**Avaya Solution & Interoperability Test Lab**

# Application Notes for Configuring Avaya Aura® Communication Manager R6.2, Avaya Aura® Session Manager R6.1 and Avaya Session Border Controller for Enterprise to Support Colt SIP Trunk Service – Issue 1.0

## Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between Colt SIP Trunk Service and an Avaya SIP enabled enterprise solution. The Avaya solution consists of Avaya Aura® Session Manager, Avaya Aura® Communication Manager and Avaya Session Border Controller for Enterprise. Colt is a member of the DevConnect Global SIP 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.

**NOTE:** This Application Note is applicable with Avaya Aura® 6.2 which is currently in Controlled Introduction. Avaya Aura® 6.2 will be Generally Available in Summer 2012.

# 1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between Colt SIP Trunk Service and an Avaya SIP enabled enterprise solution. The Avaya solution consists of Avaya Aura® Session Manager, Avaya Aura® Communication Manager Evolution Server and Avaya Session Border Controller for Enterprise. Customers using this Avaya SIP-enabled enterprise solution with the Colt 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 costs 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 Session Manager and Communication Manager. The enterprise site was configured to use the SIP Trunk Service provided by Colt.

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 the PSTN were routed to the DID numbers assigned by Colt. Incoming PSTN calls were made to H.323, SIP, Digital and Analogue telephones at the enterprise.
- Outgoing calls from the enterprise site were completed via Colt to PSTN. Outgoing calls from the enterprise to the PSTN were made from H.323, SIP, Digital and Analogue telephones.
- Calls made using G.729A and G.711A codec's.
- 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 Colt SIP Trunk Service with the following observations:

- 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.
- Inbound and Outbound fax was tested using T.38 standard.
- A Signalling manipulation had to be added to remove payload type 2 from the SDP (Session Description Protocol) as this is reserved and is rejected by Communication Manager.

## 2.3. Support

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

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

www.colt.net or Colt Local Support numbers.

| | | | |
|---|---|---|---|
| **Austria** | 0800 880 990 | **Belgium** | 0800 507 01 |
| **Germany** | 0800 111 1230 | **France** | 0800 948 888 |
| **Italy** | 192090 | **Netherlands** | 0800 265 8023 |
| **Portugal** | 808 780 222 | **Spain** | 901 888400 |
| **Switzerland** | 0800 560 560 | **UK** | 0800 136 166 |

# 3. Reference Configuration

**Figure 1** illustrates the test configuration. The test configuration shows an enterprise site connected to the Colt SIP Trunk Service. Located at the enterprise site is a Session Manager and Communication Manager. Endpoints are Avaya 9600 and 4600 series IP telephones, Avaya 2400 series Digital Telephone, an Avaya Desktop Video Device, a PC running one-X Communicator, an Analogue Telephone and Fax Machine. For security purposes, any public IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes.

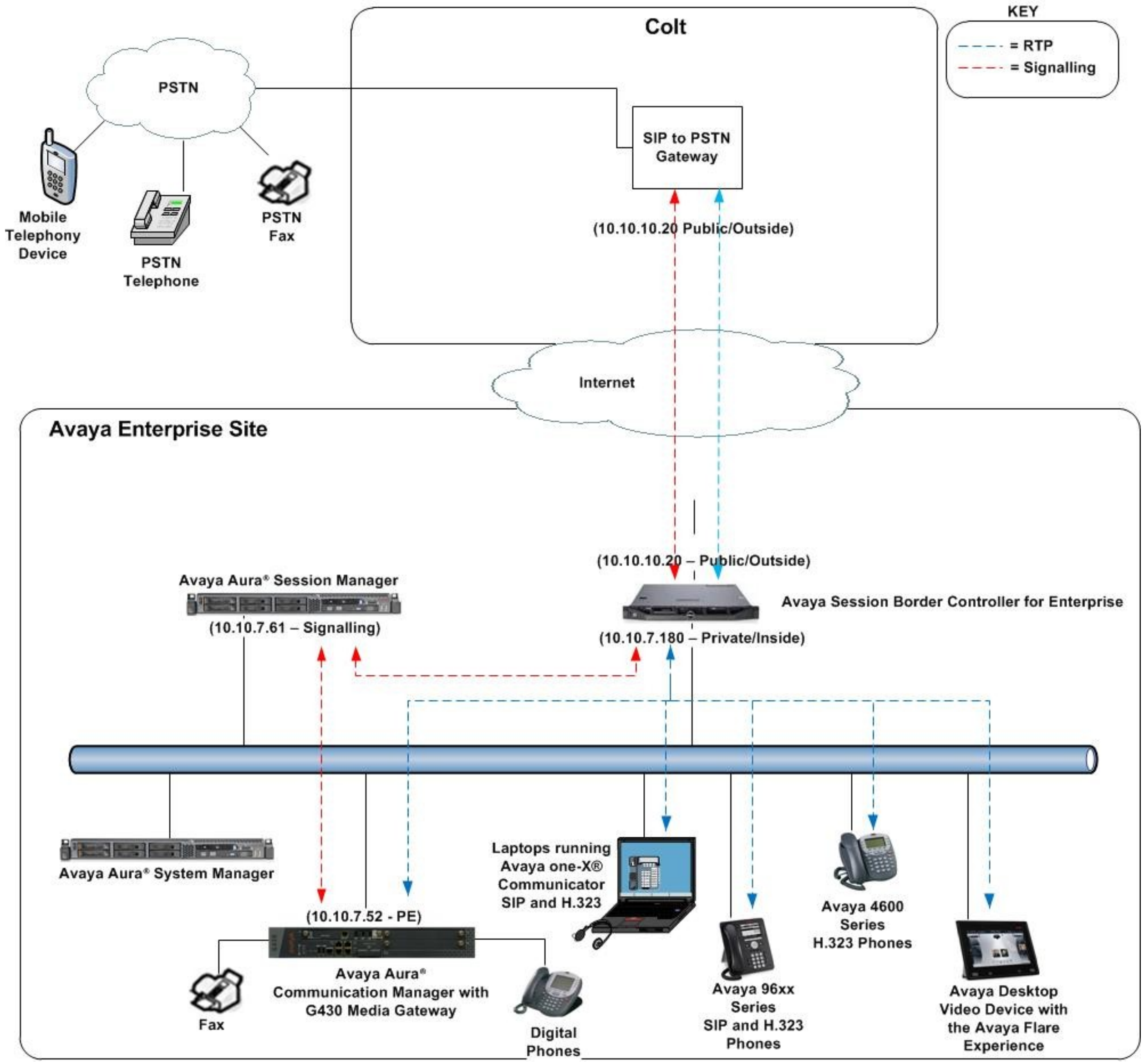



**Figure 1: Colt SIP Solution Topology**

# 4. Equipment and Software Validated

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

| Equipment/Software | Release/Version |
|---|---|
| Avaya Aura® Communication Manager running on Avaya S8800 Server | R6.2<br>(R016x.02.0.823.0) |
| Avaya G430 Media Gateway<br>MM711 Analogue<br>MM712 Digital<br>MGP Firmware | <br>HW31 FW093<br>HW07 FW009<br>30.12.1 |
| Avaya Aura® Session Manager running on Avaya S8800 Server | R6.1 (6.2.0.0.620110) |
| Avaya Aura® System Manager running on Avaya S8800 Server | Avaya Aura® System Manager R6.1<br>(6.2.0.0.15669-6.2.12.9)<br>Update revision No: 6.2.12.1.1822 |
| Avaya Session Border Controller for Enterprise running on Dell R210 | (4.0.5.Q02) |
| Avaya 9620 Phone (H.323) | 3.11 |
| Avaya 9620 Phone (SIP) | 2.6.4.0 |
| Avaya 2420 Digital Phone | N/A |
| Analog Phone | N/A |
| Avaya 4620 Phone (H.323) | 2.9 |
| Avaya one-X® Communicator | 6.1 |
| Avaya Desktop Video Device | 1.0.2 |
| Colt SIP Trunk Service<br>Sonus GSX9000<br>Sonus PSX<br>Configuration | <br>7.3.3<br>7.3.3<br>Colt6212942012 |

Note: Colt configuration kept internally for support reference.

# 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 Signaling associated with Colt SIP Trunk Service. For incoming calls, Session Manager receives SIP messages from Colt 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 signaling is routed to Session Manager. Session Manager directs the outbound SIP messages to the Colt 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 Colt 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: 24000 30
```

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, **asmV7**and **10.10.7.61** 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             10.10.7.52
 asmV7             10.10.7.61
 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 Colt were configured, namely **G.711A** and **G.729**.

