Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Communication Manager R6.3 as an Evolution Server, Avaya Aura® Session Manager R6.3 and Avaya Session Border Controller for Enterprise to support KPN VaMo1 VoIP Connect Service - Issue 1.0

Abstract

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between KPN VaMo1 (fixed mobile) VoIP Connect 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. KPN 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 KPN VaMo1 VoIP Connect Service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Session Border Controller for Enterprise (Avaya SBCE), Avaya Aura® Session Manager and Avaya Aura® Communication Manager Evolution Server. Customers using this Avaya SIP-enabled enterprise solution with KPN VaMo1 VoIP Connect are able to place and receive PSTN calls via a dedicated Internet connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks. This approach generally results in lower cost for the enterprise customer.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using an Avaya SIP telephony solution consisting of Communication Manager, Session Manager and Avaya SBCE. The enterprise site was configured to use the SIP Trunking service provided by KPN.

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

2.1. Interoperability Compliance Testing

The interoperability test included the following:

- Incoming calls to the enterprise site from PSTN phones using the SIP Trunk provided by KPN, calls made to SIP and H.323 telephones at the enterprise
- Outgoing calls from the enterprise site completed via KPN VoIP Connect to PSTN destinations, calls made from SIP and H.323 telephones
- Calls using the G.711A codec
- Fax calls to/from a group 3 fax machine to a PSTN connected fax machine using T.38
- DTMF transmission using RFC 2833 with successful Voice Mail/Vector navigation for inbound and outbound calls
- User features such as hold and resume, transfer, conference, call forwarding, etc
- Caller ID Presentation and Caller ID Restriction
- Direct IP-to-IP media (also known as “shuffling”) with SIP and H.323 telephones
- Call coverage and call forwarding for endpoints at the enterprise site
- Transmission and response of SIP OPTIONS messages sent by KPN VoIP Connect requiring Avaya response and sent by Avaya requiring KPN response

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for KPN VaMo1 VoIP Connect Service with the following observations:

- When there were no matching codecs between the SDP offer and answer on incoming calls, the Communication Manager sent a “488 Not Acceptable Here”. There was a delay on the calling PSTN phone before a tone was heard.
- When there were no matching codecs between the SDP offer and answer on outgoing calls, the network sent a “500 Server Internal Error”. A tone was heard on the calling Communication Manager extension, but the more informative response is “488 Not Acceptable Here”.
- Access to the Enterprise from a Toll-Free number was not tested as none was available for test.
- Access to Emergency Services was not tested as no test call had been booked with the Emergency Services Operator.
- Hold from the PSTN was included in the Test Plan though in this case it was not a valid test of the SIP Trunk. Although hold worked, there were no messages sent to the enterprise to indicate that the call had been placed on hold.
- When forwarding a call to the PSTN, a short delay in the establishment of the media of approximately two seconds was observed on the calling PSTN phone. This issue was also observed on calls transferred to the PSTN. This is not considered critical to certification as the delay is short and likely to be due to the configuration of the test environment.
- The transmission of outbound fax to an international destination, in this case the Avaya Lab in Galway, failed after two pages. The fax calls were made successfully and reliably to a fax machine on the KPN premises.
- Call failures were observed on calls to and from the one-X Communicator softphone when it was connected to the enterprise using SIP. These included failure to conference an H.323 extension when in “Computer” mode and failure of call transfers and conferencing when in “Other Phone” mode. All tests were successful when one-X Communicator was connected using H.323. This is considered to be an issue with one-X Communicator rather than the SIP trunk and so is not critical for certification.

2.3. Support

For technical support on KPN products please visit the website at www.kpn.com or contact an authorized KPN representative.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an Enterprise site connected to KPN VaMo1 VoIP Connect. Located at the Enterprise site is an Avaya Session Border Controller for Enterprise, Session Manager and Communication Manager. Endpoints are Avaya 96x0 series and Avaya 96x1 series IP telephones (with SIP and H.323 firmware), Avaya 46xx series IP telephones (with H.323 firmware), Avaya 16xx series IP telephones (with H.323 firmware), Avaya A175 Desktop Video Device running Flare Experience (audio only), Avaya analogue telephones and an analogue fax machine. Also included in the test configuration was an Avaya one-X® Communicator soft phone and Flare for Windows running on a laptop PC.

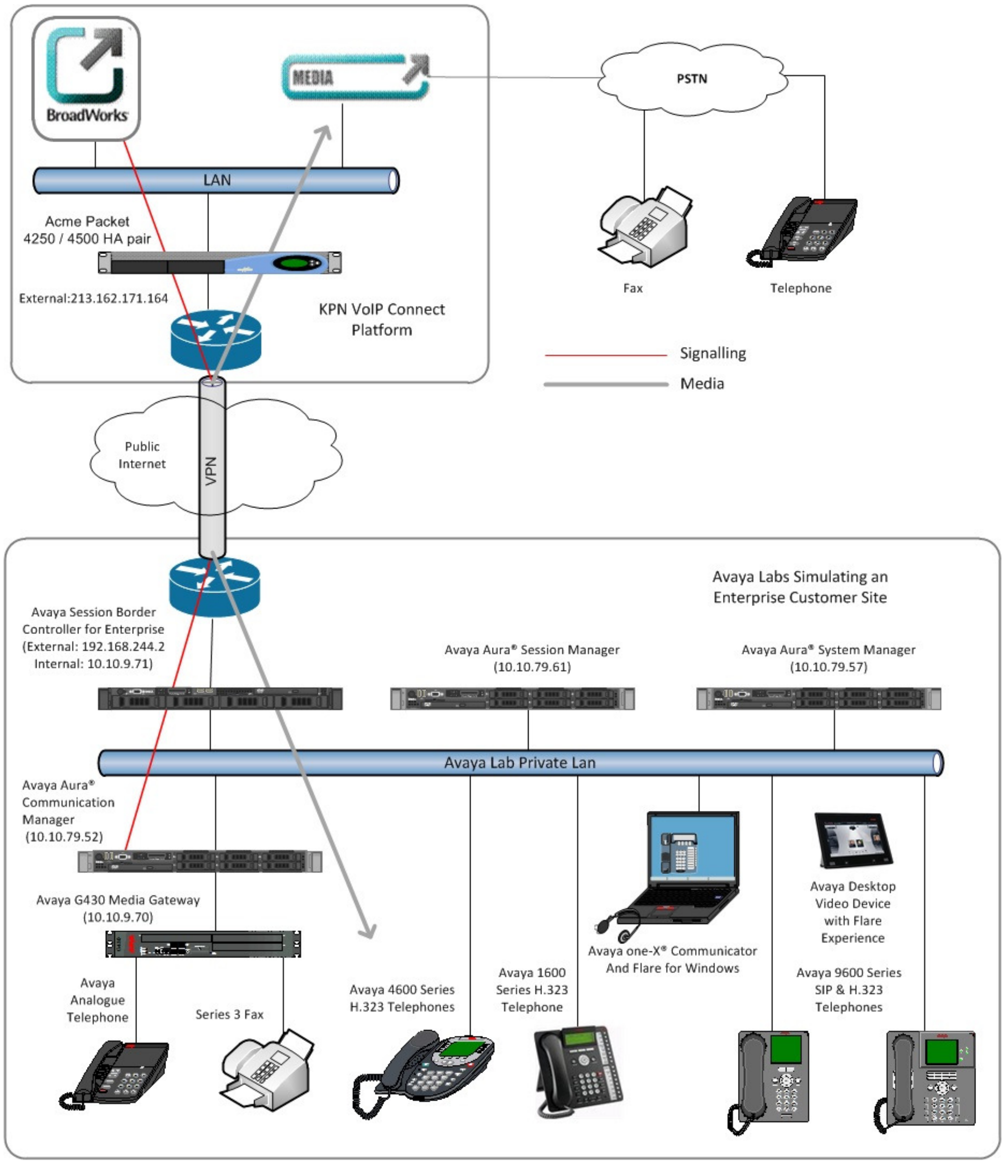


Figure 1: Test Setup KPN VaMo1 VoIP Connect to Avaya Enterprise

4. Equipment and Software Validated

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

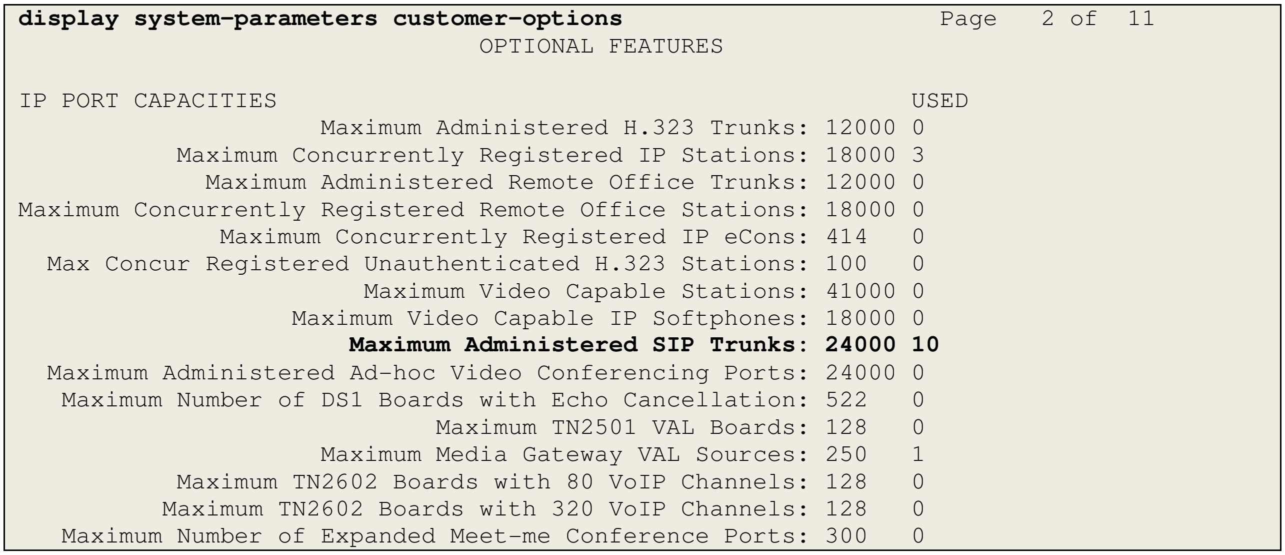
Equipment/Software	Release/Version
Avaya	
Dell PowerEdge R620 running Session Manager on VM Version 8	SM-6.3.2.0.632023-e50-00
Dell PowerEdge R620 running System Manager on VM Version 8	SMGR-6.3.0.8.5682-e50-64 (Build 5682)
Dell PowerEdge R620 running Communication Manager on VM Version 8	R016x.03.0.124.0
Avaya Session Border Controller Advanced for Enterprise Server	6.2.0.Q48
Avaya 1616 Phone (H.323)	1.302
Avaya 4621 Phone (H.323)	2.902
Avaya 96x0 Phone (H.323)	3.200
Avaya A175 Desktop Video Device (SIP)	Flare Experience Release 1.1.2
Avaya 9630 Phone (SIP)	R2.6 SP9
Avaya 9608 Phone (SIP)	R6.2 SP1
Avaya one–X® Communicator (H.323) on Lenovo T510 Laptop PC	6.1.8.06-SP8-40314
Analogue Handset	NA
Analogue Fax	NA
KPN	
as1-sbc-s-2-1 ACME Net-Net 4500	SCX6.2.0 MR-6 Patch 2 (Build 876)
as1-sbc-s-1-1 ACME Net-Net 4250	SC6.2.0 MR-6 Patch 2 (Build 876
Alcatel-Lucent-HPSS	v3.0.3
Broadsoft	v 17

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 KPN VaMo1 VoIP Connect. For incoming calls, the Session Manager receives SIP messages from the Avaya SBC for Enterprise (Avaya SBCE) and directs the incoming SIP messages to Communication Manager. Once the message arrives at Communication Manager, further incoming call treatment, such as incoming digit translations and class of service restrictions may be performed. All outgoing calls to the PSTN are processed within Communication Manager and may be first subject to outbound features such as automatic route selection, digit manipulation and class of service restrictions. Once Communication Manager selects a SIP trunk, the SIP signalling is routed to the Session Manager. The Session Manager directs the outbound SIP messages to the Avaya SBCE at the enterprise site that then sends the SIP messages to the KPN VoIP Connect 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 KPN VoIP Connect network, and any other SIP trunks used.



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

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

On **Page 4**, verify that **IP Trunks** field is set to **y.**

```
display system-parameters customer-options                      Page   4 of  11
                                OPTIONAL FEATURES

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

           IP Attendant Consoles? y
```

5.2. Administer IP Node Names

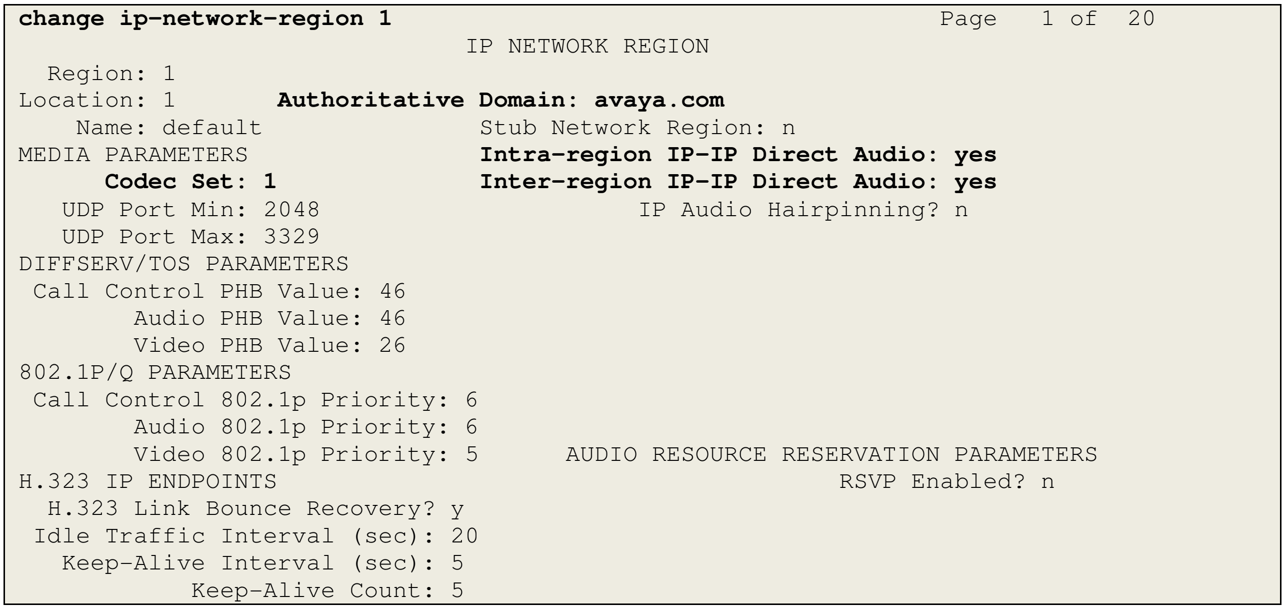
The node names defined here will be used in other configuration screens to define a SIP signalling group between Communication Manager and Session Manager. In the **IP Node Names** form, assign the node **Name** and **IP Address** for the Session Manager. In this case, **SM-SMVM1** and **10.10.79.61** are the **Name** and **IP Address** for the Session Manager SIP interface. Also note the **procr** name as this is the processor interface that Communication Manager will use as the SIP signalling interface to Session Manager.

```
display node-names ip
                                  IP NODE NAMES
    Name              IP Address
SMVM1               10.10.79.61
default             0.0.0.0
procr               10.10.79.52
procr6              ::
```

5.3. Administer IP Network Region

Use the **change ip-network-region 1** command to set the following values:

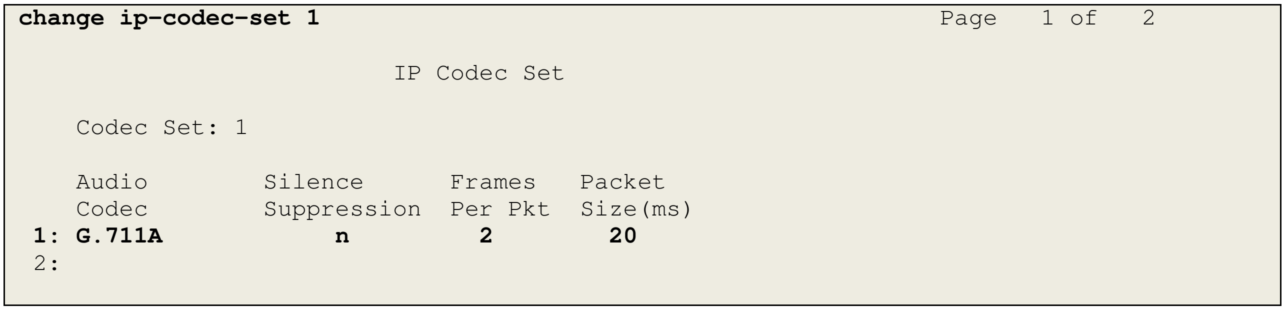
- The **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is **avaya.com**.
- By default, **IP-IP Direct Audio** (both **Intra**- and **Inter-Region**) is enabled (**yes**) to allow audio traffic to be sent directly between endpoints without using gateway VoIP resources. When a PSTN call is shuffled, the media stream is established directly between the enterprise end-point and the internal media interface of the Avaya SBCE.
- The **Codec Set** is set to the number of the IP codec set to be used for calls within the IP network region. In this case, codec set **1** is used.
- The rest of the fields can be left at default values.



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

5.4. Administer IP Codec Set

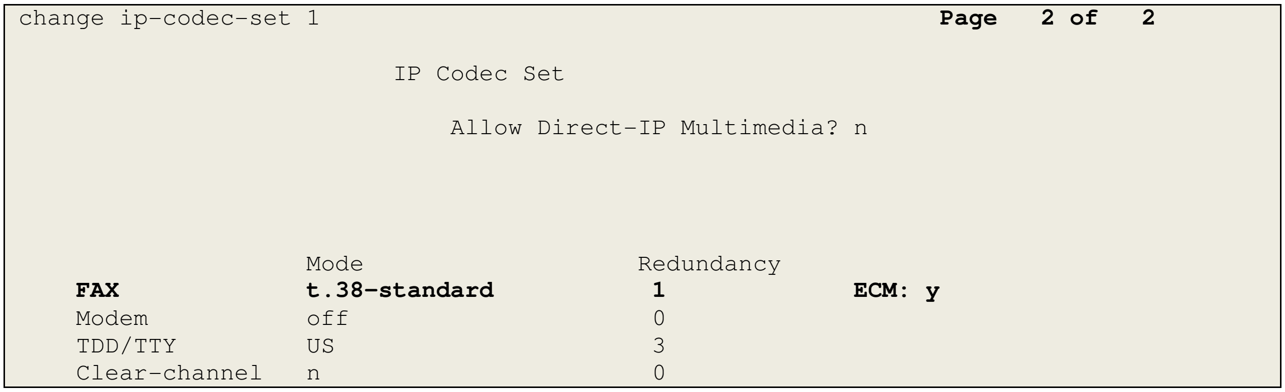
Open the **IP Codec Set** form for the codec set specified in the IP Network Region form in **Section 5.3.** Enter the list of audio codec's eligible to be used in order of preference. For the interoperability test the codec supported by KPN VoIP Connect was configured, namely **G.711A**.



