Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Communication Manager and Avaya SIP Enablement Services for British Telecom Corporate Fusion – Site Solution – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Avaya Communication Manager and Avaya SIP Enablement Services to work with British Telecom Corporate Fusion – Site Solution.

British Telecom Corporate Fusion – Site Solution is a fixed-mobile-convergence solution for the enterprise. It uses a combination of hardware, software and networking technology to integrate mobile communications with the corporate network, taking advantage of voice over wireless network across the office estate, as well as extending additional voice functionality to the mobile phone.

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 procedures for configuring Avaya Communication Manager and Avaya SIP Enablement Services (SES) to work with British Telecom (BT) Corporate Fusion – Site Solution.

The BT Corporate Fusion – Site Solution is specially designed for enterprises that wish to retain on-site ownership and control of their communication assets, and where integration with their PBX is a pre-requisite. It provides fixed-mobile-convergence (FMC) services to the enterprise users.

BT Corporate Fusion – Site Solution provides call services using a Subscriber Directory Number (SDN) which is a number that a subscriber gives to people. A profile is defined for the SDN which contains a set of devices such as PBX extensions, fixed-line phones, mobile phones, or dual-mode mobile phones.

1.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing.

The feature testing focused on verifying the following features of BT Corporate Fusion – Site Solution:

- Routing of incoming calls to a dual-mode mobile phone over wireless LAN (WLAN)
- Handoff of calls between Enterprise WLAN and public cellular networks
- Simultaneous ringing of multiple devices when an SDN is called
- Pickup of ongoing calls from another SDN device
- Features extended to outgoing calls from PBX extension or mobile phones

In addition, phone features such as call transfer and conference, sending DTMF tones during a call and handling of multiple calls by a subscriber are verified.

The serviceability testing focused on verifying the ability of BT Corporate Fusion – Site Solution to recover from adverse conditions such as resetting the Avaya Communication Manager, Avaya SES server and BT Fusion Server.

1.1.1. Call Flows

The following table lists the different kinds of call flow scenarios that may occur.

Scenario	Description
Non-subscriber to non-subscriber	Avaya Communication Manager directs the call through the SIP Trunk to the BT Fusion Server. BT Fusion Server passes the call back and terminates at the extension.

Scenario	Description
Non-subscriber to subscriber	Avaya Communication Manager directs the call through the SIP Trunk to the BT Fusion Server. BT Fusion Server bridges the incoming call leg to the subscriber’s devices.
Non-subscriber to PSTN	Avaya Communication Manager directs the call through the SIP Trunk to the BT Fusion Server. BT Fusion Server passes the call back and it is routed to the PSTN.
Subscriber to non-subscriber	Avaya Communication Manager sends the call to the BT Fusion Server through the SIP Trunk. BT Fusion Server places another call to the non-subscriber on Avaya Communication Manager and bridges the two calls.
Subscriber to subscriber	Avaya Communication Manager sends the call to the BT Fusion Server through the SIP Trunk. BT Fusion Server bridges the call leg to the subscriber on the SIP Trunk and adds a prefix before the dialed number. Due to the prefix, Avaya Communication Manager loops the call back to the BT Fusion Server. BT Fusion Server then sends the call through the Avaya Communication Manager to the destination subscriber.
Subscriber to PSTN	Avaya Communication Manager sends the call through the SIP Trunk to the BT Fusion Server. BT Fusion Server places a new call to the PSTN through Avaya Communication Manager and bridges the subscriber’s call.
PSTN to non-subscriber	Avaya Communication Manager directs the call through the SIP Trunk to the BT Fusion Server as a PBX extension number. BT Fusion Server routes the call back to Avaya Communication Manager and it terminates at the extension.
PSTN to subscriber	Avaya Communication Manager directs the call through the SIP Trunk to the BT Fusion Server as a PBX extension number. BT Fusion Server places new calls to the subscriber’s devices and bridges the calls.

A detailed description of the call flows involved in each scenario is beyond the scope of this document. A general overview of the routing strategy is described below:

- In general, all calls on the Avaya Communication Manager are routed via the Avaya SES to the BT Fusion Server, which will then make a decision on the routing of the call.
- BT Fusion Server will specify a prefix (“000” in this configuration) for any calls that are sent to the Avaya Communication Manager. The prefix is also set for pass-through calls for which BT Fusion Server does not provide any features.
- BT Fusion Server will specify a prefix (“0009” in this configuration) for all calls going out to the PSTN.
- For the special case of looped calls, BT Fusion Server will specify a prefix (“000101” in this configuration) to indicate to Avaya Communication Manager that the call should be returned to the BT Fusion Server.

1.2. Support

British Telecom customers can get support for BT Corporate Fusion – Site Solution by calling +65 6290 7100.

2. Reference Configuration

Figure 1 illustrates the test configuration used to verify the BT Corporate Fusion – Site Solution. It consists of an Avaya Communication Manager running on an Avaya S8300 Server and Avaya G700 Media Gateway, and has connections to the PSTN via an ISDN-PRI trunk. The Avaya 9600 Series IP Telephones and the PCs running Avaya IP Softphones are configured as subscribers. The BT Fusion Server provided is connected via SIP trunks to the Avaya SES Server. The Nokia E51 and E65 are installed with the BT Corporate Fusion Client software and function as dual-mode mobile phones. A generic Ethernet switch provides network connectivity to all the servers and devices and a WLAN access point provides WLAN connectivity to the dual-mode mobile phones.

