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 Zen Internet SIP Trunk Service - Issue 1.0

Abstract

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between Zen Internet SIP Trunk 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. Zen Internet is a member of the DevConnect Service Provider program.

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

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between Zen Internet's SIP Trunk 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. Customers using this Avaya SIP-enabled enterprise solution with Zen Internet SIP Trunk are able to place and receive PSTN calls via a dedicated Internet connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks. This approach generally results in lower cost for the enterprise customer.

2. General Test Approach and Test Results

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

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 Zen Internet, calls made to SIP, H.323, Digital and Analogue telephones at the enterprise.
- All inbound PSTN calls were routed to the enterprise across the SIP trunk to Zen Internet.
- Outgoing calls from the enterprise site completed via Zen Internet's SIP Trunk to PSTN destinations, calls made from SIP, H.323, Digital and Analogue telephones.
- All outbound PSTN calls were routed from the enterprise across the SIP trunk to Zen Internet.
- Inbound and outbound PSTN calls to/from Avaya One-X Communicator and Avaya Flare Experience for Windows softphones.
- Calls using G.711A and G.711MU codecs.
- Fax calls to/from a group 3 fax machine to a PSTN-connected fax machine using G711A.
- Calling Line Identification Presentation and Calling Line Identification Restriction.
- DTMF transmission using RFC 2833.
- Voice Mail/Vector navigation for inbound and outbound calls.
- User features such as hold and resume, transfer and conference.
- Call coverage and call forwarding for endpoints at the enterprise site.
- Off-net call forwarding and EC500 mobile twinning.

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for Zen Internet's SIP Trunk Service with the following observations:

- Zen Internet require the From Header to be populated with the Calling Line Identification CLID of the number that is re-directing the call or a known number on the PABX when performing call re-direction or EC500 calls. This was performed by creating a SigMa Script on the Avaya SBCE that copied the CLID details in the Diversion Header and populated it in the From Header. The details of the Sigma Script are outlined in **Section 7.2.7**.
- When performing an outbound CLID restricted call, Zen Internet require the From Header to be populated with the CLID of the set that is making the outbound call or of a known CLID on the PABX. When CLID restriction is enabled on Communication Manager, Communication Manager populates the From Header with Anonymous. In order for outbound CLID restricted call to terminate successfully, a SigMa Script was created on the Avaya SBCE to copy the CLID from the P-Asserted-Identity Header and populate it in the From Header. The details of the Sigma Script are outlined in **Section 7.2.7**.
- T.38 fax transmission is not supported by Zen Internet.
- Inbound and Outbound fax was tested successfully using G.711 pass-through. This is not a method supported by Avaya.
- No inbound toll free numbers were tested, however routing of inbound DDI numbers and the relevant number translation was successfully tested.
- Access to Emergency Services was not tested as no test call had been booked with the Emergency Services Operator.

2.3. Support

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

For technical support on Zen Internet products please contact the Zen Internet authorized representative at:

- Telephone: +44 1706 902000
- Website: http://zen.co.uk/support/phone-services.aspx

3. Reference Configuration

The following equipment in **Figure 1** illustrates the test configuration. The test configuration shows an Enterprise site connected to Zen Internet's SIP Trunk. 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 16xx series IP telephones (with H.323 firmware), Avaya A175 Desktop Video Device running Avaya 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 Avaya Flare® Experience for Windows running on a laptop PC.

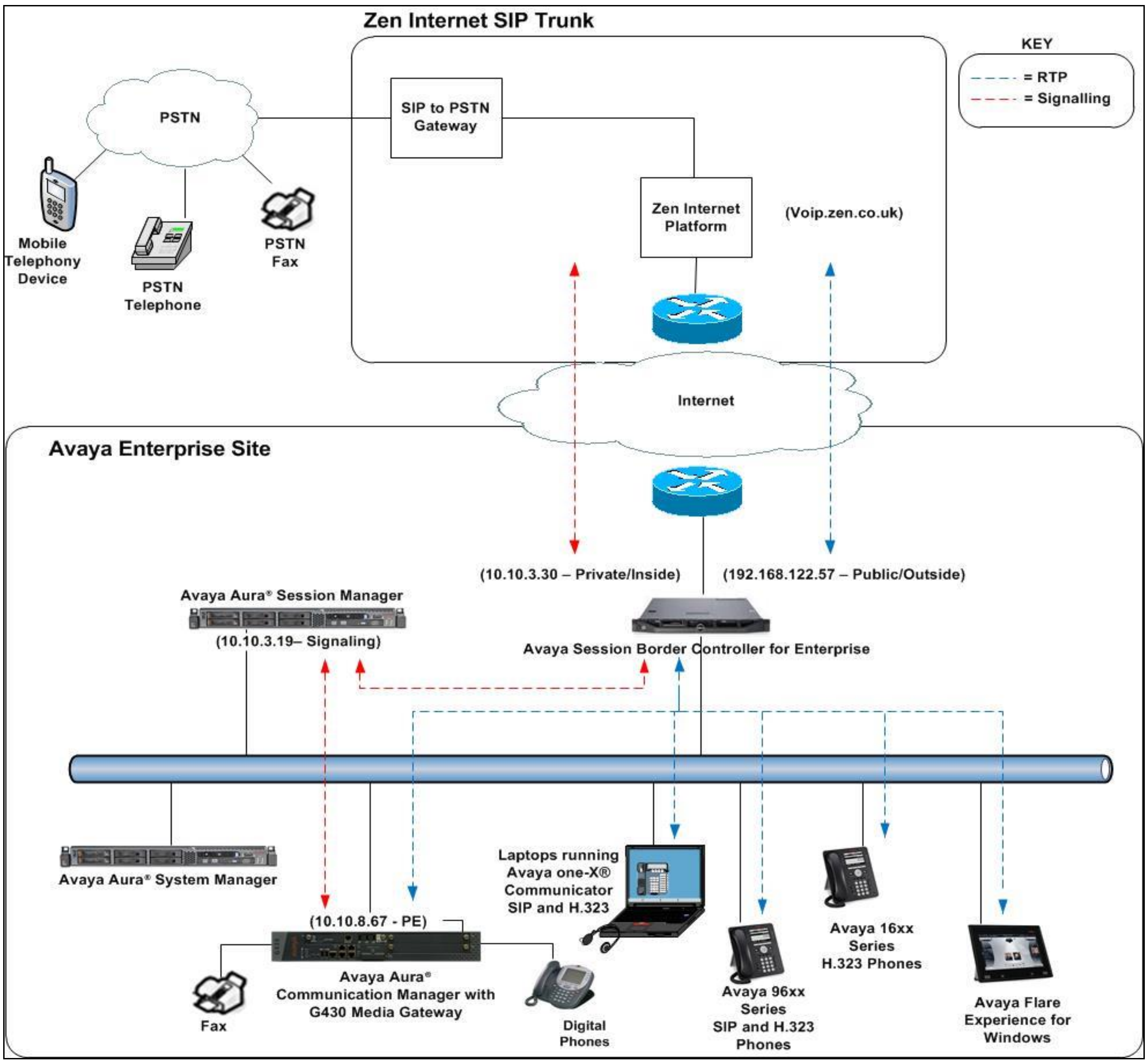


Figure 1: Test Setup Zen Internet SIP Trunk to Avaya Enterprise

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya	
Dell PowerEdge R620 running Session Manager on VM Version 8	R6.3.11 - 6.3.11.0.631103
Dell PowerEdge R620 running System Manager on VM Version 8	R6.3.11 - Build No. - 6.3.0.8.5682-6.3.8.4411 Software Update Revision No: 6.3.11.8.1.2871
Avaya S8800 Server running Communication Manager	R016x.03.0.124.0-21754
Avaya Session Border Controller for Enterprise	6.2.1.Q18
Avaya 16xx IP DeskPhone (H.323)	1.3
Avaya 9670 IP DeskPhone (H.323)	6.4
Avaya 96x0 IP DeskPhone (H.323)	6.4
Avaya 96x1 IP DeskPhone (H.323)	6.4
Avaya 96x0 IP DeskPhone (SIP)	6.4.1
Avaya 96x1 IP DeskPhone (SIP)	6.4.1
Avaya A175 Desktop Video Device with Avaya Flare® Experience	1.1.3
Avaya one–X® Communicator (H.323) on Lenovo T510 Laptop PC	6.2.4.07-FP4
Avaya Flare Experience for Windows	1.1.4.23
Avaya Digital Handset	Rel 12.0
Analogue Handset	N/A
Zen Internet	
Zen Internet SIP Trunk	Asterisk – 1.8.13.1 Proxy – 1.5.1

5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring Communication Manager for SIP Trunking. SIP trunks are established between Communication Manager and Session Manager. These SIP trunks will carry SIP signalling associated with the Zen Internet SIP Trunk. For incoming calls, 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 Session Manager. Session Manager directs the outbound SIP messages to the Avaya SBCE at the enterprise site that then sends the

SIP messages to the Zen Internet 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 Zen Internet SIP Trunk 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 Session Manager. In this case, **SM100** and **10.10.3.19** are the **Name** and **IP Address** for 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
SM100               10.10.3.19
default             0.0.0.0
procr               10.10.8.67
procr6              ::
```

5.3. Administer IP Network Region

Use the **change ip-network-region x** command where x is the desired network-region 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 Zen Internet was configured, namely **G.711A** and **G.711MU**.

```
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: G.711MU           n           2        20
```

Zen Internet SIP Trunk supports pass-through for transmission of fax. Navigate to **Page 2** and define fax properties as follows:

- Set the **FAX - Mode** to **pass-through**
- Leave **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             pass-through              0             ECM y
    Modem           off                       0
    TDD/TTY         US                        3
    Clear-channel   n                         0
```

5.5. Administer SIP Signaling Groups

This signalling group (and trunk group) will be used for inbound and outbound PSTN calls to the Zen Internet SIP Trunk 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 Session Manager (node name **SM100** 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**
- Set **Initial IP-IP Direct Media** to **n**
- 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: SM100
 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 signalling 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 Zen Internet 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

             XOIP Treatment: auto    Delay Call Setup When Accessed Via IGAR? n
```

On **Page 3**, set the **Numbering Format** field to **private**. This allows delivery of CLID in national formats.

```
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 **Mark Users as Phone** to **y**
- Set **Send Transferring Party Information** to **n**
- Set **Network Call Direction** to **n**
- Set **Send Diversion Header** to **y**
- Set **Support Request History** to **n**
- Set the **Telephone Event Payload Type** to **101** to match the value preferred by Zen Internet
- Set **Always Use re-INVITE for Display Updates** to **y**
- Set the **Identity for Calling Party Display** to **P-Asserted-Identity**

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

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

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


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

5.7. Administer Calling Party Number Information

Use the **change private-numbering x** command to configure Communication Manager to send the calling party number in the format required. 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  60              1          0333nnnnn98       11    Total Administered: 2
 4  61              1          0333nnnnn98       11    Maximum Entries: 540
```

Note: The above configuration accepts all **4** digit numbers starting with **6**, which includes all SIP and H.323 extension numbers.

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 Zen Internet's SIP Trunk. 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    10     1       pubu         n
    0900                  8    8      1       pubu         n
    118                   3    6      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 CLID 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
     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
```

5.9. Administer Incoming Digit Translation

This step configures the settings necessary to map incoming DDI calls to the proper Communication Manager extension(s). The incoming digits sent in the INVITE message from Zen Internet can be manipulated as necessary to route calls to the desired extension. In the examples used in the compliance testing, the incoming DID numbers provided by Zen Internet correlate to the internal extensions assigned within Communication Manager. The entries displayed below translates incoming DID numbers **0333xxxxxxx** to a 4 digit extension by deleting all of the incoming digits and inserting an extension. Public DID numbers have been masked for security purposes.

```
change inc-call-handling-trmt trunk-group 1                     Page   1 of   3
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 public-ntwrk    10 0333nnnnnn10       all 6010
 public-ntwrk    10 0333nnnnnn11       all 6012
 public-ntwrk    10 0333nnnnnn12       all 6102
```

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 6102. 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. **089434nnnn**)
- Set the **Trunk Selection** to **ARS**
- Set the **Config Set** to **1**

```
change off-pbx-telephone station-mapping 6102                  Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 6102             EC500         -     089434nnnn      ARS          1
-
```

Note: The phone number shown is for a mobile phone used for testing at Avaya Labs and is in national format with national dialling prefix 0. 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.

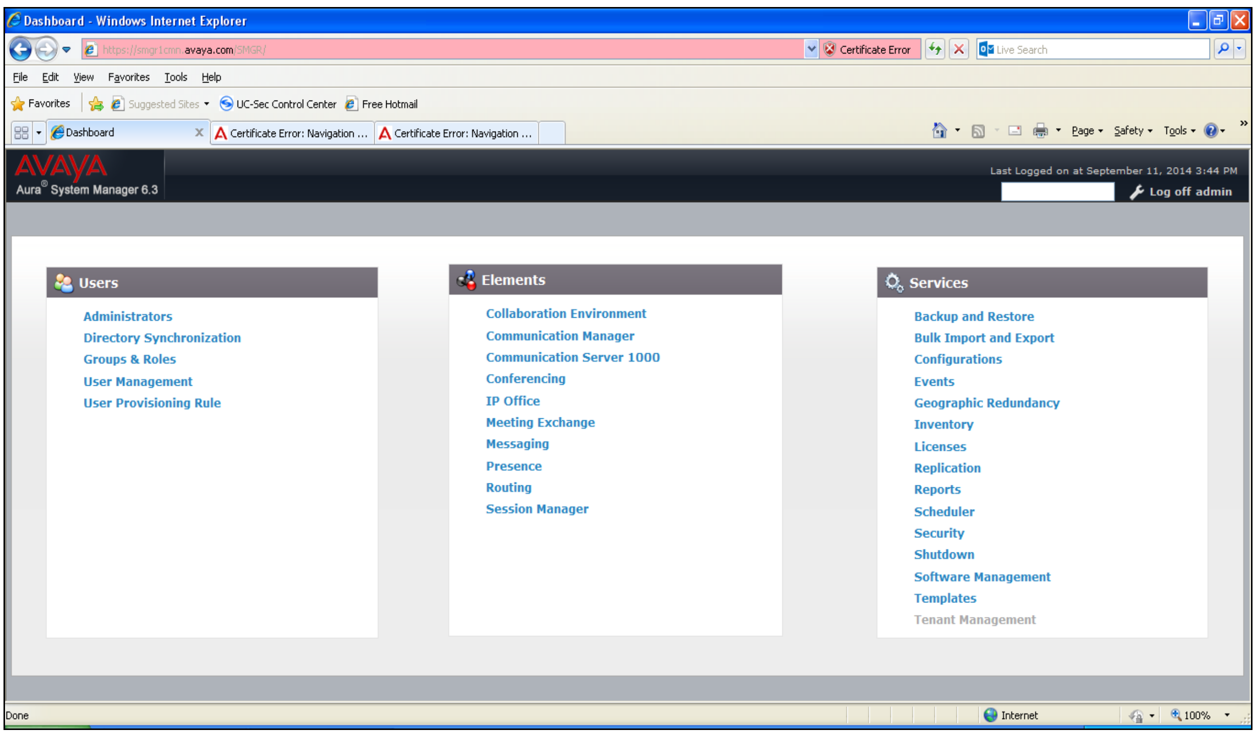
6. Configuring Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. 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

