



Avaya Solution & Interoperability Test Lab

**Application Notes for Configuring Avaya Aura®
Communication Manager 6.0.1 as an Evolution Server and
Avaya Aura® Session Manager 6.1 to support British
Telecom SIP Trunk Service 2.1.0.8 - Issue 1.0**

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between the British Telecom SIP Trunk Service and an Avaya SIP enabled enterprise solution. The Avaya solution consists of Avaya Aura® Session Manager and Avaya Aura® Communication Manager Evolution Server. British Telecom is a member of the DevConnect Service Provider program.

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

1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between British Telecom (BT) SIP Trunk Service and an Avaya SIP enabled enterprise solution. The Avaya solution consists of Avaya Aura® Session Manager and Avaya Aura® Communication Manager Evolution Server. Customers using this Avaya SIP-enabled enterprise solution with the BT 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 BT.

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 BT. Incoming PSTN calls terminated on SIP, H.323 and Digital telephones at the enterprise.
- Outgoing calls from the enterprise site were completed via BT to PSTN destinations.
- Outgoing calls from the enterprise to the PSTN were made from SIP, H.323 and Digital telephones.
- Calls to Emergency Services numbers such as 999 and 112 were made from the enterprise site via the SIP Trunk to BT.
- Calls were made using G.729, and G.711A codecs.
- Fax calls to/from a Group 3 fax machine to a PSTN connected fax machine using the T.38 mode.
- DTMF transmission using RFC 2833 with successful Voice Mail/Vector navigation for inbound and outbound calls.
- User features such as hold and resume, transfer, conference, call forwarding, etc.
- Caller ID Presentation and Caller ID Restriction.
- Direct IP-to-IP media (also known as “shuffling”) with SIP and H.323 telephones.
- Call coverage and call forwarding for endpoints at the enterprise site.
- Transmission and response of SIP OPTIONS messages sent by BT requiring Avaya response and sent by Avaya requiring BT response.

2.2. Test Results

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

- No inbound toll free numbers were tested, however routing of inbound DID numbers and the relevant number translation was successfully tested.

- G.729 annex b (silence suppression) is not supported by BT SIP Trunk Service and thus was not tested.
- G.711mu is not supported by BT SIP Trunk Service and thus was not tested
- one-X Communicator was not tested using Telecommuter mode. All one-X Communicator test cases were completed using Road Warrior mode.
- For fax calls to be successful BT requested a Min-SE value of 1800 to be set on the Communication Manager (1800 is doubled to make 3600). For incoming calls BT is attempting to use a lower value causing a negotiation of the Min-SE to occur. This increases the number of SIP messages during an inbound call setup.

2.3. Support

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

For technical support on BT products please contact an authorized BT representative.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to the BT SIP Trunk Service. Located at the enterprise site are a Session Manager and Communication Manager. Endpoints are Avaya 96xx series IP telephones (with SIP and H.323 firmware), Avaya 46xx series IP telephones (with H.323 firmware), Avaya Digital telephones and an analogue fax machine. Also included in the test configuration was an Avaya Desktop Video Device incorporating the Avaya Flare experience. For security purposes, any public IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes. Instead, public IP addresses have been replaced with private addresses and all phone numbers have been replaced with arbitrary.