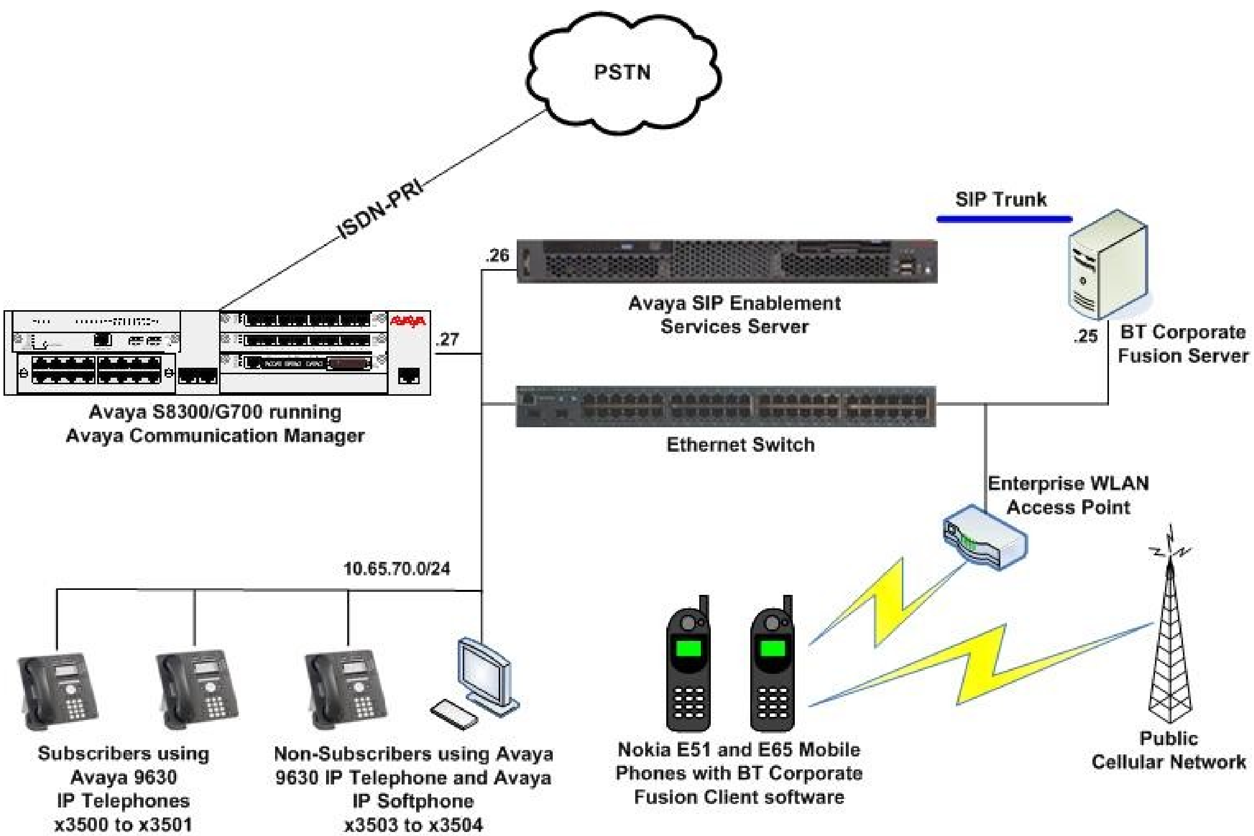


Figure 1: Test Configuration

3. Equipment and Software Validated

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

Equipment	Software
Avaya S8300 Server	Avaya Communication Manager 4.0.1 (R014x.00.1.731.2)
Avaya G700 Media Gateway - MM710BP DS1 Media Module	28.17.0 HW11, FW044
Avaya SIP Enablement Services	SES-4.0.0.0-033.6
Avaya 9630 IP Telephones	1.5 (H.323)
Avaya IP Softphone	6.0 Service Pack 4
BT Fusion Server	BT Converged Services Node 3.5 build 346

4. Configure Avaya Communication Manager

This section provides the procedures for configuring Avaya Communication Manager with BT Corporate Fusion – Site Solution. All configuration changes in Avaya Communication Manager are performed through the System Access Terminal (SAT). The highlights in the following screens indicate the parameter values used during the compliance test. Default values were used for fields not configured. After the completion of the configuration, perform a **save translation** command to make the changes permanent.

Note: The configuration on the Avaya Communication Manager requires extensive changes to the way a call is being routed. As such, it is advisable for a detail understanding of the current routing configuration of a system to be carried out before implementing the changes. Furthermore, the sample configuration described in these Application Notes is simplified for the purpose of presenting the concepts in a clear and concise manner and in no way represents the configuration for all systems. Further entries are needed to handle the calling of other destinations such as mobile and international numbers.

Step	Description
1.	Enter the **change dialplan analysis** command to add digits "8" and "9" as feature access codes (FAC).

```
change dialplan analysis                                           Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                                            Percent Full:     0

       Dialed    Total  Call    Dialed    Total  Call      Dialed    Total  Call
       String    Length Type    String    Length Type      String    Length Type
        000         7   ext
        3           4   ext
        8           1   fac
        9           1   fac
        *           3   fac
        #           3   fac
```

Step	Description
2.	Enter the **change feature-access-codes** command to add digits “8” and “9” as feature access codes (FAC).

```
change feature-access-codes                                    Page   1 of   8
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code: *00
         Abbreviated Dialing List2 Access Code: *01
         Abbreviated Dialing List3 Access Code: *02
Abbreviated Dial - Prgm Group List Access Code: *03
                      Announcement Access Code: *04
                       Answer Back Access Code: *05

      Auto Alternate Routing (AAR) Access Code: 8
    Auto Route Selection (ARS) - Access Code 1: 9      Access Code 2:
                 Automatic Callback Activation: *06     Deactivation: #06
Call Forwarding Activation Busy/DA: *07    All: *08     Deactivation: #08
   Call Forwarding Enhanced Status: *09    Act: *10     Deactivation: #10
                         Call Park Access Code: *11
                       Call Pickup Access Code: *12
CAS Remote Hold/Answer Hold-Unhold Access Code:
                  CDR Account Code Access Code:
                        Change COR Access Code:
                   Change Coverage Access Code:
                   Contact Closure   Open Code:          Close Code:
```