6.1. Log in to Avaya Aura® System Manager

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

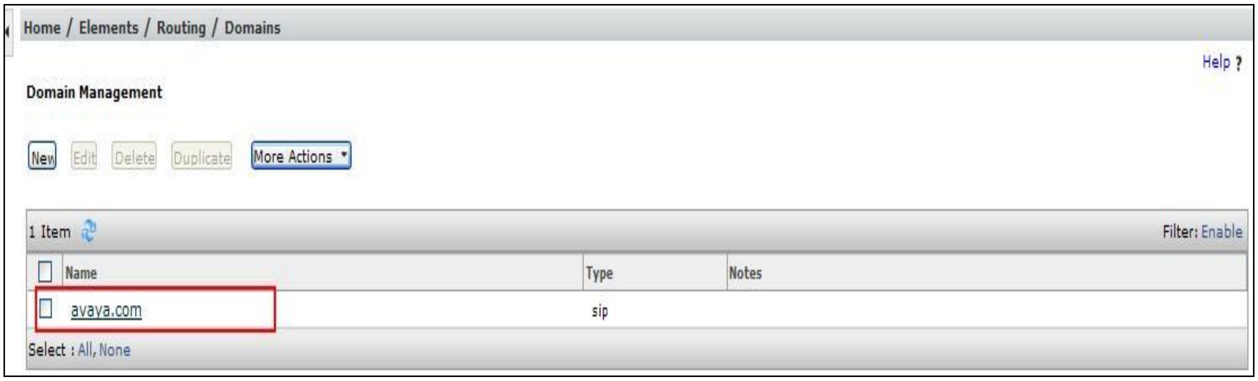


6.2. Administer SIP Domain

Create a SIP domain for each domain for which Session Manager will need to be aware in order to route calls. Expand **Elements → Routing** and select **Domains** from the left navigation menu, click **New** (not shown)**.** Enter the following values and use default values for remaining fields.

- **Name** Enter a Domain Name**.** In the sample configuration, **avaya.com** was used.
- **Type** Verify **SIP** is selected.
- **Notes** Add a brief description [Optional].

Click **Commit** to save. The screen below shows the SIP Domain defined for the sample configuration.



6.3. Administer Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management and call admission control. To add a location, navigate to **Routing →Locations** in the left-hand navigation pane and click the **New** button in the right pane (not shown). In the **General** section, enter the following values. Use default values for all remaining fields:

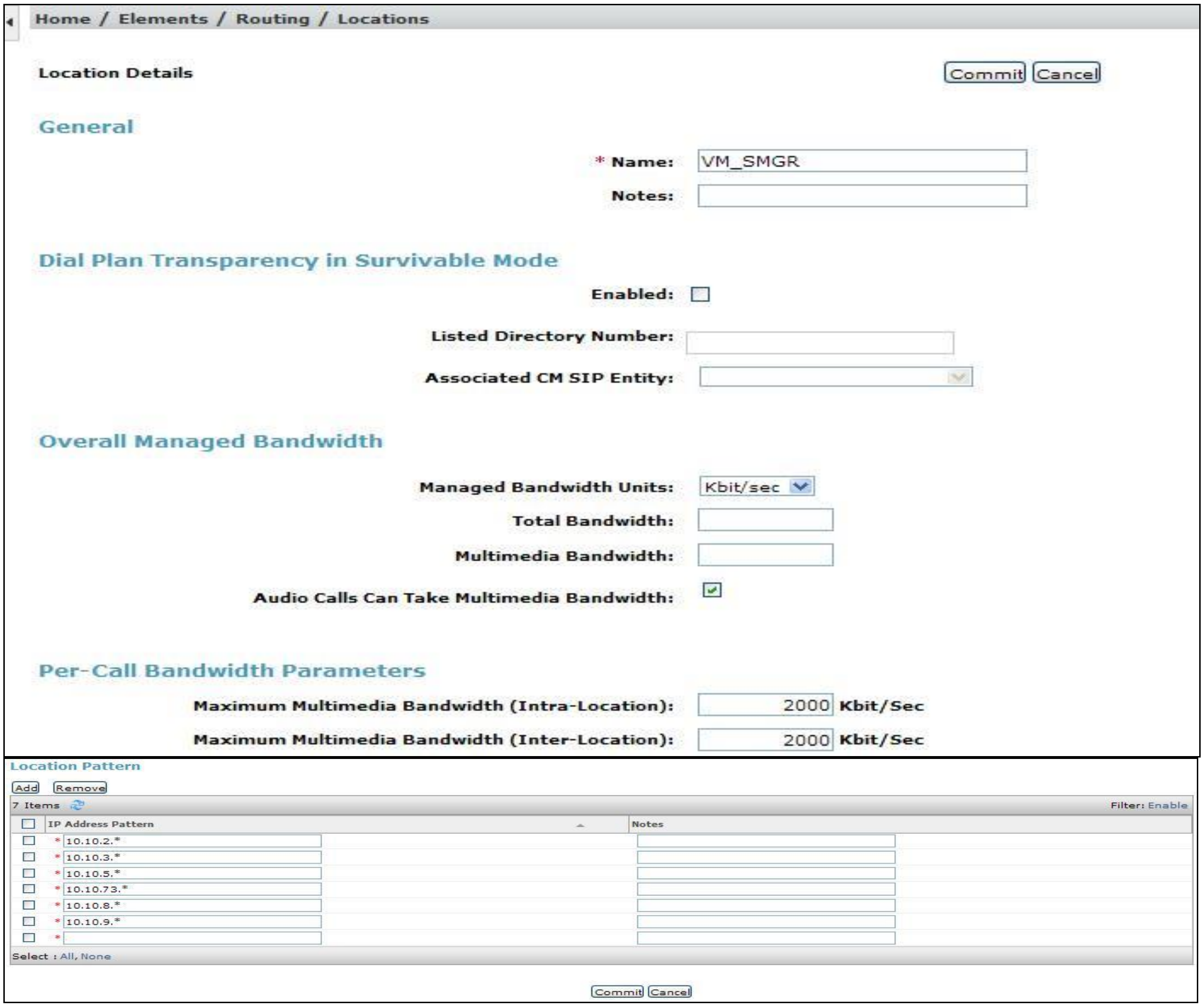
- **Name:** Enter a descriptive name for the location.
- **Notes:** Add a brief description (optional).

The Location Pattern is used to identify call routing based on IP address. Session Manager matches the IP address against the patterns defined in this section. If a call is from a SIP Entity that does not match the IP address pattern then Session Manager uses the location administered for the SIP Entity.

In the **Location Pattern** section, click **Add** and enter the following values.

- **IP Address Pattern** Enter the logical pattern used to identify the location.
- **Notes** Add a brief description [Optional].

Click **Commit** to save. The screenshot below shows the Location **VM_SMGR** defined for the compliance testing.



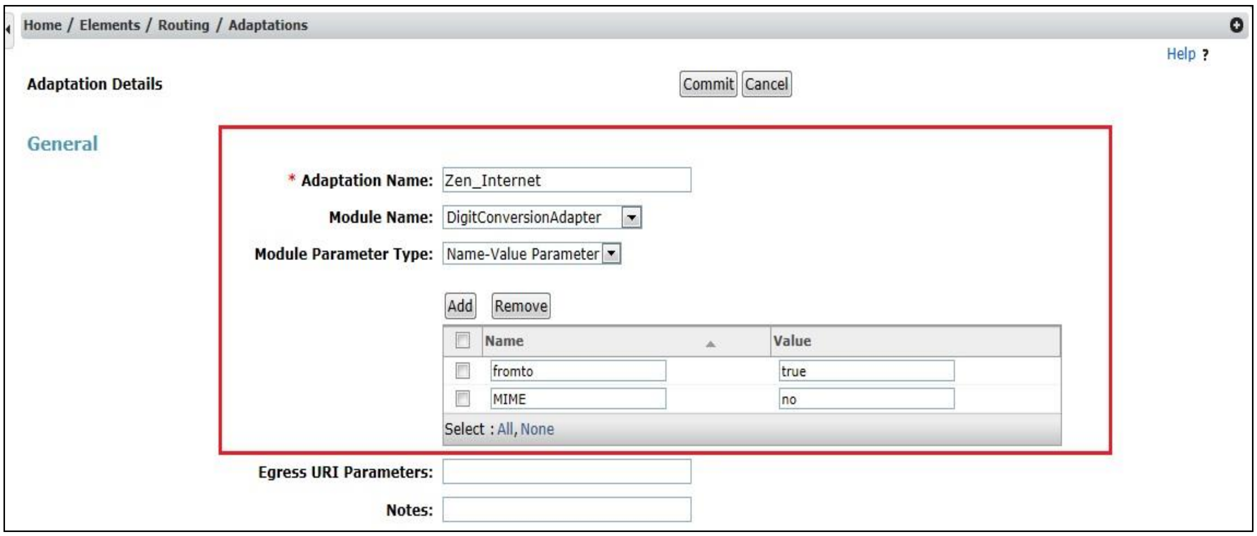
Home / Elements / Routing / Locations
Location Details
Commit
Cancel
General
* Name:
VM_SMGR
Notes:
Dial Plan Transparency in Survivable Mode
Enabled:
Listed Directory Number:
Associated CM SIP Entity:
Overall Managed Bandwidth
Managed Bandwidth Units:
Kbit/sec
Total Bandwidth:
Multimedia Bandwidth:
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
Location Pattern
Add
Remove
7 Items
Filter: Enable
IP Address Pattern
Notes
10.10.2.*
10.10.3.*
10.10.5.*
10.10.73.*
10.10.8.*
10.10.9.*
Select : All, None
Commit
Cancel

6.4. Administer Adaptations

Adaptations can be used to modify the called and calling party numbers to meet the requirements of the service. The called party number present in the SIP INVITE Request URI is modified by the **Digit Conversion** in the Adaptation. The example below was applied to the Avaya SBCE SIP Entity and was used in test to convert numbers being passed between the Avaya SBCE and Session Manager.

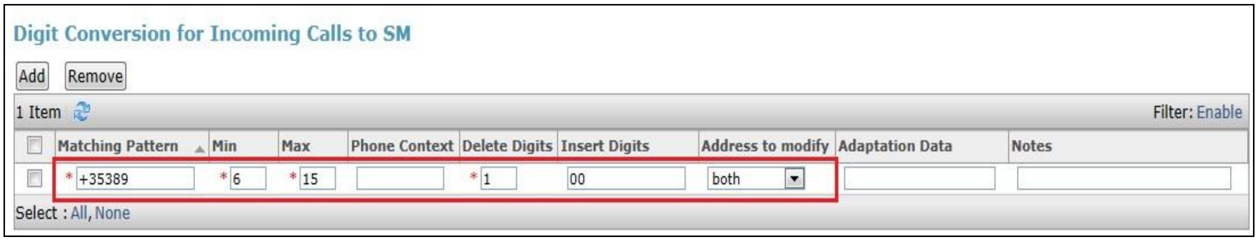
To add an adaptation, under the **Routing** tab select **Adaptations** on the left hand menu and then click on the **New** button (not shown). Under **Adaption Details →General**:

- In the **Adaptation name** field enter an informative name.
- In the **Module name** field click on the down arrow and then select the <**click to add module**> entry from the drop down list and type **DigitConversionAdapter** in the resulting New Module Name field.
- **Module Parameter Type** **MIME =no** Strips MIME message bodies on egress from Session Manager
  fromto=true Modifies from and to headers of a message



Scroll down the page and under **Digit Conversion for Incoming Calls to SM**, click the **Add** button and specify the digit manipulation to be performed as follows:

- Enter the leading digits that will be matched in the Matching Pattern field.
- In the **Min** and **Max** fields set the minimum and maximum digits allowed in the digit string to be matched.
- In the **Delete Digits** field enter the number of leading digits to be removed.
- In the **Insert Digits** field specify the digits to be prefixed to the digit string.
- In the **Address to modify** field specify the digits to manipulate by the adaptation. In this configuration the dialled number is the target so **both** have been selected.



This will ensure any incoming numbers will have the + 353 digits removed and 00 digit inserted before being presented to Communication Manager.

6.5. Administer SIP Entities

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

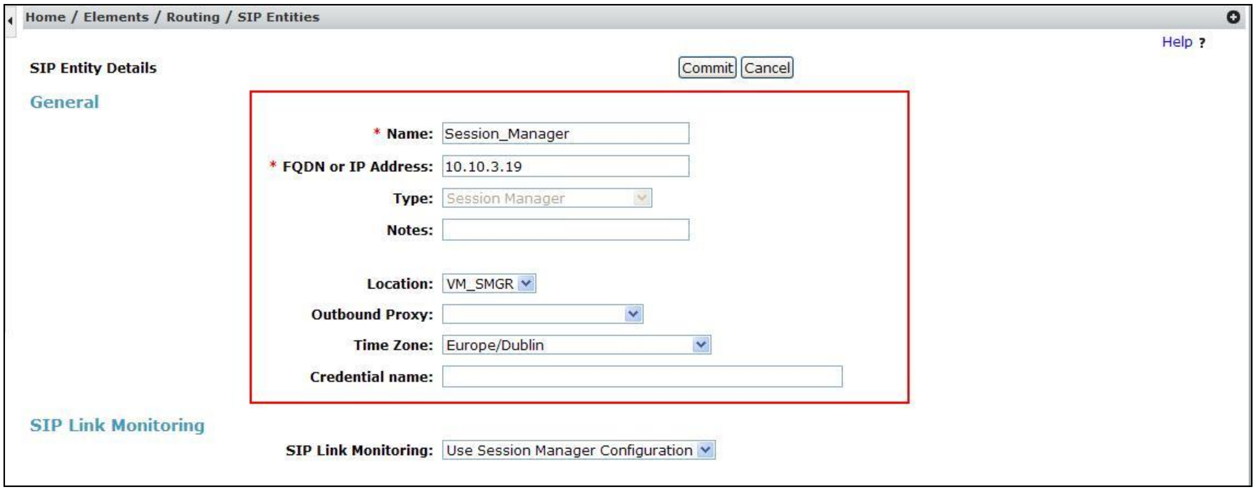
- In the **Name** field enter an informative name
- In the **FQDN or IP Address** field enter the IP address of Session Manager or the signalling interface on the connecting system
- In the **Type** field use **Session Manager** for a Session Manager SIP Entity, **CM** for a Communication Manager SIP Entity and **SIP Trunk** for the Avaya SBCE SIP Entity
- In the **Location** field select the appropriate location from the drop down menu
- In the **Time Zone** field enter the time zone for the SIP Entity

In this configuration there are four SIP Entities.