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

                          IP Codec Set

    Codec Set: 1

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

KPN VaMo1 VoIP Connect supports T.38 for transmission of fax. Navigate to **Page 2** and define T.38 fax as follows:

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



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

                          IP Codec Set

                              Allow Direct-IP Multimedia? n




                Mode                   Redundancy
    FAX         t.38-standard           1              ECM: y
    Modem       off                     0
    TDD/TTY     US                      3
    Clear-channel   n                   0
```

Note: When investigating the fax failures to international destinations, **Redundancy** was set to **1**. This results in T.38 data being sent twice to ensure successful delivery. This made no difference to the fax failures, and can be left at the default value of 0.

5.5. Administer SIP Signaling Groups

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

- Set **Group Type** to **sip**
- Set **Transport Method** to **tcp**
- Set **Peer Detection Enabled** to **y** allowing the Communication Manager to automatically detect if the peer server is a Session Manager
- Set **Near-end Node Name** to the processor interface (node name **procr** as defined in the **IP Node Names** form shown in **Section 5.2**)
- Set **Far-end Node Name** to the Session Manager (node name **SMVM1** as defined in the **IP Node Names** form shown in **Section 5.2**)
- Set **Near-end Listen Port** and **Far-end Listen Port** to **5060** (Commonly used TCP port value)
- Set **Far-end Network Region** to the IP Network Region configured in **Section 5.3**. (logically establishes the far-end for calls using this signalling group as network region **1**)
- Leave **Far-end Domain** blank (allows the CM to accept calls from any SIP domain on the associated trunk)
- Set **Direct IP-IP Audio Connections** to **y**
- Leave **DTMF over IP** at default value of **rtp-payload** (Enables **RFC2833** for DTMF transmission from the Communication Manager)

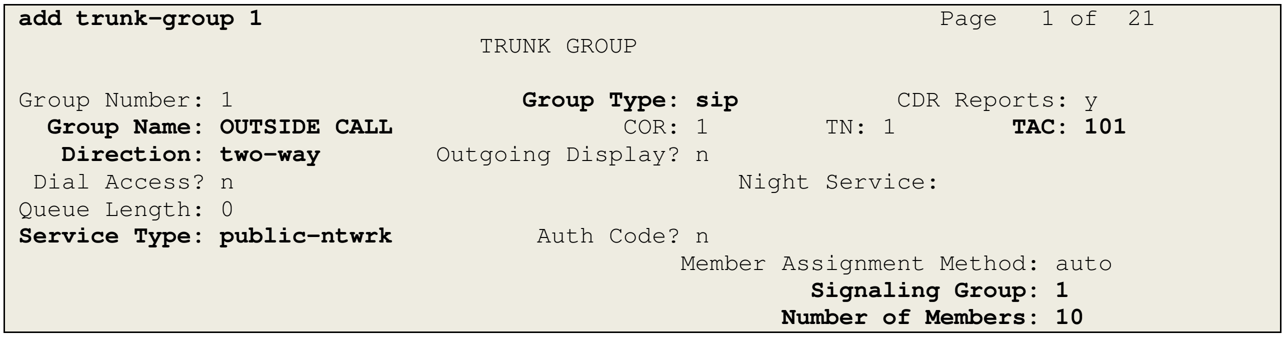
The default values for the other fields may be used.

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

 Group Number: 1             Group Type: sip
  IMS Enabled? n       Transport Method: 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

   Near-end Node Name: procr               Far-end Node Name: SMVM1
 Near-end Listen Port: 5060              Far-end Listen Port: 5060
                                      Far-end Network Region: 1

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

5.6. Administer SIP Trunk Group

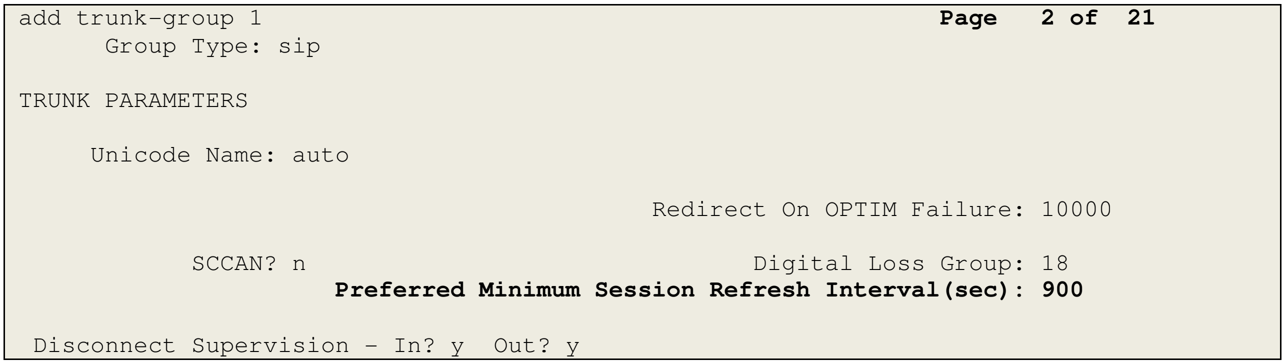
A trunk group is associated with the signaling group described in **Section 5.5**. Configure the trunk group using the **add trunk-group x** command, where **x** is an available trunk group. On **Page 1** of this form:

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



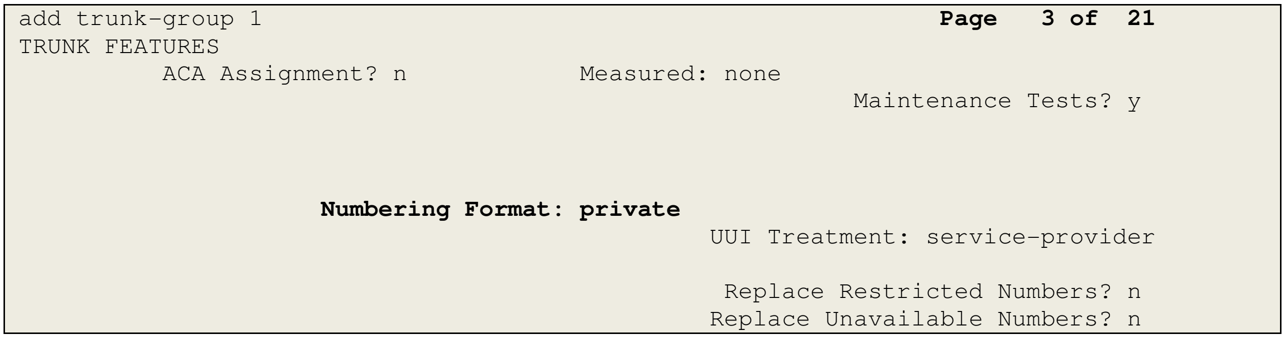
```
add trunk-group 1                                               Page   1 of  21
                                TRUNK GROUP

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

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



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

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 10000

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

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

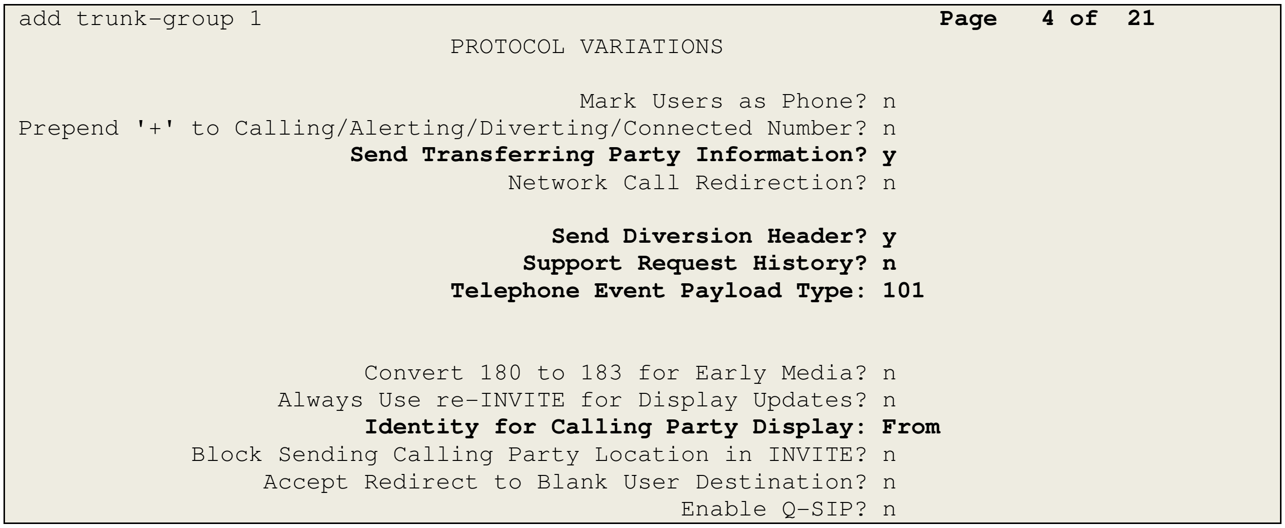
On **Page 3**, set the **Numbering Format** field to **private**. This allows delivery of CLI in formats other than E.164 with leading "+". In test, CLI was sent as the unaltered extension number and was modified using an adaptation on the Session Manager.



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



                     Numbering Format: private
                                                UUI Treatment: service-provider

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

On **Page 4** of this form:

- Set **Support Request History** to **n** as the required information for forwarded and transferred calls will be sent in the **Diversion Header** and **Transferring Party Information**
- Set **Send Transferring Party Information** to **y**
- Set **Send Diversion Header** to **y**
- Set the **Telephone Event Payload Type** to **101** to match the value preferred by KPN VoIP Connect (this Payload Type is not applied to calls from SIP end-points)
- Set the **Identity for Calling Party Display** to **From** to ensure that where CLI for incoming calls is withheld, it is not displayed on the Communication Manager extension



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

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

                                     Send Diversion Header? y
                                   Support Request History? n
                              Telephone Event Payload Type: 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-unknown-numbering** command to configure Communication Manager to send the calling party number in the format required. In test, calling party number was sent as the unaltered extension number and was modified using an adaptation on the Session Manager. This calling party number is sent in the SIP From, Contact and PAI headers, and displayed on display-equipped PSTN telephones.

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

Ext Ext             Trk         Private          Total
Len Code            Grp(s)      Prefix           Len
 4  2               1                            4     Total Administered: 1
                                                          Maximum Entries: 540
```

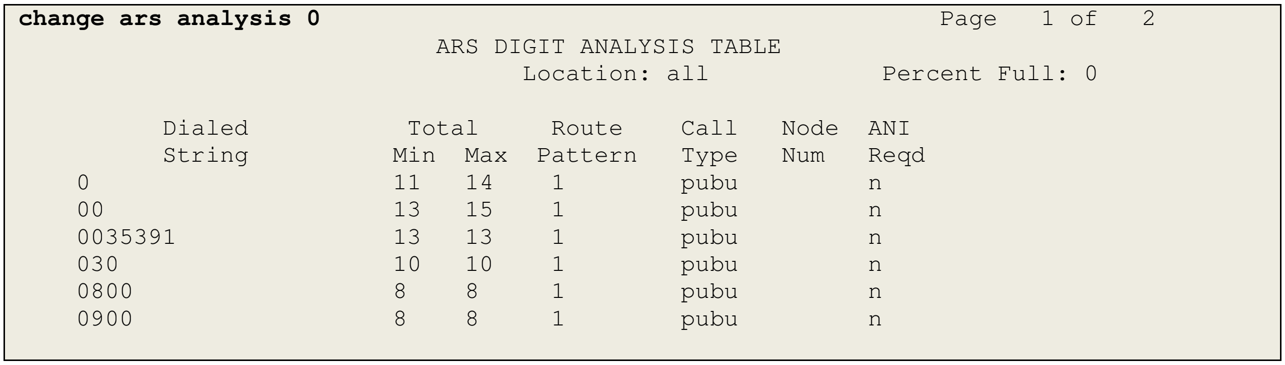
Note: The above configuration accepts all **4** digit numbers starting with **2**, which includes all SIP and H.323 extension numbers, and passes them on with no prefix.

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 KPN VaMo1 VoIP Connect. The single digit **9** was used as the ARS access code providing a facility for telephone users to dial 9 to reach an outside line. Use the **change feature-access-codes** command to configure a digit as the **Auto Route Selection (ARS) - Access Code 1**.

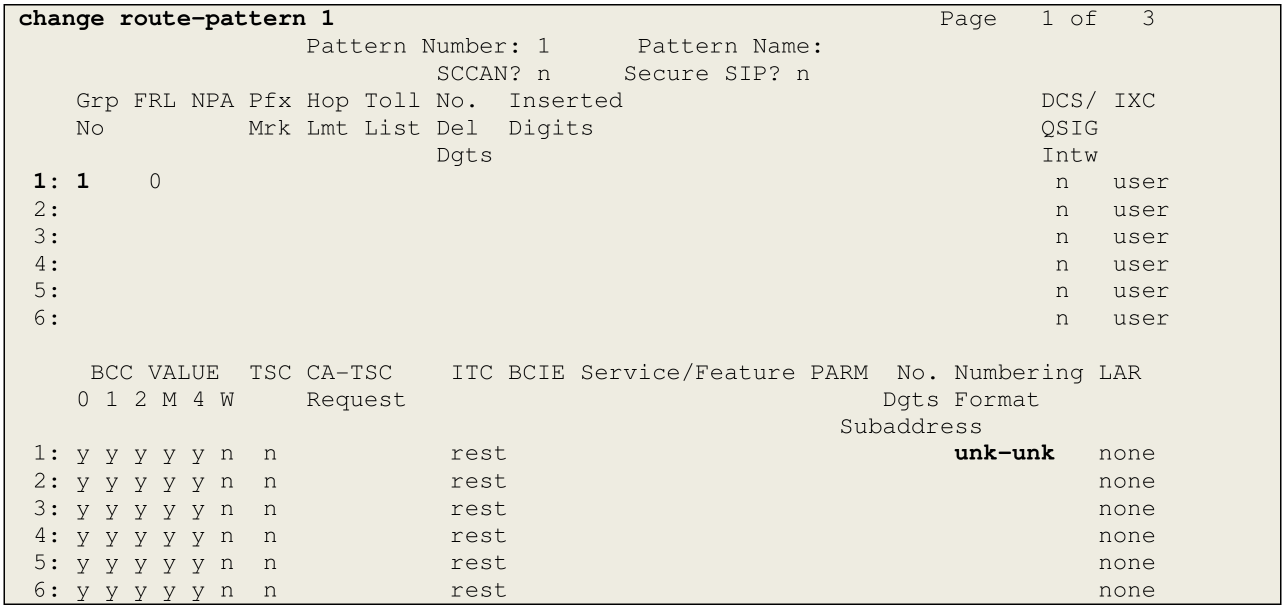
```
change feature-access-codes                                         Page   1 of  10
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated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

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



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

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    0                     11   14     1       pubu         n
    00                    13   15     1       pubu         n
    0035391               13   13     1       pubu         n
    030                   10   10     1       pubu         n
    0800                  8    8      1       pubu         n
    0900                  8    8      1       pubu         n
```

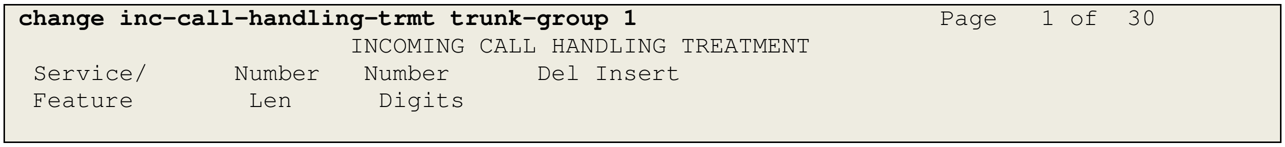
Use the **change route-pattern x** command, where **x** is an available route pattern, to add the SIP trunk group to the route pattern that ARS selects. In this configuration, route pattern **1** is used to route calls to trunk group **1**. **Numbering Format** is applied to CLI and is used to set TDM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



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

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

5.9. Administer Incoming Digit Translation

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



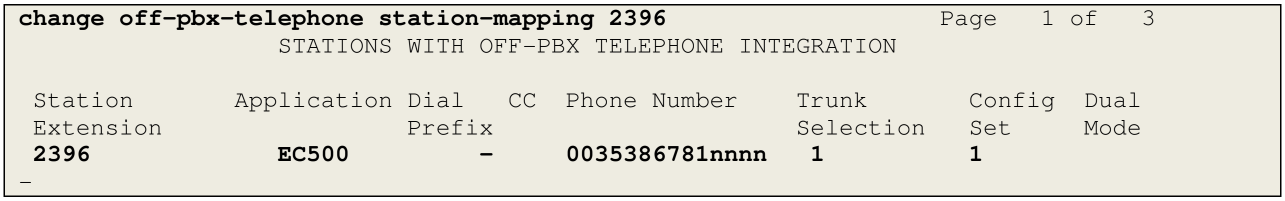
```
change inc-call-handling-trmt trunk-group 1                     Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits

```

Note: One reason for configuring the enterprise in this way is that it was found when using national number format for CLI delivery, the message waiting indicator was not successfully sent to SIP extensions when a voice mail message was available and unread.

5.10. EC500 Configuration

When EC500 is enabled on the Communication Manager station, a call to that station will generate a new outbound call from Communication Manager to the configured EC500 destination, typically a mobile phone. The following screen shows an example EC500 configuration for the user with station extension 2396. Use the command **change off-pbx-telephone station-mapping x** where **x** is the Communication Manager station.

- The **Station Extension** field will automatically populate with station extension
- For **Application** enter **EC500**
- Enter a **Dial Prefix** (e.g., 9) if required by the routing configuration
- For the **Phone Number** enter the phone that will also be called (e.g. **0035386781nnnn**)
- Set the **Trunk Selection** to **1** so that Trunk Group 1 will be used for routing
- Set the **Config Set** to **1**



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

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 2396             EC500         -     0035386781nnnn   1          1
_
```

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

Save Communication Manager changes by entering **save translation** to make them permanent.

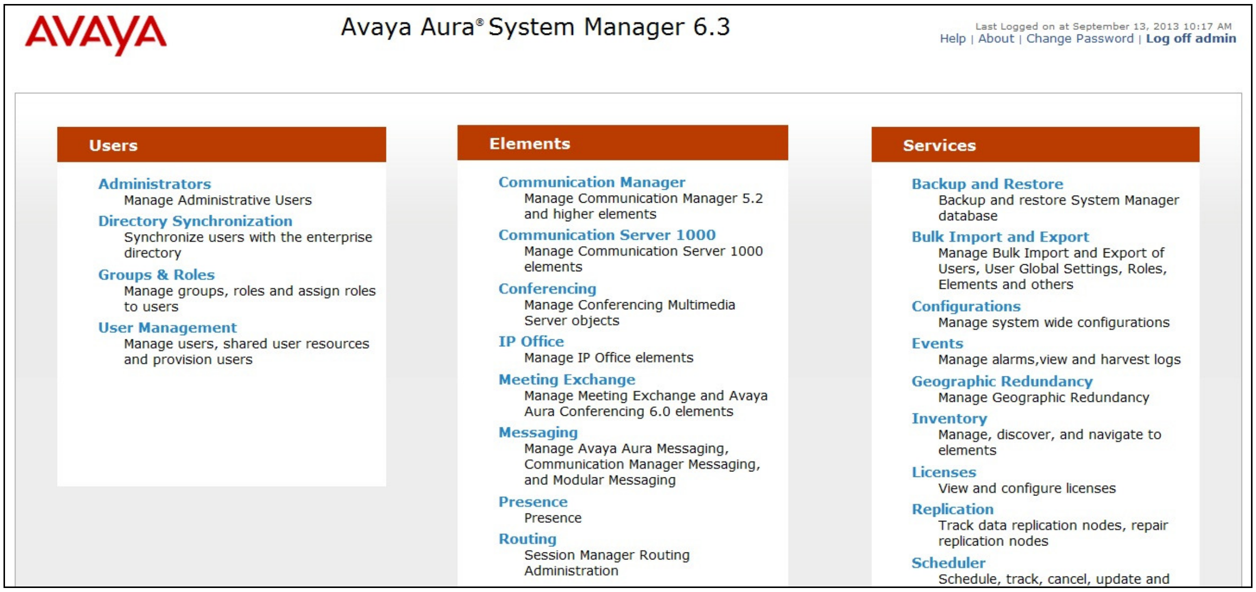
6. Configuring Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The Session Manager is configured via the System Manager. The procedures include the following areas:

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

6.1. Log in to Avaya Aura® System Manager

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



AVAYA Avaya Aura® System Manager 6.3

Last Logged on at September 13, 2013 10:17 AM
Help | About | Change Password | Log off admin

Users

Administrators
Manage Administrative Users
Directory Synchronization
Synchronize users with the enterprise directory
Groups & Roles
Manage groups, roles and assign roles to users
User Management
Manage users, shared user resources and provision users

Elements

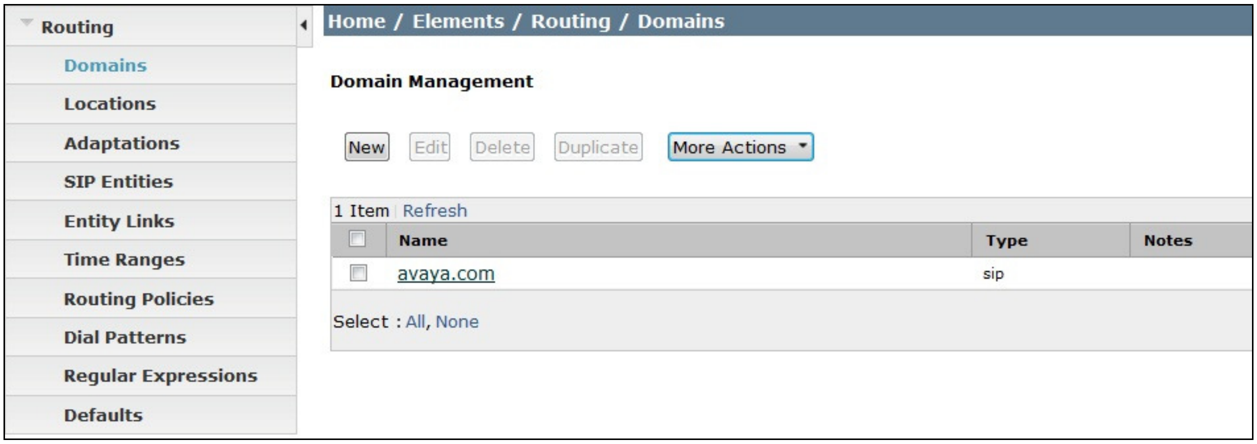
Communication Manager
Manage Communication Manager 5.2 and higher elements
Communication Server 1000
Manage Communication Server 1000 elements
Conferencing
Manage Conferencing Multimedia Server objects
IP Office
Manage IP Office elements
Meeting Exchange
Manage Meeting Exchange and Avaya Aura Conferencing 6.0 elements
Messaging
Manage Avaya Aura Messaging, Communication Manager Messaging, and Modular Messaging
Presence
Presence
Routing
Session Manager Routing Administration

Services

Backup and Restore
Backup and restore System Manager database
Bulk Import and Export
Manage Bulk Import and Export of Users, User Global Settings, Roles, Elements and others
Configurations
Manage system wide configurations
Events
Manage alarms,view and harvest logs
Geographic Redundancy
Manage Geographic Redundancy
Inventory
Manage, discover, and navigate to elements
Licenses
View and configure licenses
Replication
Track data replication nodes, repair replication nodes
Scheduler
Schedule, track, cancel, update and

6.2. Administer SIP Domain

To add the SIP domain that will be used with Session Manager, select **Routing** from the **Home** tab menu and in the resulting tab select **Domains** from left hand menu. Click the **New** button to create a new SIP domain entry. In the **Name** field enter the domain name agreed with KPN; this will be the same as specified in the Authoritative Domain specified in the IP Network Region on the 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 (not shown).