```
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
```

Colt SIP Trunk Service uses pass-through which is not a method supported by Avaya. Configure the pass-through fax protocol by setting the **Fax Mode** to **t.38-standard** on **Page 2** of the codec set form as shown below. Although during testing pass-through mode was shown to work, Avaya does not officially support this fax method**.**

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

                          IP Codec Set

                              Allow Direct-IP Multimedia? n

                    Mode                  Redundancy
    FAX             t.38-standard          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 Colt 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** (where n is the next available signaling group number) command as follows:

- 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 **asmV7**), 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 6.2.** 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 **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: asmV7
 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 **tie**
- 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: smpub                  COR: 1        TN: 1        TAC: 101
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                   Signaling Group: 1
                                                 Number of Members: 30
```

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 Colt 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: 8000

            SCCAN? n                               Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 1800
```

On **Page 3,** set the **Numbering Format** field to **private.** This prevents the number to be sent to Colt 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,** set the **Mark Users as Phone** to **y**, this field inserts a parameter to SIP requests indicating to any receiving SIP entity that the user part of the request URI should be treated as a telephone number. Set **Send Transferring Party Information** to **y,** to allow trunk to trunk transfers. Set **Telephone Event Payload Type** to **120**.

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

                      Mark Users as Phone? y
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? y
                 Network Call Redirection? n
                    Send Diversion Header? n
                  Support Request History? y
             Telephone Event Payload Type: 120
```

## 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 **1** will send the calling party number **0044xxxxxxxxxx** to Colt 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-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  1             1          0044xxxxxxxxxx  14    Maximum Entries: 240
```

## 5.8. Administer Route Selection for Outbound Calls

In these Application Notes, the Automatic Route Selection (ARS) feature will be used to route outbound calls via the SIP trunk to Colt 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 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 02                                               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 by setting **Grp No** to **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                                          none
 2: 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 Colt 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 Colt correlate to the internal extensions assigned within Communication Manager. The entries displayed below translates incoming DID numbers **44xxxxxxxxxx** 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    12  44xxxxxxxxxx    all  1306
```

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

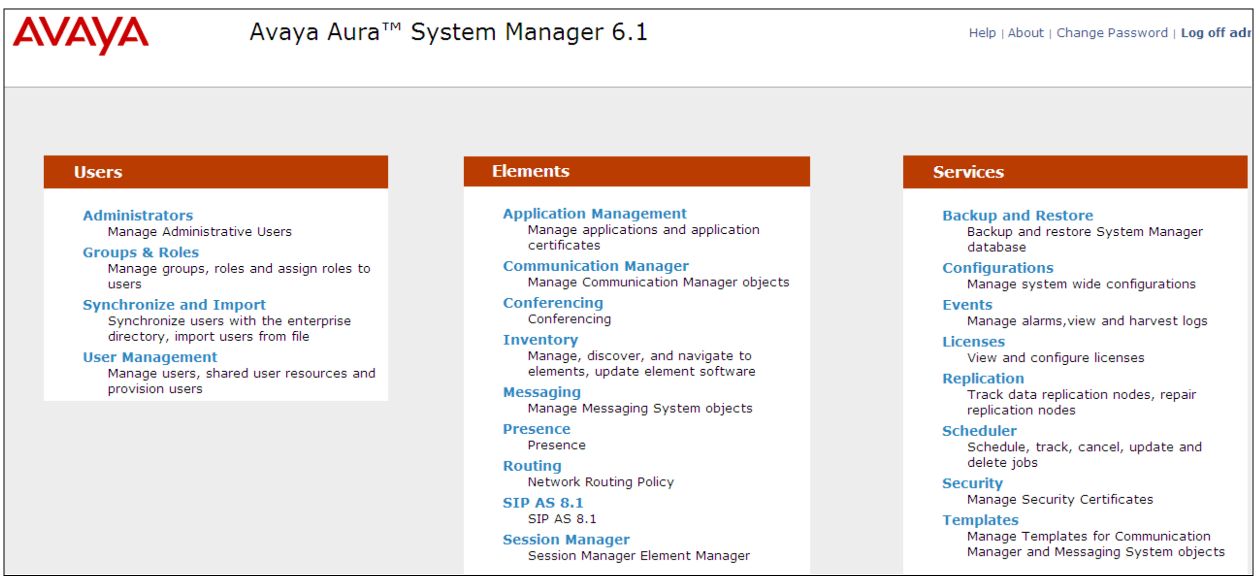
# 6. Configuring Avaya Aura® Session Manager

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

- Log in to Avaya Aura® System Manager
- Administer SIP domain
- Administrer SIP Location
- Administer SIP Entities
- Administer Entity Links
- Administer Routing Policies
- Administer Dial Patterns
- Administer Avaya Aura® Communication Manager as Managed Element
- Administer Application for Avaya Aura® Communication Manager
- Administer Application Sequence for Avaya Aura® Communication Manager
- Administer SIP Extensions

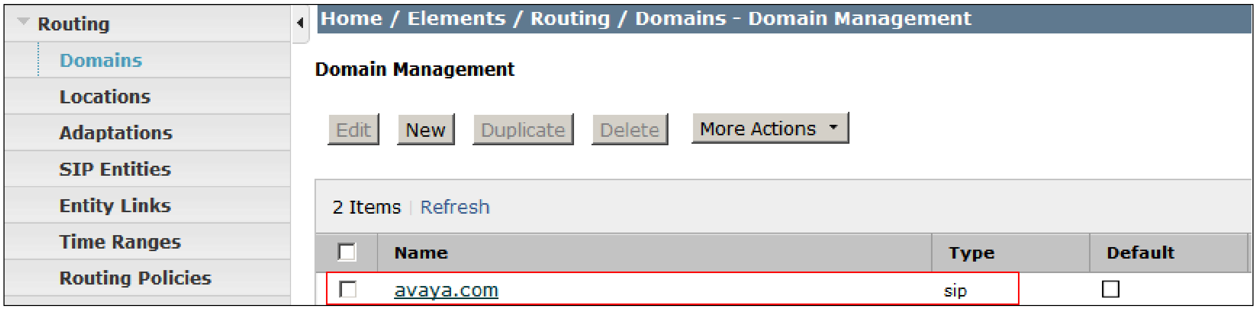
## 6.1. Log in to Avaya Aura® System Manager

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



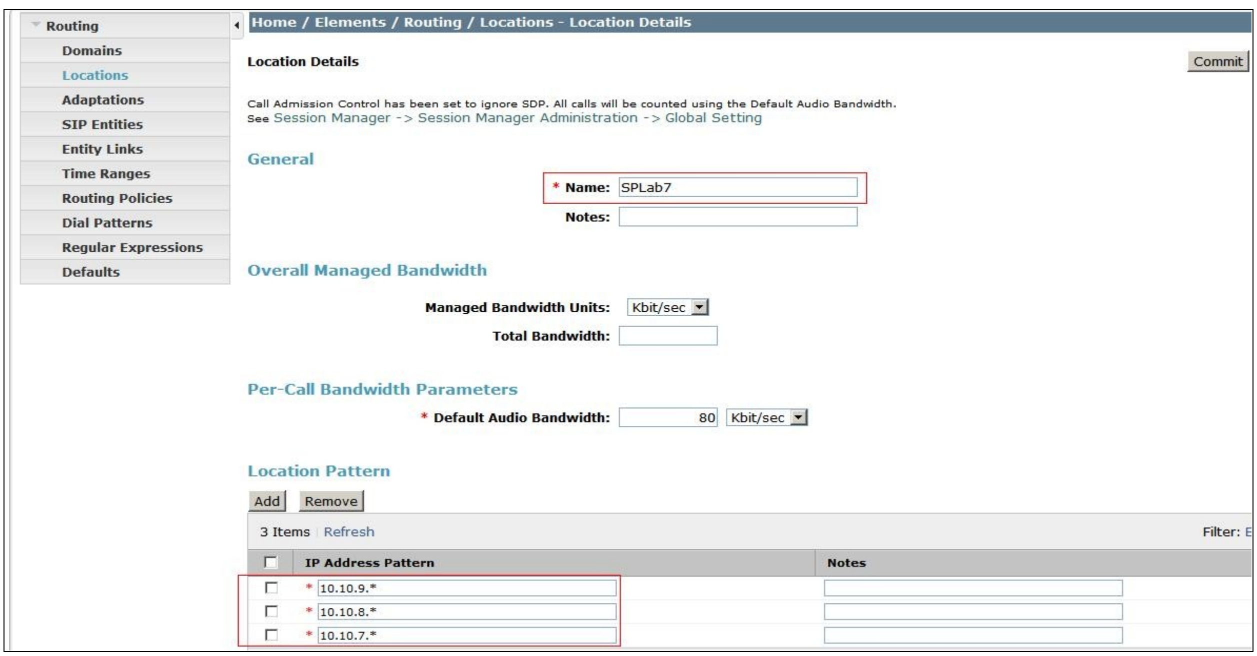
## 6.2. Administer SIP domain

To add the SIP domain that will be used with Session Manager, select **Routing** from the **Home** tab menu (not shown) and in the resulting tab select **Domains** from left hand menu. Click the **New** button to create a new SIP domain entry. In the **Name** field enter the domain name (e.g., **avaya.com**). Click **Commit** to save changes (not shown).




