® Session Manager R7.0.1; Avaya Session Border Controller for Enterprise R7.1; Endpoints as described in **Section 3**. Note that the shortened names Communication Manager, Session Manager and Avaya SBCE will be used throughout the remainder of the document. Customers using this Avaya SIP-enabled enterprise solution with the Telenet 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 Telenet 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 Telenet SIP Trunking, calls made to SIP and H.323 telephones at the enterprise.
- Outgoing calls from the enterprise site completed via Telenet SIP Trunking to PSTN destinations, calls made from SIP and H.323 telephones.
- Inbound and outbound PSTN calls to/from Avaya one-X® Communicator and Avaya Communicator for Windows soft phones.
- Calls using the G.711A Law codec.
- Fax calls to/from a group 3 fax machine to a PSTN connected fax machine using G.711 pass-through.
- In-band DTMF transmission 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 Telenet SIP Trunking requiring Avaya response and sent by Avaya requiring Telenet response.

2.2. Test Results

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

- At the time of testing, the Telenet network was configured to use national number format for the DDI range allocated for the testing. Calling party numbers were sent and received in national format with no leading zero. A Session Manager Adaptation (See **Section 6.4**) was used to convert incoming Calling Party Numbers to diallable format. The number format preferred by Avaya is E.164 with the leading plus used in SIP to indicate an international number.
- On inbound calls, a duplicate INVITE message is sent from the network after half a second even though the Avaya SBCE sends a 100 Trying immediately. The duplicate INVITE does not cause a problem for normal call setup. When a 200 OK is sent immediately however, the duplicate INVITE is rejected by Communication Manager. This is because as far as Communication Manager is concerned, the dialogue is established and it is in an incorrect state for receiving a duplicate INVITE. During testing, there were two cases where the 200 OK was sent immediately, the first was when calling the voicemail system to retrieve messages and the second was when calling the Communication Manager FNE (idle appearance) from an EC500 mobile phone to make an outbound call. Workarounds were found and tested for both these test cases and Telenet are investigating the cause of the duplicate INVITE. The workarounds used are shown in **Appendix A**.
- Duplicate 183 Session Progress messages were observed on outbound calls. These did not cause call failures but are an unnecessary addition to the signalling traffic
- When the CLI is restricted on an inbound call, the network inserts a Proxy-Require header into the SIP INVITE message. During testing this was rejected by the Session Manager and a signalling rule was used to remove it on the Avaya SBCE (See **Section 7.8**).
- When outbound calls to international destinations were put on hold, the media path was not re-established when the call was taken off hold. When the call was taken off hold, a media attribute of “sendonly” was received in the SDP of the 200 OK from the network. The problem affected all calls using hold which include call transfer and conferencing. This was believed to be an issue with the interconnect used by Telenet for international traffic and testing was carried out using national numbers.
- When making calls with the Avaya soft clients, a number of issues were experienced with initial IP-IP Direct Media turned on. These issues were as follows: Two ringbacks heard on consultative transfer and conference calls; No ringback heard on consultative transfer in “Other Phone” mode; Call failure of consultative transfer to PSTN in “Other Phone” mode. This was resolved by turning initial IP-IP Direct Media off, as shown in **Section 5.5**, which causes Communication Manager to set up calls via the Media gateway initially then “shuffle” to direct media once the call is established. A number of tests were re-run with this setting.

- Consultative transfer to internal extension by Avaya one-X® Communicator in Computer Mode failed when “Re-INVITE Handling” was used on the Avaya SBCE. This was only an issue because “Re-INVITE Handling” was a potential workaround to an issue mentioned previously where duplicate INVITE messages are received from the network after the call is answered. An alternative workaround was found.
- When making calls to CM with no available capacity on the SIP trunk, 500 Service Unavailable (Signaling Resources Unavailable) is sent from Communication Manager to the network. The network re-attempts call set-up repeatedly. Silence is heard for 30s then a tone.
- When making calls to Communication Manager when there is a signalling failure, 408 Request Timeout then 500 Server Link Monitor Status Down is sent from Session Manager to the network. The network re-attempts the call set-up repeatedly. Silence is heard for 38s then tone.

Items not tested include the following:

- DTMF transmission using telephone events as described in RFC2833 was not tested as Telenet prefer to send DTMF in-band.
- T.38 fax transmission was not tested as G.711 is the codec preferred by Telenet.
- Codecs other than G.711 were not tested for voice calls as G.711 is the codec preferred by Telenet.
- No Inbound Toll-Free access was 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 Telenet products please contact the following website: https://www2.telenet.be/en/

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an Enterprise site connected to the Telenet SIP Trunking service. Located at the Enterprise site is an Avaya SBCE, Session Manager and Communication Manager. Endpoints are Avaya 96x0 series IP telephones (with SIP and 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.

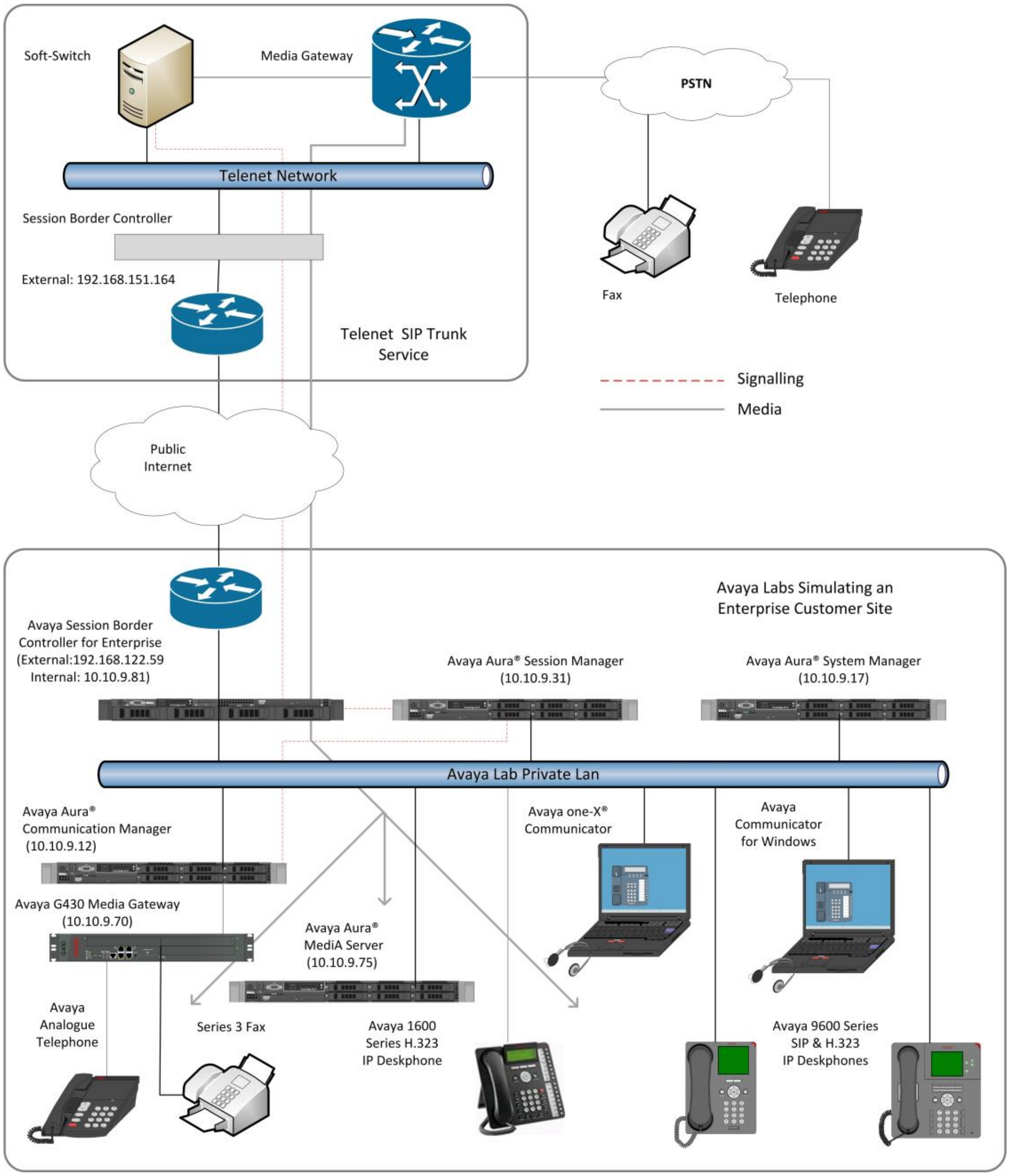

Soft-Switch
Media Gateway
PSTN
Telenet Network
Session Border Controller
External: 192.168.151.164
Fax
Telephone
Telenet SIP Trunk Service
Signalling
Media
Public Internet
Avaya Labs Simulating an Enterprise Customer Site
Avaya Session Border Controller for Enterprise (External:192.168.122.59 Internal: 10.10.9.81)
Avaya Aura® Session Manager (10.10.9.31)
Avaya Aura® System Manager (10.10.9.17)
Avaya Lab Private Lan
Avaya Aura® Communication Manager (10.10.9.12)
Avaya one-X® Communicator
Avaya Communicator for Windows
Avaya G430 Media Gateway (10.10.9.70)
Avaya Aura® MediA Server (10.10.9.75)
Avaya Analogue Telephone
Series 3 Fax
Avaya 1600 Series H.323 IP Deskphone
Avaya 9600 Series SIP & H.323 IP Deskphones

Figure 1: Test Setup Telenet SIP Trunking Service to Avaya Enterprise

4. Equipment and Software Validated

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

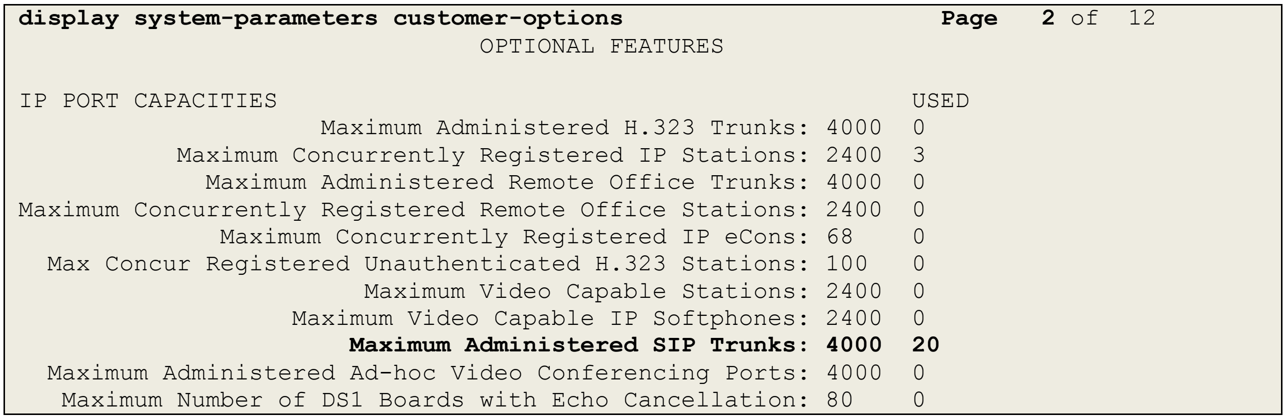
Equipment/Software	**Release/Version**
Avaya	
Avaya Aura® Session Manager	7.0.1.0.701007
Avaya Aura® System Manager	7.0.1.0.65071 – SP1
Avaya Aura® Communication Manager	7.0.1.0.0-23012 – FP1
Avaya Session Border Controller for Enterprise	7.1.0.0-04-11122
Media Server	7.7.0.334
Avaya G430 Media Gateway	37.38.0
Avaya 9600 series Handsets SIP 96x0 SIP 9608 H.323 96x0 H.323 9608 H.323 1616	 2.6.16 7.0.1.1-062716 3.2.6A 6.6.2.29 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
Telenet	
Sonus GSX9000 SBC	v09.00.08 R000
Genband CS2K	CVM17

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 Telenet 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. The Session Manager directs the outbound SIP messages to the Avaya SBCE at the enterprise site that then sends the SIP messages to the Telenet 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 Telenet 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

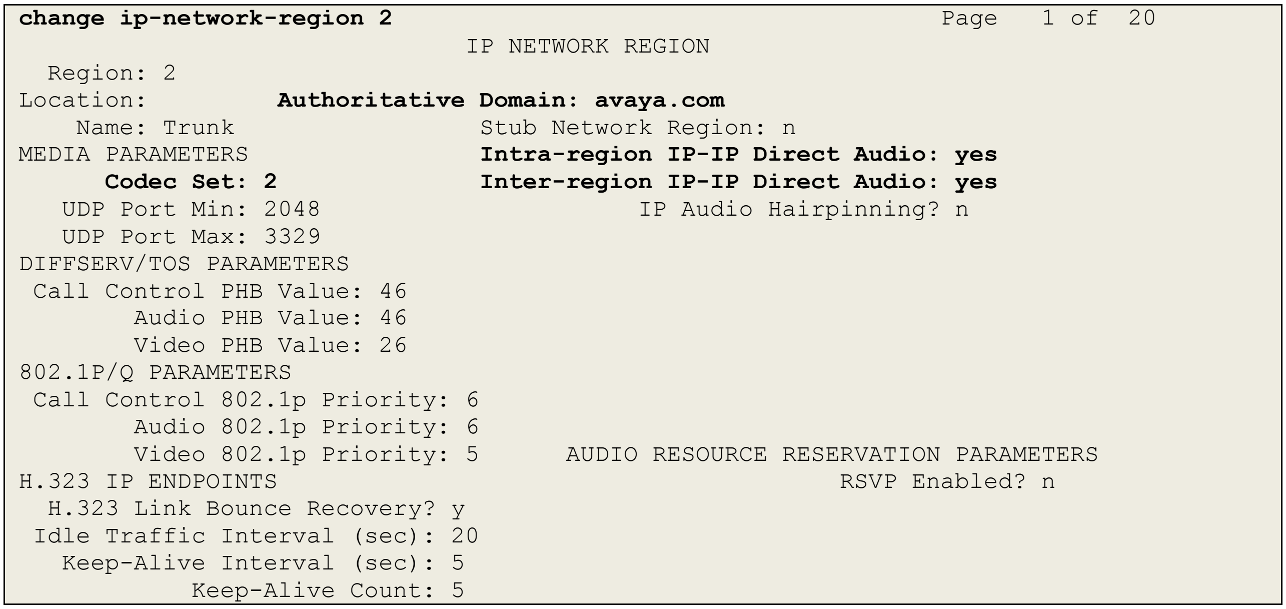
The node names defined here will be used in other configuration screens to define a SIP signalling group between Communication Manager and Session Manager. In the **IP Node Names** form, assign the node **Name** and **IP Address** for Session Manager using the **change node-names ip** command. In this 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.

```
change 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 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.

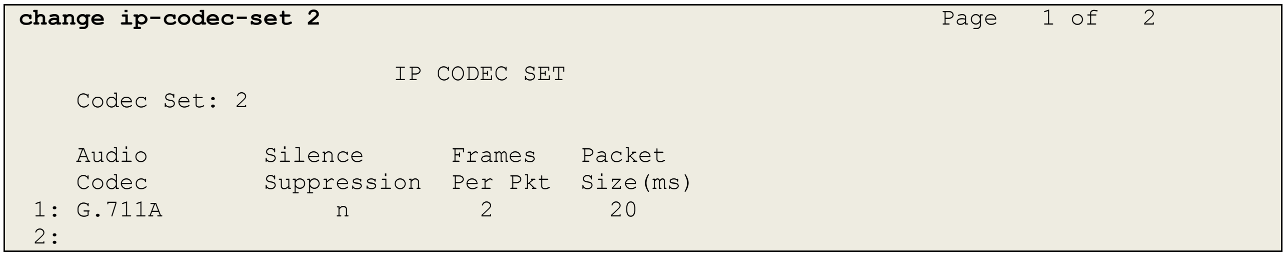


```
change ip-network-region 2                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 2
Location:         Authoritative Domain: avaya.com
    Name: Trunk                 Stub Network Region: n
MEDIA PARAMETERS                Intra-region IP-IP 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.

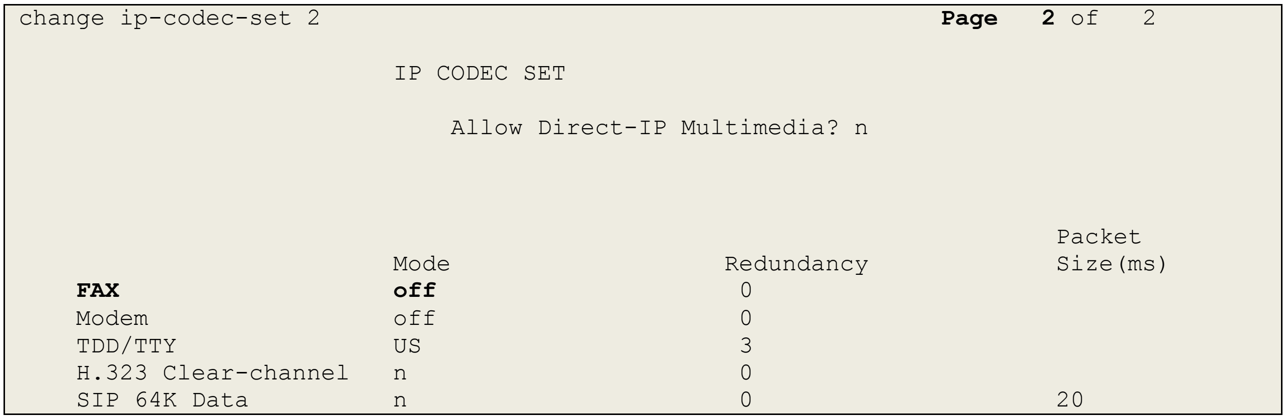
5.4. Administer IP Codec Set

Use the **change ip-codec set n** command where **n** is the codec set specified in the IP Network Region form in **Section 5.3**. Enter the list of audio codec's eligible to be used in order of preference. For the interoperability test the codec preferred by Telenet was configured, namely **G.711A**.



```
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:
```

Telenet prefers G.711 for transmission of fax. Navigate to **Page 2** and define G.711 fax by setting the **FAX Mode** to **off**. This prevents Communication Manager from taking any action when fax is detected



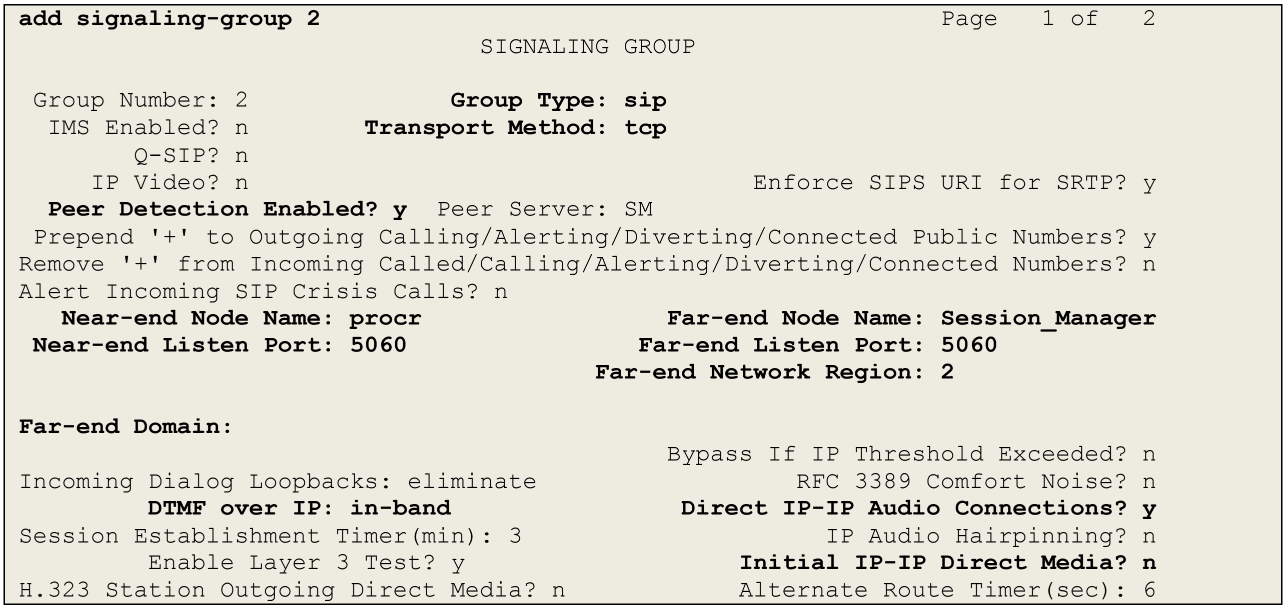
```
change ip-codec-set 2                                             Page   2 of   2

                          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
```

5.5. Administer SIP Signaling Groups

This signalling group (and trunk group) will be used for inbound and outbound PSTN calls to the Telenet SIP Trunking service. During test, this was configured to use TCP and port 5060. Configure the **Signaling Group** using the **add signaling-group n** command as follows:

- Set **Group Type** to **sip**.
- Set **Transport Method** to required protocol. Although TLS is recommended for security, **tcp** was used during testing.
- 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, during testing, **5060** was used. These must correspond to those used on the Session Manager Entity Links (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 **DTMF over IP** to **in-band** which is Telenet's preference for transmission of DTMF.
- Set **Direct IP-IP Audio Connections** to **y** to avoid unnecessary use of resources
- Leave **Initial IP-IP Direct Media** to default **n** to facilitate the use of Early Media.

The default values for the other fields may be used.