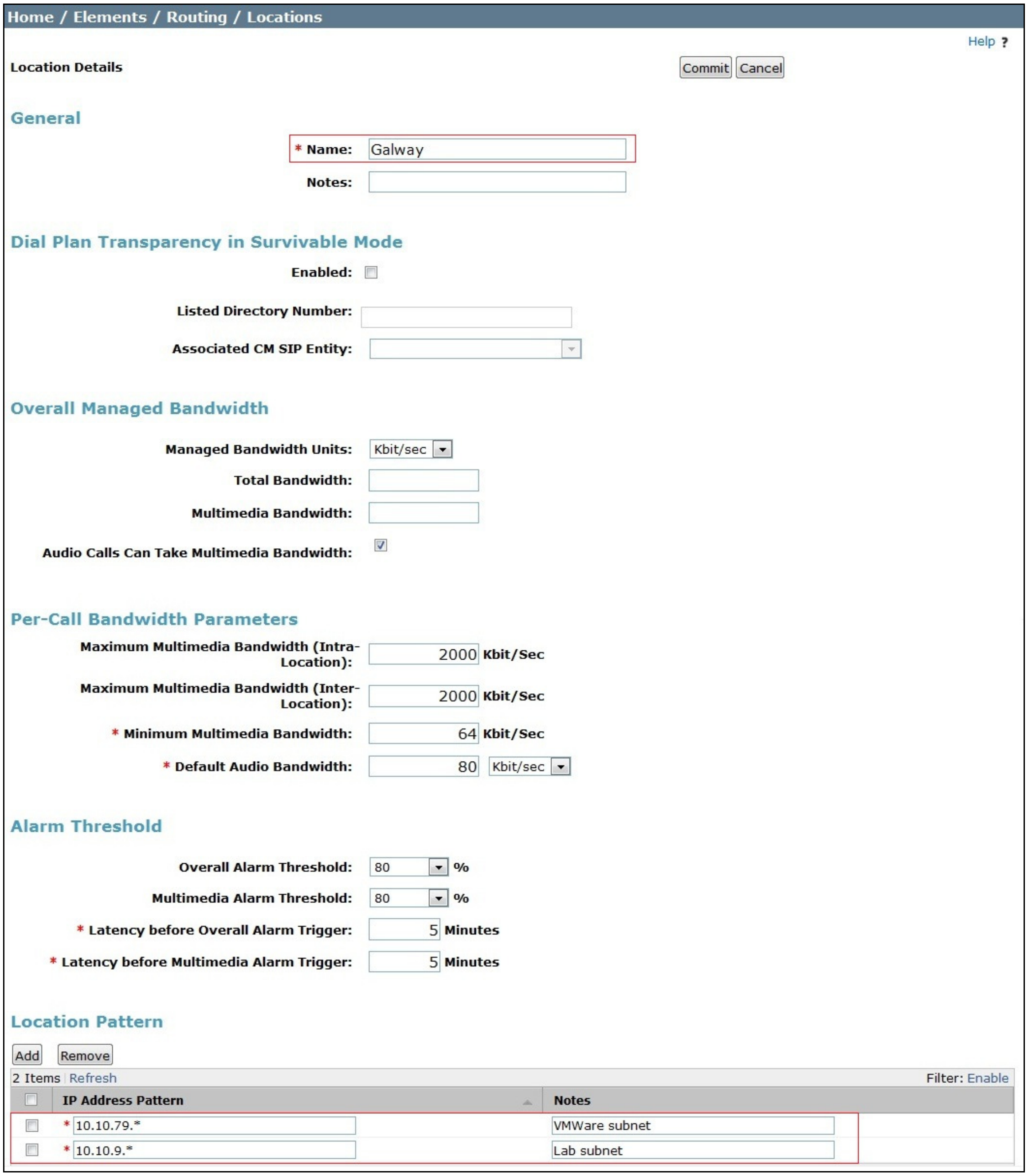


Domain Management

☐	Name	Type	Notes
☐	avaya.com	sip	

6.3. Administer Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for the purposes of bandwidth management. One location is added to the sample configuration for all of the enterprise SIP entities. On the **Routing** tab select **Locations** from the left hand menu. Under **General**, in the **Name** field, enter an informative name for the location. Scroll to the bottom of the page and under **Location Pattern**, click **Add**, then enter an **IP Address Pattern** in the resulting new row, * is used to specify any number of allowed characters at the end of the string. Below is the location configuration used for the test enterprise.



Location Details

General

* Name:	Galway
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

Location Pattern

□	IP Address Pattern	Notes
□	* 10.10.79.*	VMWare subnet
□	* 10.10.9.*	Lab subnet

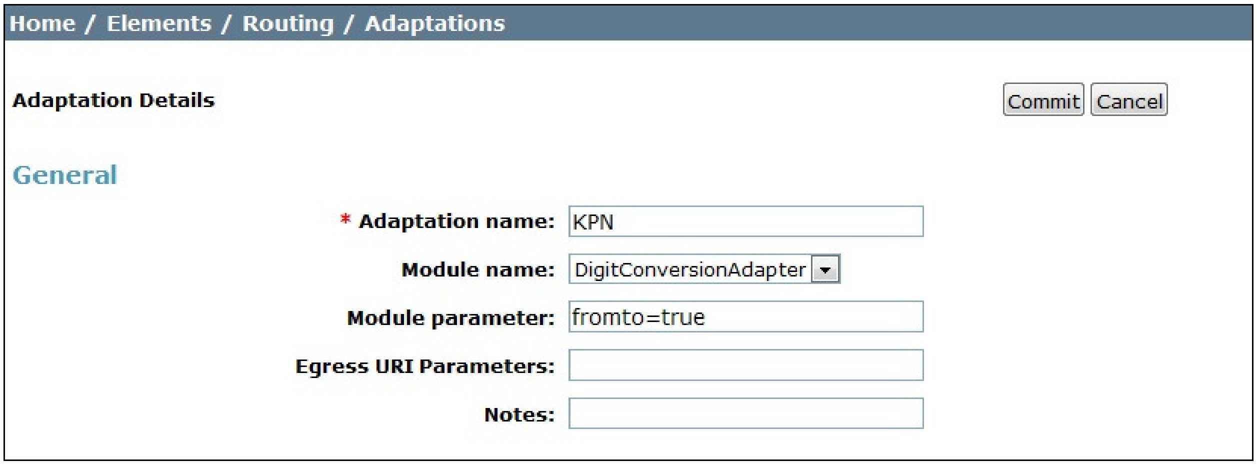
6.4. Administer Adaptations

Calls from KPN VoIP Connect are received at the enterprise in national format with a leading "0" on the Request URI. An Adaptation specific to KPN is used to convert the called number to an extension number as defined in the Communication Manager before onward routing to the Communication Manager SIP Entity and removes the requirement for incoming digit manipulation on the Communication Manager.

The adaptation is also used to convert the calling party number from an extension number to national format with leading "0".

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

- In the **Adaptation Name** field, enter a descriptive title for the adaptation.
- In the **Module Name** enter **DigitConversionAdapter**. This is used for simple digit conversion adaptations.
- In the **Module Parameter** field, enter **fromto=true**. This will apply the adaptation to the From and To headers as well as the Request URI.



* Adaptation name: KPN

Module name: DigitConversionAdapter

Module parameter: fromto=true

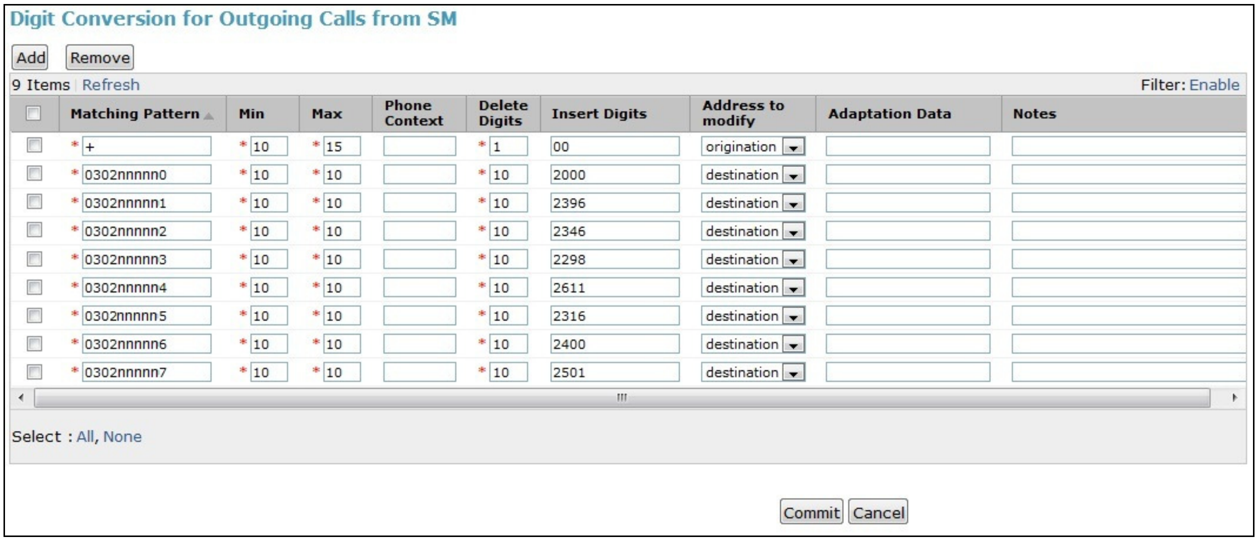
Egress URI Parameters:

Notes:

Scroll down and in the section **Digit Conversion for Outgoing Calls from SM**, click on Add. An additional row will appear. This allows information to be entered for the manipulation of numbers going from the Session Manager to the Communication Manager. This is where the incoming number is translated from national format to the extension number as defined in the Communication Manager.

The screenshot below shows a translation for each DDI number. This is not normally necessary where the extension number forms part of the national number. When this is the case, only the digits up to the extension number need to be analysed and removed.

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



Digit Conversion for Outgoing Calls from SM

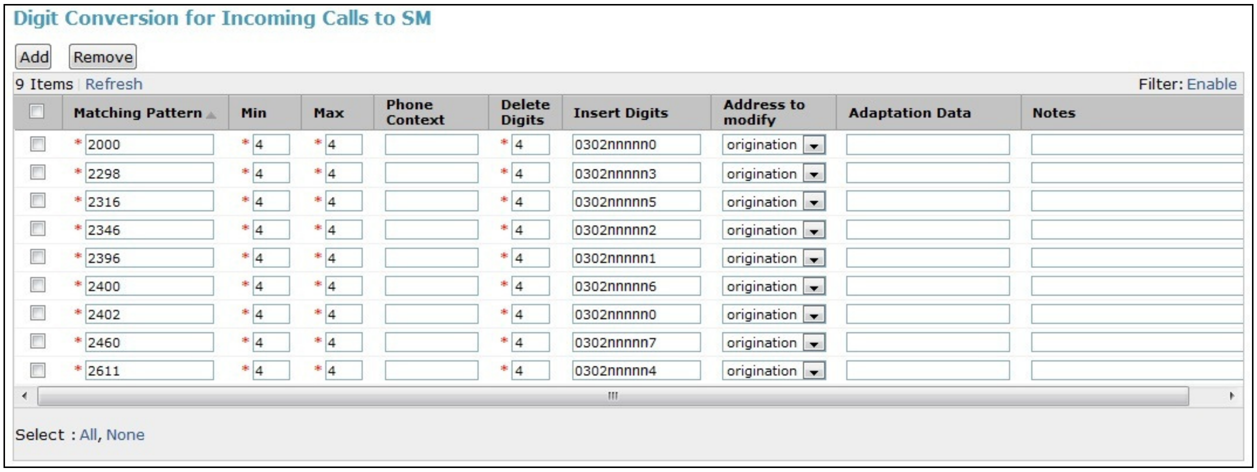
□	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
□	* +	* 10	* 15		* 1	00	origination		
□	* 0302nnnnn0	* 10	* 10		* 10	2000	destination		
□	* 0302nnnnn1	* 10	* 10		* 10	2396	destination		
□	* 0302nnnnn2	* 10	* 10		* 10	2346	destination		
□	* 0302nnnnn3	* 10	* 10		* 10	2298	destination		
□	* 0302nnnnn4	* 10	* 10		* 10	2611	destination		
□	* 0302nnnnn5	* 10	* 10		* 10	2316	destination		
□	* 0302nnnnn6	* 10	* 10		* 10	2400	destination		
□	* 0302nnnnn7	* 10	* 10		* 10	2501	destination		

Note: Included in the above configuration is a conversion to remove the leading "+" from the P-Asserted-ID and To headers and insert the international dialling prefix of 00. This was done during test so that the P-Asserted-ID of incoming calls from the EC500 mobile phone would match the entry in the **off-pbx-telephone station-mapping** table defined in **Section 5.10**

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

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

- Under **Matching Pattern** enter the minimum number of digits to identify the extension number as required. During test, this was the full extension number as each extension was translated to a national number. If the extension number forms part of the DDI number, a single digit will suffice.
- Under **Min** and **Max** enter the Minimum and Maximum digits of the extension number.
- Under **Delete Digits** enter the number of digits to delete if the full extension number does not form part of the national number, during test all had to be deleted.
- Under **Insert Digits** enter the prefix required to make up the full national number. During test, this was the entire number
- Under **Address to Modify** choose **origination** from the drop down box to apply this rule to the From and P-Asserted-ID headers only.



Digit Conversion for Incoming Calls to SM

9 Items Refresh Filter: Enable

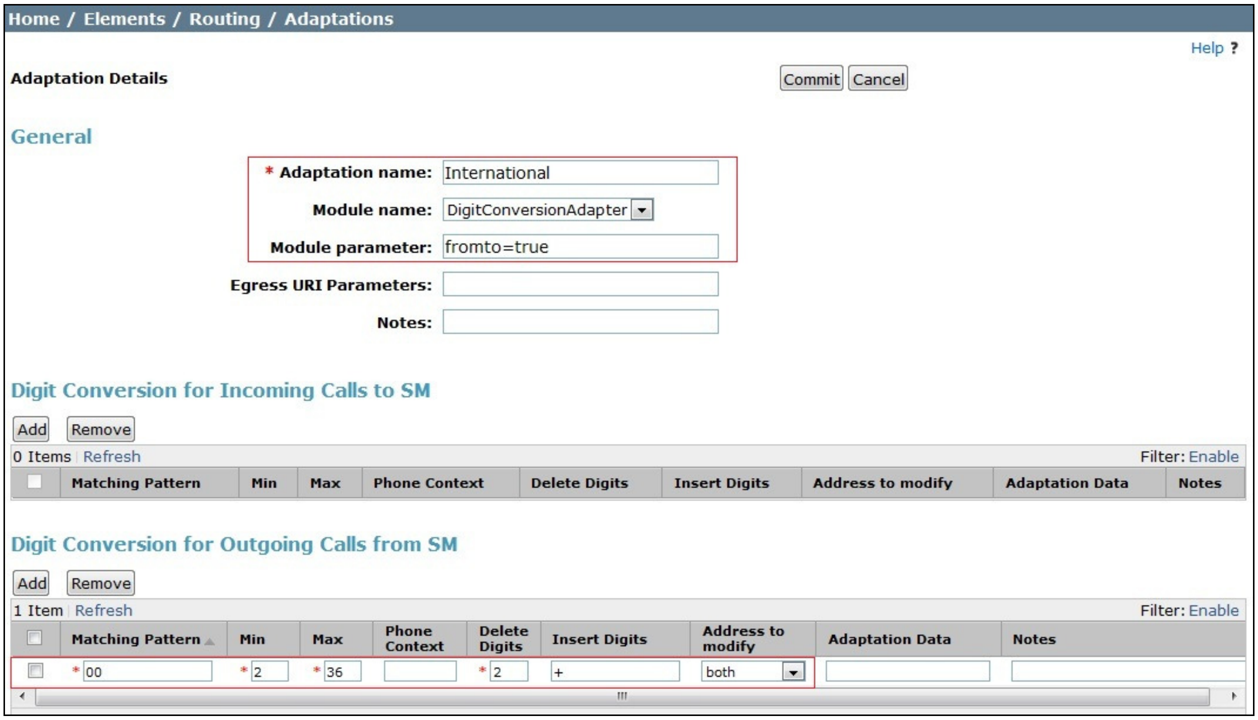
□	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
□	* 2000	* 4	* 4		* 4	0302nnnnn0	origination		
□	* 2298	* 4	* 4		* 4	0302nnnnn3	origination		
□	* 2316	* 4	* 4		* 4	0302nnnnn5	origination		
□	* 2346	* 4	* 4		* 4	0302nnnnn2	origination		
□	* 2396	* 4	* 4		* 4	0302nnnnn1	origination		
□	* 2400	* 4	* 4		* 4	0302nnnnn6	origination		
□	* 2402	* 4	* 4		* 4	0302nnnnn0	origination		
□	* 2460	* 4	* 4		* 4	0302nnnnn7	origination		
□	* 2611	* 4	* 4		* 4	0302nnnnn4	origination		

Select : All, None

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

An additional adaptation is used to convert international called numbers to E.164 format with leading "+". This adaptation is applied to the Avaya SBCE SIP Entity. On the **Routing** tab select **Adaptations** from the left-hand menu. Click on **New** (not shown).

- Under **Matching Pattern** enter the international dialling prefix to be removed, in this case **00**.
- Under **Min** and **Max** enter the Minimum and Maximum digits of the dialled number; the range set during test was large enough to accommodate any possible dialled number.
- Under **Delete Digits** enter the number required to remove the international dialling prefix, in this case **2**.
- Under **Insert Digits** enter the value required to convert the number to the standard format used for E.164 in SIP, that is the full E.164 number prefixed with a **+**.
- Under **Address to Modify** choose **destination** from the drop down box to apply this rule to the To and Request URI headers only.



Adaptation Details

General

* Adaptation name: International

Module name: DigitConversionAdapter

Module parameter: fromto=true

Egress URI Parameters:

Notes:

Digit Conversion for Incoming Calls to SM

0 Items

□	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

□	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
□	* 00	* 2	* 36		* 2	+	both		

Note: During test, the **Address to modify** was left as **both**. This is not required as the origination addresses were in national format and were unaffected by the adaptation.

6.5. Administer SIP Entities

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

Under **General**:

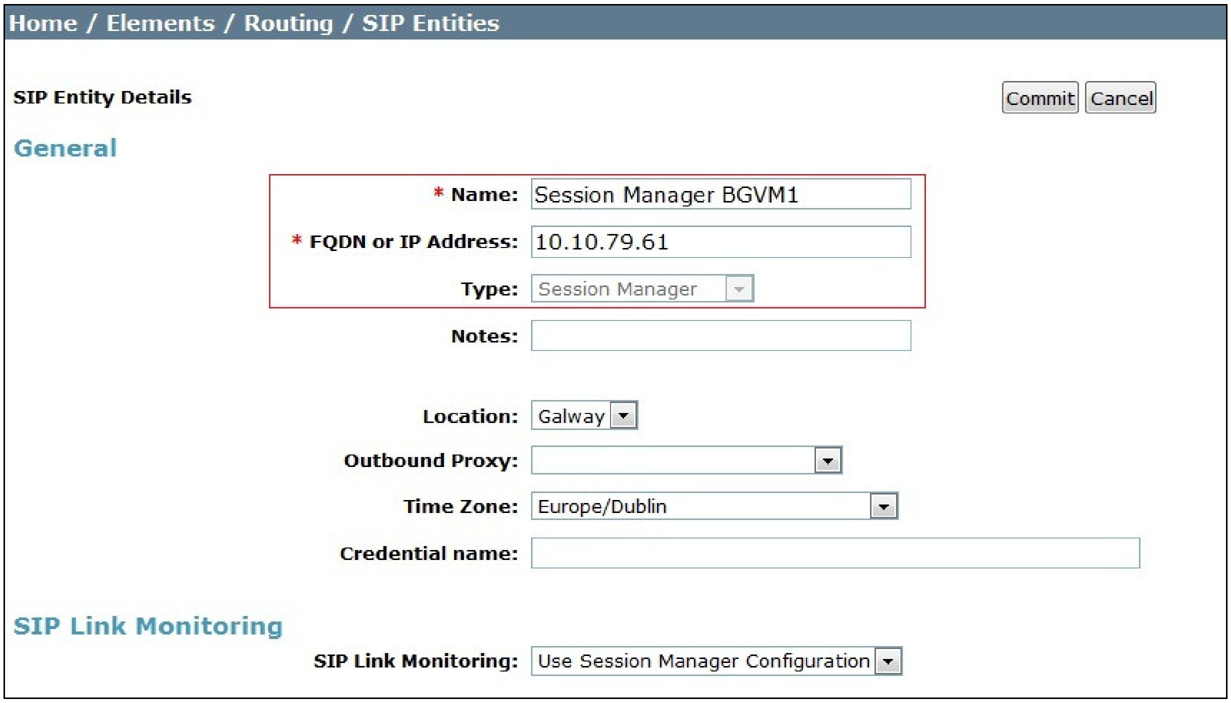
- In the **Name** field enter an informative name
- In the **FQDN or IP Address** field enter the IP address of the Session Manager or the signalling interface on the connecting system
- In the **Type** field use **Session Manager** for a Session Manager SIP entity, **CM** for a Communication Manager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

In this configuration there are three SIP Entities:

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

6.5.1. Avaya Aura® Session Manager SIP Entity

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



SIP Entity Details

General

* Name: Session Manager BGVM1

* FQDN or IP Address: 10.10.79.61

Type: Session Manager

Notes:

