Avaya Solution & Interoperability Test Lab

Application Notes for British Telecom NOAS SIP Trunk Service with Avaya Aura® Communication Manager R6.0.1, Avaya Aura® Session Manager R6.1 and Avaya Session Border Controller Advanced for Enterprise – Issue 1.0

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between British Telecom NOAS 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 Advanced for Enterprise. British Telecom 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 focused on the SIP Trunking aspect of the Avaya Session Border Controller Advanced for Enterprise. Advanced enterprise capabilities such as Remote Worker "a.k.a. Remote SIP Endpoints", dual forking, and TLS/SRTP were not tested. As a result, the Avaya Session Border Controller for Enterprise is also considered Compliance Tested for this solution.

1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between British Telecom (BT) NOAS 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 Advanced for Enterprise. Customers using this Avaya SIP-enabled enterprise solution with the British Telecom NOAS 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 Session Manager and Communication Manager. The enterprise site was configured to use the SIP Trunk Service provided by British Telecom. 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 British Telecom. 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 British Telecom to PSTN. Outgoing calls from the enterprise to the PSTN were made from H.323, SIP, Digital and Analogue telephones.
- Calls using G.729 and G.711A codec's.
- DTMF transmission using RFC 2833 with successful Vector navigation for inbound and outbound calls.
- Telephony features such as hold and resume, transfer, conference and call forwarding.
- Caller ID presentation and Caller ID restriction.
- Direct IP-to-IP media (also known as "shuffling") with SIP and H.323 telephones.
- Call coverage and call forwarding for endpoints at the enterprise site.
- Short dial numbers were tested including Directory Enquiries and the Emergency Services.
- Fax transmission using the T.38 standard.

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for the BT NOAS 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.

2.3. Support

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

For technical support on British Telecom NOAS products please contact the British Telecom authorized representative.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to the BT NOAS 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, a PC running Avaya one-X Communicator, a B179 Conference phone, 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.

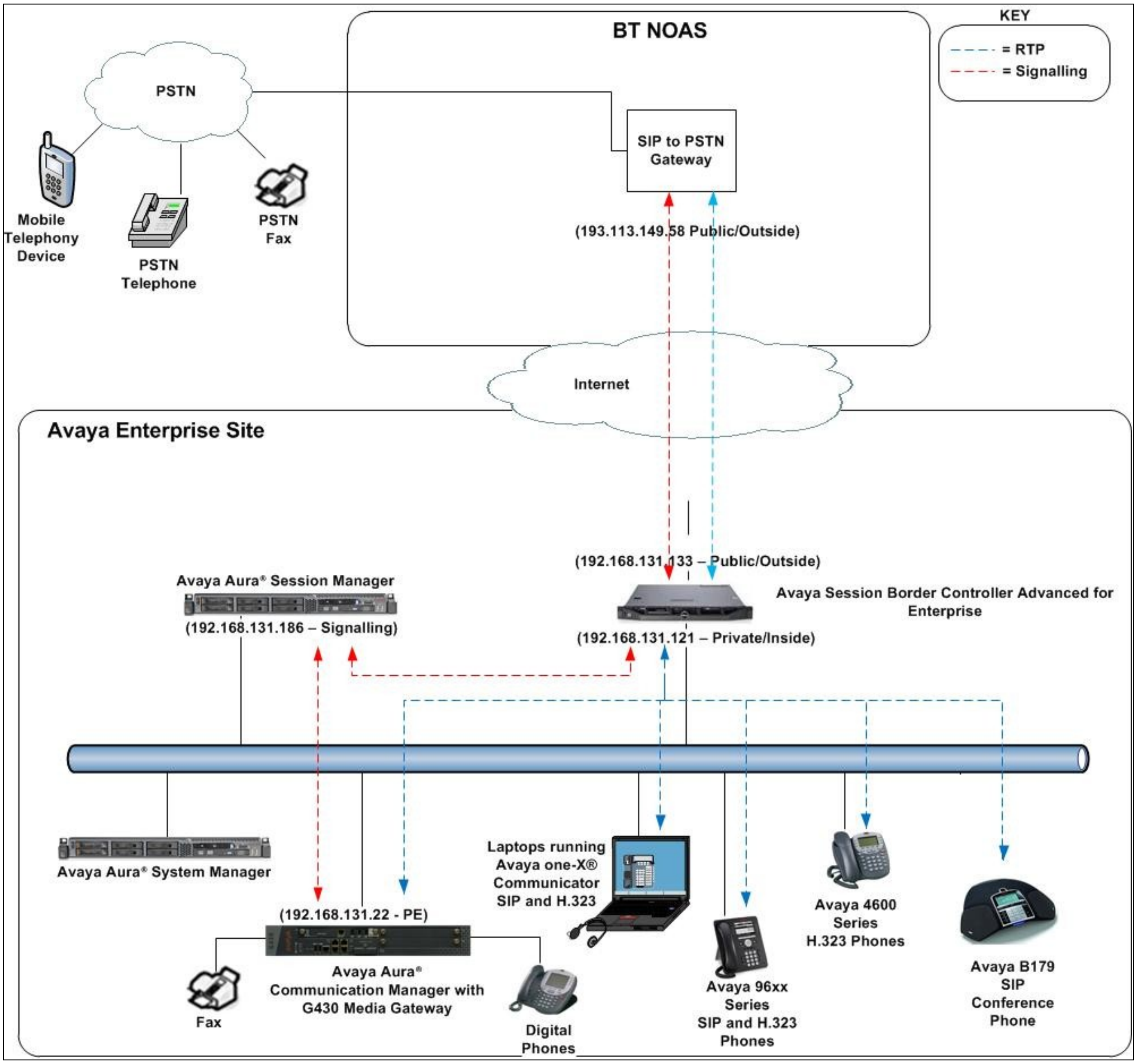


Figure 1: BT NOAS SIP Solution Topology

4. Equipment and Software Validated

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

Equipment	Software Release/Version
Avaya S8300 Server	Avaya Aura® Communication Manager R6.0.1 (R016x.00.1.510.1)
Avaya G430 Media Gateway MM711 Analogue MM712 Digital MGP Firmware	 HW31 FW093 HW07 FW009 30.12.1
Avaya S8800 Server	Avaya Aura® Session Manager R6.1 (6.1.6.0.616008)
Avaya S8800 Server	Avaya Aura® System Manager R6.1 (6.1.0.0.7345-6.1.5.606) Update revision No: 6.1.10.1.1774
Dell R310 Server	Avaya Session Border Controller Advanced for Enterprise (4.0.5.Q02)
Avaya 9620 IP Phone (H.323)	3.11
Avaya 9620 IP Phone (SIP)	2.6.4.0
Avaya 2420 Digital Phone	N/A
Analogue Phone	N/A
Avaya 4620 IP Phone (H.323)	2.9
Avaya one-X® Communicator	6.1
BT NOAS SIP Trunking	3.120.5.17

5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring Communication Manager for SIP Trunking. SIP trunks are established between Communication Manager and Session Manager. These SIP trunks will carry SIP Signaling associated with the BT NOAS SIP Trunk Service. For incoming calls, the Session Manager receives SIP messages from BT 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 the Session Manager. The Session Manager directs the outbound SIP messages to the BT 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 S8300 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 BT 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 the **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, **rom_sm6** and **192.168.131.186** 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               192.168.131.22
 rom_sm6             192.168.131.186
 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 **rom2.bt.com**
- By default, **IP-IP Direct Audio** (both **Intra-** 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 by the IP network region. In this case, codec set **3** was used

```
 change ip-network-region 1                                     Page   1 of  20
                              IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: rom2.bt.com
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 3              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 60001
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? y
  H.323 Link Bounce Recovery? y              RSVP Refresh Rate(secs): 15
 Idle Traffic Interval (sec): 20      Retry upon RSVP Failure Enabled? y
   Keep-Alive Interval (sec): 5                 RSVP Profile: guaranteed-service
            Keep-Alive Count: 5       RSVP unreserved (BBE) PHB Value: 46
```

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 BT 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
```

BT NOAS SIP Trunk Service uses t.38 or ax communication. Configure the T.38 fax protocol by setting the **Fax Mode** to **t.38-standard** on **Page 2** of the codec set form as shown below.