Step	Description
3.	Enter the **change dialplan parameters** command and set **UDP Extension Search Order** field to **udp-table-first**.

```
change dialplan parameters                                     Page   1 of   1
                             DIAL PLAN PARAMETERS

                 Local Node Number:
                   ETA Node Number:
               ETA Routing Pattern:
        UDP Extension Search Order: udp-table-first

 EXTENSION DISPLAY FORMATS
                                    Inter-Location/SAT     Intra-Location
        6-Digit Extension:          xxxxxx                 xxxxxx
        7-Digit Extension:          xxx-xxxx               xxx-xxxx
        8-Digit Extension:          xxxxxxxx               xxxxxxxx
        9-Digit Extension:          xxx-xxx-xxx            xxx-xxx-xxx
       10-Digit Extension:          xxx-xxx-xxxx           xxx-xxx-xxxx
       11-Digit Extension:          xxxx-xxx-xxxx          xxxx-xxx-xxxx
       12-Digit Extension:          xxxxxx-xxxxxx          xxxxxx-xxxxxx
       13-Digit Extension:          xxxxxxxxxxxxx          xxxxxxxxxxxxx
     AAR/ARS Internal Call Prefix:
AAR/ARS Internal Call Total Length:
```

Step	Description
4.	Enter the **change uniform-dialplan 0** command to define the **UNIFORM DIAL PLAN TABLE**. In this configuration, when the 4-digit extension starting with "3" is dialed, Avaya Communication Manager sends the call to the BT Fusion Server through the SIP trunk using Auto Alternate Routing (AAR). When BT Fusion Server needs to send the call back to the local extension, it appends "000" to the extension which is handled by the Uniform Dial Plan Table and the call is routed to the extension.

	<pre>change uniform-dialplan 0 Page 1 of 2
	                       UNIFORM DIAL PLAN TABLE
	                                                          Percent Full: 0
	
	  Matching Insert Node
	  Pattern Len Del Digits Net Conv Num
	 0003 7 3 ext n
	 3 4 0 aar n</pre>

5.	Enter the **change node-name ip** command to define the node name and IP address for the SES Server. As this is an Avaya S8300 Server, the node name **procr** is also used to define the SIP signaling interface in **Step 8**.

	<pre>change node-names ip Page 1 of 2
	                                  IP NODE NAMES
	    Name **IP Address**
	default 0.0.0.0
	procr 10.65.70.27
	ses 10.65.70.26</pre>

Step	Description
6.	Enter the **change ip-network-region r** command, where **r** is the IP network region used for the communication between Avaya Communication Manager and BT Fusion Server. Set **Authoritative Domain** to the IP address of the BT Fusion Server and **Codec Set** to the IP codec set to be used. Set **IP-IP Direct Audio** for both **Intra-region** and **Inter-region** to **yes** to allow audio traffic to be sent directly between endpoints.

```
change ip-network-region 2                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 2
Location:          Authoritative Domain: 10.65.70.25
    Name: BT Fusion
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 2              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        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
```

Step	Description
7.	Enter the **change ip-codec-set c** command, where **c** is the IP codec set used in **Step 6**. In this configuration, the **G.711MU** and **G.711A** IP codecs are used.

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

                          IP Codec Set
    Codec Set: 2

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU          n          2        20
 2: G.711A           n          2        20
 3:
 4:
 5:
 6:
 7:

     Media Encryption
 1: none
 2:
 3:
```

Step	Description
8.	Enter the **add signaling-group s** command to add the signaling group for the SIP trunk. Set **Group Type** to **sip** and **Transport Method** to **tls**. Set **Near-end Node Name** and **Far-end Node Name** to **procr** and **ses** respectively as defined in **Step 5**. Set **Far-end Domain** to the IP address of the BT Fusion Server and **Far-end Network Region** to the IP network region defined in **Step 6**. Set **Direct IP-IP Audio Connections** to allow audio traffic to be sent directly between endpoints.

```
add signaling-group 2                                           Page   1 of   1
                                SIGNALING GROUP

 Group Number: 2                Group Type: sip
                          Transport Method: tls

   Near-end Node Name: procr                  Far-end Node Name: ses
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 2
       Far-end Domain: 10.65.70.25

                                      Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections?
y
                                                     IP Audio Hairpinning? n
 Enable Layer 3 Test? n
 Session Establishment Timer(min): 3
```

Step	Description
9.	Enter the **add trunk-group t** command, where **t** is the trunk group used to connect to the BT Fusion Server. Set **Group Type** to **sip** and specify the **Group Name**. Set **TAC** to an available trunk access code as per the dial plan and **Service Type** to **tie**. Set **Signaling Group** to the signaling group created in **Step 8** and **Number of Members** to a sufficiently large value to support the number of calls between Avaya Communication Manager and BT Fusion Server.

```
change trunk-group 2                                            Page   1 of  21
                                TRUNK GROUP

Group Number: 2                   Group Type: sip          CDR Reports: n
  Group Name: SIP-acmtocsn               COR: 1        TN: 1        TAC: 702
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                     Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n

                                                  Signaling Group: 2
                                                Number of Members: 100
```

Step	Description
10.	Enter the **change aar analysis 0** command to define the AAR Digit Analysis Table. In this configuration, the 4-digit extension starting with "3" is sent to Route Pattern 2, which then routes the calls through SIP Trunk Group 2 to the BT Fusion Server. The other entry is for extensions prefixed with "101" which are to be looped back to the BT Fusion Server for further routing.