Location: Galway

Outbound Proxy:

Time Zone: Europe/Dublin

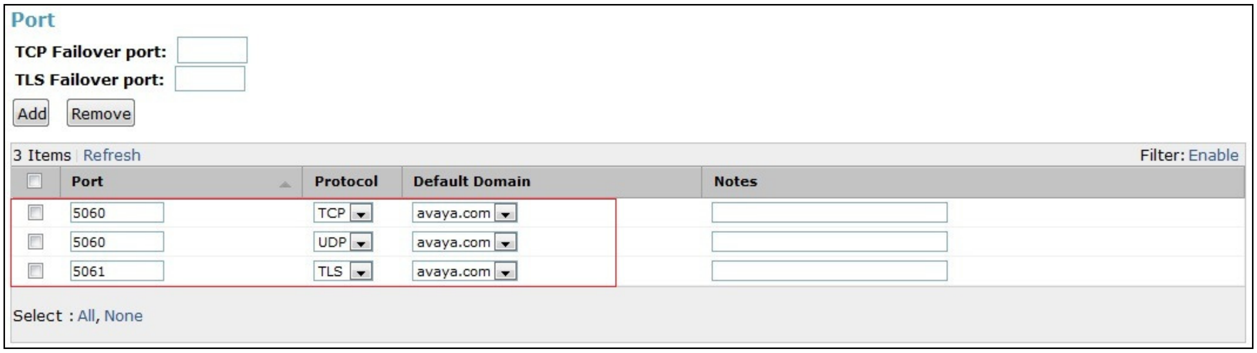
Credential name:

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



Port

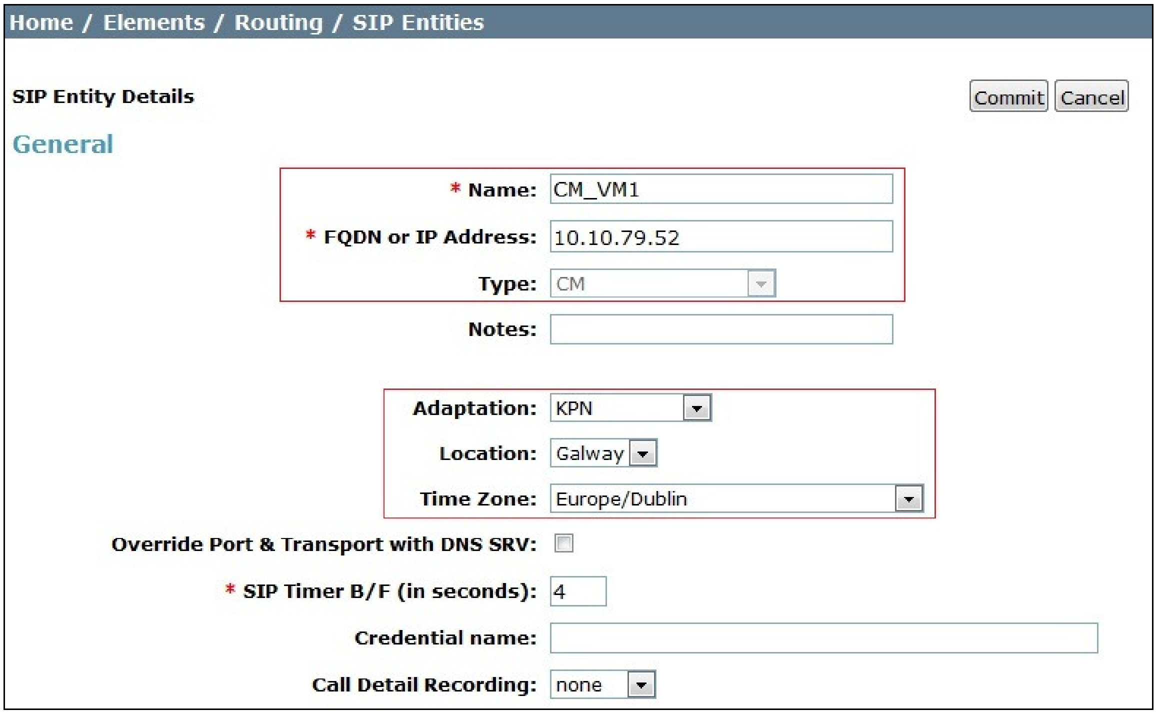
TCP Failover port:

TLS Failover port:

□	Port	Protocol	Default Domain	Notes
□	5060	TCP	avaya.com	
□	5060	UDP	avaya.com	
□	5061	TLS	avaya.com	

6.5.2. Avaya Aura® Communication Manager SIP Entity

The following screen shows the SIP entity for Communication Manager which is configured as an Evolution Server. The **FQDN or IP Address** field is set to the IP address of the interface on Communication Manager that will be providing SIP signalling. Set the **Location** to that defined in **Section 6.3**, the Adaptation to that defined in **Section 6.4** and the **Time Zone** to the appropriate time zone.



SIP Entity Details

General

* Name: CM_VM1

* FQDN or IP Address: 10.10.79.52

Type: CM

Notes:

Adaptation: KPN

Location: Galway

Time Zone: Europe/Dublin

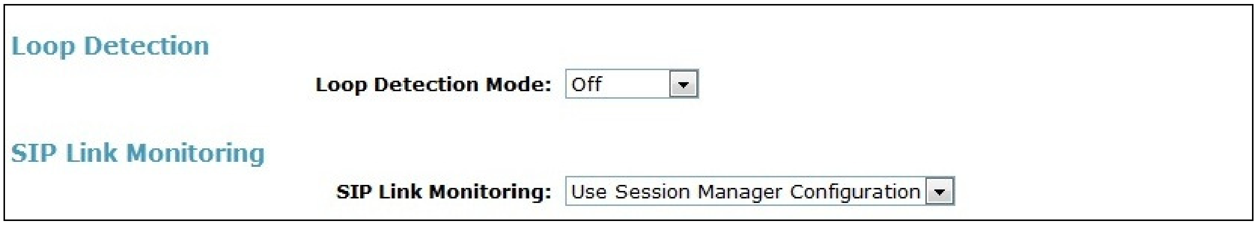
Override Port & Transport with DNS SRV: □

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

Credential name:

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

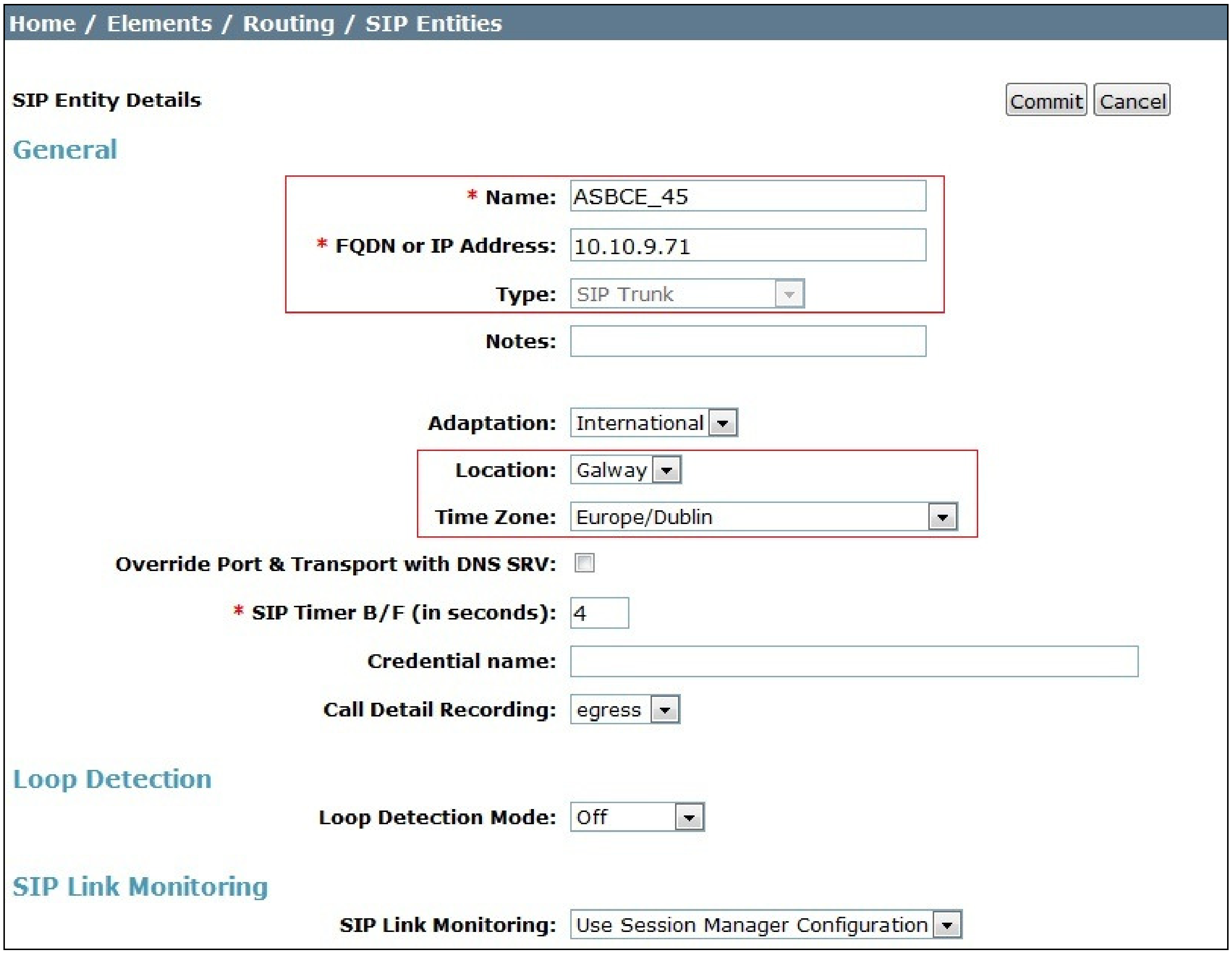
Loop Detection Mode: Off

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

6.5.3. Avaya Session Border Controller for Enterprise SIP Entity

The following screen shows the SIP Entity for the Avaya SBCE. The **FQDN or IP Address** field is set to the IP address of the Avaya SBCE private network interface (see **Figure 1**). Set the location to that defined in **Section 6.3** and the **Time Zone** to the appropriate time zone.



SIP Entity Details

General

* Name: ASBCE_45

* FQDN or IP Address: 10.10.9.71

Type: SIP Trunk

Notes:

Adaptation: International

Location: Galway

Time Zone: Europe/Dublin

Override Port & Transport with DNS SRV: □

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

Credential name:

Call Detail Recording: egress

Loop Detection

Loop Detection Mode: Off

SIP Link Monitoring

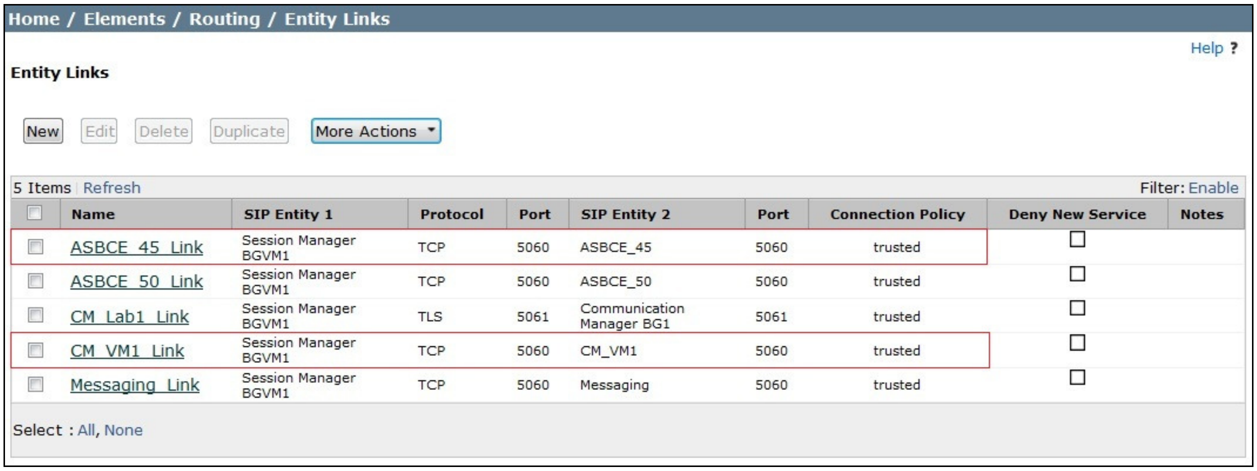
SIP Link Monitoring: Use Session Manager Configuration

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, on the **Routing** tab select **Entity Links** on the left panel menu and click on the **New** button (not shown). Fill in the following fields in the new row that is displayed.

- In the **Name** field enter an informative name
- In the **SIP Entity 1** field select **Session Manager**
- In the **Port** field enter the port number to which the other system sends its SIP requests
- In the **SIP Entity 2** field enter the other SIP Entity for this link, created in **Section 6.5**
- In the **Port** field enter the port number to which the other system expects to receive SIP requests
- Select the **Trusted** tick box to make the other system trusted
- In the **Protocol** field enter the transport protocol to be used to send SIP requests

Click **Commit** to save changes. The following screen shows the Entity Links used in this configuration.



□	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service	Notes
□	ASBCE_45_Link	Session Manager BGVM1	TCP	5060	ASBCE_45	5060	trusted	□	
□	ASBCE_50_Link	Session Manager BGVM1	TCP	5060	ASBCE_50	5060	trusted	□	
□	CM_Lab1_Link	Session Manager BGVM1	TLS	5061	Communication Manager BG1	5061	trusted	□	
□	CM_VM1_Link	Session Manager BGVM1	TCP	5060	CM_VM1	5060	trusted	□	
□	Messaging_Link	Session Manager BGVM1	TCP	5060	Messaging	5060	trusted	□	

Note: The Session Manager used for testing is also used with other test equipment. Only the Entity Links highlighted in the above screenshot are valid for this configuration.

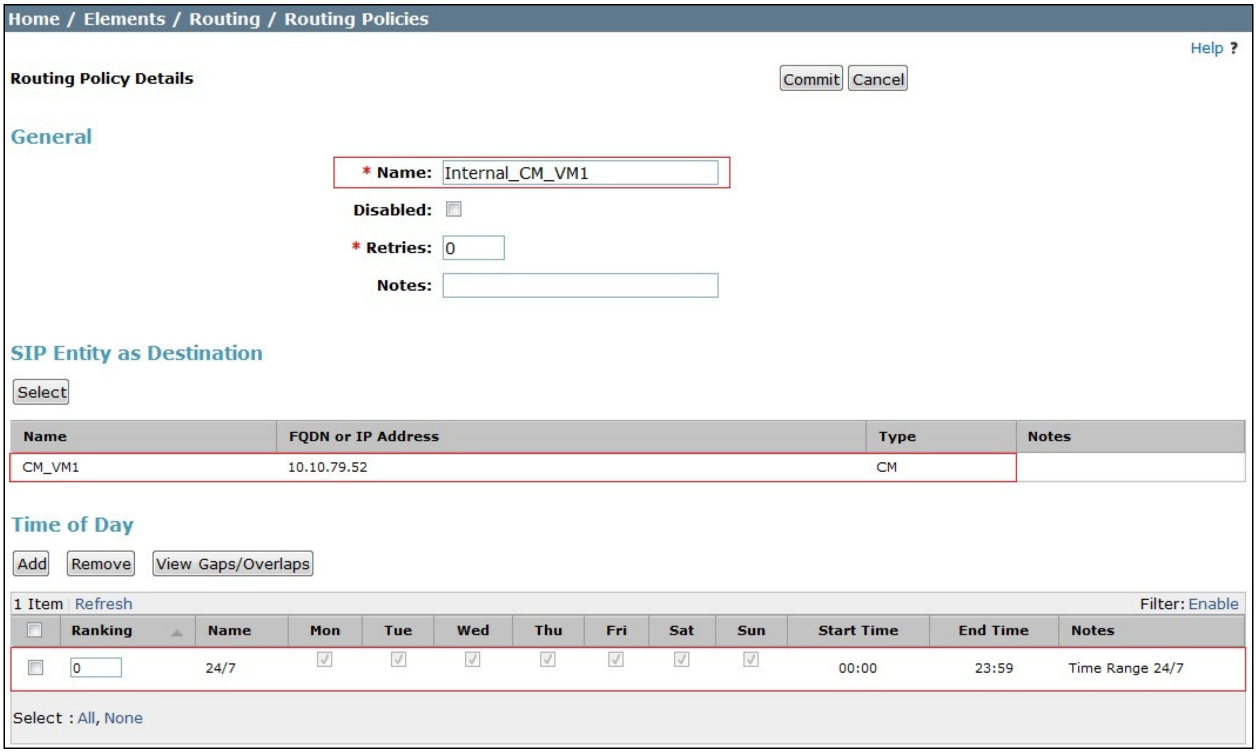
6.7. Administer Routing Policies

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

Under **General**:

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

The following screen shows the routing policy for Communication Manager.



Routing Policy Details

General

* Name: Internal_CM_VM1

Disabled: □

* Retries: 0

Notes:

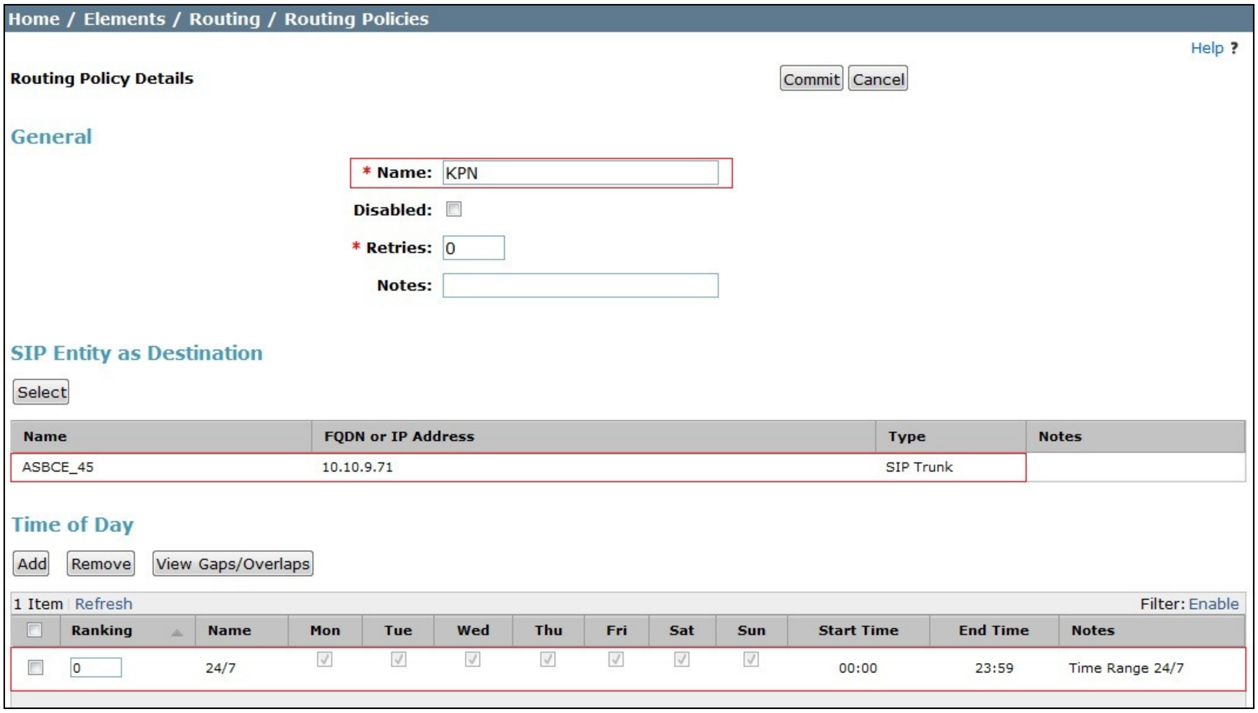
SIP Entity as Destination

Name	FQDN or IP Address	Type	Notes
CM_VM1	10.10.79.52	CM	

Time of Day

□	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

The following screen shows the routing policy for the Avaya SBCE and onward routing to the PSTN via KPN VoIP Connect.



