Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Communication Manager R6.2 as an Evolution Server, Avaya Aura® Session Manager R6.2 and Avaya Session Border Controller for Enterprise R6.2 to Support Motto VoIP SIP Trunk Service – Issue 1.0

Abstract

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

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 Trunk Service provided by Motto VoIP.

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 from the PSTN to the enterprise site were routed to DID numbers assigned by Motto VoIP. Incoming calls were made to H.323, SIP, Digital and Analogue telephones.
- Outgoing calls from the enterprise site to the PSTN were routed to PSTN numbers. Outgoing calls were made from H.323, SIP, Digital and Analogue telephones.
- Calls using G.711A, G.711MU and G.729 codec's supported by Motto VoIP.
  DTMF transmission using RFC 2833 with successful 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 was used during this test.
- Call coverage and call forwarding for endpoints at the enterprise site.

2.2. Test Results

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

- T.38 fax transmission is not supported by Motto.
- All tests were completed using H.323, SIP, Digital and Analogue phone types. The Avaya one-X® Communicator was used to test soft client functionality.
- No inbound toll free numbers were tested, however routing of inbound DID numbers and the relevant number translation was successfully tested.
- No Emergency Services numbers were tested as test calls to these numbers should be pre-arranged with the Operator.

2.3. Support

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

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

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to the Motto VoIP SIP Trunk Service. 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, Avaya analogue telephones and an analogue fax machine. Also included in the test configuration was an Avaya one-X® Communicator soft phone running on a laptop PC configured for SIP.

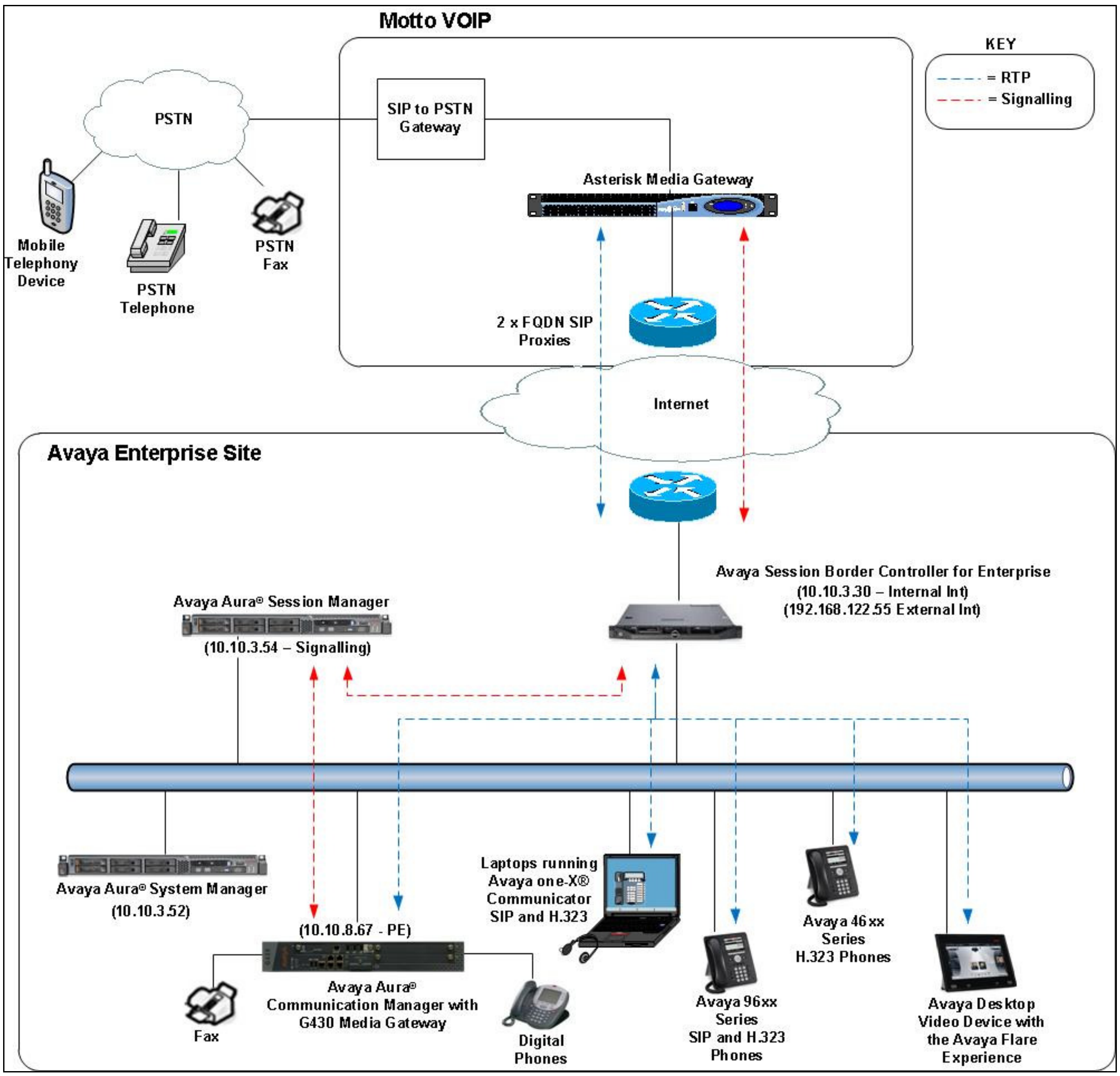


Figure 1: Test Setup Motto VoIP SIP Trunk Service to simulated Enterprise

4. Equipment and Software Validated

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

Equipment	Software
Avaya S8800 Server	Avaya Aura® Communication Manager R6.2 (R016x.02.0.823.0-20558)
Avaya G430 Media Gateway MM711 Analogue MM712 Digital MGP Firmware	 HW31 FW093 HW07 FW009 30.12.1
Avaya S8800 Server	Avaya Aura® Session Manager R6.2 SP3 (6.2.0.0.15669 -6.2.12.307)
Avaya S8800 Server	Avaya Aura® System Manager R6.2 (6.2.0.0.15669-6.2.12.9) Update revision No: 6.2.15.1.1959
Dell R310	Avaya Session Border Controller for Enterprise. (6.2.0.Q36)
Avaya 9650 Phone (H.323)	3.171B
Avaya 9621 Phone (SIP)	6.2.0.72
Avaya 2420 Digital Phone	N/A
Analog Phone	N/A
Avaya 4620 Phone (H.323)	1.2200
Avaya 9611 Phone (SIP)	6.2.0.72
Avaya one-X® Communicator	6.1.3.06-SP3-35509
Avaya A175 Desktop Video Device (SIP)	Flare Experience Release 1.1
Motto VoIP	
Proxy Servers	OpenSIPS 1.7 & OpenSIPS 1.8
Media Gateways	Asterisk 1.4.22-0Motto14

5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring Communication Manager for SIP Trunking. SIP trunks are established between Communication Manager and Session Manager. These SIP trunks will carry SIP signalling associated with the Motto VoIP SIP Trunk Service. For incoming calls, the Session Manager receives SIP messages from the Avaya SBCE and directs the incoming SIP messages to Communication Manager. Once the message arrives at Communication Manager, further incoming call treatment, such as incoming digit translations and class of service restrictions may be performed. All outgoing calls to the PSTN are processed within Communication Manager and may be first subject to outbound features such as automatic route selection, digit manipulation and class of service restrictions. Once Communication Manager selects a SIP trunk, the SIP signalling is routed to 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 Motto VoIP 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 Avaya S8800 Server and Avaya G430 Media Gateway is presumed to have been previously completed and is not discussed here.

5.1. Confirm System Features

The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity. Use the **display system-parameters customer-options** command and on **Page 2**, verify that the **Maximum Administered SIP Trunks** supported by the system is sufficient for the combination of trunks to the Motto VoIP 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: 18000 0
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 4000  10
```

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? y
                  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? n              Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                     Malicious Call Trace? y
     External Device Alarm Admin? y                 Media Encryption Over IP? n
  Five Port Networks Max Per MCC? n     Mode Code for Centralized Voice Mail? n
                Flexible Billing? n
   Forced Entry of Account Codes? y                 Multifrequency Signaling? y
      Global Call Classification? y         Multimedia Call Handling (Basic)? y
             Hospitality (Basic)? y      Multimedia Call Handling (Enhanced)? y
 Hospitality (G3V3 Enhancements)? y               Multimedia IP SIP Trunking? n
                       IP Trunks? y

           IP Attendant Consoles? y
        (NOTE: You must logoff & login to effect the permission changes.)
```

5.2. Administer IP Node Names

The node names defined here will be used in other configuration screens to define a SIP signaling group between Communication Manager and Session Manager. Type **change node-names ip** to make changes to the **IP Node Names**. In the **IP Node Names** form, assign the node **Name** and **IP Address** for the Session Manager. In this case, **SM100** and **10.10.3.55** are the **Name** and **IP Address** for the Session Manager. Also note the **procr** name as this is the interface that Communication Manager will use as the SIP signaling interface to Session Manager.

```
change node-names ip
                                   IP NODE NAMES
    Name              IP Address
 procr              10.10.8.67
 SM100              10.10.3.55
 default            0.0.0.0
```

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-region** and **Inter-region**) is set to yes to allow audio traffic to be sent directly between endpoints without using gateway VoIP resources.
- 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** was used.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avaya.com
    Name: Default NR
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 35000                     IP Audio Hairpinning? n
   UDP Port Max: 50001
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
```

5.4. Administer IP Codec Set

Use the **change ip-codec-set** command 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's supported by Motto VoIP were configured, namely **G.711A, G729** and **G711MU**.

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

Motto VoIP only supports pass-through for transmission of fax. Navigate to **Page 2** to configure pass-through by setting the **Fax Mode** to **pass-through** as shown below.

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

                          IP Codec Set

                              Allow Direct-IP Multimedia? n

                    Mode                Redundancy
    FAX             pass-through         0
    Modem           off                  0
    TDD/TTY         US                   3
    Clear-channel   n                    0
```

5.5. Administer SIP Signaling Groups

Add a signaling group and trunk group for inbound and outbound PSTN calls to Motto VoIP SIP Trunk Service and configure using TCP (Transmission Control Protocol) and tcp port of 5060. Configure the **Signaling Group** using the **add signaling-group n** command, where **n** is an available signaling group:

- Set the **Group Type** field to **sip**.
- The **Transport Method** field is set to **tcp**.
- Set the **Near-end Node Name** to the processor interface (node name **procr**). This value is taken from the **IP Node Names** form shown in **Section 5.2**.
- Set the **Far-end Node Name** to the node name defined for the Session Manager (node name (**SM100**), also shown in **Section 5.2**.
- Ensure that the recommended TCP port value of **5060** is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- In the **Far-end Network Region** field, enter the IP Network Region configured in **Section 5.3**. This field logically establishes the far-end for calls using this signaling group as network region **1**.
- The **Direct IP-IP Audio Connections** field is set to **y**.
- The **Direct IP-IP Early Media** field is set to **n**.
- The **DTMF over IP** field should remain set to the default value of **rtp-payload**. This value enables Communication Manager to send DTMF transmissions using RFC 2833.

The default values for the other fields may be used.

```
add signaling-group 1
                                SIGNALING GROUP

 Group Number: 1              Group Type: sip
                        Transport Method: tcp
  IMS Enabled? 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? n                       Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 6
```

5.6. Administer SIP Trunk Group

A trunk group is associated with the signaling group described in **Section 5.5**. Configure the trunk group using the **add trunk-group n** command, where **n** is an available trunk group. 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, i.e. **101**.
- The **Direction** is set to **two-way** to allow incoming and outgoing calls.
- Set the **Service Type** field to **public-ntwrk**.
- Specify the signaling 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: SIP to SM100                  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

                                                       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 Motto VoIP. This value defines the interval that subsequent INVITEs must be sent to keep the active session alive. For the compliance testing, the value of **1800** seconds was used.

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

TRUNK PARAMETERS
     Unicode Name: auto
                                            Redirect On OPTIM Failure: 5000
            SCCAN? n                               Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 1800
 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 prevents the number to be sent to Motto VoIP with the + used in the E164 numbering format.

```
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

                              Modify Tandem Calling Number:
```

On **Page 4** of this form:

- Set **Send Transferring Party Information** to **y** to ensure that the transferring party number is sent. This information is used by the Motto VoIP network for call transfer.
- Set **Network Call Redirection** to **n**.
- Set **Send Diversion Header** to **n** to remove the Diversion Header. This information is not used and increases the size of the INVITE unnecessarily.
- Set **Support Request History** to **n** to ensure the History-Info Header is not sent. This information is not used and increases the size of the INVITE unnecessarily.
- Set the **Telephone Event Payload Type** to **101** to match the value preferred by Motto VoIP.
- Set **Always Use re-INVITE for Display Updates** to **y** as the most effective method employed by Communication Manager of modifying an existing dialogue.
- 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 Number? n
      Send Transferring Party Information? y
                 Network Call Redirection? n
                    Send Diversion Header? n
                  Support Request History? n
             Telephone Event Payload Type: 101

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

5.7. Administer Calling Party Number Information

In this section the Calling Party Number sent when making a call using the SIP trunk is specified.

5.7.1. Set Private Numbering

Use the **change private-numbering 0** command to configure Communication Manager to send the calling party number. In the sample configuration, all stations with a **4**-digit extension beginning with **6** will send the calling party number **31457xxxxxxx** to Motto VoIP SIP Trunk Service. This calling party number will be sent in the SIP From, Contact and PAI headers, and displayed on display-equipped PSTN telephones. Public DID numbers have been masked for security purposes.

```
change private-unknown-numbering 0                          Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext            Trk        CPN             CPN
Len Code           Grp(s)     Prefix          Len
                                                 Total Administered: 1
 4  6             1           31457xxxxxx     11    Maximum Entries: 240
```

5.8. Administer Route Selection for Outbound Calls

In these Application Notes, the Automatic Route Selection (ARS) feature was used to route outbound calls via the SIP trunk to Motto VoIP SIP Trunk Service. In the sample configuration, the single digit **9** is used as the ARS access code. Avaya telephone users will dial **9** to reach an outside line. Use the **change dialplan analysis** command to define a dialed string beginning with **9** of length **1** as a feature access code (**fac**).

```
change dialplan analysis                                        Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all          Percent Full: 2

    Dialed   Total  Call      Dialed   Total  Call      Dialed   Total  Call
    String   Length Type      String   Length Type      String   Length Type
   1           3   dac
   2           4   ext
   60          4   ext
   61          4   ext
   7           1   fac
   8           4   ext
   9           1   fac
   *           3   fac
   #           3   fac
```

Use the **change feature-access-codes** command to configure or observe **9** as the **Auto Route Selection (ARS) - Access Code 1.**

```
change feature-access-codes                                    Page   1 of   9
                               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: *37
                       Answer Back Access Code: *12
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 7
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2: *99
                 Automatic Callback Activation:        Deactivation:
Call Forwarding Activation Busy/DA: *87    All: *88    Deactivation: #88
   Call Forwarding Enhanced Status:        Act:        Deactivation:
```

Use the **change ars analysis** command to configure the routing of dialed digits following the first digit 9. A small sample of dial patterns are illustrated here. Further administration of ARS is beyond the scope of these Application Notes. The example entries shown will match outgoing calls to numbers beginning **0** or **00**. Calls are sent to **Route Pattern 1**, which contains the previously configured SIP Trunk Group.

