



## **Avaya Solution & Interoperability Test Lab**

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# **Application Notes for Configuring Avaya Aura® Communication Manager Access Element and Avaya Aura® Session Manager to support British Telecom SIP Trunk Service - 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. 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 SIP (BT) Trunk Service and an Avaya SIP enabled enterprise solution. The Avaya solution consists of Avaya Aura® Session Manager and Avaya Aura® Communication Manager Access Element. Customers using this Avaya SIP-enabled enterprise solution with the British Telecom 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 were terminated on H.323 and analog 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 H.323 and analog telephones.
- Calls were made using G.729A, and G.711A codec's.
- 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 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:

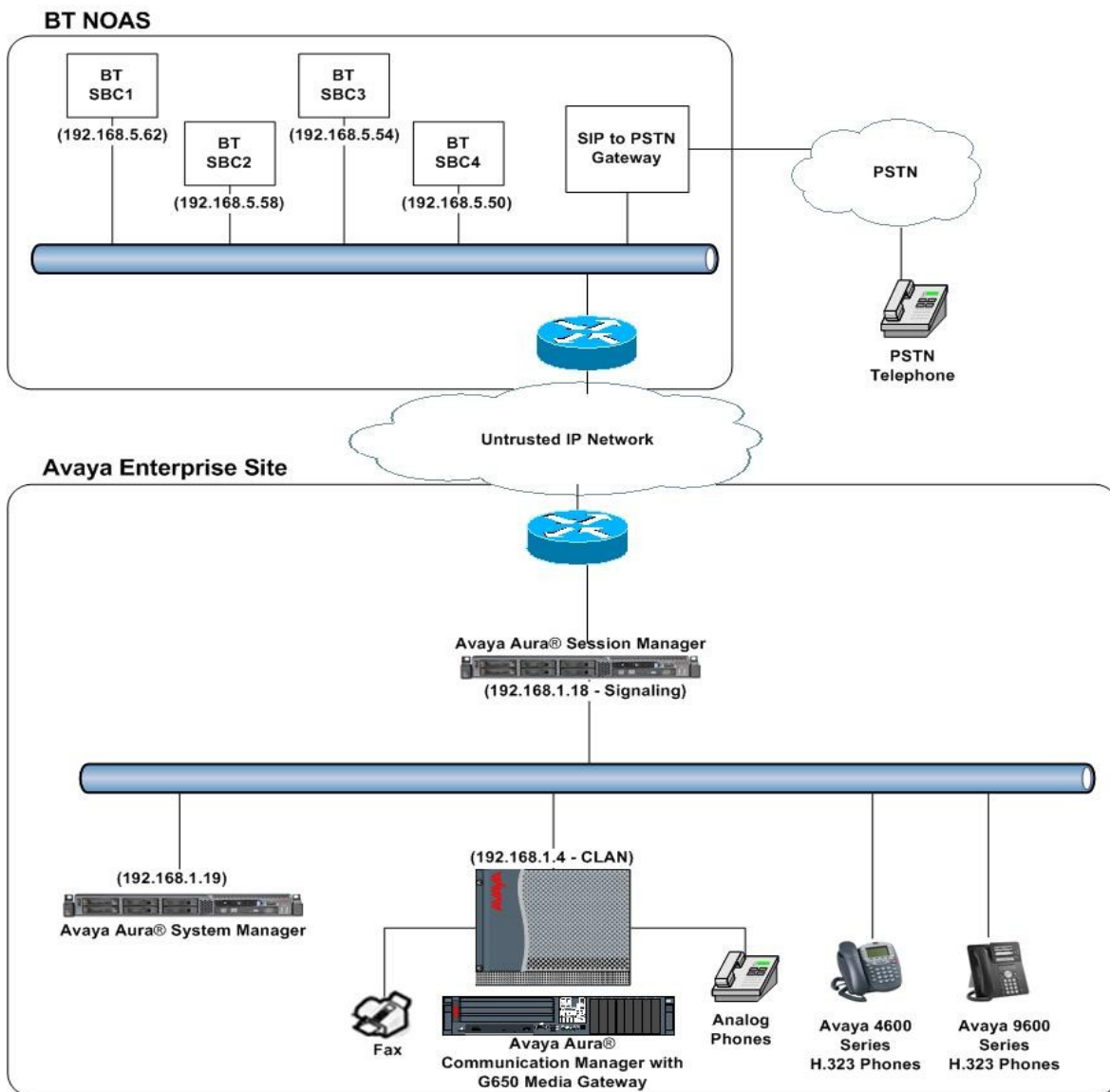
- The Caller-ID set at the enterprise is overridden by BT with a pre-configured number, if the number is withheld at the enterprise no number is presented to the called party.
- All tests were completed using H.323 or analog telephone types. No soft clients, SIP stations nor digital handsets were used throughout the testing.
- No inbound toll free numbers were tested, however routing of inbound DID numbers and the relevant number translation was successfully tested.
- Routing to emergency numbers (such as 999) was not tested.
- G729 annex b (silence suppression) is not supported by BT SIP Trunk Service and thus was not tested.
- G711mu is not supported by BT SIP Trunk Service and thus was not tested.