## 6.3. Administer Locations

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



## 6.4. 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:**

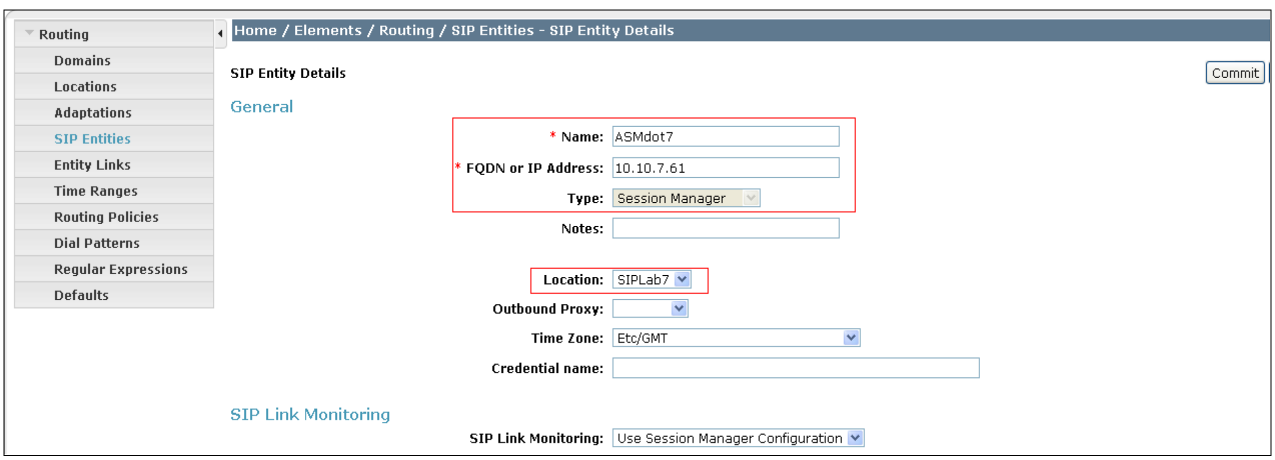
- In the **Name** field enter an informative name
- In the **FQDN or IP Address** field enter the IP address of Session Manager or the signaling interface on the connecting system
- In the **Type** field use **Session Manager** for a Session Manager SIP entity, **CM** for a Communication Manager SIP 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
- Session Border Controller SIP Entity

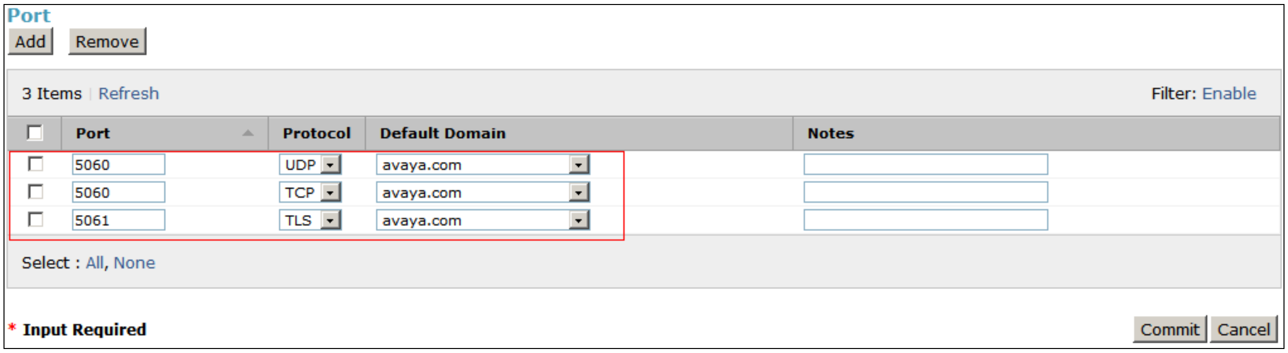
### 6.4.1. Avaya Aura® Session Manager SIP Entity

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.



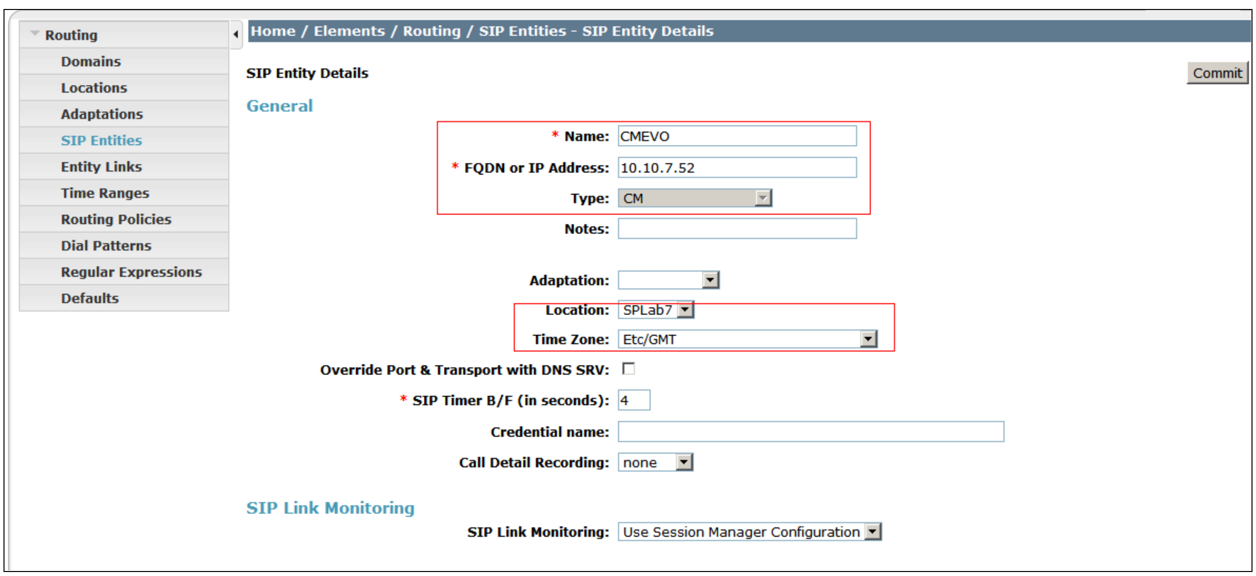
Session Manager must be configured with the port numbers and the protocols that will be used by the other SIP entities. To configure these, scroll to the bottom of the page and under **Port**, click **Add,** then edit the fields in the resulting new row.

- 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



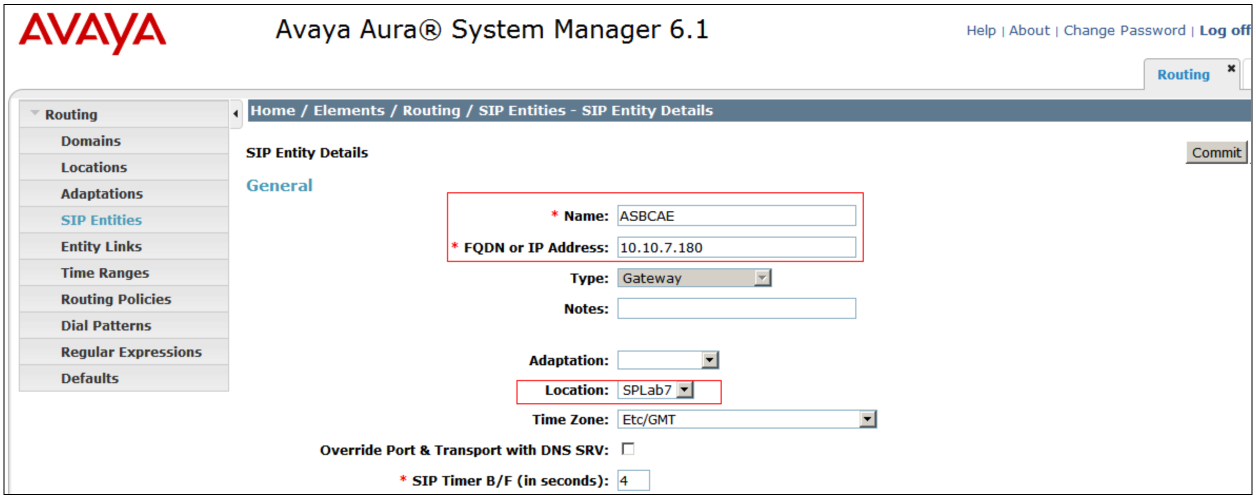

## 6.4.2. Avaya Aura® Communication Manager SIP Entity

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**.



### 6.4.3. Avaya Session Border Controller Advanced for Enterprise SIP Entities

The following screen shows the SIP entity for the Avaya Session Border Controller Advanced for Enterprise used for routing Fixed and Mobile calls. The **FQDN or IP Address** field is set to the IP address of the private interfaces administered in **Section 7** of this document.

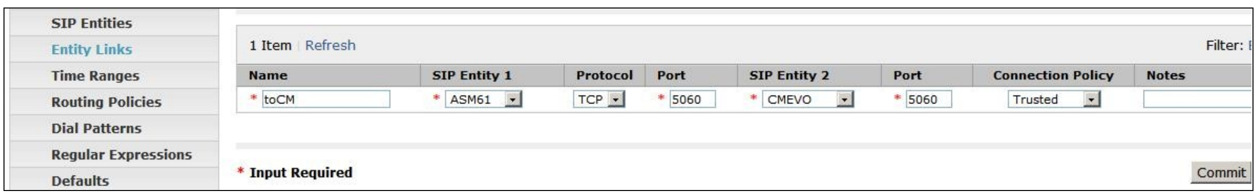


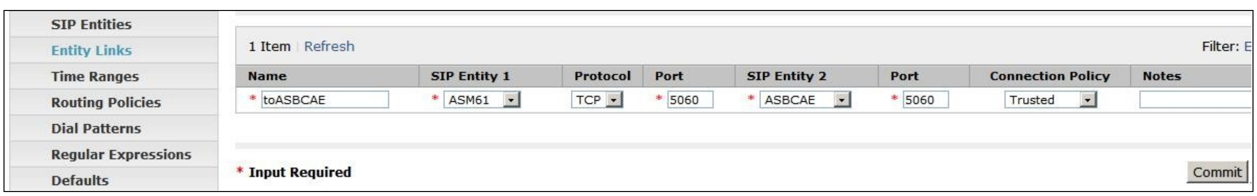
## 6.5. Administer Entity Links

A SIP trunk between a Session Manager and another system is described by an Entity Link. To add an Entity Link, select **Entity Links** on the left panel menu and click on the **New** button . Fill in the following fields in the new row that is displayed.

