



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

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# **Application Notes for Configuring SIP Trunking between the British Telecom Spain SIP trunking standard service and an Avaya Aura™ SIP Telephony Solution – Issue 1.0**

## **Abstract**

These Application Notes describe the steps to configure SIP trunking between the British Telecom (BT) Spain SIP trunking standard service and an Avaya Aura™ SIP telephony solution consisting of Avaya Aura™ Communication Manager, Avaya Aura™ SIP Enablement Services, and various Avaya telephones.

Enterprise customers with this Avaya SIP-based solution can connect via dedicated Internet access to use British Telecom as a service provider to complete PSTN calls. This allows customers with a converged network to lower PSTN telecommunication costs, easily obtain local number presence without offices in each geographic area, and easily manage their network services using web-based tools.

Information in these Application Notes has been obtained through 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 for configuring Session Initiation Protocol (SIP) trunking between the British Telecom Spain SIP trunking standard service and an Avaya Aura™ SIP telephony solution consisting of Avaya Aura™ SIP Enablement Services (SES), Avaya Aura™ Communication Manager and Avaya IP telephones using H.323 and SIP protocols as endpoints.

SIP is a standards-based communications approach designed to provide a common framework to support multimedia communication. RFC 3261 [8] is the primary specification governing this protocol. In the configuration described in these Application Notes, SIP is used as the signaling protocol between the Avaya components and the network service offered by British Telecom Spain. SIP manages the establishment and termination of connections and the transfer of related information such as the desired codec, calling party identity, etc.

## 1.1. British Telecom Spain SIP trunking standard service - IP Trunking Overview

The British Telecom (BT) Spain SIP trunking standard service with IP trunking used within these Application Notes serves as an interface between Avaya Telephones and ISDN, GSM and analog endpoints communicating with the British Telecom Spain network.

The BT SIP Trunking standard service uses UDP as transport protocol. Other transport protocols if required are covered in their respective documents.

BT Spain SIP Trunking service offers end-customers the option of minimizing costs in its access to PSTN calls. The BT Spain SIP Trunking service usually includes the trunks, low cost per calls, data network access to the BT Media Pop and a CPE (Customer Premises equipment) router with NAT Traversal feature enabled on it. There is no need for equipment other than a SIP Enabled the Avaya Aura™ Communication Manager as described in this document to make it work. There are special cases described later in this document where some modifications in the architecture are needed in order to use some features either in the BT Network or in the Avaya Aura™ Communications Manager solution. Contact a BT or Avaya Spain representative for more information.

## 1.2. Support

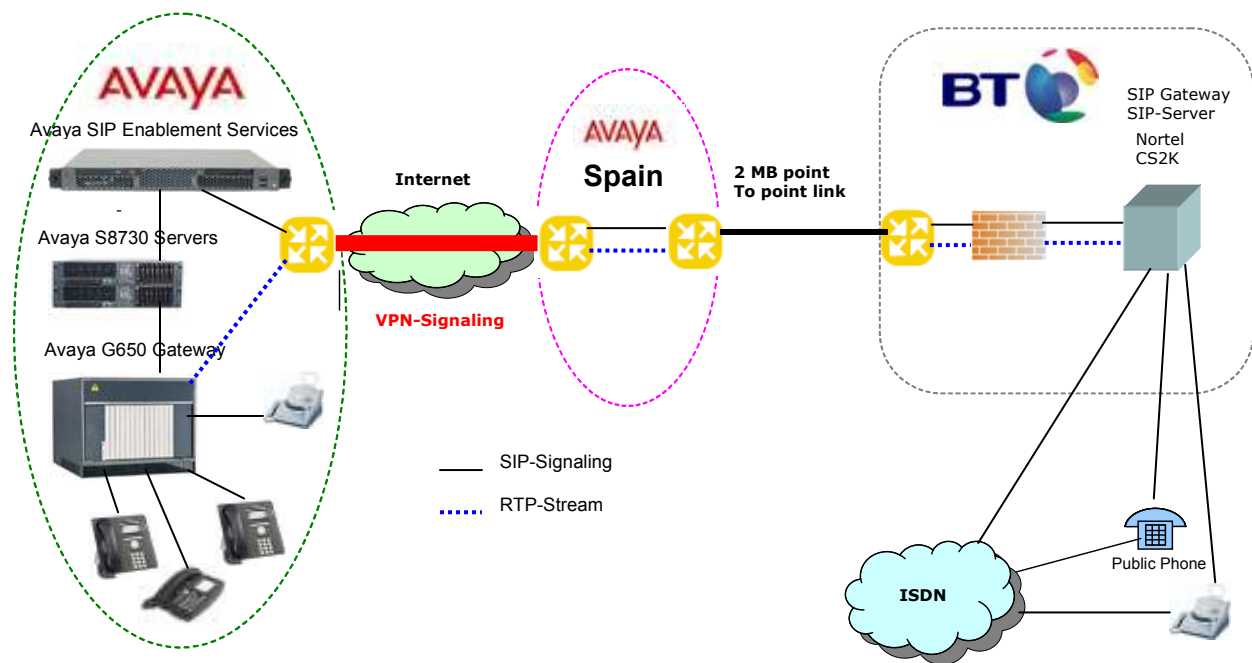
BT Spain: **Phone:** 901111109  
**Phone from out of Spain:** +34 912706009  
**Email:** [901111109.callbt@bt.com](mailto:901111109.callbt@bt.com)  
**Fax:** 912706243  
**Webpage:** <http://www.bt.es>

## 2. SIP Trunking Configuration

**Figure 1** illustrates a sample Avaya SIP telephony solution connected to British Telecom Spain SIP trunking service using SIP trunking. This is the configuration that was used during the DevConnect compliance testing.