```
change aar analysis 0                                          Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                                           Percent Full:    3

          Dialed             Total     Route     Call    Node   ANI
          String            Min  Max  Pattern    Type    Num    Reqd
    101                     7    7      2        aar            n
    3                       4    4      2        aar            n
```

Step	Description
11.	Enter the **change route-pattern 2** command to define the Route Pattern 2 Table. This route pattern is used to route the calls through SIP Trunk Group 2 to the BT Fusion Server.

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

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

Step	Description
12.	Enter the **change ars analysis 0** command to define the ARS Digit Analysis Table. In this configuration, the 8-digit PSTN number starting with "6" is sent to route pattern 5, which then routes the calls through SIP Trunk Group 2 to the BT Fusion Server. The other entry "000" is used to do the actual routing of the call to the PSTN and is explained in **Step 14**.

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

          Dialed              Total      Route      Call    Node   ANI
          String            Min  Max    Pattern     Type    Num    Reqd
    000                     11   11       1         pubu              n
    6                        8    8       5         locl              n
```

Step	Description
13.	Enter the **change route-pattern 5** command to define the Route Pattern 5 Table. This route pattern is used to route the calls through SIP Trunk Group 2 to the BT Fusion Server. Note that the digit "9" is inserted to identify that the call is to be routed to the PSTN.