- In the **Name** field enter an informative name
- In the **SIP Entity 1** field select the Session Manager Sip Entity, in the example below it is **ASM61**
- 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 the **Trusted** tick box to make the other system trusted
- In the **Protocol** field enter the transport protocol to be used to send SIP requests

Click **Commit** (not shown) to save changes. The following screen shows the Entity Links used in this configuration.



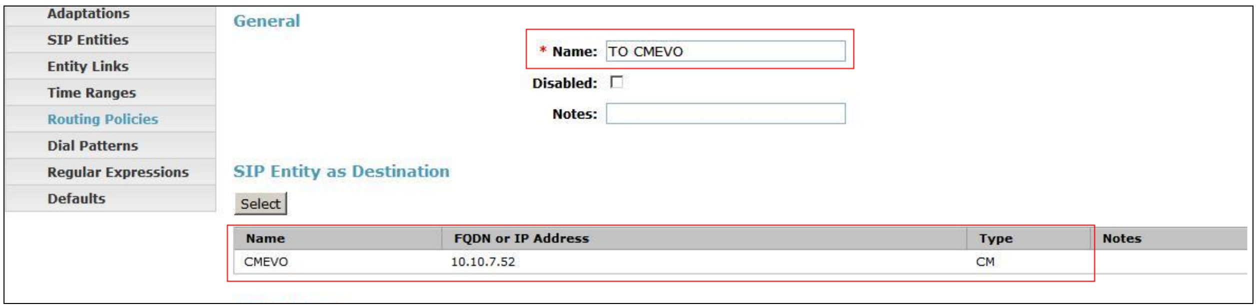
## 6.6. Administer Routing Policies

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).

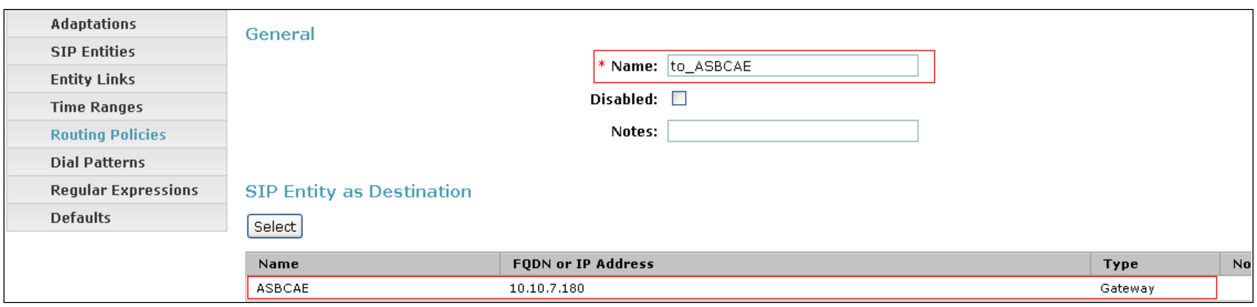
Under **General**:

- 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 Session Border Controller Advanced for Enterprise:



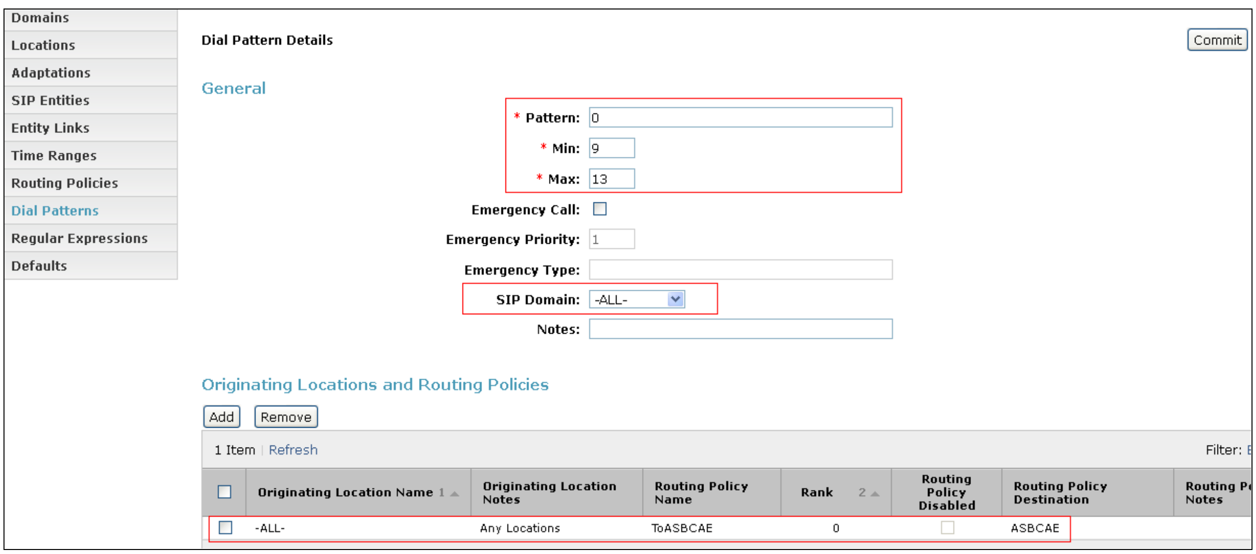
## 6.7. Administer Dial Patterns

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).

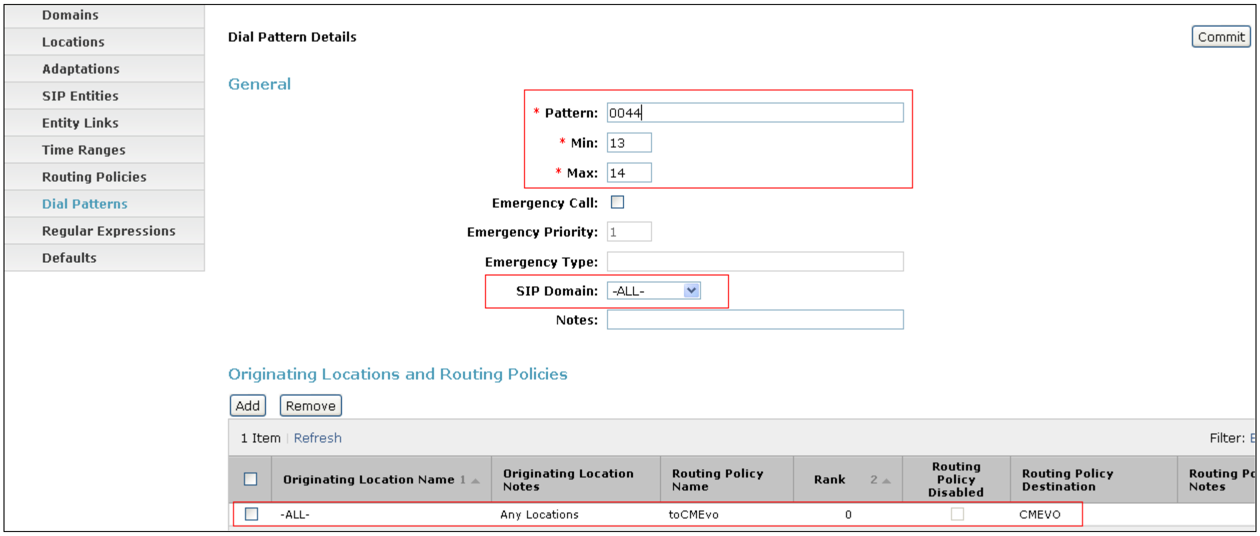
Under **General**:

- 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 6.7.** Click **Select** button to save (not shown). The following screens show an example dial pattern configured for Colt SIP Trunk Service.



The following screen shows an example dial pattern configured for Communication Manager.

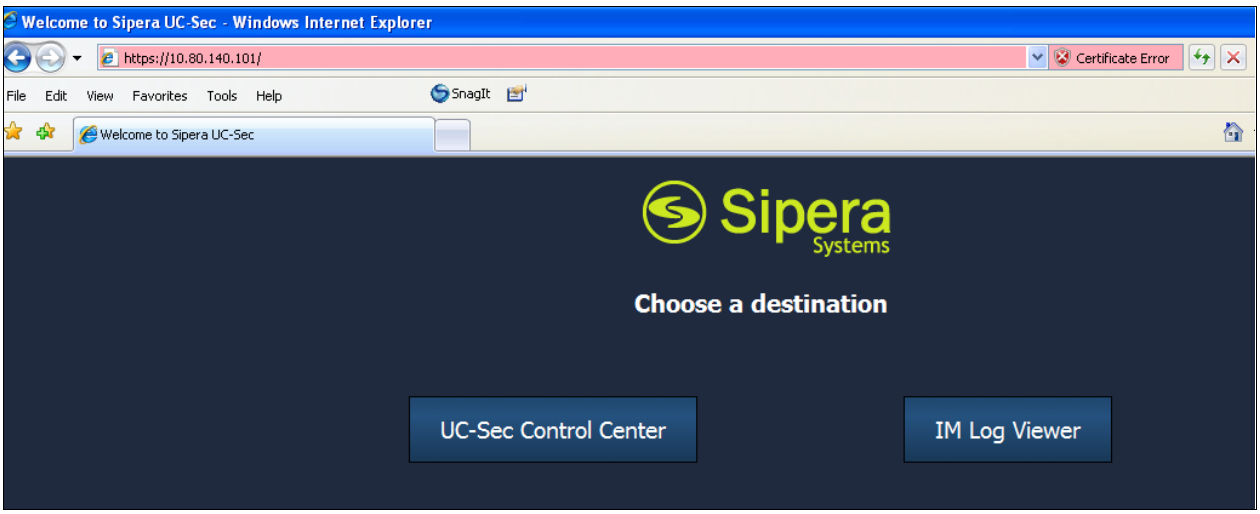


# 7. Avaya Session Border Controller Advanced for Enterprise Configuration

This section provides the procedures for configuring Session Border Controller for Enterprise (E-SBC).