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

          Dialed           Total      Route     Call   Node  ANI
          String         Min  Max  Pattern     Type   Num   Reqd
    0                    10    11    1         pubu         n
    00                   13    14    1         pubu         n
```

Use the **change route-pattern** command 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.

```
change route-pattern 1                                              Page   1 of   3
                    Pattern Number: 1   Pattern Name: tosm100
                             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 DID calls to the proper Communication Manager extension(s). The incoming digits sent in the INVITE message from Motto VoIP 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 Motto VoIP correlate to the internal extensions assigned within Communication Manager. The **change inc-call-handling-trmt trunk-group 1** command is used to translate numbers **+31457nnnnn0** to **+31457nnnnn5** to the 4 digit extension by deleting **all** of the incoming digits and inserting the extension number. Note that the significant digits beyond the city code have been obscured.

```
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    11 31457nnnnn0     all 6100
 public-ntwrk    11 31457nnnnn1     all 6102
 public-ntwrk    11 31457nnnnn2     all 6003
 public-ntwrk    11 31457nnnnn3     all 6004
 public-ntwrk    11 31457nnnnn4     all 6104
 public-ntwrk    11 31457nnnnn5     all 6006
```

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 6100. 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. **0035386nnnnnnn**).
- 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 6100                    Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 6100             EC500         -     0035386nnnnnnn   1          1
                                -
```

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

6. Configuring Avaya Aura® Session Manager

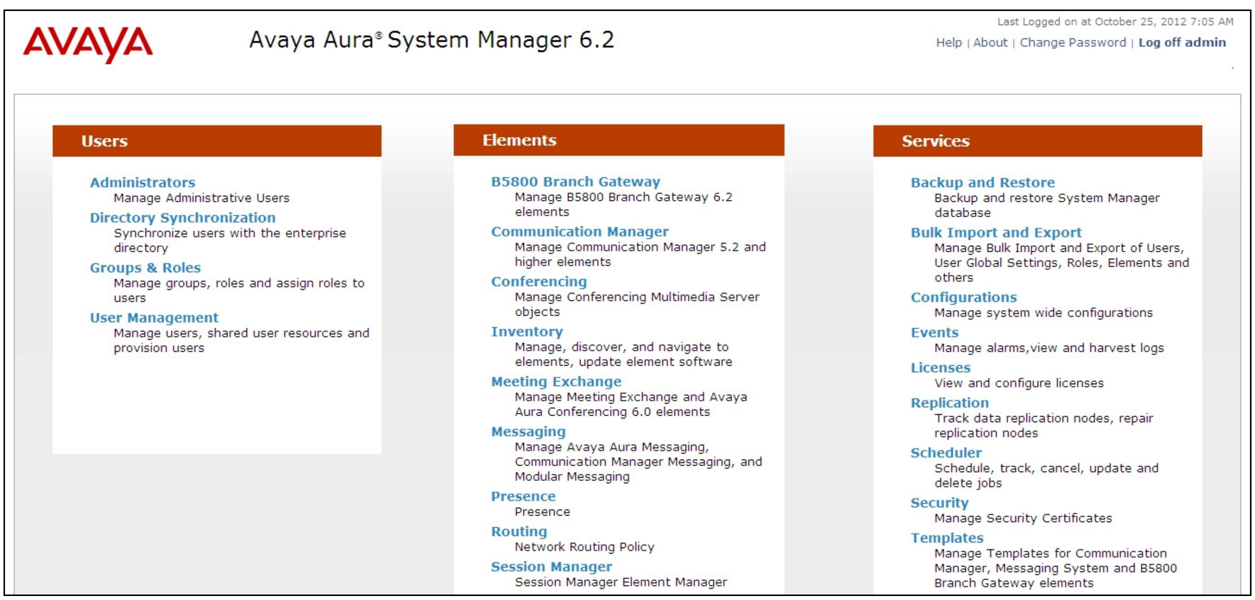
This section provides the procedures for configuring Session Manager. Session Manager is configured via System Manager. The procedures include the following areas:

- Log in to Avaya Aura® System Manager.
- Administer SIP domain.
- Administer SIP Location.
- Administer Adaptations.
- Administer SIP Entities.
- Administer Entity Links.
- Administer Routing Policies.
- Administer Dial Patterns.

It may not be necessary to create all the items above when creating a connection to the service provider since some of these items would have already been defined as part of the initial Session Manager installation. This includes items such as certain SIP domains, locations, SIP entities, and Session Manager itself. However, each item should be reviewed to verify the configuration.

6.1. Log in to Avaya Aura® System Manager

Session Manager configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL https://<ip-address>/SMGR, where <ip-address> is the IP address of System Manager. Log in with the appropriate credentials and click on **Log On** (not shown). The screen shown below is then displayed.



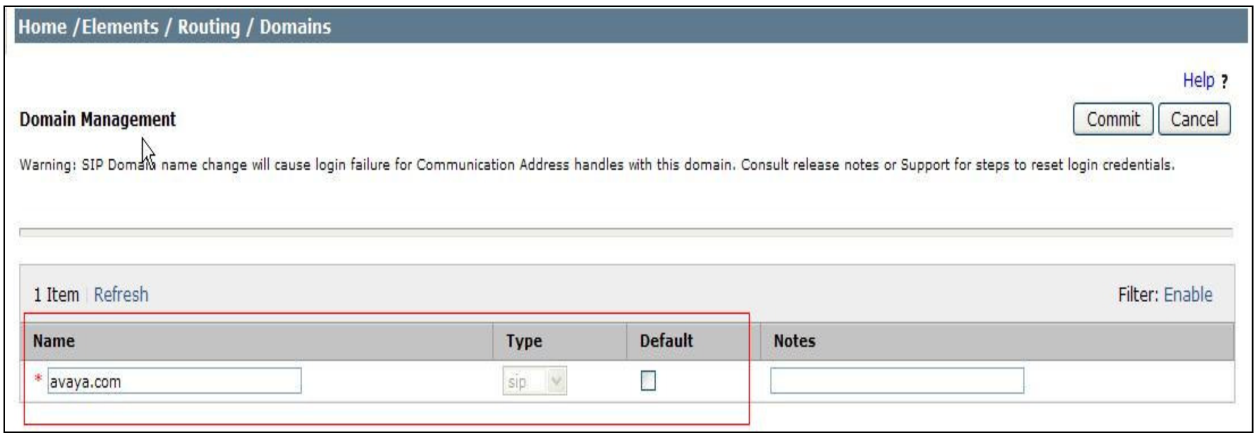
Most of the configuration items are performed in the Routing Element. Click on **Routing** in the Elements column shown above to bring up the Introduction to Network Routing Policy screen (not shown).

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

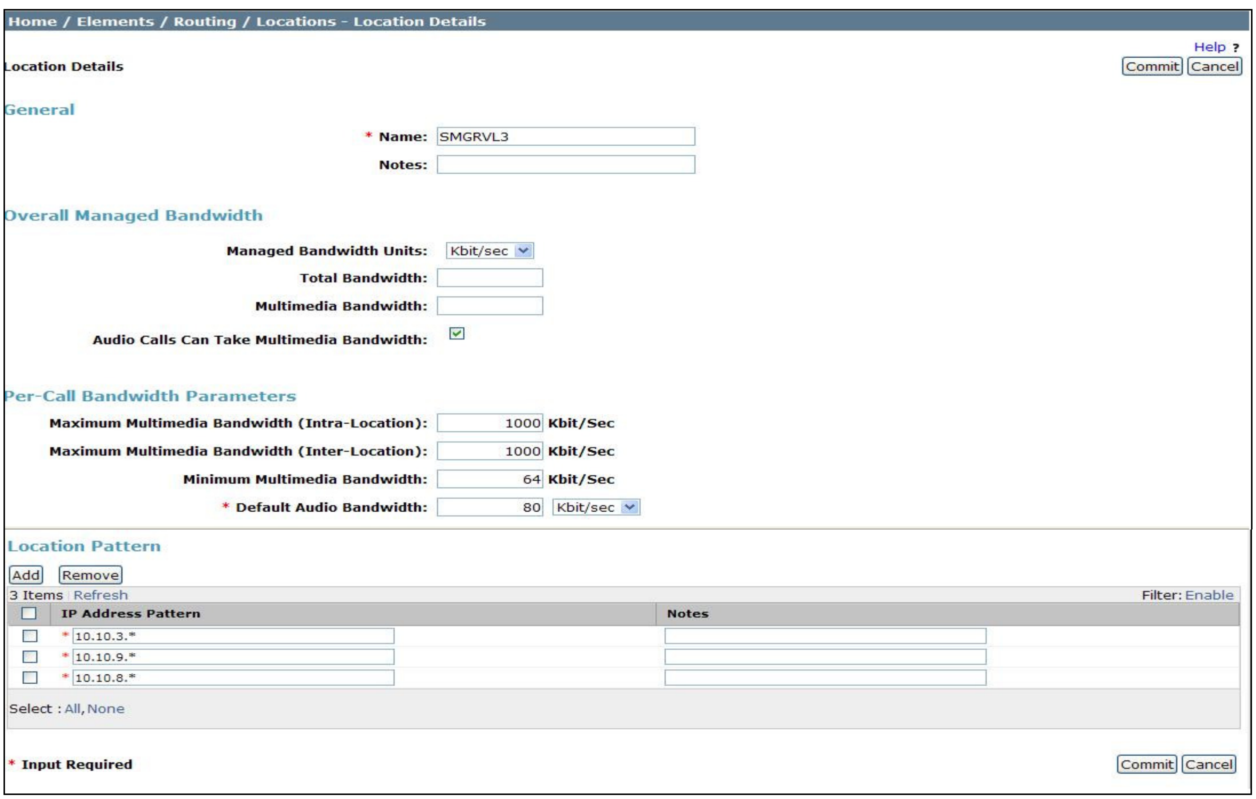
Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management and call admission control. To add a location, navigate to **Routing →Locations** in the left-hand navigation pane and click the **New** button in the right pane (not shown). 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 **SMGRVL3** defined for the compliance testing.