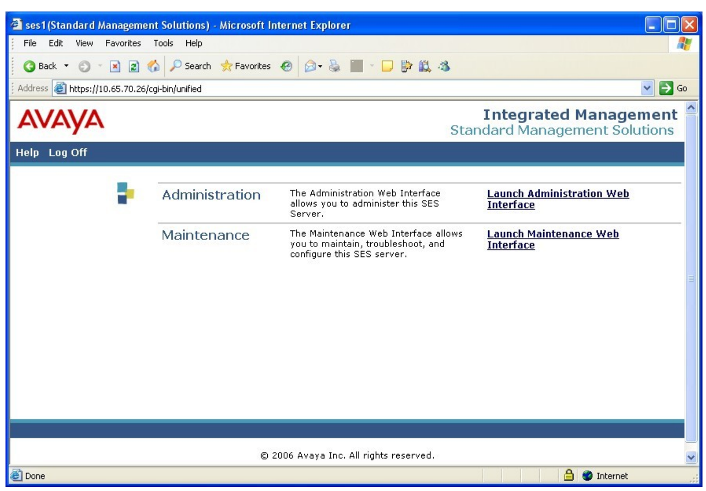
```
change route-pattern 5                                          Page   1 of   3
                    Pattern Number: 5   Pattern Name: ACM-to-CSN PSTN
                             SCCAN? n      Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                              DCS/ IXC
    No          Mrk Lmt List Del  Digits                                QSIG
                             Dgts                                       Intw
 1: 2    0                         9                                     n
user
 2:                                                                      n
user
 3:                                                                      n
user
 4:                                                                      n
user
 5:                                                                      n
user
 6:                                                                      n
user

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

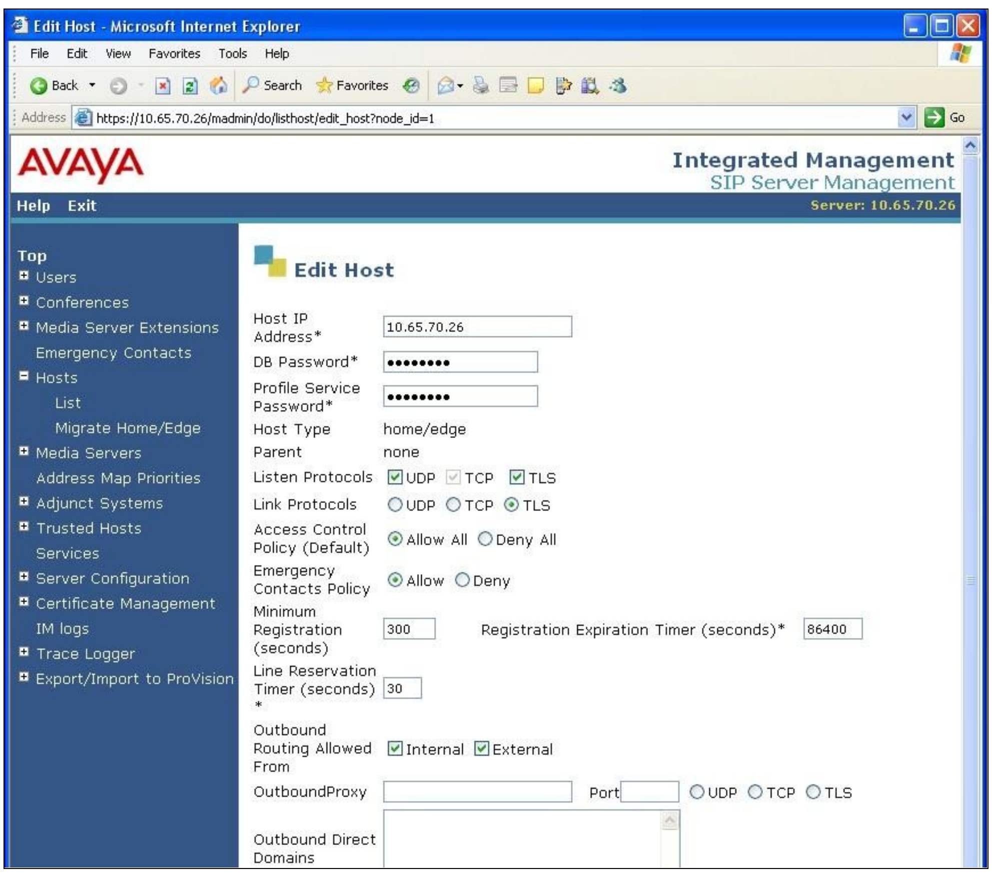
Step	Description
14.	Enter the **change inc-call-handling-trmt trunk-group g** command, where **g** is the SIP trunk group defined in **Step 9**. For calls that are to be routed to the PSTN, BT Fusion Server adds the prefix “0009” to the number. The entry below matches the prefix and changes it to “9000”, where “9” is the ARS Feature Access Code. The next digits “000” are then matched in the ARS Digit Analysis Table as shown in **Step 12** and the 8-digit PSTN number is sent to route pattern 1, which then routes the call to the PSTN. The second entry is to handle looped calls which are to be routed back to the BT Fusion Server for further routing. The digit “8” is inserted so that the AAR Digit Analysis Table in **Step 10** is used to route the call.

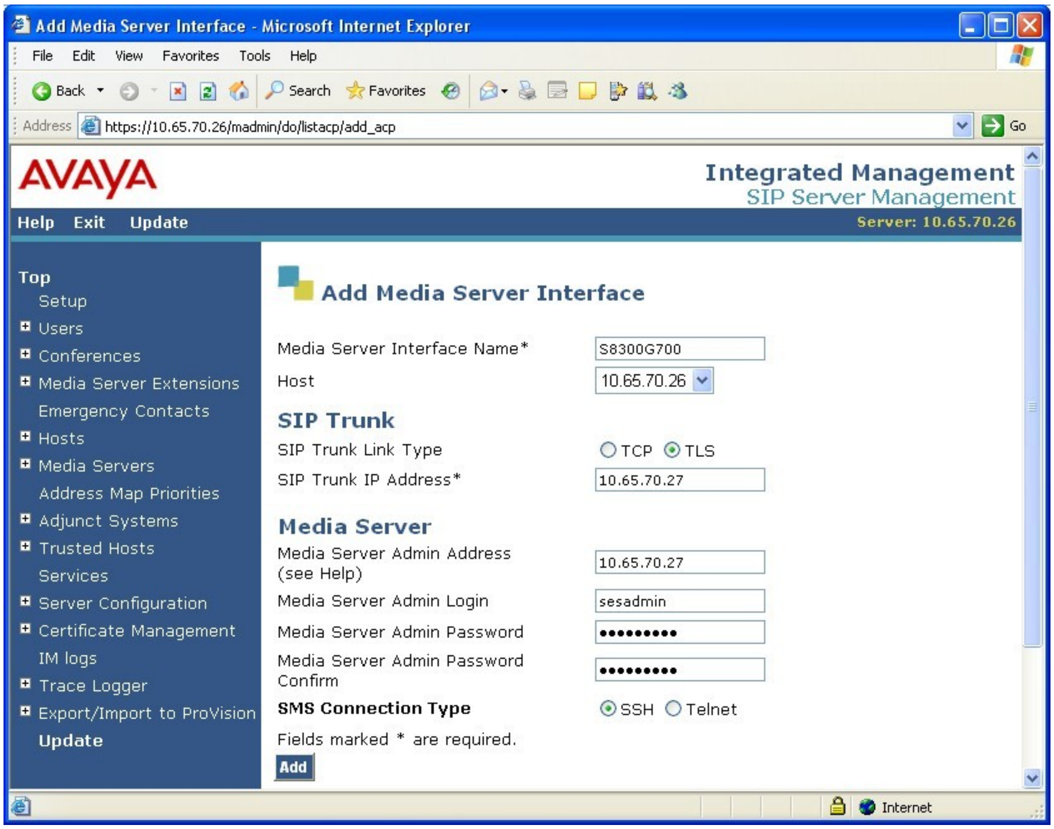
```
change inc-call-handling-trmt trunk-group 2                          Page   1 of   3
                       INCOMING CALL HANDLING TREATMENT
 Service/      Called    Called      Del Insert
 Feature        Len      Number
 tie             12 0009              4   9000
 tie             10 000101            3   8
```

5. Configure Avaya SIP Enablement Services

This section covers the administration of Avaya SES. Avaya SES is configured via an Internet browser using the Administration web interface. It is assumed that Avaya SES software and the license file have already been installed on Avaya SES. During the software installation, the **initial_setup** script is run on the Linux shell of the server to specify the IP network properties of the server along with other parameters. For information on these installation tasks, refer to [3].

Step	Description
1.	Access the Avaya SES Administration web interface, by entering **http://<ip-addr>/admin** as the URL in an Internet browser, where <ip-addr> is the IP address of Avaya SES server. Log in with the appropriate credentials and then select the **Launch Administration Web Interface** link from the main screen. 

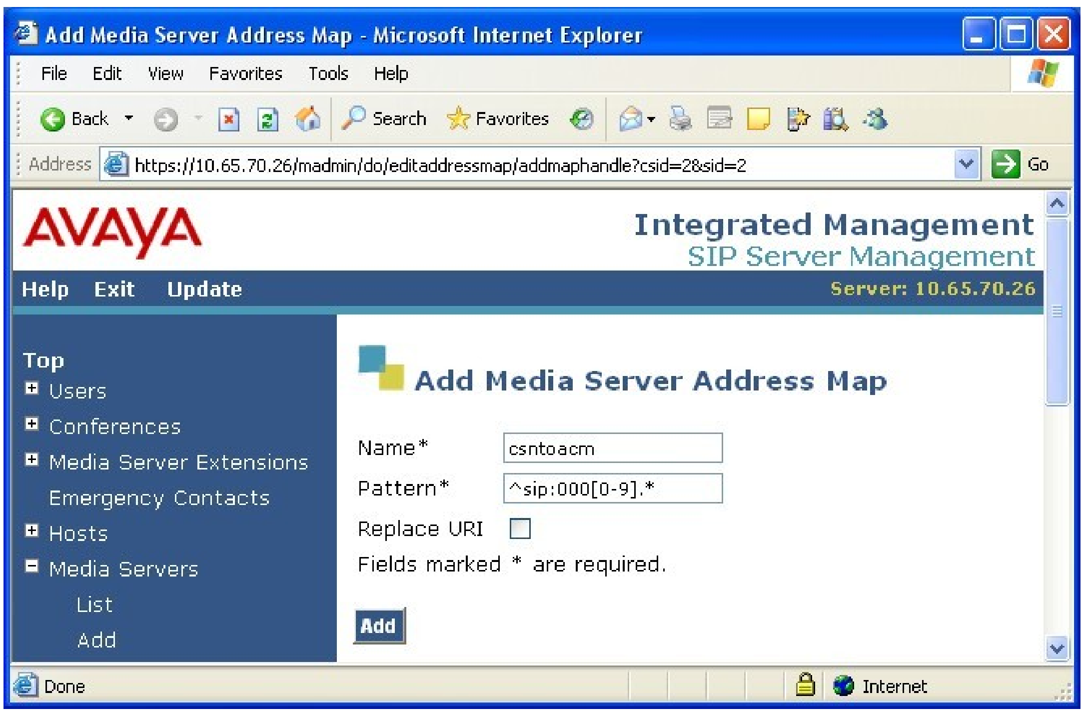
Step	Description