## 7.1. Accessing UC-Sec Control Centre

Access the web interface by typing **https://x.x.x.x** (where x.x.x.x is the management IP of the E-SBC).




Select **UC-Sec Control Center** and enter the **Login ID** and **Password**.

## 7.2. Global Profiles

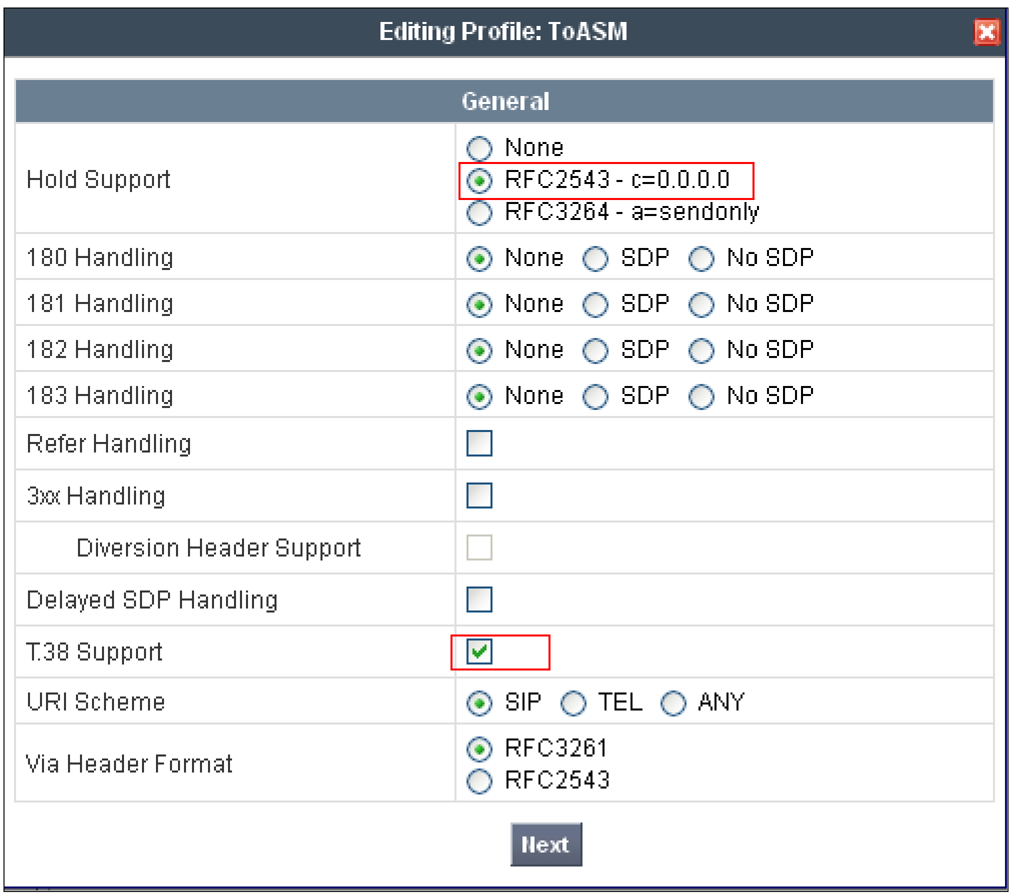
When selected, Global Profiles allows for configuration of parameters across all UC-Sec appliances.

### 7.2.1. Server Internetworking Avaya Side

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 Profile.** Enter **Profile Name: ToASM** and click **Next** (not Shown).

- Check **Hold Support** to **RFC2543 – c=0.0.0.0**
- Check **T.38 Support**

All other options on the **General** tab can be left at default. Click on **Next** on the following screens and then **Finish**.

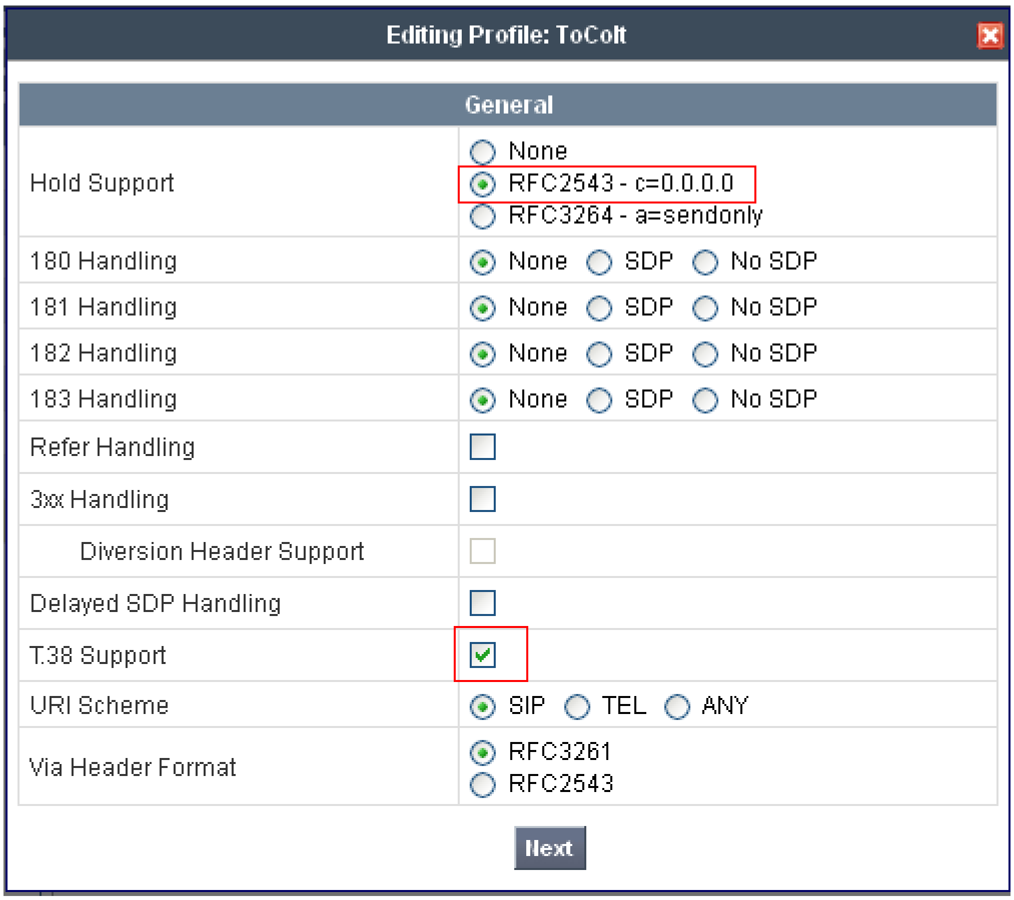


### 7.2.2. Server Internetworking – Colt side

Server Internetworking allows the configuration and management of 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 Profile.** Enter profile name**: ToColt** and click on **Next**.

- Check **Hold Support** to **RFC2543 – c=0.0.0.0**
- Check **T.38 Support**

All other options on the **General** tab can be left at default. Click on **Next** on the following screens and then **Finish**.

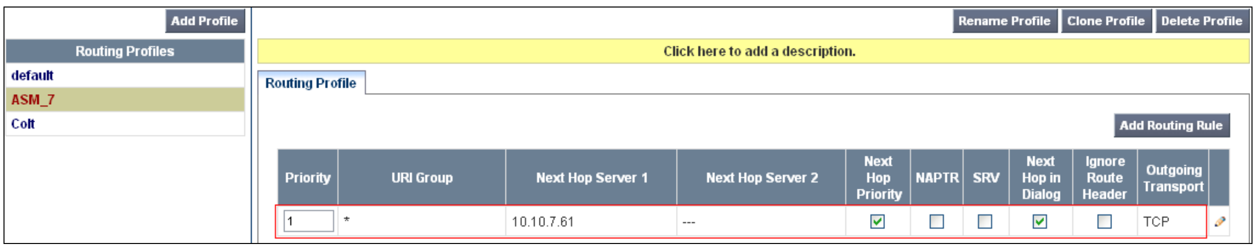


### 7.2.3. Routing – Avaya side

The Routing Profile allows the management of parameters related to routing SIP signaling messages. From the lefthand menu select **Global Profiles → Routing** and click on **Add Profile**.