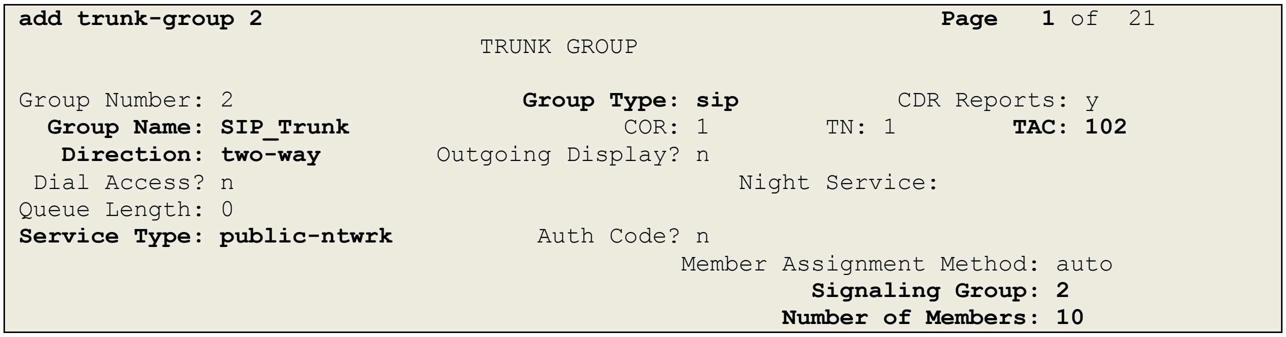


```
add signaling-group 2                                           Page   1 of   2
                                SIGNALING GROUP

 Group Number: 2              Group Type: sip
  IMS Enabled? n        Transport Method: 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: 5060                Far-end Listen Port: 5060
                                        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: in-band                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

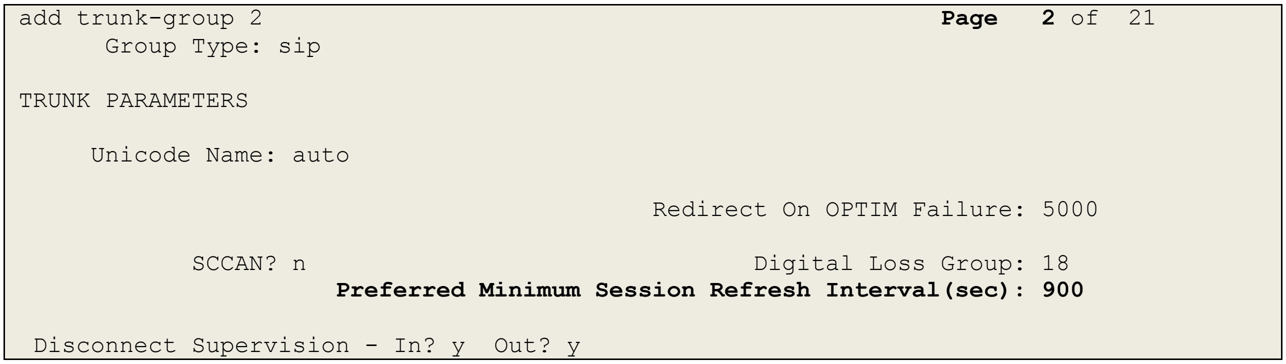
A trunk group is associated with the signaling group described in **Section 5.5**. Configure the trunk group using the **add trunk-group n** command, where **n** is an available trunk group for the SIP Trunk. On **Page 1** of this form:

- 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.



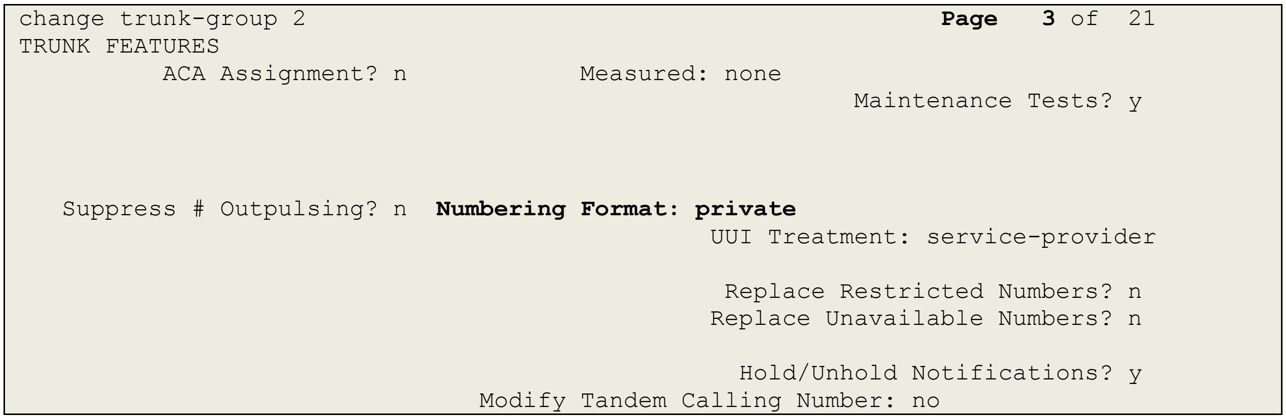
```
add trunk-group 2                                               Page   1 of  21
                                TRUNK GROUP

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



```
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
```

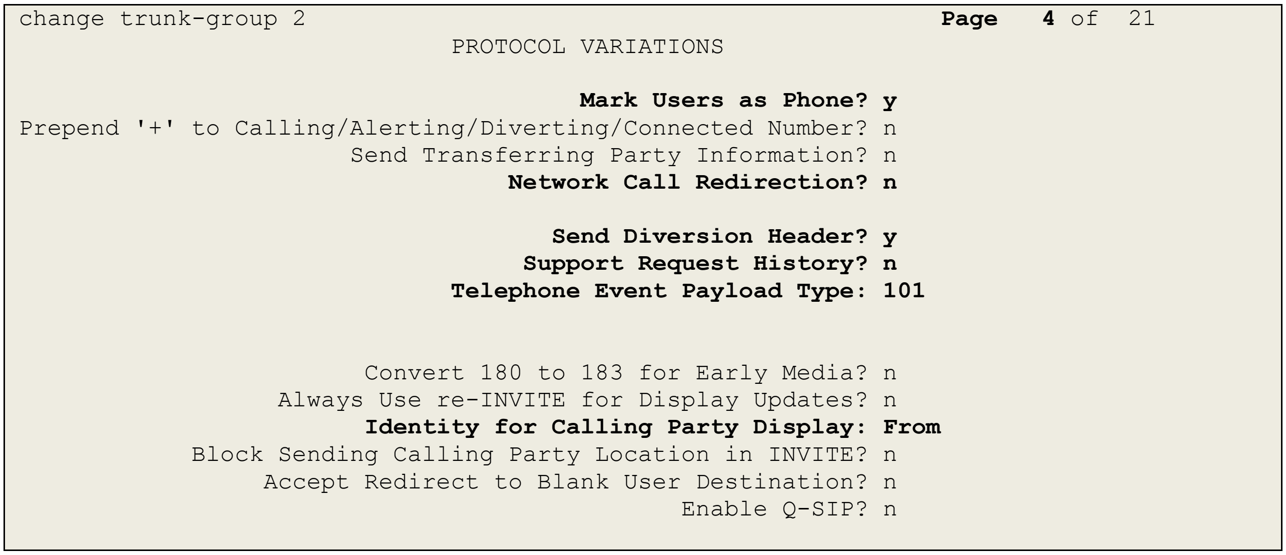
On **Page 3**, set the **Numbering Format** field to **private** if national numbering is to be used as was the case during testing. If E.164 with preceding “+” is to be used, select public.



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



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

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n

                                                  Hold/Unhold Notifications? y
                                Modify Tandem Calling Number: no
```

On **Page 4** of this form:

- Set **Mark Users as Phone** to **y** as required by Telenet
- Set **Network Call Redirection** to **n** as redirection using “302 Moved Temporarily” or REFER are not supported.
- Set **Send Diversion Header** to **y** so that the DDI number assigned to the extension is passed for forwarded calls.
- Set **Support Request History** to **n**.
- Set the **Telephone Event Payload Type** to **101** to match the value preferred by Telenet (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.



```
change 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? n
                                  Network Call Redirection? n

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


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

5.7. Administer Calling Party Number Information

Use the **change private-numbering** command to configure Communication Manager to send the calling party number in national format with no leading zero. 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: 9
 4  2000              2           329nnnn0          8        Maximum Entries: 540
 4  2001              2           329nnnn8          8
 4  2291              2           329nnnn2          8
 4  2316              2           329nnnn3          8
 4  2391              2           329nnnn1          8
 4  2400              2           329nnnn4          8
 4  2401              2           329nnnn7          8
 4  7000              2           329nnnn5          8
```

Use the **change public-unknown-numbering** command if it has been agreed with Telenet to use E.164 numbering. Communication Manager automatically prefixes a "+" to the numbers when this table is used. This table is also used for the Contact header in SIP responses for incoming calls even where private has been selected in the Numbering format in the Trunk settings.

```
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: 9
 4  2                1                      4        Maximum Entries: 240
 4  2000             2         32329nnnn0   10
 4  2001             2         32329nnnn8   10    Note: If an entry applies to
 4  2291             2         32329nnnn2   10    a SIP connection to Avaya
 4  2316             2         32329nnnn3   10    Aura(R) Session Manager,
 4  2391             2         32329nnnn1   10    the resulting number must
 4  2400             2         32329nnnn4   10    be a complete E.164 number.
 4  2401             2         32329nnnn7   10
 4  7000             2         32329nnnn5   10    Communication Manager
                                                  automatically inserts
                                                  a '+' digit in this case.
```

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

5.8. Administer Route Selection for Outbound Calls

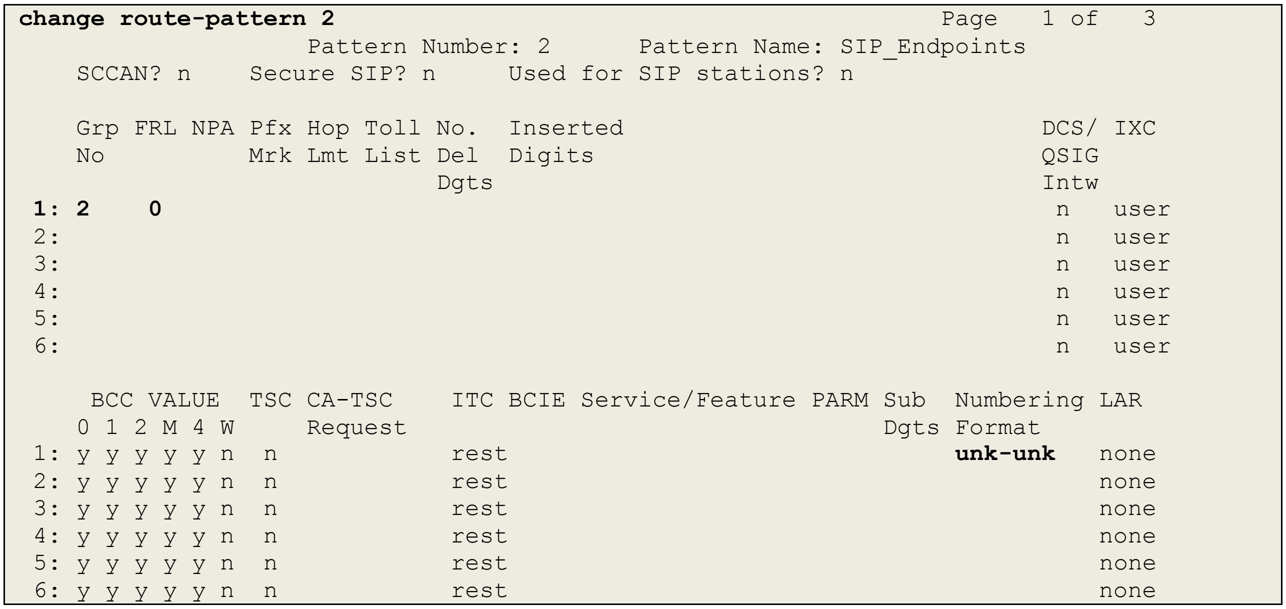
In the test environment, the Automatic Route Selection (ARS) feature was used to route outbound calls via the SIP trunk to the Telenet network. The single digit **9** was used as the ARS access code providing a facility for telephone users to dial 9 to reach an outside line. Use the **change feature-access-codes** 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 with leading **0**. Note that exact maximum number lengths should be used where possible to reduce post-dial delay, the example shows international numbers with country code **353** for Ireland and area code **91** for Galway. 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
```

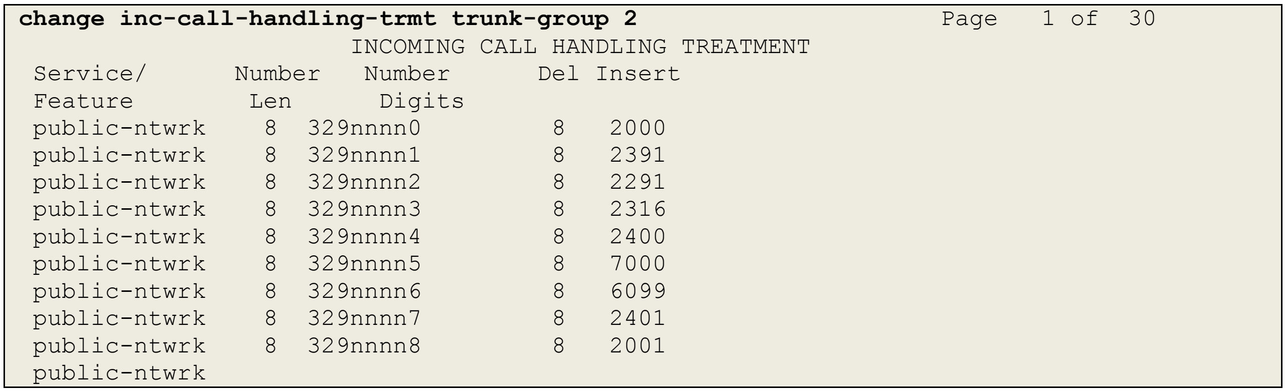
Use the **change route-pattern n** command, where **n** is an available route pattern, to add the SIP trunk group to the route pattern that ARS selects. In this configuration, route pattern **2** is used to route calls to trunk group **2**. **Numbering Format** is applied to CLI and is used to set TDM 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 Telenet 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, 8 digits are received with no preceding zero. All digits are deleted and the extension number is inserted. Note that some of the DDI digits have been obscured.