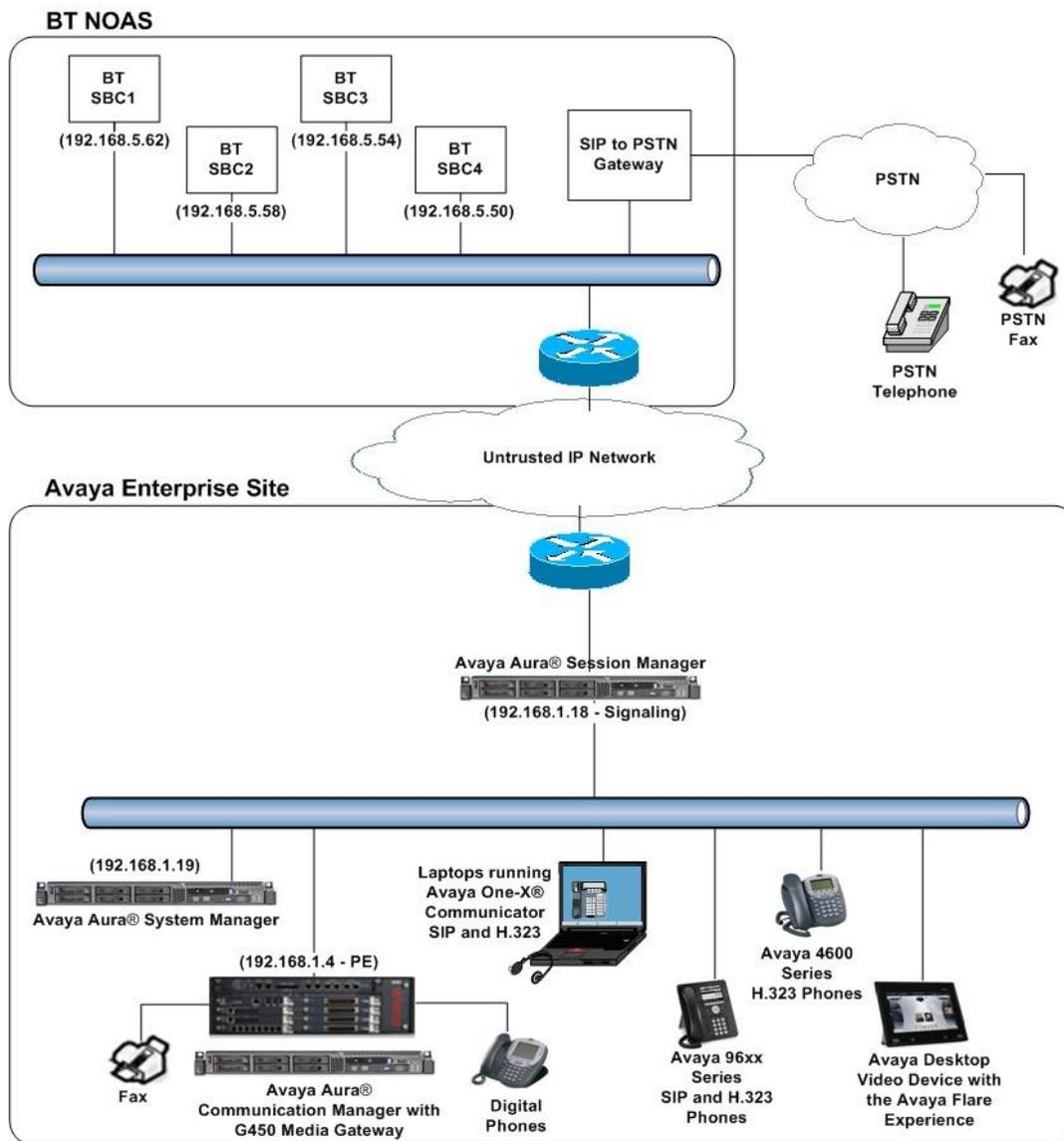


Figure 1: BT Sample Configuration

4. Equipment and Software Validated

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

Equipment	Software
Avaya S8800 Server	Avaya Aura® Communication Manager 6.0.1 (R016x.00.1.510.1) Service Pack 18621 (System Platform 6.0.2.1.5)
Avaya G450 Media Gateway	FW 31.17.1
Avaya S8800 Server	Avaya Aura® Session Manager 6.1 (6.1.0.0.610023)
Avaya S8800 Server	Avaya Aura® System Manager 6.1 (System Platform 6.0.2.0.5, Template 6.1.4.0)
Avaya 9620 Phone (H.323)	3.1.1
Avaya 9620 Phone (SIP)	2.6.1
Avaya 9621 Phone (H.323)	s9621_41HAL_R6_0r58_V4r52
Avaya 4621 Phone (H.323)	2.9.1
Avaya Desktop Video Device, A175, incorporating the Avaya Flare experience	1.0.0
one-X® Communicator (SIP)	6.0.1.16-SP1-25226
one-X® Communicator (H.323)	6.0.1.16-SP1-25226
Digital Phone 2420	N/A
BT SIP Trunk Service	2.1.0.8

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 BT 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 S8800 Server and Avaya G450 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 10
      Maximum Concurrently Registered IP Stations: 18000 4
      Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
      Maximum Concurrently Registered IP eCons: 113 0
      Max Concur Registered Unauthenticated H.323 Stations: 100 0
      Maximum Video Capable Stations: 18000 0
      Maximum Video Capable IP Softphones: 0 0
      Maximum Administered SIP Trunks: 24000 24
      Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
```

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

```
isplay 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? y                                         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? y
    IP Trunks? y

IP Attendant Consoles? y
```

5.2. Administer IP Node Names

The node names defined here will be used in other configuration screens to define a SIP signaling group between Communication Manager and Session Manager. Use the **change node-name ip** command and assign the node **Name** and **IP Address** for the Session Manager. In this case, **rom_sm6** and **192.168.1.18** are the **Name** and **IP Address** for the Session Manager. Also note the **procr** name as this is the processor 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         192.168.1.4
rom_sm6      192.168.1.18
default        0.0.0.0
```

5.3. Administer IP Network Region

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

- The **Authoritative Domain** field is configured to match the domain name configured on Session Manager (**Section 6.2**). In this configuration, the domain name is **rom2.bt.com**.
- By default, **IP-IP Direct Audio** (both **Intra-** and **Inter-Region**) is enabled to allow audio traffic to be sent directly between endpoints without using gateway VoIP resources. When a PSTN call is shuffled the enterprise endpoint will talk directly to the public interface of the BT Session Border Controller.
- The **Codec Set** is set to the number of the IP codec set to be used for calls within the IP network region. In this case, codec set **1** is used which is configured in **Section 5.4**.

```
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: 1      Inter-region IP-IP Direct Audio: yes
      UDP Port Min: 2048      IP Audio Hairpinning? n
      UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
Call Control PHB Value: 46
      Audio PHB Value: 46
      Video PHB Value: 26
802.1P/Q PARAMETERS
Call Control 802.1p Priority: 6
      Audio 802.1p Priority: 6
      Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS      RSVP Enabled? n
H.323 Link Bounce Recovery? y
Idle Traffic Interval (sec): 20
Keep-Alive Interval (sec): 5
Keep-Alive Count: 5
```

5.4. Administer IP Codec Set

Enter the **change ip-codec-set x** command to configure the codec set specified in the **IP Network Region** form and enter the list of audio codecs to be used in order of preference. For the interoperability test, the codecs supported by BT were configured, namely G.729 and G.711A. During compliance testing, other codec set configurations were also verified.

```
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.729      n            2          20
2: G.711A     n            2          20
3: G.723     n            2          20
```

BT SIP Trunk Service supports the T.38 fax protocol. 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

This signaling group (and trunk group) will be used for inbound and outbound PSTN calls to BT SIP Trunk Service and will be configured using TLS (Transport Layer Security) and the default TLS port of 5061. Configure the **Signaling Group** using the **add signaling-group x** command as follows:

- Set the **Group Type** field to **sip**.
- The **Transport Method** field is set to **tls** (Transport Layer Security).
- The **Peer Detection Enabled** field should be set to **y** allowing the Communication Manager to automatically detect if the peer server is a Session Manager.
- Set the **Near-end Node Name** to the processor interface (node name **procr**) as shown in **Section 5.2**.
- Set the **Far-end Node Name** to the node name defined for the Session Manager (node name **rom_sm6**), as shown in **Section 5.2**.
- Ensure that the recommended TLS port value of **5061** is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- In the **Far-end Network Region** field, enter the IP Network Region configured in **Section 5.3**. This field logically establishes the **far-end** for calls using this signaling group as network region 1.
- Set the **Far-end Domain** field to the domain of the enterprise.
- 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.