```
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 BT NOAS 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 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 **rom_sm6**), 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 **Far-end Domain** is set as **rom2.bt.com**
- 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 4
                                SIGNALING GROUP

 Group Number: 4              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n                                                SIP Enabled LSP? n
     IP Video? y          Priority Video? n        Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM



   Near-end Node Name: procr                  Far-end Node Name: rom_sm6
 Near-end Listen Port: 5060                 Far-end Listen Port: 5060
                                         Far-end Network Region: 1

Far-end Domain: rom2.bt.com
                                             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): 5                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                     Initial IP-IP Direct Media? y
H.323 Station Outgoing Direct Media? y              Alternate Route Timer(sec): 6
```

5.6. Administer SIP Trunk Group

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

- Set the **Group Type** field to **sip**
- Choose a descriptive **Group Name**
- Specify a trunk access code (**TAC**) consistent with the dial plan, i.e. **104**
- 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 4                                                Page   1 of  21
                                TRUNK GROUP

Group Number: 4                    Group Type: sip           CDR Reports: y
  Group Name: sip trunk to Rom SM6        COR: 1        TN: 1        TAC: 104
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                     Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 4
                                                     Number of Members: 4
```

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 BT to prevent unnecessary SIP messages during call setup.

```
add trunk-group 4                                                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 **public.** This allows the number to be sent to BT with the + used in the E164 numbering format.

```
add trunk-group 4                                               Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                        Maintenance Tests? y

                     Numbering Format: public
                                                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: 101
```

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 Unknown Numbering

Use the **change public-unknown-numbering** command to configure Communication Manager to send the calling party number. In the sample configuration, all stations with a **4**-digit extension beginning with **3** will send the calling party number **44207xxxxxxx** to BT NOAS 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 public-unknown-nmbering 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  3             4         44207xxxxxxx    12   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 BT NOAS 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 4**, 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                    11    11     4         pubu         n
    00                   13    13     4         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                                        none
 2: y y y y y n  n            rest                                        none
```

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

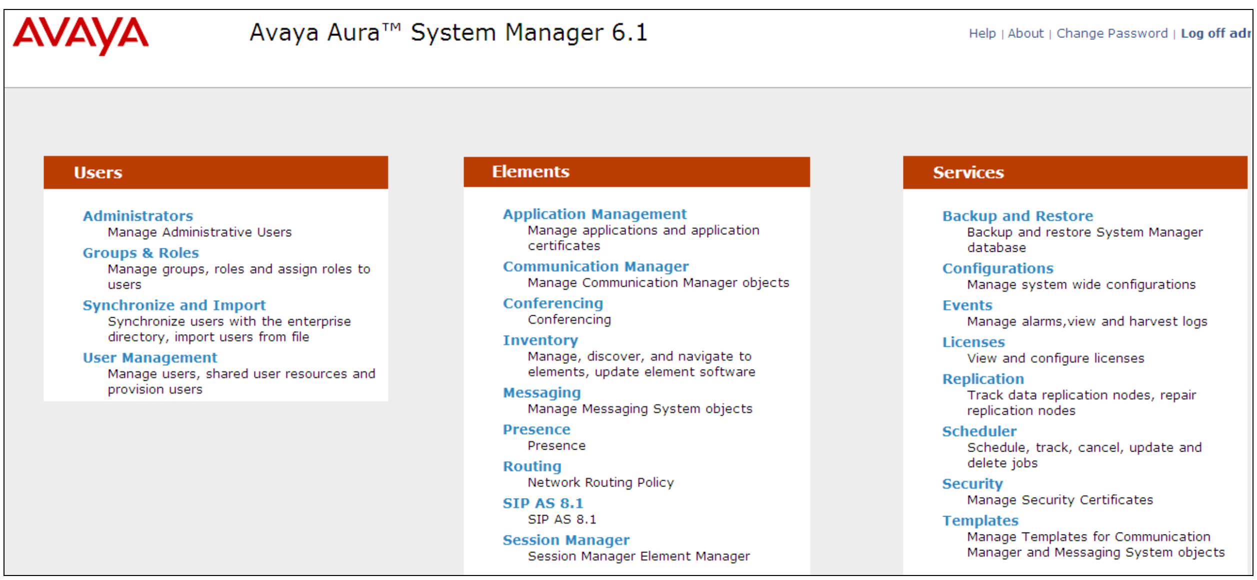
6. Configuring Avaya Aura® Session Manager

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

- Log into Avaya Aura® System Manager
- Administer SIP domain
- Administer Locations
- Administer Adaptations
- 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.

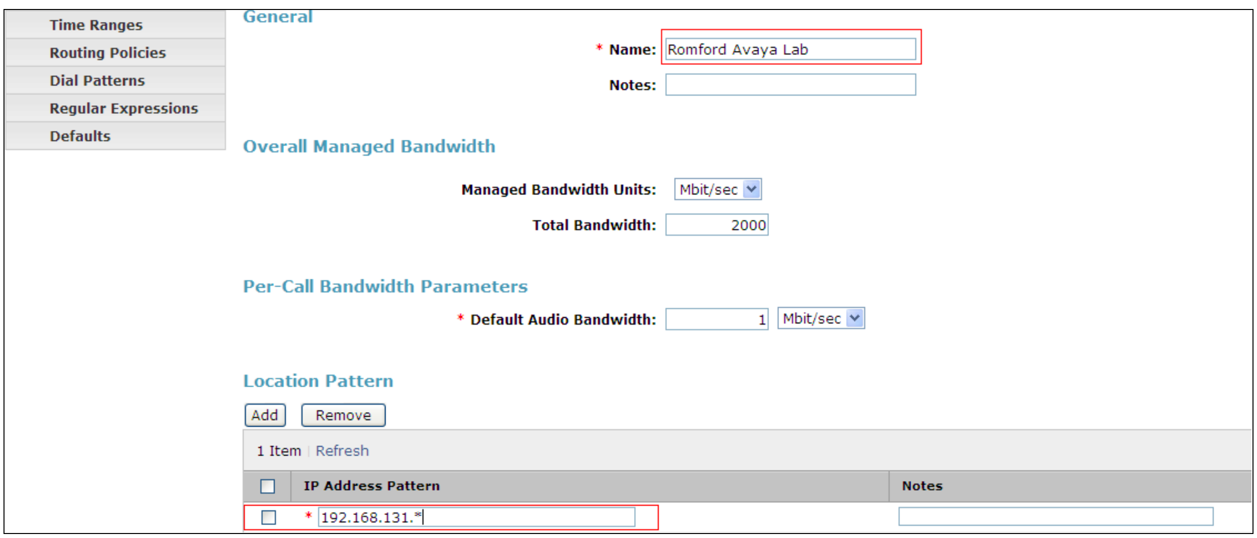


6.2. Administer SIP domain

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 (not shown) to create a new SIP domain entry. In the **Name** field, enter the domain name (e.g., **rom2.bt.com**). Click **Commit** to save changes.

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. Below is the location configuration used for the simulated enterprise.



6.4. Administer Adaptations

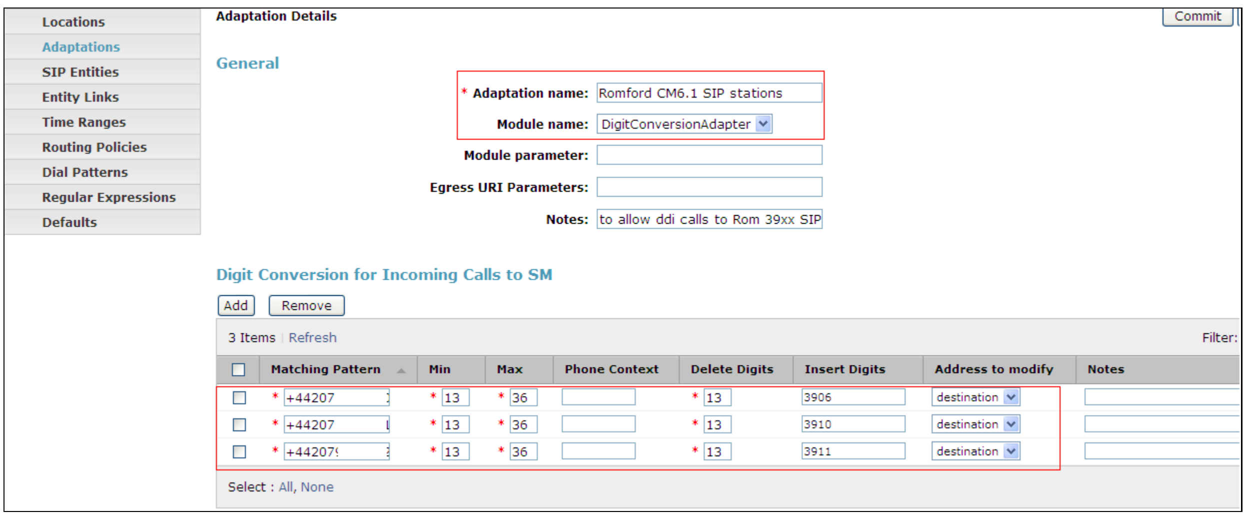
BT use a Session Manager Adaptation to present calls to the Communication Manager. This is used in place of the incoming call handling administered using the SAT terminal.