- Session Manager SIP Entity
- Communication Manager SIP Entity
- 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 Session Manager SIP signalling interface and **Type** is **Session Manager**. Set the **Location** to that defined in **Section 6.3** and the **Time Zone** to the appropriate time.

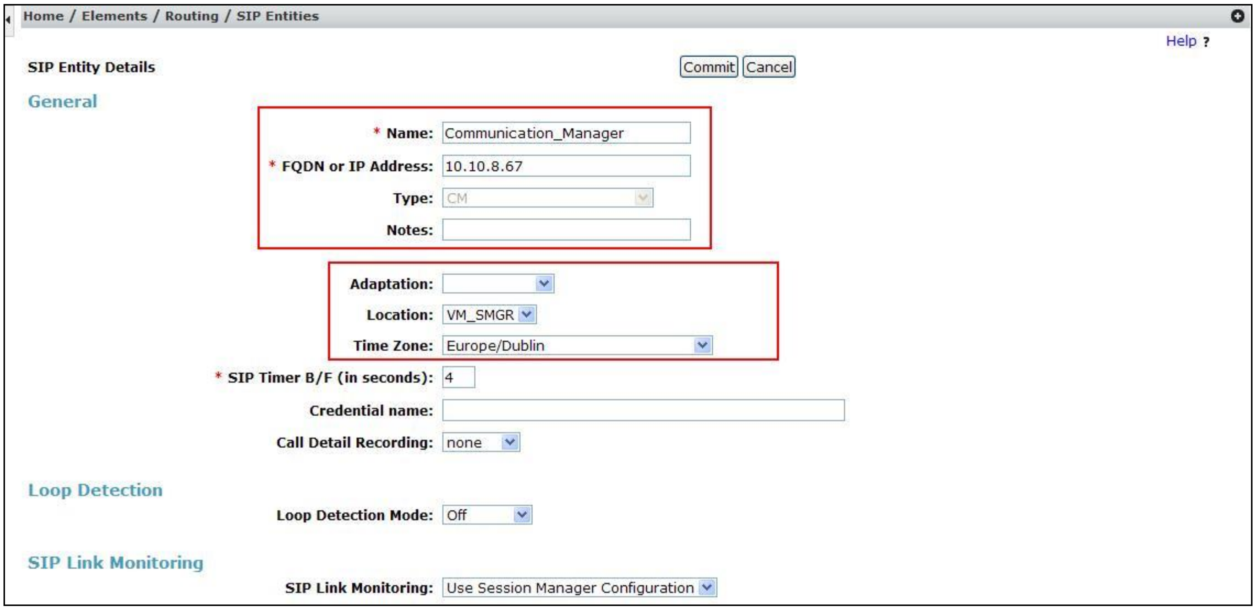


Session Manager must be configured with the port numbers on the protocols that will be used by the other SIP entities. To configure these scroll to the bottom of the page and under **Port**, click **Add**, then edit the fields in the resulting new row.

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

6.5.2. Avaya Aura® Communication Manager SIP Entity

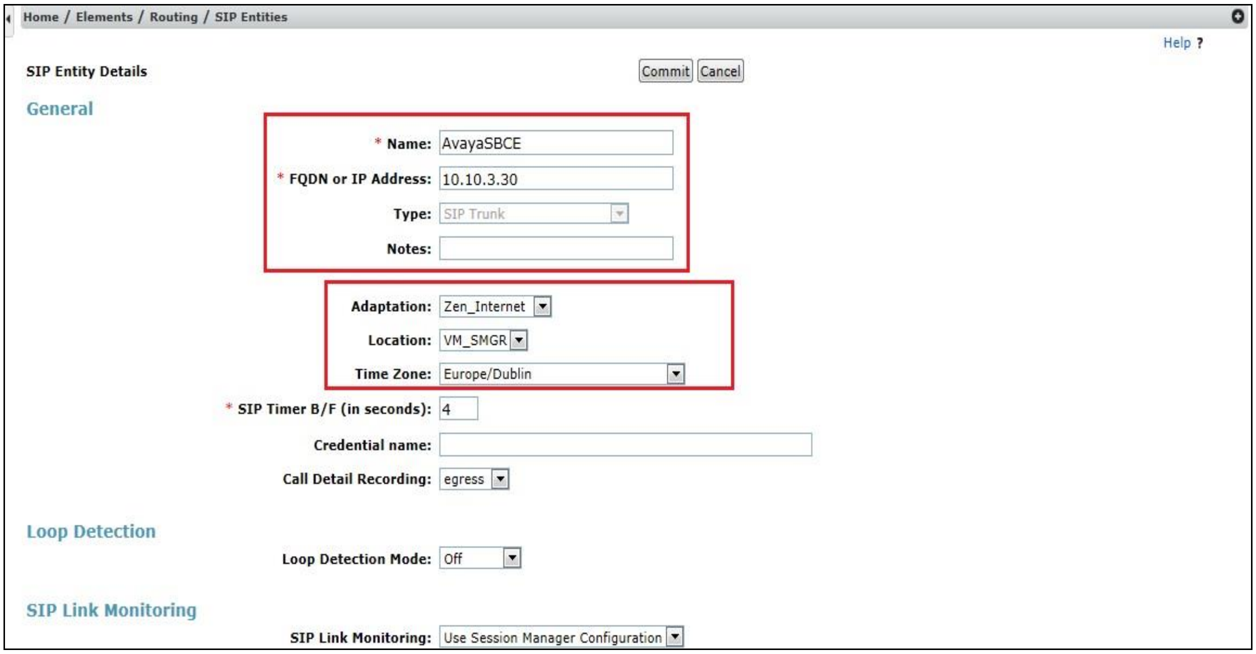
The following screen shows the SIP entity for Communication Manager which is configured as an Evolution Server. The **FQDN or IP Address** field is set to the IP address of the interface on Communication Manager that will be providing SIP signalling and **Type** is **CM**. Set the **Location** to that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



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

6.5.3. Avaya Session Border Controller for Enterprise SIP Entity

The following screen shows the SIP Entity for the Avaya SBCE. The **FQDN or IP Address** field is set to the IP address of the Avaya SBCE private network interface (see **Figure 1**). Set **Type** to **SIP Trunk** and **Adaptation** to that defined in **Section 6.4**. Set the **Location** to that defined in **Section 6.3** and the **Time Zone** to the appropriate time zone.

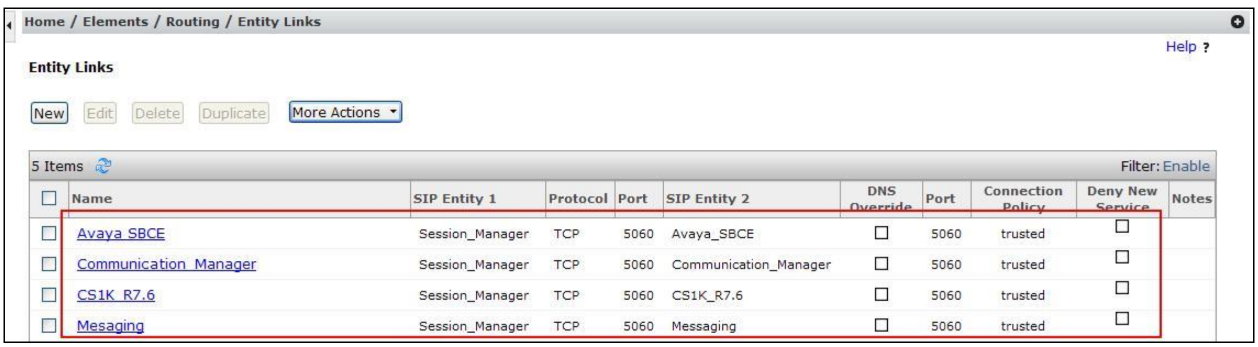


6.6. Administer Entity Links

A SIP trunk between a Session Manager and another system is described by an Entity Link. To add an Entity Link, select **Entity Links** on the left panel menu and click on the **New** button (not shown). Fill in the following fields in the new row that is displayed.

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

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



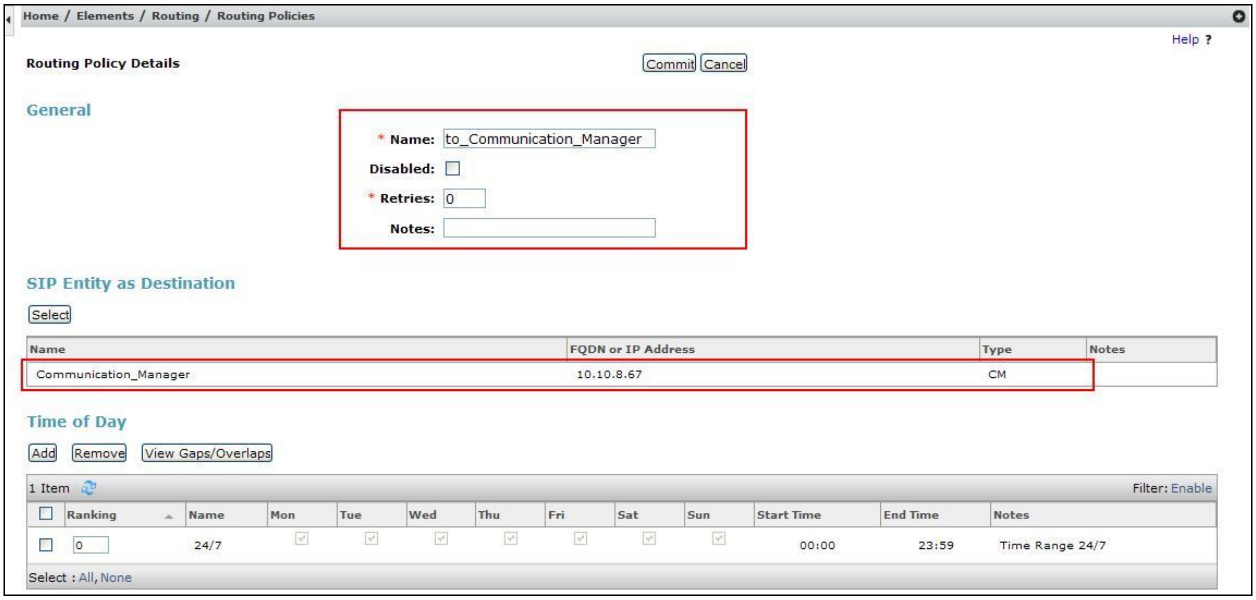
6.7. Administer Routing Policies

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

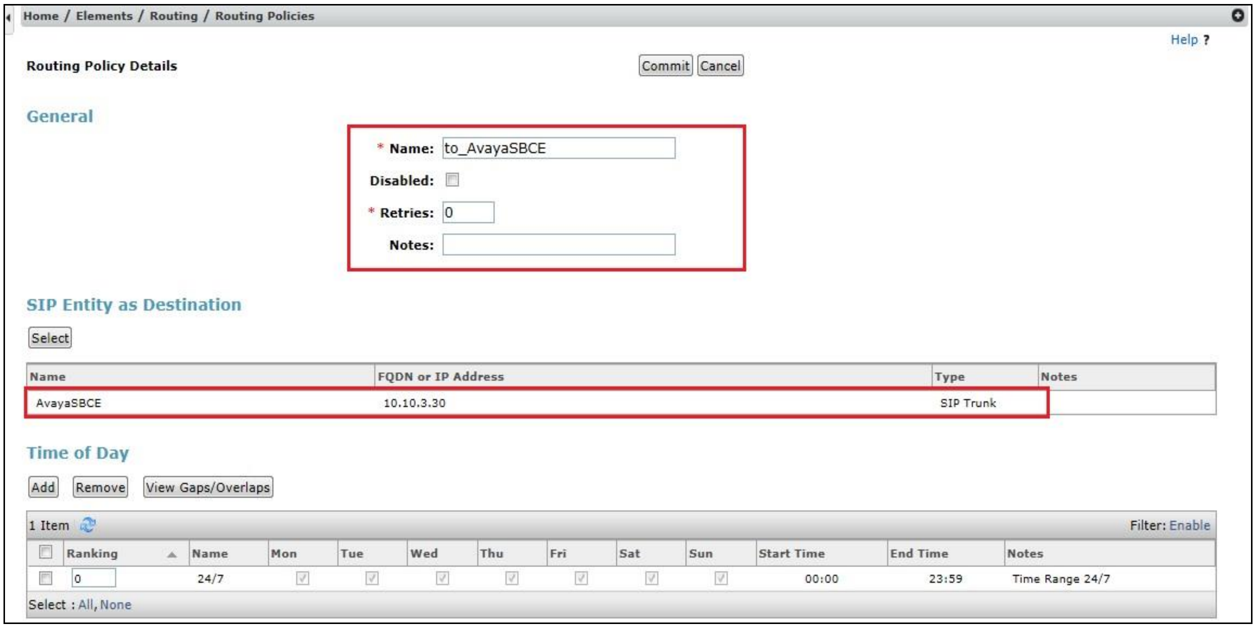
Under **General**:

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

The following screen shows the routing policy for Communication Manager.



The following screen shows the Routing Policy for the Avaya SBCE.