2.	Verify the Avaya SES Host information using the **Edit Host** page. In this compliance test, the Avaya SES **Host Type** is **home/edge.** This means that a single Avaya SES routes SIP messages between the BT Fusion and Avaya Communication Manager. Navigate to the **Edit Host** page by following the **Hosts** link in the left navigation pane and then clicking on the **Edit** option under the **Commands** section of the **List Hosts** screen (not shown). On the **Edit Host** screen: • Verify that the IP address of this Avaya SES server is in the **Host IP Address** field. • Verify that the **UDP**, **TCP** and **TLS** checkboxes are enabled as **Listen Protocols**. • Verify that **TLS** is selected as the **Link Protocol**. • Ensure that the **Outbound Proxy** and **Outbound Direct Domains** fields are left blank. • Default values for the remaining fields may be used. • Click the **Update** button only if changes are necessary. Otherwise, exit the **Edit Host** page by selecting the **Top** link on the left navigation bar. 

Step	Description
3.	Expand the **Media Servers** option within Avaya SES **SIP Server Management** page, and select **Add** (not shown) to display the **Add Media Server Interface** page. This will create the Avaya SES side of the SIP trunk previously created in Avaya Communication Manager in **Section 4**, **Step 9**. Enter the following information: • Enter any descriptive name in the **Media Server Interface Name** field. • Select the Avaya SES IP address displayed in **Step 2** in the **Host** field. • Select **TLS** (Transport Link Security) for the **SIP Trunk Link Type**. TLS provides encryption at the transport layer between Avaya Communication Manager and the Avaya SES. • Enter the IP address of the processor interface **procr** shown in **Section 4**, **Step 5** in the **SIP Trunk IP Address** field. • Click **Add** and the click **Update**. 

Step	Description
4.	Calls from the BT Fusion Server arriving at Avaya SES are routed to the appropriate Avaya Communication Manager for termination services. This routing is specified in a Media Server Address Map configured on Avaya SES as shown in **Step 5**. This routing compares the Uniform Resource Identifier (URI) of an incoming INVITE message to the pattern configured in the Media Server Address Map**,** and if there is a match, the call is routed to the designated Avaya Communication Manager. The URI usually takes the form of *sip:user@domain*, where *domain* can be a domain name or an IP address. Patterns must be specific enough to uniquely route incoming calls to the proper destination if there are multiple Avaya Communication Manager systems supported by the Avaya SES server. In this configuration, BT Fusion Server will set the *user* portion of the SIP URI to contain the prefix "000" for all calls to the Avaya Communication Manager. The Avaya SES will forward the messages with matching patterns to the appropriate Media Server Interface. An example of a SIP URI in an INVITE message for a call to be terminated to an extension would be: **INVITE sip:0003501@10.65.70.26 SIP/2.0** For a call to be routed to the PSTN, the SIP URI in an INVITE message would be: **INVITE sip:000968728599@10.65.70.26 SIP/2.0** The user portion in this case is the 8 digit DID number "68728599" with the prefix "0009".

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>5.</td><td>
To configure a Media Server Address Map:

Select Media Servers in the left pane of the Administration web interface. This will display the List Media Servers screen (not shown).