On the **Routing** tab, select **Adaptations** from the lefthand menu.Click on **New** (not shown).

- For the **Adaptation Name** give the adaption a descriptive title
- For the **Module Name** enter **DigitConversionAdapter**

In the section **Digit Conversion for Outgoing Calls to SM.**

- Under **Matching Pattern** enter **+44207xxxxxxx**
- Under **Min** and **Max** enter the Minimum and Maximum digits expected
- Under **Delete Digits** enter **13** to remove the whole number
- Under **Insert Digits** enter **3xxx**
- Under **Address to Modify** choose **destination** from the drop down box



6.5. Administer SIP Entities

A SIP Entity must be added for each SIP-based telephony system supported by a SIP connection to the Session Manager. To add a SIP Entity, 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:**

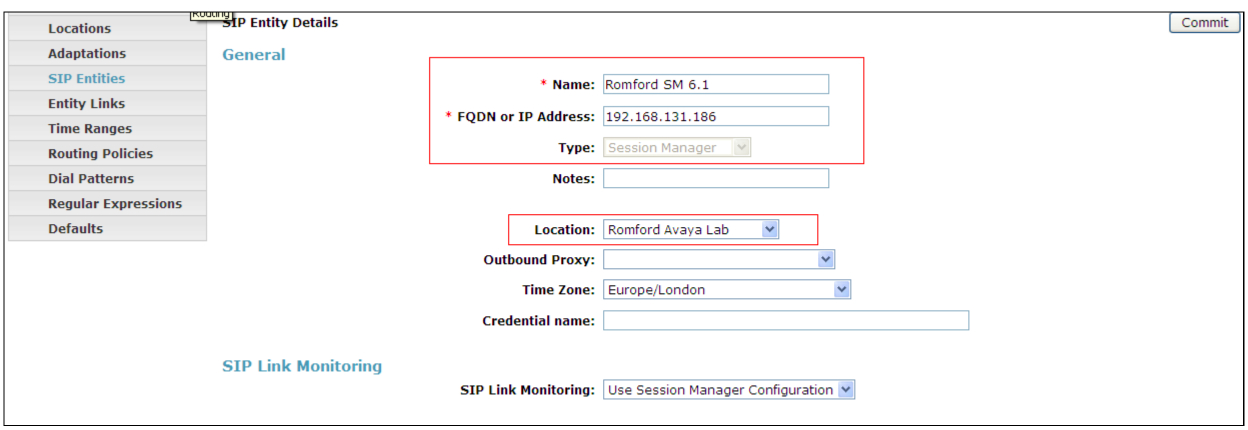
- In the **Name** field enter an informative name
- In the **FQDN or IP Address** field enter the IP address of Session Manager or the signaling interface on the connecting system
- In the **Type** field use **Session Manager** for a Session Manager SIP entity, **CM** for a Communication Manager SIP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

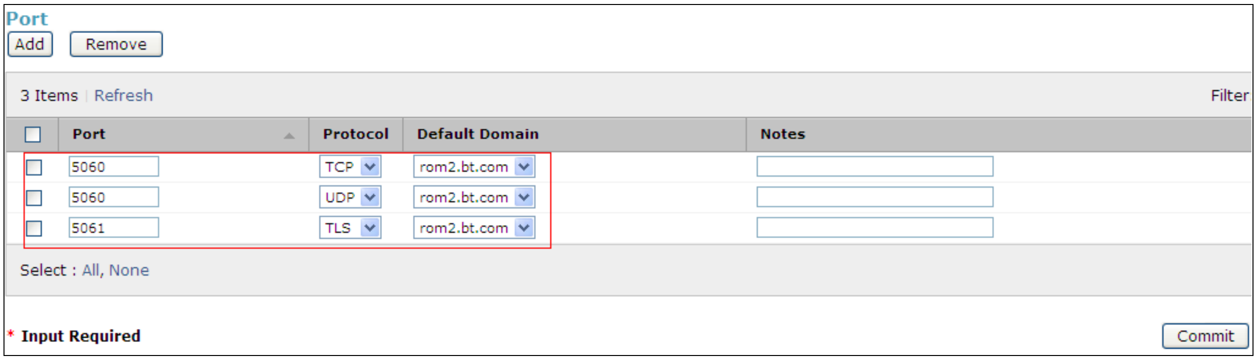
6.5.1. Avaya Aura® Session Manager SIP Entity

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



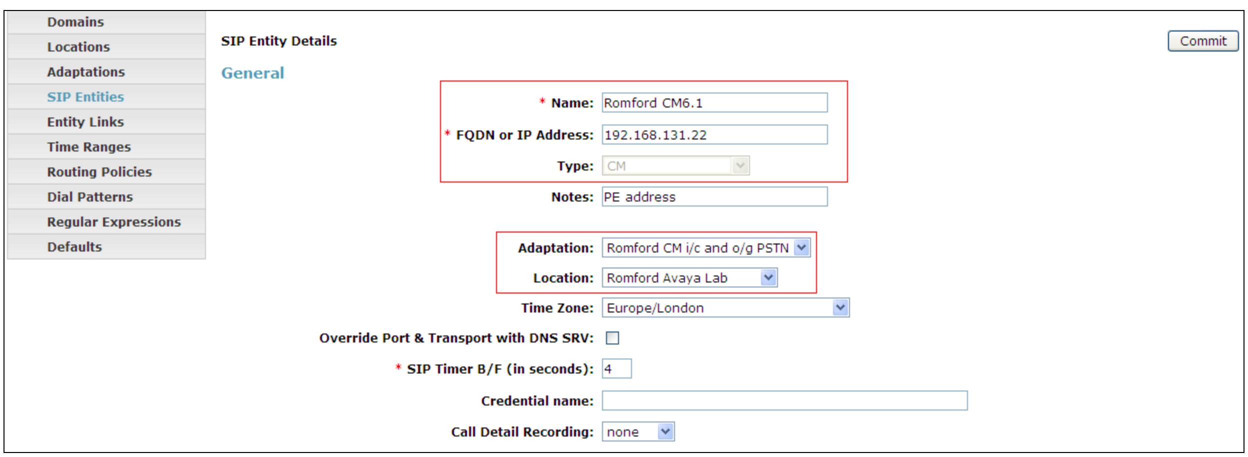
The Session Manager must be configured with the port numbers on the protocols that will be used by the other SIP entities. To configure these scroll to the bottom of the page and under **Port**, click **Add,** then edit the fields in the resulting new row.

- In the **Port** field enter the port number on which the system listens for SIP requests
- In the **Protocol** field enter the transport protocol to be used for SIP requests
- In the **Default Domain** field from the drop down menu, select **rom2.bt.com** as the default domain



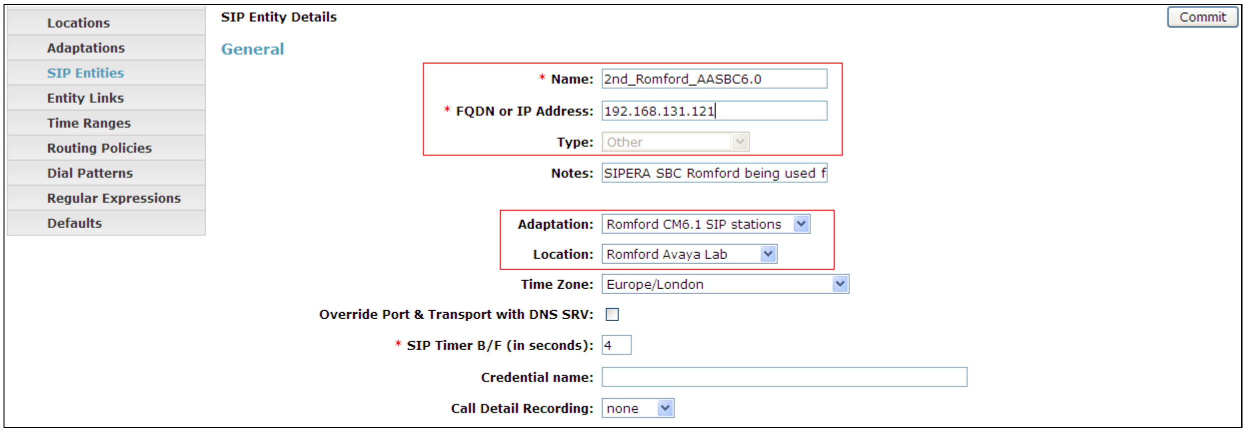
6.5.2. Avaya Aura® Communication Manager SIP Entity

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**.