6.4. 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, 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:**

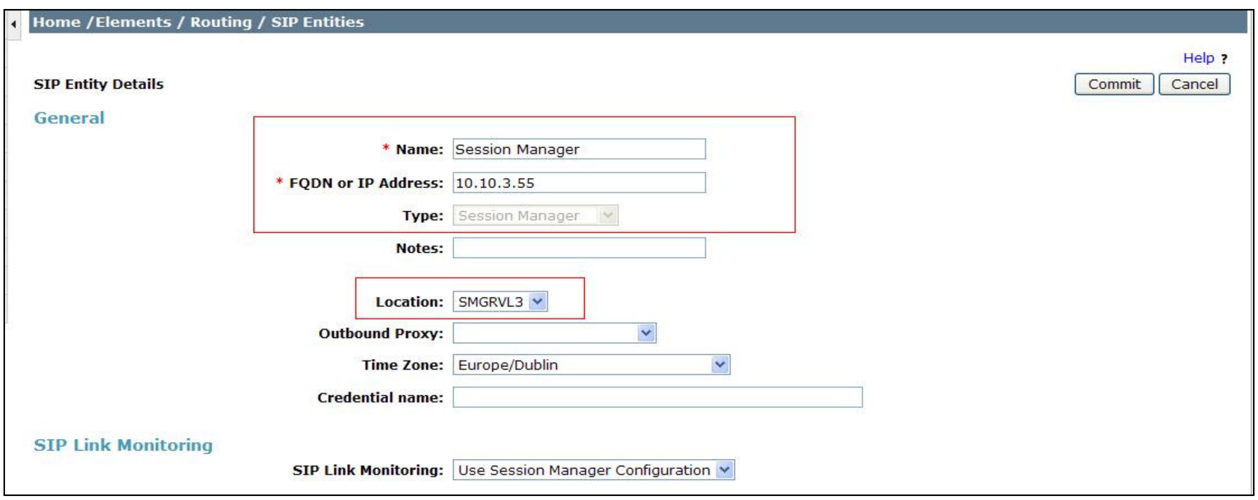
- In the **Name** field enter an informative name.
- In the **FQDN or IP Address** field enter the IP address of Session Manager or the signaling interface on the connecting system.
- In the **Type** field use **Session Manager** for a Session Manager SIP entity, **CM** for a Communication Manager SIP entity and **Gateway** for the SBC 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 three SIP Entities.

- Session Manager SIP Entity.
- Communication Manager SIP Entity.
- Avaya SBCE SIP Entity.

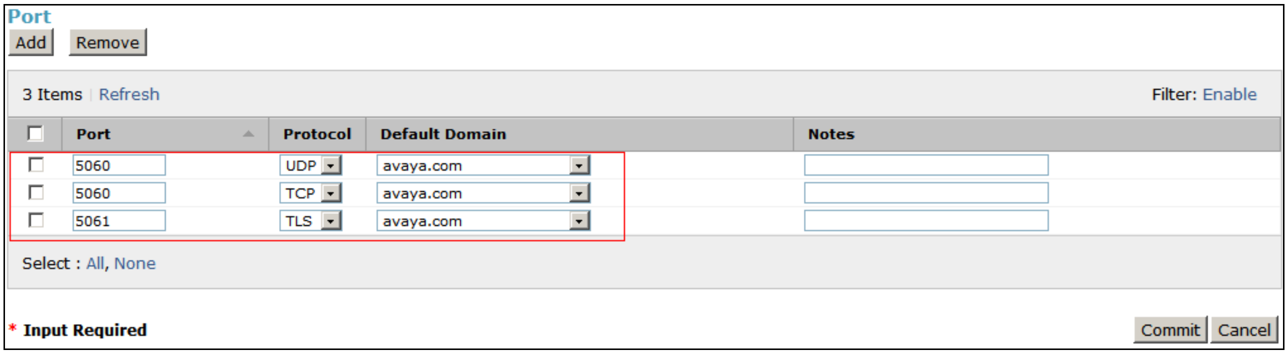
6.4.1. Avaya Aura® Session Manager SIP Entity

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 signaling interface.



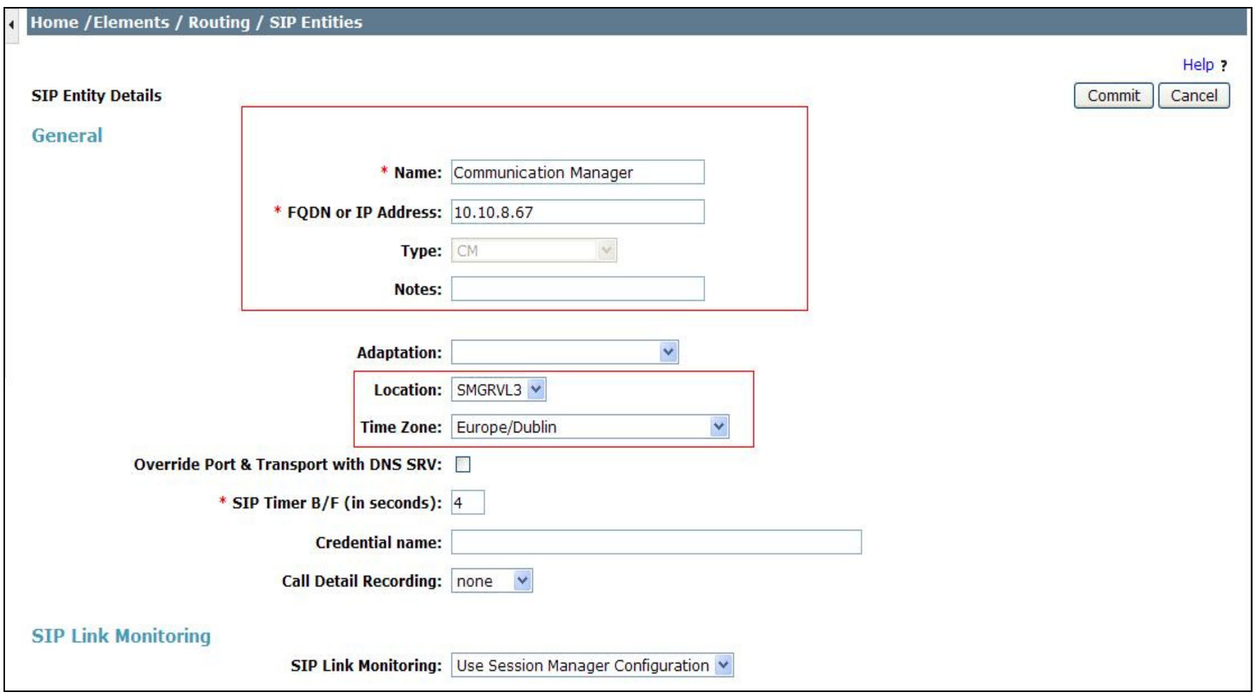
Session Manager must be configured with the port numbers on the protocols that will be used by the other SIP entities. To configure these scroll to the bottom of the page and under **Port**, click **Add**, then edit the fields in the resulting new row.

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



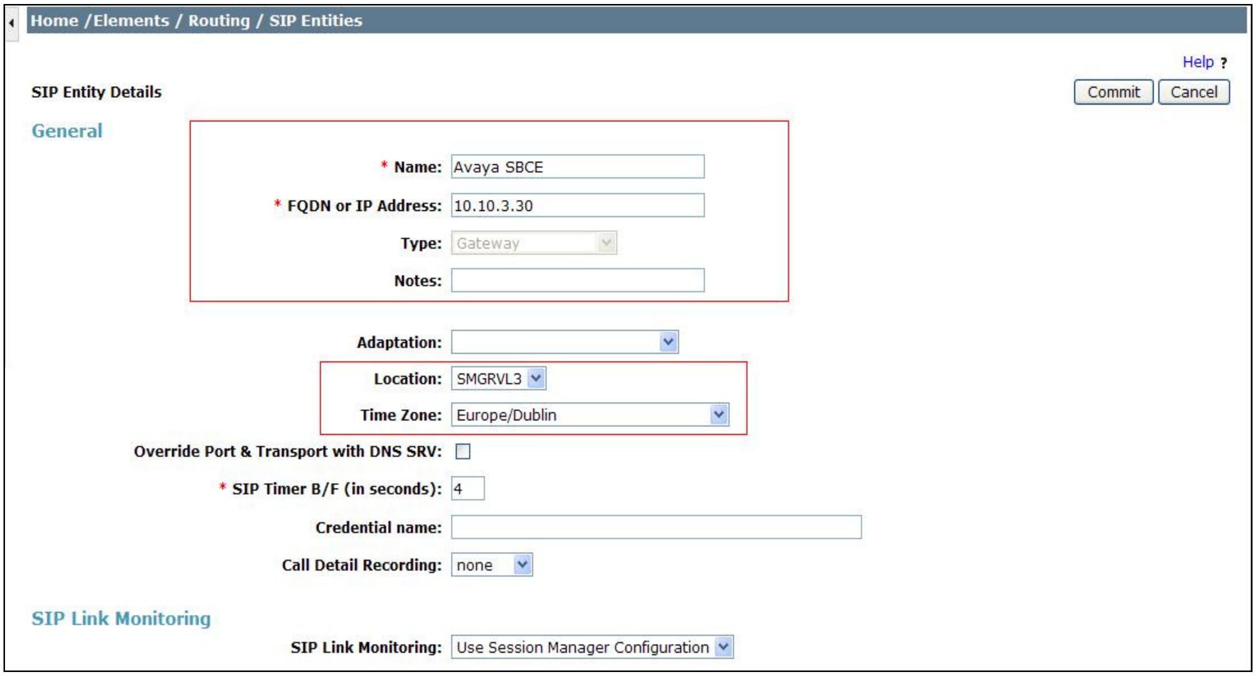
6.4.2. Avaya Aura® Communication Manager SIP Entity

The following screens show the SIP entity for Communication Manager. The **FQDN or IP Address** field is set to the IP address of the Interface that will be providing SIP signaling. The entity **Type** is set to **CM**. Set the location to that defined in **Section 6.3** and the **Time Zone** to the appropriate time zone.



6.4.3. Avaya Session Border Controller for Enterprise SIP Entities

The following screen shows the SIP entity for the Avaya SBCE used for routing calls. The **FQDN or IP Address** field is set to the IP address of the private interfaces administered in **Section 7** of this document. Set the location to that defined in **Section 6.3** and the **Time Zone** to the appropriate time zone.

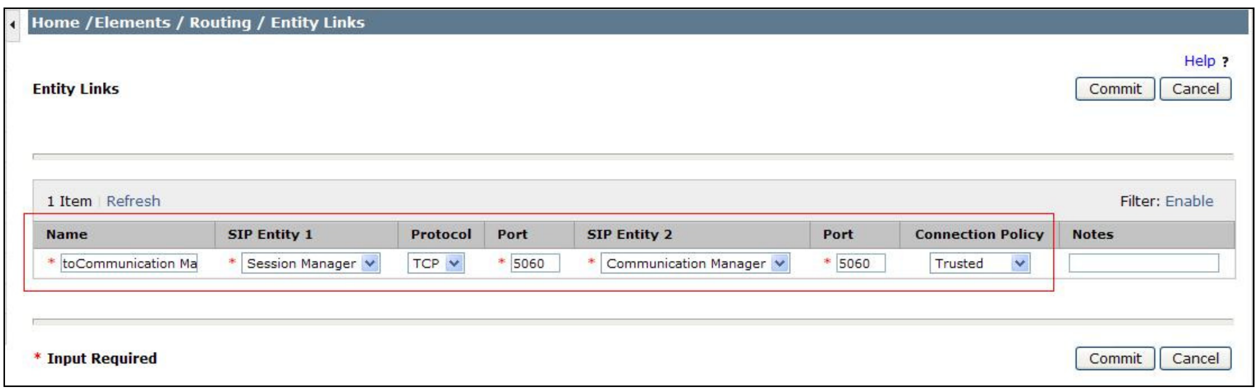


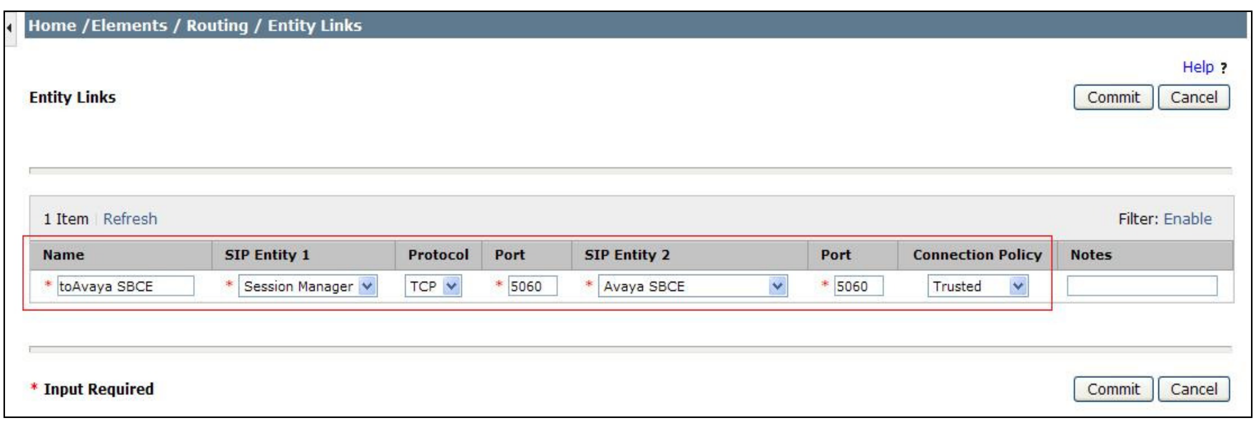
6.5. Administer Entity Links

A SIP trunk between 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 **SessionManager**.
- 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 6.4**.
- 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.
- 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.





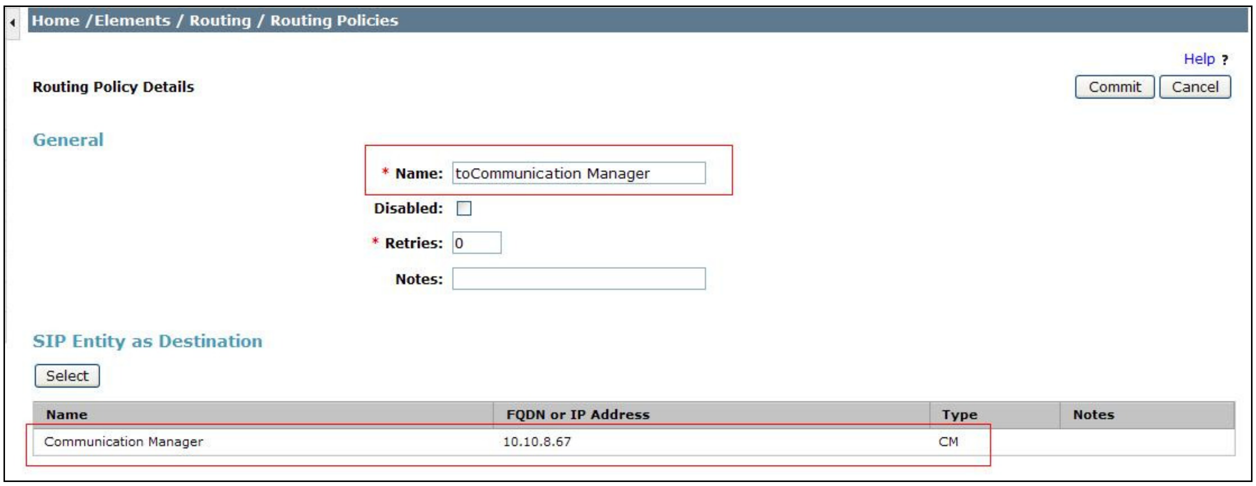
6.6. Administer Routing Policies

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).

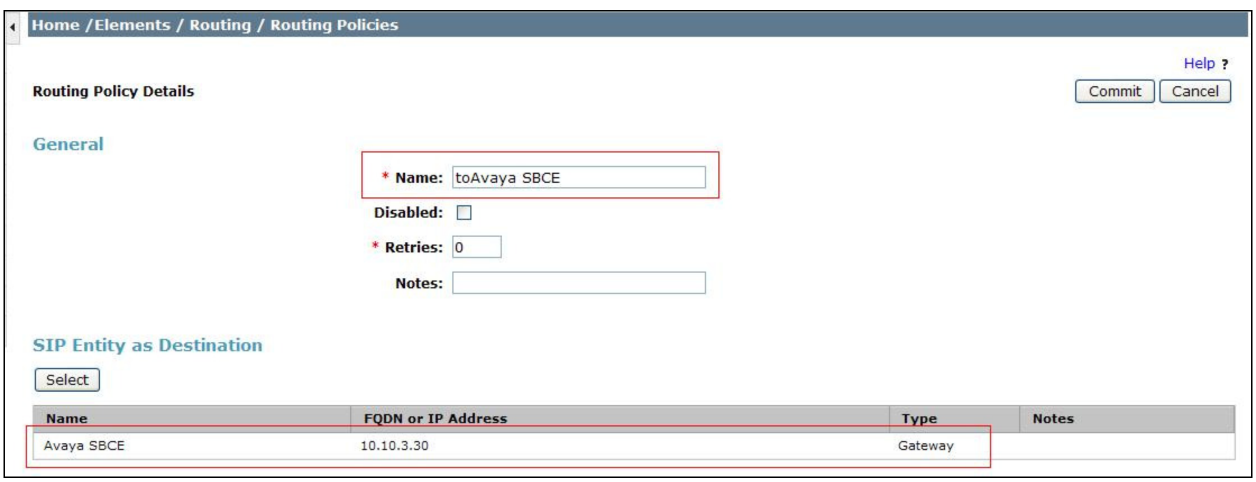
Under **General**:

- 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.

The following screen shows the routing policy for Communication Manager:



The following screens show the routing policy for Avaya SBCE:



6.7. Administer Dial Patterns

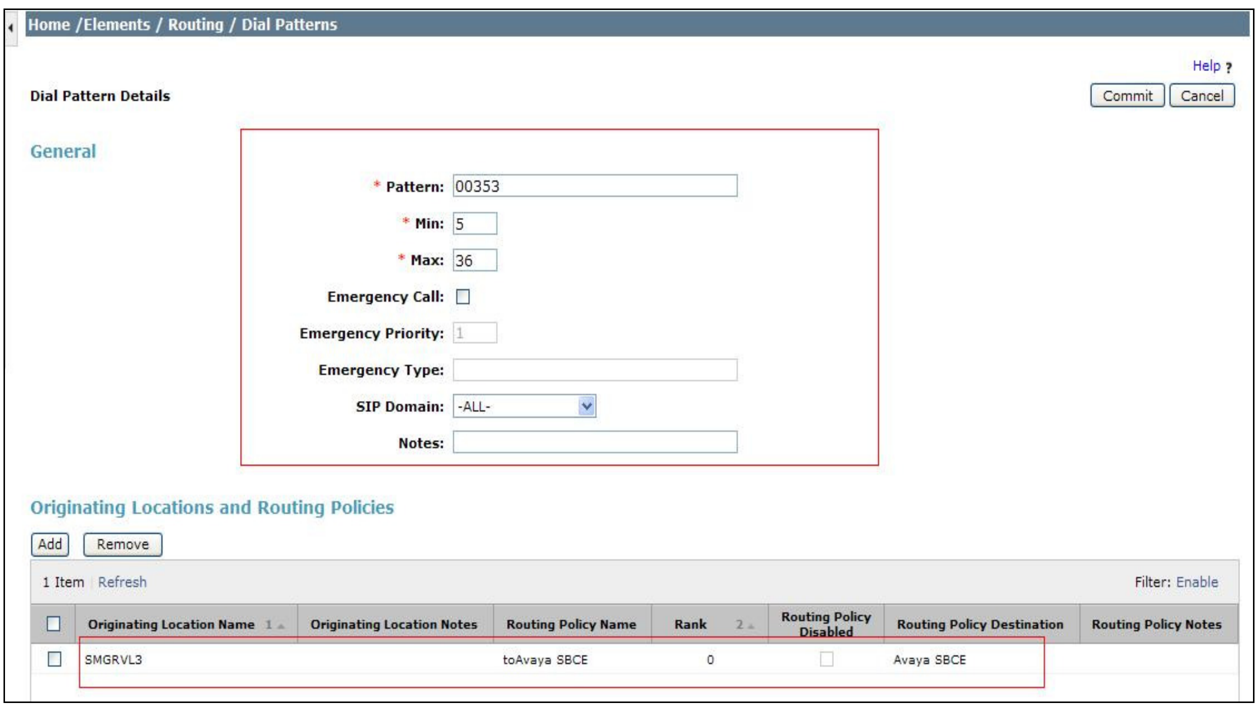
A dial pattern must be defined to direct calls to the appropriate telephony system. To configure a dial pattern select **Dial Patterns** on the left panel menu and then click on the **New** button (not shown).

Under **General**:

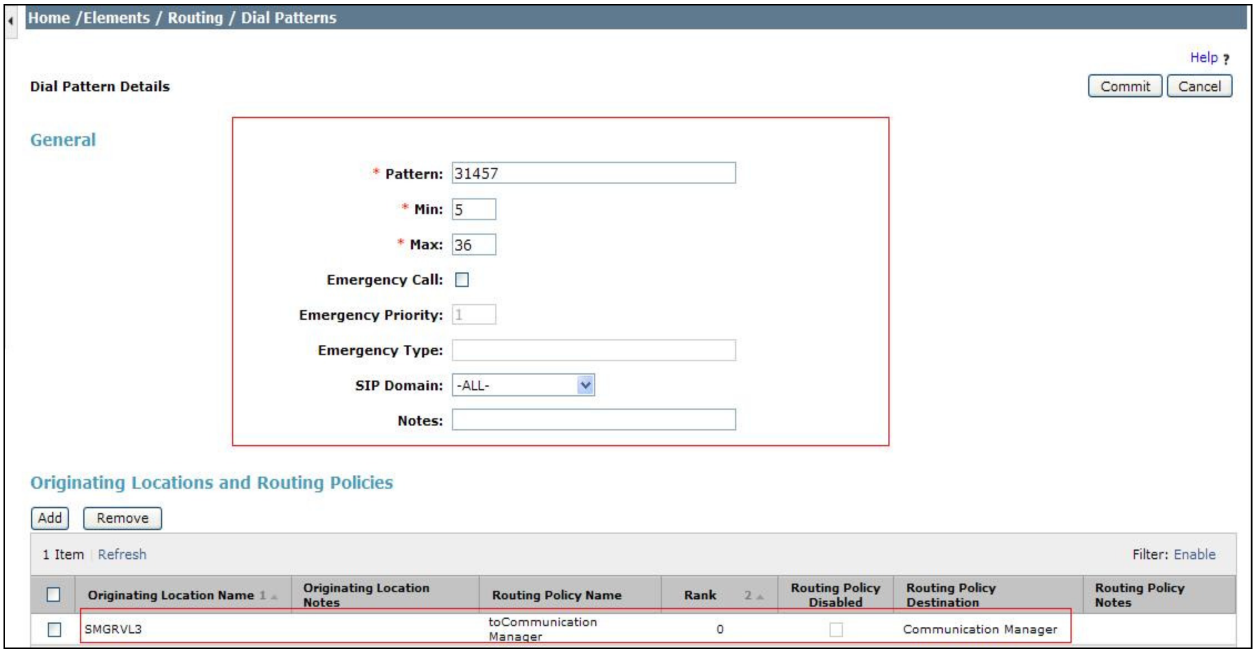
- In the **Pattern** field enter a dialed number or prefix to be matched.
- In the **Min** field enter the minimum length of the dialed number.
- In the **Max** field enter the maximum length of the dialed number.
- In the **SIP Domain** field select **–ALL-**.

Under **Originating Locations and Routing Policies**. Click **Add**, in the resulting screen (not shown) under **Originating Location** select **Locations** created in **Section 6.3** and under **Routing Policies** select one of the routing policies defined in **Section 6.6**. Click **Select** button to save (not shown).

The following screen shows an example dial pattern configured for the Avaya SBCE which will route the calls out to the PSTN via the Motto VoIP SIP Trunk Service.



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