Click on the Map link (not shown) associated with the appropriate media server, added in Step 3, to display the List Media Server Address Map screen (not shown).
Click on the Add Map In New Group link (not shown).
Enter any descriptive name in the Name field.
Enter the regular expression to be used for the pattern matching in the Pattern field. The pattern specification (without the double quotes) is: “^sip:000[0-9].*”.
Uncheck the Replace URI checkbox.
Click the Add button once the form is completed.



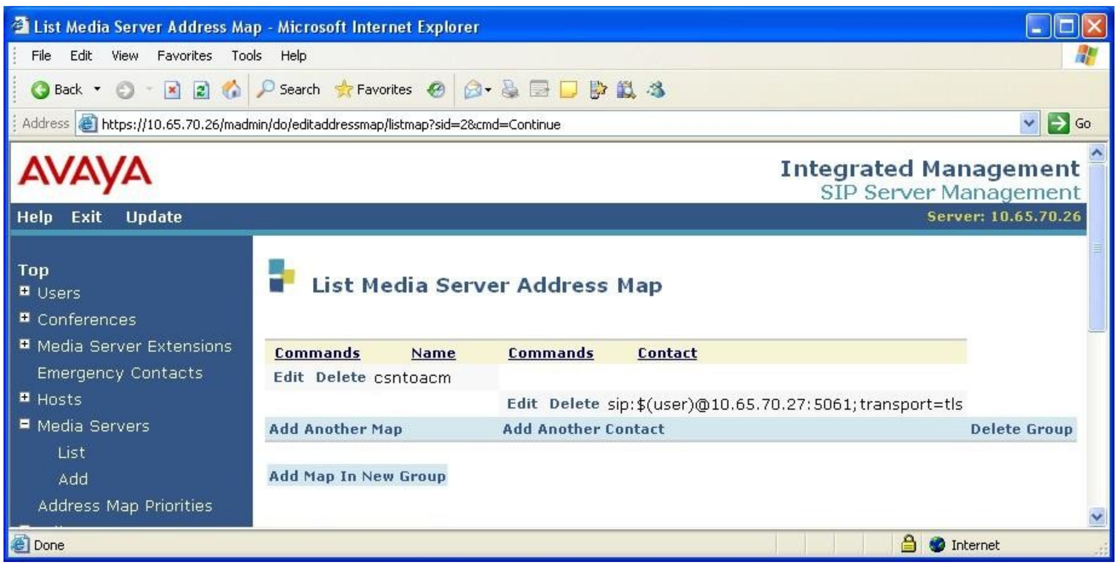
</td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>6.</td><td>

After configuring the media server address map, the **List Media Server Address Map** screen appears as shown below. Note that after the first **Media Server Address Map** is added, the **Media Server Contact** is created automatically. For the **Media Server Address Map** added in **Step 5**, the following contact was created:

sip:$(user)@10.65.70.27:5061;transport=tls

The contact specifies the processor address of the Avaya S8300 Server and the transport protocol used to send SIP signaling messages. The incoming number sent in the user part of the original request URI is substituted for $(user).



</td></tr>
<tr><td>7.</td><td>

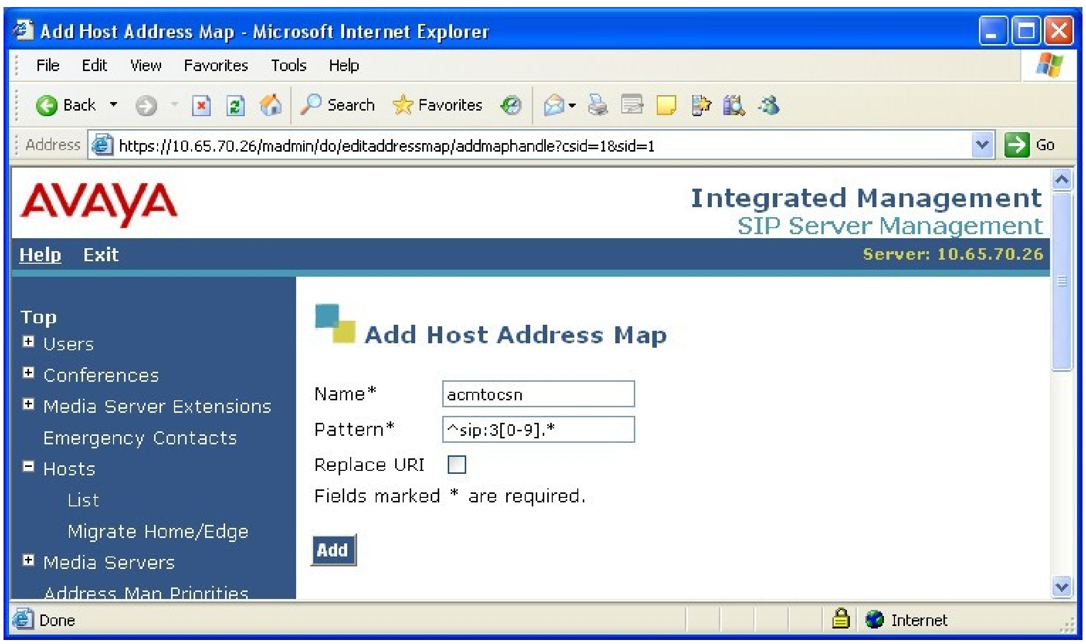
Calls from Avaya Communication Manager are sent to the BT Fusion Server using Host Address Maps within Avaya SES. As with the inbound media server address maps, these Host Address Maps use pattern matching on the SIP URI to direct messages to the corresponding contact address (e.g., BT Fusion Server) as shown in **Step 8**.

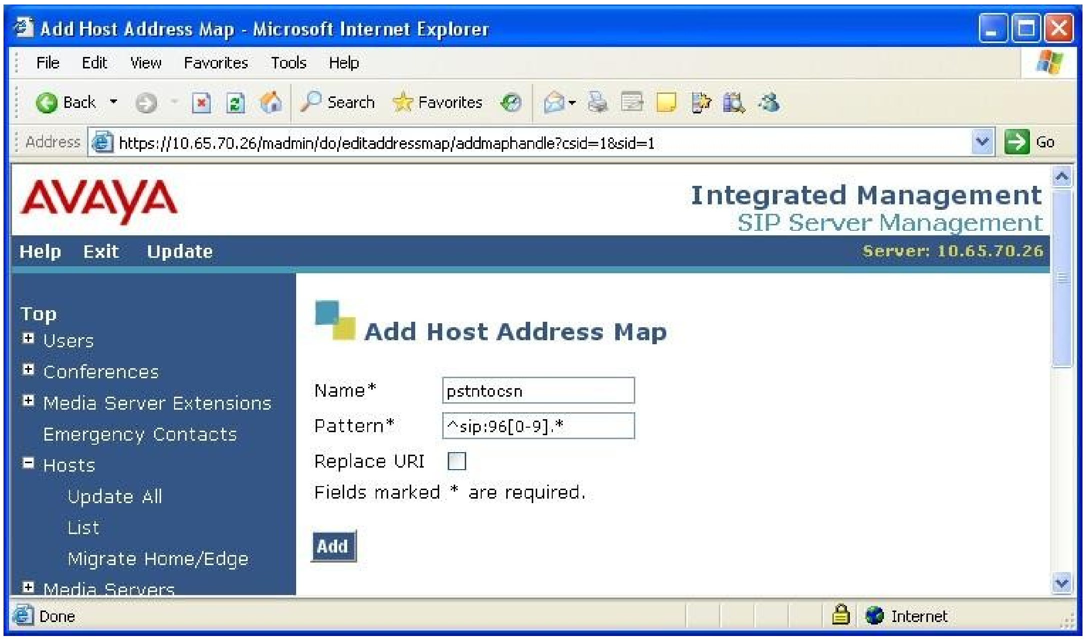
In this configuration, the Avaya SES routing rule for the SIP trunk group will be to send all calls on the Avaya Communication Manager to the BT Fusion Server. To perform this, several dialing patterns will be created in the Avaya SES.

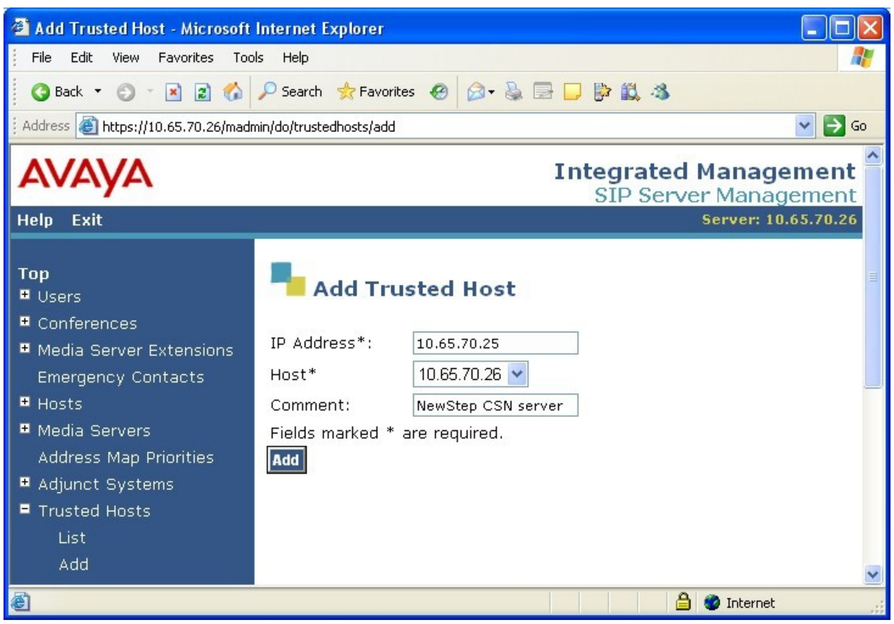
The first dialing pattern “^sip:3[0-9].*” is for numbers matching the user extension on Avaya Communication Manager which are routed to the BT Fusion Server using Uniform Dialing Plan.

The second dialing pattern “^sip:96[0-9].*” is for numbers to be routed to the PSTN, e.g. 968728599. The digit “9” was inserted in Route Pattern 5 in **Section 4**, **Step 10**.

</td></tr>
</table>

Step	Description
8.	To configure a **Host Address Map**: • Access the **Add Host Address Map** screen by selecting the **Hosts** link in the left pane of the Administration web interface and then clicking on the **Map** link [noth shown] associated with the appropriate host. The **List Host Address Map** screen is displayed [not shown]. • From this screen, click the **Add Map In New Group** link [not shown] to display the **Add Host Address Map** screen. • Enter any descriptive name in **Name** field. • Specify an appropriate pattern for the call type. In this example, the pattern used is "^sip:3[0-9].*". • Uncheck the **Replace URI** checkbox to allow further routing by the BT Fusion Server. • Click the **Add** button. 

Step	Description
9.	Repeat **Step 8** to configure another **Host Address Map** for the second dialing pattern described in **Step 7**. 

Step	Description