6.5.3. 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 calls. The **FQDN or IP Address** field is set to the IP address of the private interfaces administered in **Section 7** of this document.



6.6. Administer Entity Links

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

- 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 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.

1 Item | Refresh

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy
* Romford SM 6.1_Ro	* Romford SM 6.1	TCP	* 5060	* Romford CM6.1	* 5060	Trusted

1 Item | Refresh

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy
* Romford SM 6.1 to l	* Romford SM 6.1	UDP	* 5060	* Romford AASBC 6.0	* 5060	Trusted

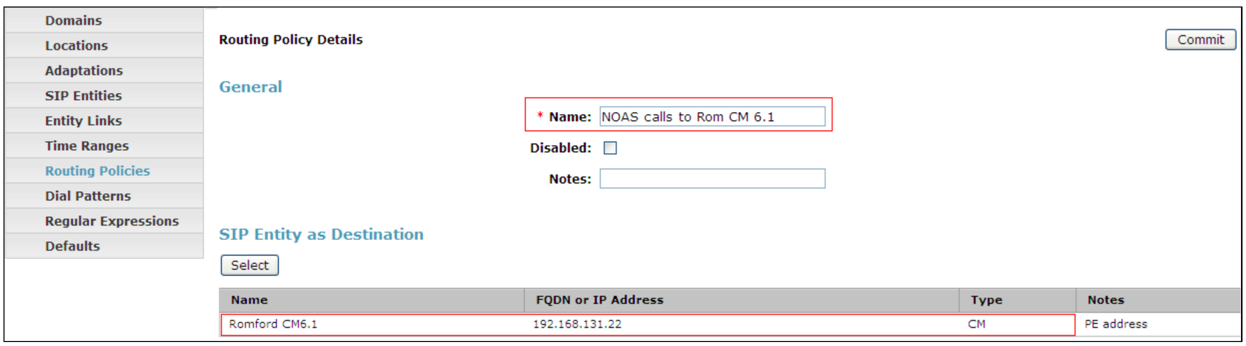
6.7. Administer Routing Policies

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

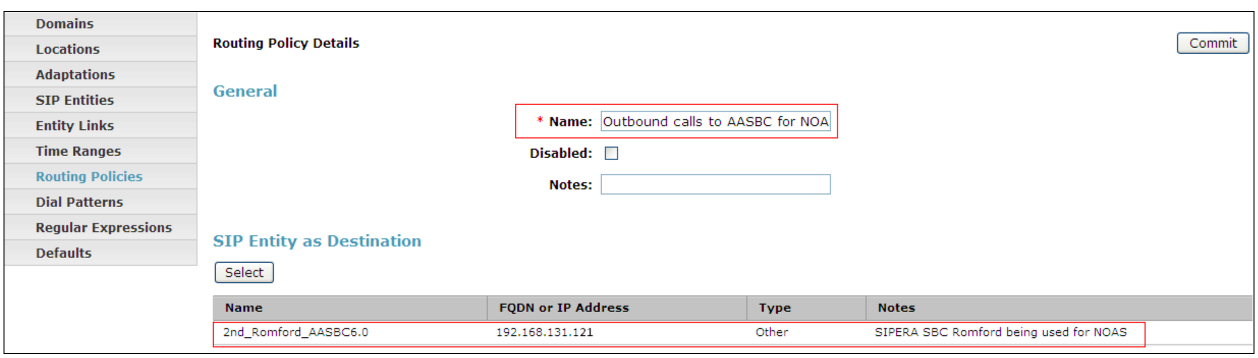
Under **General**:

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

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:



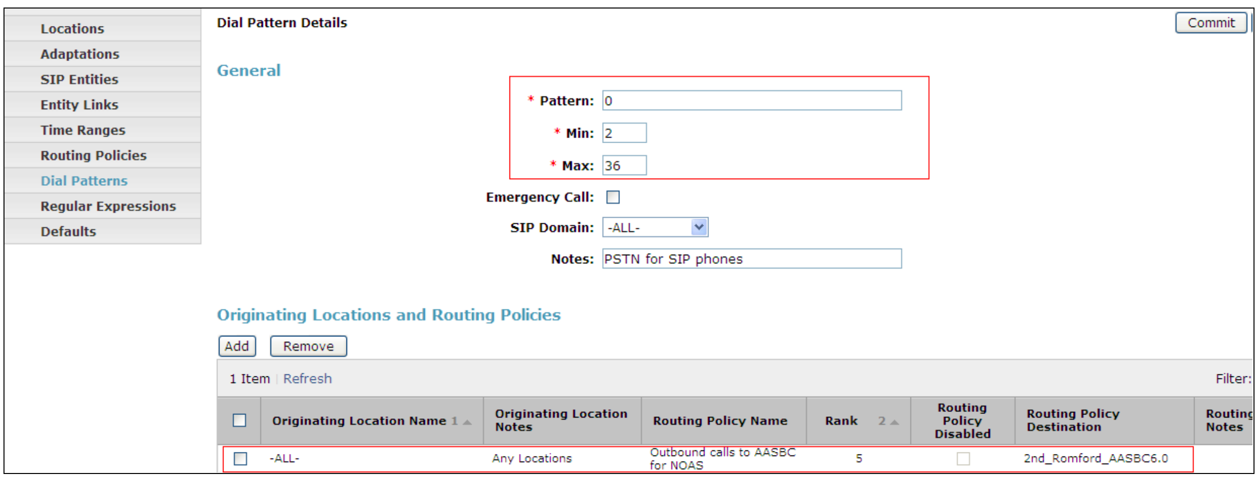
6.8. Administer Dial Patterns

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

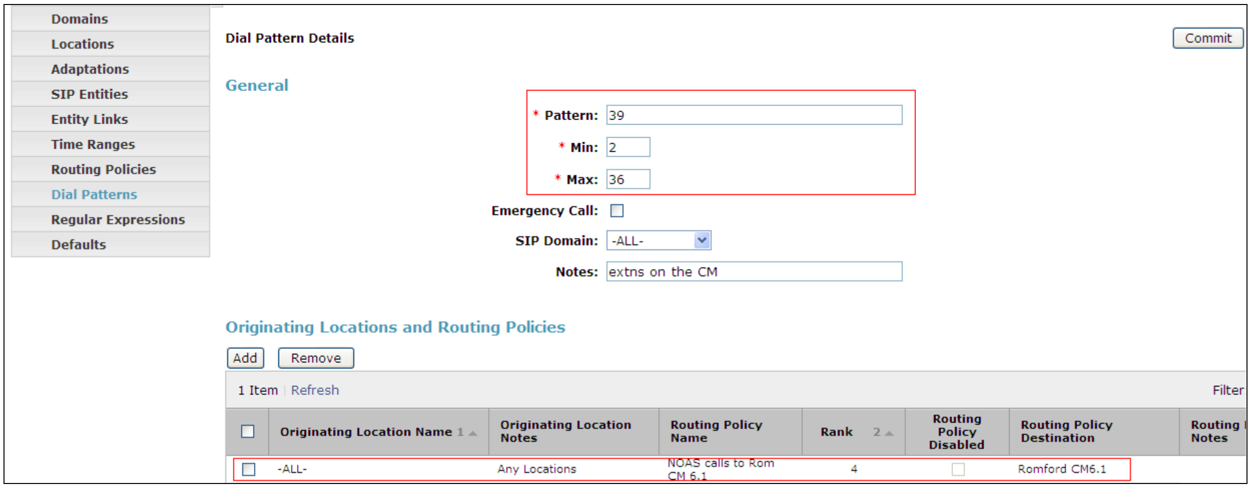
Under **General**:

- In the **Pattern** field enter a 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 BT NOAS SIP Trunk Service.