## 2.3. Support

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 9600 series IP telephones, Avaya 4600 series IP telephones (with H.323 firmware), an Analog Telephone and a 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. Instead, public IP addresses have been replaced with private addresses and all phone numbers have been replaced with arbitrary numbers that bear no relevance to the test configuration.



**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 S8730 Server with G650 Media Gateway	Avaya Aura® Communication Manager 5.2.1 (R015x.02.1.016.4) Service Pack 18475
Avaya G650 Media Gateway TN799DP CLAN TN2602AP MedPro	HW01 FW015 HW08 FW031
Avaya S8800 Server	Avaya Aura® Session Manager 6.1 (6.1.0.0.610023)
Avaya S8800 Server	Avaya Aura® System Manager 6.1
Avaya 9620 Phone (H.323)	3.11
Avaya 4621 Phone (H.323)	2.9.1
Analog Phone	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 G650 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 20
      Maximum Concurrently Registered IP Stations: 18000 8
      Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
      Maximum Concurrently Registered IP eCons: 2 0
      Max Concur Registered Unauthenticated H.323 Stations: 0 0
      Maximum Video Capable H.323 Stations: 0 0
      Maximum Video Capable IP Softphones: 0 0
      Maximum Administered SIP Trunks: 420 98
      Maximum Administered Ad-hoc Video Conferencing Ports: 0 0
```

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

```
display system-parameters customer-options                               Page 4 of 11
                                OPTIONAL FEATURES

Emergency Access to Attendant? y                                       IP Stations? y
  Enable 'dadmin' Login? y
  Enhanced Conferencing? y                                           ISDN Feature Plus? 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. In the **IP Node Names** form, assign the node **Name** and **IP Address** for the Session Manager. In this case, **romsm6sm100** and **192.168.1.18** are the **Name** and **IP Address** for the Session Manager. Also note the **cm5clan2** name as this is the he C-LAN interface that Communication Manager will use as the SIP signaling interface to Session Manager.

```
change node-names ip
                                IP NODE NAMES

Name          IP Address
cm5clan2    192.168.1.4
romsm6sm100 192.168.1.18
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 (**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 end point will talk directly to the outside 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** will be used (**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: Default NR
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
```

### 5.4. Administer IP Codec Set

Open the **IP Codec Set** form for the codec set specified in the **IP Network Region** form. 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. 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.711A      n              2          20
2: G.729      n              2          20
3: G.711MU    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
<b>FAX</b>	<b>t.38-standard</b>	<b>0</b>
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 TCP (Transport control Protocol) and the default 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** (Transport Control Protocol).
- Set the **Near-end Node Name** to the CLAN interface (node name **cm5clan2**). 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 **romsm6sm100**), shown in **Section 5.2**.
- Ensure that the recommended TCP port value of **5060** is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- In the **Far-end Network Region** field, enter the IP Network Region configured in **Section 5.3**. This field logically establishes the far-end network region for calls using this signaling group as network region 1.
- Set the **Far-end Domain** field to the domain of the enterprise.
- 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 **Direct IP-IP Audio Connections** field is set to **y**.
- The default values for the other fields may be used.

```
add signaling-group 35
                                SIGNALING GROUP

Group Number: 35                Group Type: sip
                                Transport Method: tcp

IMS Enabled? n

Near-end Node Name: cm5clan2    Far-end Node Name: romsm6sm100
Near-end Listen Port: 5060      Far-end Listen Port: 5060
                                Far-end Network Region: 1

Far-end Domain: rom2.bt.com

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

## 5.6. Administer SIP Trunk Group

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

- Set the **Group Type** field to **sip**.
- Choose any 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 configured in **Section 5.5**.
- Specify the **Number of Members** supported by this SIP trunk group.