6.8. Administer Dial Patterns

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

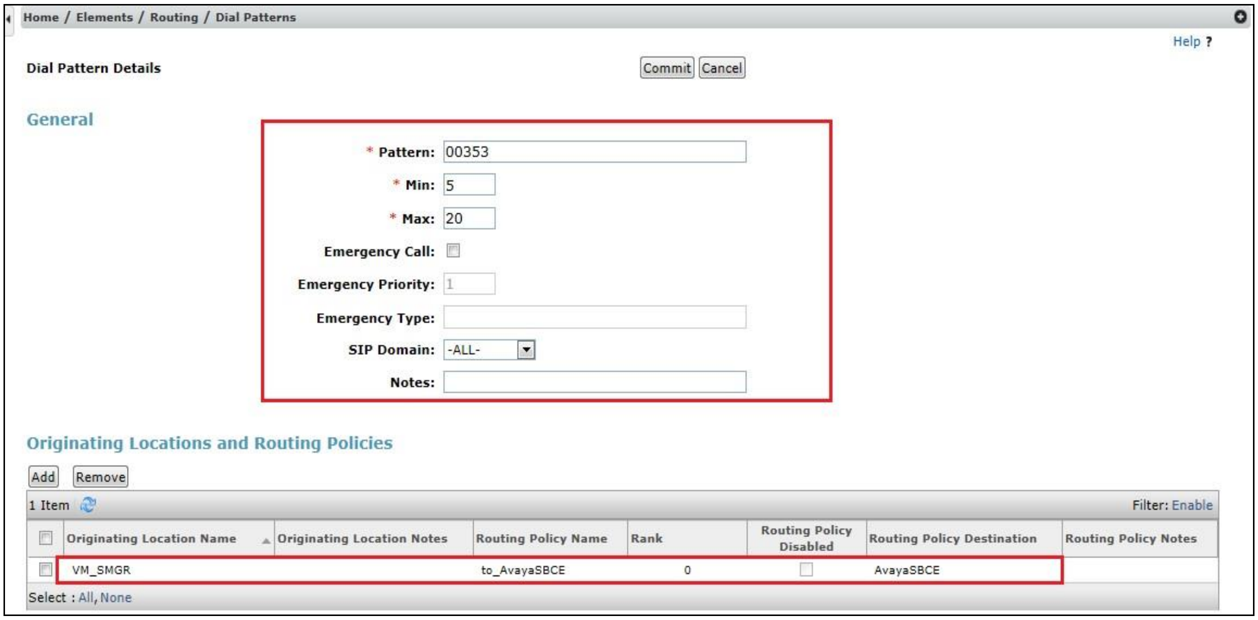
Under **General**:

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

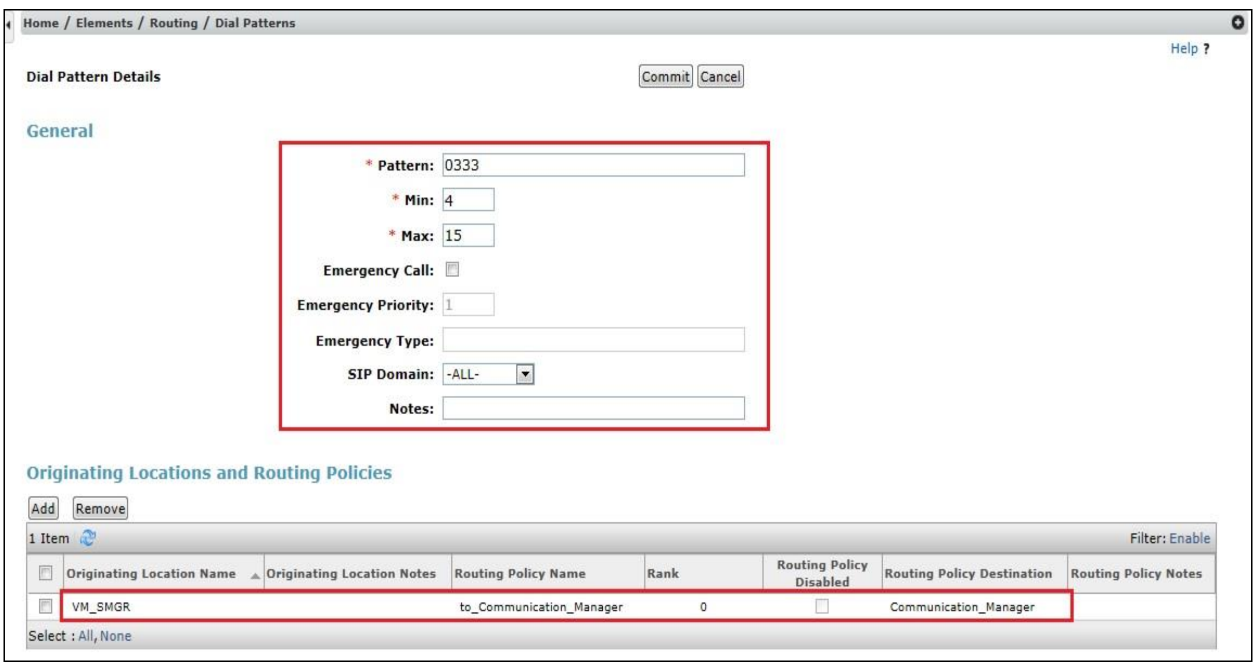
Under **Originating Locations and Routing Policies**:

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

The following screen shows an example dial pattern configured for the Avaya SBCE.



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

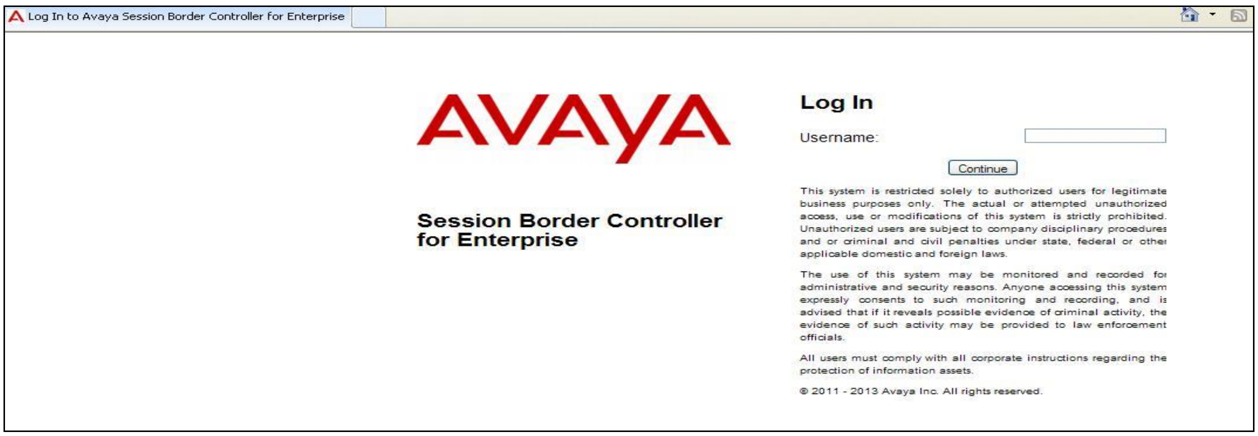


7. Configure Avaya Session Border Controller for Enterprise

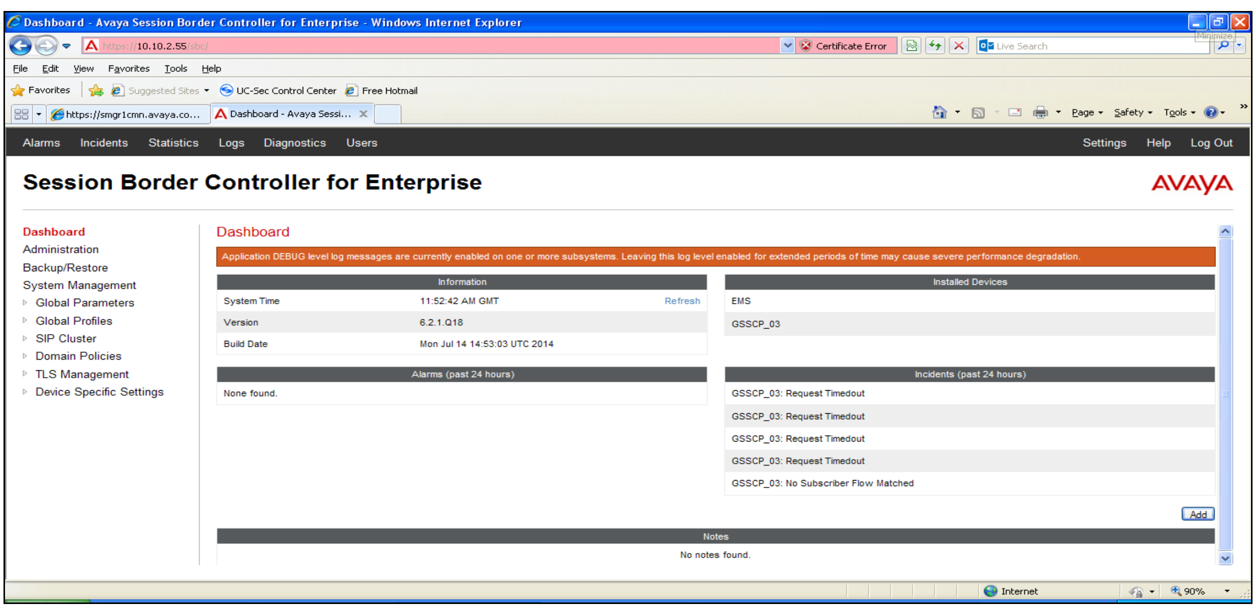
This section describes the configuration of Session Border Controller for Enterprise (Avaya SBCE). The Avaya SBCE provides security and manipulation of signalling to provide an interface to the Service Provider's SIP Trunk that is standard where possible and adapted to the Service Provider's SIP implementation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

7.1. Access Avaya Session Border Controller for Enterprise

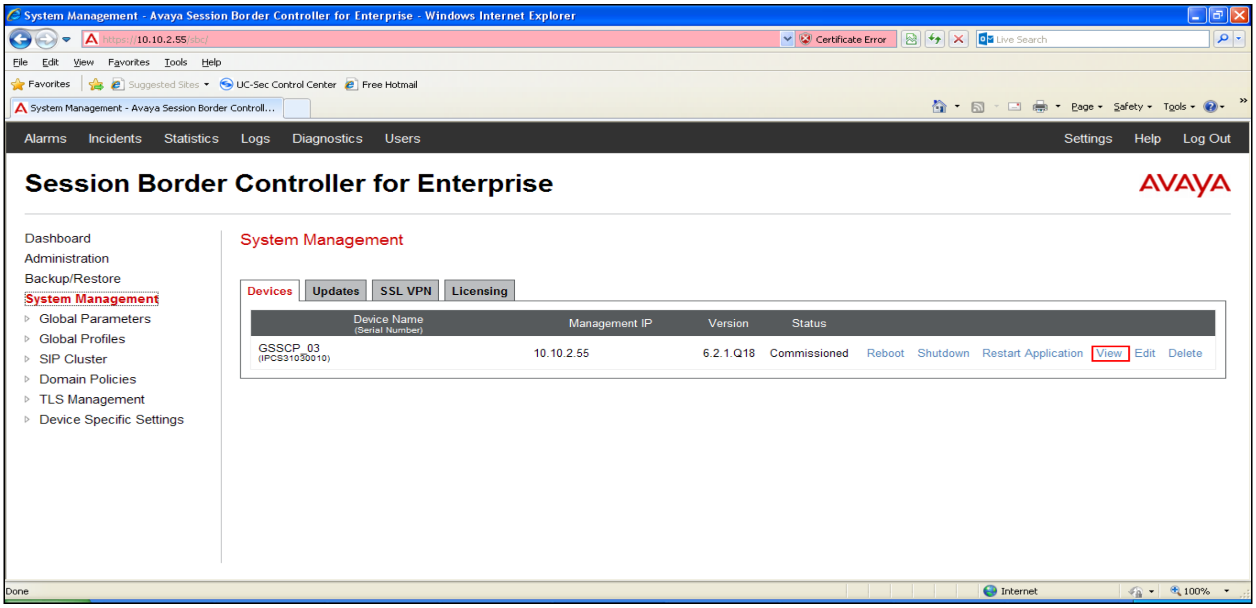
Access the Avaya SBCE using a web browser by entering the URL **https://<ip-address>**, where **<ip-address>** is the management IP address configured at installation and enter the **Username** and **Password**.



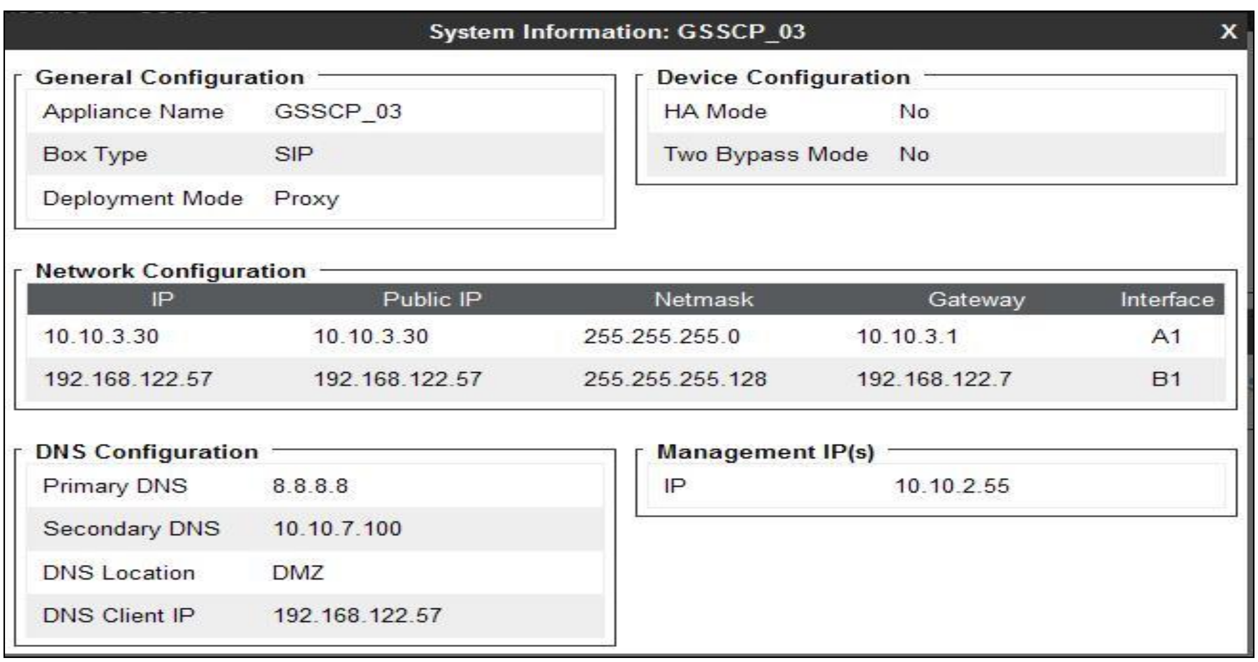
Once logged in, a dashboard is presented with a menu on the left-hand side. The menu is used as a starting point for all configuration of the Avaya SBCE.



To view system information that was configured during installation, navigate to **System Management**. A list of installed devices is shown in the right pane. In the case of the sample configuration, a single device named **GSSCP_03** is shown. To view the configuration of this device, click **View** (the third option from the right).



The System Information screen shows the **Appliance Name**, **Device Configuration** and **DNS Configuration** information.