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

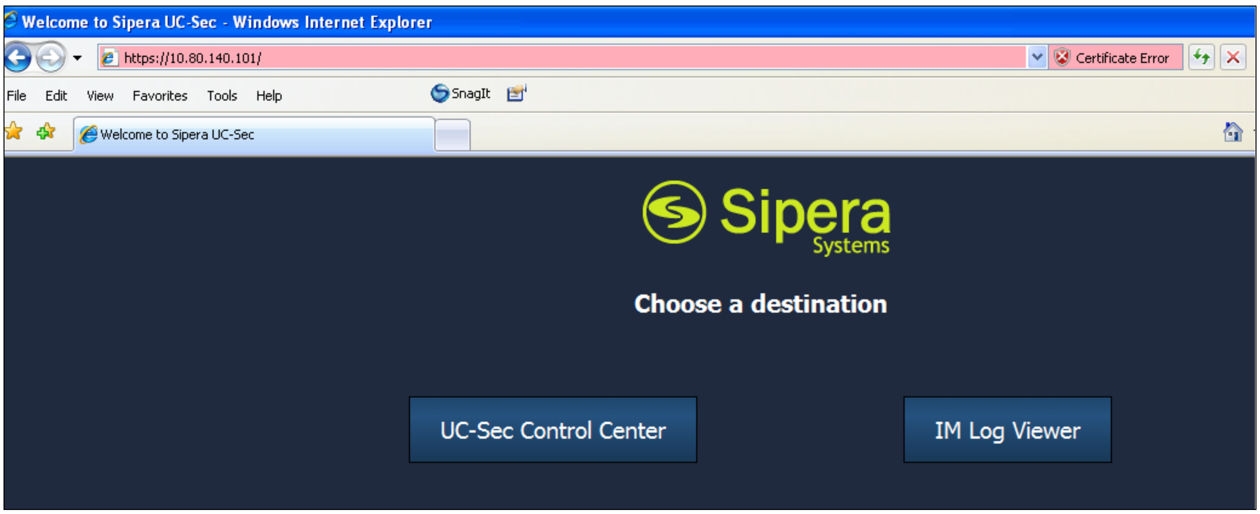


7. Avaya Session Border Controller Advanced for Enterprise Configuration

This section provides the procedures for configuring Session Border Controller Advanced or Enterprise.

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

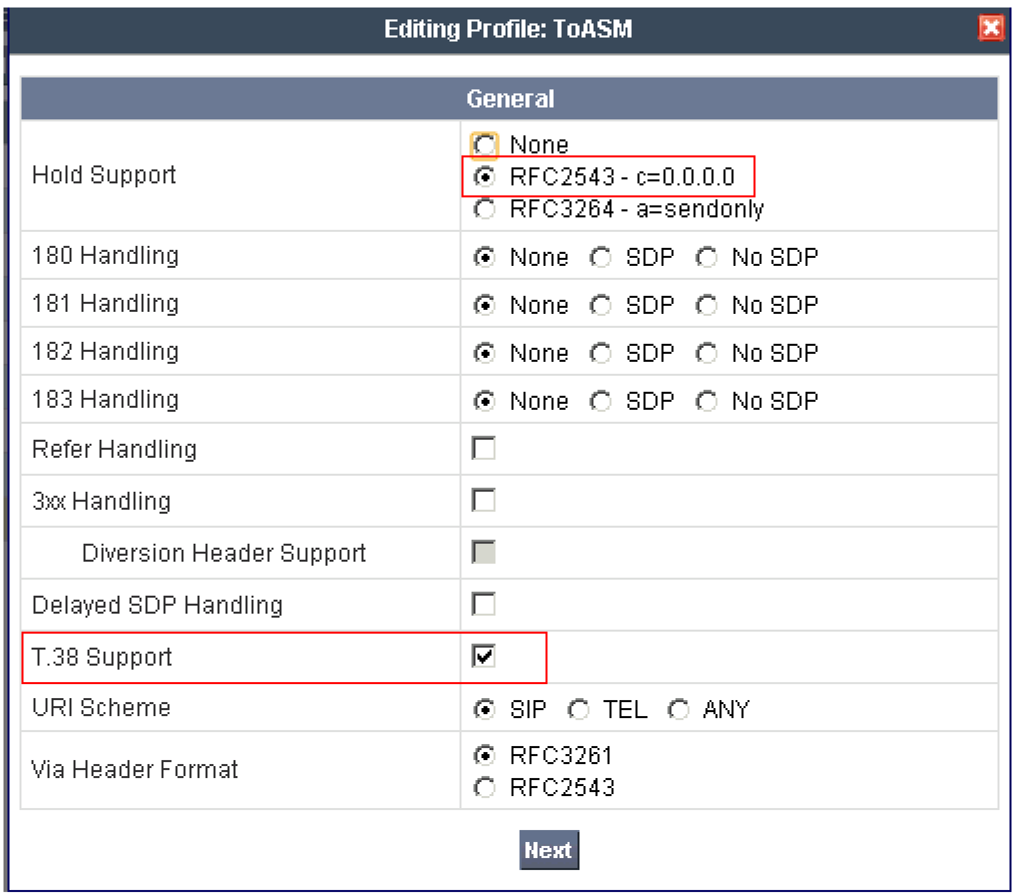
When selected, Global Profiles allows for configuration of parameters across all UC-Sec appliances.

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 lefthand menu select **Global Profiles → Server Interworking** and click on **Add Profile.**

- Enter profile name**: ToASM** and click **Next**
- Check **Hold Support= RFC2543**
- 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**.

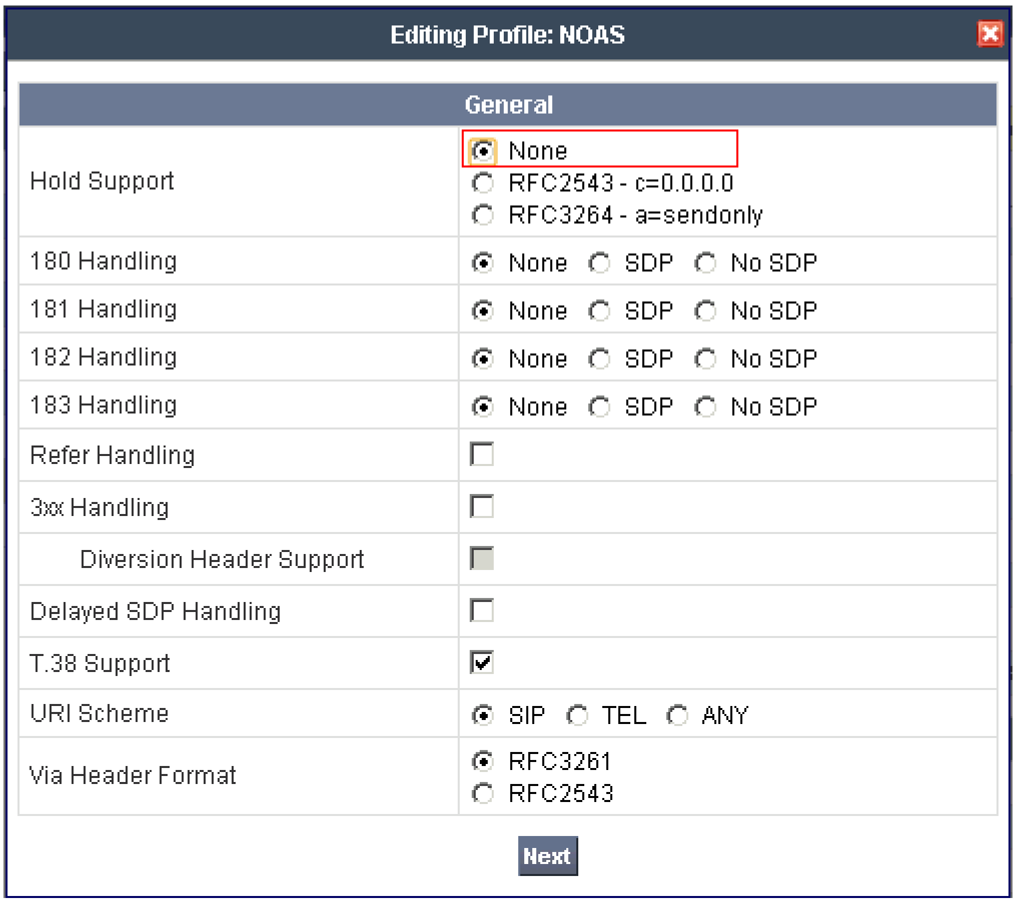


7.2.2. Server Internetworking – BT NOAS Side

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 Profile.**

- Enter profile name**: NOAS** and click on **Next**
- Check **Hold Support= None**
- 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**.