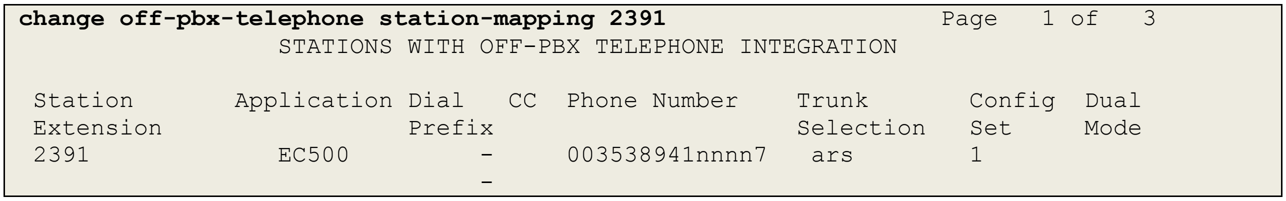


```
change inc-call-handling-trmt trunk-group 2                     Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 public-ntwrk    8  329nnnn0         8   2000
 public-ntwrk    8  329nnnn1         8   2391
 public-ntwrk    8  329nnnn2         8   2291
 public-ntwrk    8  329nnnn3         8   2316
 public-ntwrk    8  329nnnn4         8   2400
 public-ntwrk    8  329nnnn5         8   7000
 public-ntwrk    8  329nnnn6         8   6099
 public-ntwrk    8  329nnnn7         8   2401
 public-ntwrk    8  329nnnn8         8   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 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, 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 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         -     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.

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

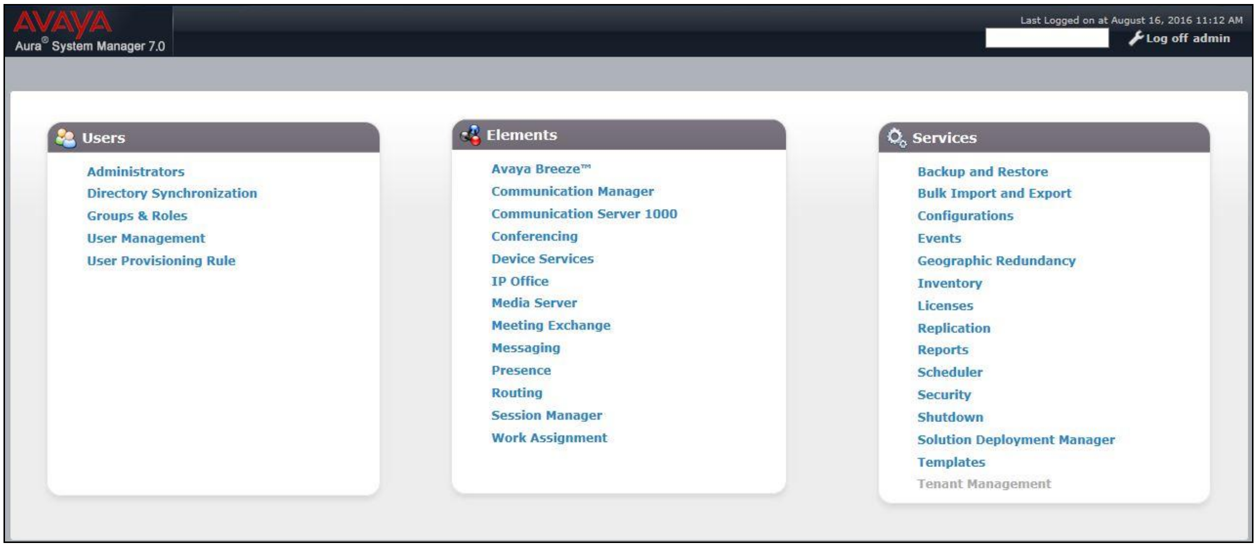
6. Configuring Avaya Aura® Session Manager

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

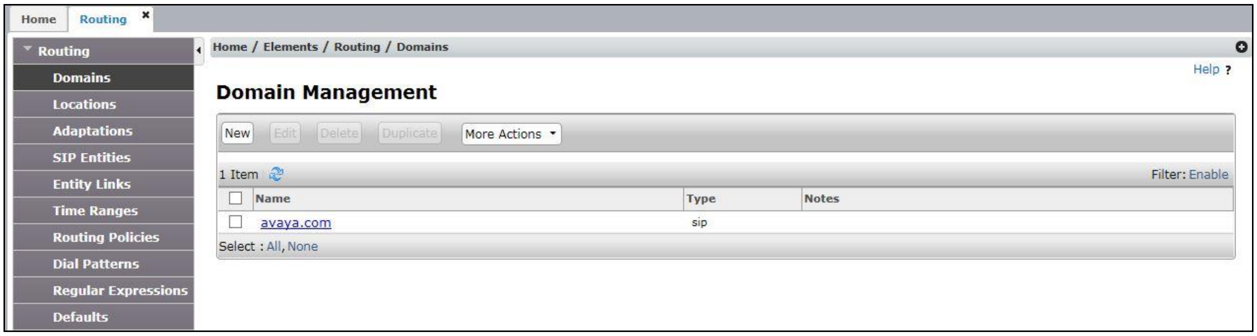
6.1. Log in to Avaya Aura® System Manager

Access the System Manager using a web browser and 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** screen 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 **Elements**, **Home** screen 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 Telenet; 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.



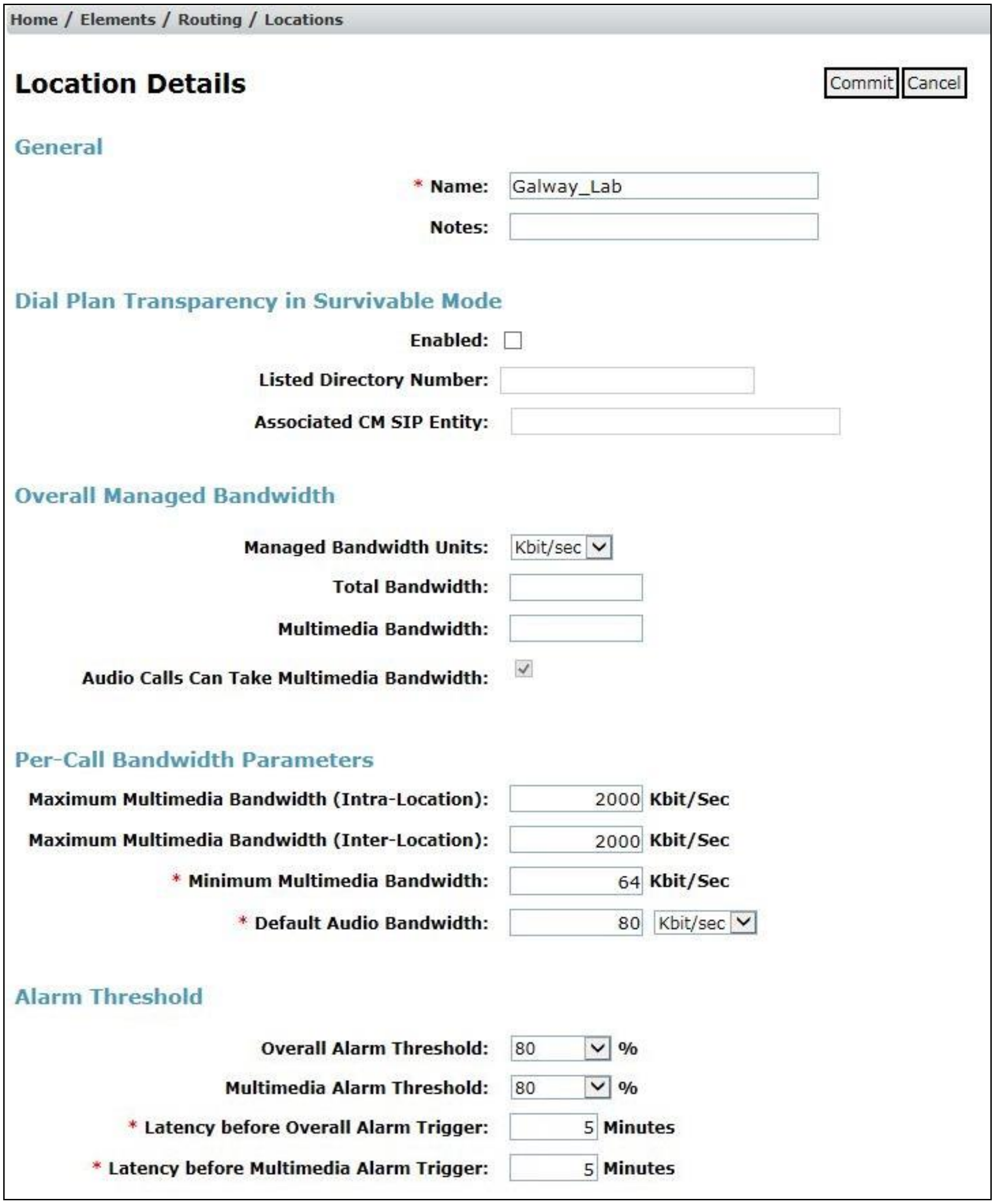
Domain Management

☐	Name	Type	Notes
☐	avaya.com	sip	

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 and Session manager routing. One location is added to the sample configuration for all of the enterprise SIP entities and another for the Telenet SIP Trunk. 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. Define bandwidth requirements, during testing these were left at default values.



Location Details

General

* Name: Galway_Lab

Notes:

Dial Plan Transparency in Survivable Mode

Enabled: □

Listed Directory Number:

Associated CM SIP Entity:

Overall Managed Bandwidth

Managed Bandwidth Units: Kbit/sec

Total Bandwidth:

Multimedia Bandwidth:

Audio Calls Can Take Multimedia Bandwidth: ☑

Per-Call Bandwidth Parameters

Maximum Multimedia Bandwidth (Intra-Location): 2000 Kbit/Sec

Maximum Multimedia Bandwidth (Inter-Location): 2000 Kbit/Sec

* Minimum Multimedia Bandwidth: 64 Kbit/Sec

* Default Audio Bandwidth: 80 Kbit/sec

Alarm Threshold

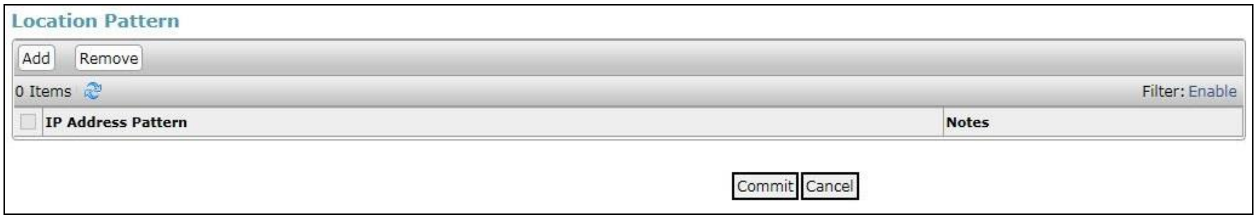
Overall Alarm Threshold: 80 %

Multimedia Alarm Threshold: 80 %

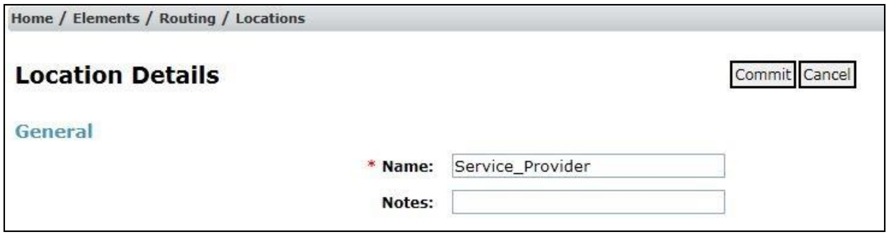
* Latency before Overall Alarm Trigger: 5 Minutes

* Latency before Multimedia Alarm Trigger: 5 Minutes

The location pattern is a way of using subnets to further refine the location information, this may be useful for endpoints that could be logged in from different subnets. This was not used during testing. If required, 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.



Although routing based on location was not used on Session manager during testing, a separate location was defined for the Telenet SIP Trunk. The bandwidth parameters were left at default values and are not shown here.



Location Details

General

* Name: Service_Provider

Notes:

6.4. Administer Adaptations

Session Manager Adaptations can be used to alter parameters in the SIP message headers, during compliance testing, two were used. One Adaptation was used on the Communication Manager SIP Entity to convert the Calling Party Numbers sent from Session Manager to diallable formats for display on Communication Manager extensions. The other was used on the Avaya SBCE SIP Entity to remove Avaya proprietary headers from messages sent from Session Manager.

6.4.1. Communication Manager

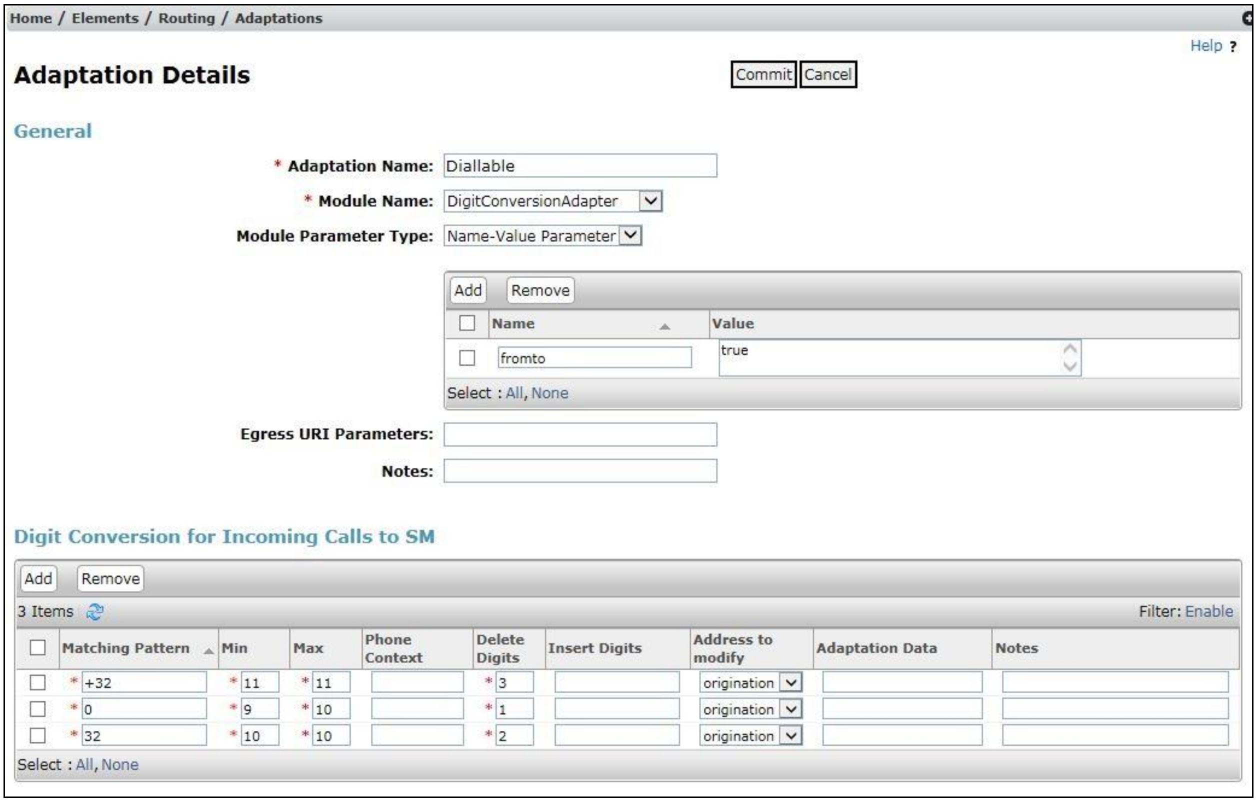
Calling Party Numbers were received from the network with no leading zeros, so the Adaption was used to analyse the numbers and prefix national numbers with a single zero and international numbers with two zeros. In addition, there were messages where the Calling Party Number of the Communication Manager extension was sent in a format other than that preferred by Telenet, these were as follows:

- E.164 with leading "+"
- E.164 with no leading "+"
- National with leading "0"

The Adaptation was used to change the Calling Party Number format of both incoming and outgoing calls in both SIP Request and Response messages. To achieve this, digits were converted in both incoming calls to Session Manager and Outgoing calls from Session 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. During testing **Diallable** was used.
- 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 **fromto**.
- In the **Value** field, type **true**.
- 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 Communication Manager.



Adaptation Details

General

* Adaptation Name: Diallable

* Module Name: DigitConversionAdapter

Module Parameter Type: Name-Value Parameter

☐	Name	Value
☐	fromto	true

Egress URI Parameters:

Notes:

Digit Conversion for Incoming Calls to SM

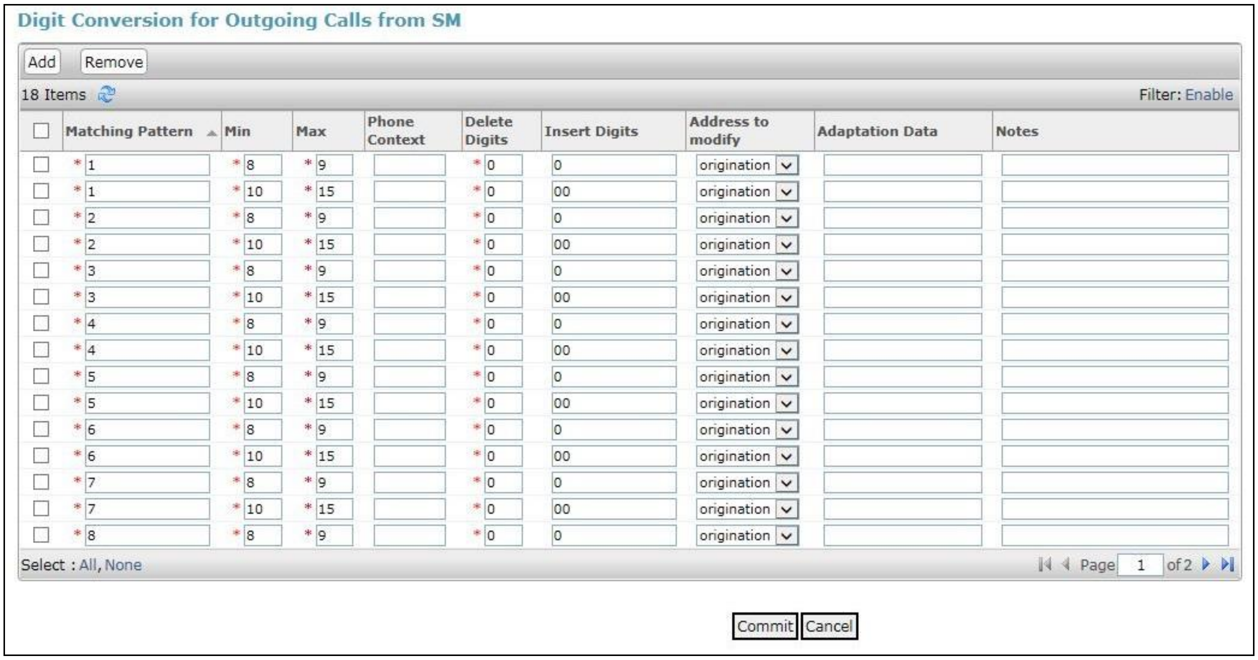
3 Items | Filter: Enable

☐	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
☐	* +32	* 11	* 11		* 3		origination		
☐	* 0	* 9	* 10		* 1		origination		
☐	* 32	* 10	* 10		* 2		origination		

The screenshot shows how the calling party numbers in messages coming from Communication Manager were analysed for testing. The three formats described on the previous page are converted to national format with no leading zero.

Scroll down to **Digit Conversion for Outgoing Calls from SM**. Digit conversion is used to convert to diallable format which is a single leading zero for national numbers and two leading zeros for international format. During testing, the only way to distinguish between national and international numbers was by number length.

Click on **Add** and enter the first digit of the Calling Party Number, during test all possible numbers were analysed though not all are shown in the screenshot as they were on the next page of the table. Define number lengths for both national and international numbers and prefix as appropriate.



Digit Conversion for Outgoing Calls from SM

□	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
□	* 1	* 8	* 9		* 0	0	origination		
□	* 1	* 10	* 15		* 0	00	origination		
□	* 2	* 8	* 9		* 0	0	origination		
□	* 2	* 10	* 15		* 0	00	origination		
□	* 3	* 8	* 9		* 0	0	origination		
□	* 3	* 10	* 15		* 0	00	origination		
□	* 4	* 8	* 9		* 0	0	origination		
□	* 4	* 10	* 15		* 0	00	origination		
□	* 5	* 8	* 9		* 0	0	origination		
□	* 5	* 10	* 15		* 0	00	origination		
□	* 6	* 8	* 9		* 0	0	origination		
□	* 6	* 10	* 15		* 0	00	origination		
□	* 7	* 8	* 9		* 0	0	origination		
□	* 7	* 10	* 15		* 0	00	origination		
□	* 8	* 8	* 9		* 0	0	origination		

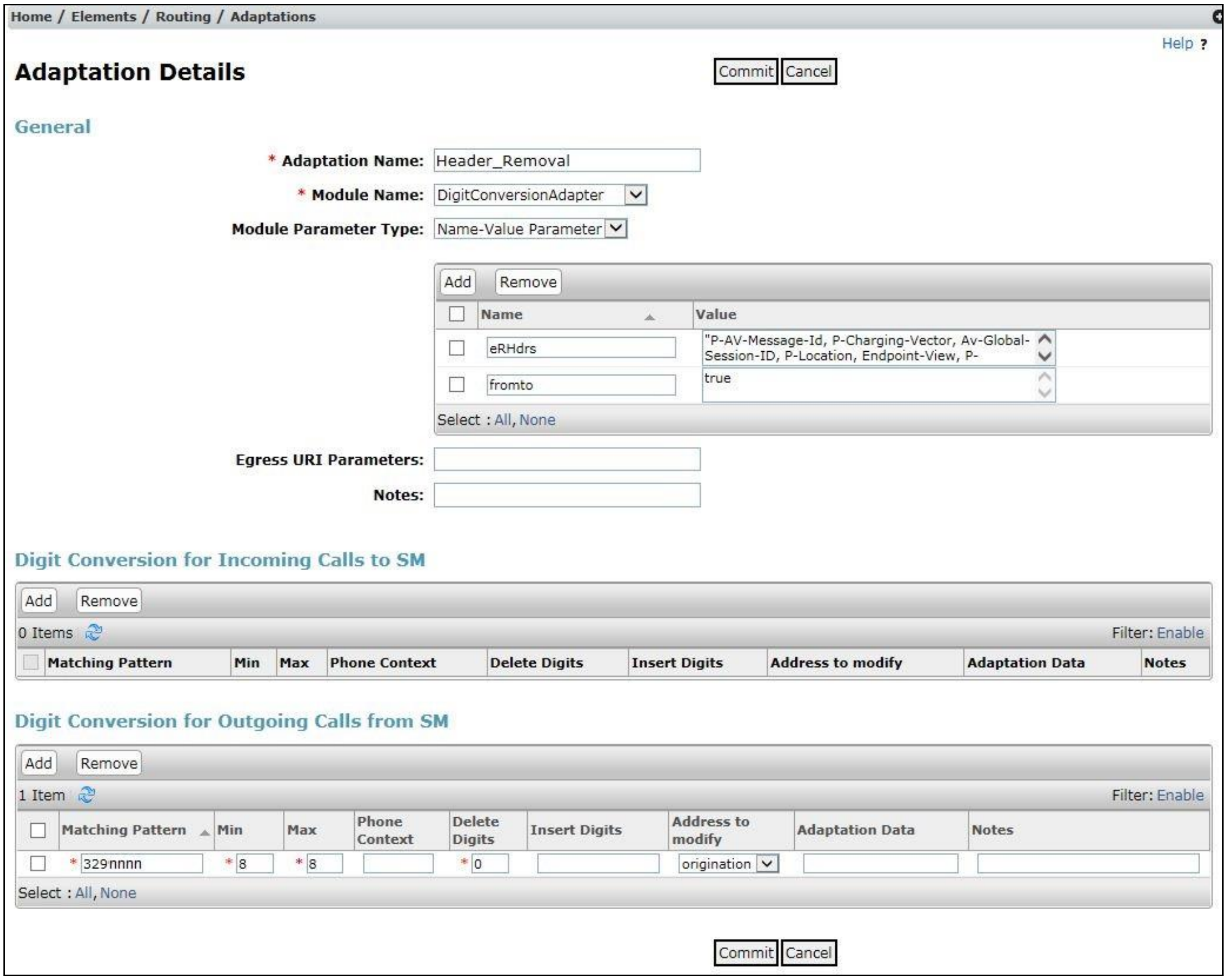
If E.164 numbering is used, a similar Adaptation could be used to analyse Belgian numbers and replace the preceding "+32" with "0". It could also be used to analyse international numbers and replace the preceding "+" with "00".

6.4.2. Avaya SBCE

Communication Manager and Session Manager make use of Avaya proprietary SIP headers to facilitate the full suite of Avaya functionality within the enterprise. These are not required on the SIP trunk however, and make the SIP messages unnecessarily large. A Session Manager Adaptation is used to remove proprietary headers.

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

- In the **Adaptation Name** field, enter a descriptive title for the adaptation.
- In the **Module Name** drop down menu, select **DigitConversionAdapter**. 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, Alert-Info"**.
- In **Digit Conversion for Outgoing Calls from SM**, specify the common digits in the Calling Party Number from Communication Manager in the **Matching Pattern** such that a match is found for all outgoing calls.



Adaptation Details

General

* Adaptation Name:	Header_Removal
* Module Name:	DigitConversionAdapter
Module Parameter Type:	Name-Value Parameter

□	Name	Value
□	eRHdrs	"P-AV-Message-Id, P-Charging-Vector, Av-Global-Session-ID, P-Location, Endpoint-View, P-
□	fromto	true

Select : All, None

Egress URI Parameters:	
Notes:	

Digit Conversion for Incoming Calls to SM

0 Items

Filter: Enable

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

Digit Conversion for Outgoing Calls from SM

1 Item

Filter: Enable

□	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
□	* 329nnnn	* 8	* 8		* 0		origination		

Select : All, None

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.

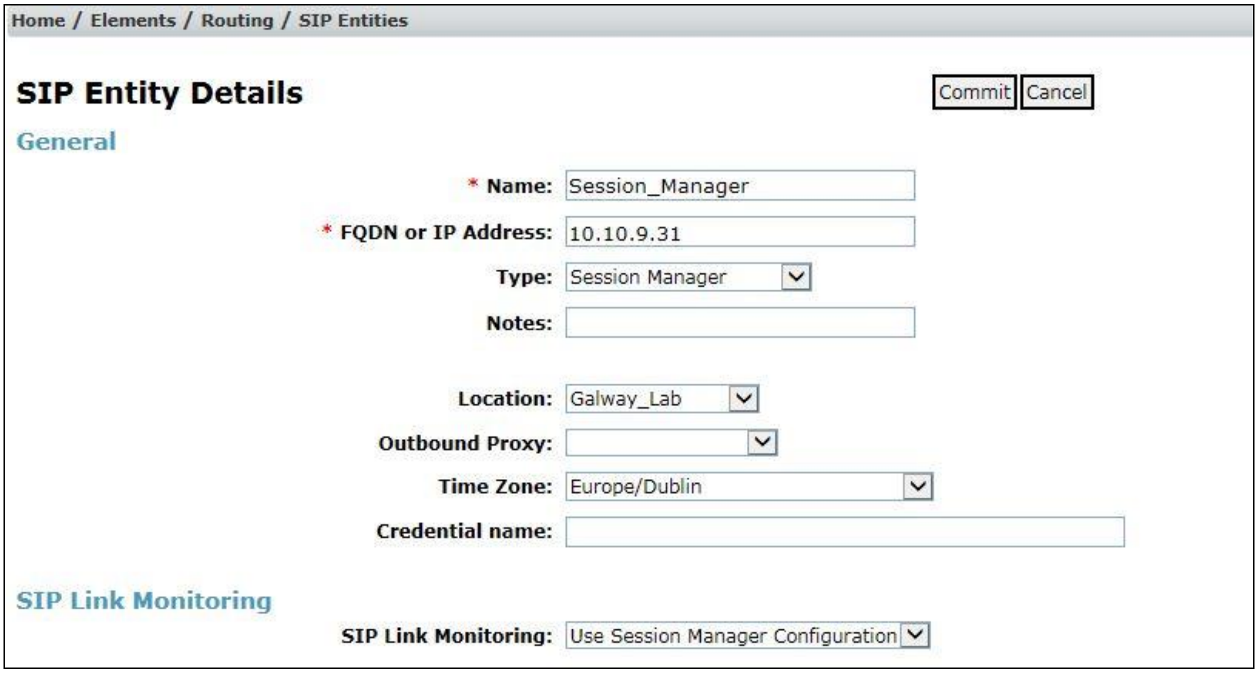
In this configuration there are four SIP Entities:

- 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.

There is also a SIP Entity for Avaya Aura® Messaging but that is not described in this document.

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.



Home / Elements / Routing / SIP Entities

SIP Entity Details

General

* Name: Session_Manager

* FQDN or IP Address: 10.10.9.31

Type: Session Manager

Notes:

Location: Galway_Lab

Outbound Proxy:

Time Zone: Europe/Dublin

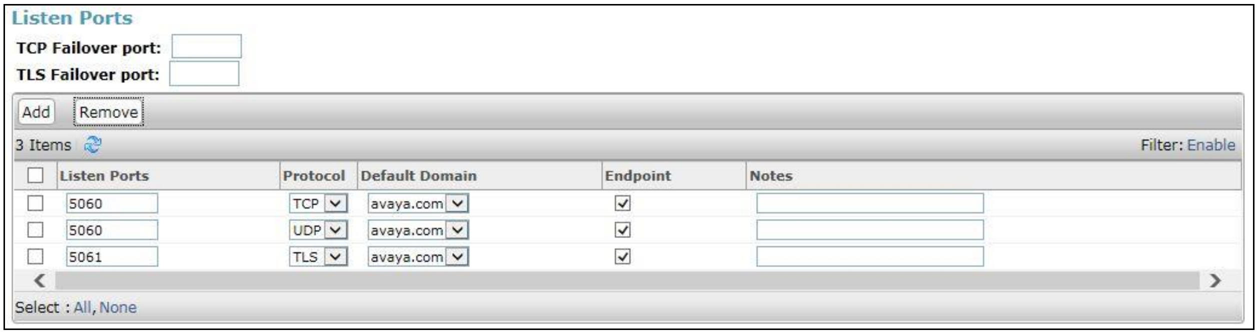
Credential name:

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

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.
- Click on **Commit**.



Listen Ports

TCP Failover port:

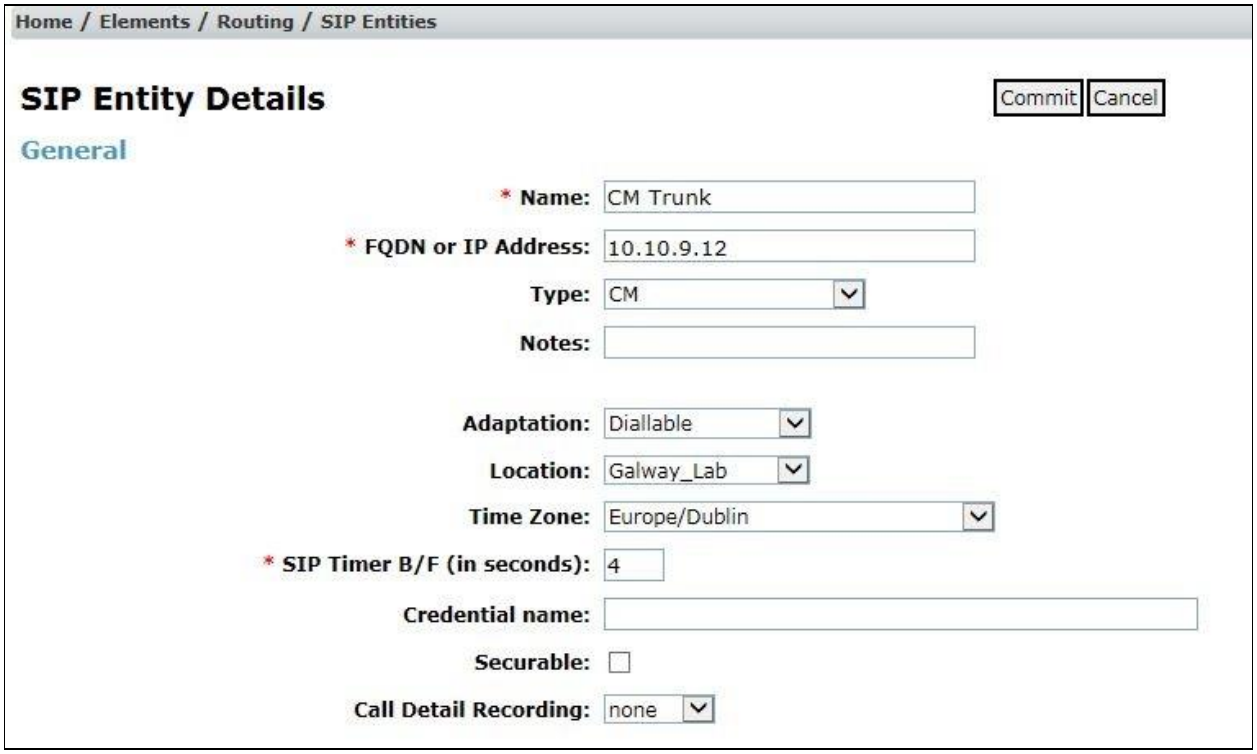
TLS Failover port:

3 Items

□	Listen Ports	Protocol	Default Domain	Endpoint	Notes
□	5060	TCP	avaya.com	☑	
□	5060	UDP	avaya.com	☑	
□	5061	TLS	avaya.com	☑	

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**.



SIP Entity Details

General

* Name: CM Trunk

* FQDN or IP Address: 10.10.9.12

Type: CM

Notes:

Adaptation: Diallable

Location: Galway_Lab

Time Zone: Europe/Dublin

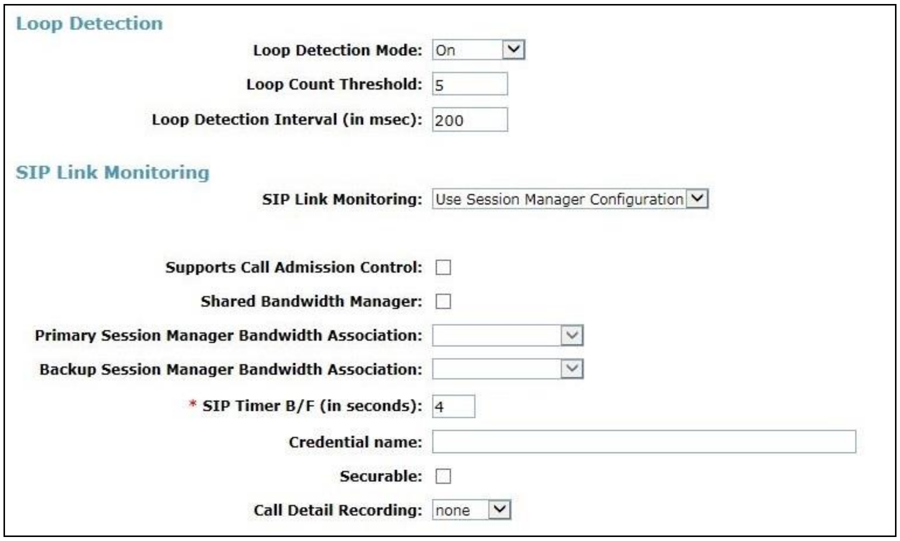
* SIP Timer B/F (in seconds): 4

Credential name:

Securable: □

Call Detail Recording: none