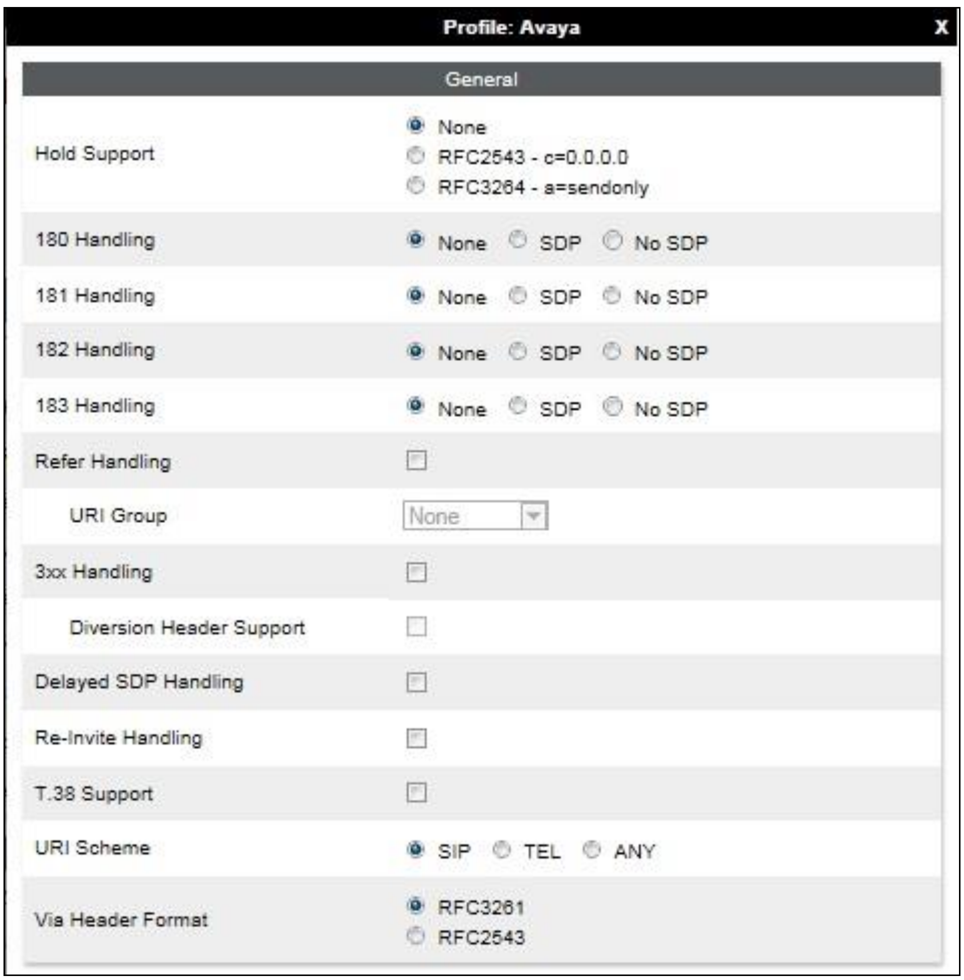
7.2. Global Profiles

When selected, Global Profiles allows for configuration of parameters across all UC-Sec appliances.

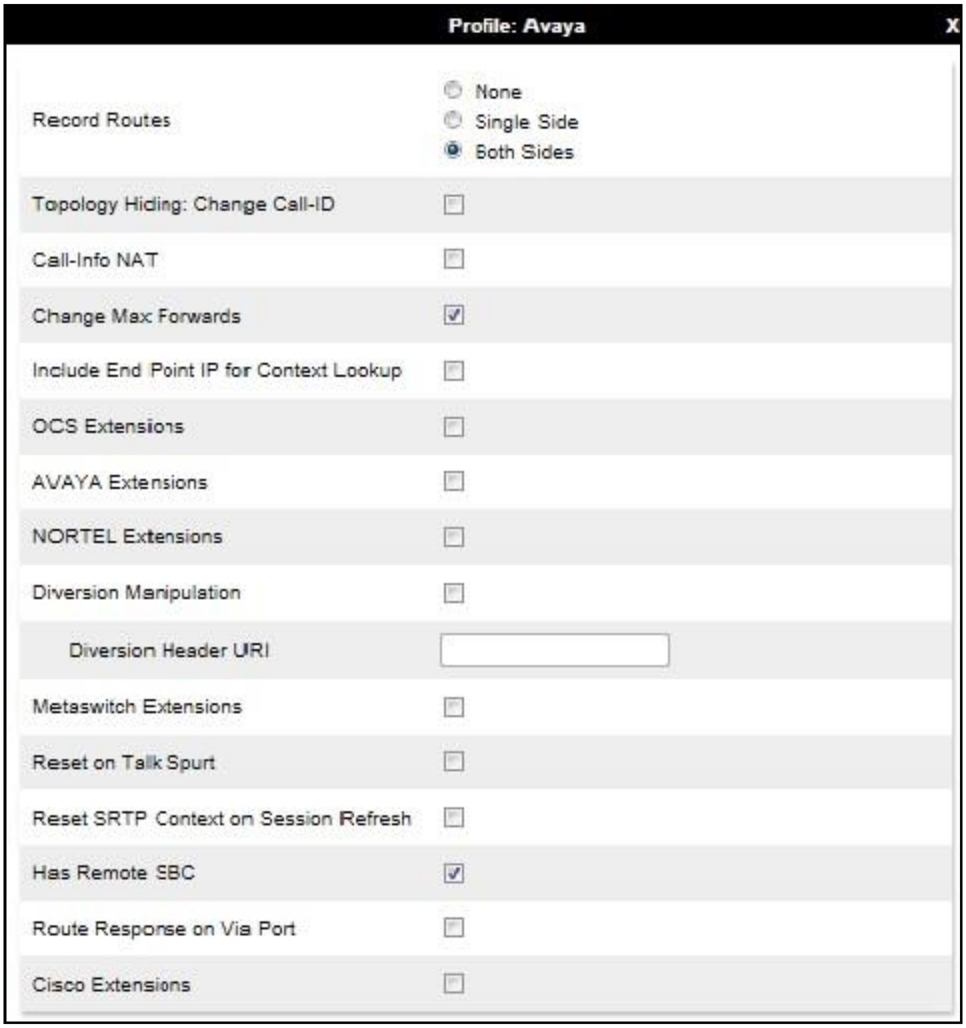
7.2.1. Server Interworking - Avaya

Server Interworking allows the configuration and management of various SIP call server-specific capabilities such as call hold and T.38. From the left-hand menu select **Global Profiles → Server Interworking** and click on **Add Profile.**

- Enter profile name such as **Avaya** and click **Next** (Not Shown)
- Check **Hold Support=None**
- All other options on the **General** Tab can be left at default



Default values can be used for the **Advanced Settings** window. Click **Finish**



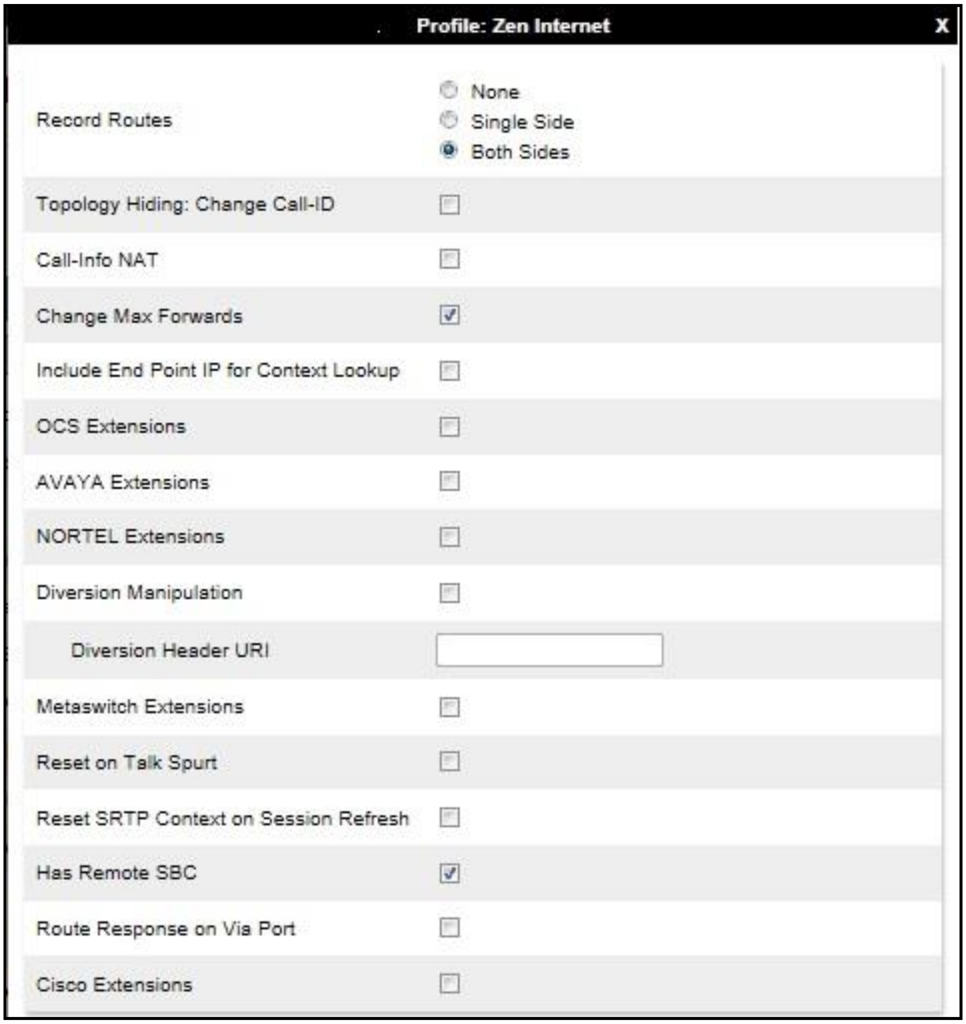
7.2.2. Server Interworking – Zen Internet

Server Interworking allows the configuration and management of various SIP call server-specific capabilities such as call hold and T.38. From the left-hand menu select **Global Profiles → Server Interworking** and click on **Add Profile**.

- Enter profile name such as **Zen Internet** and click **Next** (Not Shown)
- Check **Hold Support = None**
- All other options on the **General** Tab can be left at default

Click on **Next** on the following screens and then **Finish**

Default values can be used for the **Advanced Settings** window. Click **Finish**.



7.2.3. Routing

Routing profiles define a specific set of packet routing criteria that are used in conjunction with other types of domain policies to identify a particular call flow and thereby ascertain which security features will be applied to those packets. Parameters defined by Routing Profiles include packet transport settings, name server addresses and resolution methods, next hop routing information, and packet transport types.

Routing information is required for routing to Session Manager on the internal side and Zen Internet addresses 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.

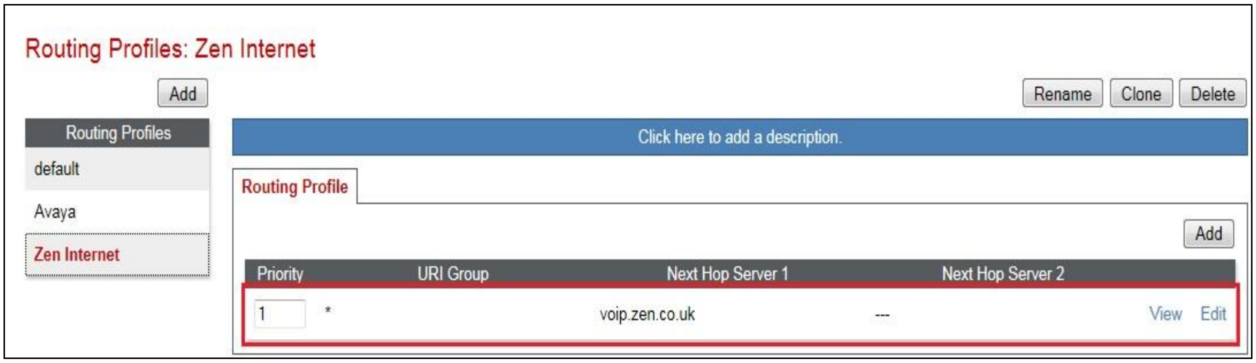
Create a Routing Profile for both Session Manager and Zen Internet SIP trunk. To add a routing profile, navigate to **Global Profiles → Routing** and select **Add Profile**. Enter a **Profile Name** and click **Next** to continue.
In the new window that appears, enter the following values. Use default values for all remaining fields:

- **URI Group:** Select "*****" from the drop down box
- **Next Hop Server 1:** Enter the Domain Name or IP address of the Primary Next Hop server, e.g. Session Manager
- **Next Hop Server 2:** (Optional) Enter the Domain Name or IP address of the secondary Next Hop server
- **Routing Priority Based on Next Hop Server**: Checked (not shown)
- **Outgoing Transport:** Choose the protocol used for transporting outgoing signalling packets (not shown)

Click **Finish**.

The following screen shows the Routing Profile to Session Manager.

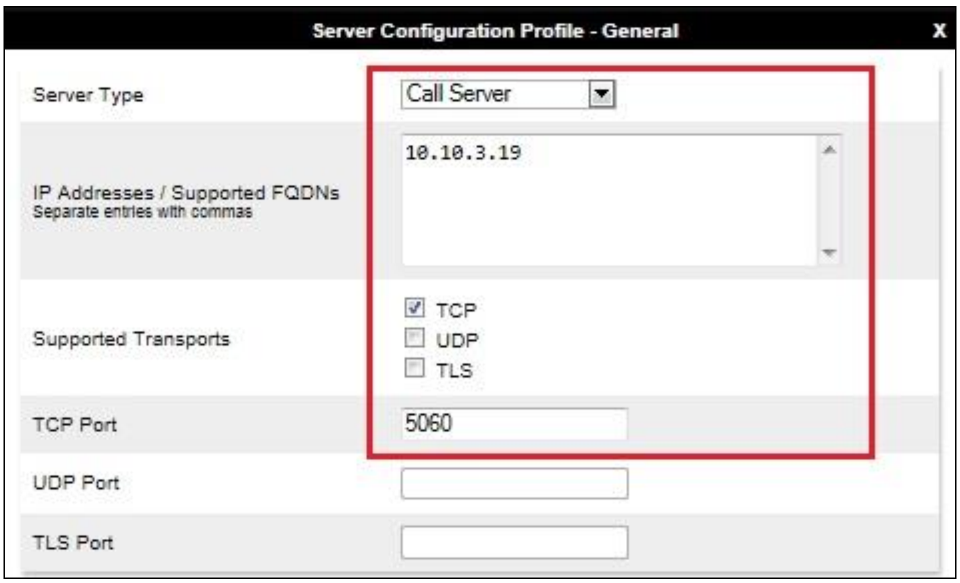
The following screen shows the Routing Profile to Zen Internet SIP Trunk.