Default values were used for other fields.

```
add signaling-group 4                               Page 1 of 1
                                                    SIGNALING GROUP
Group Number: 4                                   Group Type: sip
IMS Enabled? n                                   Transport Method: tls
  Q-SIP? n                                       SIP Enabled LSP? n
  IP 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: 5061                       Far-end Listen Port: 5061
                                                    Far-end Network Region: 1

Far-end Domain: rom2.bt.com

Incoming Dialog Loopbacks: eliminate             Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload                       RFC 3389 Comfort Noise? n
Session Establishment Timer(min): 5              Direct IP-IP Audio Connections? y
  Enable Layer 3 Test? y                         IP Audio Hairpinning? n
H.323 Station Outgoing Direct Media? y         Initial IP-IP Direct Media? n
                                                    Alternate Route Timer(sec): 6
```

5.6. Administer SIP Trunk Group

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

- Set the **Group Type** field to **sip**.
- Choose a descriptive **Group Name**.
- Specify a trunk access code (**TAC**) consistent with the dial plan.
- The **Direction** is set to **two-way** to allow incoming and outgoing calls.
- Set the **Service Type** field to **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. Also note that the value for **Redirect On OPTIM Failure** was set to **8000** to allow additional set-up time for calls destined for an EC500 destination.

```
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**.

```

add trunk-group 4                                     Page 3 of 21
TRUNK FEATURES
  ACA Assignment? n                                Measured: both
                                                    Maintenance Tests? y

  Numbering Format: public
  UUI Treatment: service-provider
  Replace Restricted Numbers? n
  Replace Unavailable Numbers? n
  
```

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 **101**, the value preferred by BT.

```

add display trunk-group 4                             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

  Convert 180 to 183 for Early Media? n
  Always Use re-INVITE for Display Updates? y
  Identity for Calling Party Display: P-Asserted-Identity
  Enable Q-SIP? n
  
```

5.7. Administer Calling Party Number Information

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 **39** will send the calling party number **4420711111x** to BT SIP Trunk Service, where **x** is the last digit of the 4-digit extension. This calling party number will be sent in the SIP From, Contact and PAI headers, and displayed on display-equipped PSTN telephones.

```

change public-unknown-numbering 0                     Page 1 of 2
  NUMBERING - PUBLIC/UNKNOWN FORMAT
  Total
  Ext Ext      Trk   CPN      Total
  Len Code    Grp(s) Prefix   CPN
  Len Code    Grp(s) Prefix   Len
  4 39      4    4420711111 12
  Total Administered: 1
  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 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 10
                                FEATURE ACCESS CODE (FAC)
    Abbreviated Dialing List1 Access Code:
    Abbreviated Dialing List2 Access Code:
    Abbreviated Dialing List3 Access Code:
    Abbreviated Dial - Prgm Group List Access Code:
    Announcement Access Code:
    Answer Back Access Code:
    Auto Alternate Routing (AAR) Access Code: 7
    Auto Route Selection (ARS) - Access Code 1: 9      Access Code 2:
```

Use the **change ars analysis** command to configure the routing of dialed digits following the first digit 9. A small sample of dial patterns is 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 **0207** or **0208**. Calls are sent to **Route Pattern 5**.

```
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
0207        4   11     5       pubu     n        n
0208        4   11     5       pubu     n        n
```

Use the **change route-pattern x** command to add the SIP trunk group to the route pattern that ARS selects. In this configuration, route pattern **5** is used to route calls to trunk group **4**.

```
change route-pattern 5                                       Page 1 of 3
                                Pattern Number: 5           Pattern Name: sip trk to SM6
                                SCCAN? n                   Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted      DCS/  IXC
    No   Mrk Lmt List Del  Digits  Dgts      QSIG
    1:  4   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      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 BT 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 BT correlate to the internal extensions assigned within Communication Manager. The entries displayed below translates incoming DID numbers 0207111111 and 0208111111 to a 4 digit extension by deleting all of the incoming digits and inserting an extension.

```
change inc-call-handling-trmt trunk-group 1 Page 1 of 3
      INCOMING CALL HANDLING TREATMENT
Service/      Number  Number  Del Insert
Feature       Len    Digits
public-ntwrk  11    0207111111  all  3936
public-ntwrk  11    0208111111  all  3934
```

5.10. EC500 Configuration

When EC500 is enabled on the Communication Manager station, a call to that station will generate a new outbound call from Communication Manager to the configured EC500 destination, typically a mobile phone. The following screen shows an example EC500 configuration for the user with station extension 3910. Use the command **change off-pbx-telephone station mapping x**, where x is the Communication Manager station.

- The **Station Extension** field will automatically populate.
- For **Application** enter **EC500**.
- Enter a **Dial Prefix** (e.g., 9) if required by the routing configuration.
- For the **Phone Number** field enter the phone that will also be called (e.g., **0788011111**).
- Set the **Trunk Selection** to **ars** so that the ARS tables will be used to determine how Communication Manager will route to the Phone Number destination.
- Set the **Config Set** to **1**.
- Other parameters can retain default value.

```
change off-pbx-telephone station-mapping 3910 Page 1 of 3
      STATIONS WITH OFF-PBX TELEPHONE INTEGRATION
Station      Application  Dial  CC  Phone Number  Trunk  Config  Dual
Extension    EC500       Prefix  -   0788011111   Selection  Set    Mode
3910         EC500       -       -   0788011111   ars       1      -
```

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

6. Configuring Avaya Aura® Session Manager

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

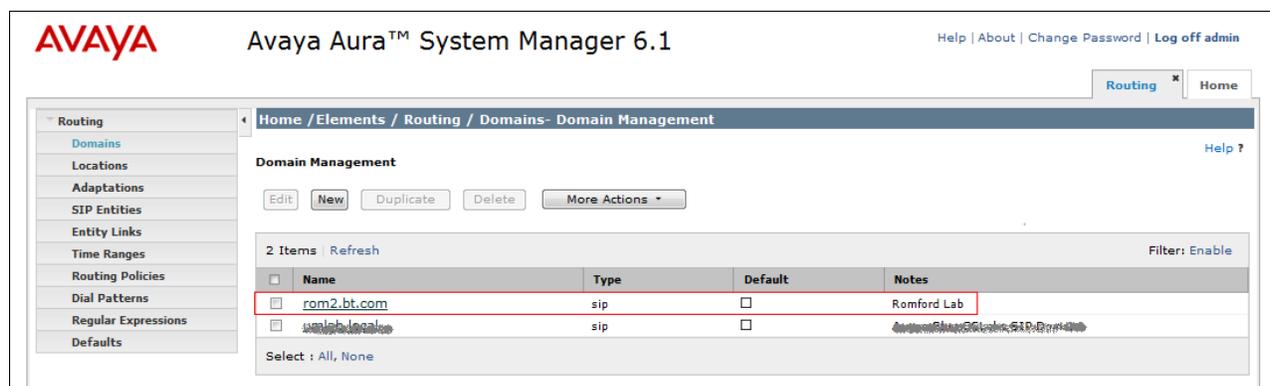
- Log in to Avaya Aura® System Manager
- Administer SIP domain
- Administer Adaptations
- Administer SIP Entities
- Administer Entity Links
- Administer Routing Policies
- Administer Dial Patterns
- Administer Avaya Aura® Communication Manager as Managed Element
- Administer Application for Avaya Aura® 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 **SIP 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., **rom2.bt.com**) and optionally a description for the domain in the **Notes** field. Click **Commit** (not shown) to save changes.