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



Loop Detection

Loop Detection Mode: On

Loop Count Threshold: 5

Loop Detection Interval (in msec): 200

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

Supports Call Admission Control: □

Shared Bandwidth Manager: □

Primary Session Manager Bandwidth Association:

Backup Session Manager Bandwidth Association:

* SIP Timer B/F (in seconds): 4

Credential name:

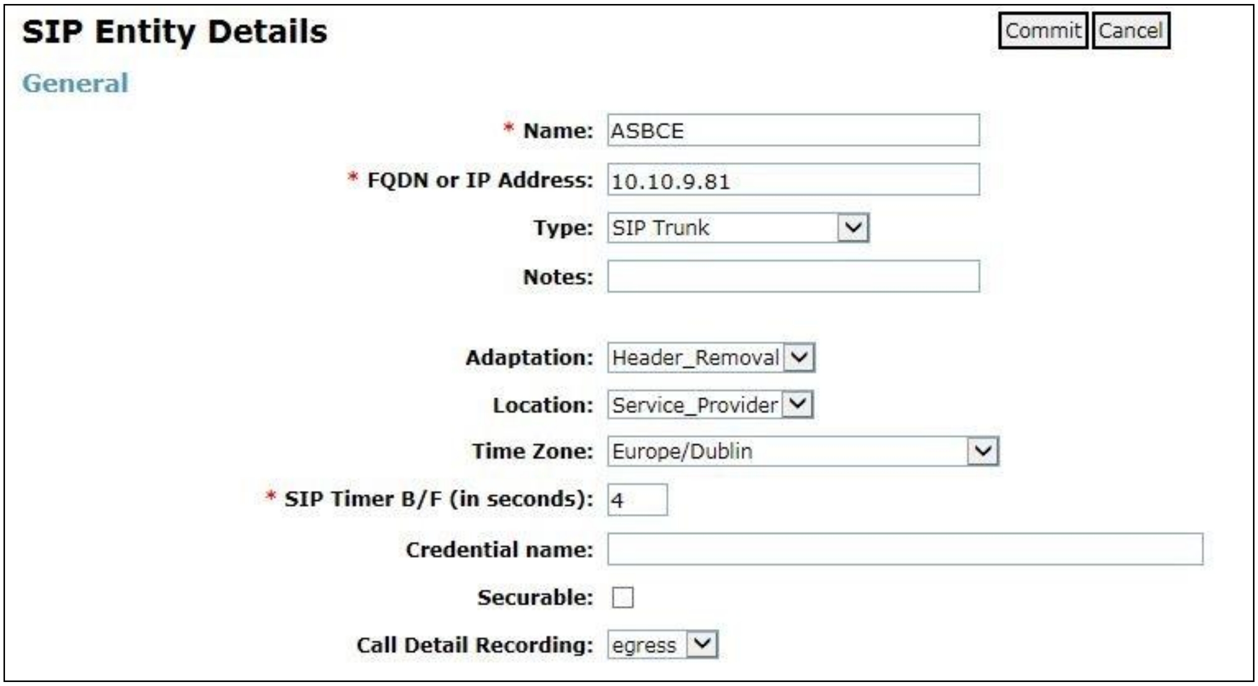
Securable: □

Call Detail Recording: none

Note: The Adaptation assigned is that defined in **Section 6.4.1**. Note also that a second SIP Entity for Communication Manager is required for SIP Endpoints. In the test environment this is named "CM_SIP_Endpoints". The parameters are the same apart from the Adaptation, and the two are assigned to different Entity Links, as described in **Section 6.6**, so that different ports can be used. It is these different ports that distinguish between traffic for SIP Endpoints and traffic for the SIP Trunk.

6.5.3. Avaya Session Border Controller for Enterprise SIP Entity

The screenshot 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 6.4.2**, the **Location** to that defined in **Section 6.3** for the SIP Trunk, and the **Time Zone** to the appropriate time zone.

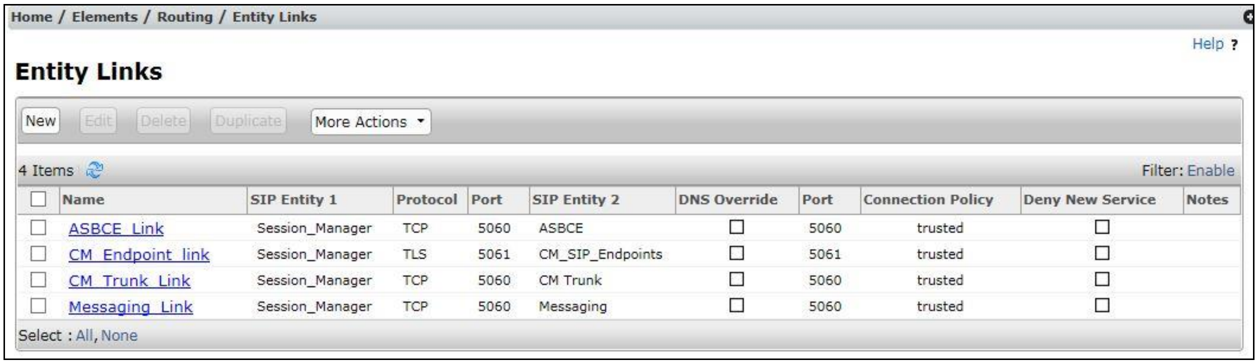


* Name:	ASBCE
* FQDN or IP Address:	10.10.9.81
Type:	SIP Trunk
Notes:	
Adaptation:	Header_Removal
Location:	Service_Provider
Time Zone:	Europe/Dublin
* SIP Timer B/F (in seconds):	4
Credential name:	
Securable:	□
Call Detail Recording:	egress

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. Fill in the following fields in the new row that is displayed (not shown).

- 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** (not shown) to save changes. The screenshot shows the Entity Links used in this configuration.



Entity Links

□	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_Endpoint_link	Session_Manager	TLS	5061	CM_SIP_Endpoints	□	5061	trusted	□	
□	CM_Trunk_Link	Session_Manager	TCP	5060	CM Trunk	□	5060	trusted	□	
□	Messaging_Link	Session_Manager	TCP	5060	Messaging	□	5060	trusted	□	

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.

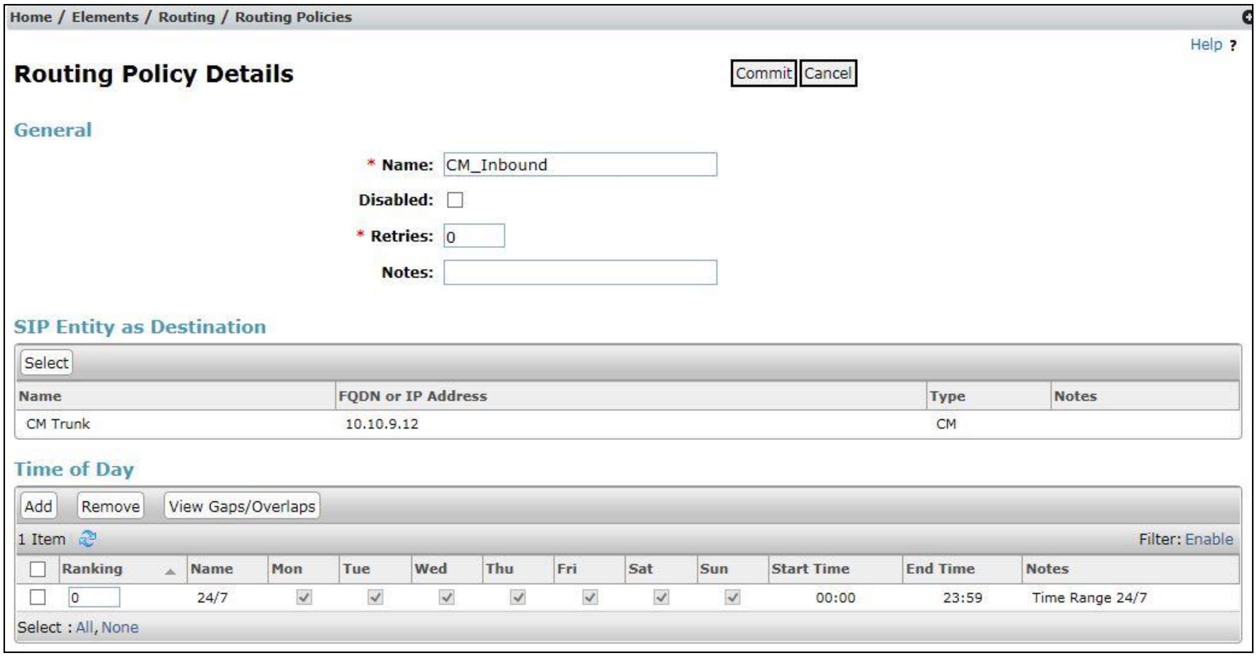
6.7. Administer Routing Policies

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, defined in **Section 6.5**, to which this routing policy applies (not shown).
- Under **Time of Day**, click **Add**, and then select the time range. **24/7** is provided as a default.

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



Routing Policy Details

General

* Name:	CM_Inbound
Disabled:	□
* Retries:	0
Notes:	

SIP Entity as Destination

Name	FQDN or IP Address	Type	Notes
CM Trunk	10.10.9.12	CM	

Time of Day

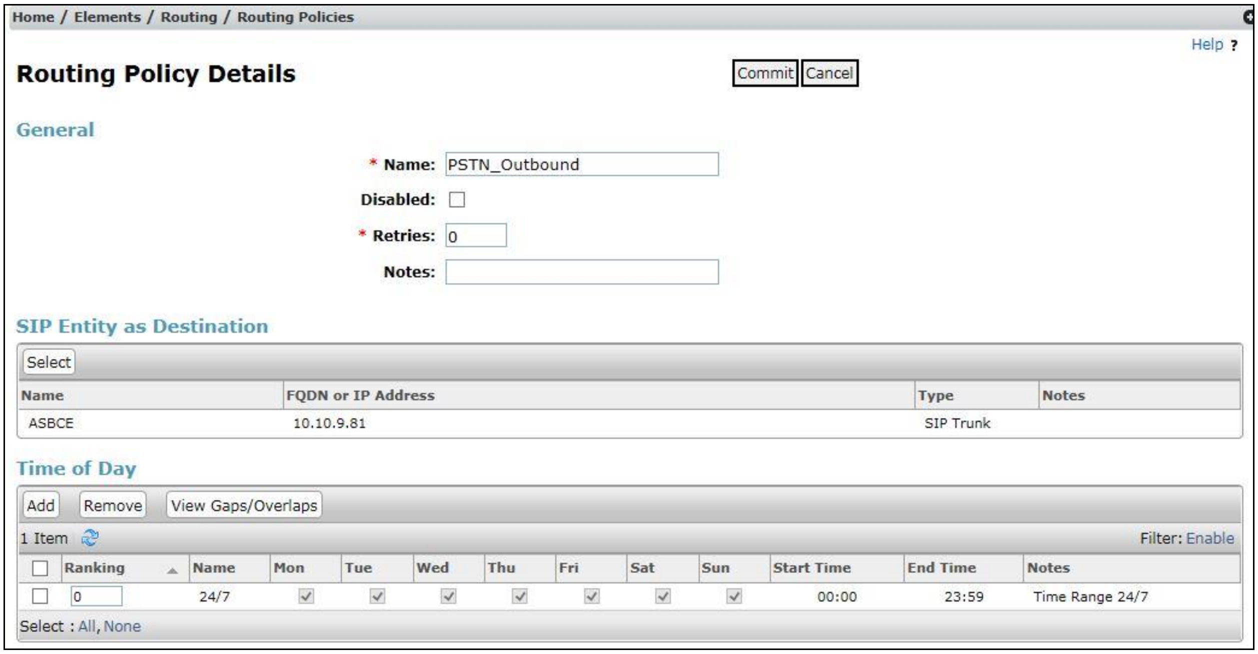
1 Item

Filter: Enable

□	Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
□	0	24/7	☑	☑	☑	☑	☑	☑	☑	00:00	23:59	Time Range 24/7

Select : All, None

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



Routing Policy Details

General

* Name:	PSTN_Outbound
Disabled:	□
* Retries:	0
Notes:	

SIP Entity as Destination

Name	FQDN or IP Address	Type	Notes
ASBCE	10.10.9.81	SIP Trunk	

Time of Day

1 Item

Filter: Enable

□	Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
□	0	24/7	☑	☑	☑	☑	☑	☑	☑	00:00	23:59	Time Range 24/7

Select : All, None

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).

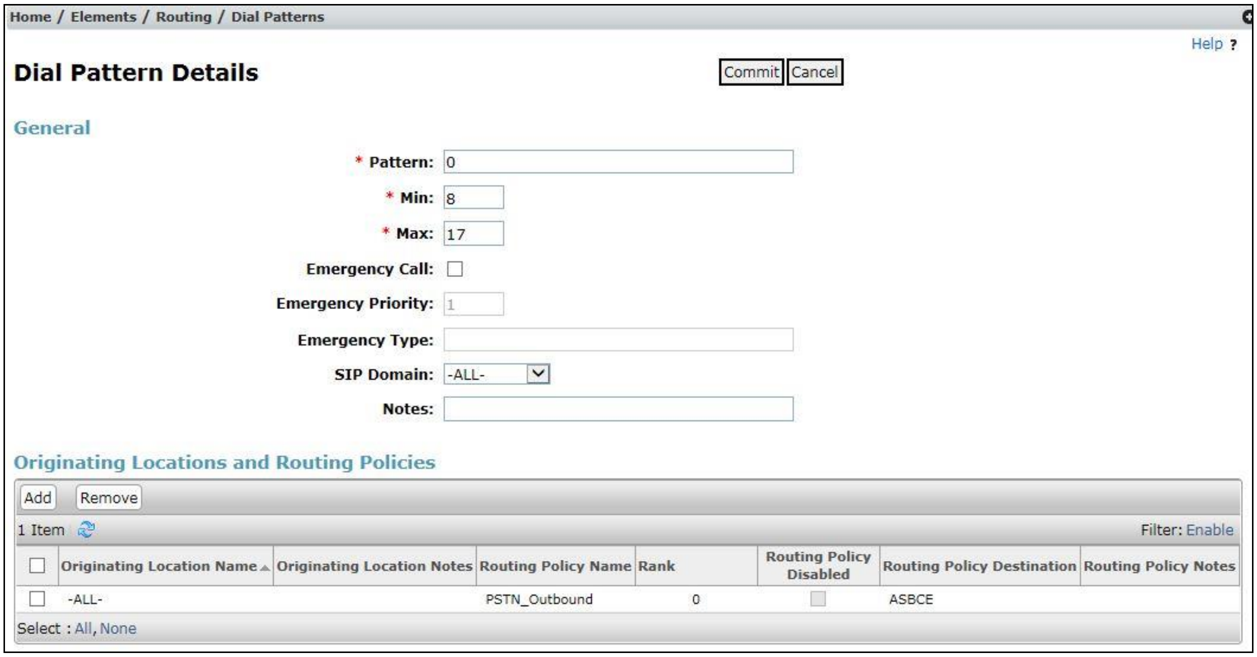
Under **General**:

- 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 one of the locations 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 all calls starting with zero to the PSTN via Telenet SIP Trunking.



Dial Pattern Details

General

* Pattern: 0

* Min: 8

* Max: 17

Emergency Call: □

Emergency Priority: 1

Emergency Type:

SIP Domain: -ALL-

Notes:

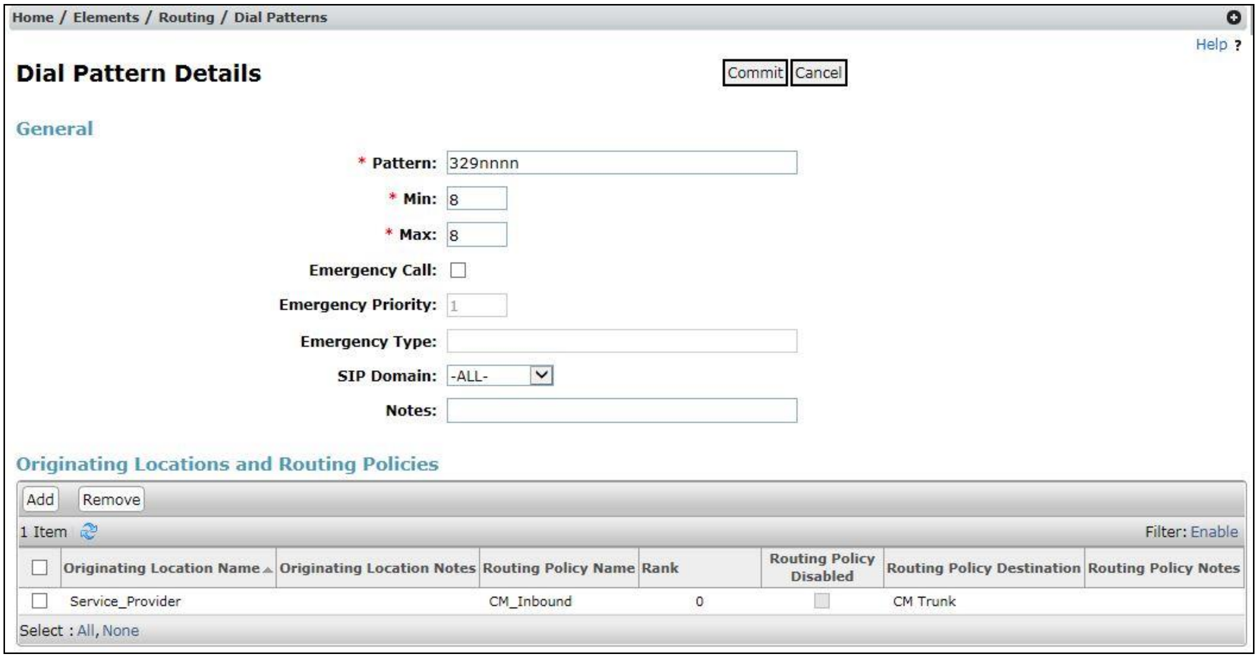
Originating Locations and Routing Policies

1 Item

□	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
□	-ALL-		PSTN_Outbound	0	□	ASBCE	

Note: Additional dial patterns (not shown) will be required for PSTN numbers that do not start with zero, for example directory enquiries. This was tested with a dial pattern for 4 digit numbers starting with 1.

The next screenshot shows the test dial pattern configured for Communication Manager. This 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.



Dial Pattern Details

General

* Pattern:	329nnnn
* Min:	8
* Max:	8
Emergency Call:	□
Emergency Priority:	1
Emergency Type:	
SIP Domain:	-ALL-
Notes:	

Originating Locations and Routing Policies

1 Item

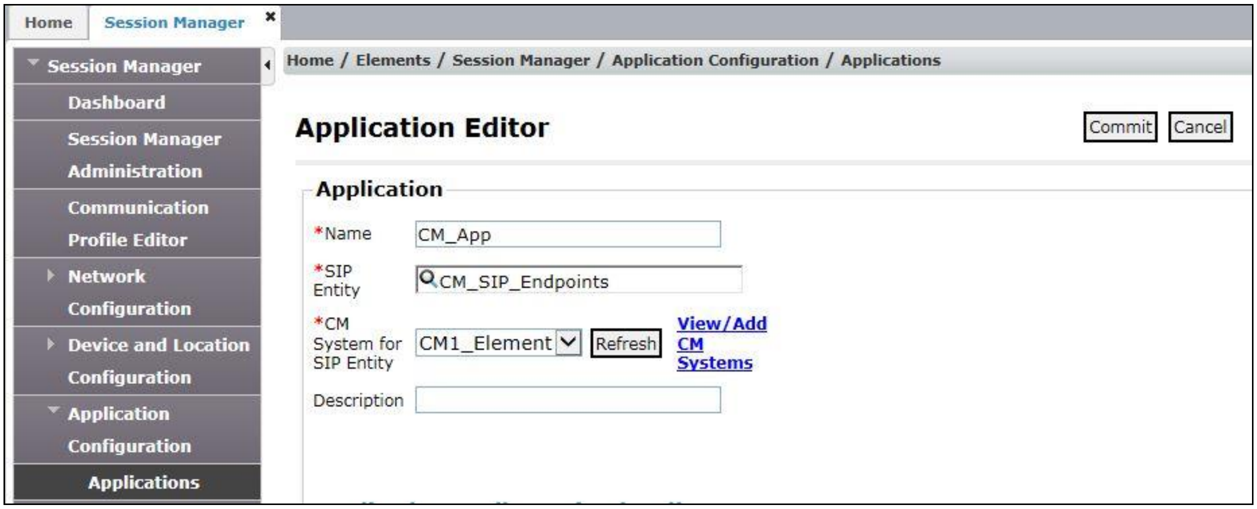
□	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
□	Service_Provider		CM_Inbound	0	□	CM Trunk	

Note: A specific location for the SIP Trunk was used for routing to Communication Manager. If required, an additional policy could be added to route calls differently if they originated within the enterprise. This may be useful if there is a requirement to route calls from one Communication Manager DDI number to another via the network.

6.9. Administer Application for Avaya Aura® Communication Manager

From the **Home** screen select **Session Manager** from the Elements 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 Endpoints described in **Section 6.5**.
- In the **CM System for SIP Entity** field select the appropriate Communication Manager from the System Manager inventory and select **Commit** to save the configuration.


Home
Session Manager
Session Manager
Dashboard
Session Manager Administration
Communication Profile Editor
Network Configuration
Device and Location Configuration
Application Configuration
Applications
Home / Elements / Session Manager / Application Configuration / Applications
Commit
Cancel
Refresh
View/Add CM Systems

Application Editor

Application

*Name: CM_App

*SIP Entity: CM_SIP_Endpoints

*CM System for SIP Entity: CM1_Element

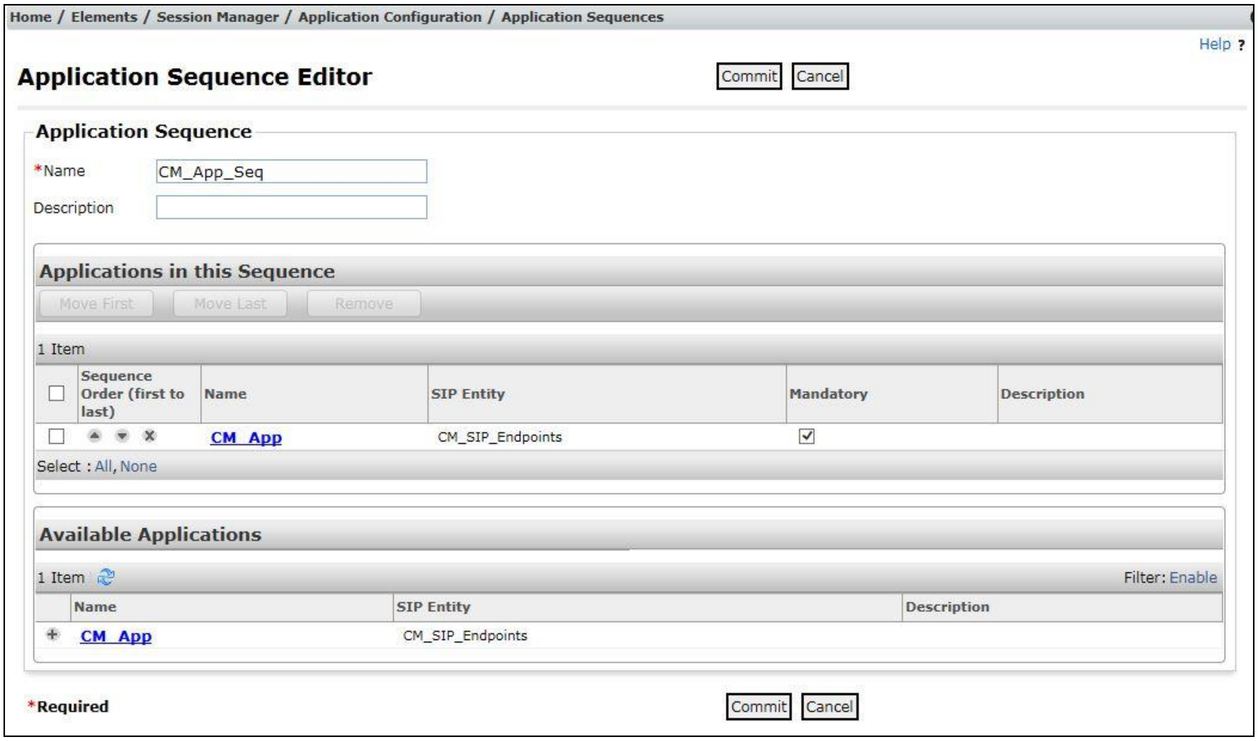
Description:

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**.



Home / Elements / Session Manager / Application Configuration / Application Sequences

Help ?

Application Sequence Editor

Commit Cancel

Application Sequence

*Name: CM_App_Seq

Description:

Applications in this Sequence

Move First | Move Last | Remove

1 Item

□	Sequence Order (first to last)	Name	SIP Entity	Mandatory	Description
□	▲ ▼ ✕	CM_App	CM_SIP_Endpoints	☑	

Select : All, None

Available Applications

1 Item | Filter: Enable

	Name	SIP Entity	Description
+	CM_App	CM_SIP_Endpoints	

*Required

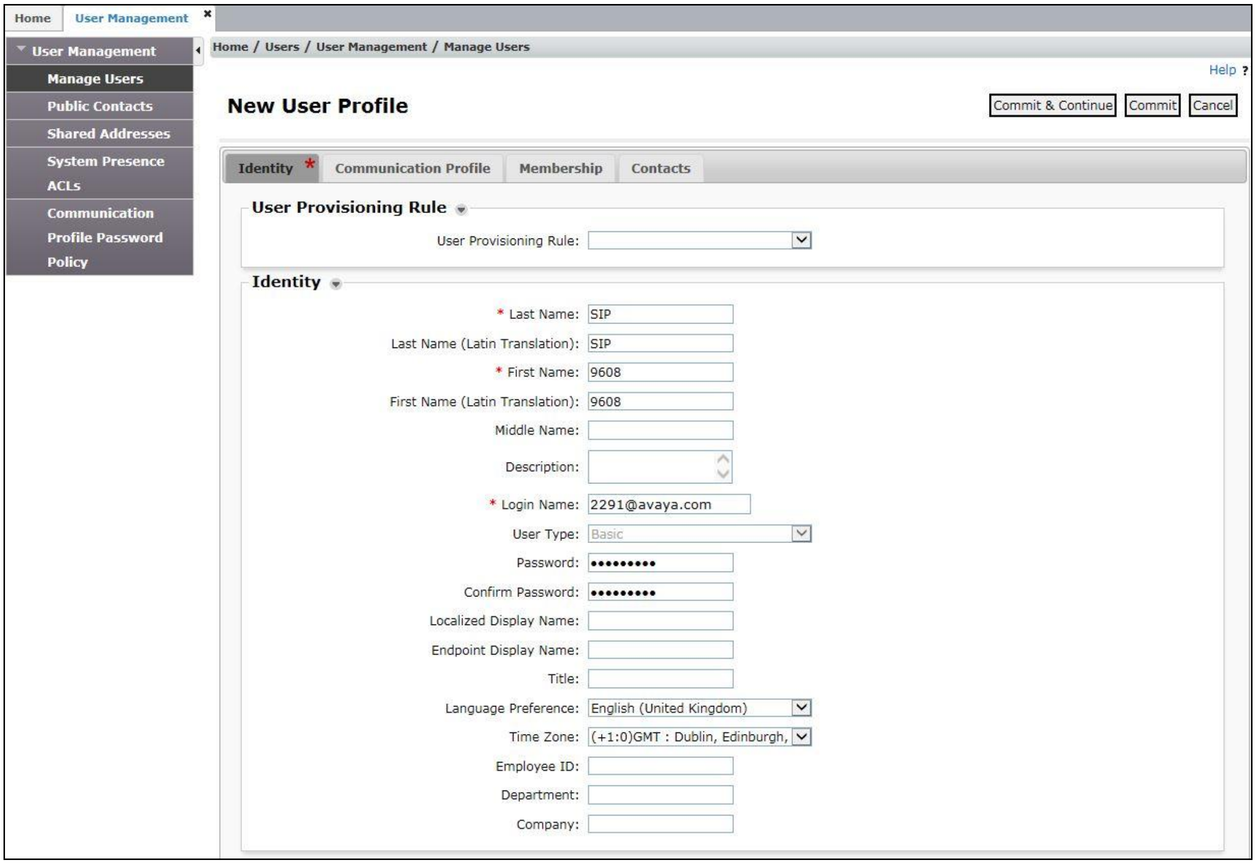
Commit Cancel

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** screen select **User Management** from the **Users** 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.



New User Profile

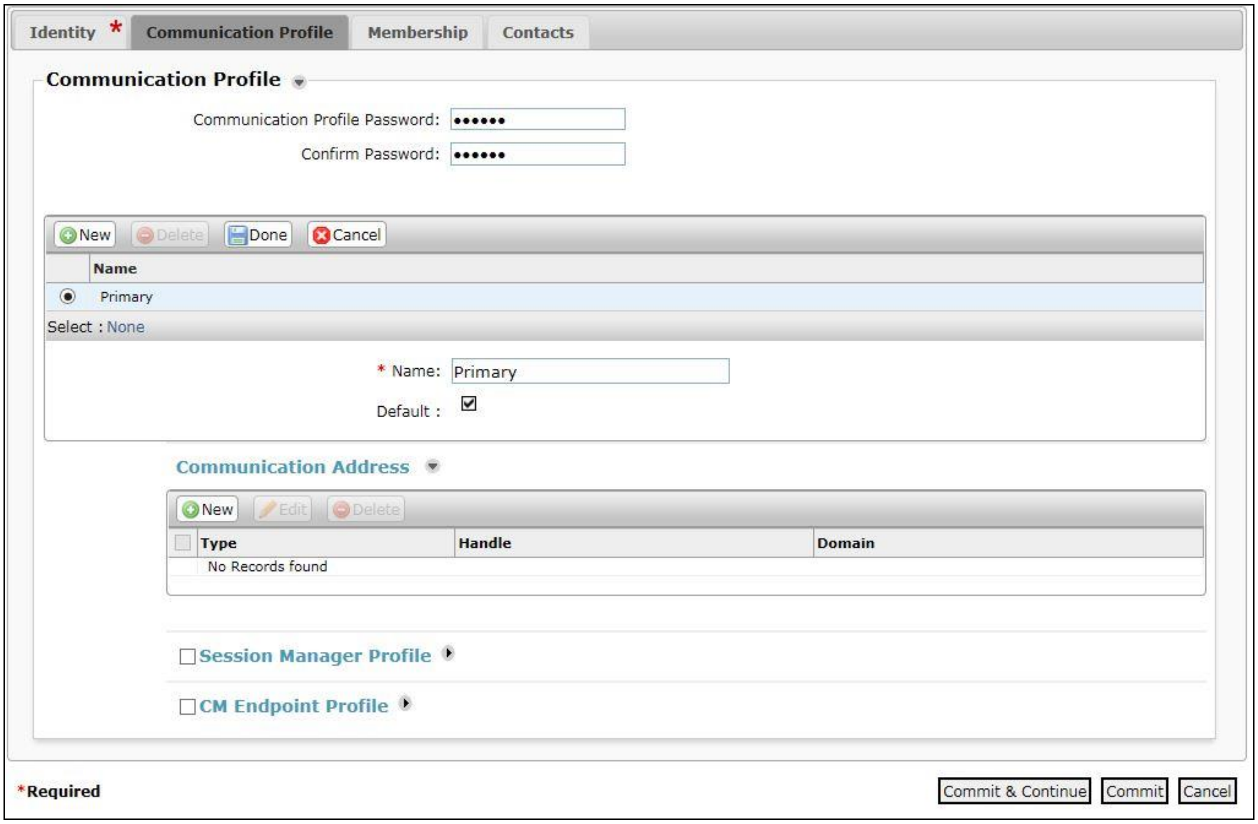
User Provisioning Rule

User Provisioning Rule:	

Identity

* Last Name:	SIP
Last Name (Latin Translation):	SIP
* First Name:	9608
First Name (Latin Translation):	9608
Middle Name:	
Description:	
* Login Name:	2291@avaya.com
User Type:	Basic
Password:	•••••••••
Confirm Password:	•••••••••
Localized Display Name:	
Endpoint Display Name:	
Title:	
Language Preference:	English (United Kingdom)
Time Zone:	(+1:0)GMT : Dublin, Edinburgh,
Employee ID:	
Department:	
Company:	

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



Communication Profile

Communication Profile Password: ••••••

Confirm Password: ••••••

	Name
●	Primary

* Name: Primary

Default : ☑

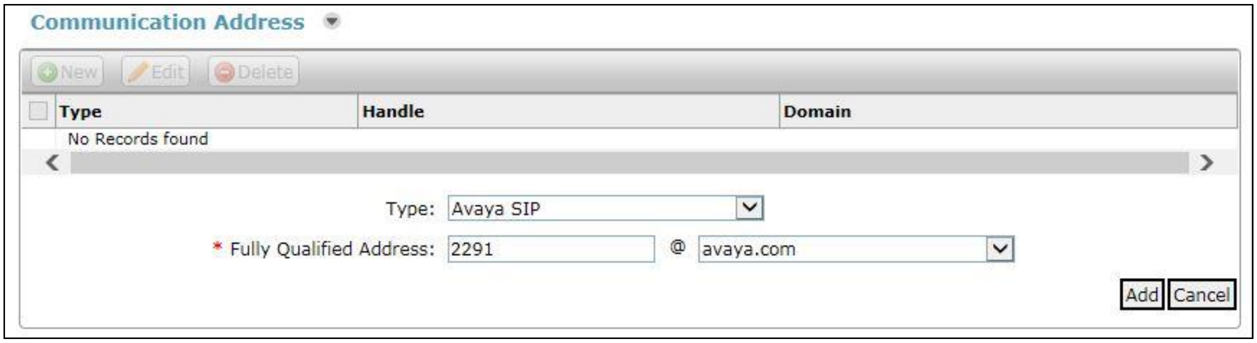
Communication Address

□	Type	Handle	Domain
	No Records found		

□ Session Manager Profile

□ CM Endpoint Profile

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.



Communication Address

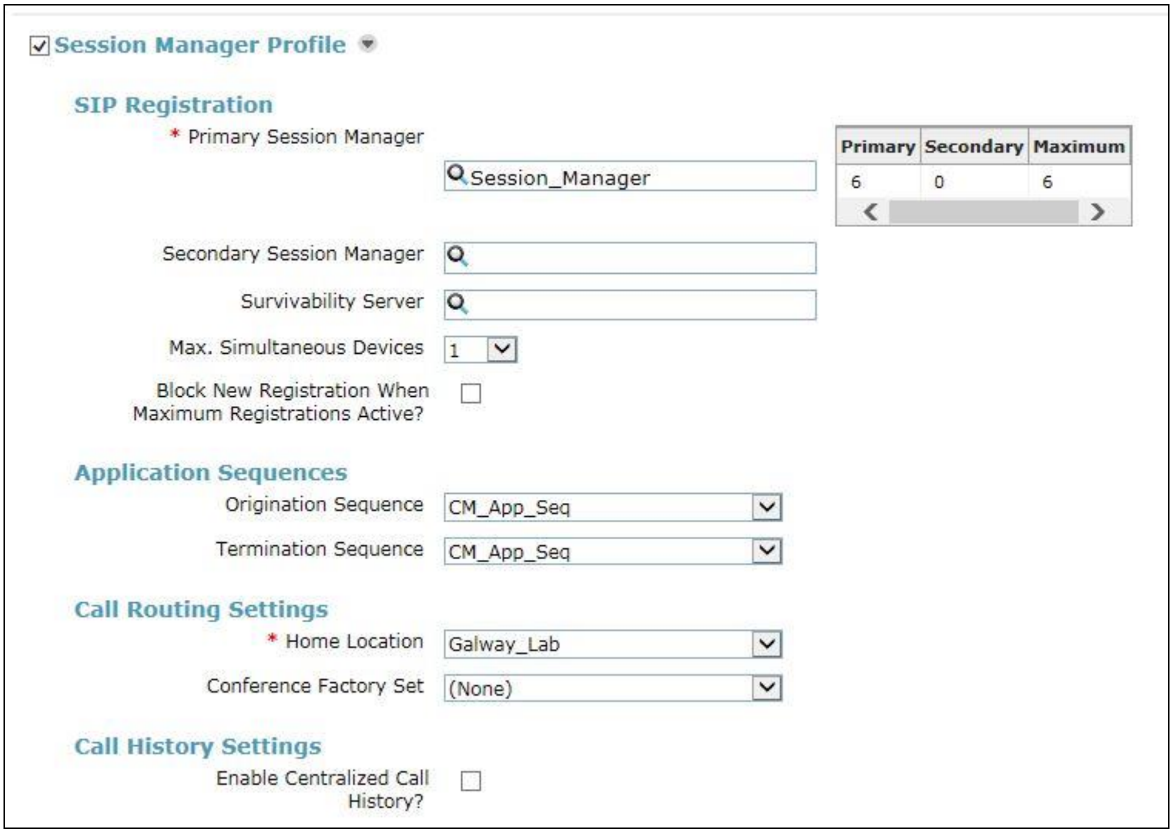
□	Type	Handle	Domain
	No Records found		

Type: Avaya SIP

* Fully Qualified Address: 2291 @ avaya.com

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.



☑ Session Manager Profile

SIP Registration

* Primary Session Manager: Session_Manager

Primary	Secondary	Maximum
6	0	6

Secondary Session Manager:

Survivability Server:

Max. Simultaneous Devices: 1

Block New Registration When Maximum Registrations Active? □

Application Sequences

Origination Sequence: CM_App_Seq

Termination Sequence: CM_App_Seq

Call Routing Settings

* Home Location: Galway_Lab

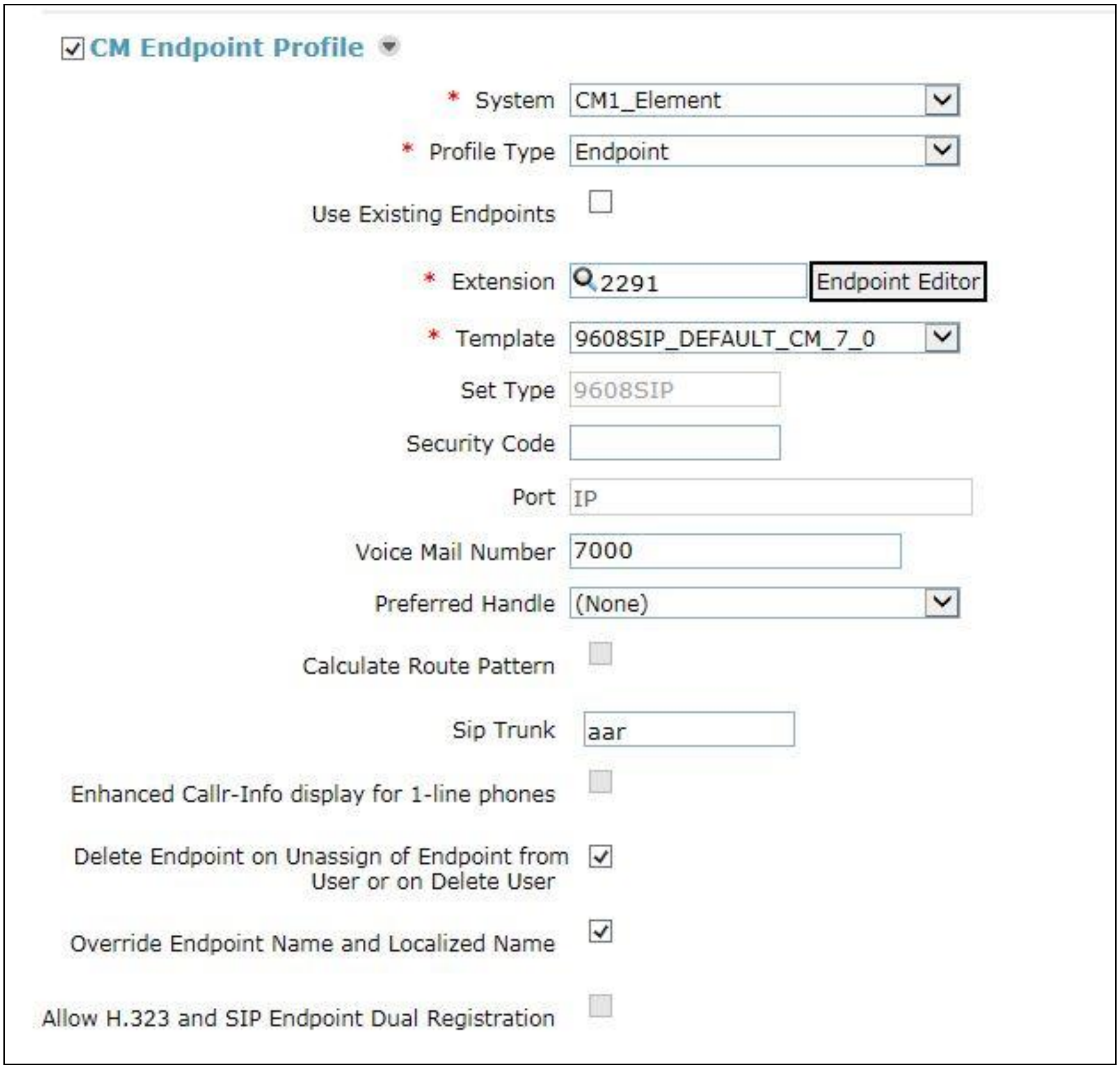
Conference Factory Set: (None)

Call History Settings

Enable Centralized Call History? □

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.
- Enter a **Voice Mail Number** if required. In the test environment, this was **7000**
- Select the **Delete Endpoint on Unassign of Endpoint from User or on Delete User** check box.



☑ CM Endpoint Profile

* System	CM1_Element
* Profile Type	Endpoint
Use Existing Endpoints	□
* Extension	2291 [Endpoint Editor]
* Template	9608SIP_DEFAULT_CM_7_0
Set Type	9608SIP
Security Code	
Port	IP
Voice Mail Number	7000
Preferred Handle	(None)
Calculate Route Pattern	□
Sip Trunk	aar
Enhanced Callr-Info display for 1-line phones	□
Delete Endpoint on Unassign of Endpoint from User or on Delete User	☑
Override Endpoint Name and Localized Name	☑
Allow H.323 and SIP Endpoint Dual Registration	□

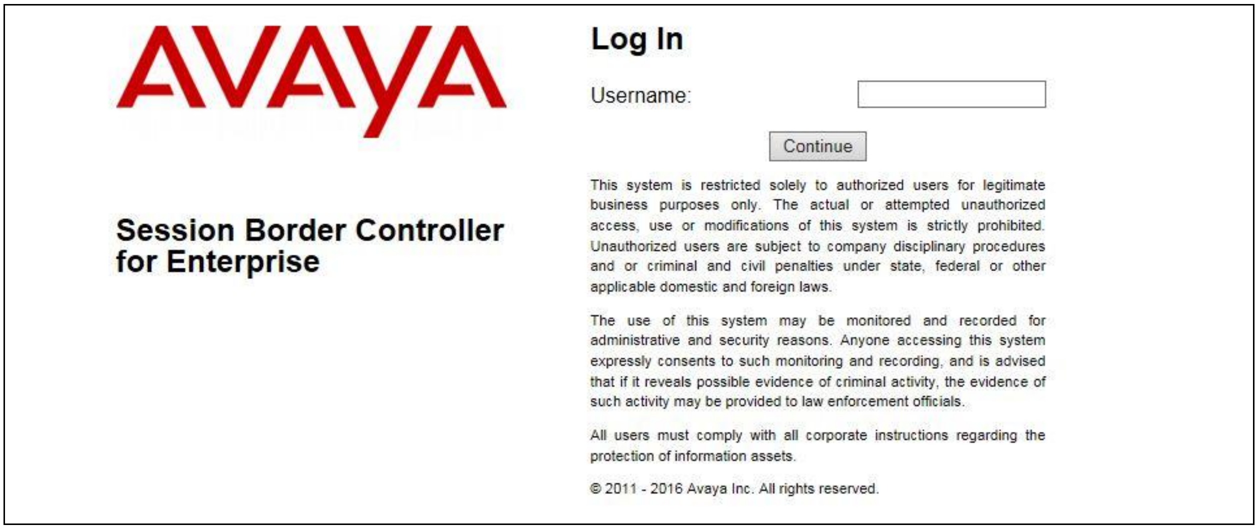
Select **Commit** (Not Shown) to save changes and the System Manager will add Communication Manager user configuration automatically.

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.



This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use or modifications of this system is strictly prohibited. Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal or other applicable domestic and foreign laws.

The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials.

All users must comply with all corporate instructions regarding the protection of information assets.

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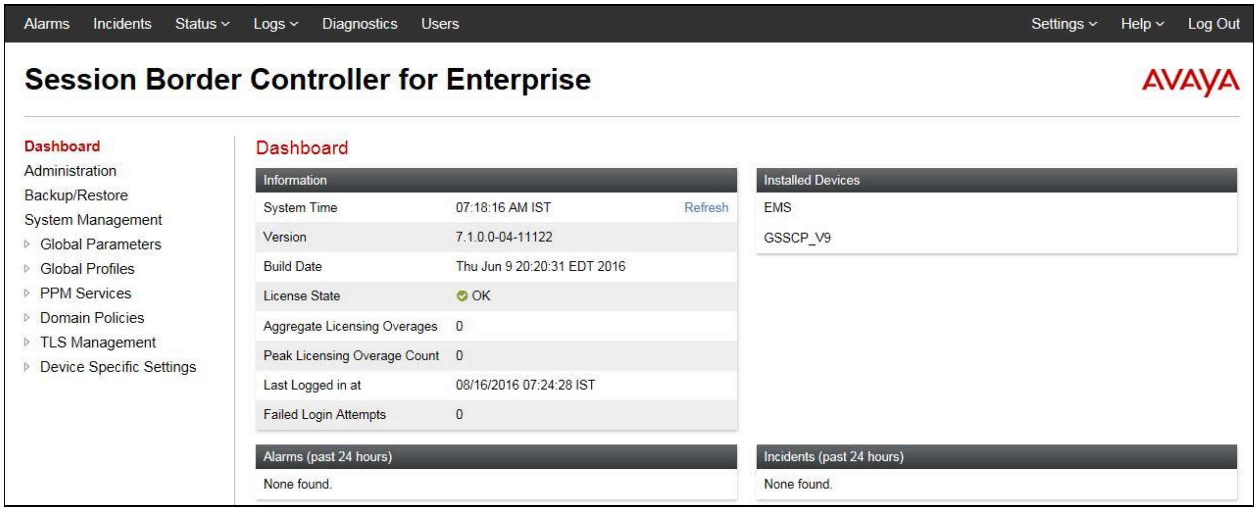