Routing Policy Details

General

* Name: KPN

Disabled: □

* Retries: 0

Notes:

SIP Entity as Destination

Name	FQDN or IP Address	Type	Notes
ASBCE_45	10.10.9.71	SIP Trunk	

Time of Day

1 Item Refresh 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

6.8. Administer Dial Patterns

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

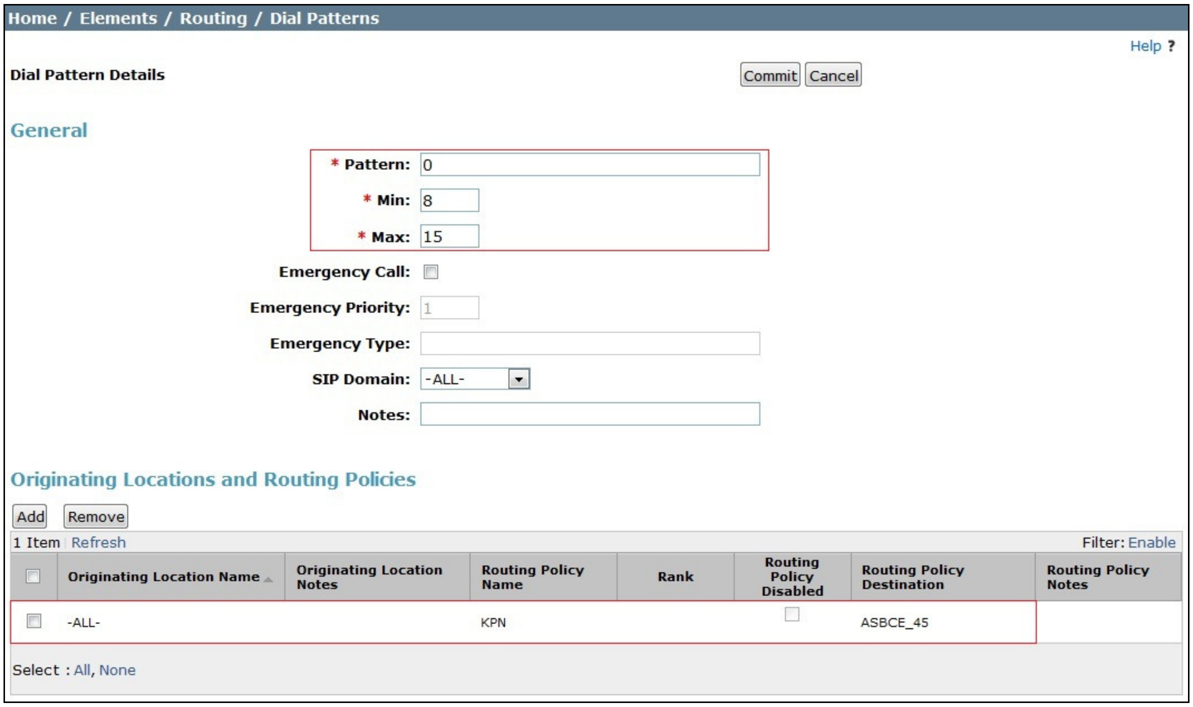
Under **General**:

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

Under **Originating Locations and Routing Policies**:

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

The following screen shows an example dial pattern configured for the Avaya SBCE which will route the calls out to KPN VoIP Connect.



Dial Pattern Details

General

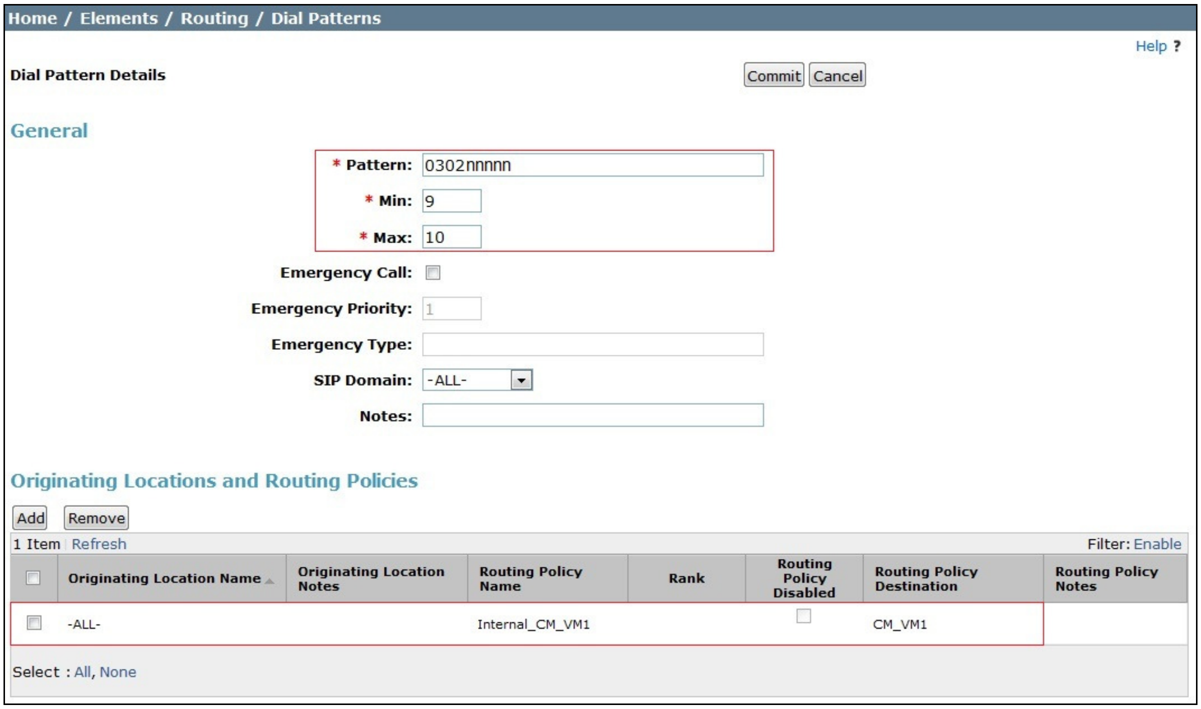
* Pattern:	0
* Min:	8
* Max:	15
Emergency Call:	□
Emergency Priority:	1
Emergency Type:	
SIP Domain:	-ALL-
Notes:	

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-		KPN		□	ASBCE_45	

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



Dial Pattern Details

General

* Pattern:	0302nnnnn
* Min:	9
* Max:	10
Emergency Call:	□
Emergency Priority:	1
Emergency Type:	
SIP Domain:	-ALL-
Notes:	

Originating Locations and Routing Policies

1 Item

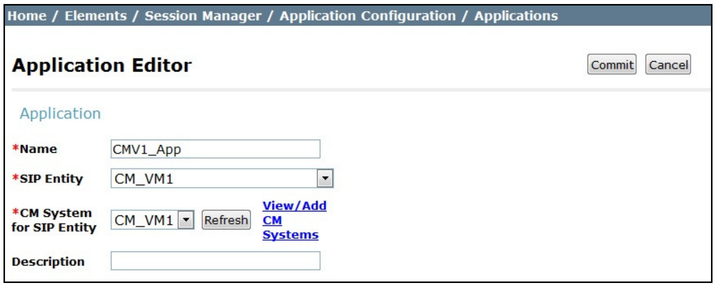
□	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
□	-ALL-		Internal_CM_VM1		□	CM_VM1	

Note: The pattern to be matched has been obscured.

6.9. Administer Application for Avaya Aura® Communication Manager

From the **Home** tab select **Session Manager** from the menu. In the resulting tab from the left panel menu select **Application Configuration → Applications** and click **New**.

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



Home / Elements / Session Manager / Application Configuration / Applications

Application Editor

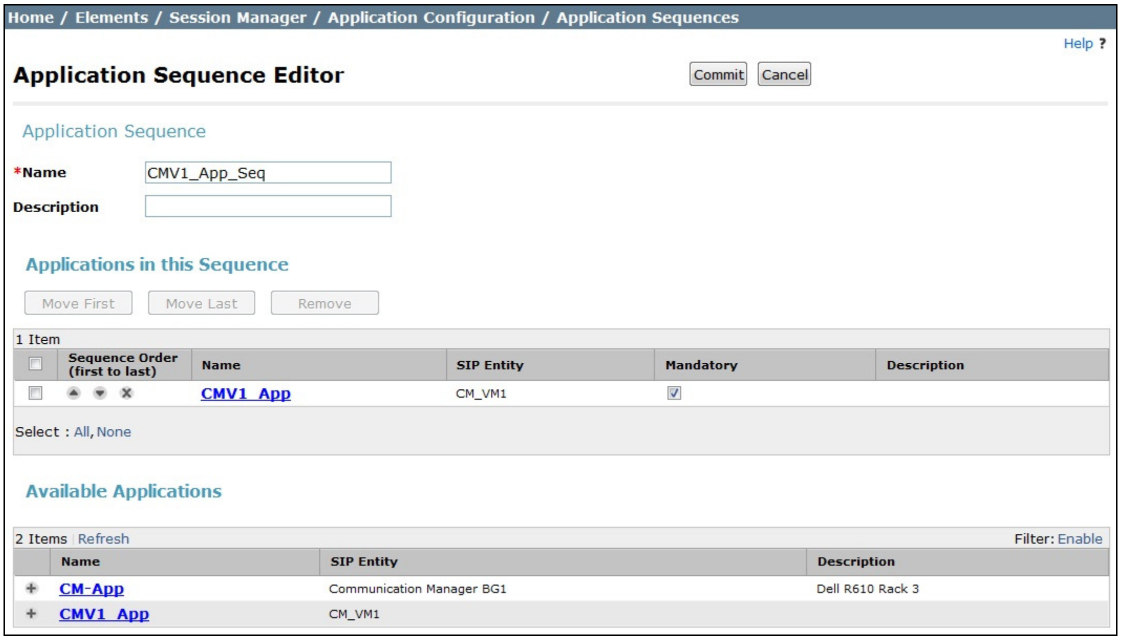
Application

*Name	CMV1_App
*SIP Entity	CM_VM1
*CM System for SIP Entity	CM_VM1 Refresh View/Add CM Systems
Description	

6.10. Administer Application Sequence for Avaya Aura® Communication Manager

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

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



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

Application Sequence Editor

Application Sequence

*Name	CMV1_App_Seq
Description	

Applications in this Sequence

1 Item

☐	Sequence Order (first to last)	Name	SIP Entity	Mandatory	Description
☐	▲ ▼ ✕	CMV1_App	CM_VM1	☑	

Select : All, None

Available Applications

2 Items | Refresh   Filter: Enable

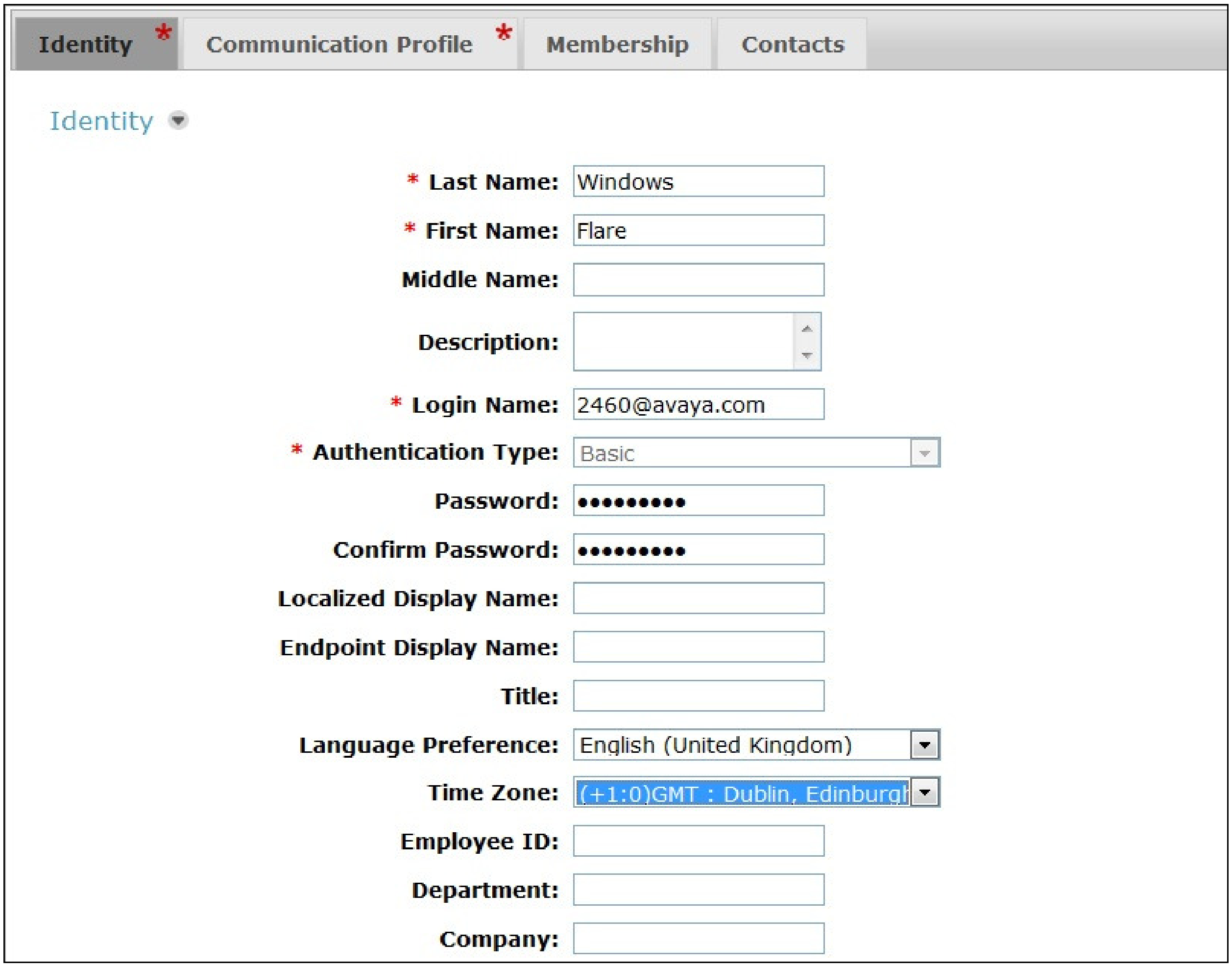
	Name	SIP Entity	Description
+	CM-App	Communication Manager BG1	Dell R610 Rack 3
+	CMV1_App	CM_VM1	

6.11. Administer SIP Extensions

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

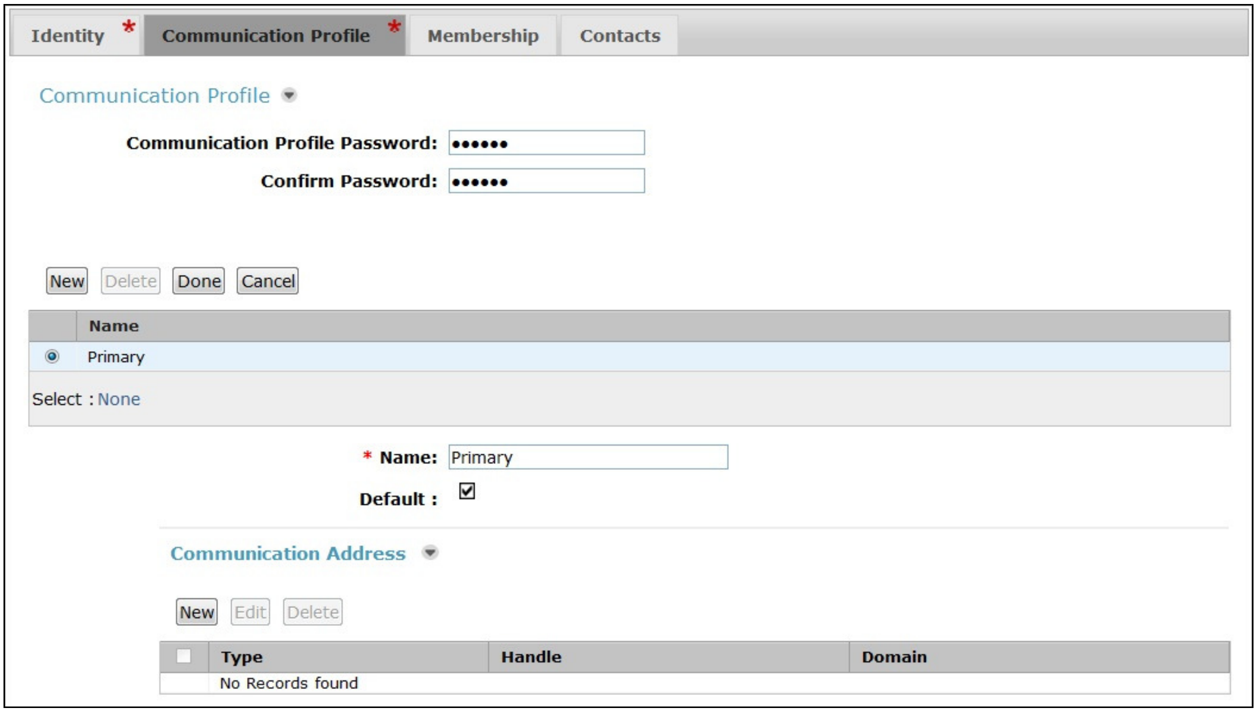
On the **Identity** tab:

- Enter the user's name in the **Last Name** and **First Name** fields
- In the **Login Name** field enter a unique system login name in the form of user@domain e.g. **2460@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



* Last Name:	Windows
* First Name:	Flare
Middle Name:	
Description:	
* Login Name:	2460@avaya.com
* Authentication 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:	

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



Communication Profile Password: ••••••

Confirm Password: ••••••

	Name
●	Primary

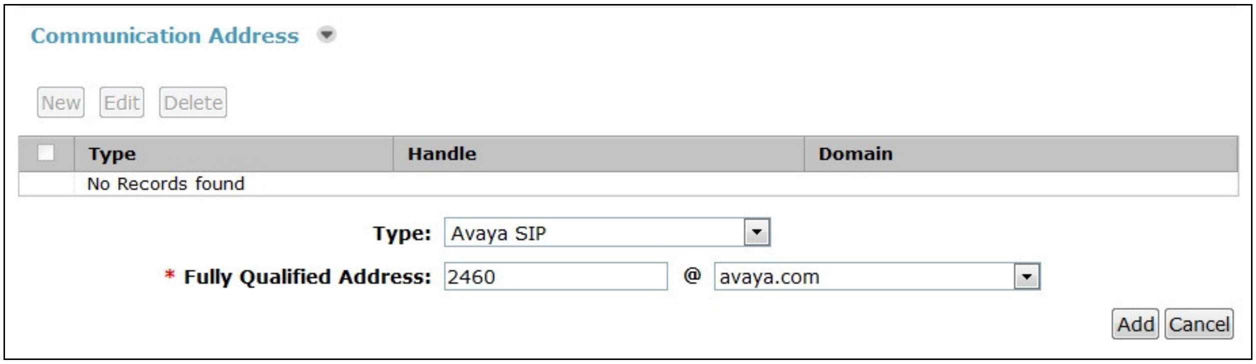
Select : None

* Name: Primary

Default : ☑

□	Type	Handle	Domain
	No Records found		

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.



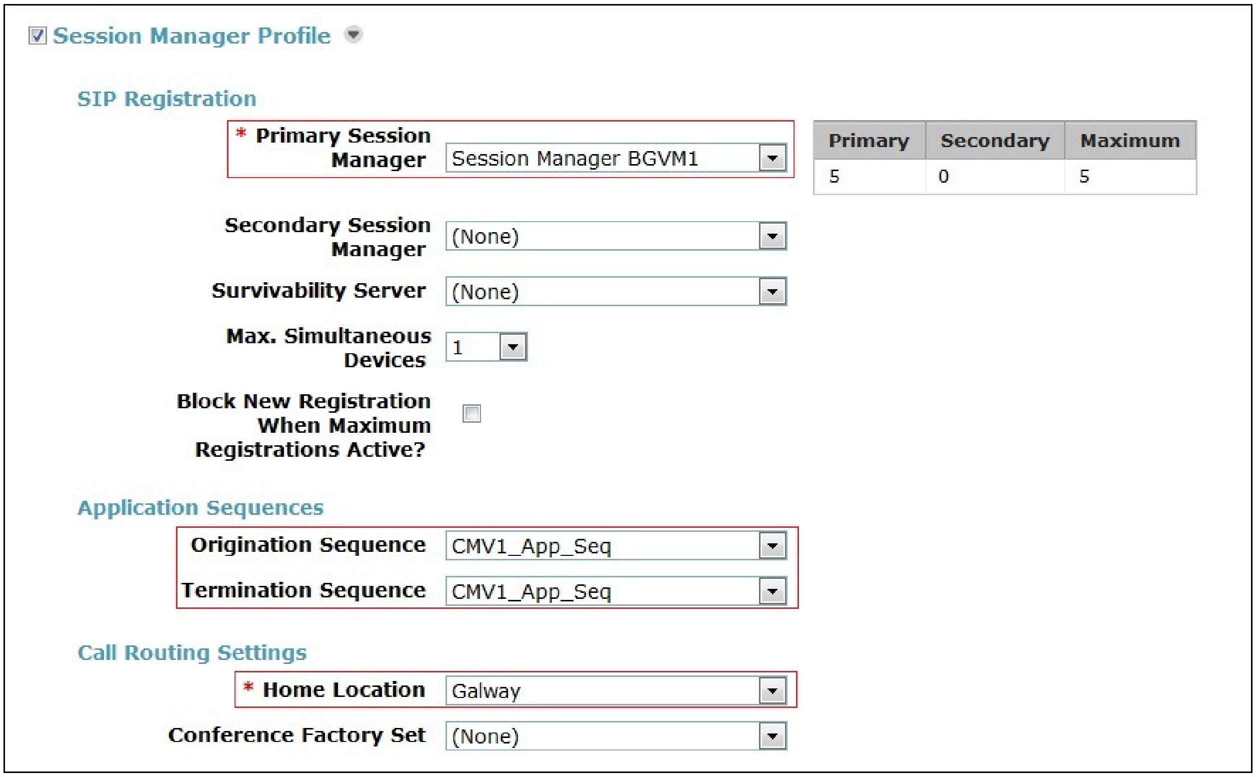
□	Type	Handle	Domain
	No Records found		

Type: Avaya SIP

* Fully Qualified Address: 2460 @ 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 Application Sequence** field configured in **Section 6.10**
- Select the appropriate application sequence from the drop-down menu in the **Termination Application 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 BGVM1

Primary	Secondary	Maximum
5	0	5

Secondary Session Manager	(None)
Survivability Server	(None)
Max. Simultaneous Devices	1
Block New Registration When Maximum Registrations Active?	□

Application Sequences

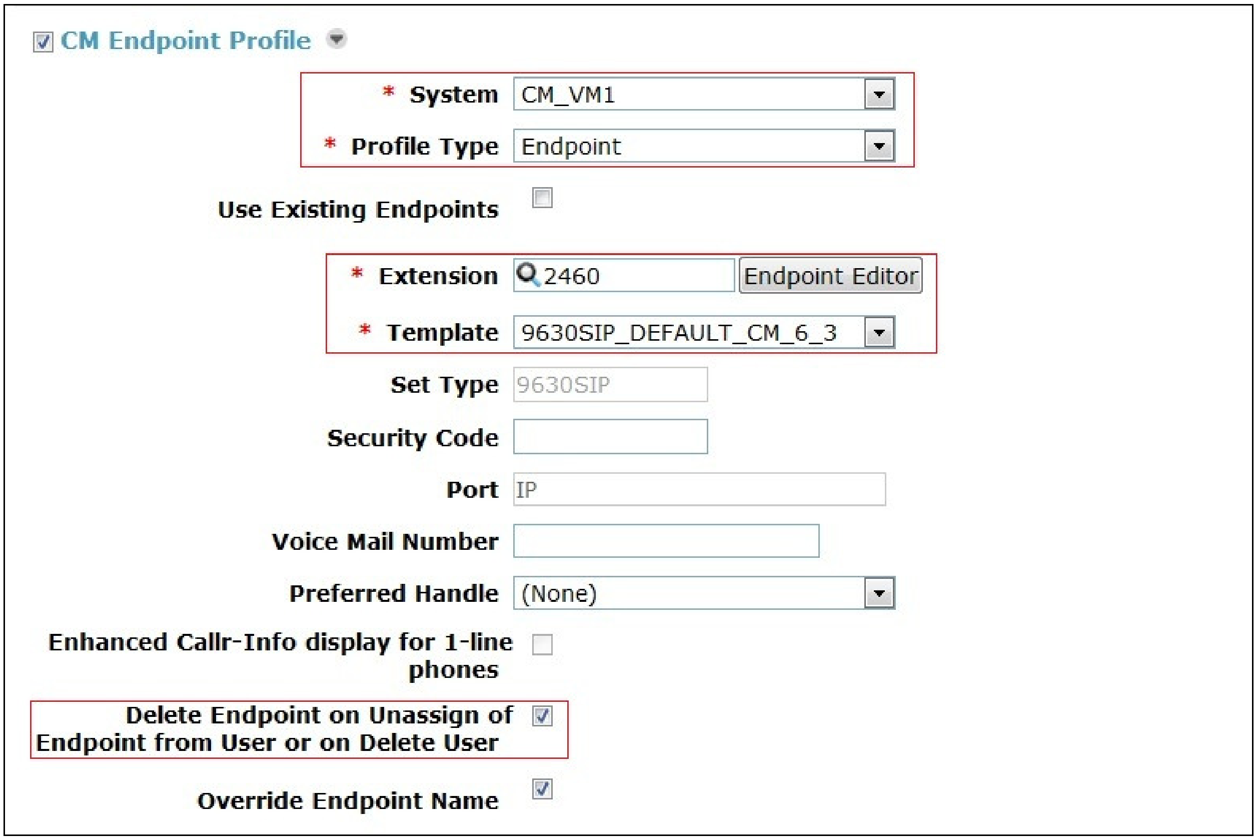
Origination Sequence	CMV1_App_Seq
Termination Sequence	CMV1_App_Seq

Call Routing Settings

* Home Location	Galway
Conference Factory Set	(None)

Expand the CM **Endpoint Profile** section.