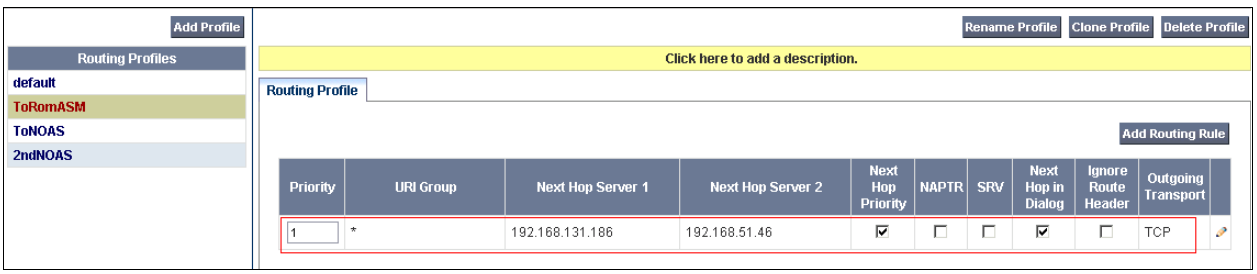


7.2.3. Routing – Avaya side

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

- Enter Profile Name: **ToRomASM**
- Click **Next** (not shown)
- **Next Hop Server 1: 192.168.131.186 (Session Manager IP address)**
- **Next Hop Server 2: 192.168.51.46 (Session Manager backup IP address)**
- Select **Routing Priority Based on Next Hop Server**
- Select **Use Next Hop for In-Dialog Messages**
- **Outgoing Transport: TCP**

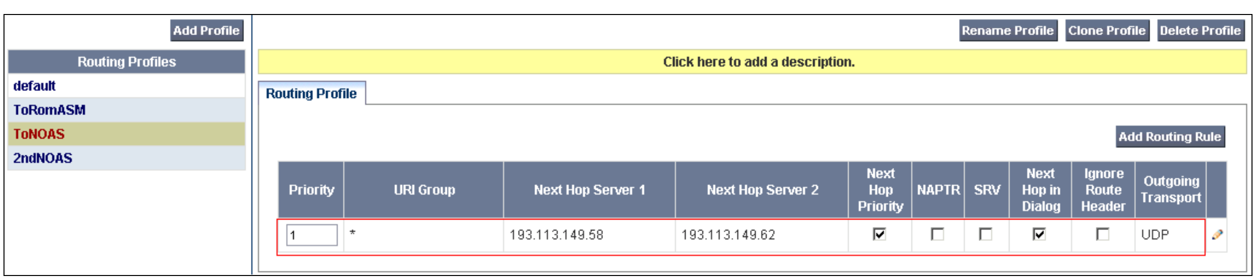
Click **Finish** (not shown).



7.2.4. Routing – BT NOAS side

The Routing Profile allows you to manage 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: **ToNOAS**
- Click **Next**
- **Next Hop Server 1: 193.113.149.58 (IP Address provided by BT)**
- **Next Hop Server 1: 193.113.149.62 (IP Address provided by BT)**
- Select **Routing Priority Based on Next Hop Server**
- Select **Use Next Hop for In-Dialog Messages**
- **Outgoing Transport: UDP**
- Click **Finish** (not shown)



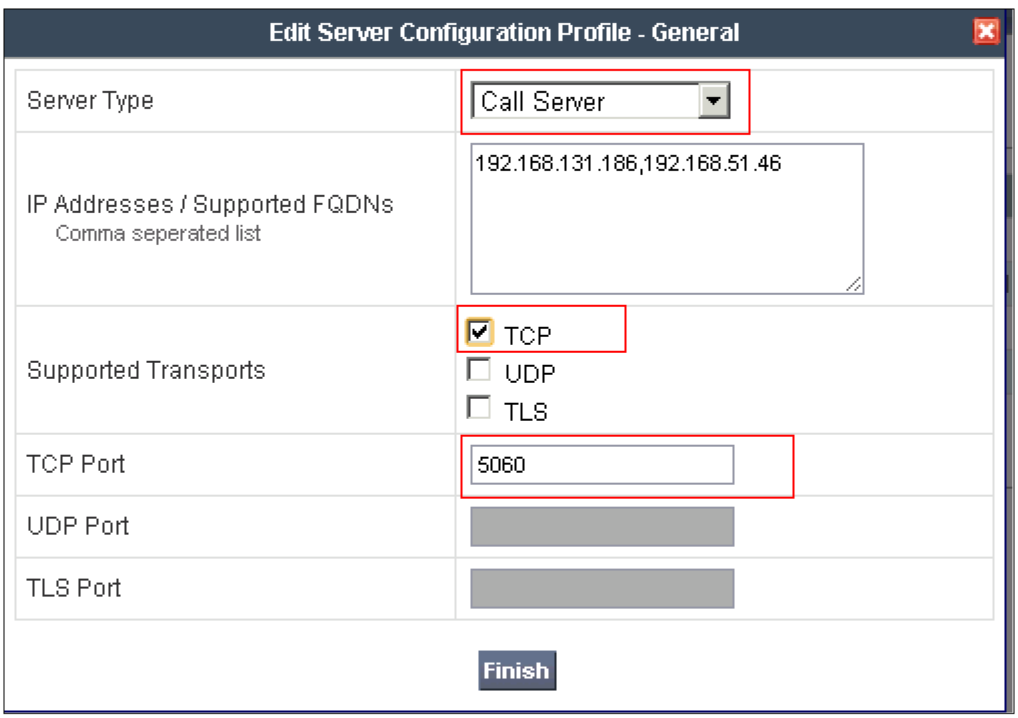
7.2.5. Server Configuration– Avaya SM

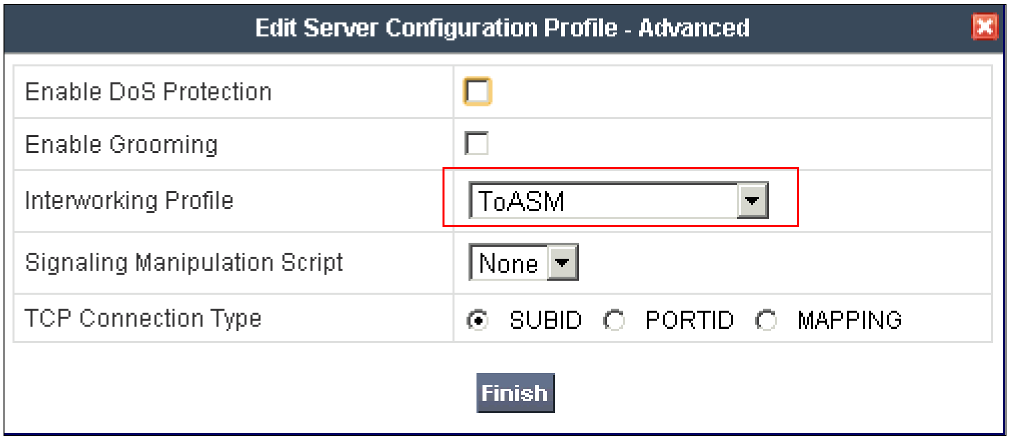
The **Server Configuration** screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. Together, these tabs allow you to configure and manage various SIP call server-specific parameters such as TCP and UDP port assignments, IP Server type, heartbeat signaling parameters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

- **Enter profile name: ASM_CallServer**

On the **Add Server Configuration Profile** Tab:

- Select Server Type**: Call Server**
- **IP Address: 192.168.131.186,192.168.51.46 (Session Manager IP Addresses)**
- **Supported Transports**: Check **TCP**
- **TCP Port:5060**
- Click on **Next** for the **Authentication** and **Heartbeat** tabs.
- On the **Advanced** Tab
- Select **ToASM** for Interworking Profile
- Click **Next**
- Click **Finish**




Edit Server Configuration Profile - Advanced
Enable DoS Protection
Enable Grooming
Interworking Profile
ToASM
Signaling Manipulation Script
None
TCP Connection Type
SUBID
PORTID
MAPPING
Finish

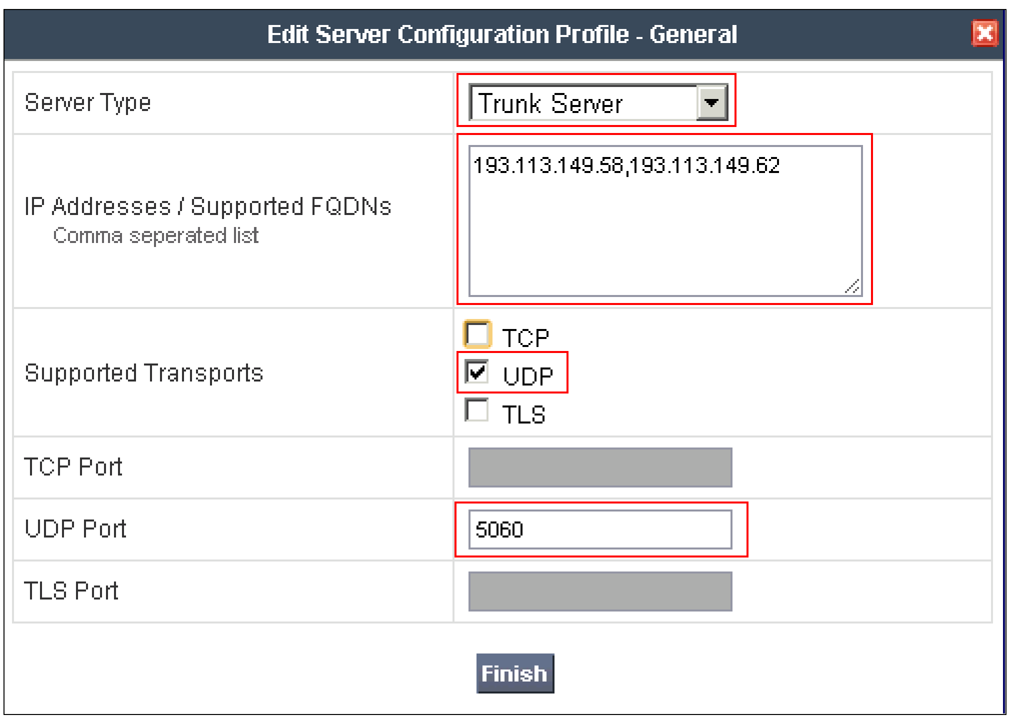
7.2.6. Server Configuration– BT NOAS side