Once logged in, a dashboard is presented with a menu on the left-hand side. The menu is used as a starting point for all configuration of the Avaya SBCE.



Information		
System Time	07:18:16 AM IST	Refresh
Version	7.1.0.0-04-11122	
Build Date	Thu Jun 9 20:20:31 EDT 2016	
License State	✓ OK	
Aggregate Licensing Overages	0	
Peak Licensing Overage Count	0	
Last Logged in at	08/16/2016 07:24:28 IST	
Failed Login Attempts	0	

Installed Devices
EMS
GSSCP_V9

Alarms (past 24 hours)

None found.

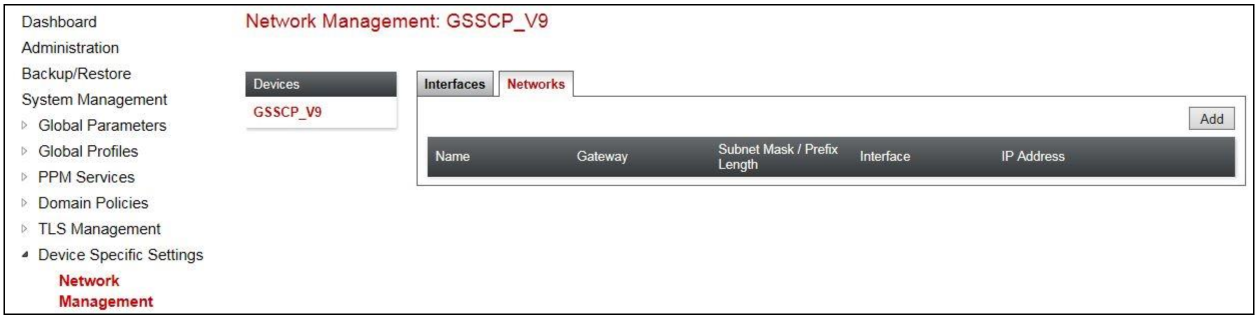
Incidents (past 24 hours)

None found.

7.2. Define Network Management

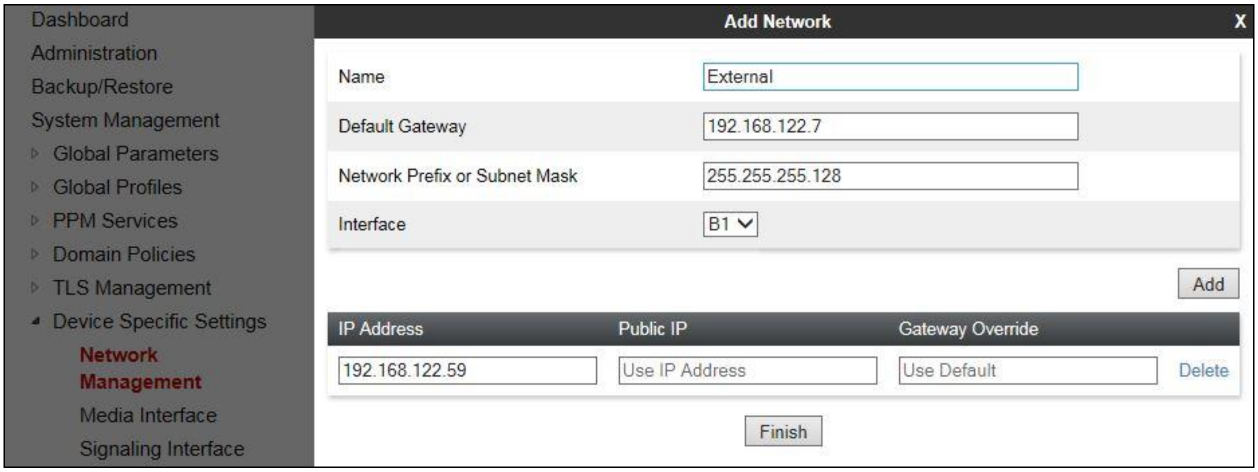
Network information is required on the Avaya SBCE to allocate IP addresses and masks to the interfaces. Note that only the **A1** and **B1** interfaces are used, typically the **A1** interface is used for the internal side and **B1** is used for external. Each side of the Avaya SBCE can have only one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



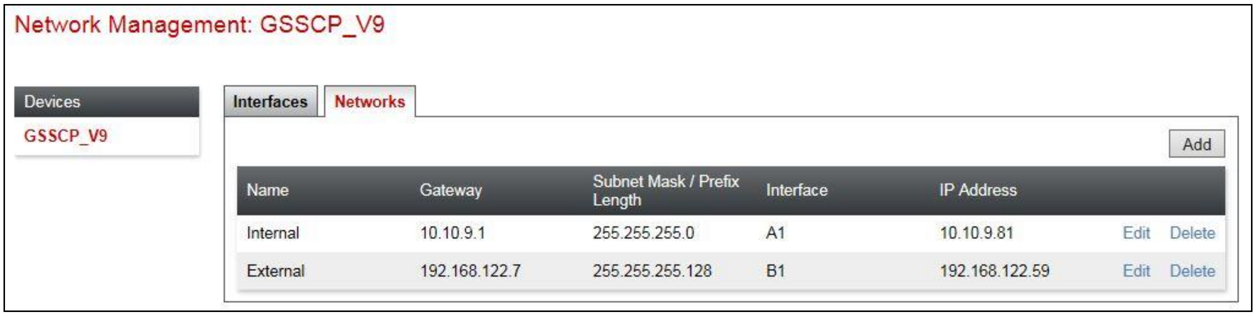
Name	External
Default Gateway	192.168.122.7
Network Prefix or Subnet Mask	255.255.255.128
Interface	B1

IP Address	Public IP	Gateway Override
192.168.122.59	Use IP Address	Use Default

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.

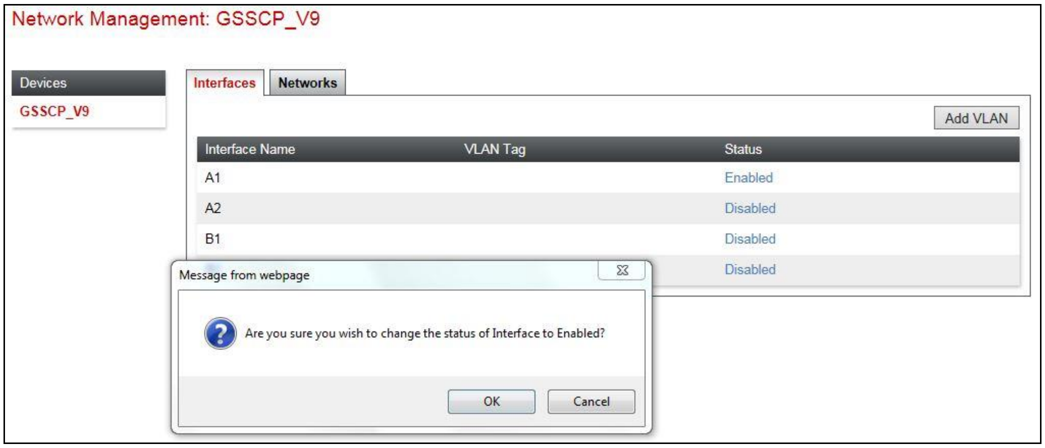
The following screenshot shows the completed Network Management configuration:



Network Management: GSSCP_V9

Name	Gateway	Subnet Mask / Prefix Length	Interface	IP Address		
Internal	10.10.9.1	255.255.255.0	A1	10.10.9.81	Edit	Delete
External	192.168.122.7	255.255.255.128	B1	192.168.122.59	Edit	Delete

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.



Network Management: GSSCP_V9

Interface Name	VLAN Tag	Status
A1		Enabled
A2		Disabled
B1		Disabled
		Disabled

Message from webpage

Are you sure you wish to change the status of Interface 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 (not shown) will appear that will indicate when the restart is complete.

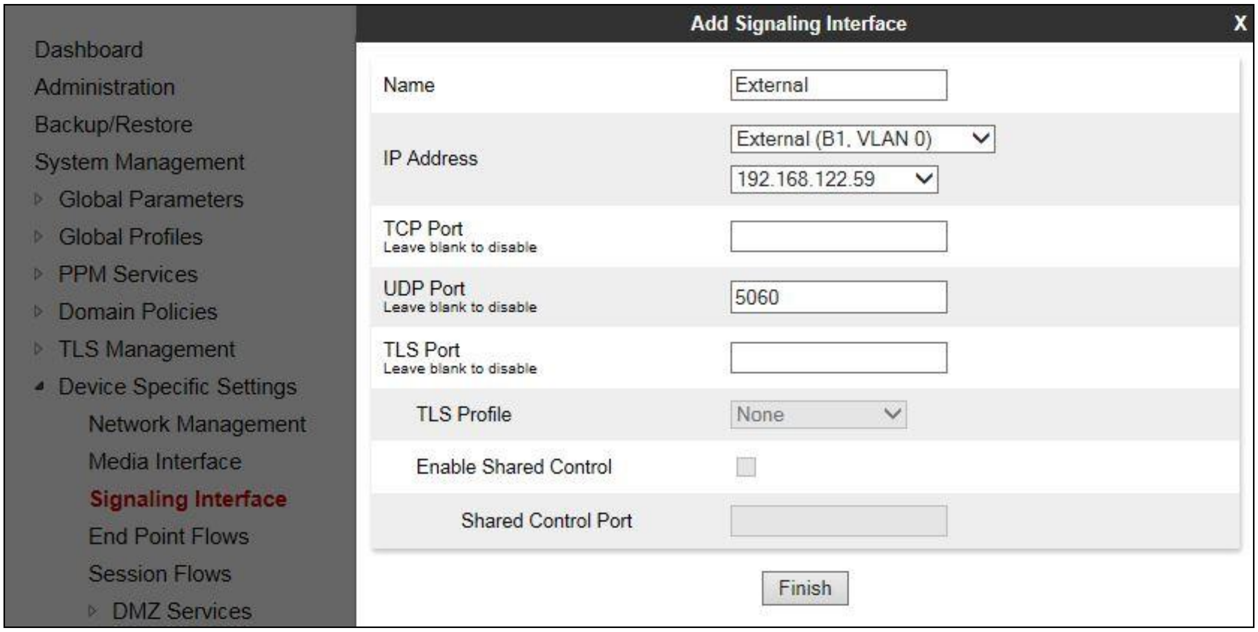
7.3. Define Interfaces

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 Telenet SIP Trunk. A signalling and media interface was required on both the internal and external sides of the Avaya SBCE. 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** 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** (not shown) 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 **192.168.122.59** for the Avaya SBCE interface on the SIP Trunk.
- Enter the UDP port number in the **UDP Port** field, **5060** is used for the Telenet SIP Trunk.
- Click on **Finish**



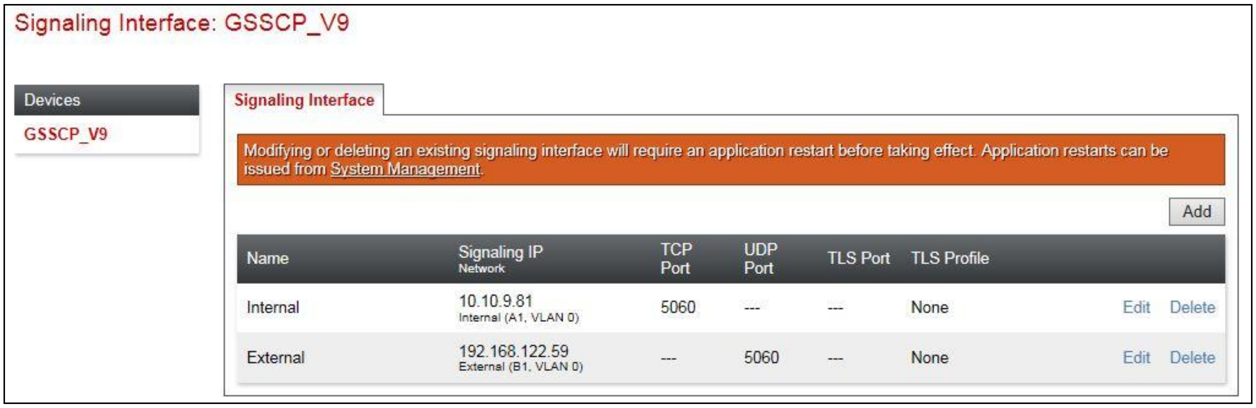
Add Signaling Interface

Name	External
IP Address	External (B1, VLAN 0) 192.168.122.59
TCP Port Leave blank to disable	
UDP Port Leave blank to disable	5060
TLS Port Leave blank to disable	
TLS Profile	None
Enable Shared Control	□
Shared Control Port	

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:



Signaling Interface: GSSCP_V9

Modifying or deleting an existing signaling interface will require an application restart before taking effect. Application restarts can be issued from System Management.

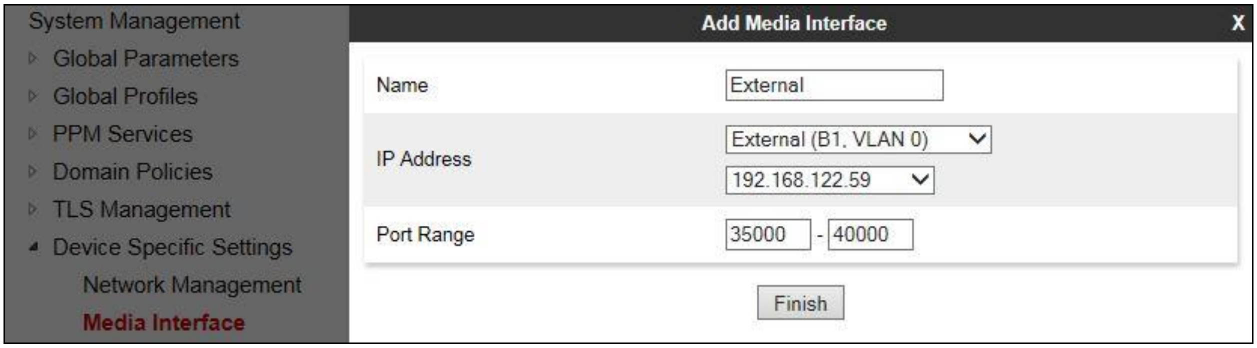
Name	Signaling IP Network	TCP Port	UDP Port	TLS Port	TLS Profile		
Internal	10.10.9.81 Internal (A1, VLAN 0)	5060	---	---	None	Edit	Delete
External	192.168.122.59 External (B1, VLAN 0)	---	5060	---	None	Edit	Delete

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 **192.168.122.59**.
- Define the RTP **Port Range** for the media path with the Telenet SIP Trunking, during testing this was left at default values of **35000** - **40000**.



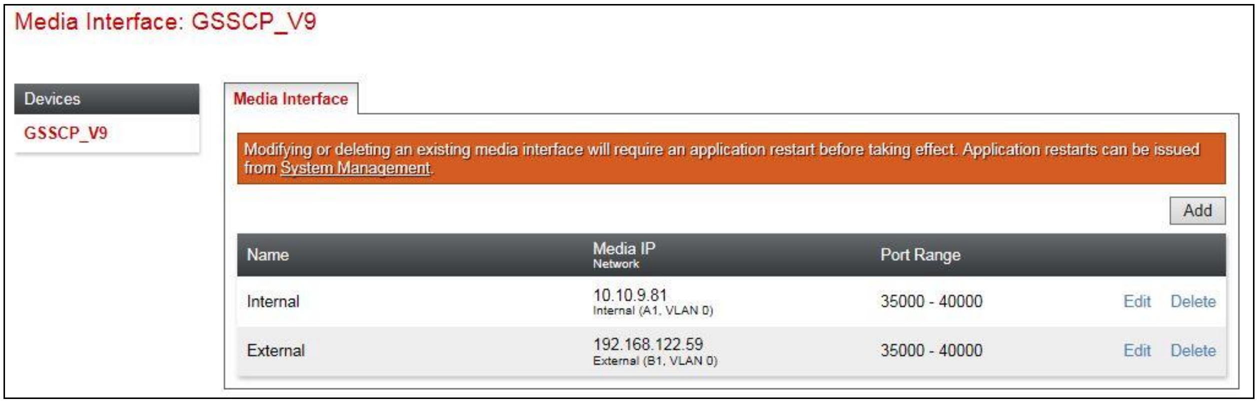
Add Media Interface

Name	External
IP Address	External (B1, VLAN 0) 192.168.122.59
Port Range	35000 - 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:



Media Interface: GSSCP_V9

Modifying or deleting an existing media interface will require an application restart before taking effect. Application restarts can be issued from System Management.

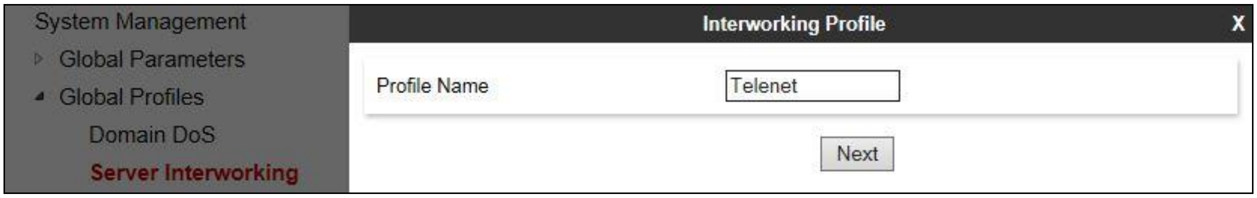
Name	Media IP Network	Port Range		
Internal	10.10.9.81 Internal (A1, VLAN 0)	35000 - 40000	Edit	Delete
External	192.168.122.59 External (B1, VLAN 0)	35000 - 40000	Edit	Delete

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 Telenet SIP Trunking service 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.

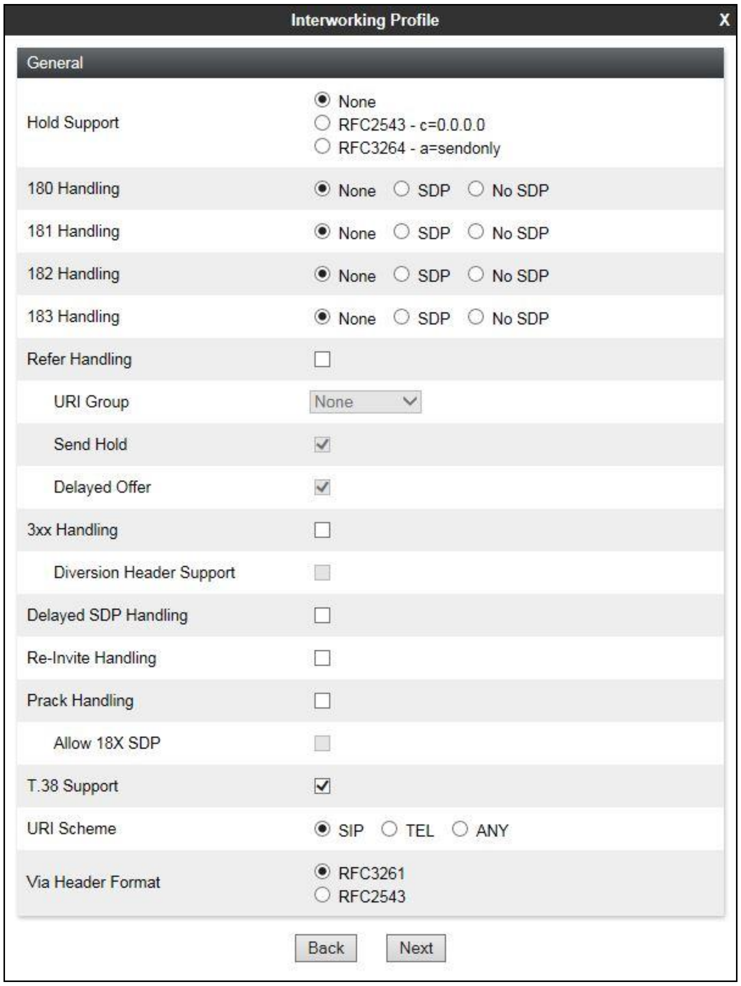
To define server interworking on the Avaya SBCE, navigate to **Global Profiles → Server Interworking** in the main menu on the left hand side. To define Server Interworking for the Telenet SIP Trunking service, click on **Add** (not shown). A pop-up menu is generated. In the **Name** field enter a descriptive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



Profile Name: Telenet

Telenet’s preferred method of fax transmission is via G.711 so T.38 support is not necessary. This document shows how to define it however, as it may be required in the future.

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

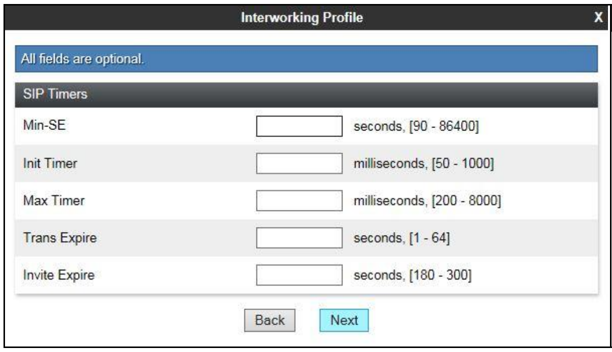


Interworking Profile X

General

Hold Support	● None ○ RFC2543 - c=0.0.0.0 ○ RFC3264 - a=sendonly
180 Handling	● None ○ SDP ○ No SDP
181 Handling	● None ○ SDP ○ No SDP
182 Handling	● None ○ SDP ○ No SDP
183 Handling	● None ○ SDP ○ No SDP
Refer Handling	□
URI Group	None
Send Hold	☑
Delayed Offer	☑
3xx Handling	□
Diversion Header Support	□
Delayed SDP Handling	□
Re-Invite Handling	□
Prack Handling	□
Allow 18X SDP	□
T.38 Support	☑
URI Scheme	● SIP ○ TEL ○ ANY
Via Header Format	● RFC3261 ○ RFC2543

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

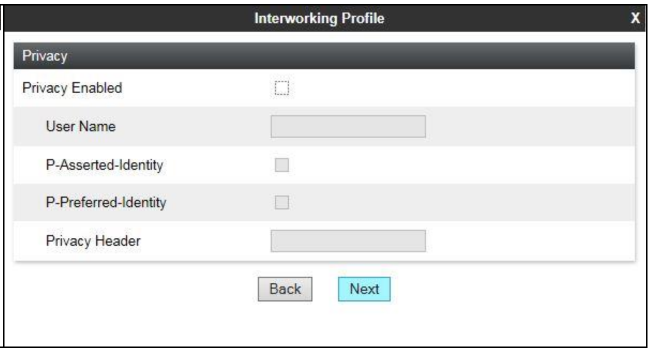


Interworking Profile X

All fields are optional.

SIP Timers

Min-SE		seconds, [90 - 86400]
Init Timer		milliseconds, [50 - 1000]
Max Timer		milliseconds, [200 - 8000]
Trans Expire		seconds, [1 - 64]
Invite Expire		seconds, [180 - 300]

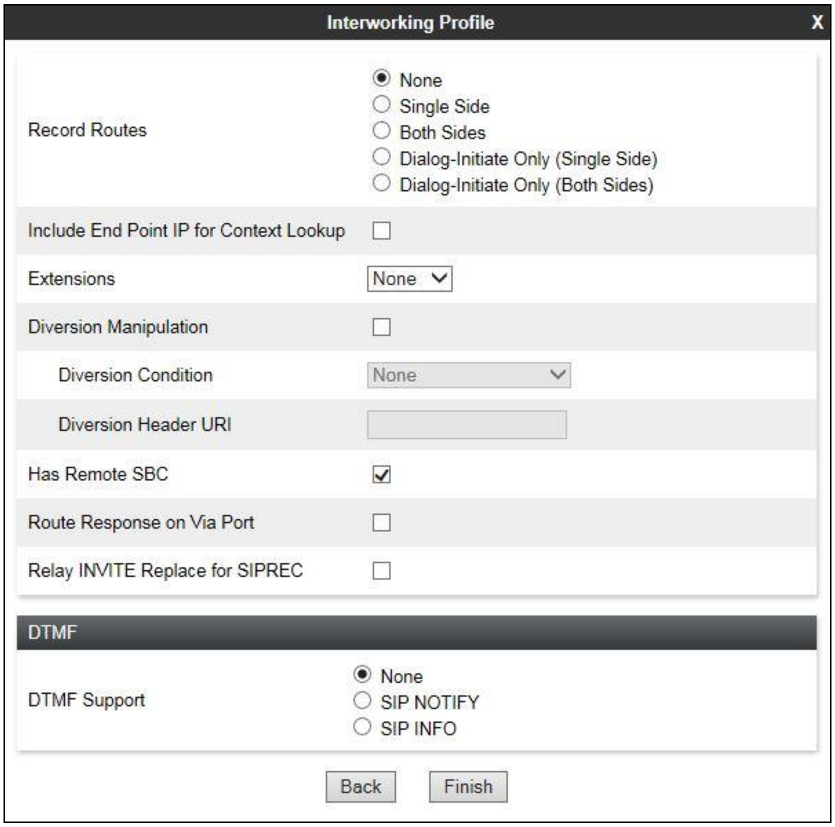


Interworking Profile X

Privacy

Privacy Enabled	□
User Name	
P-Asserted-Identity	□
P-Preferred-Identity	□
Privacy Header	

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**



Interworking Profile

Record Routes	● None ○ Single Side ○ Both Sides ○ Dialog-Initiate Only (Single Side) ○ Dialog-Initiate Only (Both Sides)
Include End Point IP for Context Lookup	□
Extensions	None
Diversion Manipulation	□
Diversion Condition	None
Diversion Header URI	
Has Remote SBC	☑
Route Response on Via Port	□
Relay INVITE Replace for SIPREC	□

DTMF

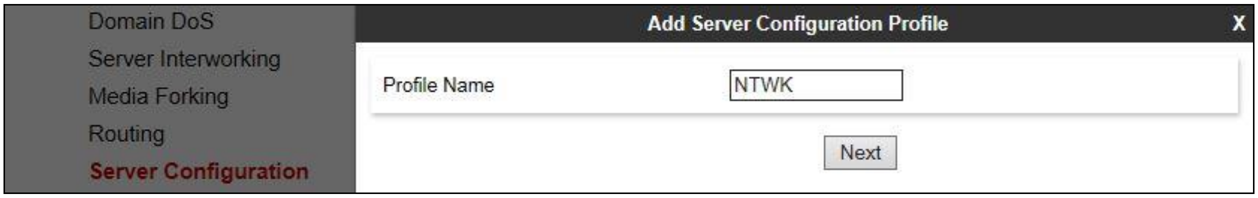
DTMF Support	● None ○ SIP NOTIFY ○ SIP INFO

Repeat the process to define Server Interworking for Session Manager using the same parameter settings apart from **Record Routes** which is set to **Both Sides** as the Session Manager uses the Record-Route header.

7.5. Define Servers

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

To define the Telenet SIP Trunking Server, navigate to **Global Profiles → Server Configuration** in the main menu on the left hand side. Click on **Add** (not shown) and enter an appropriate name in the pop-up menu.

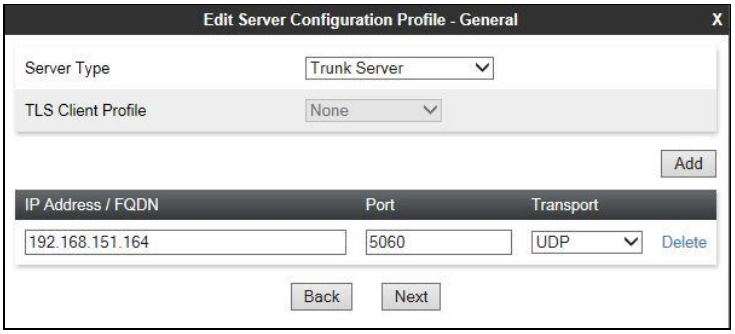


Add Server Configuration Profile

Profile Name: NTWK

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 Telenet SIP Trunking IP address.
- In the **Port** box, enter the port to be used for the SIP Trunk.
- In the **Transport** drop down menu, select **UDP**.
- Click on **Next**.

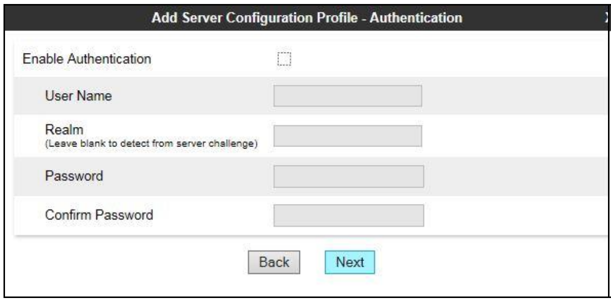


Edit Server Configuration Profile - General

Server Type	Trunk Server
TLS Client Profile	None

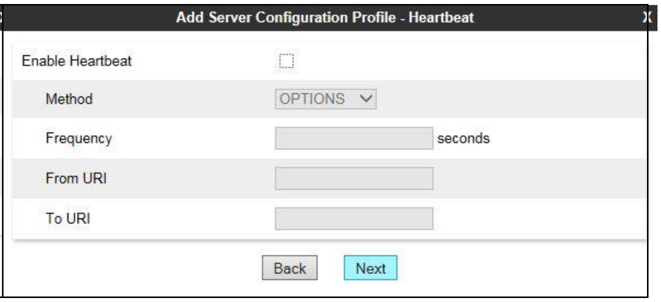
IP Address / FQDN	Port	Transport	
192.168.151.164	5060	UDP	Delete

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



Add Server Configuration Profile - Authentication

Enable Authentication	□
User Name	
Realm (Leave blank to detect from server challenge)	
Password	
Confirm Password	

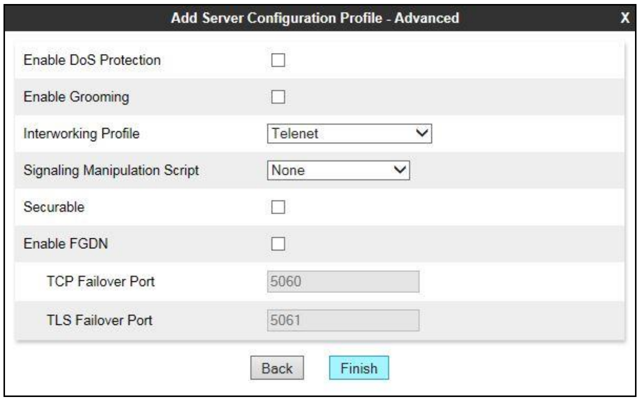


Add Server Configuration Profile - Heartbeat

Enable Heartbeat	□
Method	OPTIONS
Frequency	seconds
From URI	
To URI	

Click on **Next** again to get to the final dialogue box. This contains the **Advanced** settings:

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



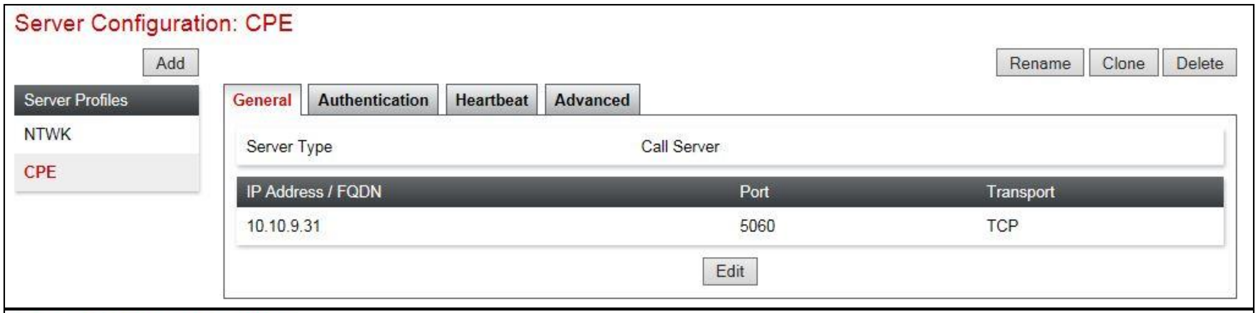
Add Server Configuration Profile - Advanced

Enable DoS Protection	□
Enable Grooming	□
Interworking Profile	Telenet
Signaling Manipulation Script	None
Securable	□
Enable FGDN	□
TCP Failover Port	5060
TLS Failover Port	5061

Use the process described 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 (not shown).
- 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 (not shown).

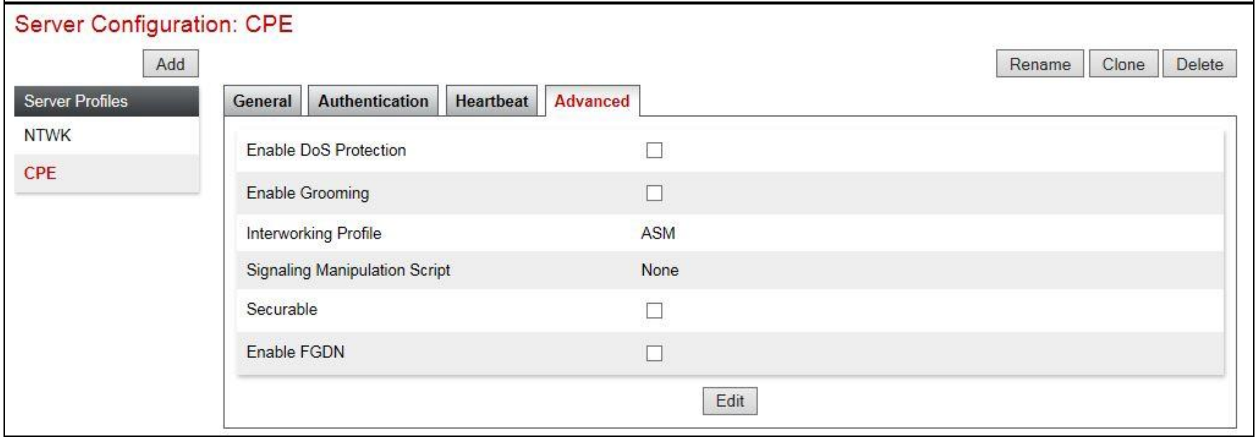
The following screenshots show the **General** and **Advanced** tabs of the completed Server Configuration:



Server Configuration: CPE

Server Type	Call Server

IP Address / FQDN	Port	Transport
10.10.9.31	5060	TCP

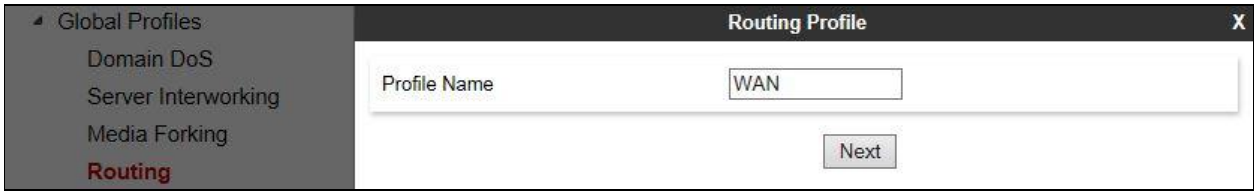


Server Configuration: CPE

Enable DoS Protection	□
Enable Grooming	□
Interworking Profile	ASM
Signaling Manipulation Script	None
Securable	□
Enable FGDN	□

7.6. Define Routing

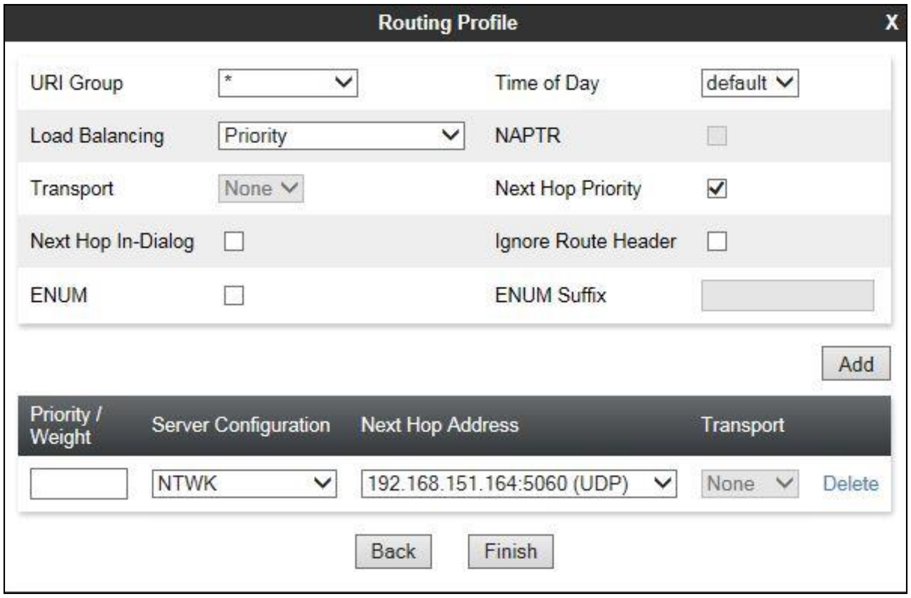
Routing information is required for routing to the Telenet SIP Trunking on the external side and Session Manager on the internal side. The IP addresses and ports defined here will be used as the destination addresses for signalling. To define routing to Telenet SIP Trunking, navigate to **Global Profiles → Routing** in the main menu on the left hand side. Click on **Add** (not shown) and enter an appropriate name in the dialogue box.



Profile Name: WAN

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

- During testing, **Load Balancing** was not required and was left at the default value of **Priority**.
- Click on **Add** to specify an IP address for the SIP Trunk.
- Assign a priority in the **Priority / Weight** field, during testing **1** was used.