The screenshot displays the Avaya Aura System Manager 6.1 web interface. The top navigation bar includes the Avaya logo, the title "Avaya Aura™ System Manager 6.1", and links for "Help | About | Change Password | Log off admin". The main content area is titled "Domain Management" and features a left-hand navigation menu with options like "Routing", "Domains", "Locations", "Adaptations", "SIP Entities", "Entity Links", "Time Ranges", "Routing Policies", "Dial Patterns", "Regular Expressions", and "Defaults". The "Routing" menu is expanded, showing "Domains" as the selected option. The "Domain Management" section includes buttons for "Edit", "New", "Duplicate", "Delete", and "More Actions". Below these buttons is a table with 2 items, showing a list of SIP domains. The first row is highlighted with a red border and contains the following data:

Name	Type	Default	Notes
rom2.bt.com	sip	<input type="checkbox"/>	Romford Lab
www.bt.com	sip	<input type="checkbox"/>	www.bt.com SIP Domain

At the bottom of the table, there is a "Select" dropdown menu set to "All, None".

6.3. Administer Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside, for the purposes of bandwidth management. One location is added to the sample configuration for 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.

The screenshot displays the Avaya Aura System Manager 6.1 interface. The left-hand navigation menu includes options such as Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Home / Elements / Routing / Locations- Location Details'. Under the 'General' section, the 'Name' field is set to 'Romford Avaya Lab'. The 'Overall Managed Bandwidth' section shows 'Managed Bandwidth Units' as 'Mbit/sec' and 'Total Bandwidth' as '1000'. The 'Per-Call Bandwidth Parameters' section shows 'Default Audio Bandwidth' as '80 Kbit/sec'. The 'Location Pattern' section contains a table with one entry:

IP Address Pattern	Notes
192.168.1.	Romford Avaya Lab

The 'Add' and 'Remove' buttons are visible above the table. The 'Filter: Enable' option is also present.

6.4. Administer Adaptations

In order to ensure that the E.164 numbering format is used between the enterprise and BT SIP Trunk Service, an adaptation module is used to perform some digit manipulation. This adaptation is applied to the Communication Manager SIP entity. To add an adaptation, on the **Routing** tab select **Adaptations** on the left hand menu and then click on the **New** button (not shown). Under **General**:

- In the **Adaptation Name** field enter an informative name.
- In the **Module Name** field select <click to add module> from the drop down list and enter **DigitConversionAdapter** in the resulting **New Module Name** field (not shown).

AVAYA Avaya Aura™ System Manager 6.1 Help | About | Change Password | Log off admin

Routing * Home

Home / Elements / Routing / Adaptations- Adaptation Details

Adaptation Details

General

* Adaptation name: Romford CM i/c and o/g PSTN

Module name: DigitConversionAdapter

Module parameter:

Egress URI Parameters:

Notes: For calls into and out of the Romf

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

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

This will ensure any destination numbers received from Communication Manager are converted to the E.164 numbering format before being processed by Session Manager

Digit Conversion for Incoming Calls to SM

Add Remove

5 Items Refresh Filter: Enable

<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Notes
<input type="checkbox"/>	*0207	*4	*36		*1	++44	destination	converts 0207 to ++44207 for N
<input type="checkbox"/>	*0208	*4	*36		*1	++44	destination	converts 0208 to ++44208 for N
<input type="checkbox"/>	*07	*2	*13		*1	++44	destination	converts 07 numbers to ++447 fo

Select : All, None

Under **Digit Conversion for Outgoing Calls from SM** click the **Add** button and specify the digit manipulation to be performed as follows:

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

This will ensure any destination numbers will have the + symbol and international dialing code removed before being presented to Communication Manager.

Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Notes
*+44207	*6	*36		*3	0	destination	converts +44207 to 0207 for Cj
*+44208	*6	*36		*3	0	destination	converts +44208 to 0208 for Cj

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 **Other** for the BT SIP Trunk Service SIP entities.
- 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 types of SIP Entity.

- Session Manager SIP Entity
- Communication Manager SIP Entity
- BT SIP Trunk Service SIP Entities

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 screenshot shows the Avaya Aura System Manager 6.1 interface. The breadcrumb navigation is Home / Elements / Routing / SIP Entities - SIP Entity Details. The left sidebar lists various configuration options, with SIP Entities selected. The main area displays the 'SIP Entity Details' for 'Romford SM 6.1'. The 'General' tab is active, showing fields for Name, FQDN or IP Address, Type, Notes, Location, Outbound Proxy, Time Zone, and Credential name. The 'SIP Link Monitoring' section is set to 'Use Session Manager Configuration'. The fields for Name, FQDN or IP Address, Location, and Time Zone are highlighted with red boxes.

Field	Value
Name	Romford SM 6.1
FQDN or IP Address	192.168.1.18
Type	Session Manager
Notes	
Location	Romford Avaya Lab
Outbound Proxy	
Time Zone	Europe/London
Credential name	

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.

The screenshot shows the 'Port' configuration section. It includes 'Add' and 'Remove' buttons, a '3 Items Refresh' indicator, and a 'Filter: Enable' option. A table lists three configured ports. The table has columns for Port, Protocol, Default Domain, and Notes. The first two rows are highlighted with a red box.

Port	Protocol	Default Domain	Notes
5060	TCP	rom2.bt.com	
5060	UDP	rom2.bt.com	
5061	TLS	rom2.bt.com	

6.5.2. Avaya Aura® Communication Manager SIP Entity

The following screens show the SIP entity for Communication Manager which is configured as an Evolution Server. The **FQDN or IP Address** field is set to the IP address of the interface on Communication Manager that will be providing SIP signaling. For the **Adaptation** field, select the adaptation module previously defined for dial plan digit manipulation in **Section 6.4**.