7.2.4. Server Configuration– Avaya Aura® Session Manager

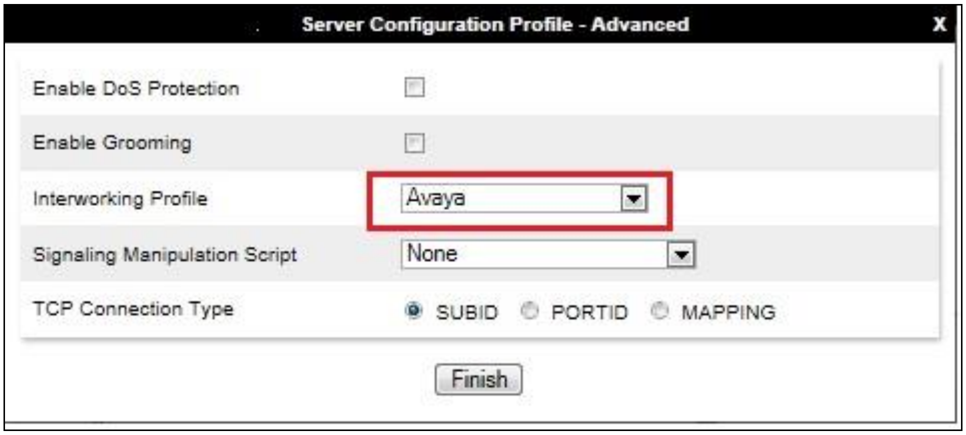
Servers are defined for each server connected to the Avaya SBCE. In this case, Zen Internet is connected as the Trunk Server and Session Manager is connected as the Call Server.
The **Server Configuration** screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. Together, these tabs allow the configuration and management of various SIP call server-specific parameters such as TCP and UDP port assignments, IP Server type, heartbeat signalling parameters and some advanced options. From the left-hand menu select **Global Profiles → Server Configuration** and click on **Add Profile** and enter a descriptive name. On the **Add Server Configuration Profile** tab, set the following:

- Select **Server Type** to be **Call Server**
- Enter **IP Addresses / Supported FQDNs** to **10.10.3.19** (Session Manager IP Address)
- For **Supported Transports,** check **TCP**
- **TCP Port:5060**
- Click on **Next** (not shown) to use default entries on the **Authentication** and **Heartbeat** tabs



On the **Advanced** tab:

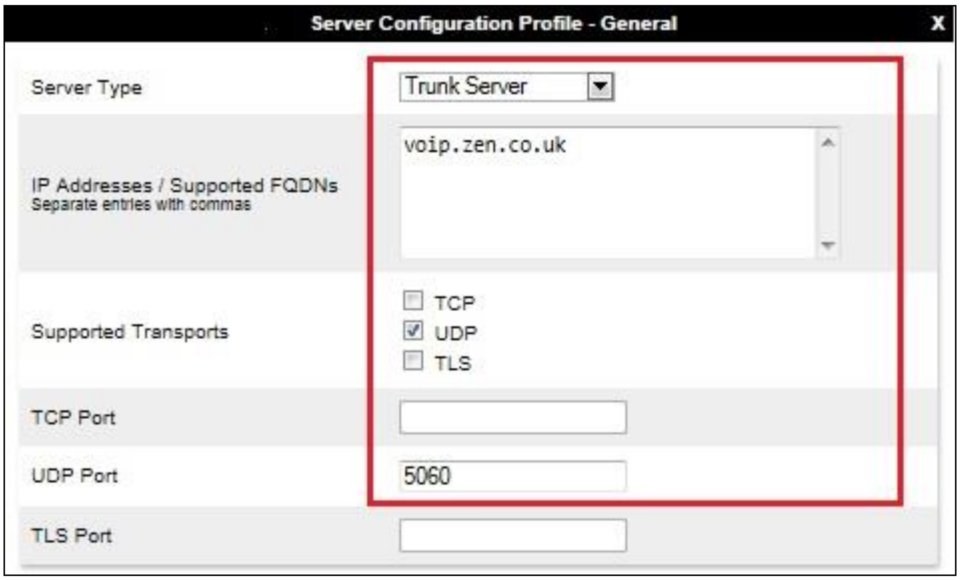
- Select **Avaya** for **Interworking Profile** defined in **Section 7.2.1**
- Click **Finish**



7.2.5. Server Configuration – Zen Internet

To define the Zen Internet SBC as a Trunk Server, navigate to **Global Profiles → Server Configuration** and click on **Add Profile** and enter a descriptive name. On the **Add Server Configuration Profile** tab, click on **Edit** and set the following:

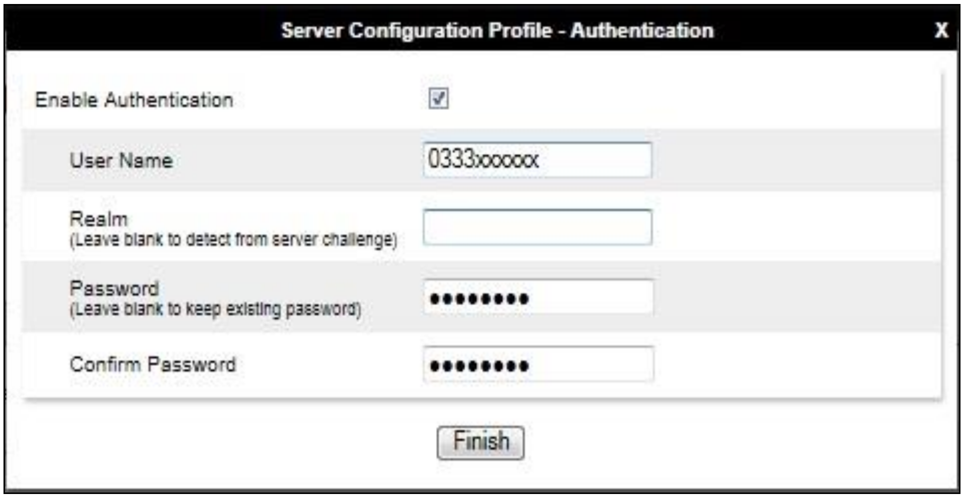
- Select **Server Type** as **Trunk Server**
- Set **IP Addresses/Supported FQDNs** to **voip.zen.co.uk** (Zen Internet SIP Trunk)
- **Supported Transports**: Check **UDP**
- **UDP Port: 5060**
- Click **Next** (not shown)



In the new window that appears, enter the following values as Zen Internet require authentication to connect to their network:

- **Enabled Authentication:** Checked
- **User Name:** Enter username provided by the Service Provider
- **Realm:** Enter realm details provided by the Service Provider
- **Password** Enter password provided by the Service Provider
- **Confirm Password** Re-enter password provided by the Service Provider

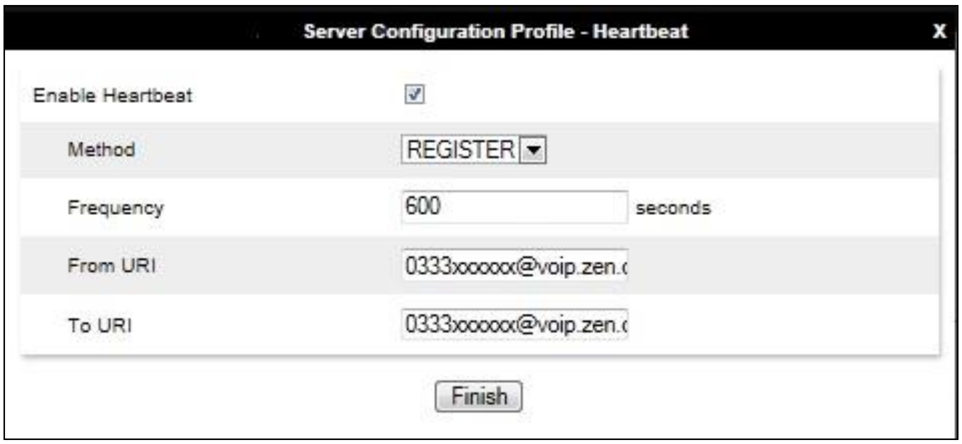
Click **Next** to continue (not shown).



In the new window that appears, enter the following values.

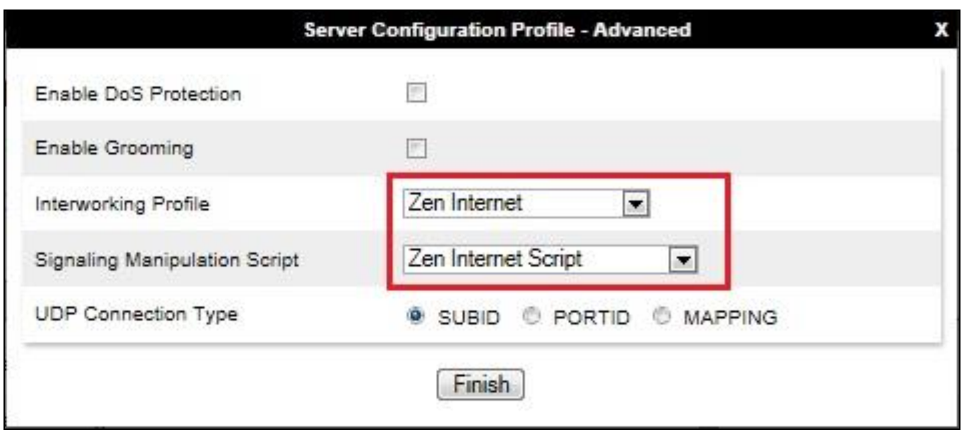
- **Enabled Heartbeat:** Checked
- **Method:** Select **REGISTER** from the drop-down box
- **Frequency:** Choose the desired frequency in seconds the Avaya SBCE will send SIP REGISTER messages
- **From URI:** Enter a URI to be sent in the FROM header for SIP REGISTER messages
- **TO URI:** Enter a URI to be sent in the TO header for SIP REGISTER messages

Click **Next** to continue (not shown).



On the **Advanced** tab:

- Select **Zen Internet** for **Interworking Profile** as defined in **Section 7.2.2**
- Select **Zen Internet Script** defined in **Section 7.2.7** from the Signaling Manipulation Script drop down menu
- Click **Finish**

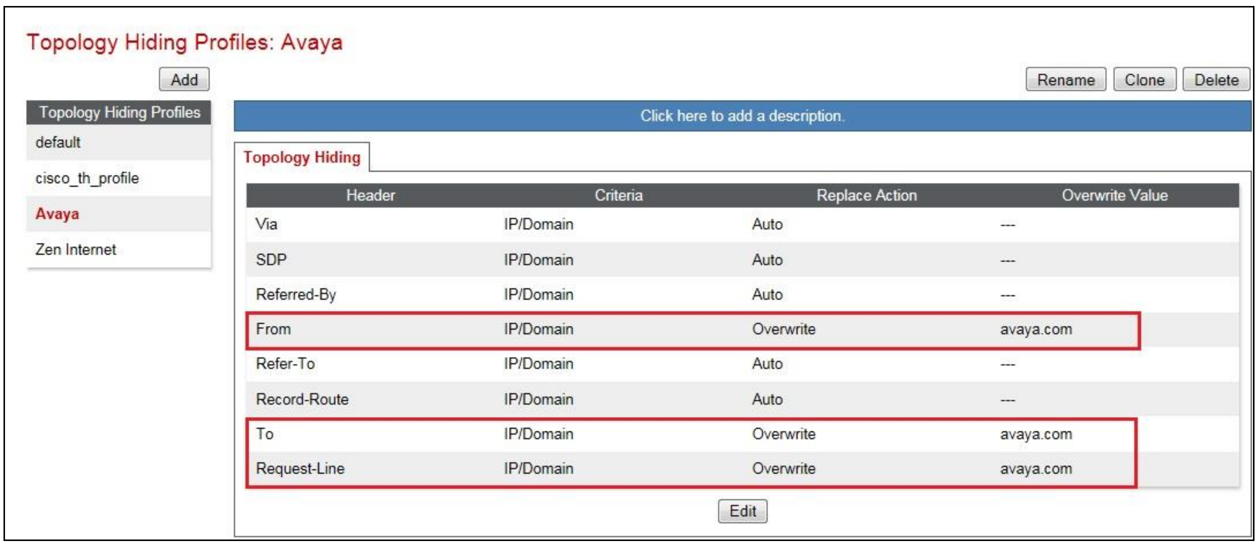


7.2.6. Topology Hiding

Topology hiding is used to hide local information such as private IP addresses and local domain names. The local information can be overwritten with a domain name or IP addresses. The default **Replace Action** is **Auto**, this replaces local information with IP addresses, generally the next hop. Topology hiding has the advantage of presenting single Via and Record-Route headers externally where multiple headers may be received from the enterprise, particularly from 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.

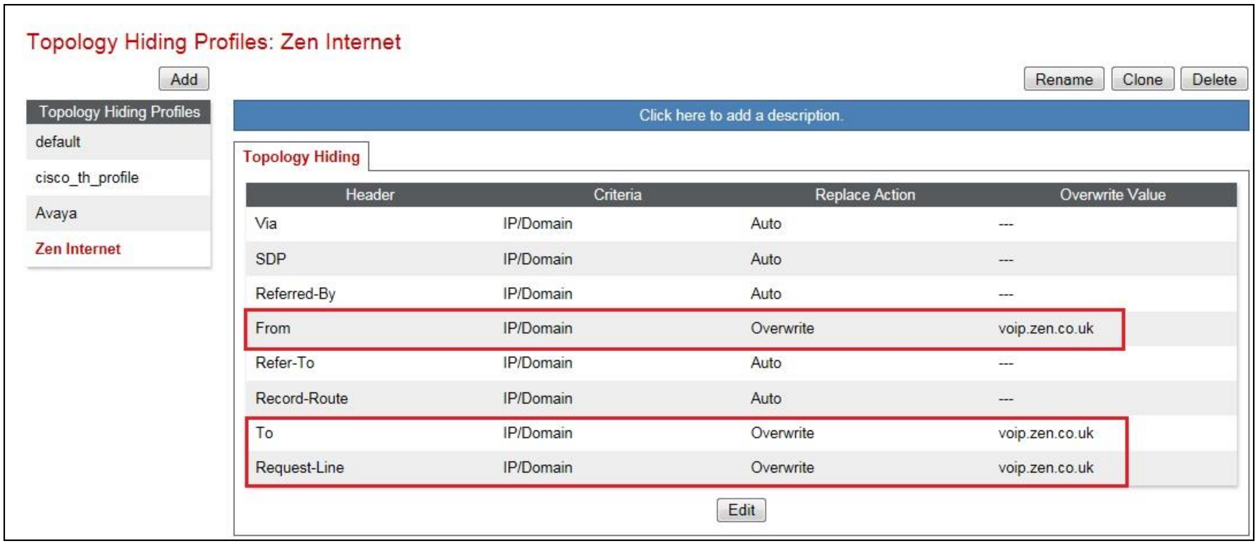
To define Topology Hiding for Session Manager, navigate to **Global Profiles → Topology Hiding** from menu on the left hand side. Click on **Add Profile** and enter details in the **Topology Hiding Profile** pop-up menu (not shown).

- Enter a descriptive Profile Name such as **Avaya**
- If the required Header is not shown, click on **Add Header** (not shown)
- Under the **Header** field for **To**, **From** and **Request Line**, select **IP/Domain** under **Criteria** and **Overwrite** under **Replace Action**. For **Overwrite Value**, insert **avaya.com**.
- Click **Finish** (not shown)



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

- Enter a descriptive **Profile Name** such as **Zen Internet** and click **Next**
- If the required Header is not shown, click on **Add Header**
- Under the **Header** field for **To**, **From** and **Request Line**, select **IP/Domain** under **Criteria** and **Overwrite** under **Replace Action**. For **Overwrite Value**, insert **voip.zen.co.uk**
- Click **Finish** (not shown)



7.2.7. Signaling Manipulation

The Signaling Manipulation feature allows the ability to add, change and delete any of the headers in a SIP message. This feature will add the ability to configure such manipulation in a highly flexible manner using a proprietary scripting language called SigMa. The SigMa scripting language is designed to express any of the SIP header manipulation operations to be done by the Avaya SBCE.