```
add trunk-group 35                                     Page 1 of 21
                                     TRUNK GROUP
Group Number: 35                                     Group Type: sip           CDR Reports: y
  Group Name: asm6-sm100                             COR: 1                   TN: 1           TAC: 135
  Direction: two-way                                 Outgoing Display? n
Dial Access? n                                       Night Service:
Queue Length: 0
Service Type: tie                                     Auth Code? n
                                               Signaling Group: 35
                                               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 35                                     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 35                                     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
  
```

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 trunk-group 35                                     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

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 **38** will send the calling party number **02071111111** to BT 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.

```

change public-unknown-numbering 0                     Page 1 of 2
NUMBERING - PUBLIC/UNKNOWN FORMAT
Ext  Ext      Trk      CPN      Total
Len  Code     Grp(s)  Prefix   CPN
                                           Len
                                           Total Administered: 1
                                           Maximum Entries: 240
4   38       35       02071111111  11
  
```

## 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 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 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 35.

```

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      35      pubu      n
0208          4      11      35      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 **35** is used to route calls to trunk group **35**.

```

change route-pattern 35                                     Page 1 of 3
                Pattern Number: 35  Pattern Name: asm6-sm100
                SCCAN? n          Secure SIP? n
  Grp FRL NPA Pfx Hop Toll No.  Inserted          DCS/ IXC
  No          Mrk Lmt List Del  Digits          QSIG
                Dgts
1: 35    0
2:
3:
4:
5:
6:
                DCS/ IXC
                n    user
                n    user
                n    user
                n    user
                n    user
                n    user

  BCC VALUE TSC CA-TSC ITC BCIE Service/Feature PARM No. Numbering LAR
  0 1 2 M 4 W      Request      Dgts Format
                Subaddress
1: y y y y y n n          rest          none
2: y y y y y n n          rest          none