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



☑ CM Endpoint Profile

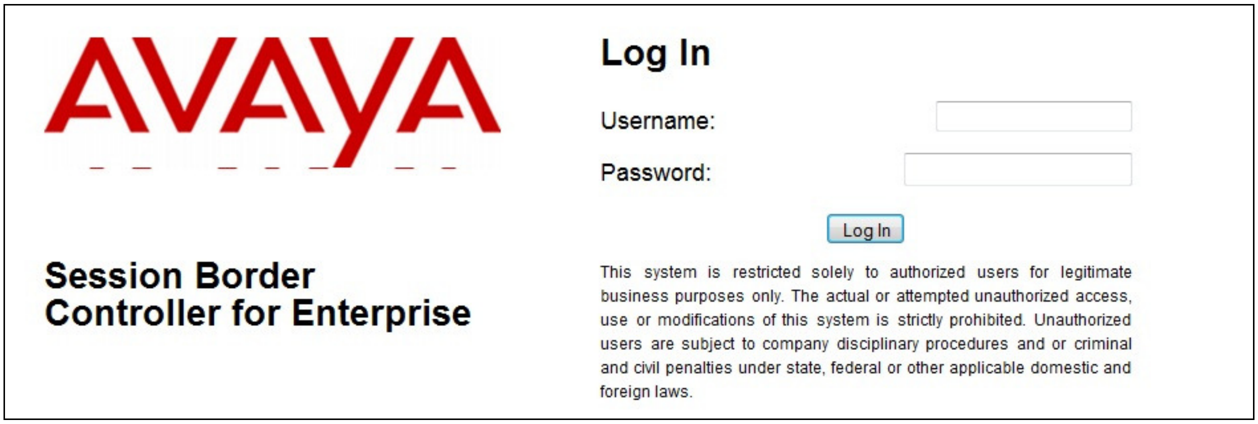
* System	CM_VM1
* Profile Type	Endpoint
Use Existing Endpoints	□
* Extension	2460
* Template	9630SIP_DEFAULT_CM_6_3
Set Type	9630SIP
Security Code	
Port	IP
Voice Mail Number	
Preferred Handle	(None)
Enhanced Callr-Info display for 1-line phones	□
Delete Endpoint on Unassign of Endpoint from User or on Delete User	☑
Override Endpoint Name	☑

7. Configure Avaya Session Border Controller for Enterprise

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

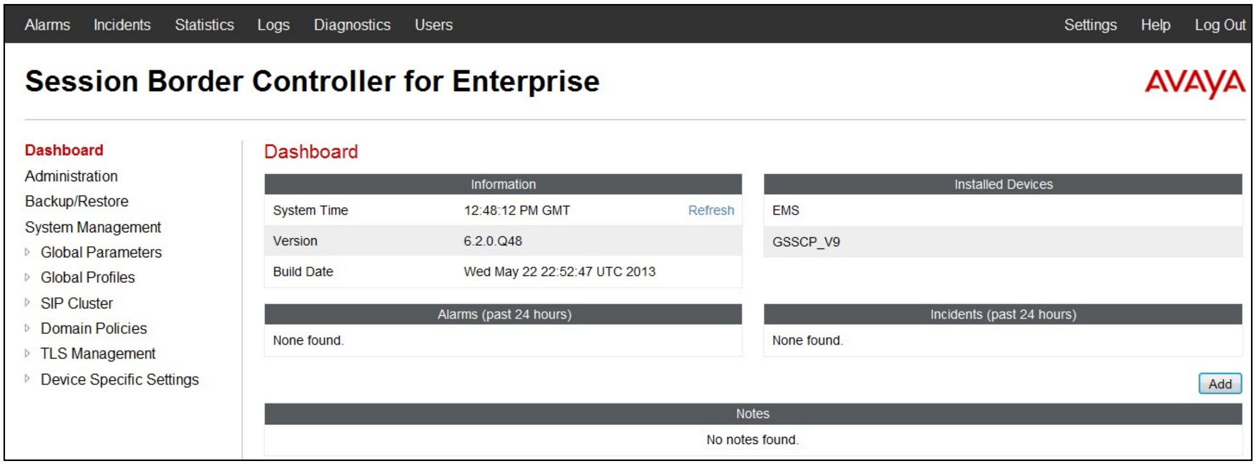
7.1. Access Avaya Session Border Controller for Enterprise

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



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.

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.



Dashboard

Information		
System Time	12:48:12 PM GMT	Refresh
Version	6.2.0.Q48	
Build Date	Wed May 22 22:52:47 UTC 2013	

Installed Devices
EMS
GSSCP_V9

Alarms (past 24 hours)
None found.

Incidents (past 24 hours)
None found.

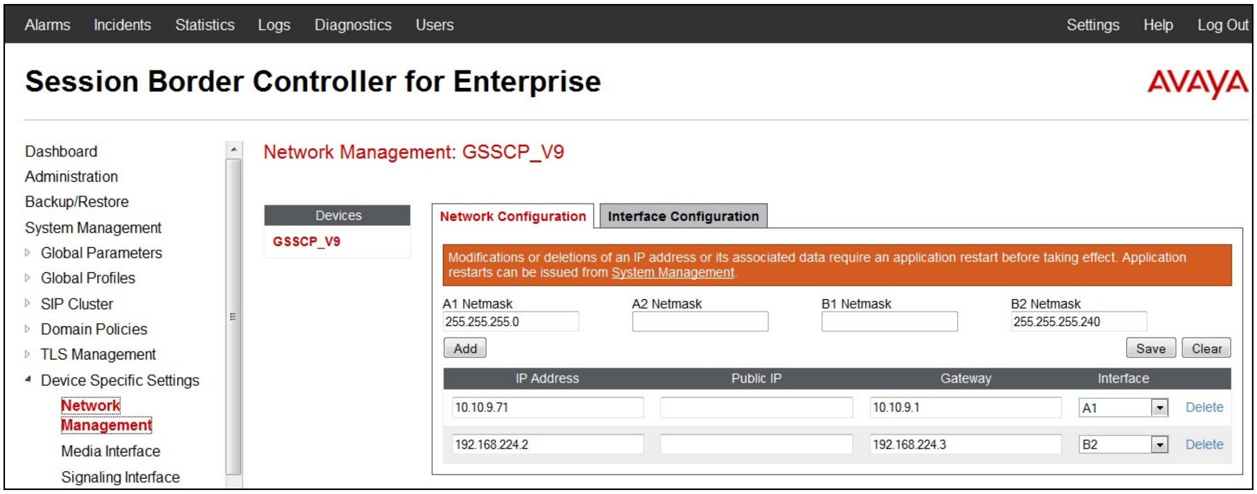
Notes
No notes found.

7.2. Define Network Information

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

To define the network information, navigate to **Device Specific Settings → Network Management** in the main menu on the left hand side and click on **Add**. Enter details in the blank box that appears at the end of the list

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



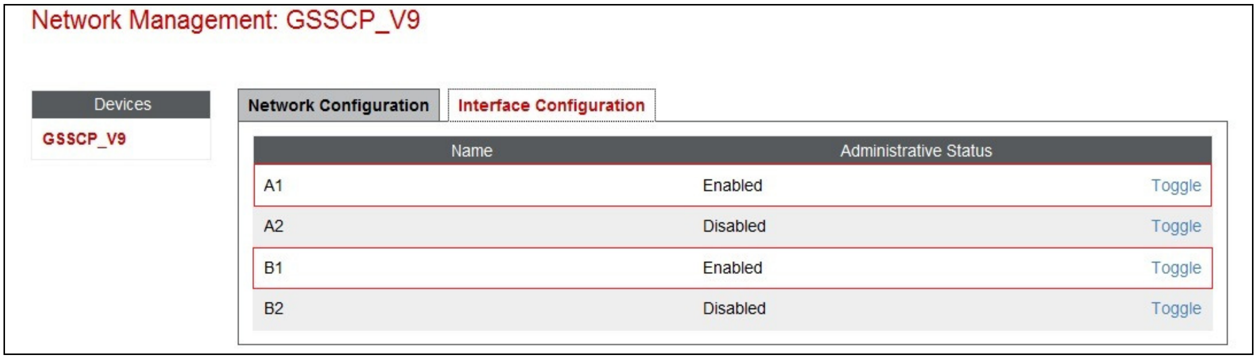
Network Management: GSSCP_V9

Modifications or deletions of an IP address or its associated data require an application restart before taking effect. Application restarts can be issued from System Management.

A1 Netmask	A2 Netmask	B1 Netmask	B2 Netmask
255.255.255.0			255.255.255.240

IP Address	Public IP	Gateway	Interface	
10.10.9.71		10.10.9.1	A1	Delete
192.168.224.2		192.168.224.3	B2	Delete

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



Network Management: GSSCP_V9

Name	Administrative Status	
A1	Enabled	Toggle
A2	Disabled	Toggle
B1	Enabled	Toggle
B2	Disabled	Toggle

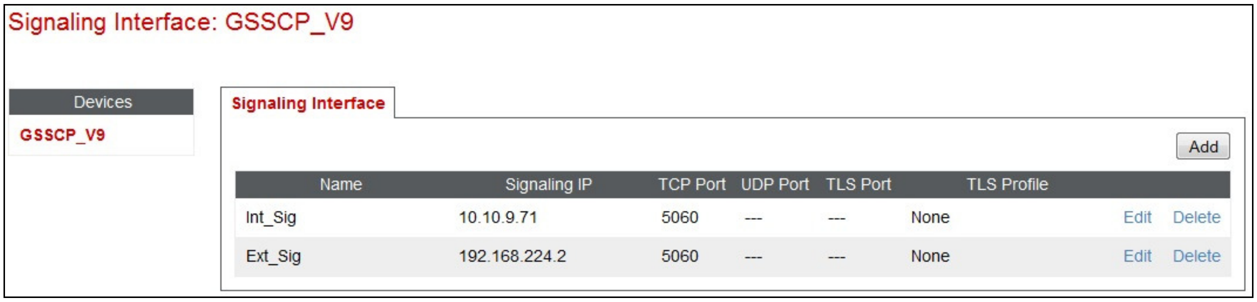
7.3. Define Interfaces

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

7.3.1. Signalling Interfaces

To define the signalling interfaces on the Avaya SBCE, navigate to **Device Specific Settings → Signaling Interface** (not shown) in the main menu on the left hand side. Details of transport protocol and ports for the internal and external SIP signalling are entered here.

- Select **Add** and enter details of the internal signalling interface in the pop-up menu (not shown)
- In the **Name** field enter a descriptive name for the internal signalling interface
- For **Signaling IP**, select an **internal** signalling interface IP address defined in **Section 7.2**
- Select **TCP** port number, **5060** is used for the Session Manager
- Select **Add** and enter details of the external signalling interface in the pop-up menu (not shown)
- In the **Name** field enter a descriptive name for the external signalling interface
- For **Signaling IP**, select an **external** signalling interface IP address defined in **Section 7.2**
- Select **TCP** port number, **5060** is used for KPN VoIP Connect

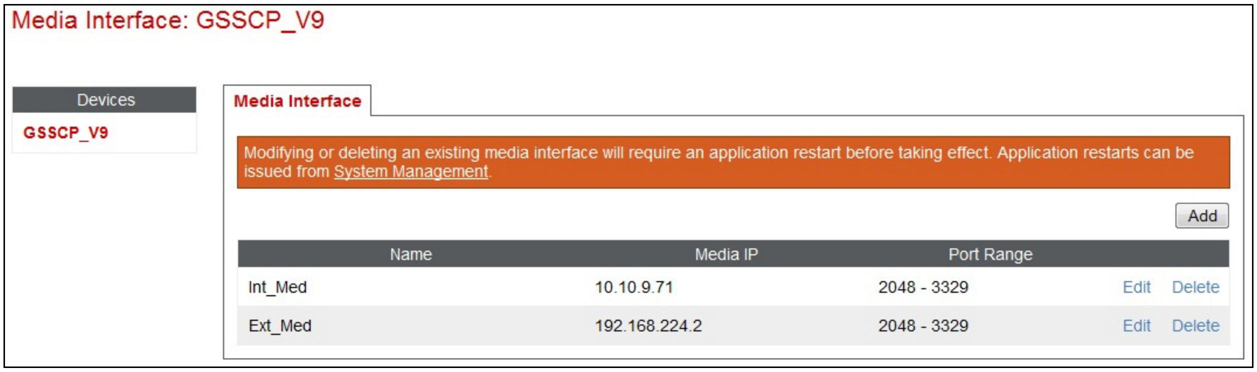


Name	Signaling IP	TCP Port	UDP Port	TLS Port	TLS Profile		
Int_Sig	10.10.9.71	5060	---	---	None	Edit	Delete
Ext_Sig	192.168.224.2	5060	---	---	None	Edit	Delete

7.3.2. Media Interfaces

To define the media interfaces on the Avaya SBCE, navigate to **Device Specific Settings → Media Interface** in the main menu on the left hand side. Details of the RTP and SRTP port ranges for the internal and external media streams are entered here. The IP addresses for media can be the same as those used for signalling.

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



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	Port Range		
Int_Med	10.10.9.71	2048 - 3329	Edit	Delete
Ext_Med	192.168.224.2	2048 - 3329	Edit	Delete

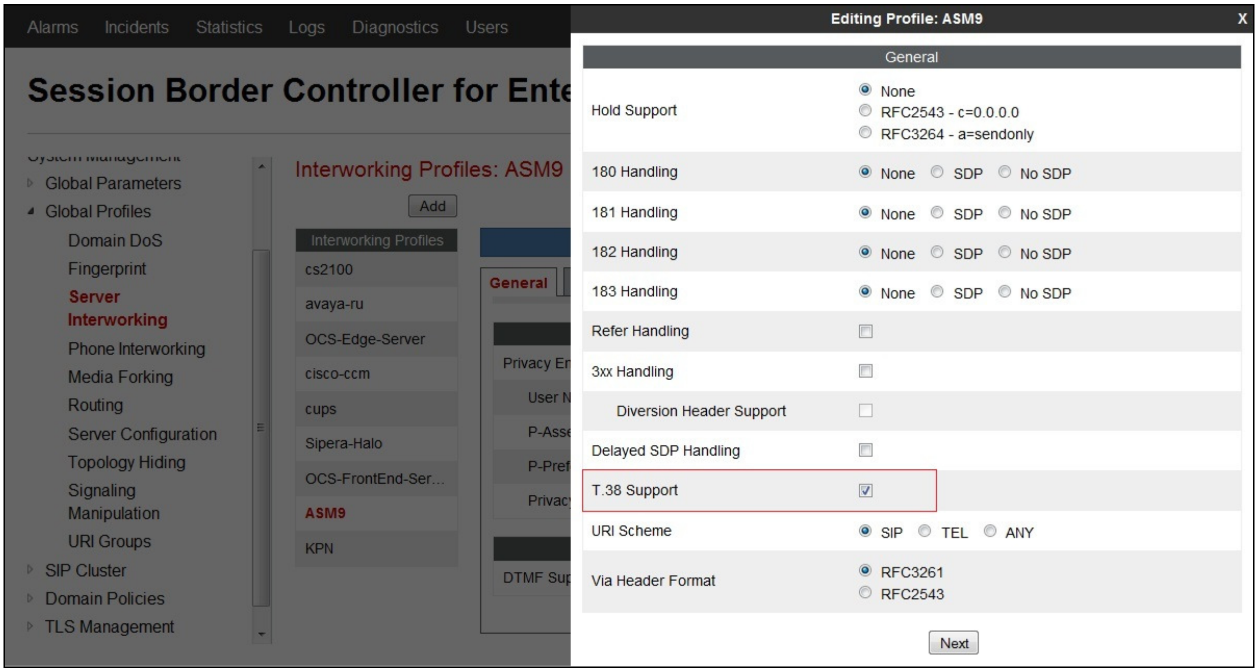
Note: During test the port ranges for the internal and external media interfaces were defined as the default values used by the Communication Manager,

7.4. Define Server Interworking

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

To define server interworking on the Avaya SBCE, navigate to **Global Profiles → Server Interworking** in the main menu on the left hand side. To define Server Interworking for the Session Manager, highlight the **avaya-ru** profile which is a factory setting appropriate for Avaya equipment and select **Clone Profile**. A pop-up menu is generated headed **Clone Profile** (not shown)

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



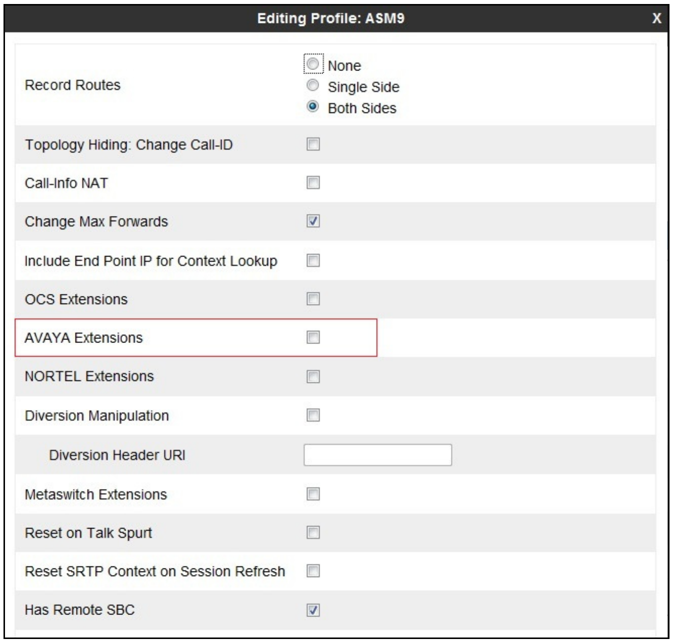
Editing Profile: ASM9

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	□
3xx Handling	□
Diversion Header Support	□
Delayed SDP Handling	□
T.38 Support	☑
URI Scheme	● SIP ○ TEL ○ ANY
Via Header Format	● RFC3261 ○ RFC2543

Next

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



Editing Profile: ASM9

Record Routes	○ None ○ Single Side ● Both Sides
Topology Hiding: Change Call-ID	□
Call-Info NAT	□
Change Max Forwards	☑
Include End Point IP for Context Lookup	□
OCS Extensions	□
AVAYA Extensions	□
NORTEL Extensions	□
Diversion Manipulation	□
Diversion Header URI	
Metaswitch Extensions	□
Reset on Talk Spurt	□
Reset SRTP Context on Session Refresh	□
Has Remote SBC	☑

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

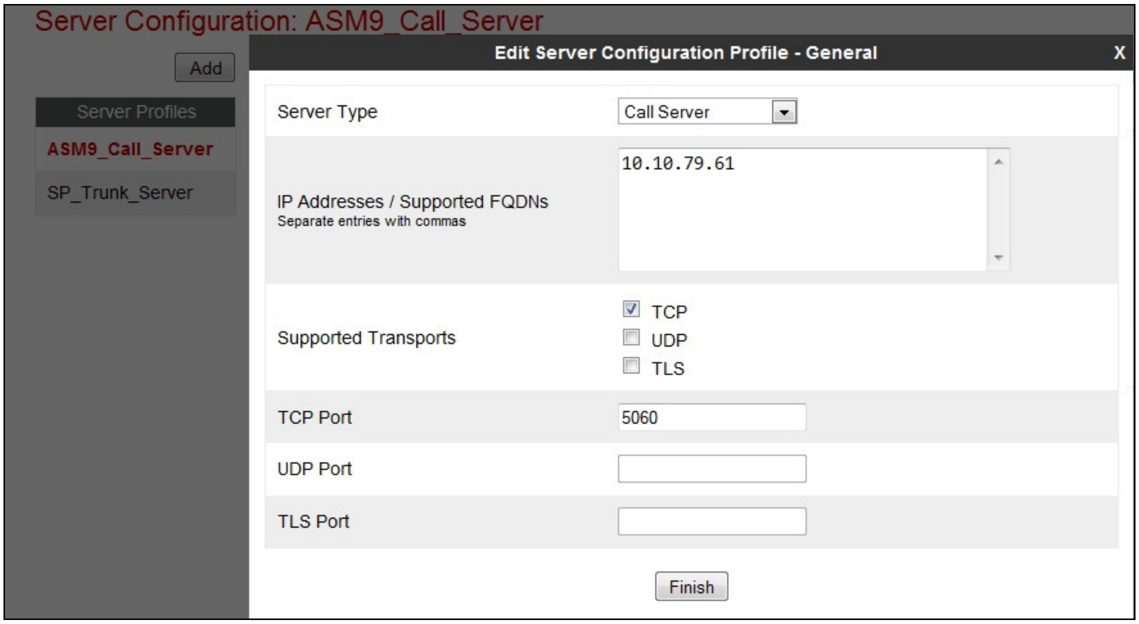
- In the **Clone Name** field enter a descriptive name for server interworking profile for KPN VoIP Connect and click **Finish** – in test **KPN** was used
- Select **Edit** and enter details in the pop-up menu
- Check the **T.38** box
- Select **Next** three times and **Finish**

7.5. Define Servers

Servers are defined for each server connected to the Avaya SBCE. In this case, KPN VoIP Connect is connected as the Trunk Server and the Session Manager is connected as the Call Server. To define the Session Manager, navigate to **Global Profiles → Server Configuration** in the main menu on the left hand side.