10.	The final step to complete the SIP trunk administration on Avaya SES is to designate BT Fusion Server as a trusted host. As a trusted host, Avaya SES will not issue SIP authentication challenges for incoming requests from the designated IP address. If multiple Avaya SES proxies are used, then each Avaya SES and BT Fusion Server pair must be added as a trusted host. To configure a Trusted Host: • Select **Trusted Hosts** in the left pane of the Administration web interface. This will display the **Manage Trusted Hosts** screen [not shown]. • Click on the **Add Managed Trusted Hosts** link [not shown]. Enter the IP address of the BT Fusion Server in the **IP Address** field as the Trusted Host which in this configuration is "10.65.70.25". • Select the IP address of the Avaya SES server in the **Host** field which in this configuration is "10.65.70.26". • Press the **Add** button and click **Continue** at the next screen [not shown]. 
11.	After completion the required configurations for Avaya SES, click Update to

6. Configure BT Fusion Server

The configuration of the BT Fusion Server is performed as part of the services delivered through the BT Corporate Fusion – Site Solution and will not be covered in these Application Notes.

7. General Test Approach and Test Results

All feature and serviceability test cases were performed manually.

All test cases were executed and passed.

8. Verification Steps

The following steps may be used to verify the configuration:

- From the Avaya Communication Manager SAT, use the **status signaling-group** command to verify that the SIP signaling group configured in **Section 4**, **Step 8** is in-service.
- From the Avaya Communication Manager SAT, use the **status trunk-group** command to verify that the SIP trunk group configured in **Section 4**, **Steps 9** is in-service.
- Verify that calls can be placed to/from a subscriber's telephone extension.
- From the Avaya Communication Manager SAT, use the **list trace tac** command to verify that the calls are routed to the expected trunks.

9. Conclusion

These Application Notes describe the configuration steps required for Avaya Communication Manager and Avaya SIP Enablement Services to interoperate with British Telecom Corporate Fusion – Site Solution. All feature and serviceability test cases were completed successfully.

10. Additional References

This section references the Avaya documentation relevant to these Application Notes. The following Avaya product documentation is available at http://support.avaya.com.

[1] *Administrator Guide for Avaya Communication Manager, February 2007*, Issue 3.1, Document Number 03-300509.
[2] *Feature Description and Implementation for Avaya Communication Manager*, February 2007, Issue 5, Document Number 555-245-205
[3] *Installing and Administering SIP Enablement Services*, May 2007, Issue 4, Document Number 03-600768
[4] *SIP Support in Avaya Communication Manager Running on the Avaya S8300, S8400, S8500 series and S8700 series Media Server*, May 2007, Issue 7, Document Number 555-245-206.

Information on British Telecom Corporate Fusion – Site Solution is available by selecting **Products and services > Corporate Fusion** on the following website:

http://www.globalservices.bt.com/

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.