The screenshot displays the Avaya Aura System Manager 6.1 interface. The top navigation bar includes the Avaya logo, the title "Avaya Aura™ System Manager 6.1", and links for "Help | About | Change Password | Log off admin". Below the navigation bar, there are tabs for "Routing", "Session Manager", and "Home". The main content area is titled "SIP Entity Details" and is currently on the "General" tab. A left-hand sidebar lists various configuration categories: Routing, Domains, Locations, Adaptations, SIP Entities (highlighted), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The "SIP Entity Details" section contains several fields: "Name" (Romford CM6.1), "FQDN or IP Address" (192.168.1.4), "Type" (CM), and "Notes" (PE address). Below these, there are three dropdown menus: "Adaptation" (Romford CM i/c and o/g PSTN), "Location" (Romford Avaya Lab), and "Time Zone" (Europe/London). Further down, there is a checkbox for "Override Port & Transport with DNS SRV" (unchecked), a field for "SIP Timer B/F (in seconds)" (4), a "Credential name" field, and a "Call Detail Recording" dropdown (none). At the bottom, there is a "SIP Link Monitoring" section with a dropdown menu set to "Use Session Manager Configuration".

6.5.3. BT SIP Trunk Service SIP Entities

BT has multiple SBCs in their network as illustrated in **Figure 1**. Each SBC must be added to Session Manager as a SIP entity. As an example the following screen shows the SIP Entity for BT SBC1. The **FQDN or IP Address** field is set to the IP address of the SBC1 (see **Figure 1**).

AVAYA Avaya Aura™ System Manager 6.1

Routing / Home / Elements / Routing / SIP Entities- SIP Entity Details

SIP Entity Details

General

* Name: NOAS SBC Birm2

* FQDN or IP Address: 192.168.5.62

Type: Other

Notes: Primary SIP inbound / outbound

Adaptation:

Location: NOAS SIP Service

Time Zone: Europe/London

Override Port & Transport with DNS SRV:

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

Credential name:

Call Detail Recording: none

SIP Link Monitoring: Use Session Manager Configuration

The following screen shows all 4 BT SBC SIP Entities after they have been configured.

AVAYA Avaya Aura™ System Manager 6.1

Routing / Session Manager / Home / Elements / Routing / SIP Entities - SIP Entities

SIP Entities

34 Items | Refresh Filter: Enable

Name	FQDN or IP Address	Type	Notes
NOAS_SBC_Birm1	192.168.5.58	Other	Second choice inbound/outbound calls
NOAS_SBC_Birm2	192.168.5.62	Other	Primary SIP inbound / outbound calls
NOAS_SBC_Man1	192.168.5.50	Other	fourth choice inbound / outbound calls
NOAS_SBC_Man2	192.168.5.54	Other	third choice inbound / outbound calls
Romford_CM6_1	192.168.1.4	CM	PE address
Romford_SM_6_1	192.168.1.18	Session Manager	

Select : All, None Page 2 of 3

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. In the resulting screen, fill in the following fields displayed in the new row.

- In the **Name** field enter an informative name.
- In the **SIP Entity 1** field select **Romford SM 6.1**.
- In the **Port** field enter the port number to which the other system sends its SIP requests.
- In the **SIP Entity 2** field enter the other SIP Entity for this link, created in **Section 6.5**.
- In the **Port** field enter the port number to which the other system expects to receive SIP requests.
- Select the **Trusted** tick box to make the other system trusted.
- In the **Protocol** field enter the transport protocol to be used to send SIP requests.

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

The screenshot shows the Avaya Aura System Manager 6.1 interface. The left sidebar contains a navigation menu with options like Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Entity Links' and includes a table with 37 items. The table has the following columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, Trusted, and Notes. A red box highlights the following rows in the table:

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
Romford SM 6.1 to Romford CM 6.1	Romford SM 6.1	TLS	5061	Romford CM6.1	5061	✓	
Romford SM 6.1 NOAS SBC Birm1	Romford SM 6.1	UDP	5060	NOAS SBC Birm1	5060	✓	
RomSM6.1toBirm2	Romford SM 6.1	UDP	5060	NOAS SBC Birm2	5060	✓	
RomSM6.1toMan1	Romford SM 6.1	UDP	5060	NOAS SBC Man1	5060	✓	
RomSM6.1toMan2	Romford SM 6.1	UDP	5060	NOAS SBC Man2	5060	✓	

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 in the resulting window (not shown) select the appropriate SIP entity to which this routing policy applies.
- Under **Time of Day**, click **Add**, and then select the time range.

The following screen shows the routing policy for Communication Manager

The screenshot displays the Avaya Aura System Manager 6.1 interface. The breadcrumb trail is 'Home / Elements / Routing / Routing Policies - Routing Policy Details'. The left sidebar shows 'Routing Policies' selected. The main content area is titled 'Routing Policy Details' and includes a 'Help ?' link, 'Commit', and 'Cancel' buttons.

General

* Name: NOAS calls to Rom CM 6.1

Disabled:

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
Romford CM6.1	192.168.1.4	CM	PE address

Time of Day

Add Remove View Gaps/Overlaps

1 Item Refresh Filter: Enable

<input type="checkbox"/>	Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="checkbox"/>	4	24/7	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7						

Select : All, None

As an example, the following screen shows the routing policy for BT SBC1, a routing policy must be added for each SBC. Note the **Ranking** given to the time range in this routing policy is set to **10**. Each SBC routing policy will have a different ranking; this is to define a priority order for the routing policies when they are added to a dial pattern in **Section 6.8**. The rankings are set in blocks of ten for clarity in this Application Note.

The screenshot shows the Avaya Aura System Manager 6.1 interface. The breadcrumb trail is: Home / Elements / Routing / Routing Policies - Routing Policy Details. The page title is "Routing Policy Details".

General

- Name: SIP Trunk Calls to Birm2
- Disabled:
- Notes: a

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
NOAS SBC Birm2	192.168.5.62	Other	Primary SIP inbound / outbound calls

Time of Day

Add Remove View Gaps/Overlaps

1 Item Refresh Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
10	24/7	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7						

Select : All, None

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 the domain configured in **Section 6.2**.