```

### 5.9. Administer Incoming Digit Translation

This step configures the settings necessary to map incoming DID calls to the proper Communication Manager extension(s). The incoming digits sent in the INVITE message from 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 02071111111 and 02081111111 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   12    02071111111  all  3802
public-ntwrk   12    02081111111  all  3803

```

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:

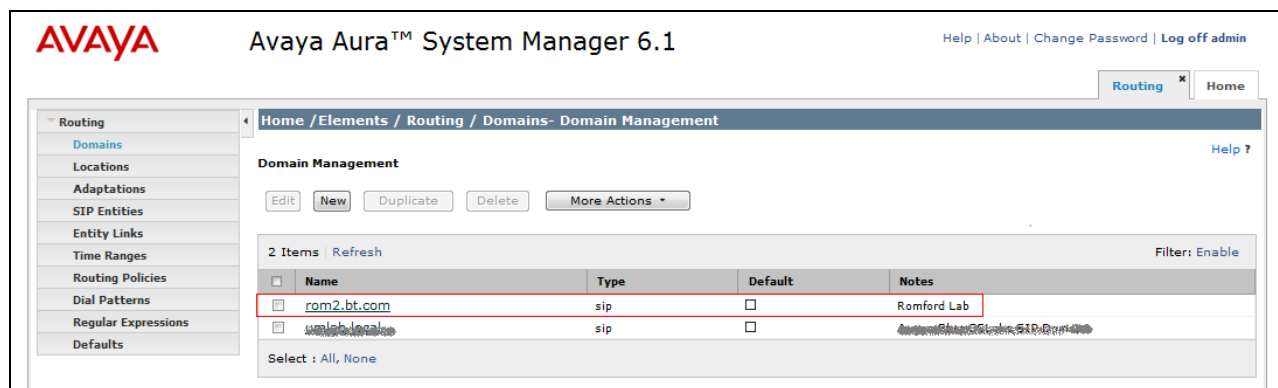
- Log in to Avaya Aura® Session Manager
- Administer SIP domain
- Administer Locations
- Administer Adaptations
- Administer SIP Entities
- Administer Entity Links
- Administer Routing Policies
- Administer Dial Patterns

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

Access 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 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 shows a table of SIP domains. The table has columns for "Name", "Type", "Default", and "Notes". Two domains are listed: "rom2.bt.com" and "rom2.bt.com". The "rom2.bt.com" entry is highlighted with a red box, and its "Notes" field contains "Romford Lab".

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

### 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. Under 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 breadcrumb trail is Home / Elements / Routing / Locations- Location Details. The left-hand navigation menu includes Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Location Details' and contains the following sections:

- General:** The 'Name' field is set to 'Romford Avaya Lab'. There is a 'Notes' field below it.
- Overall Managed Bandwidth:** 'Managed Bandwidth Units' is set to 'Mbit/sec' and 'Total Bandwidth' is set to '1000'.
- Per-Call Bandwidth Parameters:** 'Default Audio Bandwidth' is set to '80 Kbit/sec'.
- Location Pattern:** An 'Add' button is visible. Below it, a table lists the configured patterns:

IP Address Pattern	Notes
*192.168.1.*	Romford Avaya Lab

At the bottom of the table, there is a 'Select' dropdown menu with options for 'All' and 'None'.



## 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, under 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

AVAYA Avaya Aura™ System Manager 6.1

Help | About | Change Password | Log off admin

Routing x 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 **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 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 is titled 'SIP Entity Details' and 'General'. The configuration fields are as follows:

- Name: Romford SM 6.1
- FQDN or IP Address: 192.168.1.18
- Type: Session Manager
- Notes: (empty)
- Location: Romford Avaya Lab
- Outbound Proxy: (empty)
- Time Zone: Europe/London
- Credential name: (empty)
- SIP Link Monitoring: Use Session Manager Configuration

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 this sample configuration TCP was used to connect to Communication Manager and UDP was used to connect to the BT SIP Trunk Service.
- 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 with 'Add' and 'Remove' buttons. Below is a table with 3 items and a 'Filter: Enable' option. The table has columns for Port, Protocol, Default Domain, and Notes. A red box highlights the first three rows of the table.

<input type="checkbox"/>	Port	Protocol	Default Domain	Notes
<input type="checkbox"/>	5060	TCP	rom2.bt.com	
<input type="checkbox"/>	5060	UDP	rom2.bt.com	
<input type="checkbox"/>	5061	TLS	rom2.bt.com	

Select : All, None

## 6.5.2. Avaya Aura® Communication Manager SIP Entity

The following screens show the SIP entity for Communication Manager which is configured as an Access Element. The **FQDN or IP Address** field is set to the IP address of the CLAN 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 breadcrumb trail is Home / Elements / Routing / SIP Entities - SIP Entity Details. The left sidebar shows a navigation menu with 'SIP Entities' selected. The main content area is titled 'SIP Entity Details' and 'General'. Two red boxes highlight specific configuration fields: the first box encloses the 'Name' (Romford CM5.2), 'FQDN or IP Address' (192.168.1.4), and 'Type' (CM) fields; the second box encloses the 'Adaptation' (Romford CM i/c and o/g PSTN), 'Location' (Romford Avaya Lab), and 'Time Zone' (Europe/London) dropdown menus. Other visible fields include 'Notes' (CLAN address), 'Override Port & Transport with DNS SRV' (unchecked), 'SIP Timer B/F (in seconds)' (4), 'Credential name' (empty), 'Call Detail Recording' (none), and 'SIP Link Monitoring' (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

Help | About | Change Password | Log off admin

Routing x Home

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

SIP Entity Details

Commit Cancel Help ?

General

\* Name: NOAS SBC Birm2

\* FQDN or IP Address: 192.168.5.62

Type: Other

Notes: Primary SIP inbound / outbound

Adaptation: [dropdown]

Location: NOAS SIP Service

Time Zone: Europe/London

Override Port & Transport with DNS SRV:

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

Credential name: [text box]

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

Help | About | Change Password | Log off admin

Routing x Home

Home / Elements / Routing / SIP Entities- SIP Entities

SIP Entities

Edit New Duplicate Delete More Actions

32 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 CMS.2	192.168.1.4	CM	CLAN address
Romford SM 6.1	192.168.1.18	Session Manager	

Select : All, None < Previous Page 2 of 3 Next >

## 6.6. Administer Entity Links

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

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

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

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

Routing Home