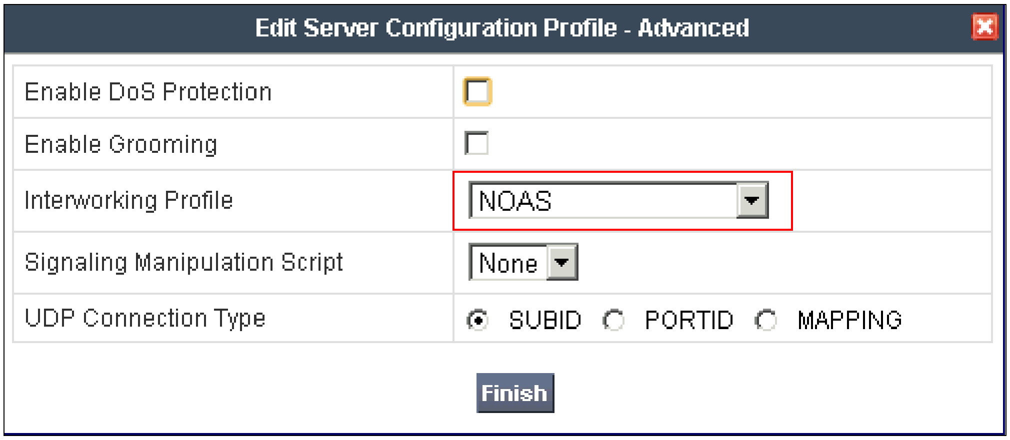
The **Server Configuration** screen contains fourtabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. Together, these tabs allow you to configure and manage various SIP call server-specific parameters such as TCP and UDP port assignments, server type, heartbeat signaling parameters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

- **Name: ToNOAS**

On the **Add Server Configuration Profile** Tab:

- Click on **Edit**
- Select Server Type**: Trunk Server**
- **IP Address: 193.113.149.58,193.113.149.62 (BT Trunk Server)**
- **Supported Transports**: Check **UDP**
- **UDP Port: 5060**
- Click **Next**
- Click on **Next** for the **Authentication** and **Heartbeat** tabs.
- On the **Advanced** Tab
- Select **NOAS** for Interworking Profile
- Click **Next**
- Click **Finish**



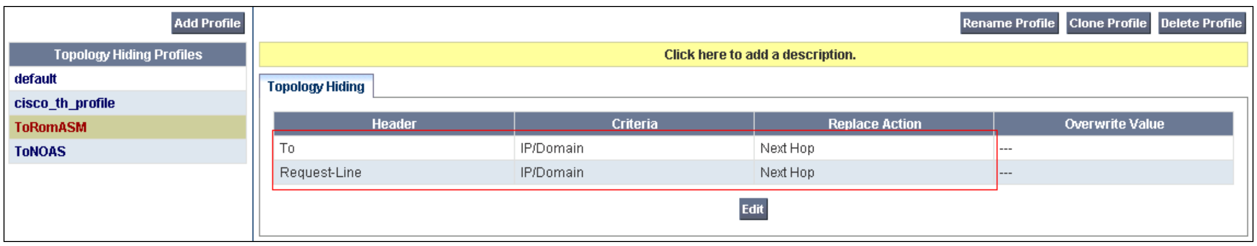

Edit Server Configuration Profile - Advanced
Enable DoS Protection
Enable Grooming
Interworking Profile
NOAS
Signaling Manipulation Script
None
UDP Connection Type
SUBID
PORTID
MAPPING
Finish

7.2.7. Topology Hiding – Avaya side

The **Topology Hiding** screen allows you to manage 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 **Header To** and **Request Line** select **IP/Domain** under **Criteria** and **Next Hop** under **Replace Action.**
- Click **Finish**

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

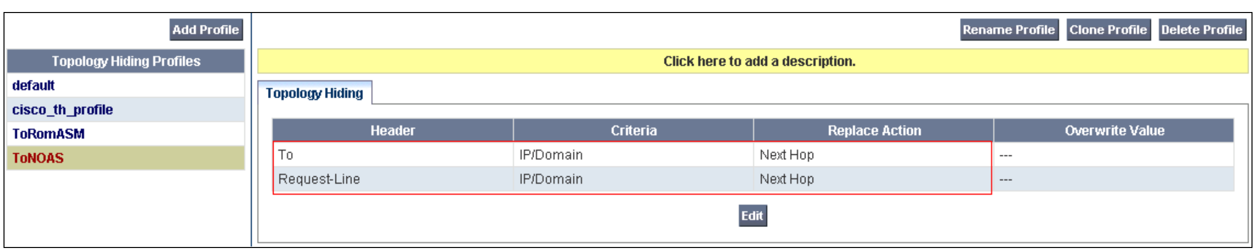


7.2.8. Topology Hiding – BT side

The **Topology Hiding** screen allows you to manage 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: ToNOAS**
- For the **Header To** and **Request Line** select **IP/Domain** under **Criteria** and **NextHop** under **Replace Action**
- Click **Finish**

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

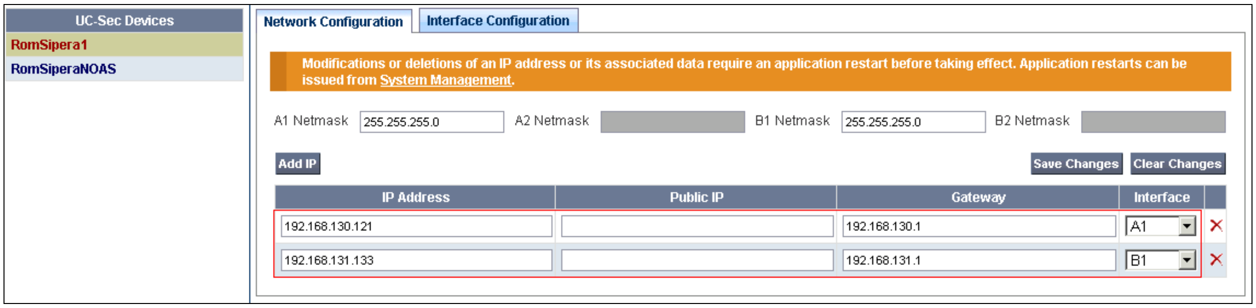


7.3. Device Specific Settings

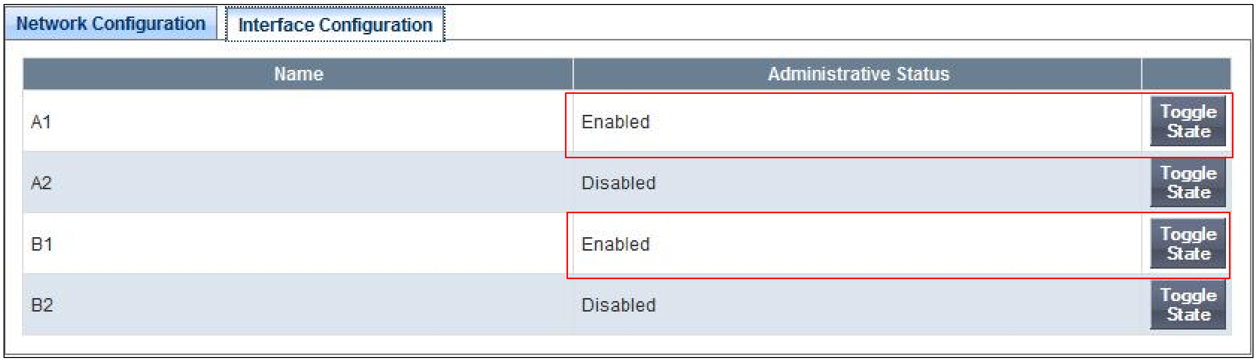
7.3.1. Network Configuration

The Network Configuration 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 phy**sical int**erface **used in the Interface** column