Under **Originating Locations and Routing Policies**, click **Add** in the resulting screen (not shown). Under **Originating Location** select **ALL** and under **Routing Policies** select one of the routing policies defined in **Section 6.7**. Click **Select** button to save. The following screen shows an example dial pattern configured for BT SIP Trunk Service. Note the ranking for each routing policy as applied in **Section 6.7**. The routing policy with the lowest rank will be selected first, if this route is unavailable or does not respond then the routing policy with the next lowest rank will be selected and so on. This allows for redundant routing within Session Manager.

The screenshot displays the Avaya Aura System Manager 6.1 interface for configuring a dial pattern. The top navigation bar shows 'Home / Elements / Routing / Dial Patterns - Dial Pattern Details'. The left sidebar contains a menu with 'Dial Patterns' highlighted. The main content area is titled 'Dial Pattern Details' and includes a 'General' section with the following fields:

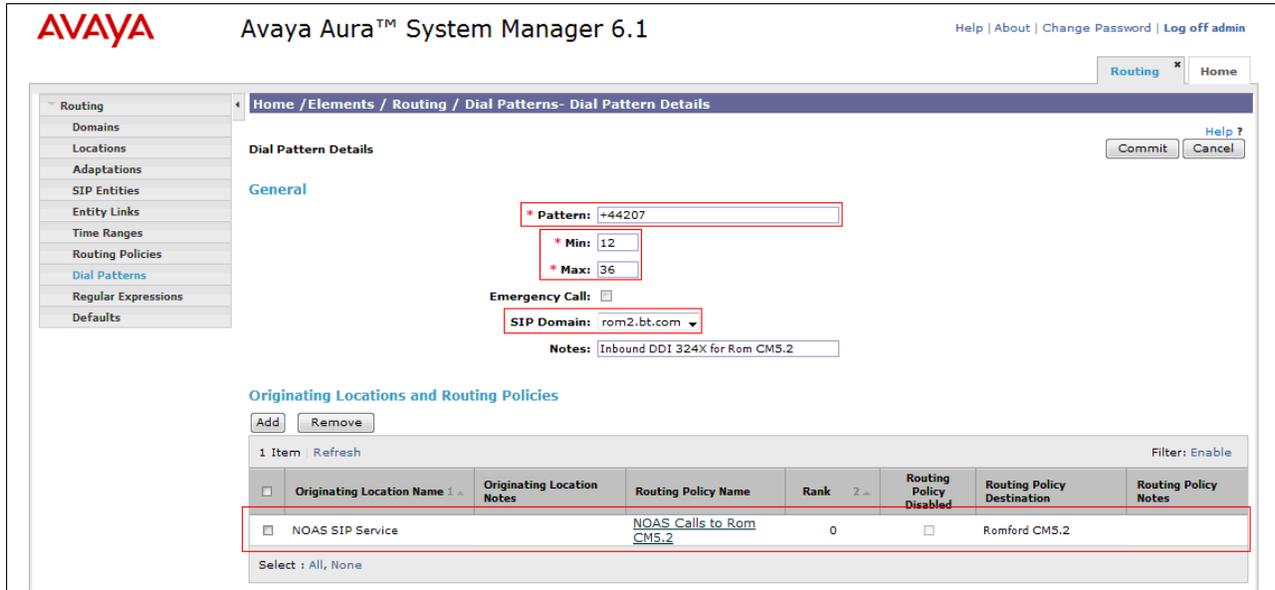
- Pattern:** +447
- Min:** 4
- Max:** 14
- Emergency Call:**
- SIP Domain:** -ALL-
- Notes:** Mobile UK via SIP

Below the form is the 'Originating Locations and Routing Policies' section, which includes an 'Add' button and a table with 5 items. The table has the following columns: Originating Location Name, Originating Location Notes, Routing Policy Name, Rank, Routing Policy Disabled, Routing Policy Destination, and Routing Policy Notes.

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-	Any Locations	SIP Trunk Calls to Birm2	10	<input type="checkbox"/>	NOAS SBC Birm2	a
<input type="checkbox"/>	-ALL-	Any Locations	SIP Trunk calls to Birm1	20	<input type="checkbox"/>	NOAS SBC Birm1	b
<input type="checkbox"/>	-ALL-	Any Locations	SIP Trunk calls to Man2	30	<input type="checkbox"/>	NOAS SBC Man2	c
<input type="checkbox"/>	-ALL-	Any Locations	SIP Trunk calls to Man1	40	<input type="checkbox"/>	NOAS SBC Man1	d

At the bottom of the table, there is a 'Select' dropdown menu set to 'All, None'.

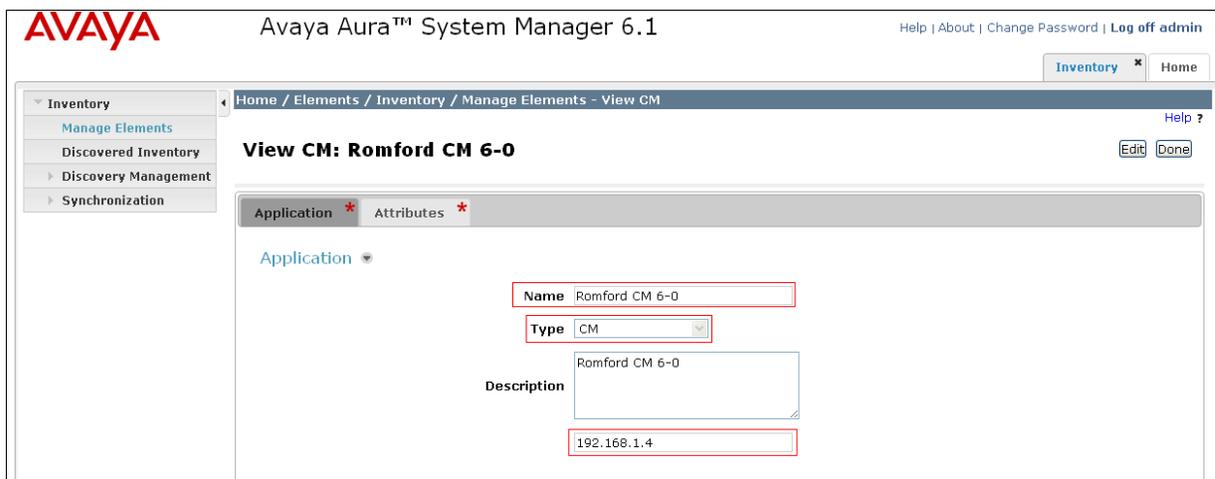
The following screen shows an example dial pattern configured for the Communication Manager.