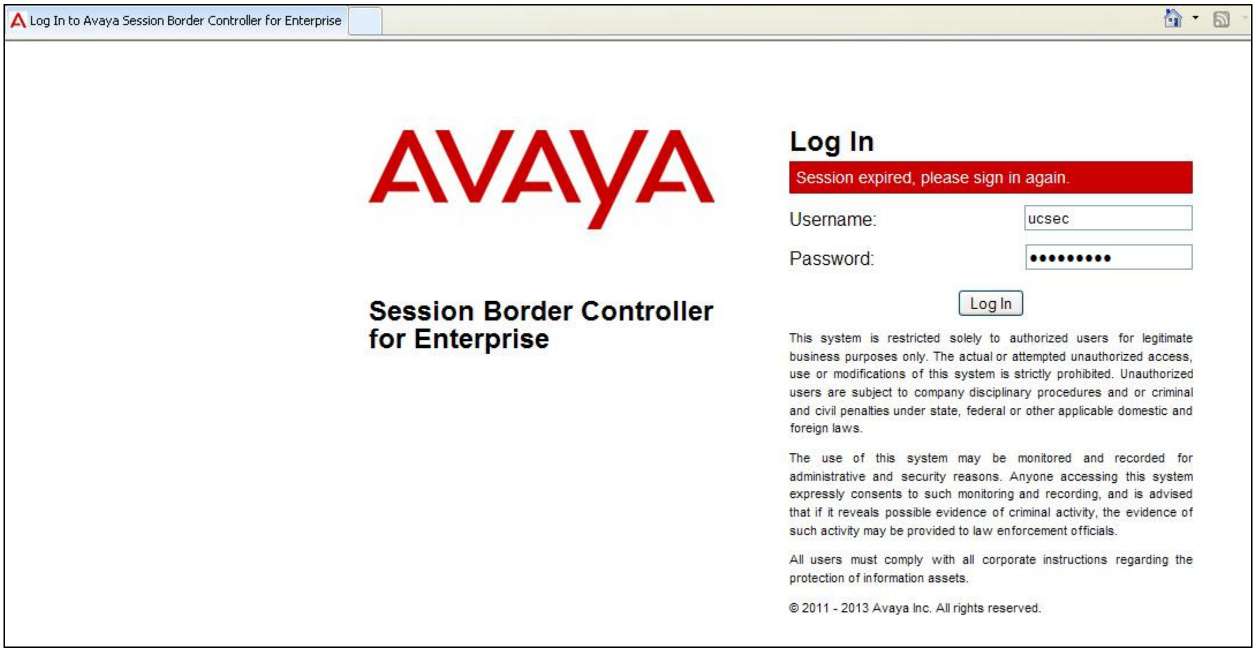


7. Configure Avaya Session Border Controller for Enterprise

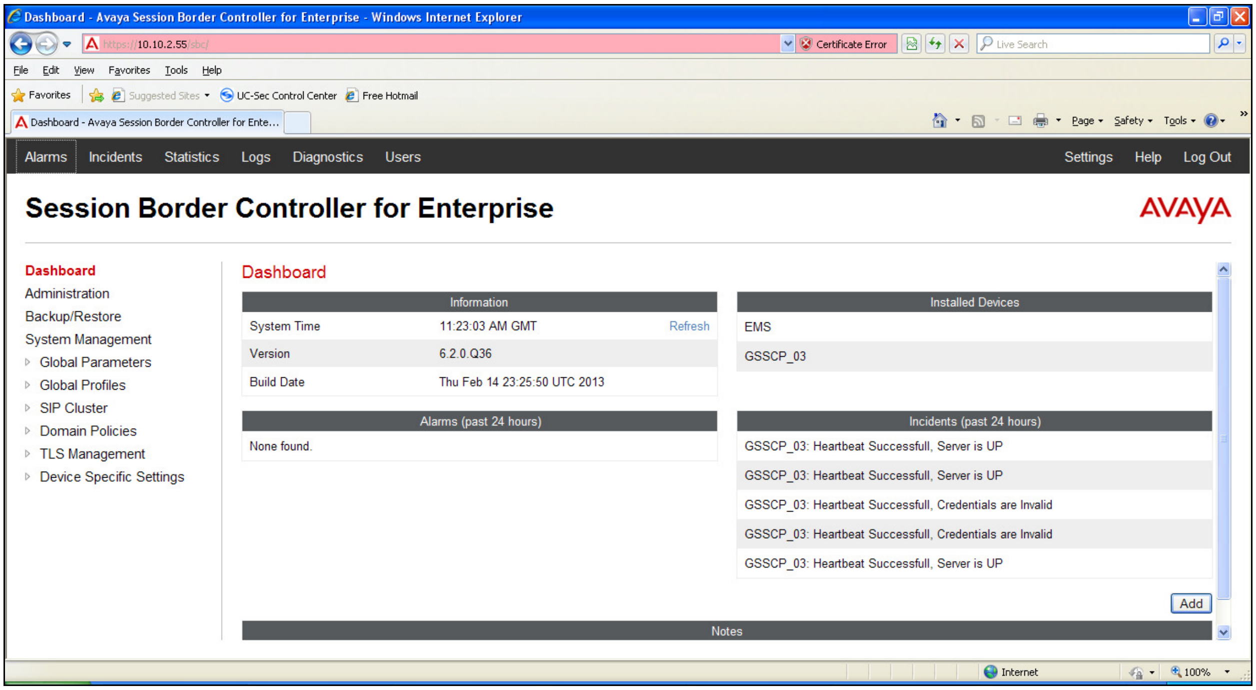
This section describes the configuration of the Avaya SBCE.

7.1. Accessing Avaya Session Border Controller for Enterprise

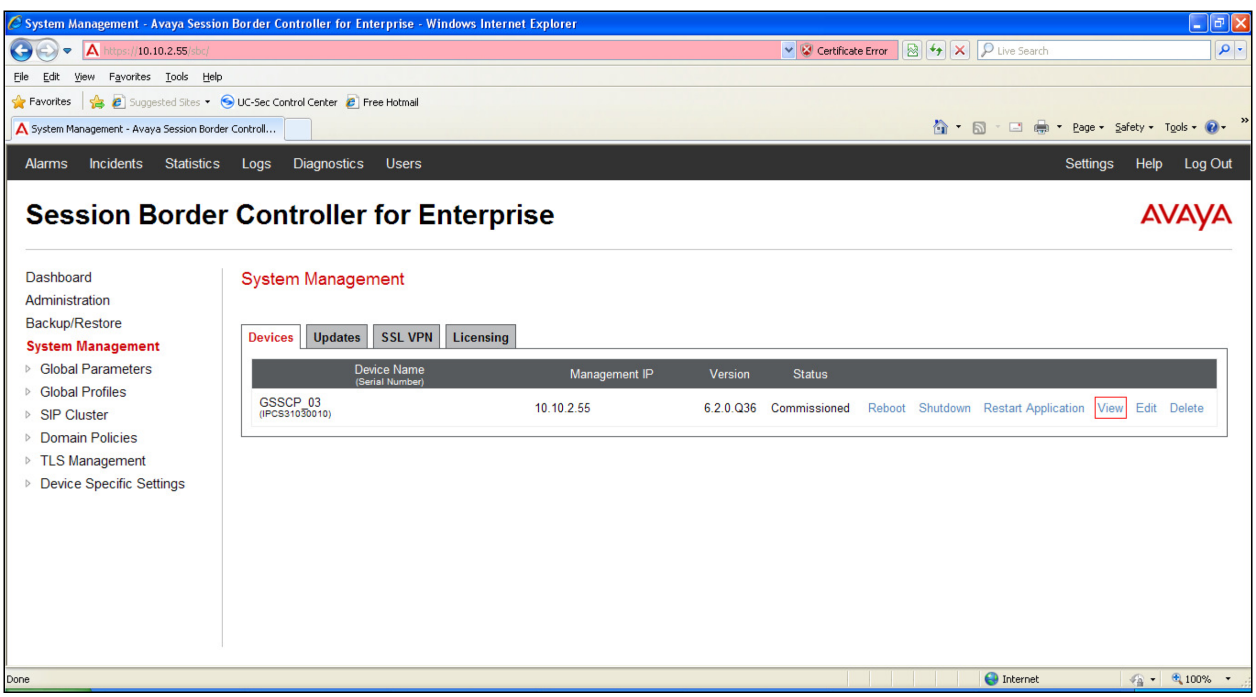
Access the Avaya SBCE using a web browser by entering the URL **https://<ip-address>**, where **<ip-address>** is the management IP address configured at installation and enter the **Username** and **Password**.



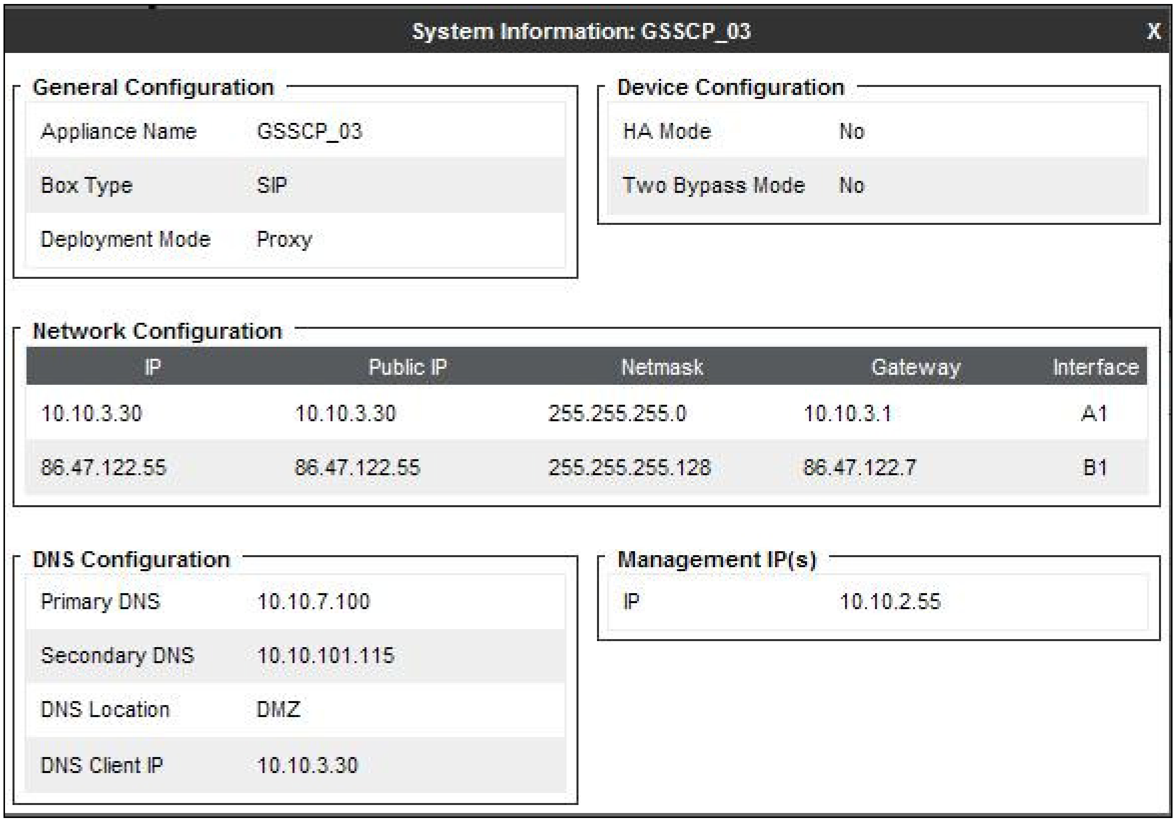
The main page of the Avaya SBCE will appear.



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 Settings** and **DNS Configuration** information.



7.2. Global Profiles

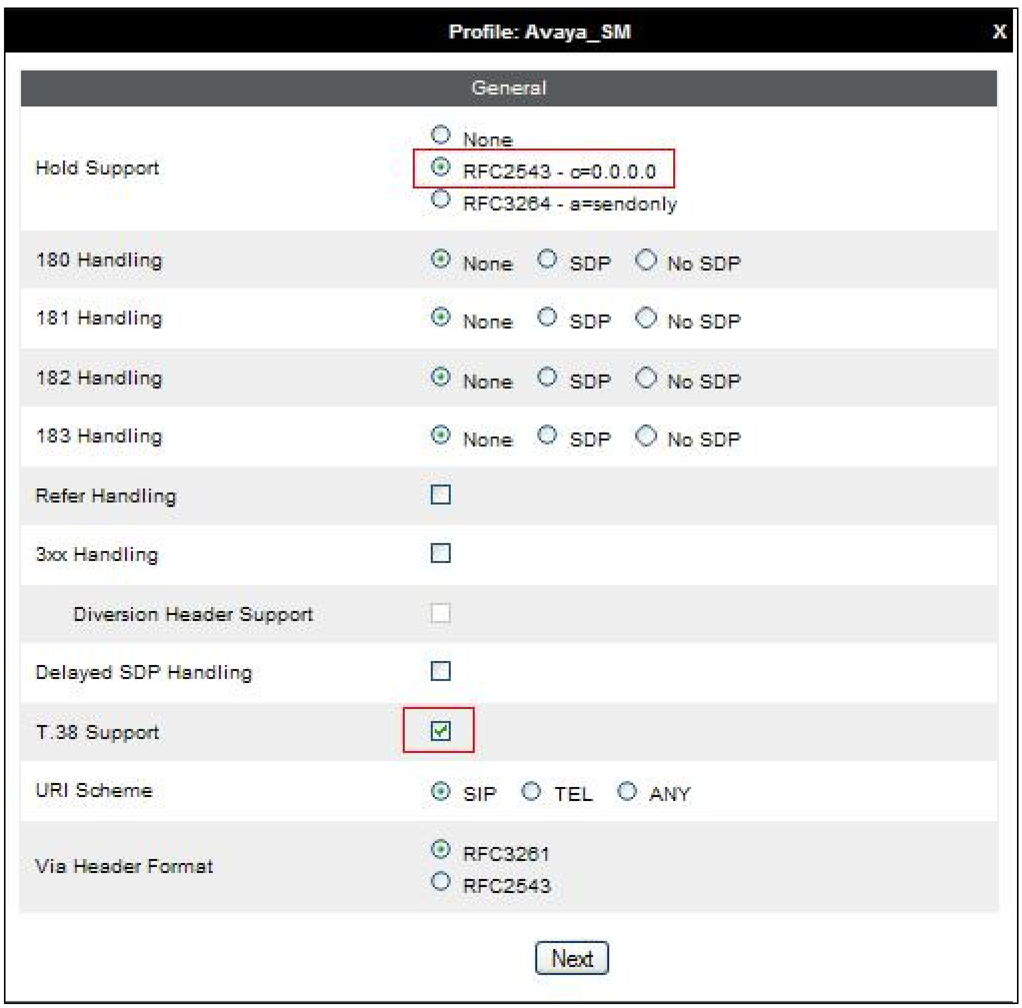
When selected, Global Profiles allows for configuration of parameters across all Avaya SBCE appliances.

7.2.1. Server Internetworking Avaya

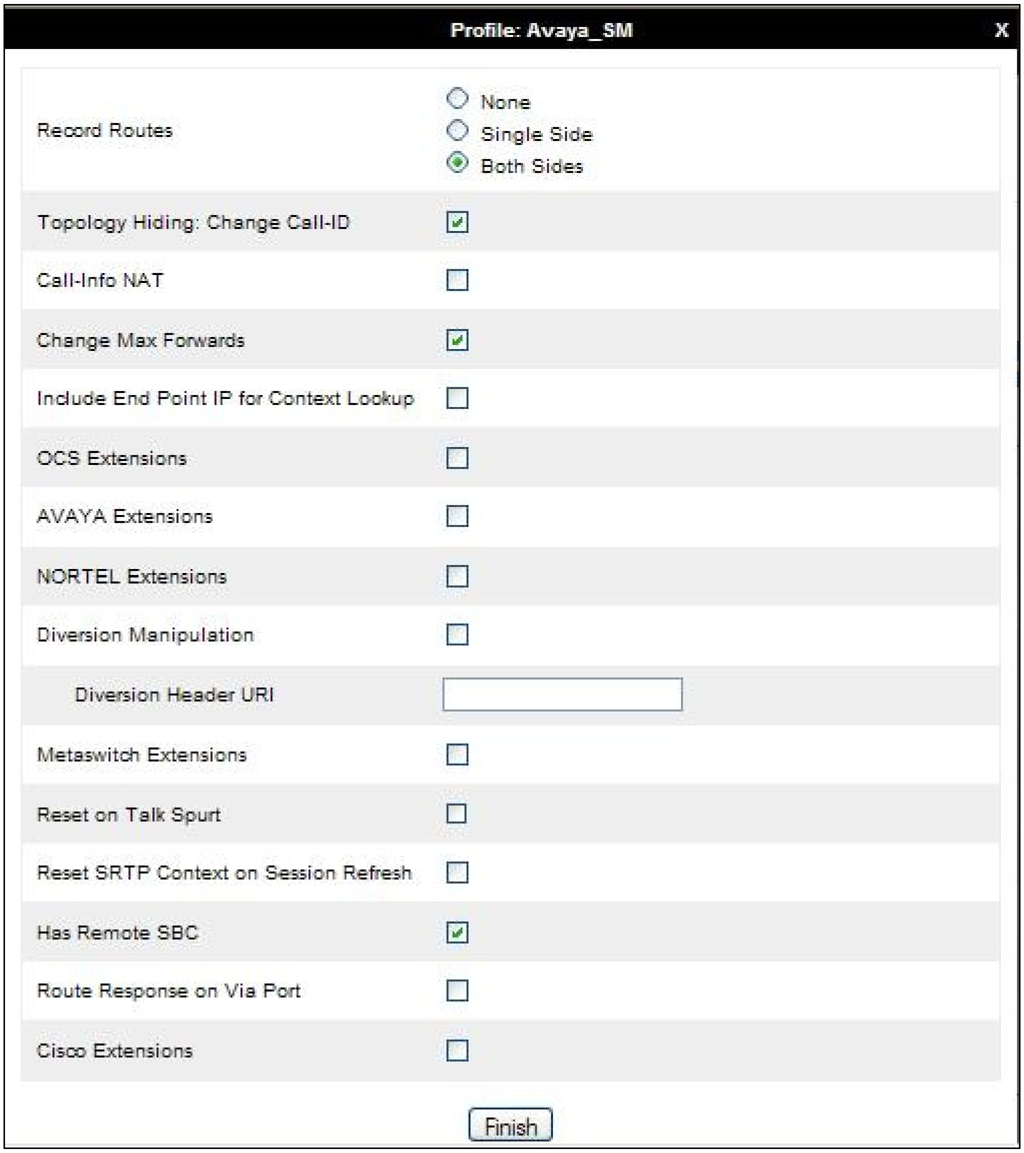
Server Internetworking allows you to configure and manage various SIP call server-specific capabilities such as call hold and T.38. From the left-hand menu select **Global Profiles → Server Interworking** and click on **Add**.

- Enter profile name such as **Avaya_SM** and click **Next** (Not Shown)
- Check **Hold Support= RFC2543**
- Check **T.38 Support** (not required but checked to avoid restriction on Avaya SBCE)
- 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**

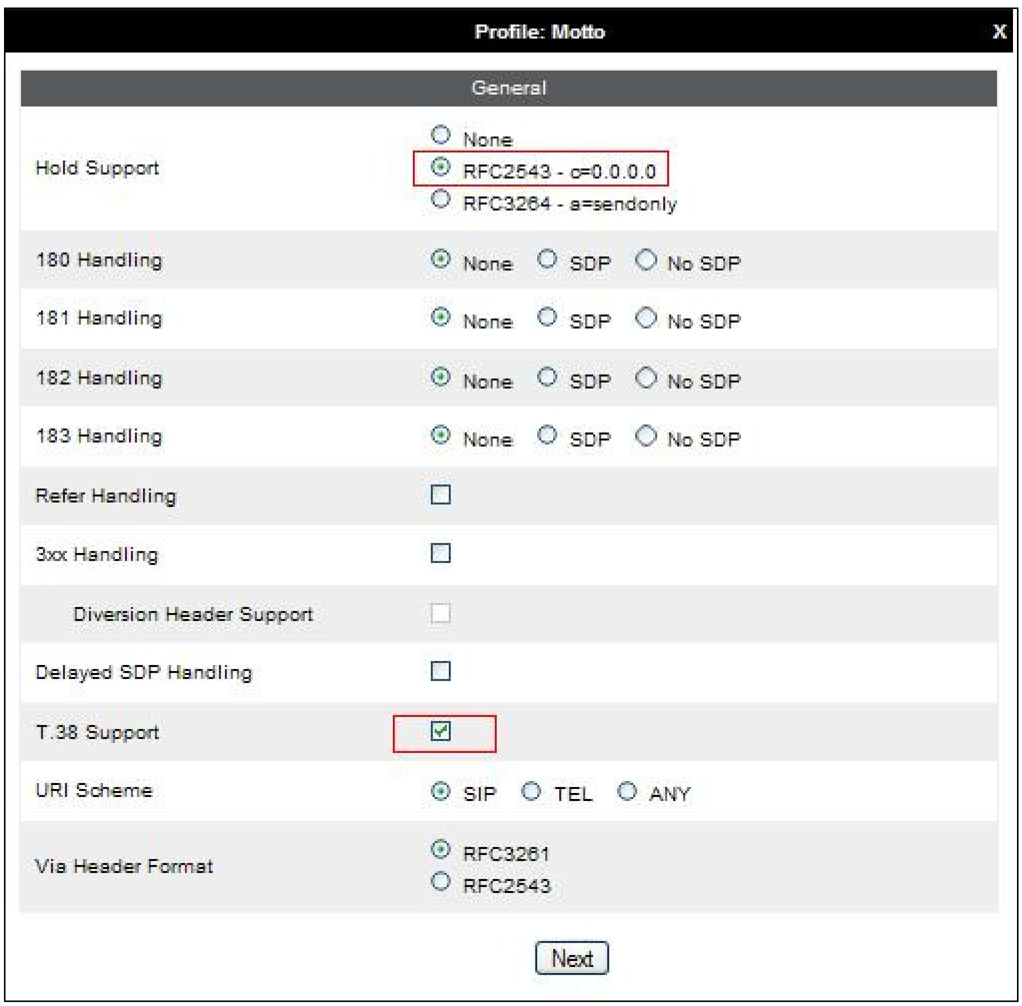


7.2.2. Server Internetworking – Motto VoIP

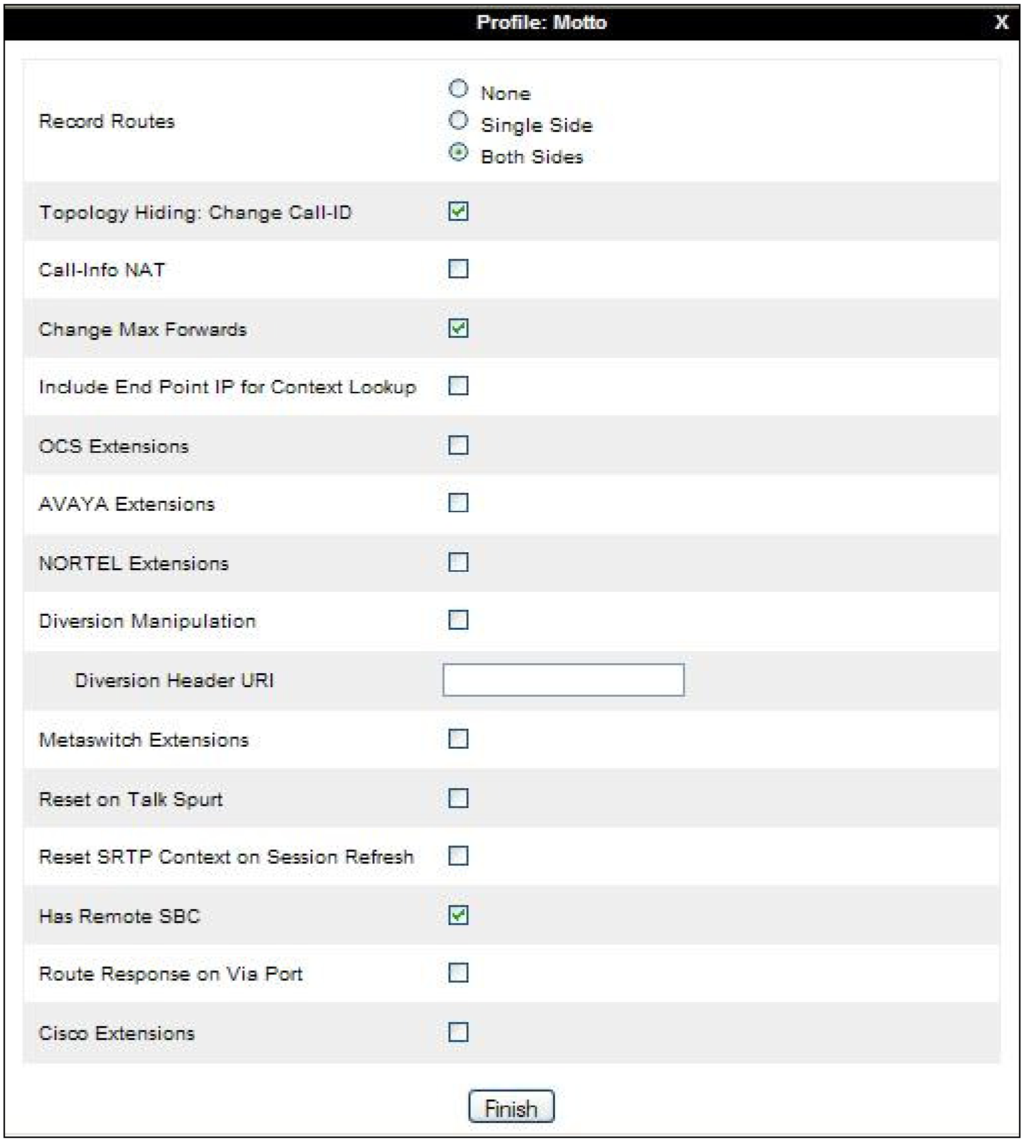
Server Internetworking allows you to configure and manage various SIP call server-specific capabilities such as call hold and T.38. From the lefthand menu select **Global Profiles → Server Interworking** and click on **Add**.

- Enter profile name such as **Motto** and click **Next** (Not Shown)
- Check **Hold Support= RFC2543**
- Check **T.38 Support** (not required but checked to avoid restriction on Avaya SBCE)
- 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.

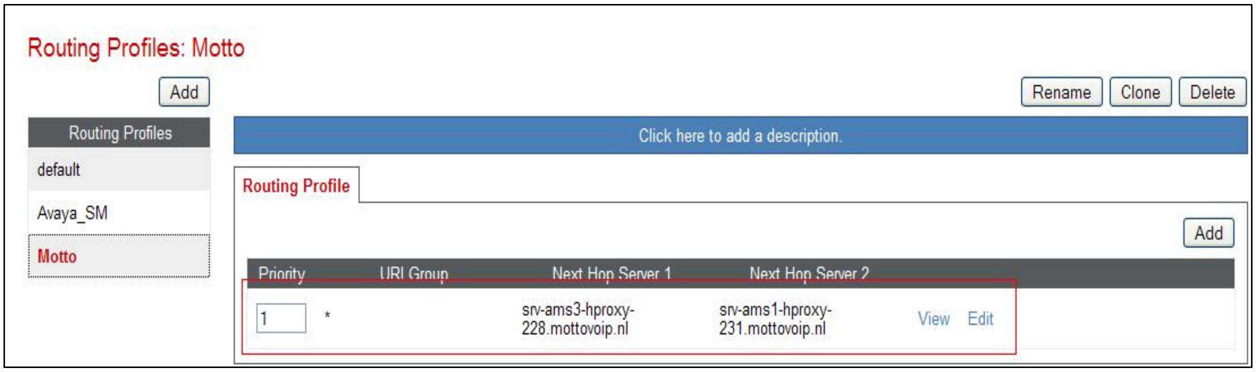
Create a Routing Profile for Session Manager and a Routing Profile for Motto VoIP. To add a routing profile, navigate to **Global Profiles → Routing** and select **Add**. Enter a **Profile Name** and click **Next** to continue.
In the new window that appears, enter the following values. Use default values 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
- **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
- **Use Next Hop for In-Dialog Messages**: Select only if there is no secondary Next Hopserver
- **Outgoing Transport:** Choose the protocol used for transporting outgoing signaling packets

Click **Finish**.

The following screen shows the Routing Profile to Session Manager.

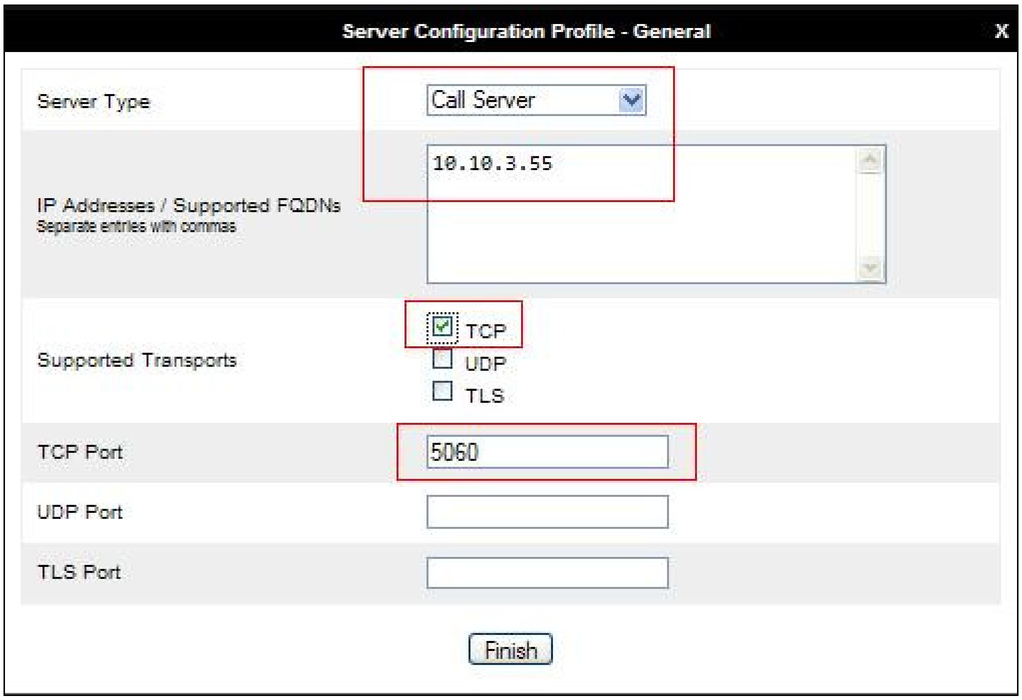
The following screen shows the Routing Profile to Motto VoIP.



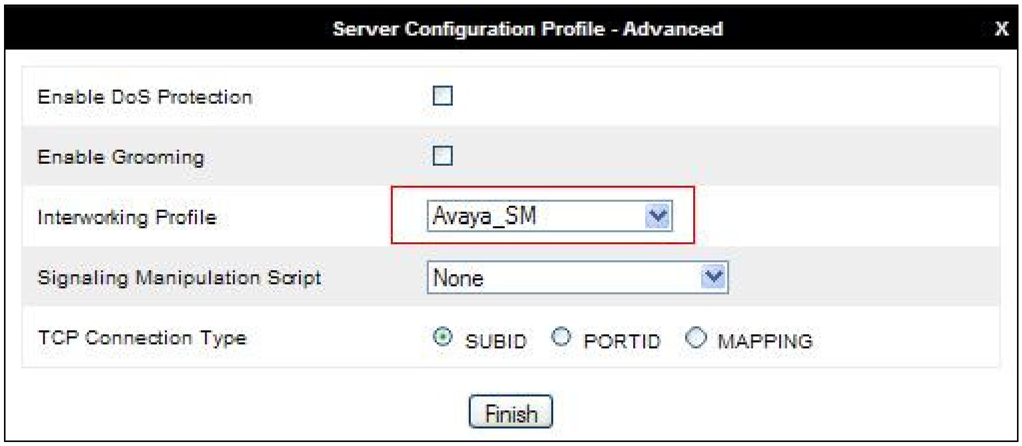
7.2.4. Server Configuration– Avaya Aura® Session Manager

The **Server Configuration** screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. Together, these tabs allow you to configure and manage various SIP call server-specific parameters such as TCP and UDP port assignments, IP Server type, heartbeat signaling parameters and some advanced options. From the lefthand menu select **Global Profiles → Server Configuration** and click on **Add**. Enter **Profile Name: Avaya_SM**. On the **Add Server Configuration Profile** tab, set the following:

- Select **Server Type** to be **Call Server**
- Enter **IP Addresses / Supported FQDNs** to **10.10.3.55** (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.



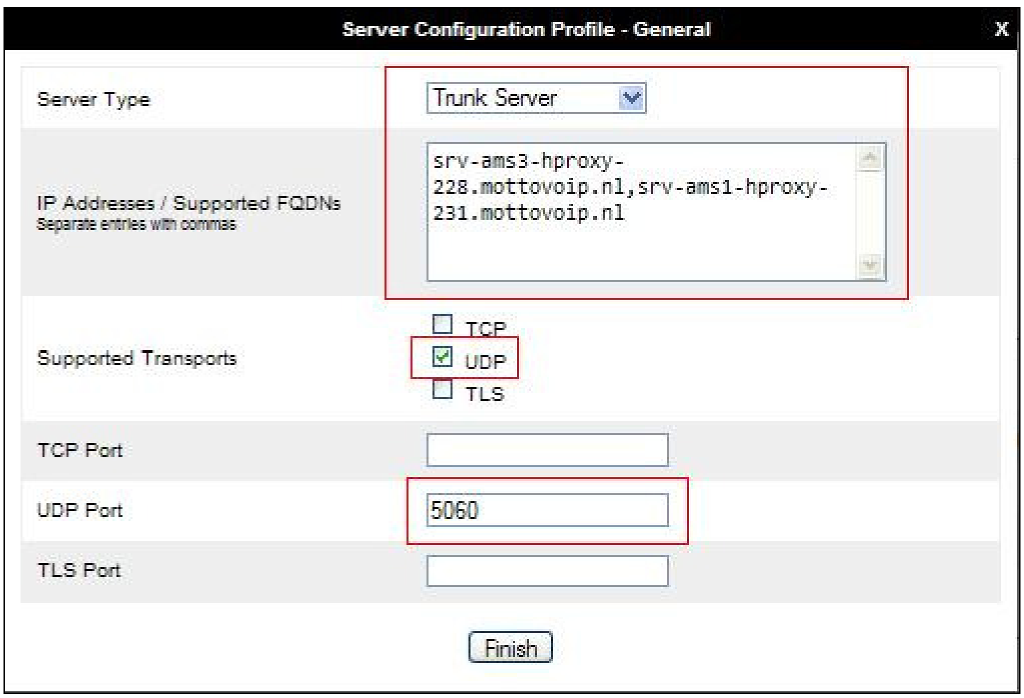
On the **Advanced** tab:
- Select **Avaya_SM** for **Interworking Profile**
- Click **Finish**



7.2.5. Server Configuration – Motto VoIP