- Select the Server Configuration defined in **Section 7.5** in the **Server Configuration** drop down menu. This automatically populates the **Next Hop Address** field
- Click **Finish**.



Routing Profile

URI Group	*	Time of Day	default
Load Balancing	Priority	NAPTR	□
Transport	None	Next Hop Priority	☑
Next Hop In-Dialog	□	Ignore Route Header	□
ENUM	□	ENUM Suffix	

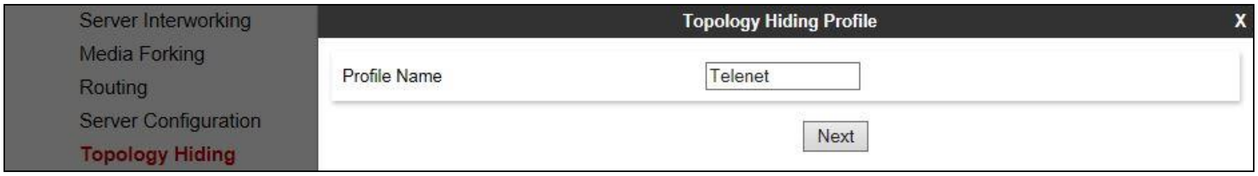
Priority / Weight	Server Configuration	Next Hop Address	Transport	
	NTWK	192.168.151.164:5060 (UDP)	None	Delete

Repeat the process for the Routing Profile for Session Manager. In the test environment, this was called “LAN” and the Server Configuration was “CPE”.

7.7. Topology Hiding

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

To define Topology Hiding for Telenet SIP Trunking, navigate to **Global Profiles → Topology Hiding** in the main menu on the left hand side. Click on **Add** (not shown) to bring up a dialogue box, assign an appropriate name and click on **Next** to configure Topology Hiding for each header as required:

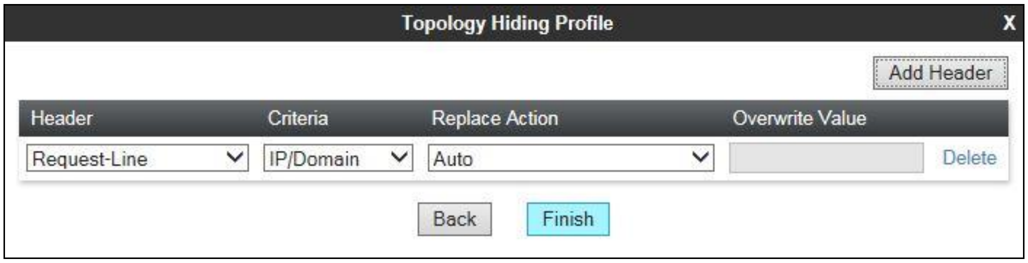


Topology Hiding Profile

Profile Name	Telenet

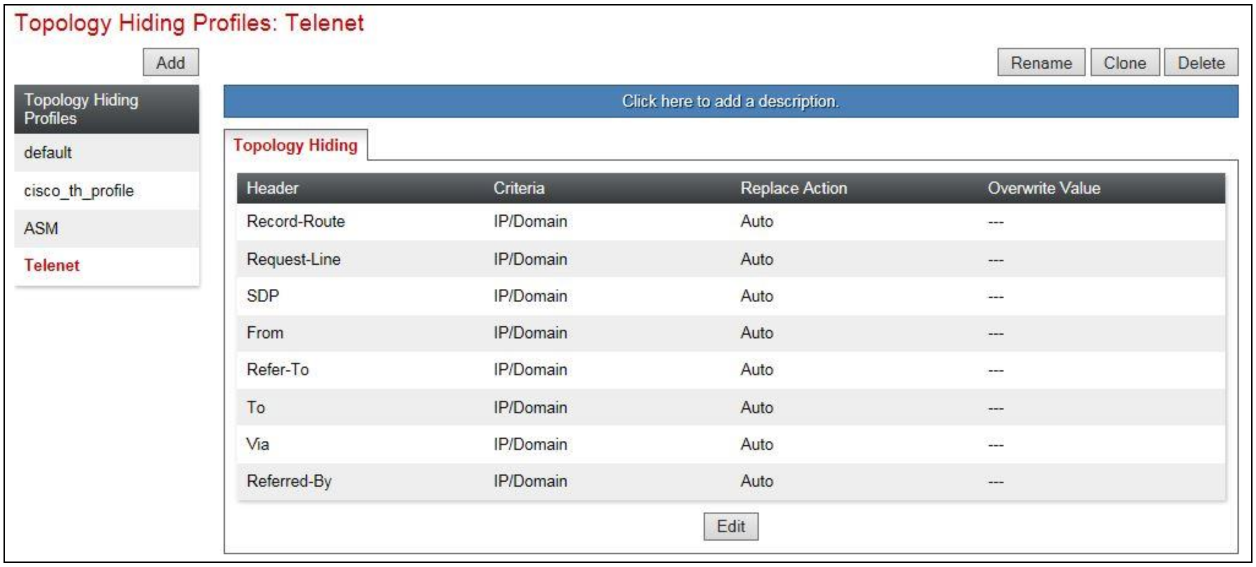
Enter details in the **Topology Hiding Profile** pop-up menu.

- 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.



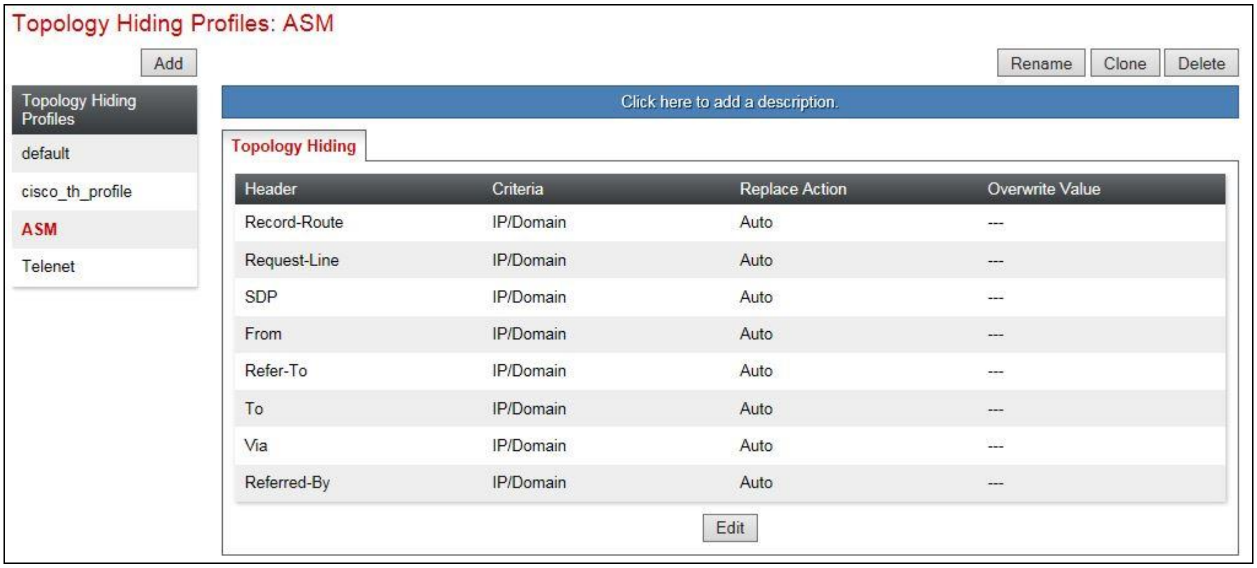
Header	Criteria	Replace Action	Overwrite Value
Request-Line	IP/Domain	Auto	

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



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

To define Topology hiding for Session Manager, follow the same process. This can be simplified by cloning the profile defined for Telenet SIP Trunking. Do this by highlighting the profile defined for Telenet and clicking on **Clone**. Enter an appropriate name for Session Manager and click on **Next** (not shown). Make any changes where required, in the test environment the settings were left at the same values.



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

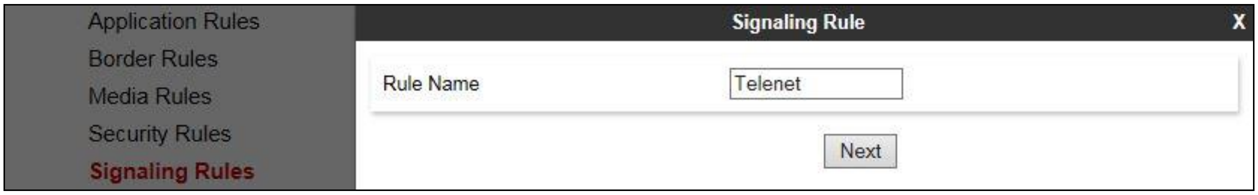
7.8. End Point Policy Groups

End Point Policy Groups are used to bring together a number of different rules for use in a server flow described in **Section 7.9**. During testing of Telenet SIP Trunking, a Signalling Rule was used so an End Point Policy Group was also required to apply it to the Server Flow

7.8.1. Signalling Rules

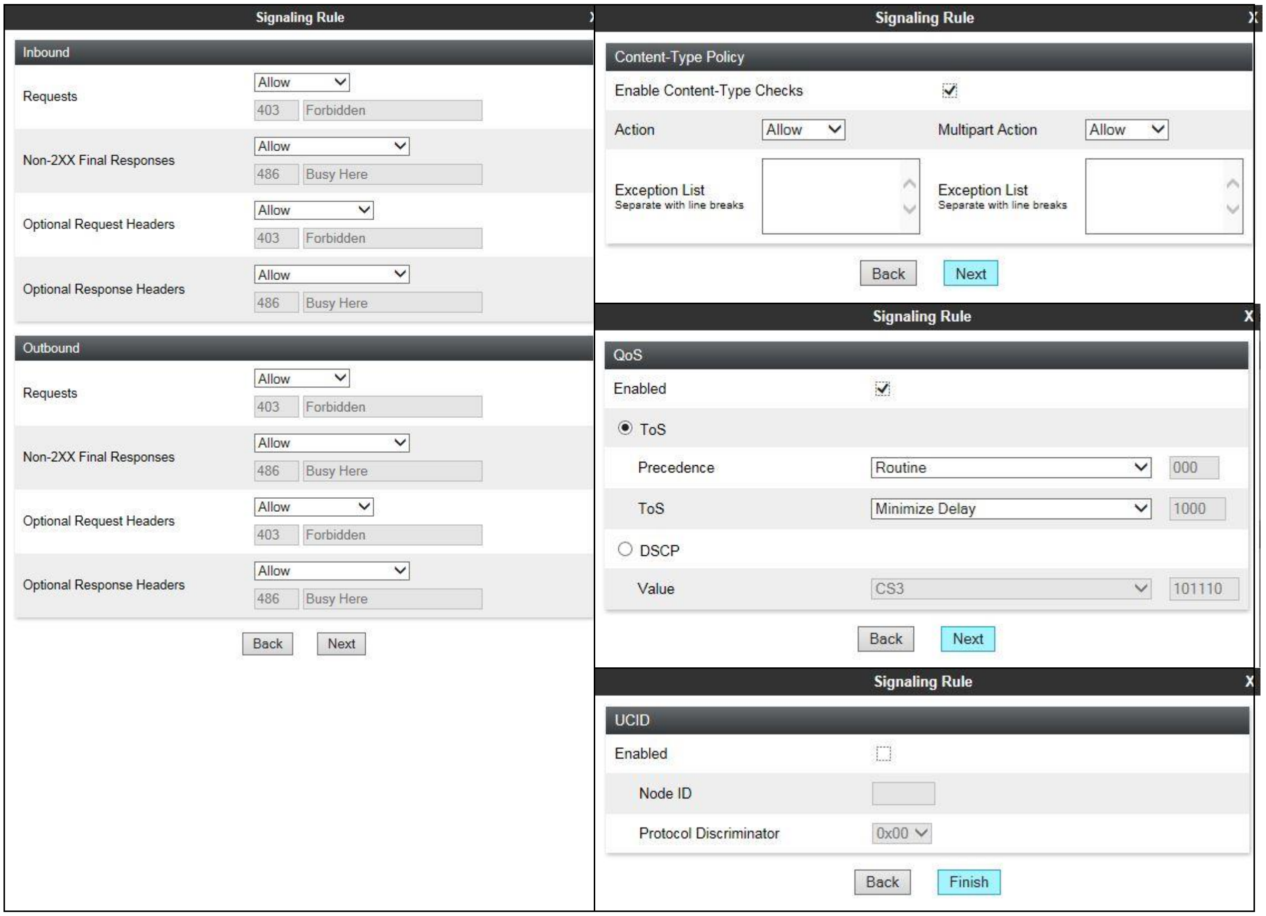
Signalling rules are a mechanism on the Avaya SBCE to handle any incompatible signalling that may be encountered on the SIP Trunk of a particular Service Provider. In the case of the Telenet SIP Trunk, it was found that Session Manager rejected the Proxy-Require header in the INVITE message of incoming calls with CLI Restricted (See **Section 2.2**). A signalling Rule was used to remove the Proxy-Require header.

To define a signalling rule to remove the Proxy-Require header, navigate to **Domain Policies →Signaling Rules** in the main menu on the left hand side. Click on **Add** (not shown) and enter details in the Signaling Rule pop-up box. In the **Rule Name** field enter a descriptive name for the signalling rule, in this case **Trunk**.



Rule Name: Telenet

Click on **Next** 3 times leaving the settings at default values then click on **Finish.**



Inbound

Requests	Allow	403	Forbidden
Non-2XX Final Responses	Allow	486	Busy Here
Optional Request Headers	Allow	403	Forbidden
Optional Response Headers	Allow	486	Busy Here

Outbound

Requests	Allow	403	Forbidden
Non-2XX Final Responses	Allow	486	Busy Here
Optional Request Headers	Allow	403	Forbidden
Optional Response Headers	Allow	486	Busy Here

Content-Type Policy

Enable Content-Type Checks ☑

Action: Allow   Multipart Action: Allow

Exception List (Separate with line breaks)   Exception List (Separate with line breaks)

QoS

Enabled ☑

● ToS

Precedence: Routine 000

ToS: Minimize Delay 1000

○ DSCP

Value: CS3 101110

UCID

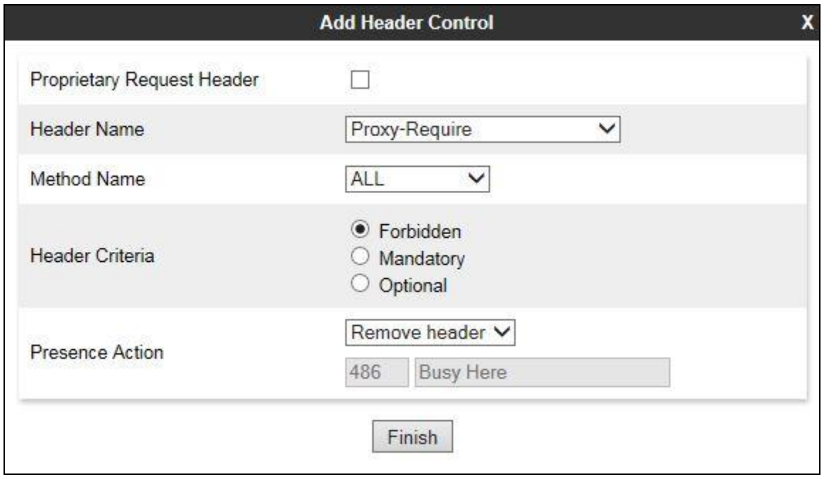
Enabled □

Node ID

Protocol Discriminator: 0x00

Once the rule is created, it is edited to provide the required functionality. To edit the rule, navigate to **Domain Policies →Signaling Rules** in the main menu on the left hand side and highlight the rule.

- Click on the **Request Headers** tab and then click on **Add In Header Control** for Headers to be deleted from SIP INVITE messages coming from the Telenet SIP Trunk.
- Select a **Proxy-Require** from the **Header Name** drop down menu
- Select **ALL** from the **Method Name** drop down menu.
- Check the **Forbidden** button in the **Header Criteria** menu.
- Select **Remove Header** from the **Presence Action** drop down menu.



Proprietary Request Header □

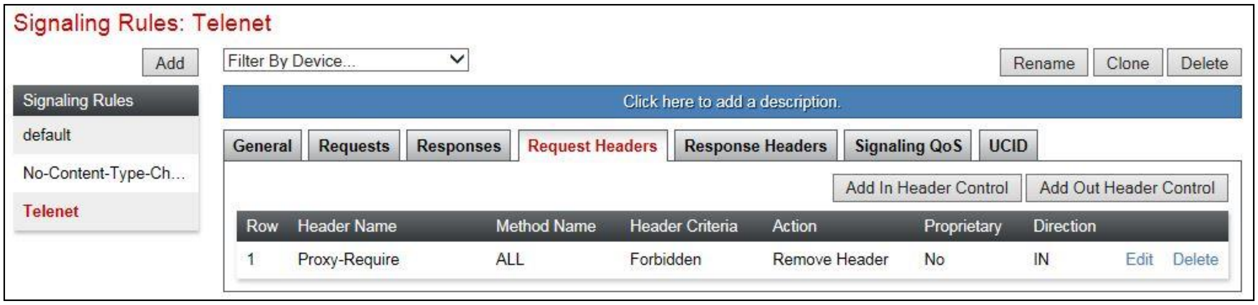
Header Name: Proxy-Require

Method Name: ALL

Header Criteria: ● Forbidden ○ Mandatory ○ Optional

Presence Action: Remove header; 486 Busy Here

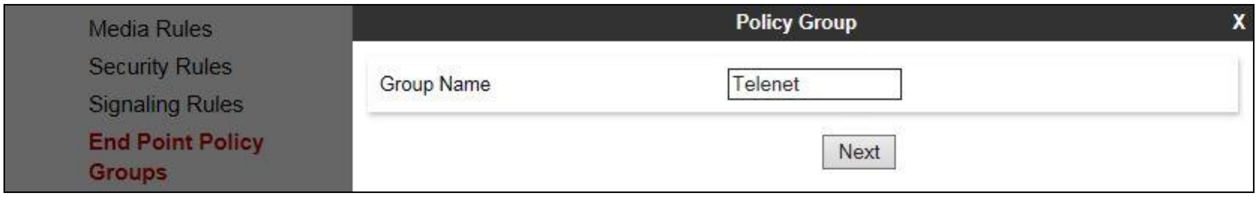
The screenshot shows the Request Headers tab of the Signalling Rule which removes the **Proxy-Require** header from INVITE messages coming **IN** from the Telenet SIP Trunk:



Row	Header Name	Method Name	Header Criteria	Action	Proprietary	Direction		
1	Proxy-Require	ALL	Forbidden	Remove Header	No	IN	Edit	Delete

7.8.2. End Point Policy Groups

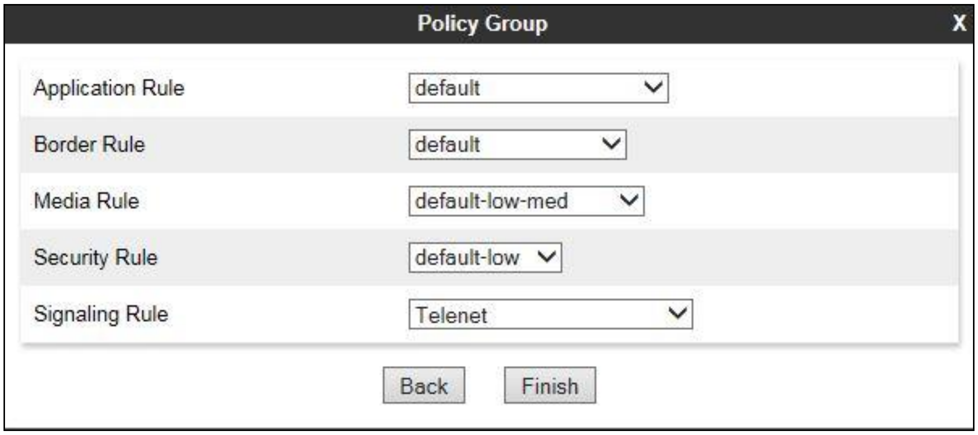
End Point Policy Groups are required to implement the signalling rules. To define one for use in the SIP Trunk server flow to remove the Proxy-Require header, navigate to **Domain Policies → End Point Policy Groups** in the main menu on the left hand side. Click on **Add** and enter an appropriate name in the pop-up box.



Group Name: Telenet

Click on **Next** to configure the Policy Set. Enter details as follows:.

- Leave the **Application Rule**, **Border Rule**, **Media Rule** and **Security Rule** at their default values.
- Select the **Signaling Rule** created in the previous section in the drop down menu, in this case **Trunk**.
- Click on **Finish**.



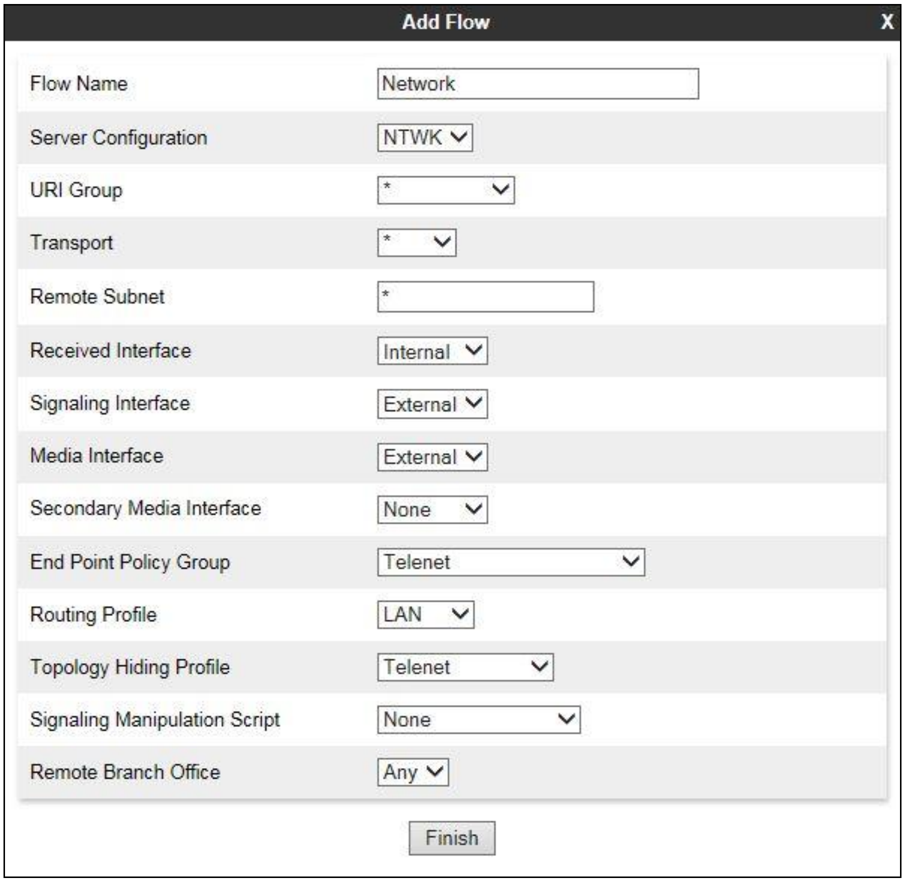
Application Rule	default
Border Rule	default
Media Rule	default-low-med
Security Rule	default-low
Signaling Rule	Telenet

7.9. Server Flows

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

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

- Click on the **Server Flows** tab (not shown).
- Select **Add Flow** (not shown) and enter details in the pop-up menu.
- In the **Flow Name** field enter a descriptive name for the server flow for the Telenet SIP Trunk, in the test environment **Network** was used.
- In the **Server Configuration** drop-down menu, select the server configuration for the Telenet 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 Telenet SIP Trunk defined in **Section 7.7** and click **Finish**.

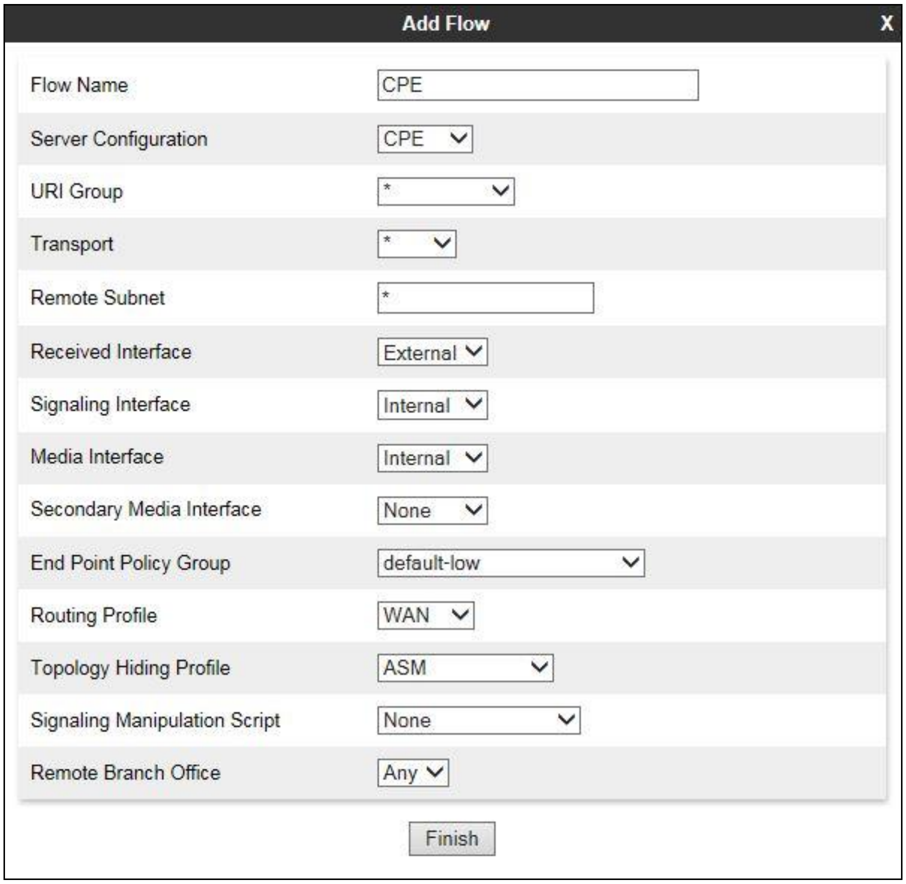


Add Flow

Flow Name	Network
Server Configuration	NTWK
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Internal
Signaling Interface	External
Media Interface	External
Secondary Media Interface	None
End Point Policy Group	Telenet
Routing Profile	LAN
Topology Hiding Profile	Telenet
Signaling Manipulation Script	None
Remote Branch Office	Any

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

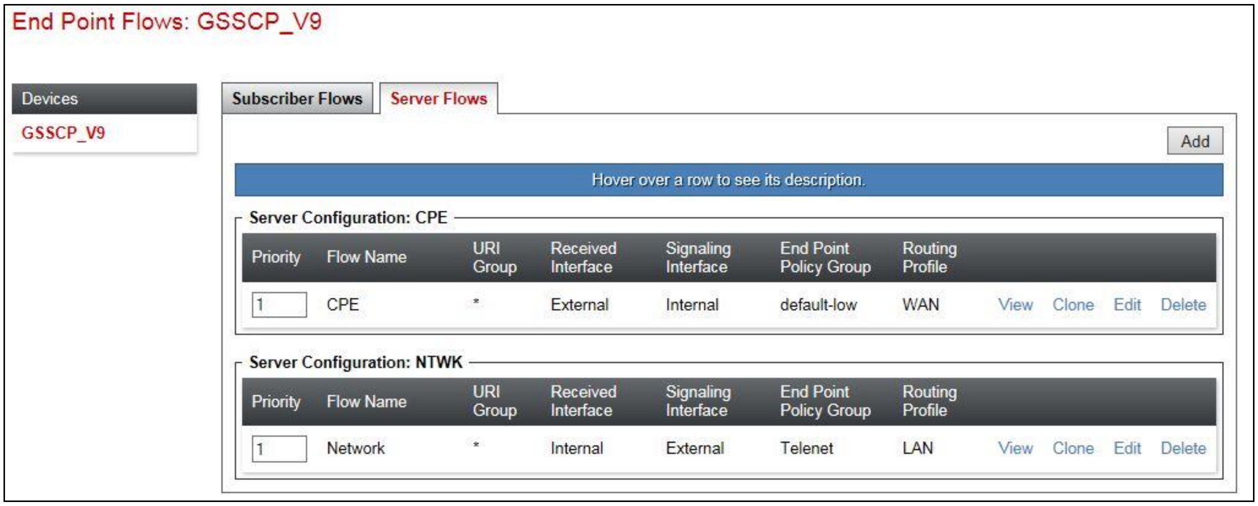
- Click on the **Server Flows** tab (not shown).
- Select **Add Flow** (not shown) and enter details in the pop-up menu.
- In the **Flow Name** field enter a descriptive name for the server flow for Session Manager, in the test environment **CPE** was used.
- In the **Server Configuration** drop-down 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 Telenet SIP Trunking 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**.



Add Flow

Flow Name	CPE
Server Configuration	CPE
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	External
Signaling Interface	Internal
Media Interface	Internal
Secondary Media Interface	None
End Point Policy Group	default-low
Routing Profile	WAN
Topology Hiding Profile	ASM
Signaling Manipulation Script	None
Remote Branch Office	Any

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


End Point Flows: GSSCP_V9
Devices
GSSCP_V9
Subscriber Flows
Server Flows
Add
Hover over a row to see its description.

Server Configuration: CPE

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile				
1	CPE	*	External	Internal	default-low	WAN	View	Clone	Edit	Delete

Server Configuration: NTWK

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile				
1	Network	*	Internal	External	Telenet	LAN	View	Clone	Edit	Delete

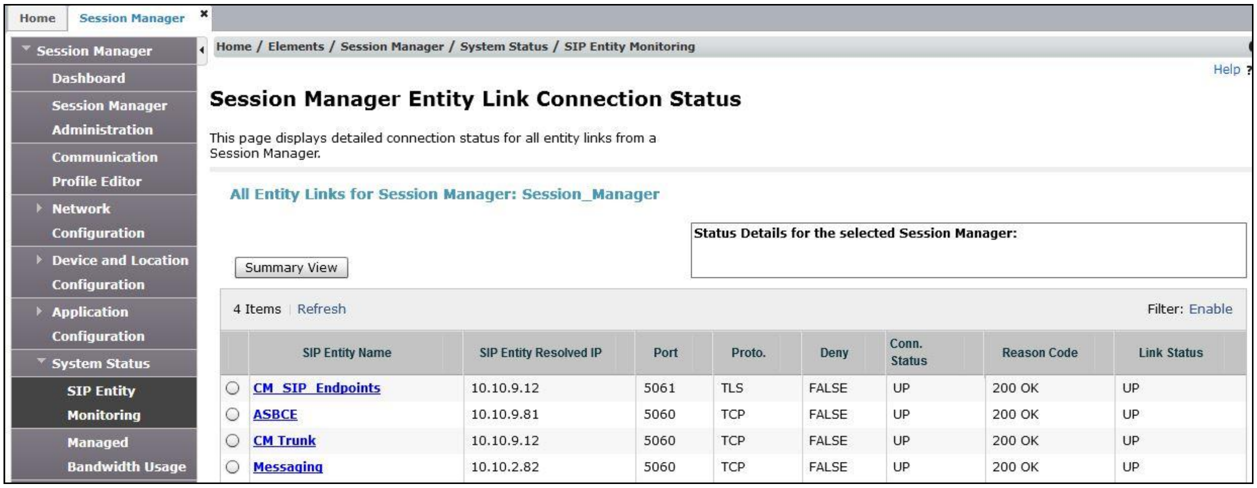
8. Configure the Telenet SIP Trunking Equipment

The configuration of the Telenet 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 Telenet equipment and system configuration please contact an authorized Telenet 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** screen 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**.


Home
Session Manager
Session Manager
Dashboard
Session Manager Administration
Communication Profile Editor
Network Configuration
Device and Location Configuration
Application Configuration
System Status
SIP Entity Monitoring
Managed Bandwidth Usage
Home / Elements / Session Manager / System Status / SIP Entity Monitoring
Help ?
Summary View
Refresh
Filter: Enable

Session Manager Entity Link Connection Status

This page displays detailed connection status for all entity links from a Session Manager.

All Entity Links for Session Manager: Session_Manager

Status Details for the selected Session Manager:

4 Items

	SIP Entity Name	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
○	CM_SIP_Endpoints	10.10.9.12	5061	TLS	FALSE	UP	200 OK	UP
○	ASBCE	10.10.9.81	5060	TCP	FALSE	UP	200 OK	UP
○	CM Trunk	10.10.9.12	5060	TCP	FALSE	UP	200 OK	UP
○	Messaging	10.10.2.82	5060	TCP	FALSE	UP	200 OK	UP

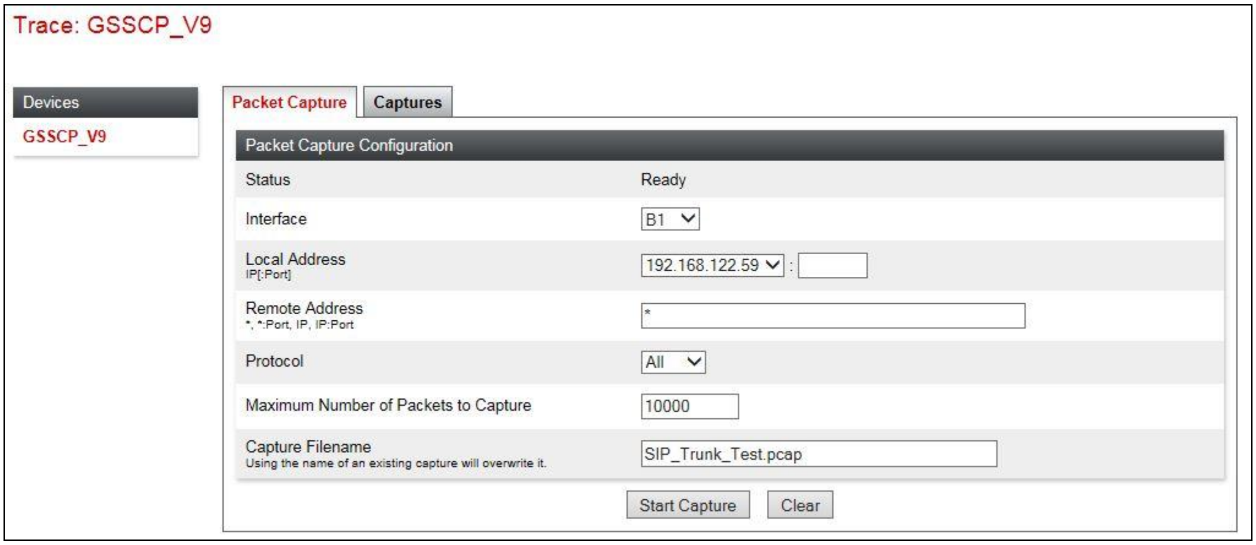
2. From Communication Manager SAT interface run the command **status trunk n** where **n** is the previously configured SIP trunk. Observe if all channels on the trunk group display **in-service/idle**.