Click on **Add** and enter details in the pop-up menu

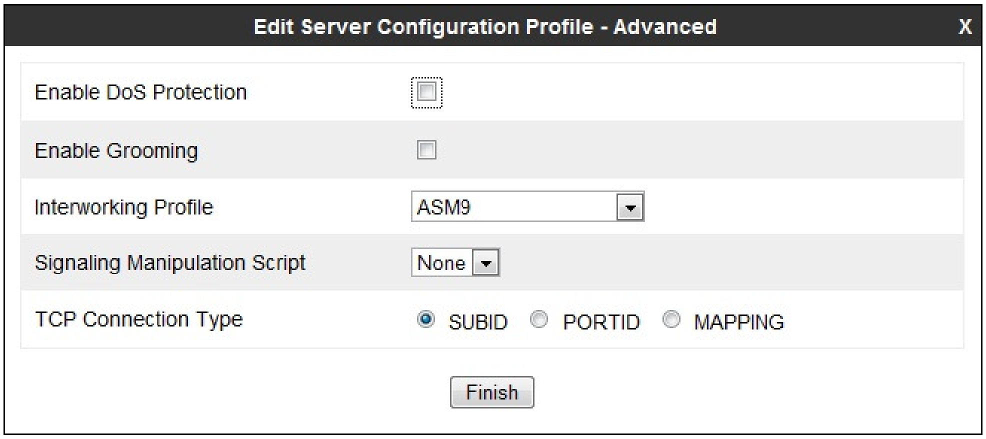
- In the **Profile Name** field enter a descriptive name for the Session Manager and click **Next** (not shown)
- In the **Server Type** drop down menu, select **Call Server**
- In the **IP Addresses / Supported FQDNs** box, type the Session Manager SIP interface address which is the same as that defined on the Communication Manager in **Section 5.2**
- Check **TCP** in **Supported Transports**
- Define the **TCP** port for SIP signalling, **5060** is used for the Session Manager and click **Finish**



Edit Server Configuration Profile - General

Server Type	Call Server
IP Addresses / Supported FQDNs Separate entries with commas	10.10.79.61
Supported Transports	☑ TCP □ UDP □ TLS
TCP Port	5060
UDP Port	
TLS Port	

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

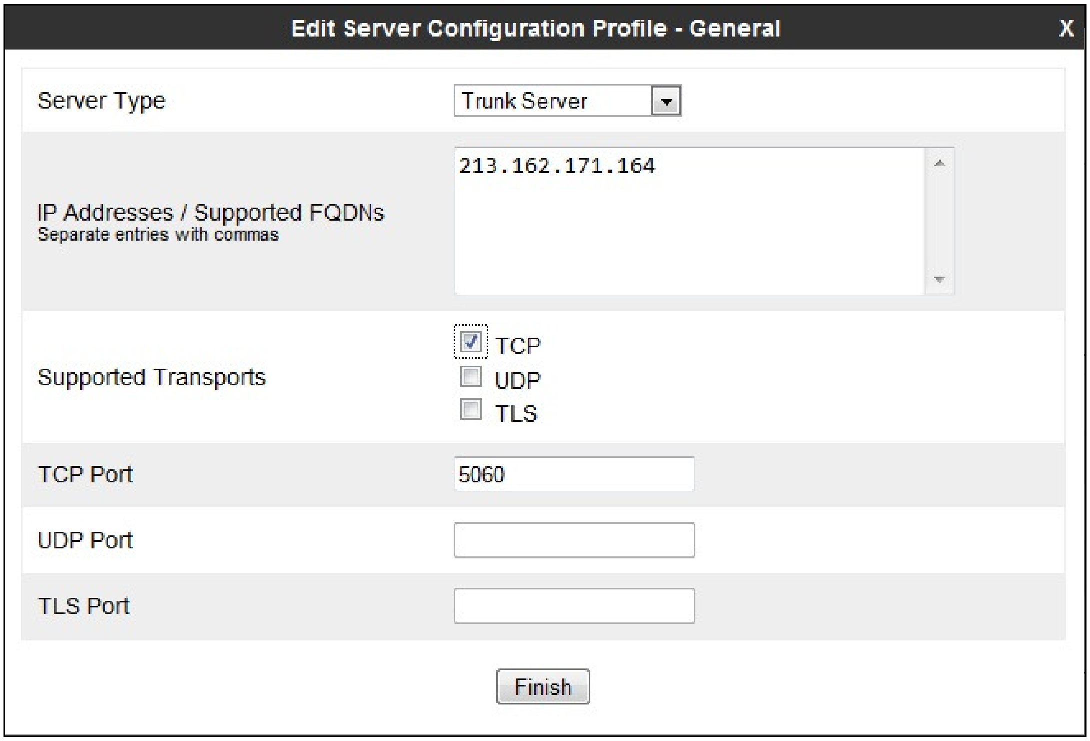


Edit Server Configuration Profile - Advanced

Enable DoS Protection	□
Enable Grooming	□
Interworking Profile	ASM9
Signaling Manipulation Script	None
TCP Connection Type	● SUBID ○ PORTID ○ MAPPING

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

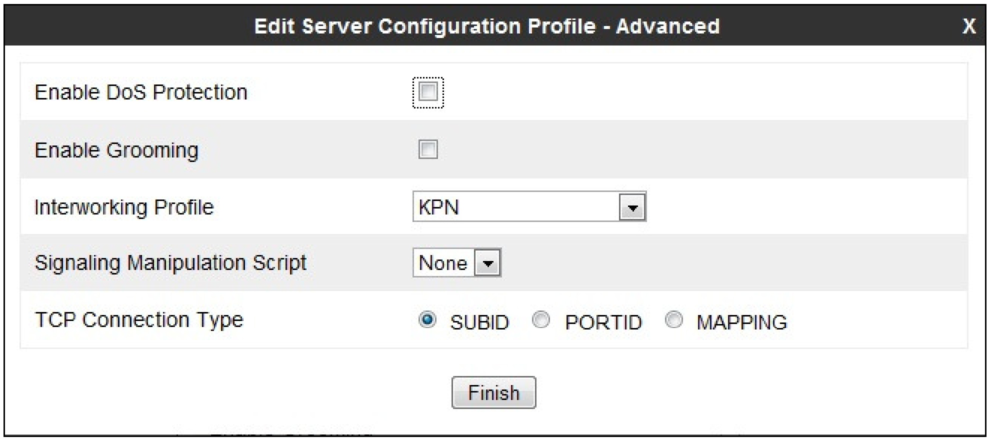
- In the **Profile Name** field enter a descriptive name for KPN VoIP Connect and click **Next** (not shown)
- In the **Server Type** drop down menu, select **Trunk Server**
- In the **IP Addresses / Supported FQDNs** box, type the IP address of KPN VoIP Connect
- Check **TCP** in **Supported Transports**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



Edit Server Configuration Profile - General

Server Type	Trunk Server
IP Addresses / Supported FQDNs Separate entries with commas	213.162.171.164
Supported Transports	☑ TCP □ UDP □ TLS
TCP Port	5060
UDP Port	
TLS Port	

- Click **Next** again then select the **Interworking Profile** for the KPN VoIP Connect defined in **Section 7.4** from the drop down menu



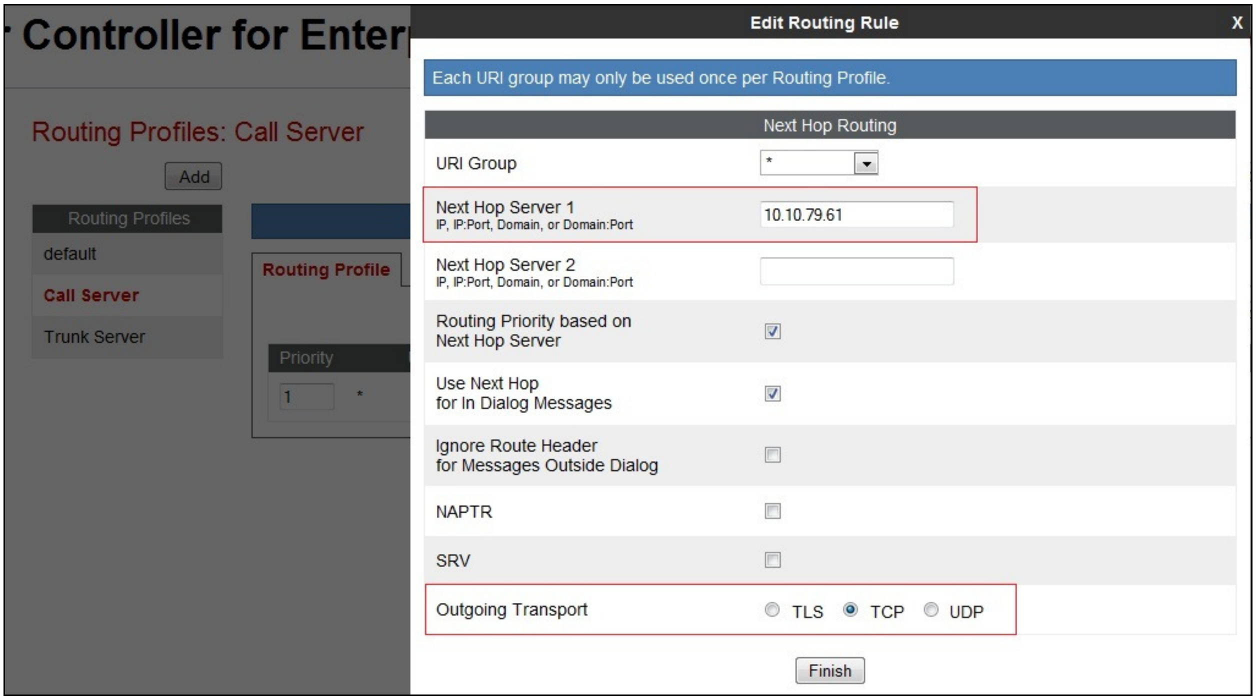
Edit Server Configuration Profile - Advanced

Enable DoS Protection	□
Enable Grooming	□
Interworking Profile	KPN
Signaling Manipulation Script	None
TCP Connection Type	● SUBID ○ PORTID ○ MAPPING

7.6. Define Routing

Routing information is required for routing to the Session Manager on the internal side and KPN VoIP Connect on the external side. The IP addresses and ports defined here will be used as the destination addresses for signalling. If no port is specified in the **Next Hop IP Address**, default 5060 is used. To define routing to the Session Manager, navigate to **Global Profiles → Routing** in the main menu on the left hand side (not shown). Click on **Add** and enter details in the **Routing Profile** pop-up menu.

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



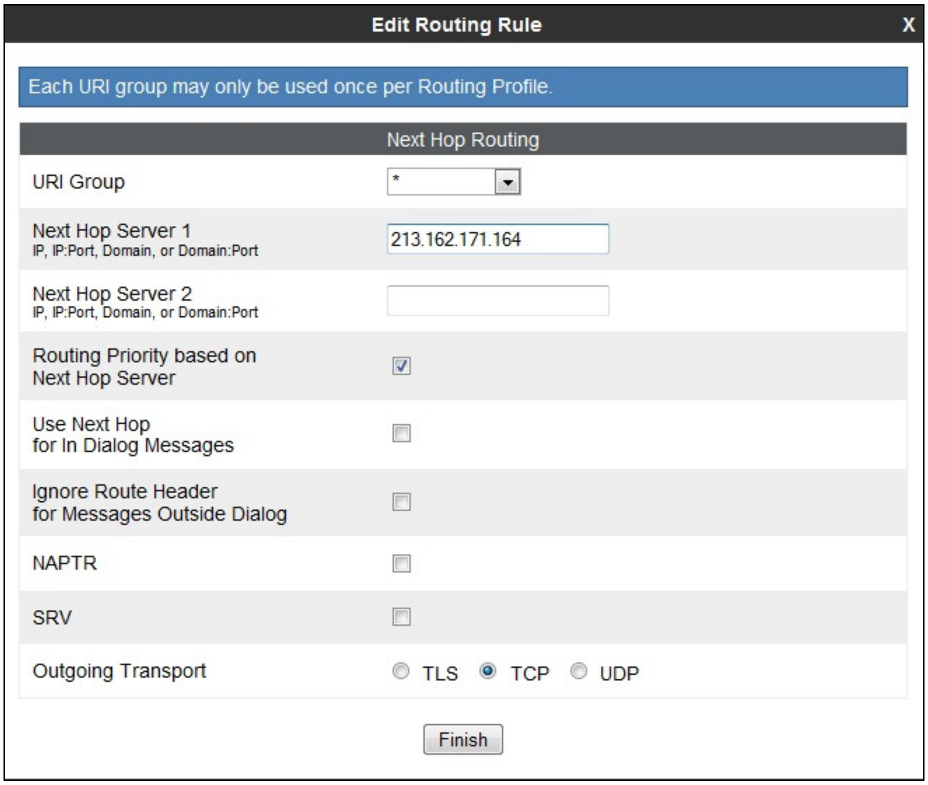
Edit Routing Rule

Each URI group may only be used once per Routing Profile.

Next Hop Routing	
URI Group	*
Next Hop Server 1 IP, IP:Port, Domain, or Domain:Port	10.10.79.61
Next Hop Server 2 IP, IP:Port, Domain, or Domain:Port	
Routing Priority based on Next Hop Server	☑
Use Next Hop for In Dialog Messages	☑
Ignore Route Header for Messages Outside Dialog	☐
NAPTR	☐
SRV	☐
Outgoing Transport	○ TLS ● TCP ○ UDP

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

- In the **Profile Name** field enter a descriptive name for KPN VoIP Connect, in this case a generic name of **Trunk Server** was used, and click **Next**
- Enter the KPN VoIP Connect IP address and port in the **Next Hop Server 1** field
- Select **TCP** for the **Outgoing Transport**
- Click **Finish**



Each URI group may only be used once per Routing Profile.

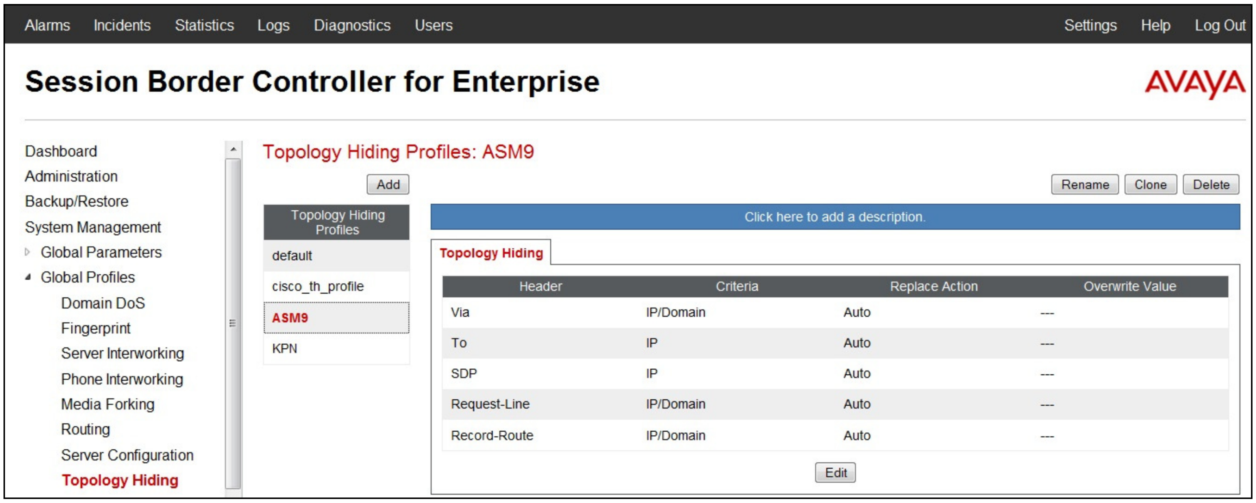
Next Hop Routing	
URI Group	*
Next Hop Server 1 IP, IP:Port, Domain, or Domain:Port	213.162.171.164
Next Hop Server 2 IP, IP:Port, Domain, or Domain:Port	
Routing Priority based on Next Hop Server	☑
Use Next Hop for In Dialog Messages	□
Ignore Route Header for Messages Outside Dialog	□
NAPTR	□
SRV	□
Outgoing Transport	○ TLS ● TCP ○ UDP

7.7. Topology Hiding

Topology hiding is used to hide local information such as private IP addresses and local domain names. The local information can be overwritten with a domain name or IP address. The default **Replace Action** is **Auto**; this replaces local information with IP addresses, generally the next hop. Topology hiding has the advantage of presenting single Via and Record-Route headers externally where multiple headers may be received from the enterprise, particularly from the Session Manager. In some cases where Topology Hiding can't be applied, in particular the Contact header, IP addresses are translated to the Avaya SBCE external addresses using NAT.