When performing an outbound CLID restricted call, Zen Internet require the From Header to be populated with the CLID of the set that is making the outbound call or of a known CLID on the PABX. When CLID restriction is enabled on Communication Manager, Communication Manager populates the From Header with Anonymous. In order for outbound CLID restricted call to terminate successfully, a SigMa script was created on the Avaya SBCE to copy the CLID from the P-Asserted-Identity Header and populate the From Header with the P-Asserted-Identity Header CLID.

When performing call re-direction or EC500 calls, Zen Internet require the From Header to be populated with the CLID of the set that is performing the call re-direction/EC500 call or of a known CLID on the PABX. In order for call redirection/EC500 to complete successfully, a SigMa script was created on the Avaya SBCE to copy the CLID from the Diversion Header and populate the From Header with the Diversion Header CLID.

To create a new Signaling Manipulation,navigate to **Global Profiles → Signaling Manipulation** from the main menu on the left hand side. Click on **Add Script** and enter a title in the script editor (not shown). The script text is displayed on the next page below.

```
within session "ALL"
{
 act on message where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING" and
%METHOD="INVITE"
 {
 if(%HEADERS["From"][1].URI.USER.regex_match("^anonymous"))then
 {
 %DivUser = %HEADERS["P-Asserted-Identity"][1].URI.USER;
 %HEADERS["From"][1].URI.USER = %DivUser;
 }
 }
 }

within session "ALL"
{
 act on message where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING" and
%METHOD="INVITE"
 {
 if(exists(%HEADERS["Diversion"][1]))then
 {
 %DivUser = %HEADERS["Diversion"][1].URI.USER;
 %HEADERS["From"][1].URI.USER = %DivUser;
 %HEADERS["From"][1].DISPLAY_NAME = %DivUser;
 %HEADERS["Contact"][1].URI.USER = %DivUser;
 %HEADERS["Contact"][1].DISPLAY_NAME = %DivUser;
 %HEADERS["P-Asserted-Identity"][1].URI.USER = %DivUser;
 %HEADERS["P-Asserted-Identity"][1].DISPLAY_NAME = %DivUser;
 }
 }
 }
```

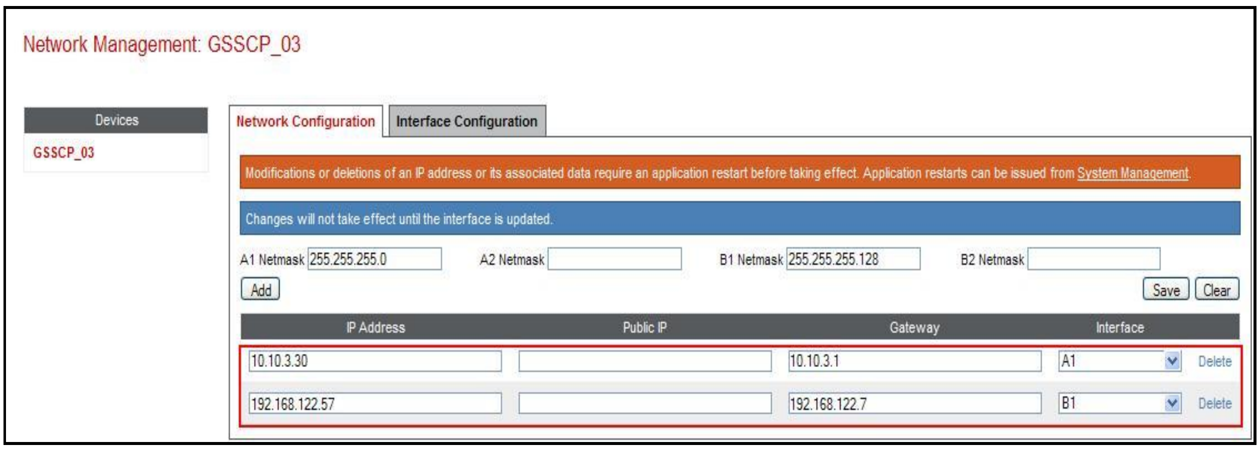
Once entered and saved, the script appears as shown in the following screenshot.

7.3. Define Network Information

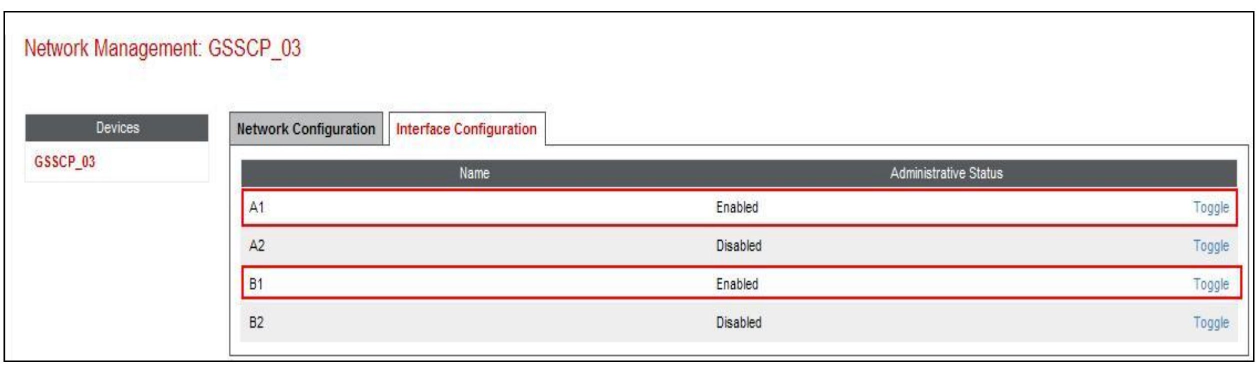
Network information is required on the Avaya SBCE to allocate IP addresses and masks to the interfaces. Note that only the **A1** and **B1** interfaces are used, typically the **A1** interface is used for the internal side and **B1** is used for external. Each side of the Avaya SBCE can have only one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

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

- Click on **Add**
- Define the internal IP address and assign to interface **A1**
- Select **Save Changes** to save the information
- Click on **Add**
- Define the external IP address and assign to interface **B1**
- Select **Save Changes** 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)



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



7.4. Define Interfaces

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

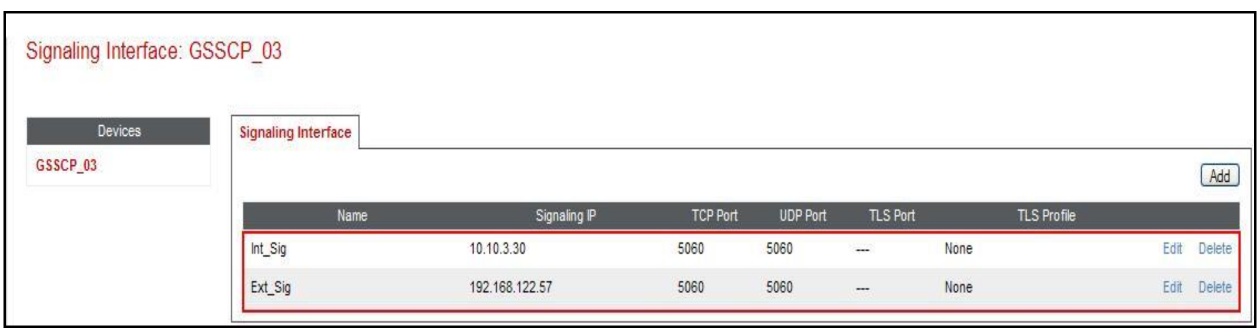
7.4.1. Signalling Interfaces

To define the signalling interfaces on the Avaya SBCE, navigate to **Device Specific Settings → Signaling Interface** from the menu on the left hand side. Details of transport protocol and ports for the internal and external SIP signalling are entered here.

The Signaling Interface screen allows the IP address and ports to be set for transporting signalling messages over the SIP trunk. The Avaya SBCE listens for SIP requests on the defined ports. Create a Signaling Interface for both the inside and outside IP interfaces. To create a new Signaling Interface, navigate to **Device Specific Settings → Signaling Interface** and click **Add**.

- **Name**: **Int_Sig**
- **Signaling IP**: **10.10.3.30** (Internal address for calls toward Session Manager)
- **TCP Port**: **5060**
- **UDP Port**: **5060**
- Click **Finish**
- Select **Add**
- **Name**: **Ext_Sig**
- **Signaling IP: 192.168.122.57** (External address for calls toward Zen Internet)
- **TCP Port**: **5060**
- **UDP Port**: **5060**
- Click **Finish**

The following screen shows the signalling interfaces created in the sample configuration for the inside and outside IP interfaces.

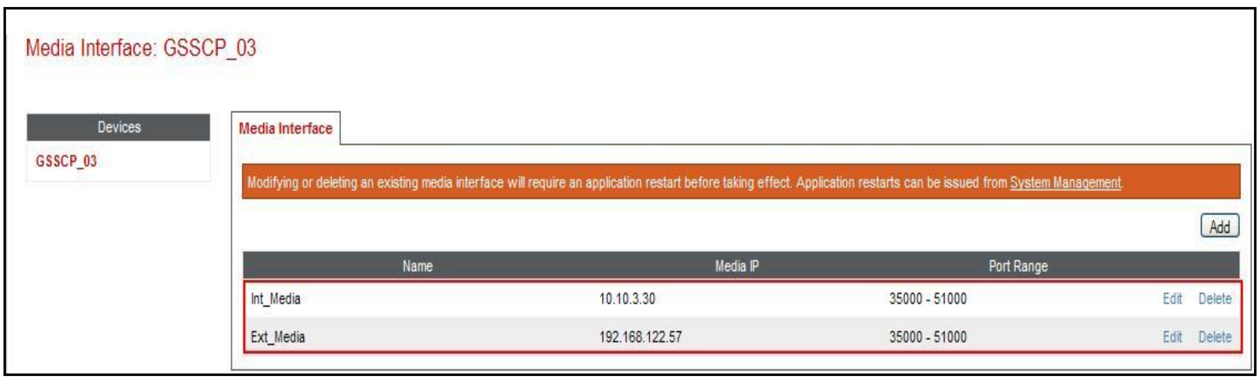


7.4.2. Media Interfaces

The Media Interface screen allows the IP address and ports to be set for transporting Media over the SIP trunk. The Avaya SBCE listens for SIP media on the defined ports.
To create a new Media Interface, navigate to **Device Specific Settings → Media Interface**.

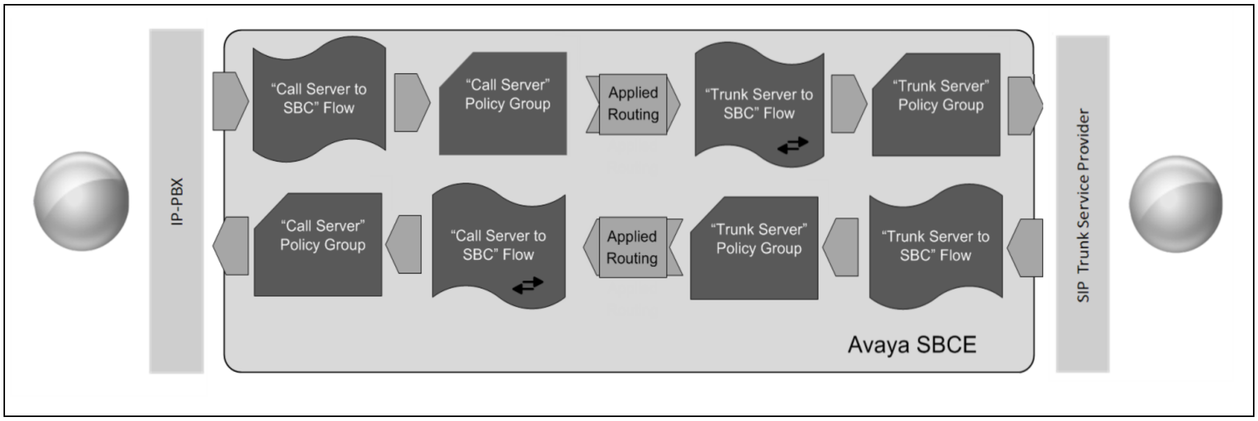
- Select **Add**
- **Name**: **Int_Media**
- **Media IP**: **10.103.30** (Internal address for calls toward Session Manager)
- **Port Range**: **35000-51000**
- Click **Finish**
- Select **Add**
- **Name**: **Ext_Media**
- **Media IP**: **192.168.122.57** (External address for calls toward Zen Internet)
- **Port Range**: **35000-51000**
- Click **Finish**

The following screen shows the Media Interfaces created in the sample configuration for the inside and outside IP interfaces.



7.5. Server Flows

Server Flows combine the previously defined profiles into outgoing flows from Session Manager to Zen Internet's SIP Trunk and incoming flows from Zen Internet's SIP Trunk to Session Manager. This configuration ties all the previously entered information together so that signalling can be routed from Session Manager to the PSTN via the Zen Internet network and vice versa. The following screen illustrates the flow through the Avaya SBCE to secure a SIP Trunk call.

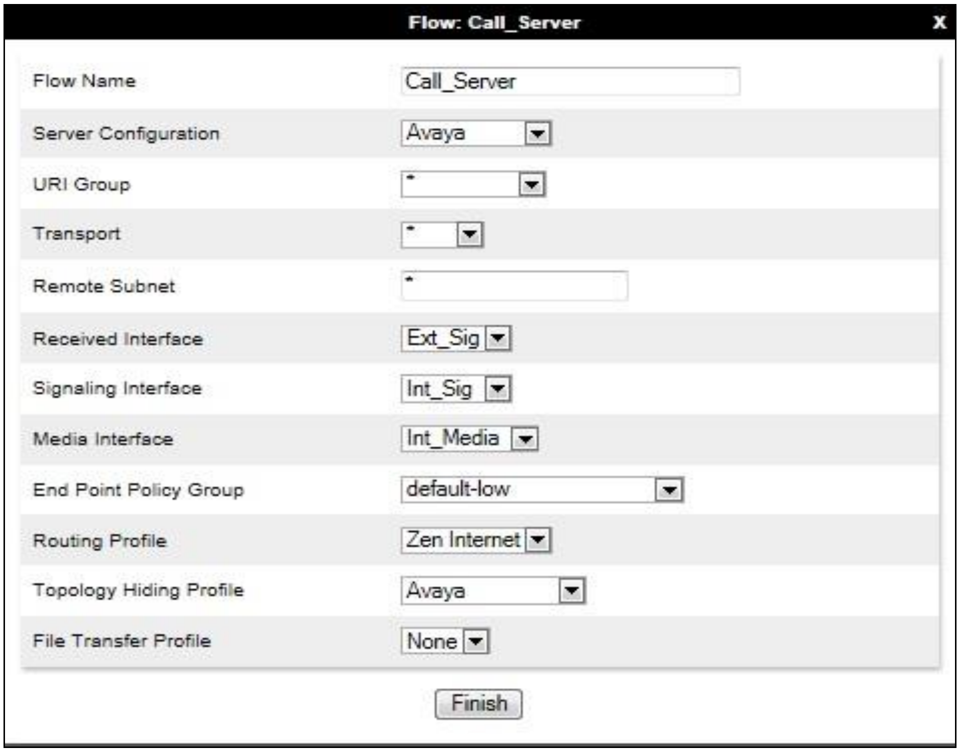


To create a Server Flow, navigate to **Device Specific Settings → End Point Flows**. Select the **Server Flows** tab and click **Add Flow**.