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

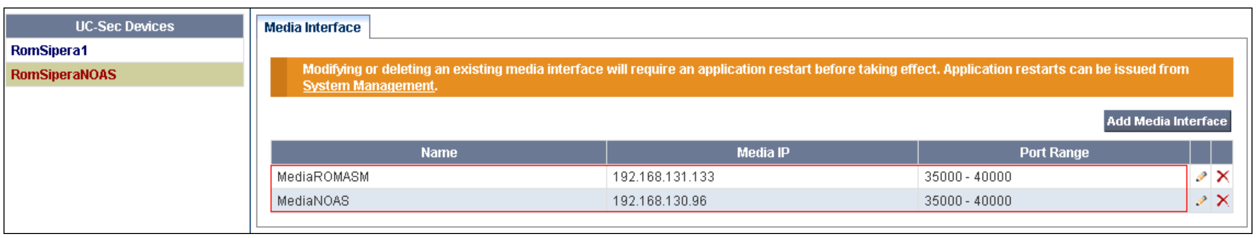


7.3.2. Media Interfaces

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**: **MediaROMASM**
- **Media IP**: **192.168.131.133**(Internal Address for calls toward Session Manager)
- **Port Range**: **35000-40000**
- Click **Finish**
- Select **Add Media Interface**
- **Name**: **MediaNOAS**
- **Media IP**: **192.168.130.96**(External Address for calls toward BT trunk)
- **Port Range**: **35000-40000**
- Click **Finish**
- Select **Add Media Interface**

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

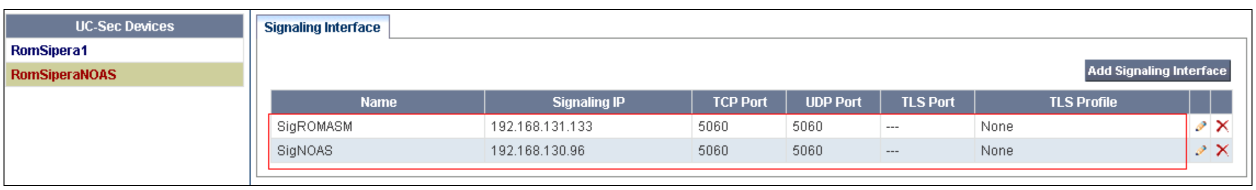


7.3.3. Signaling Interfaces

The Signaling 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**: **SigROMASM**
- **Media IP**: **192.168.131.133** (Internal Address for calls toward Session Manager)
- **TCP Port**: **5060**
- **UDP Port**: **5060**
- Click **Finish**
- Select **Add Media Interface**
- **Name**: **SigNOAS**
- **Media IP: 192.168.130.96**(External Address for calls toward BT)
- **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

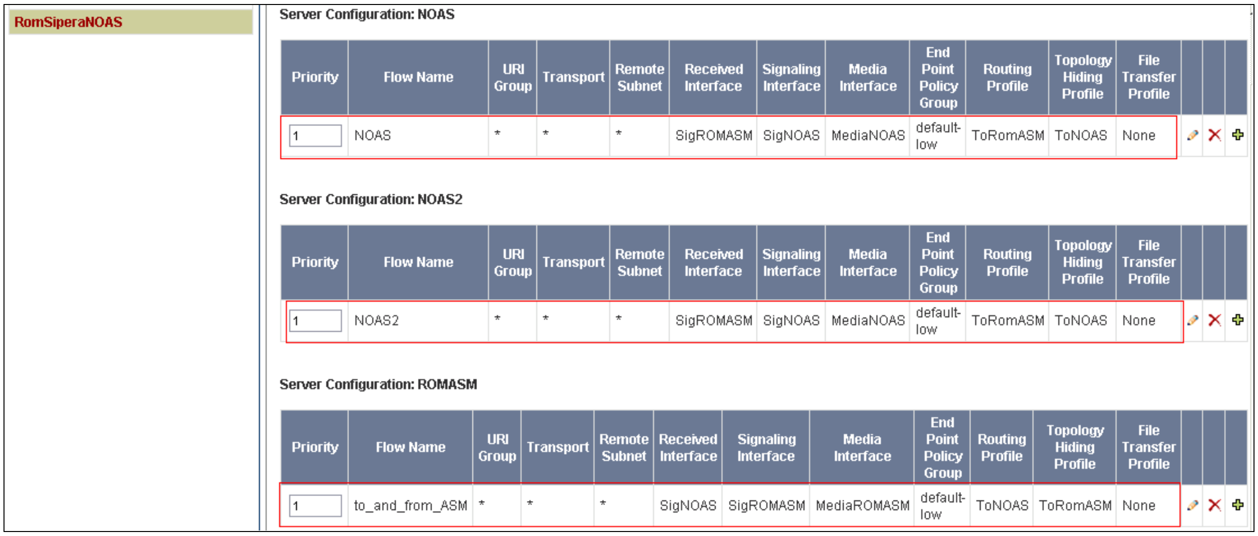
To add the settings for Fixed call flow to Session Manager Click on select **Add Flow.**

- **Name**: **Callserver**
- **Server Configuration**: **ROMASM**
- **URI Group: ***
- **Transport**: *****
- **Remote Subnet**: *****
- **Received Interface**: **SigNOAS**
- **Signaling Interface**: **SigROMASM**
- **Media Interface**: **MediaROMASM**
- **End Point Policy Group**: **default-low**
- **Routing Profile**: **ToNOAS**
- **Topology Hiding Profile**: **ToROMASM**
- **File Transfer Profile**: **None**
- Click **Finish**

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

- **Name**: **TrunkServer**
- **Server Configuration**: **NOAS**
- **URI Group**: *****
- **Transport**: *****
- **Remote Subnet**: *****
- **Received Interface**: **SigROMASM**
- **Signaling Interface**: **SigNOAS**
- **Media Interface**: **MediaNOAS**
- **End Point Policy Group**: **default-low**
- **Routing Profile**: **ToRomASM**
- **Topology Hiding Profile**: **ToNOAS**
- **File Transfer Profile**: **None**
- Click **Finish**

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



RomSiperaNOAS

Server Configuration: NOAS

Priority	Flow Name	URI Group	Transport	Remote Subnet	Received Interface	Signaling Interface	Media Interface	End Point Policy Group	Routing Profile	Topology Hiding Profile	File Transfer Profile
1	NOAS	*	*	*	SigROMASM	SigNOAS	MediaNOAS	default-low	ToRomASM	ToNOAS	None

Server Configuration: NOAS2

Priority	Flow Name	URI Group	Transport	Remote Subnet	Received Interface	Signaling Interface	Media Interface	End Point Policy Group	Routing Profile	Topology Hiding Profile	File Transfer Profile
1	NOAS2	*	*	*	SigROMASM	SigNOAS	MediaNOAS	default-low	ToRomASM	ToNOAS	None

Server Configuration: ROMASM

Priority	Flow Name	URI Group	Transport	Remote Subnet	Received Interface	Signaling Interface	Media Interface	End Point Policy Group	Routing Profile	Topology Hiding Profile	File Transfer Profile
1	to_and_from_ASM	*	*	*	SigNOAS	SigROMASM	MediaROMASM	default-low	ToNOAS	ToRomASM	None

8. BT NOAS Configuration

The configuration required by BT to allow the tests to be carried out is not covered in this document and any further information required shown should be obtained through the local BT representative.

9. Verification Steps

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

- 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.**

This is the SIP Entity link to the Communication Manager:

2 Items | Refresh Filter: Enable

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
▶ Show	Romford SM 6.1	192.168.131.22	5060	TCP	Up	200 OK	Up
▶ Show	Leeds SM6.1	192.168.131.22	5061	TLS	Up	200 OK	Up

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

2 Items | Refresh Filter: Enable

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
▶ Show	Romford SM 6.1	192.168.131.133	5060	TCP	Up	200 OK	Up
▶ Show	Leeds SM6.1	192.168.131.133	5060	TCP	Up	200 OK	Up

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

```
status trunk 4

                              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 the BT NOAS SIP Trunk Service. The testing was successfully performed with BT, refer to **Section 2.2** for test results.

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, *D*ocument Number 555-245-205.

[5] *Upgrading Avaya Aura® System Manager toRelease6.0.1*, August 2011.

[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.