- Enter **Profile Name**: **ASM_7**
- Hit **Next** (not shown)
- Set **Next Hop Server 1** to **10.10.7.61** (Session Manager IP address)
- Select **Routing Priority Based on Next Hop Server**
- Select use **Next Hop in Dialog** Messages
- Set **Outgoing Transport** to **TCP**
- Click **Finish** (not shown)

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

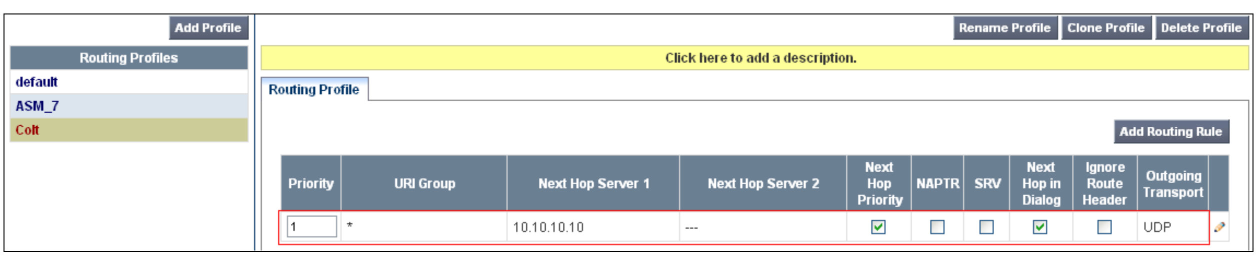



### 7.2.4. Routing – Colt side

The Routing Profile allows the management of parameters related to routing SIP signaling messages. A routing profile must be set for Fixed and Mobile calls. From the lefthand menu select **Global Profiles → Routing** and click on **Add Profile**.

- Enter **Profile Name** as **Colt**
- Hit **Next**
- Set **Next Hop Server 1** to **10.10.10.10** (IP Address provided by Colt)
- Select **Routing Priority Based on Next Hop Server**
- Select use **Next Hop in Dialog** Messages
- Set **Outgoing Transport** to **UDP**
- Click **Finish** (not shown)

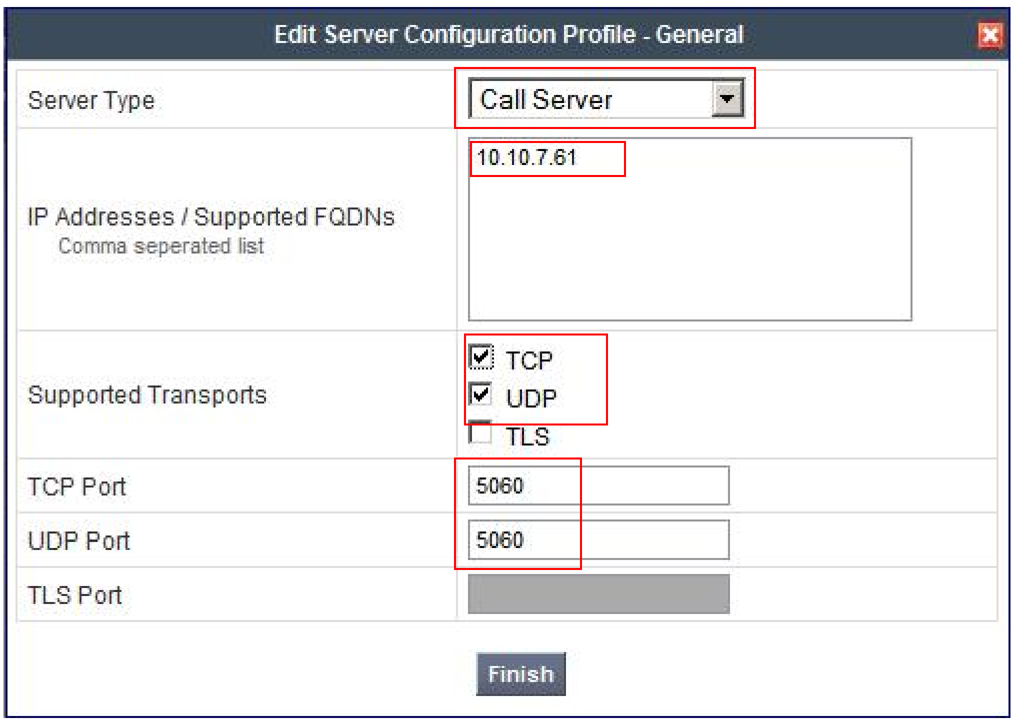
The screen below is a result of the details configured above.



### 7.2.5. Server Configuration– Avaya Aura® Session Manager

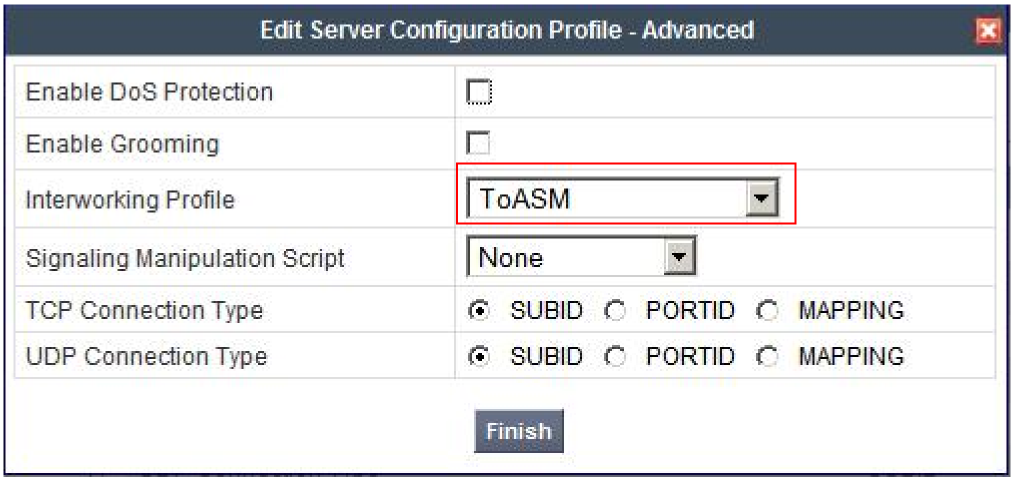
The **Server Configuration** screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. Together, these tabs allow the configuration and management of various SIP call server-specific parameters such as TCP and UDP port assignments, IP Server type, heartbeat signaling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

- Select **Server Type** to be **Call Server**
- Enter **IP Addresses / Supported FQDNs** to **10.10.7.61** (Session Manager IP Address)
- For **Supported Transports,** check **UDP** and **TCP**
- **TCP Port:5060**
- **UDP Port: 5060**
- Click on **Next** (not shown) to use deault entries on the **Authentication** and **Heartbeat** tabs



On the **Advanced** tab

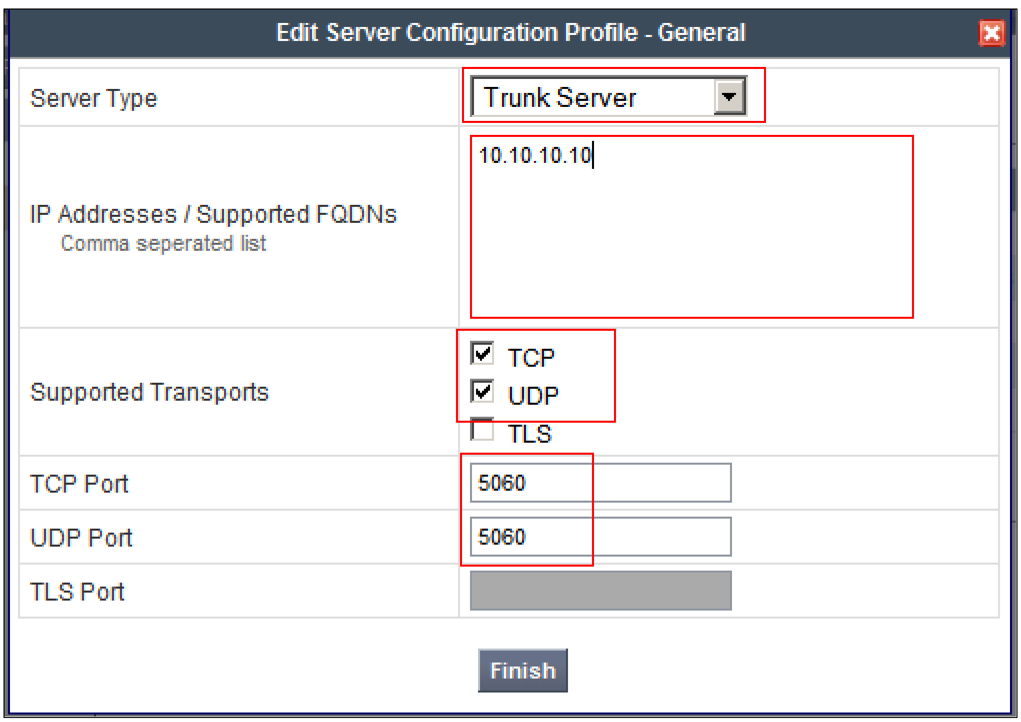
- Select **ToASM** for **Interworking Profile**
- Click **Finish**