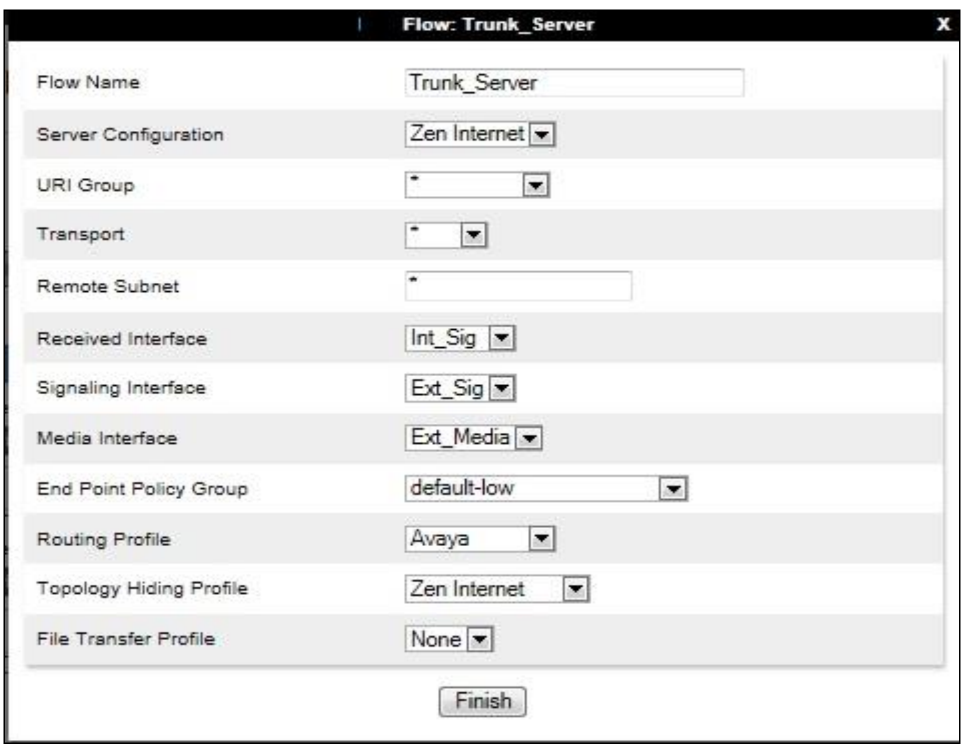
- **Flow Name:** Enter a descriptive name
- **Server Configuration:** Select a Server Configuration created in **Section 7.2.4** and **7.2.5** and assign to the Flow
- **Received Interface:** Select the Signaling Interface the Server Configuration is allowed to receive SIP messages from
- **Signaling Interface:** Select the Signaling Interface used to communicate with the Server Configuration
- **Media Interface:** Select the Media Interface used to communicate with the Server Configuration
- **End Point Policy Group:** Select the policy assigned to the Server Configuration
- **Routing Profile:** Select the profile the Server Configuration will use to route SIP messages
- **Topology Hiding Profile:** Select the profile to apply toward the Server Configuration

Click **Finish** to save and exit.

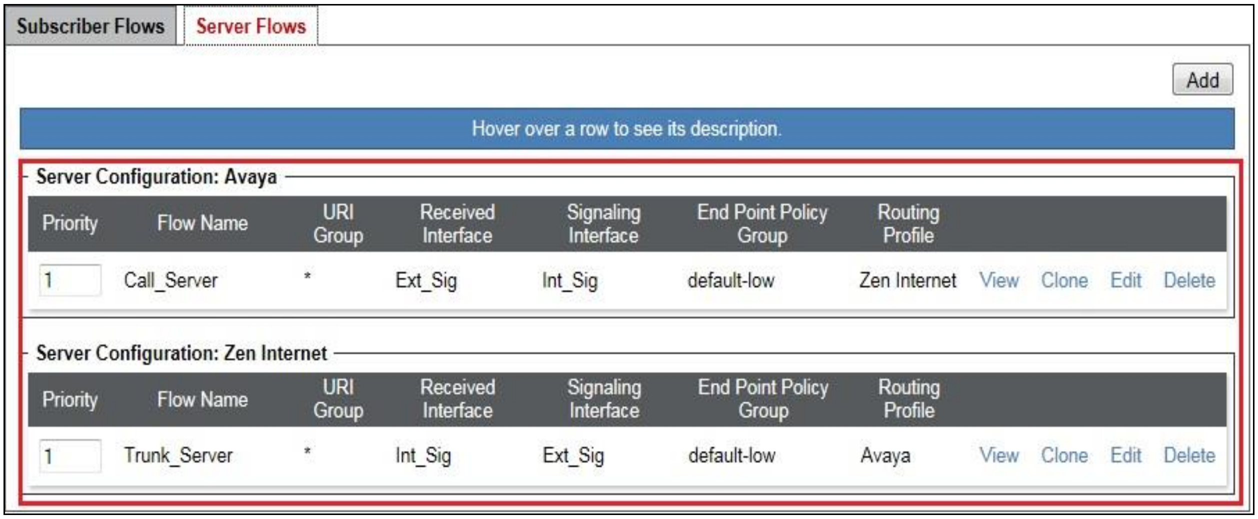
The following screen shows the Server Flow for Session Manager.



The following screen shows the Server Flow for Zen Internet.



This configuration ties all the previously entered information together so that calls can be routed from Session Manager to Zen Internet SIP Trunk service and vice versa. The following screenshot shows all configured flows.



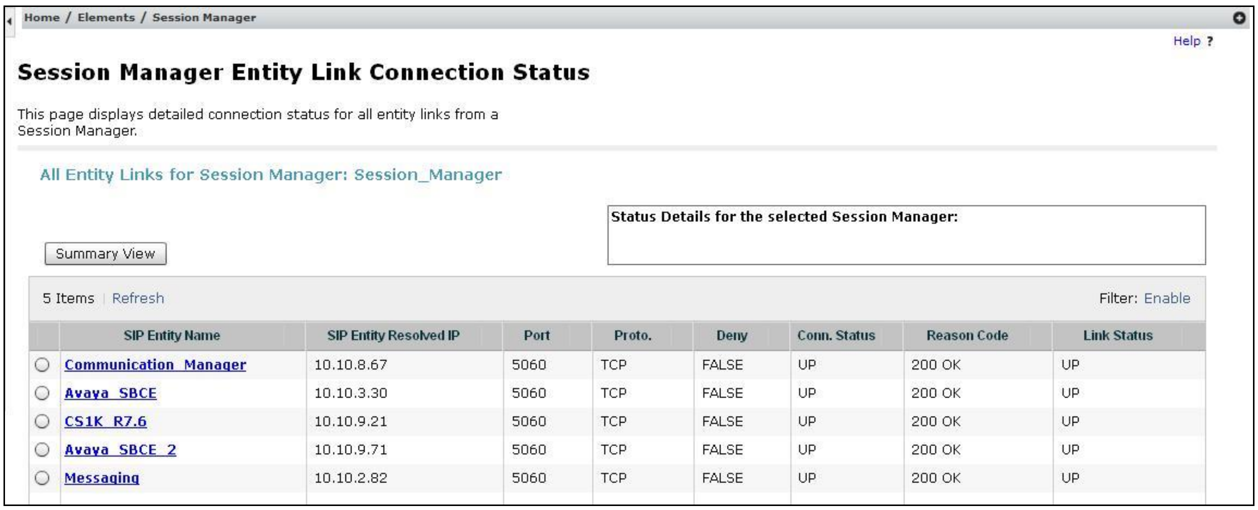
8. Configure Zen Internet SIP Trunk Equipment

The configuration of the Zen Internet equipment used to support Zen Internet's SIP Trunk is outside of the scope of these Application Notes and will not be covered. To obtain further information on Zen Internet equipment and system configuration please contact an authorized Zen Internet representative.

9. Verification Steps

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

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



Home / Elements / Session Manager

Help ?

Session Manager Entity Link Connection Status

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

All Entity Links for Session Manager: Session_Manager

Status Details for the selected Session Manager:

Summary View

5 Items | Refresh   Filter: Enable

	SIP Entity Name	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
○	Communication_Manager	10.10.8.67	5060	TCP	FALSE	UP	200 OK	UP
○	Avaya_SBCE	10.10.3.30	5060	TCP	FALSE	UP	200 OK	UP
○	CS1K_R7.6	10.10.9.21	5060	TCP	FALSE	UP	200 OK	UP
○	Avaya_SBCE_2	10.10.9.71	5060	TCP	FALSE	UP	200 OK	UP
○	Messaging	10.10.2.82	5060	TCP	FALSE	UP	200 OK	UP

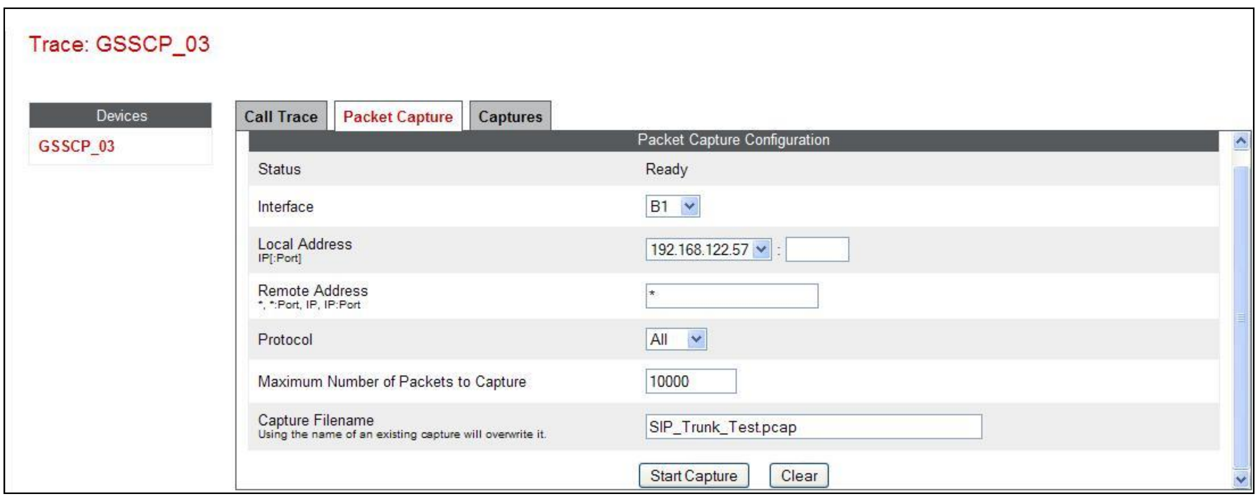
2. From the Communication Manager SAT interface run the command **status trunk n** where **n** is a previously configured SIP trunk. Observe if all channels on the trunk group display **in-service/idle**.

```
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
0001/010 T00010   in-service/idle    no
```

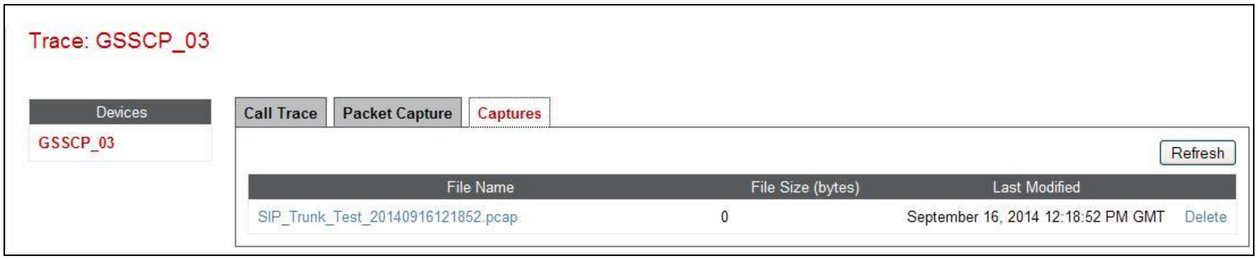
3. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active.
4. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active.
5. Verify that the user on the PSTN can end an active call by hanging up.
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 Session Manager via the Avaya SBCE to the network SBCs are receiving a response.

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

- Select the SIP Trunk interface from the **Interface** drop down menu
- Select the signalling interface IP address from the **Local Address** drop down menu
- Enter the IP address of the network SBC in the **Remote Address** field or enter a * to capture all traffic
- Specify the **Maximum Number of Packets to Capture**, 10000 is shown as an example
- Specify the filename of the resultant pcap file in the **Capture Filename** field
- Click on **Start Capture**



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



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

10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Communication Manager R6.3 as an Evolution Server, Avaya Aura® Session Manager R6.3 and Avaya Session Border Controller for Enterprise to Zen Internet's SIP Trunk Service. Zen Internet's SIP Trunk 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 2014
[2] *Administering Avaya Aura® System Platform*, Release 6.3, May 2014
[3] *Avaya Aura® Communication Manager using VMware® in the Virtualized Environment Deployment Guide*, April 2014
[4] *Avaya Aura® Communication Manager 6.3 Documentation library*, August 2014
[5] *Avaya Aura® System Manager using VMware® in the Virtualized Environment Deployment Guide* Release 6.3 April 2014
[6] *Implementing Avaya Aura® System Manager* Release 6.3, May 2014
[7] *Upgrading Avaya Aura® System Manager to 6.3* May 2014
[8] *Administering Avaya Aura® System Manager* Release 6.3, May 2014
[9] *Avaya Aura® Session Manager using VMware® in the Virtualized Environment Deployment Guide* Release 6.3 August 2014
[10] *Implementing Avaya Aura® Session Manager* Release 6.3, May 2014
[11] *Upgrading Avaya Aura® Session Manager* Release 6.3, May 2014
[12] *Administering Avaya Aura® Session Manager* Release 6.3, June 2014
[13] *Installing Avaya Session Border Controller for Enterprise*, Release 6.2 June 2014
[14] *Upgrading Avaya Session Border Controller for Enterprise* Release 6.2 July 2014
[15] *Administering Avaya Session Border Controller for Enterprise* Release 6.2 March 2014
[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.