6.9. Administer Avaya Aura® Communication Manager as a Managed Element

From the Home tab select **Inventory** from the menu. In the resulting tab from the left panel menu select **Manage Elements** and click **New** (not shown). On the **Application** tab, enter values in the following fields and use defaults for the remaining fields:

- In the **Name** field enter a descriptive name.
- In the **Type** field select CM from the drop-down menu.
- In the **Node** enter the IP address of the Communication Manager.



On the **Attributes** tab, under the **Attributes** heading, enter values in the following fields and use defaults for the remaining fields:

- In the **Login** field enter a login name for Communication Manager (SAT SSH login).
- In the **Password** field enter the password for Communication Manager (SAT SSH password).
- Select the **Is SSH Connection** check box if SSH is to be used.
- In the **Port** field enter the port number to use for SAT access.

Select **Commit** (not shown), to synchronize System Manager with the Communication Manager in the background.

The screenshot shows the Avaya Aura System Manager 6.1 interface. The top navigation bar includes the Avaya logo, the title 'Avaya Aura™ System Manager 6.1', and links for 'Help | About | Change Password | Log off admin'. Below the navigation bar, there are tabs for 'Inventory' and 'Home'. The main content area is titled 'View CM: Romford CM 6-0' and includes a breadcrumb trail 'Home / Elements / Inventory / Manage Elements - View CM'. A sidebar on the left contains a tree view with 'Inventory' expanded, showing 'Manage Elements', 'Discovered Inventory', 'Discovery Management', and 'Synchronization'. The main content area has two tabs: 'Application' and 'Attributes', with 'Attributes' selected. Under the 'Attributes' tab, there are sections for 'SNMP Attributes' and 'Attributes'. The 'Attributes' section contains a 'Version' field with radio buttons for 'None', 'V1', and 'V3'. Below this is a red-bordered box containing the following fields: 'Login' (text input with value 'romsystemngr'), 'Password' (password input with masked characters), 'Is SSH Connection' (checkbox checked), and 'Port' (text input with value '5022'). Below the red box is an 'Alternate IP Address' field.

6.10. Administer Application for Communication Manager

From the Home tab select Session Manager from the menu. In the resulting tab from the left panel menu select **Application Configuration** → **Applications** and click **New** (not shown) and configure as follows:

- In the **Name** field enter a name for the application.
- In the **SIP Entity** field select the SIP entity for the Communication Manager configured in **Section 6.5.2**.
- In the **CM System for SIP Entity** field select the managed element for the Communication Manager configured in **Section 6.9**.

Select **Commit** to save the configuration.

The screenshot displays the Avaya Aura System Manager 6.1 interface. The top navigation bar includes the Avaya logo, the title 'Avaya Aura™ System Manager 6.1', and user options like 'Help', 'About', 'Change Password', and 'Log off admin'. The breadcrumb trail shows the path: Home / Elements / Session Manager / Application Configuration / Applications - Applications. The left sidebar menu is expanded to 'Applications'. The main content area is titled 'Application Editor' and contains a form for creating a new application. The form fields are: Name (non-IMS SIP users), SIP Entity (Romford CM6.1), CM System for SIP Entity (Romford CM 6-0), and Description (Romford CM 6-0). A red box highlights the Name, SIP Entity, and CM System for SIP Entity fields. The interface also includes a 'Commit' button and a 'Cancel' button.

6.11. Administer Application Sequence for Communication Manager

From the left panel navigate to **Session Manager** → **Application Configuration** → **Application Sequences** and click on **New** (not shown).

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

Select **Commit**.

The screenshot shows the Avaya Aura System Manager 6.1 interface. The left sidebar contains a navigation menu with categories like Session Manager, Network Configuration, and Application Configuration. The main area is titled 'Application Sequence Editor' and contains the following elements:

- Application Sequence** section: A form with a ***Name** field containing 'non-IMS SIP users application sq' and an empty **Description** field.
- Applications in this Sequence** section: A table with one item, 'non-IMS SIP users', which is highlighted with a red box. The table has columns for Sequence Order, Name, SIP Entity, Mandatory, and Description.
- Available Applications** section: A table with one item, 'non-IMS SIP users', which is highlighted with a red box. The table has columns for Name, SIP Entity, and Description.

Sequence Order (first to last)	Name	SIP Entity	Mandatory	Description
<input type="checkbox"/>	non-IMS SIP users	Romford CM6.1	<input checked="" type="checkbox"/>	Romford CM 6-0

Name	SIP Entity	Description
non-IMS SIP users	Romford CM6.1	Romford CM 6-0

6.12. Administer SIP Extensions

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

On the **Identity** tab:

- Enter the user's name in the **Last Name** and **First Name** fields.
- In the **Login Name** field enter a unique system login name in the form of user@domain (e.g. **3936@rom2.bt.com**) which is used to create the user's primary handle.
- The **Authentication Type** should be **Basic**.
- In the **Password/Confirm Password** fields enter an alphanumeric password.

The screenshot shows the Avaya Aura System Manager 6.1 interface. The page title is "Avaya Aura™ System Manager 6.1" with navigation links for "Help", "About", "Change Password", and "Log off admin". The breadcrumb trail is "Home / Users / User Management / Manage Users - New User Profile". The main content area is titled "New User Profile" and has tabs for "Identity", "Communication Profile", "Membership", and "Contacts". The "Identity" tab is active. The form fields are as follows:

- * Last Name: Bloggs
- * First Name: Joe
- Middle Name: (empty)
- Description: (empty)
- * Login Name: 3936@rom2.bt.com
- * Authentication Type: Basic (dropdown)
- * Password: (masked with dots)
- * Confirm Password: (masked with dots)
- Localized Display Name: (empty)
- Endpoint Display Name: (empty)

Buttons for "Commit" and "Cancel" are visible in the top right of the form area.

On the **Communication Profile** tab enter a numeric **Communication Profile Password** and confirm it, then click on the show/hide button for **Communication Address** and click **New**. For the **Type** field select **Avaya SIP** from the drop-down menu. In the **Fully Qualified Address** field, enter an extension number and select the relevant domain from the drop-down menu. Click the **Add** button.