### 7.2.6. Server Configuration– Colt side

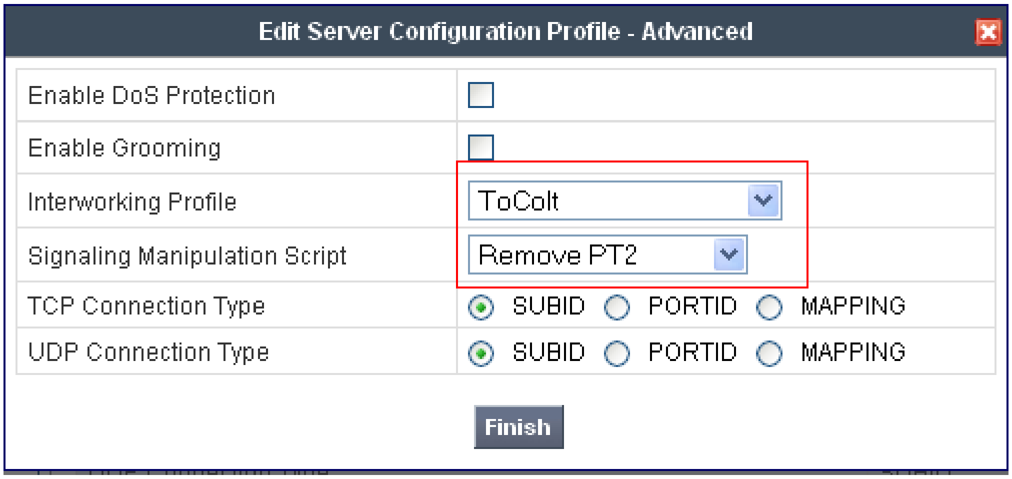
The **Server Configuration** screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. Together, these tabs allow the configuration and management of various SIP call server-specific parameters such as TCP and UDP port assignments, server type, heartbeat signaling parameters and some advanced options. From the left-hand menu select **Global Profiles → Server Configuration** and click on **Add Profile.** Enter N**ame as Colt_TS.** On the **Add Server Configuration Profile** tab, click on **Edit** and set the following:

- Select **Server Type** as **Trunk Server**
- Set **IP Address** to **10.10.10.10** (Colt Trunk Server )
- **Supported Transports**: Check **UDP and TCP**
- **UDP and TCP Port: 5060**
- Hit **Next**
- Click on **Next** (not shown) to use default entries on the **Authentication** and **Heartbeat** tabs



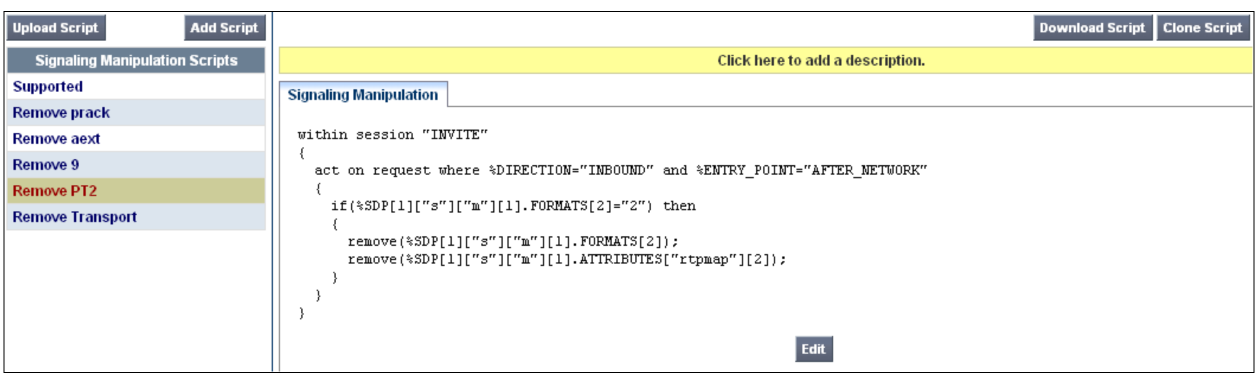
On the **Advanced** tab

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




### 7.2.7. Signaling Manipulation

Calls coming in from the colt network use the payload type 2 for the G.726 codec and this is a reserved type that is rejected by Communication Manager. The following sigma Script must be written and set in the Server Configuration Profile in **Section 7.2.6**.

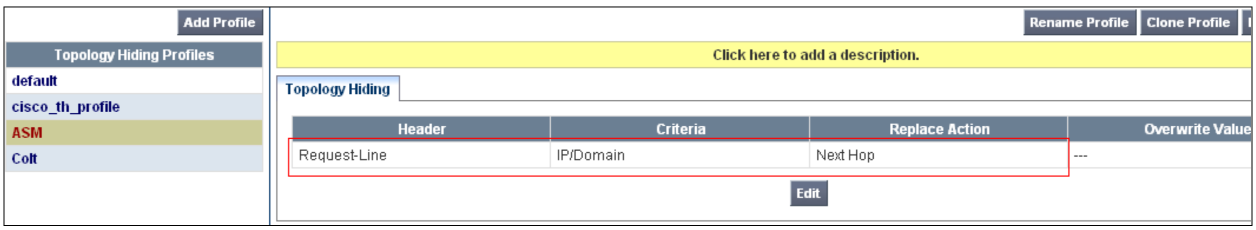


### 7.2.8. Topology Hiding – Avaya side

The **Topology Hiding** screen allows the management of 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. From the left-hand menu select **Global Profiles → Topology Hiding**.

- Click **default** profile and select **Clone Profile**
- Enter Profile Name**: ASM**
- For the **Request-Line** select **IP/Domain** under **Criteria** and **Next Hop** under **Replace Action**
- Remove all other entries
- Click **Finish**

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

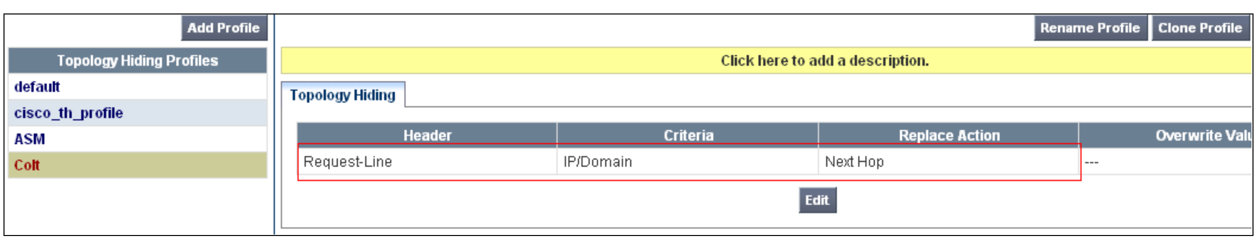



### 7.2.9. Topology Hiding – Colt side

The **Topology Hiding** screen allows the management of 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. From the left-hand menu select **Global Profiles → Topology Hiding**.

- Click **default** profile and select **Clone Profile**
- **Enter Profile Name: Colt**
- For the Header **Request-Line** select **IP/Domain** under **Criteria** and **Next Hop** under **Replace Action**
- Remove all other entries
- Click **Finish**

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

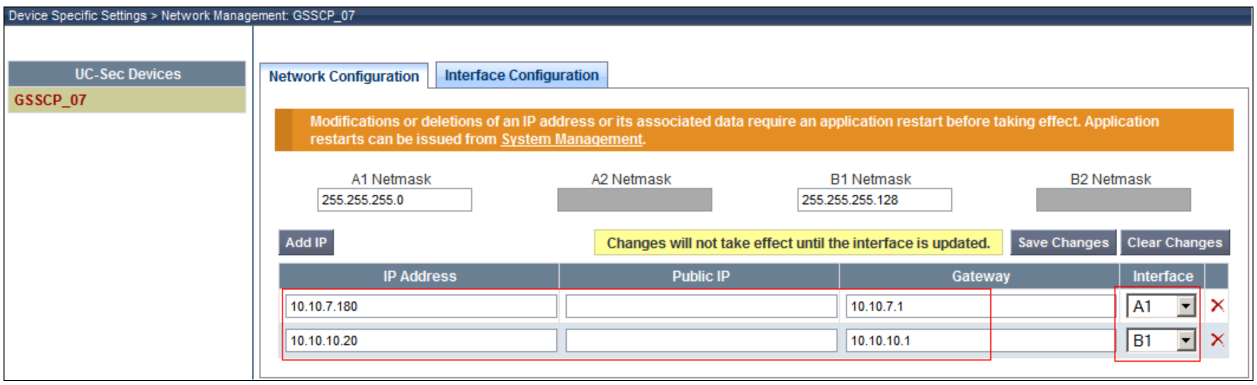


## 7.3. Device Specific Settings

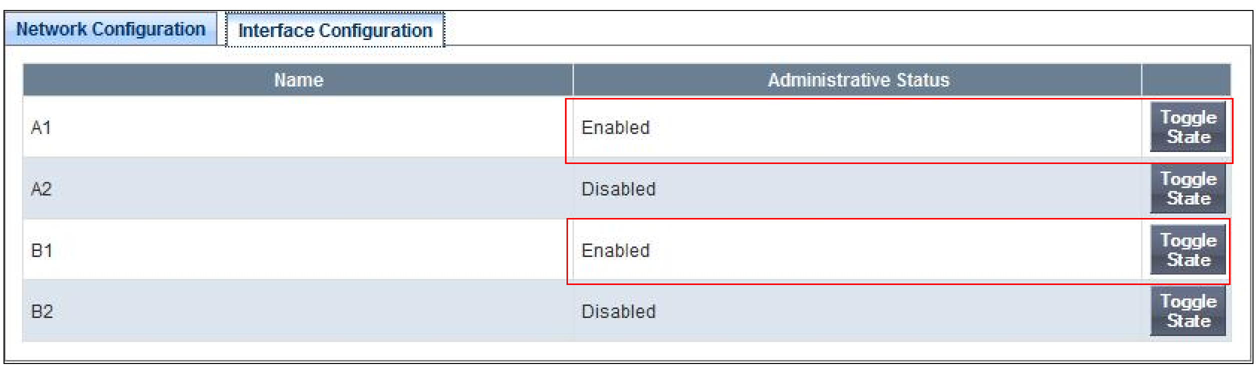
This section is used to configure the Interfaces used or the transportation of SIP messaging between the enterprise and the service provider

### 7.3.1. Network Management

The **Network Management** feature allows the public and private interface addresses and state to be set. From the left-hand menu select **Device Specific Settings → Network Management**. Enter in the **IP Address** and **Gateway Address** for both the Inside and the Outside interfaces Select the physical interface used in the **Interface** column.