Home / Elements / Routing / Entity Links - Entity Links Help ?

Entity Links

Edit New Duplicate Delete More Actions

29 Items Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
Romford SM6.1 to Romford CM5.2	Romford SM 6.1	TCP	5060	Romford CM5.2	5060	<input checked="" type="checkbox"/>	
Romford SM 6.1 to Romford CM 6.1	Romford SM 6.1	TLS	5061	Romford CM6.1	5061	<input checked="" type="checkbox"/>	
RomSM6.1toBirm1	Romford SM 6.1	UDP	5060	NOAS SBC Birm1	5060	<input checked="" type="checkbox"/>	
RomSM6.1toBirm2	Romford SM 6.1	UDP	5060	NOAS SBC Birm2	5060	<input checked="" type="checkbox"/>	
RomSM6.1toMan1	Romford SM 6.1	UDP	5060	NOAS SBC Man1	5060	<input checked="" type="checkbox"/>	
RomSM6.1toMan2	Romford SM 6.1	UDP	5060	NOAS SBC Man2	5060	<input checked="" type="checkbox"/>	

Select: All, None < Previous Page 2 of 2 Next >

## 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 the 'Routing Policies' menu item selected. The main content area is titled 'Routing Policy Details' and includes a 'Commit' button and a 'Cancel' button. The 'General' section shows the 'Name' field with the value 'NOAS Calls to Rom CM5.2'. The 'SIP Entity as Destination' section has a 'Select' button. Below this is a table with one row: Romford CM5.2, 192.168.1.4, CM, CLAN address. The 'Time of Day' section has 'Add', 'Remove', and 'View Gaps/Overlaps' buttons. Below this is a table with one row: Ranking 0, Name 24/7, Mon, Tue, Wed, Thu, Fri, Sat, Sun, Start Time 00:00, End Time 23:59, Notes Time Range 24/7.

Name	EQDN or IP Address	Type	Notes
Romford CM5.2	192.168.1.4	CM	CLAN address

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

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. Lower number means higher ranking.

The screenshot displays the 'Routing Policy Details' page in Avaya Aura System Manager 6.1. The page title is 'Home / Elements / Routing / Routing Policies- Routing Policy Details'. The left sidebar shows a navigation menu with 'Routing Policies' selected. The main content area is divided into sections: 'General', 'SIP Entity as Destination', and 'Time of Day'.

**General Section:**

- Name:** SIP Trunk Calls to Birm2
- Disabled:**
- Notes:** (empty text box)

**SIP Entity as Destination Section:**

Name	FQDN or IP Address	Type	Notes
NOAS SBC Birm2	192.168.5.62	Other	

**Time of Day Section:**

Buttons: 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"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<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 'Dial Pattern Details' configuration page in Avaya Aura System Manager 6.1. The page is titled 'Home / Elements / Routing / Dial Patterns - Dial Pattern Details'. The left navigation menu includes options like Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is divided into several sections:

- General:** Contains form fields for 'Pattern' (value: +44208), 'Min' (value: 6), 'Max' (value: 36), 'Emergency Call' (checkbox), 'SIP Domain' (dropdown menu showing 'rom2.bt.com'), and 'Notes' (text: 'Outer London via SIP').
- Originating Locations and Routing Policies:** Features an 'Add' and 'Remove' button, a '4 Items Refresh' link, and a table with columns: 'Originating Location Name', 'Originating Location Notes', 'Routing Policy Name', 'Rank', 'Routing Policy Disabled', 'Routing Policy Destination', and 'Routing Policy Notes'. The table lists four entries with ranks 10, 20, 30, and 40. A red box highlights the 'Rank' column.
- Denied Originating Locations:** Features an 'Add' and 'Remove' button and shows '0 Items Refresh'.

At the bottom right, there are 'Commit' and 'Cancel' buttons. A legend at the bottom left indicates '\* Input Required'.

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	
<input type="checkbox"/> -ALL-	Any Locations	SIP Trunk calls to Birm1	20	<input type="checkbox"/>	NOAS SBC Birm1	
<input type="checkbox"/> -ALL-	Any Locations	SIP Trunk calls to Man2	30	<input type="checkbox"/>	NOAS SBC Man2	
<input type="checkbox"/> -ALL-	Any Locations	SIP Trunk calls to Man1	40	<input type="checkbox"/>	NOAS SBC Man1	

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

The screenshot displays the 'Dial Pattern Details' configuration page in Avaya Aura System Manager 6.1. The 'General' section includes the following fields:

- Pattern:** +44207
- Min:** 12
- Max:** 36
- Emergency Call:**
- SIP Domain:** rom2.bt.com
- Notes:** Inbound DDI 324X for Rom CM5.2

The 'Originating Locations and Routing Policies' section shows a table with one item:

Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/> NOAS SIP Service		NOAS Calls to Rom CM5.2	0	<input type="checkbox"/>	Romford CM5.2	

## 7. 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 displays the 'SIP Entity, Entity Link Connection Status' page in Avaya Aura System Manager 6.1. The page title is 'SIP Entity, Entity Link Connection Status' and it shows 'All Entity Links to SIP Entity: Romford CM5.2'. A table displays the connection status for the selected entity:

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	5060	TCP	Up	200 OK	Up

- 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 35

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 remains 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.

## 8. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Communication Manager Access Element 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.

## 9. 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*, May 2009, Document Number 03-300509.
- [4] *Avaya Aura® Communication Manager Feature Description and Implementation*, May 2009, 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*, January 2011, Document Number 03-603473
- [7] *Administering Avaya Aura® Session Manager*, March 2011, Document Number 03-603324.
- [8] RFC 3261 *SIP: Session Initiation Protocol*, <http://www.ietf.org/>

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