Identity * Communication Profile * Membership Contacts

Communication Profile ▾

Communication Profile Password: ●●●●●●

Confirm Password: ●●●●●●

New Delete Done Cancel

Name
Primary

Select : None

* Name: Primary

Default :

Communication Address ▾

New Edit Delete

Type	Handle	Domain
No Records found		

Type: Avaya SIP ▾

* Fully Qualified Address: 3936 @ rom2.bt.com ▾

Add Cancel

Click the show/hide button next to **Session Manager Profile**:

- Make sure the **Session Manager** check box is checked.
- Select the appropriate Session Manager instance from the drop-down menu in the **Primary Session Manager** field.
- Select the appropriate application sequence from the drop-down menu in the **Origination Application Sequence** field configured in **Section 6.11**.
- Select the appropriate application sequence from the drop-down menu in the **Termination Application Sequence** field configured in **Section 6.11**.
- Select the appropriate location from the drop-down menu in the **Home Location** field as configured in **Section 6.3**.

Session Manager Profile

* Primary Session Manager Romford SM 6.1

Primary	Secondary	Maximum
5	0	5

Secondary Session Manager (None)

Primary	Secondary	Maximum

Origination Application Sequence non-IMS SIP users application sq

Termination Application Sequence non-IMS SIP users application sq

Survivability Server (None)

* Home Location Romford Avaya Lab

Click the show/hide button next to **Endpoint Profile**.

- Select the Communication Manager SIP Entity from the **System** drop-down menu.
- Select **Endpoint** from the drop-down menu for **Profile Type**.
- Enter the extension in the **Extension** field.
- Select the desired template from the **Template** drop-down menu.
- For the **Port** field select **IP**.
- Select the **Delete Endpoint on Unassign of Endpoint from User or on Delete User** check box.
- Select **Commit** to save changes and the System Manager will add the Communication Manager user configuration automatically.

Endpoint Profile ▾

* System Romford CM 6-0 ▾

* Profile Type Endpoint ▾

Use Existing Endpoints

* Extension 3936 Endpoint Editor

* Template DEFAULT_9620SIP_CM_6_0 ▾

Set Type 9620SIP

Security Code ●●●●●●

* Port IP

Voice Mail Number

Delete Endpoint on Unassign of Endpoint from User or on Delete User.

Messaging Profile ▾

*Required

Commit Cancel

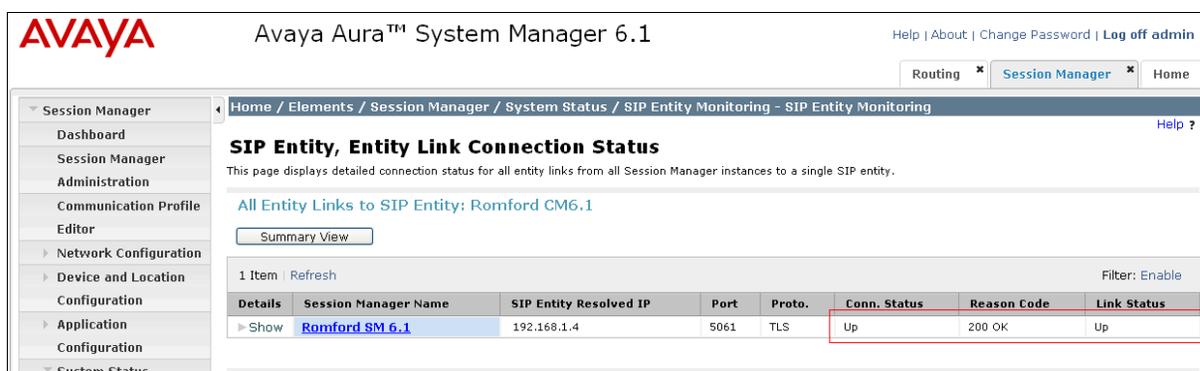
7. Service Provider Configuration

The configuration of the BT equipment used to support the BT SIP trunk service is outside of the scope for these application notes and will not be covered. To obtain further information on BT equipment and system configuration please contact an authorised BT representative.

8. Verification Steps

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

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



The screenshot shows the Avaya Aura System Manager 6.1 interface. The breadcrumb trail is: Home / Elements / Session Manager / System Status / SIP Entity Monitoring - SIP Entity Monitoring. The page title is "SIP Entity, Entity Link Connection Status". Below the title, it says "This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity." The main content area shows "All Entity Links to SIP Entity: Romford CM6.1" and a "Summary View" button. Below that, there is a table with 1 item. The table has columns: Details, Session Manager Name, SIP Entity Resolved IP, Port, Proto., Conn. Status, Reason Code, and Link Status. The row shows: Show, Romford SM 6.1, 192.168.1.4, 5061, TLS, Up, 200 OK, Up. The "Up" status in the Conn. Status and Link Status columns is highlighted with a red box.

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
Show	Romford SM 6.1	192.168.1.4	5061	TLS	Up	200 OK	Up

2. From the 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
          Port      Busy
0004/001  T00001    in-service/idle  no
0004/002  T00007    in-service/idle  no
0004/003  T00008    in-service/idle  no
0004/004  T00009    in-service/idle  no
0004/005  T00010    in-service/idle  no
```

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

9. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Communication Manager Evolution Server and Avaya Aura® Session Manager to BT SIP Trunk Service. BT SIP Trunk Service is a SIP-based Voice over IP solution providing businesses a flexible, cost-saving alternative to traditional hardwired telephony trunks.

10. 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, June 2010.
- [2] *Administering Avaya Aura® System Platform*, Release 6, June 2010.
- [3] *Administering Avaya Aura® Communication Manager*, August 2010, Document Number 03-300509.
- [4] *Avaya Aura® Communication Manager Feature Description and Implementation*, August 2010, Document Number 555-245-205.
- [5] *Installing and Upgrading Avaya Aura® System Manager Release 6.1*, November 2010.
- [6] *Installing and Configuring Avaya Aura® Session Manager*, April 2011, Document Number 03-603473
- [7] *Administering Avaya Aura® Session Manager*, May 2011, Document Number 03-603324.
- [8] RFC 3261 *SIP: Session Initiation Protocol*, <http://www.ietf.org/>

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