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

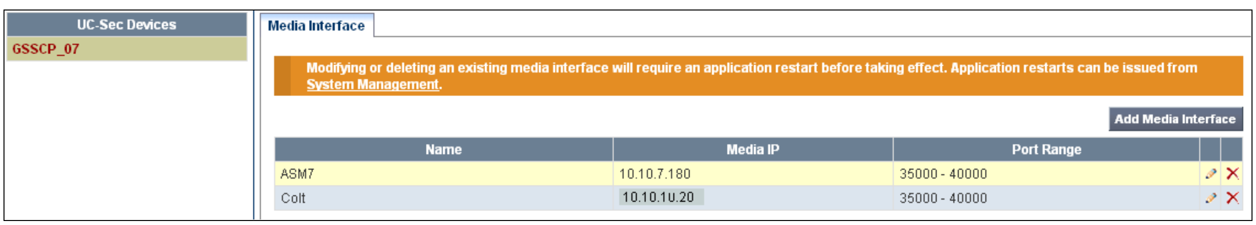


## 7.3.2. Media Interfaces

The Media Interfaces feature allows the IP Address and ports to be set for transporting Media over the SIP trunk. From the left-hand menu select **Device Specific Settings → Media Interface**.

- Select **Add Media Interface**
- **Name**: **ASM7**
- **Media IP**: **10.10.7.180** (Internal Address for calls toward Session Manager)
- **Port Range**: **35000-40000**
- Click **Finish**
- Select **Add Media Interface**
- **Name**: **Colt**
- **Media IP**: **10.10.10.20** (External Address for calls toward Colt trunk)
- **Port Range**: **35000-40000**
- Click **Finish**
- Select **Add Media Interface**

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

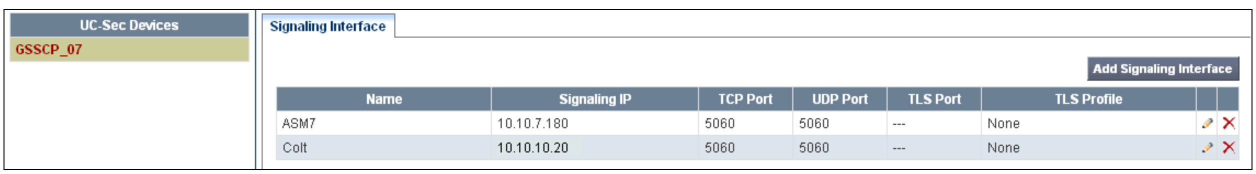


### 7.3.3. Signalling Interfaces

The Signalling Interfaces feature allows the IP Address and ports to be set for transporting Media over the SIP trunk. From the left-hand menu select Device Specific Settings → Signalling Interface.

- Select **Add Signaling Interface**
- **Name**: **ASM7**
- **Media IP**: **10.10.7.180** (Internal Address for calls toward Session Manager)
- **TCP Port**: **5060**
- **UDP Port**: **5060**
- Click **Finish**
- Select **Add Signaling Interface**
- **Name**: **Colt**
- **Media IP: 10.10.10.20** (External Address for calls toward Colt)
- **TCP Port**: **5060**
- **UDP Port**: **5060**
- Click **Finish**

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



### 7.3.4. End Point Flows

The End Point Flows allow the Interfaces, Policies and Profiles administered to be used to transport the SIP traffic. From the left-hand menu select Device Specific Settings → Endpoint Flows.

- Select the **Server Flows** tab

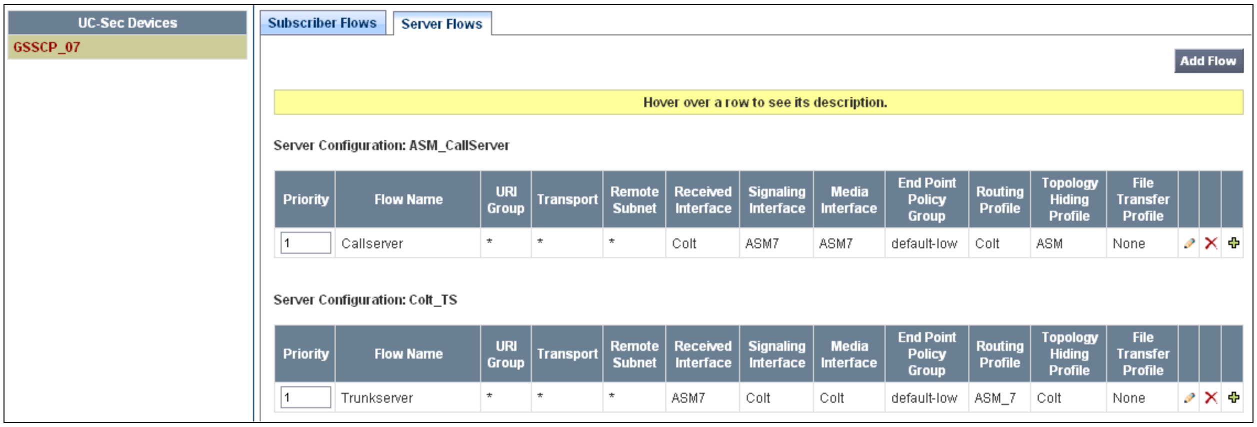
To add the settings for Fixed call flow to Session Manager click on **Add Flow**.

- **Name**: **Callserver**
- **Server Configuration**: **ASM7_CallServer**
- **URI Group: ***
- **Transport**: *****
- **Remote Subnet**: *****
- **Received Interface**: **Colt**
- **Signaling Interface**: **ASM7**
- **Media Interface**: **ASM7**
- **End Point Policy Group**: **default-low**
- **Routing Profile**: **Colt**
- **Topology Hiding Profile**: **ASM**
- **File Transfer Profile**: **None**
- Click **Finish**

To add the settings for Fixed call flow to Colt select **Add Flow**.

- **Name**: **TrunkServer**
- **Server Configuration**: **Colt_TS**
- **URI Group**: *****
- **Transport**: *****
- **Remote Subnet**: *****
- **Received Interface**: **ASM7**
- **Signaling Interface**: **Colt**
- **Media Interface**: **Colt**
- **End Point Policy Group**: **default-low**
- **Routing Profile**: **ASM_7**
- **Topology Hiding Profile**: **Colt**
- **File Transfer Profile**: **None**
- Click **Finish**

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



# 8. Colt Configuration

The configuration of the Colt equipment for interoperability with the Avaya Enterprise Site is not covered in this document. Any further information required can be obtained through the local Colt representative.

# 9. Verification Steps

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

- From the 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.**

This is the SIP Entity link to the Communication Manager.

Summary View

1 Item | Refresh Filter: Er

| Details | Session Manager Name | SIP Entity Resolved IP | Port | Proto. | Conn. Status | Reason Code | Link Statu |
|---|---|---|---|---|---|---|---|
| ▶ Show | ASMdot7 | 10.10.7.52 | 5060 | TCP | Up | 200 OK | Up |

This is the SIP Entity link to the Avaya Session Border Controller Advanced for Enterprise.

Summary View

1 Item | Refresh Filter: Enable

| Details | Session Manager Name | SIP Entity Resolved IP | Port | Proto. | Conn. Status | Reason Code | Link Status |
|---|---|---|---|---|---|---|---|
| ▶ Show | ASMdot7 | 10.10.7.180 | 5060 | TCP | Up | 420 Bad Extension | Up |

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 T00007   in-service/idle     no
0001/003 T00008   in-service/idle     no
0001/004 T00009   in-service/idle     no
0001/005 T00010   in-service/idle     no
```

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

# 10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Communication Manager, Avaya Aura® Session Manager and the Avaya Session Border Controller Advanced for Enterprise to Colt SIP Trunk Service. The testing was successfully performed with Colt, refer to **Section 2.2** for more details.

# 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 6.03, February 2011.

[2] *Administering Avaya Aura® System Platform*, Release 6.03, February 2011.

[3] *Administering Avaya Aura® Communication Manager*, August 2010, Document Number 03-300509.

[4] *Avaya Aura® Communication Manager Feature Description and Implementation,* May 2009, Document Number 555-245-205.

[5] *Upgrading Avaya Aura® System Manager toRelease6.2*, 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] 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.