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

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

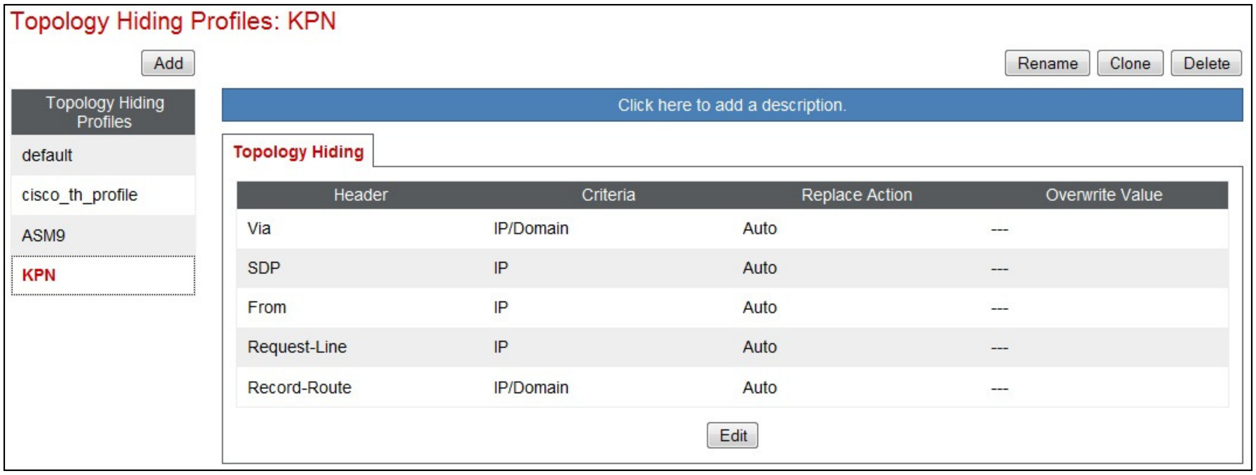


Header	Criteria	Replace Action	Overwrite Value
Via	IP/Domain	Auto	---
To	IP	Auto	---
SDP	IP	Auto	---
Request-Line	IP/Domain	Auto	---
Record-Route	IP/Domain	Auto	---

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

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

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



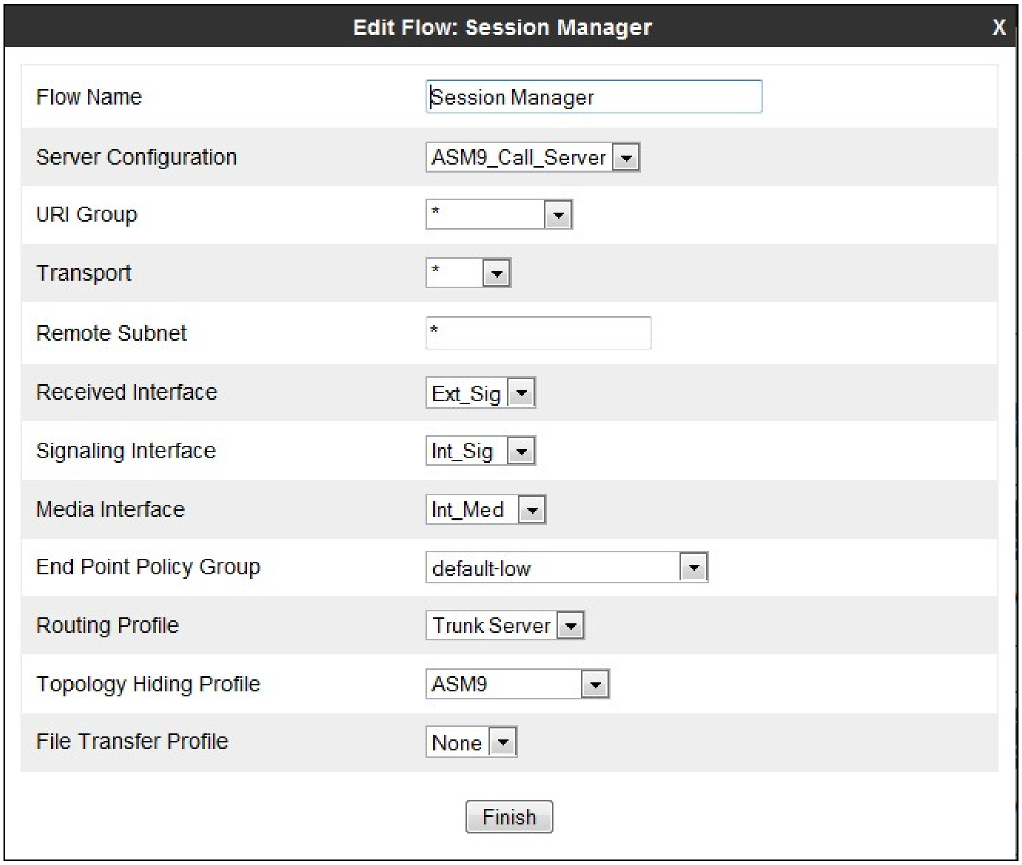
Header	Criteria	Replace Action	Overwrite Value
Via	IP/Domain	Auto	---
SDP	IP	Auto	---
From	IP	Auto	---
Request-Line	IP	Auto	---
Record-Route	IP/Domain	Auto	---

7.8. Server Flows

Server Flows combine the previously defined profiles into an outgoing flow from the Session Manager to KPN VoIP Connect and an incoming flow from KPN VoIP Connect to the Session Manager. This configuration ties all the previously entered information together so that signalling can be routed from the Session Manager to KPN VoIP Connect and vice versa.

To define a Server Flow for the Session Manager, navigate to **Device Specific Settings → End Point Flows** (not shown).

- Click on the **Server Flows** tab (not shown).
- Select **Add Flow** and enter details in the pop-up menu (not shown).
- In the **Flow Name** field enter a descriptive name for the server flow for the Session Manager; in this case **Session Manager** was used.
- In the **Server Configuration** drop down menu, select the Server defined in **Section 7.5** for the Session Manager
- In the **Received Interface** drop-down menu, select the external SIP signalling interface defined in **Section 7.3**. This is the interface that signalling bound for the Session Manager is received on.
- In the **Signaling Interface** drop-down menu, select the internal SIP signalling interface defined in **Section 7.3**. This is the interface that signalling bound for the Session Manager is sent on.
- In the **Media Interface** drop-down menu, select the internal media interface defined in **Section 7.3**. This is the interface that media bound for the Session Manager is sent on.
- In the **Routing Profile** drop-down menu, select the routing profile of KPN VoIP Connect defined in **Section 7.6**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of the Session Manager defined in **Section 7.7** and click **Finish**.

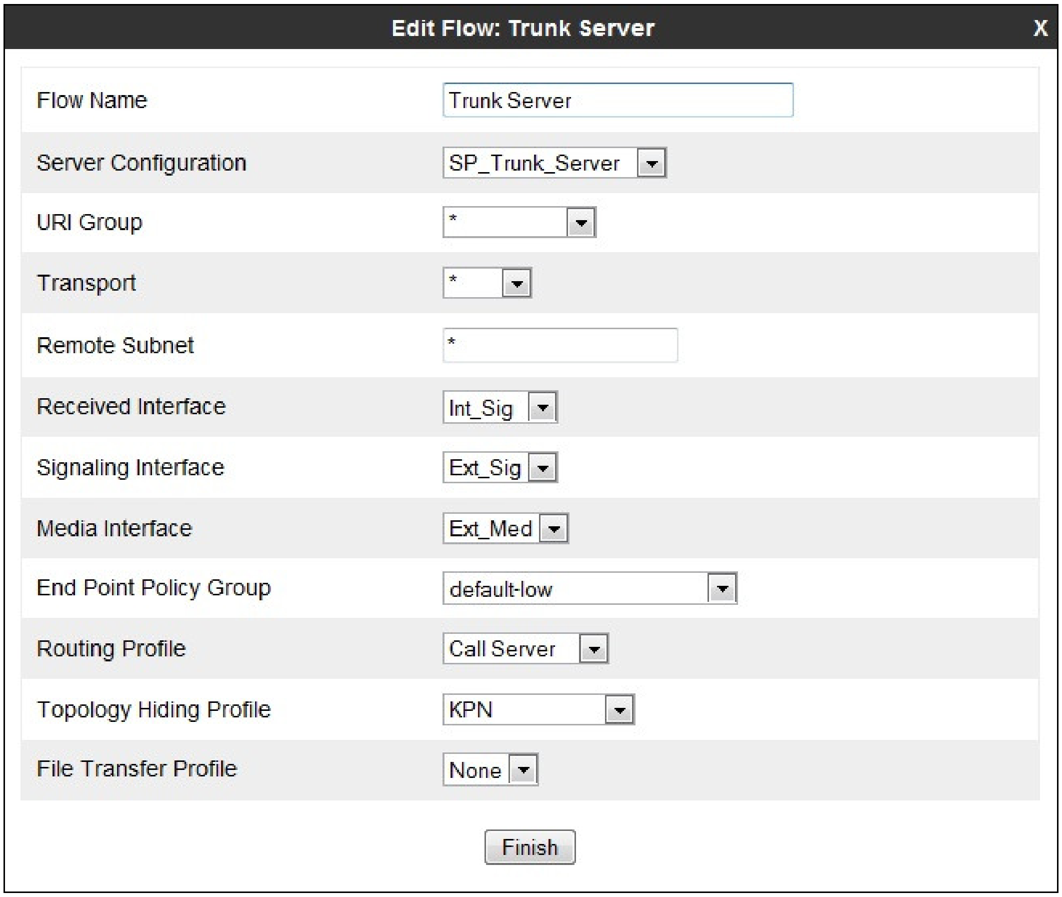


Edit Flow: Session Manager

Flow Name	Session Manager
Server Configuration	ASM9_Call_Server
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Ext_Sig
Signaling Interface	Int_Sig
Media Interface	Int_Med
End Point Policy Group	default-low
Routing Profile	Trunk Server
Topology Hiding Profile	ASM9
File Transfer Profile	None

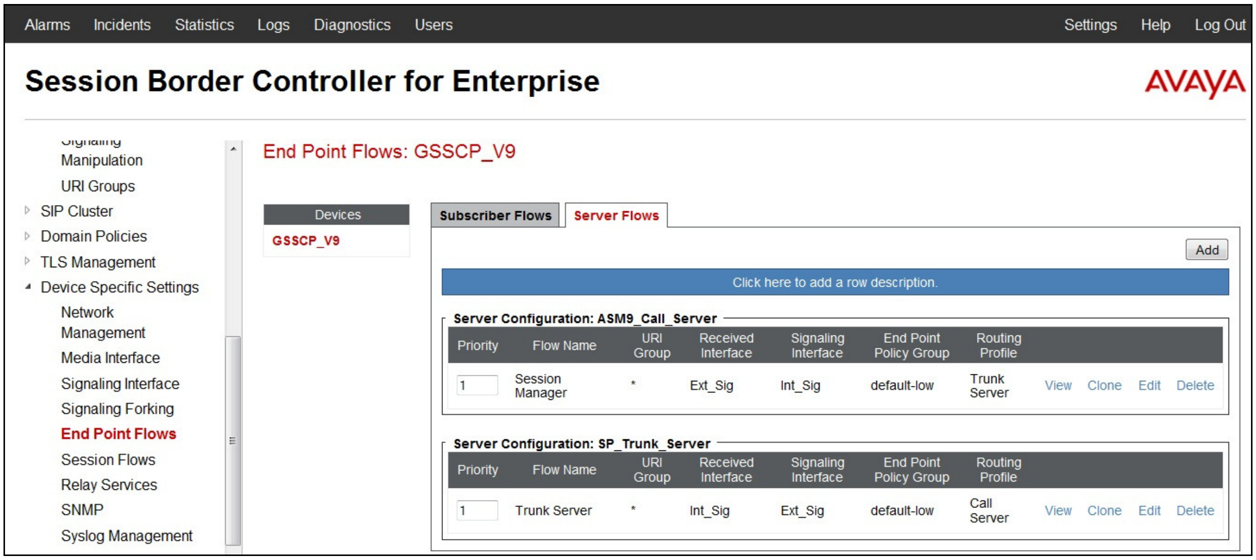
To define a Server Flow for KPN VoIP Connect, navigate to **Device Specific Settings → End Point Flows** (not shown).

- Click on the **Server Flows** tab (not shown).
- Select **Add Flow** and enter details in the pop-up menu (not shown).
- In the **Flow Name** field enter a descriptive name for the server flow for KPN VoIP Connect; in this case a generic name of **Trunk Server** was used.
- In the **Server Configuration** drop down menu, select the Server defined in **Section 7.5** for KPN VoIP Connect
- 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 KPN VoIP Connect 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 KPN VoIP Connect 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 KPN VoIP Connect is sent on.
- In the **Routing Profile** drop-down menu, select the routing profile of the Session Manager defined in **Section 7.6**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of the KPN VoIP Connect defined in **Section 7.7** and click **Finish.**



Flow Name	Trunk Server
Server Configuration	SP_Trunk_Server
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Int_Sig
Signaling Interface	Ext_Sig
Media Interface	Ext_Med
End Point Policy Group	default-low
Routing Profile	Call Server
Topology Hiding Profile	KPN
File Transfer Profile	None

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



Server Configuration: ASM9_Call_Server

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile				
1	Session Manager	*	Ext_Sig	Int_Sig	default-low	Trunk Server	View	Clone	Edit	Delete

Server Configuration: SP_Trunk_Server

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile				
1	Trunk Server	*	Int_Sig	Ext_Sig	default-low	Call Server	View	Clone	Edit	Delete

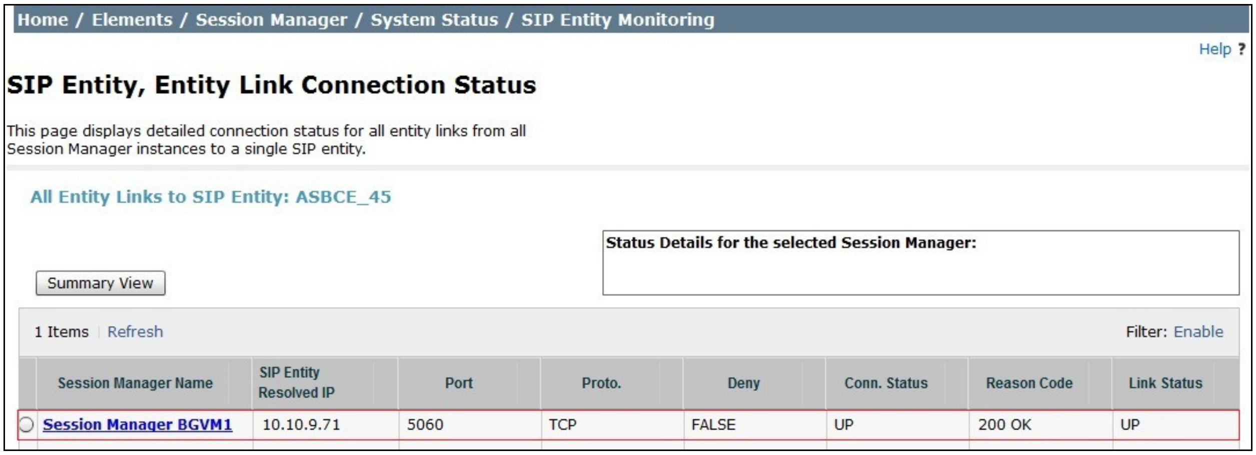
8. Configure KPN VoIP Connect Equipment

The configuration of the KPN equipment used to support the KPN VaMo1 VoIP Connect Service is outside of the scope of these Application Notes and will not be covered. To obtain further information on KPN equipment and system configuration please contact an authorised KPN representative.

9. Verification Steps

This section provides steps that may be performed to verify that the solution is configured correctly.

1. From System Manager **Home** tab click on **Session Manager** and navigate to **Session Manager → System Status → SIP Entity Monitoring**. Select the relevant SIP Entity from the list and observe if the **Conn Status** and **Link Status** are showing as **up**.



Home / Elements / Session Manager / System Status / SIP Entity Monitoring
Help ?

SIP Entity, Entity Link Connection Status

This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.

All Entity Links to SIP Entity: ASBCE_45

Status Details for the selected Session Manager:

Summary View
1 Items | Refresh
Filter: Enable

	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
○	Session Manager BGVM1	10.10.9.71	5060	TCP	FALSE	UP	200 OK	UP

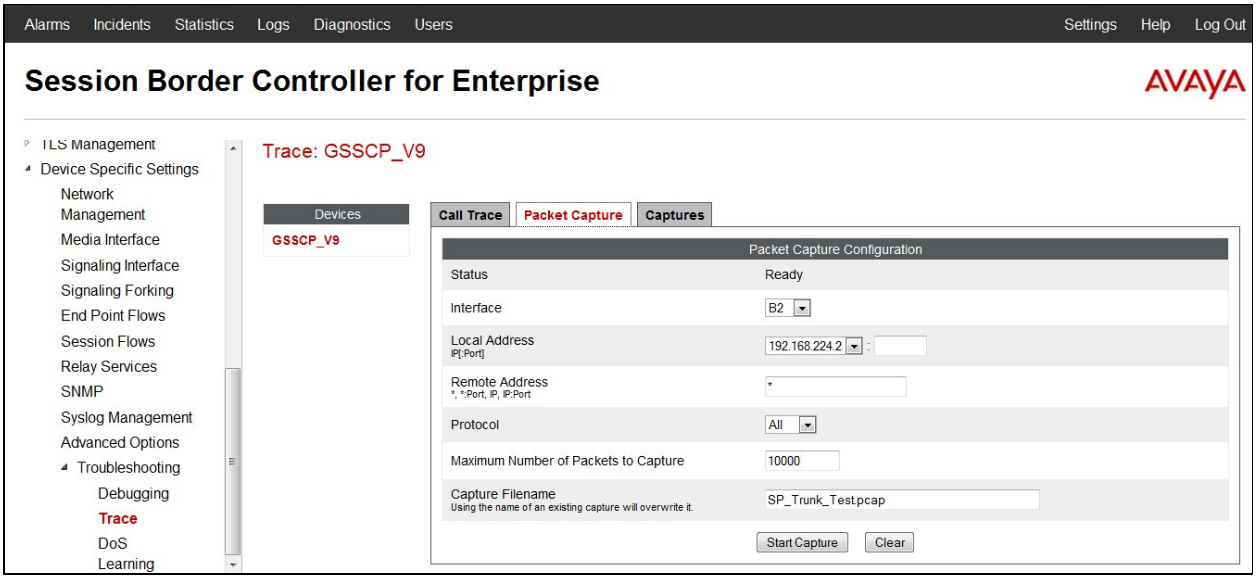
2. From the Communication Manager SAT interface run the command **status trunk n** where **n** is a previously configured SIP trunk. Observe if all channels on the trunk group display **in-service/idle**.

```
status trunk 1

                              TRUNK GROUP STATUS

Member    Port     Service State      Mtce Connected Ports
                                      Busy

0001/001 T00001   in-service/idle    no
0001/002 T00002   in-service/idle    no
0001/003 T00003   in-service/idle    no
0001/004 T00004   in-service/idle    no
0001/005 T00005   in-service/idle    no
0001/006 T00006   in-service/idle    no
0001/007 T00007   in-service/idle    no
0001/008 T00008   in-service/idle    no
0001/009 T00009   in-service/idle    no
0001/010 T00010   in-service/idle    no
```

3. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active.
4. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active.
5. Verify that the user on the PSTN can end an active call by hanging up.
6. Verify that an endpoint at the enterprise site can end an active call by hanging up.

7. Should issues arise with the SIP trunk, use the Avaya SBCE trace facility to check that the OPTIONS requests sent from the Session Manager via the Avaya SBCE to KPN VoIP Connect are receiving a response.

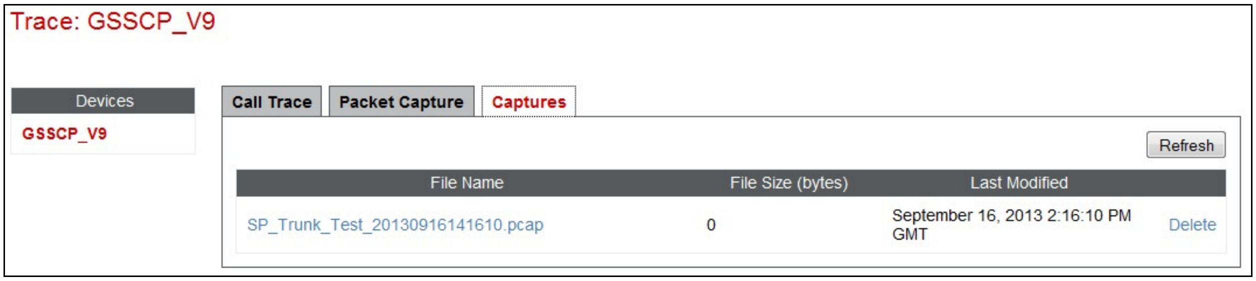
   To define the trace, navigate to **Device Specific Settings → Advanced Options → Troubleshooting → Trace** in the main menu on the left hand side and select the **Packet Capture** tab.
   - Select the SIP Trunk interface from the **Interface** drop down menu
   - Select the signalling interface IP address from the **Local Address** drop down menu
   - Enter the IP address of KPN VoIP Connect 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**



Trace: GSSCP_V9

Packet Capture Configuration	
Status	Ready
Interface	B2
Local Address IP[:Port]	192.168.224.2 :
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.	SP_Trunk_Test.pcap

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



Trace: GSSCP_V9

File Name	File Size (bytes)	Last Modified	
SP_Trunk_Test_20130916141610.pcap	0	September 16, 2013 2:16:10 PM GMT	Delete

The trace is viewed as a standard pcap file in Wireshark. If the SIP trunk is working correctly, a SIP response in the form of a 200 OK will be seen from KPN VoIP Connect.

10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Communication Manager R6.3 as an Evolution Server, Avaya Aura® Session Manager R6.3 and Avaya Session Border Controller for Enterprise to KPN VaMo1 VoIP Connect Service. KPN VaMo1 VoIP Connect Service is a SIP-based Voice over IP solution providing businesses a flexible, cost-saving alternative to traditional hardwired telephony trunks. The service was successfully tested with a number of observations listed in **Section 2.2**.

11. Additional References

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

[1] *Installing and Configuring Avaya Aura® System Platform*, Release 6.3, May 2013.

[2] *Administering Avaya Aura® System Platform*, Release 6.3, May 2013.

[3] *Avaya Aura® Communication Manager using VMware® in the Virtualized Environment Deployment Guide*, May 2013

[4] *Avaya Aura® Communication Manager 6.3 Documentation library*, August 2013.

[5] *Avaya Aura® System Manager using VMware® in the Virtualized Environment Deployment Guide* Release 6.3 May 2013

[6] *Implementing Avaya Aura® System Manager* Release 6.3, May 2013

[7] *Upgrading Avaya Aura® System Manager to 6.3.2*, May 2013.

[8] *Administering Avaya Aura® System Manager* Release 6.3, May 2013

[9] *Avaya Aura® Session Manager using VMware® in the Virtualized Environment Deployment Guide* Release 6.3 May 2013

[10] *Implementing Avaya Aura® Session Manager* Release 6.3, May 2013

[11] *Upgrading Avaya Aura® Session Manager* Release 6.3, May 2013

[12] *Administering Avaya Aura® Session Manager* Release 6.3, June 2013,

[13] *Installing Avaya Session Border Controller for Enterprise*, Release 6.2 June 2013

[14] *Upgrading Avaya Session Border Controller for Enterprise* Release 6.2 July 2013

[15] *Administering Avaya Session Border Controller for Enterprise* Release 6.2 March 2013

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

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.