The **Server Configuration** screen contains fourtabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. Together, these tabs allow you to configure and manage various SIP call server-specific parameters such as TCP and UDP port assignments, server type, heartbeat signaling parameters and some advanced options. From the left-hand menu select **Global Profiles → Server Configuration** and click on **Add**. Enter Name as **Motto**. On the **Add Server Configuration Profile** tab, set the following:

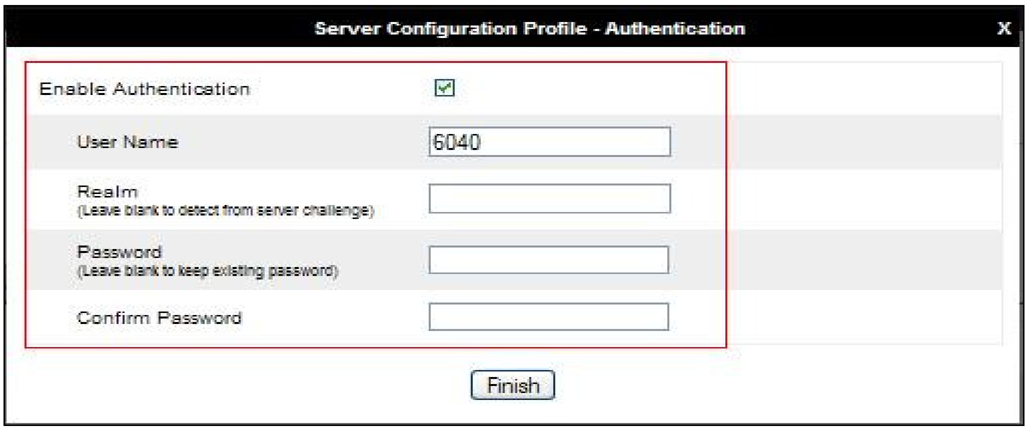
- Select **Server Type** as **Trunk Server**
- Enter the **FQDNs** of the SIP proxies to Motto VoIP
- **Supported Transports**: Check **UDP**
- **UDP Port: 5060**
- Click on **Next** (not shown)



In the new window that appears, enter the following values as Motto VoIP 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

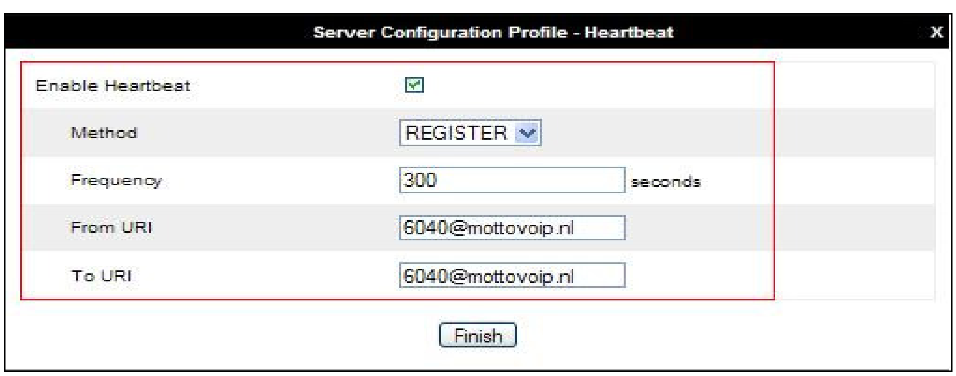
Click **Finish** to continue.



In the new window that appears, enter the following values. Use default values for all remaining fields:

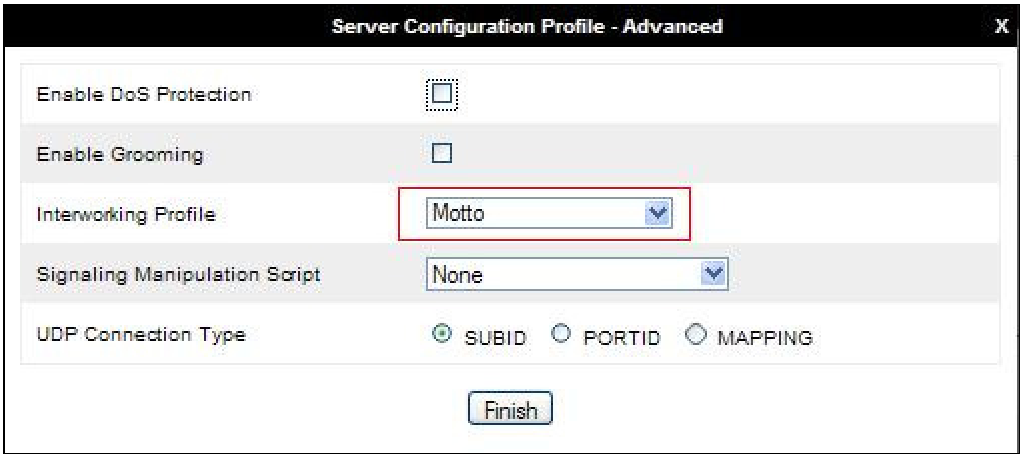
- **Enabled Heartbeat:** Checked
- **Method:** Select **REGISTER** from the drop-down box
- **Frequency:** Choose the desired frequency in seconds the Avaya SBCE will send SIP REGISTERS
- **From URI:** Enter an URI to be sent in the FROM header for SIP REGISTERS
- **TO URI:** Enter an URI to be sent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

Click **Next** to continue.



On the **Advanced** tab:

- Select **Motto** for **Interworking Profile**
- Click **Finish**

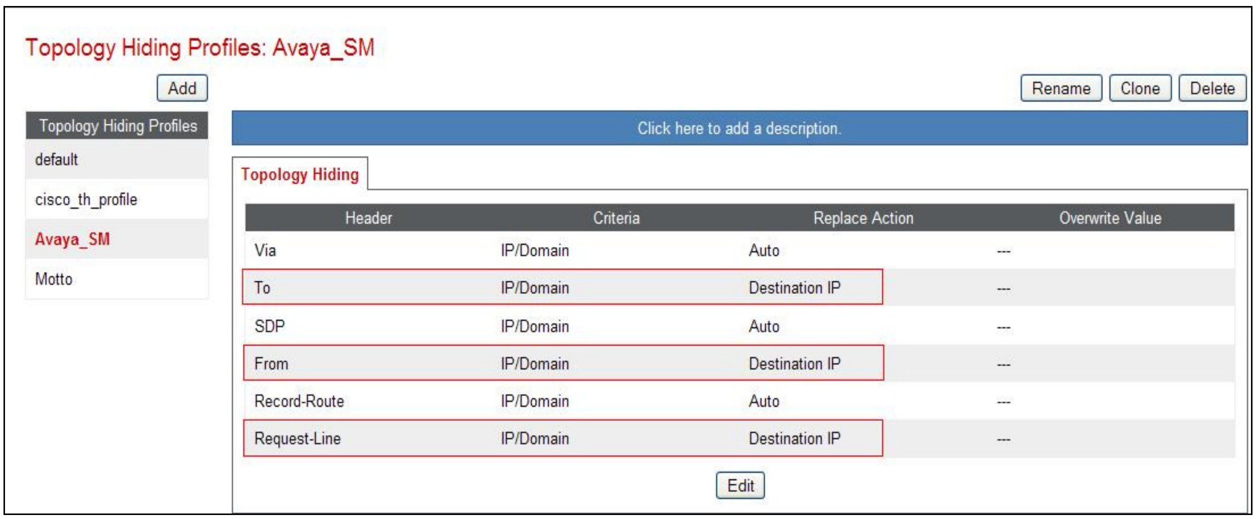


7.2.6. Topology Hiding – Avaya

The **Topology Hiding** screen manages how various source, destination and routing information in SIP and SDP message headers are substituted or changed to maintain the integrity of the network. It hides the topology of the enterprise network from external networks. Navigate to **Global Profiles → Topology Hiding** (not shown).

- Click **default** profile and select **Clone** (not shown)
- Enter Profile Name : **Avaya_SM**
- Under the **Header** field for **To**, **From** and **Request Line**, select **IP/Domain** under **Criteria** and **Destination IP** under **Replace Action**
- Click **Finish** (not shown)

The screen below is a result of the details configured above.

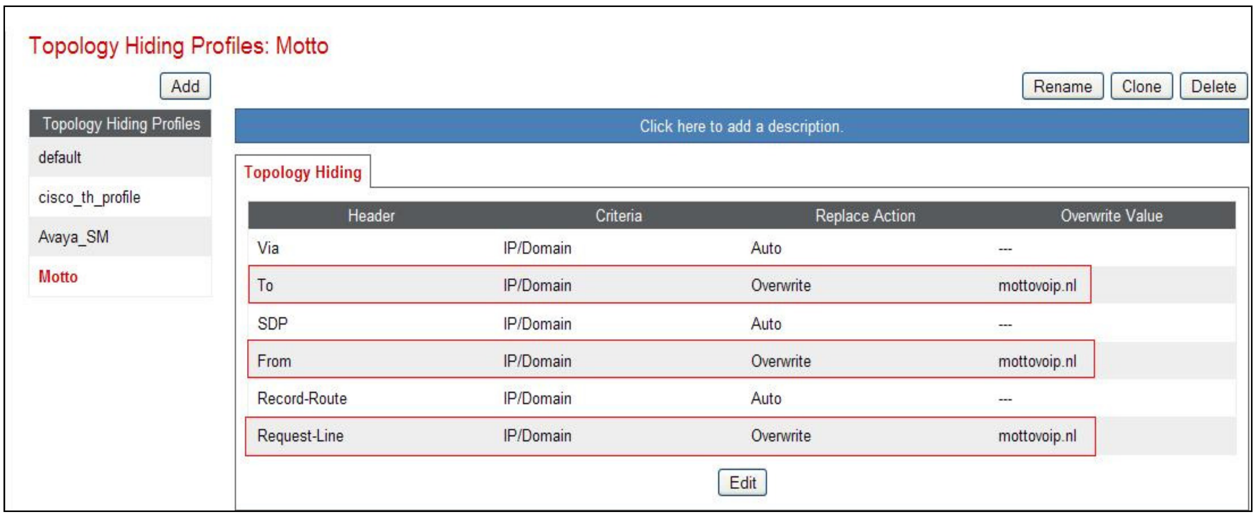


7.2.7. Topology Hiding – Motto VoIP

The **Topology Hiding** screen manages how various source, destination and routing information in SIP and SDP message headers are substituted or changed to maintain the integrity of the network. It hides the topology of the enterprise network from external networks. Navigate to **Global Profiles → Topology Hiding** (not shown).

- Click **default** profile and select **Clone** (not shown)
- Enter Profile Name : **Motto**
- Under the **Header** field for **To, From** and **Request Line**, select **IP/Domain** under **Criteria** and **Overwrite** under **Replace Action**. For **Override Value** type **mottovoip.nl**
- Click **Finish** (not shown)

The screen below is a result of the details configured above.



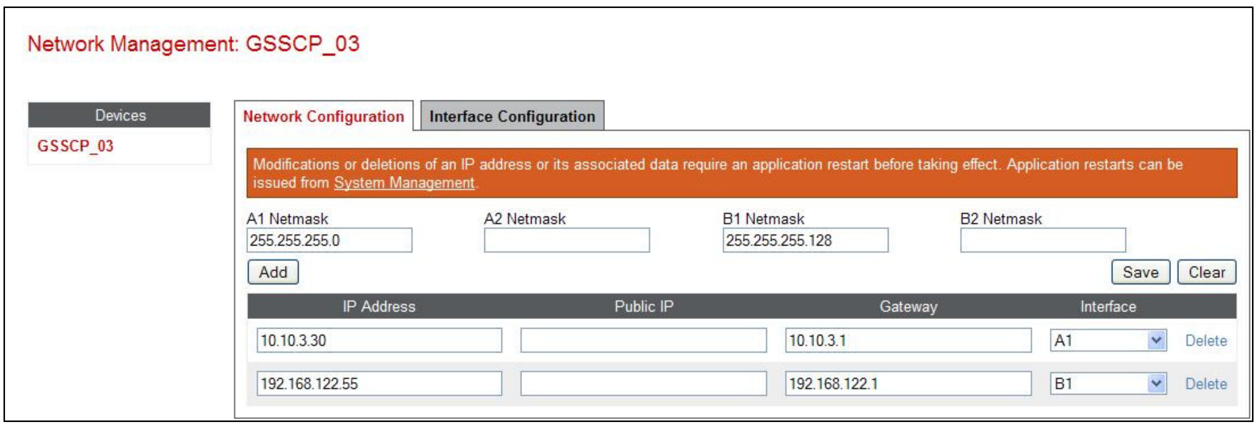
7.3. Device Specific Settings

The Device Specific Settings feature allows aggregation of system information to be viewed, and various device-specific parameters to be managed to determine how a particular device will function when deployed in the network.

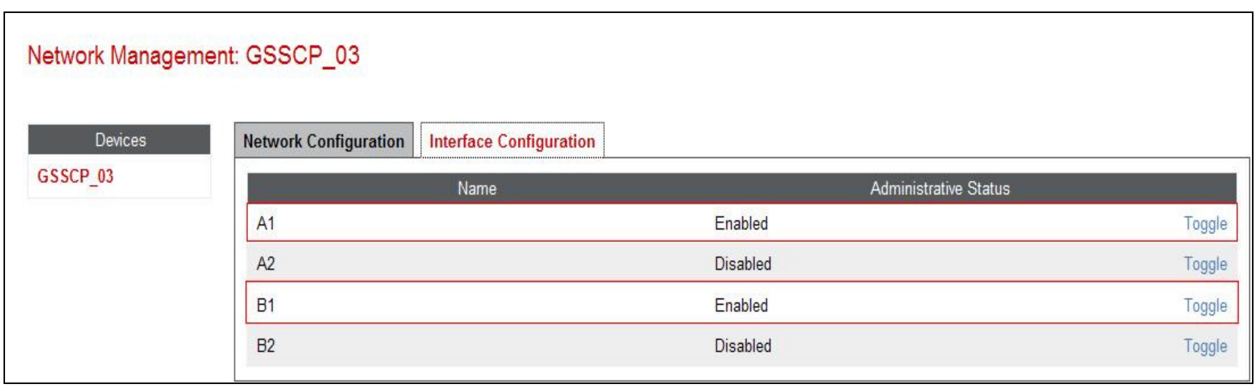
7.3.1. Network Management

The Network Management screen is where the network interface settings are configured and enabled. During the installation process of the Avaya SBCE, certain network-specific information is defined such as device IP address(es), public IP address(es), netmask, gateway, etc. to interface the device to the network. It is this information that populates the various Network Management tab displays, which can be edited as needed to optimize device performance and network efficiency.