```
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 a trace on the Avaya SBCE, navigate to **Device Specific Settings → Advanced Options → Troubleshooting → Trace** in the main menu on the left hand side and select the **Packet Capture** tab.

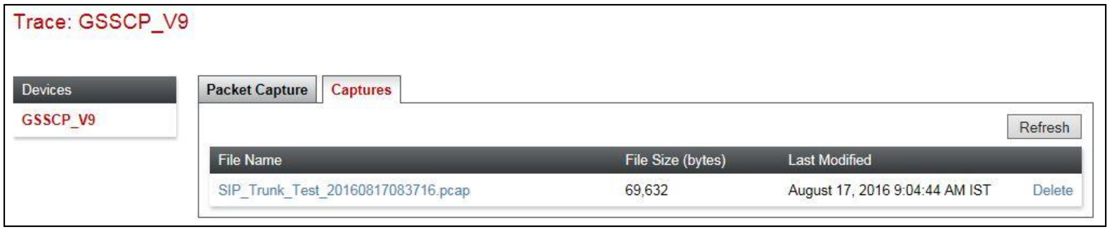
- Select the SIP Trunk interface from the **Interface** drop down menu.
- Select the signalling interface IP address or **All** from the **Local Address** drop down menu.
- Enter the IP address of the network SBC in the **Remote Address** field or enter a ***** to capture all traffic.
- Specify the **Maximum 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**.



Packet Capture Configuration

Status	Ready
Interface	B1
Local Address IP[:Port]	192.168.122.59 :
Remote Address *, *:Port, IP, IP:Port	*
Protocol	All
Maximum Number of Packets to Capture	10000
Capture Filename Using the name of an existing capture will overwrite it.	SIP_Trunk_Test.pcap

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



File Name	File Size (bytes)	Last Modified	
SIP_Trunk_Test_20160817083716.pcap	69,632	August 17, 2016 9:04:44 AM IST	Delete

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 Telenet network.

10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Communication Manager R7.0.1, Avaya Aura® Session Manager R7.0.1 and Avaya Session Border Controller for Enterprise R7.1 to Telenet SIP Trunking. The Telenet 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. 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*, Aug 2016.

[3] *Deploying Avaya Aura® applications from System Manager*, Aug 2016

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

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

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

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

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

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

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

[13] *Deploying Avaya Session Border Controller for Enterprise*, Release 7.1, Jun 2016

[14] *Upgrading Avaya Session Border Controller for Enterprise*, Release 7.1, Aug 2016

[15] *Administering Avaya Session Border Controller for Enterprise*, Release 7.1, Jun 2016

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

12. Appendix A

This appendix contains vector examples for the workarounds used for the duplicate INVITE issue described in **Section 2.2**. The issue arose when the duplicate INVITE was received after the call was answered. This only happened in two cases during testing: Calls to the voicemail system to retrieve messages; Calls from an EC500 mobile to the idle appearance FNE for making outbound calls. In both cases, vectors were used to delay the call answer as described below:

Voicemail Retrieval:

```
change vector 5                                                 Page   1 of   6
                                  CALL VECTOR

    Number: 5               Name: Messaging
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n          Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    2   secs hearing ringback
02 route-to     number 7000             with cov n if unconditionally
03 stop
```

The vector lines work as follows:

1. Wait for two seconds hearing ringback.
2. Route to voicemail system (extension number 7000).

A Vector Directory Number (VDN) is assigned to invoke the vector, and the DDI assigned to voicemail retrieval is pointed towards the VDN in the incoming trunk settings.

Idle Appearance FNE:

```
display vector 1                                                Page   1 of   6
                                  CALL VECTOR

    Number: 1               Name: Idle_appearance
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n          Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    2   secs hearing ringback
02 collect      16   digits after announcement 6902     for none
03 set          digits = digits CATL  3
04 route-to     digits with coverage n
05 stop
```

The vector lines work as follows:

1. Wait for two seconds hearing ringback.
2. Play an announcement, during testing “Please dial the number you require” was used. Collect up to 16 digits (works for less with an inter-digit timeout that can be cancelled with “#”).

3. Prefix the dialled number with the extension number assigned for the idle appearance FNE. A single digit extension was used as the total maximum length of 16 digits could be restrictive for international numbers.
4. Route to the complete number. This routes to the extension number assigned to the FNE with the digits dialled by the user.

The following shows the entry for the FNE:

```
display off-pbx-telephone feature-name-extensions set 1         Page   2 of   2
     EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME

       Exclusion (Toggle On/Off):
      Extended Group Call Pickup:
          Held Appearance Select:
          Idle Appearance Select: 3
              Last Number Dialed:
```

Note: This changes the customer experience of this function as follows:

1. Customer rings the number for the FNE.
2. The call is answered and an announcement is played – this happens even if the caller is not an EC500 mobile.
3. After the announcement there is silence while the vector is collecting digits – dial tone is not played.
4. The caller enters the number.
5. If the caller is a registered EC500 mobile, the call is routed.
6. If the caller is not an EC500 mobile, the call is cleared without an announcement.
7. A maximum number of 15 digits can be dialled including ARS access code and national / international dialling prefixes. If an extension number longer than a single digit is used for the FNE, this number is further reduced.

A Vector Directory Number (VDN) is assigned to invoke the vector, and the DDI assigned to the idle appearance FNE is pointed towards the VDN in the incoming trunk settings.

Another workaround was successfully tested for the FNE that prevented duplicate INVITE messages from being sent from the Avaya SBCE to the Session Manager. This was a setting in the Avaya SBCE for “Re-Invite Handling”. Unfortunately this caused other test cases to fail as described in **Section 2.2**.

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