Navigate to **Device Specific Settings → Network Management** and verify the IP addresses assigned to the interfaces and that the interfaces are enabled. The following screen shows the private interface is assigned to **A1** and the external interface is assigned to **B1**.



Select the **Interface Configuration** Tab and use the **Toggle** button to enable the interfaces.

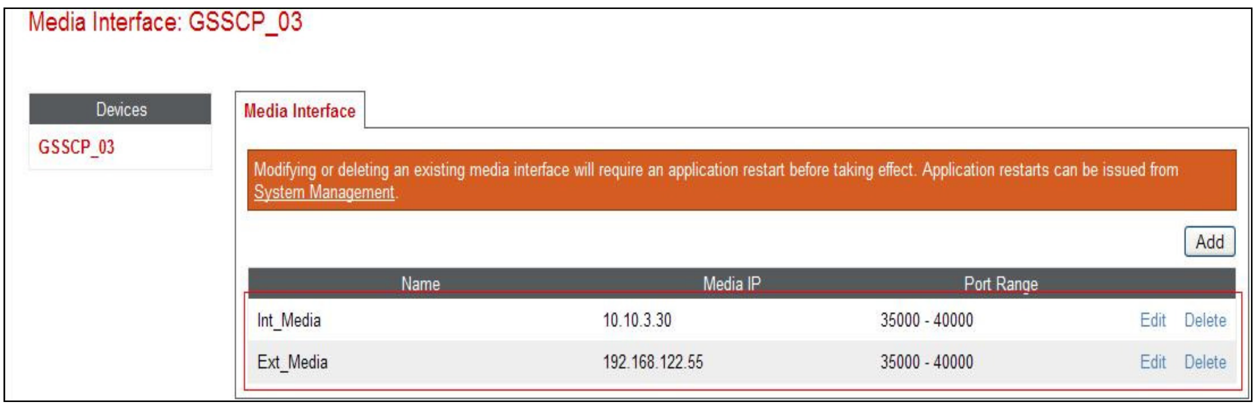


7.3.2. Media Interface

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**: **10.10.3.30** (Internal address for calls toward Communication Manager)
- **Port Range**: **35000-40000**
- Click **Finish**
- Select **Add**
- **Name**: **Ext_Media**
- **Media IP**: **192.168.122.55** (External address for calls toward Motto VoIP)
- **Port Range**: **35000-40000**
- Click **Finish**

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

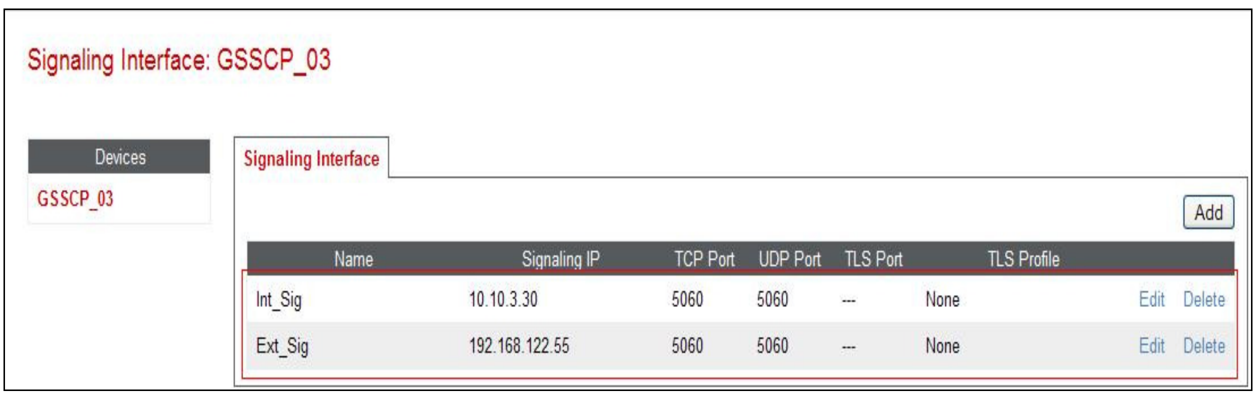


7.3.3. Signalling Interface

The Signalling Interface screen allows the IP Address and ports to be set for transporting signaling 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**.

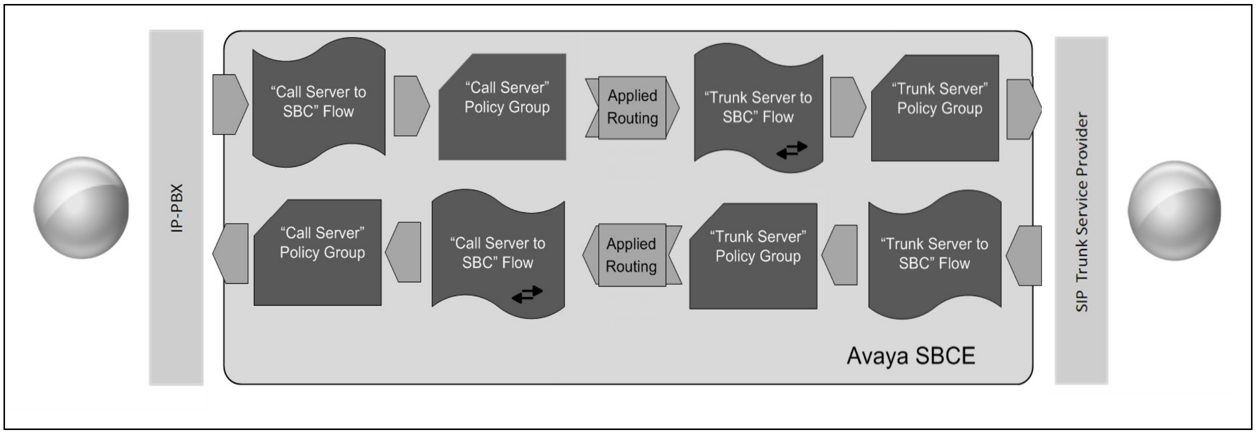
- **Name**: **Int_Sig**
- **Signaling IP**: **10.10.3.30** (Internal address for calls toward Communication Manager)
- **TCP Port**: **5060**
- **UDP Port**: **5060**
- Click **Finish**
- Select **Add**
- **Name**: **Ext_Sig**
- **Signaling IP: 192.168.122.55** (External address for calls toward Motto VoIP)
- **TCP Port**: **5060**
- **UDP Port**: **5060**
- Click **Finish**

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



7.3.4. End Point Flows

When a packet is received by Avaya SBCE, the content of the packet (IP addresses, URIs, etc.) is used to determine which flow it matches. Once the flow is determined, the flow points to a policy which contains several rules concerning processing, privileges, authentication, routing, etc. Once routing is applied and the destination endpoint is determined, the policies for this destination endpoint are applied. The context is maintained, so as to be applied to future packets in the same flow. The following screen illustrates the flow through the Avaya SBCE to secure a SIP Trunk call.

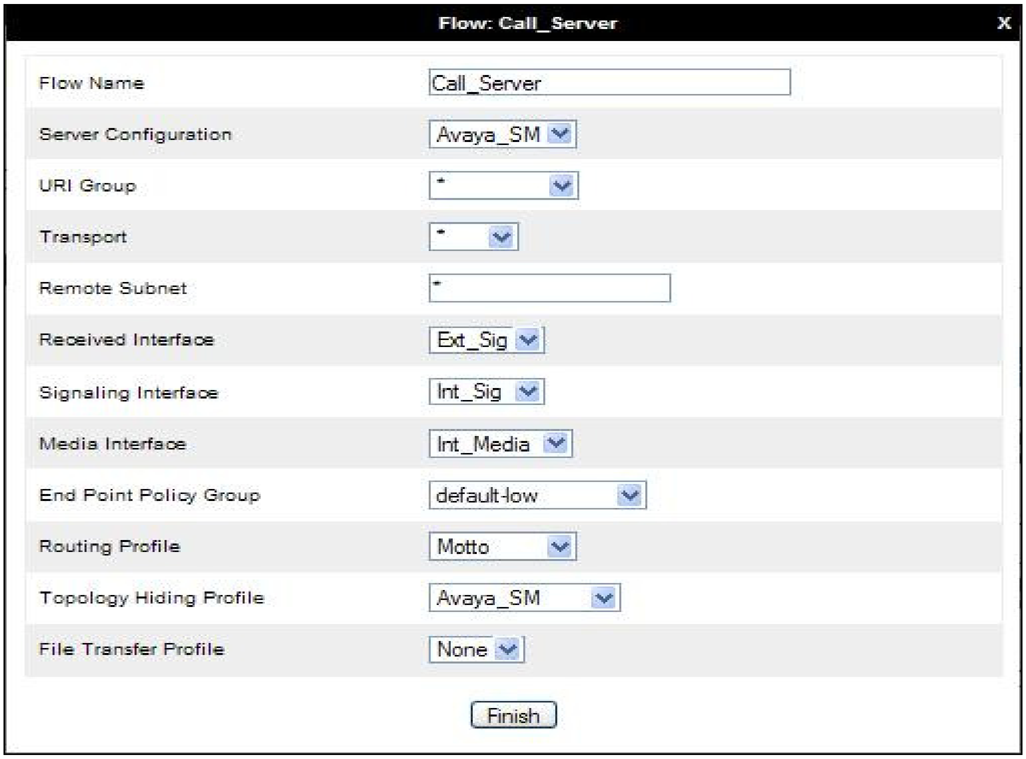


To create a Server Flow, navigate to **Device Specific Settings → End Point Flows**. Select the **Server Flows** tab and click **Add Flow**.

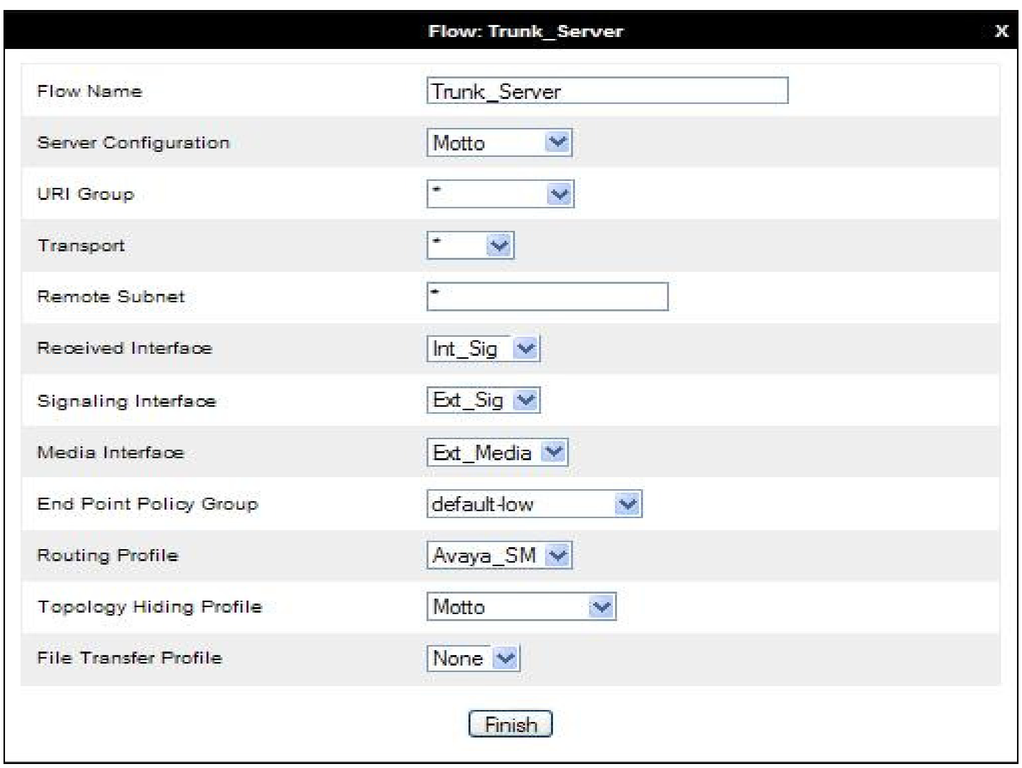
- **Flow Name:** Enter a descriptive name
- **Server Configuration:** Select a Server Configuration created in **Section 7.2.4** and **7.2.5** and assign to the Flow
- **Received Interface:** Select the Signaling Interface the Server Configuration is allowed to receive SIP messages from
- **Signaling 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 to
- **Topology Hiding Profile:** Select the profile to apply toward the Server Configuration

Click **Finish** to save and exit.

The following screen shows the Sever Flow for Session Manager.



The following screen shows the Sever Flow for Motto VoIP.



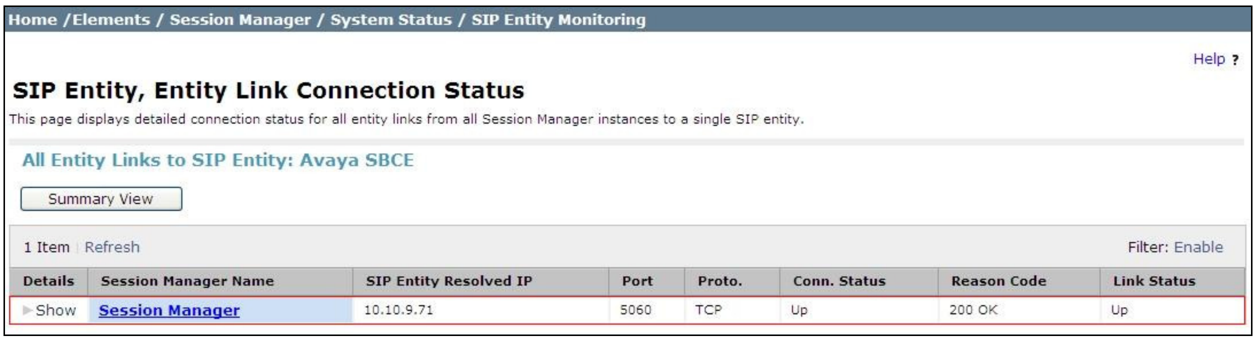
8. Motto VoIP SIP Trunk Configuration

The configuration of the Motto VoIP equipment used to support the Motto VoIP SIP Trunk Service is outside of the scope of these Application Notes and will not be covered. To obtain further information on Motto VoIP equipment and system configuration please contact an authorized Motto VoIP 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**. The screenshot shows the status of the Entity Link for the Avaya SBCE



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: Avaya SBCE

Summary View

1 Item | Refresh Filter: Enable

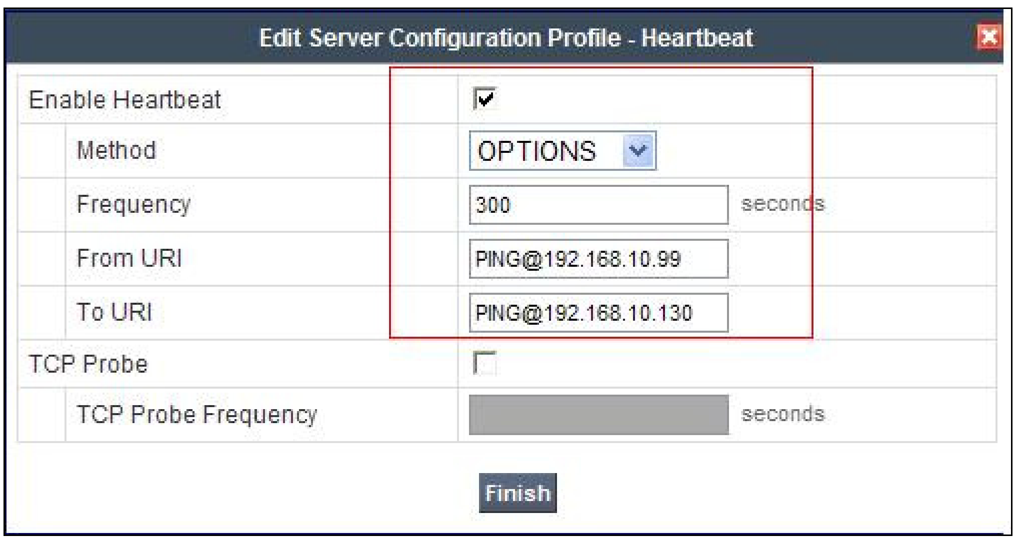
Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
▶Show	Session Manager	10.10.9.71	5060	TCP	Up	200 OK	Up

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
```

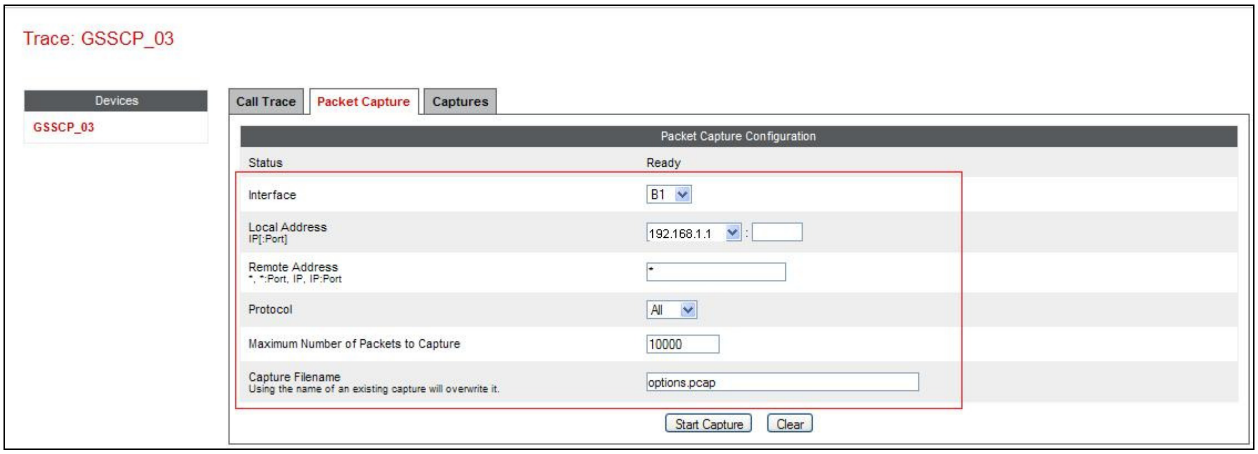
3. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active.

4. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active for more than 35 seconds. This time period is included to verify that proper routing of the SIP messaging has satisfied SIP protocol timers.
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, check from the Avaya SBCE using OPTIONS. This is done by defining the heartbeat in the Server configuration then running a trace. To define the heartbeat, navigate to **Global Profiles →Server Configuration** in the menu on the left hand side and click on the Trunk Server profile. Select the **Heartbeat** tab and click on **Edit**
   - Check the **Enable Heartbeat** box
   - Select **OPTIONS** from the **Method** drop down menu
   - Enter the **Frequency** in seconds, for convenience this can be set to the minimum value of **60** seconds
   - Enter the **From URI** in Fully Qualified Domain Name format
   - Enter the **To URI** in FQDN format
   - Click on **Finish**



To define the trace, navigate to **Device Specific Settings →Troubleshooting → Trace** in the 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 Service Provider's 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 200 OK response will be seen from the Service Provider.

10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Communication Manager, Avaya Aura® Session Manager and Avaya Session Border Controller for Enterprise to Motto VoIP SIP Trunk Service. The service was successfully tested with observations listed in **Section 2.2**.

11. 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 6.2,* March 2012.

[2] *Administering Avaya Aura® System Platform Release 6.2,* February 2012.

[3] *Administering Avaya Aura® Communication Manager*, Release 6.2, February 2012.

[4] *Avaya Aura® Communication Manager Feature Description and Implementation*, February 2012, Document Number 555-245-205.

[5] *Implementing Avaya Aura® System Manager Release 6.2*, March 2012.

[6] *Implementing Avaya Aura® Session Manager*, February 2012, Document Number 03-603473.

[7] *Administering Avaya Aura® Session Manager*, February 2012, Document Number 03-603324.

[8} *Avaya One-X® Communicator Getting Started,* November 2009, Document Number 03-600758.

[9] *Installing Avaya Session Border Controller for Enterprise*, Release 6.2

[10 *Administering Avaya Session Border Controller for Enterprise*, Release 6.2]

[11] *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.