The simulated customer configuration included:

- Two Avaya S8730 Media Server with an Avaya G650 Media Gateway. The S8730 served as the processor for Avaya Aura™ Communication Manager.
- Avaya Aura™ SIP Enablement Services operating on a SES Combined Home/Edge server (S8500C).
- Avaya 96xx Series IP telephones using the H.323 / SIP software bundle.
- Avaya Analog phones
- Fax machines



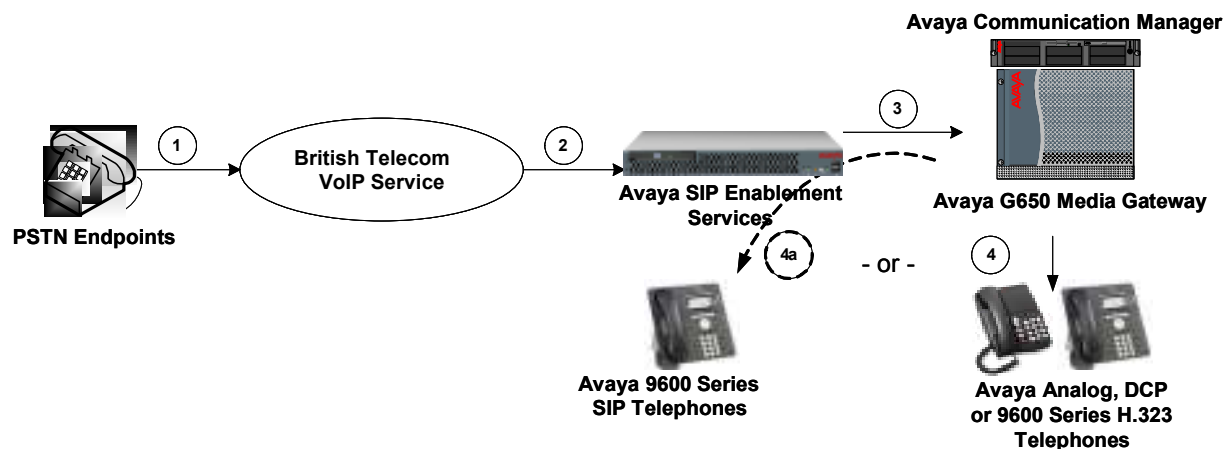
**Figure 1: Avaya Aura™ SIP Telephony Solution using British Telecom Spain SIP trunking service**

## 2.1. Call Flows

To better understand how calls are routed between the BT Spain network and the enterprise site shown in **Figure 1** using SIP trunks, two call flows are described in this section.

The first call scenario illustrated in **Figure 2** is a PSTN call to the enterprise site terminating on a telephone supported by Communication Manager.

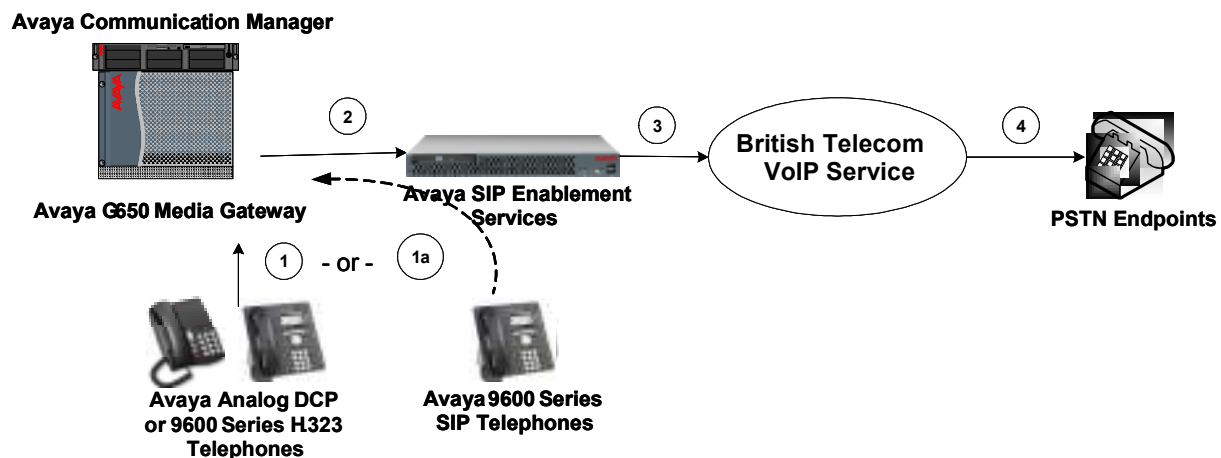
1. A user on the BT Spain network dials a British Telecom Spain SIP trunking service provided number that is assigned to a Communication Manager telephone at the enterprise site.
2. Based on the DID number, the British Telecom Spain SIP trunking service offers the call to SES using SIP signaling messages sent over the managed access facility. Note that the assignment of the number and the address of the SES server were previously established during the ordering and provisioning of the service.
3. SES routes the call to the Avaya S8730 Media Server running Communication Manager over a SIP trunk.
4. Communication Manager terminates the call to an Avaya H.323/Analog/DCP telephone as shown in **Figure 2**.  
- or -  
4.a. Inbound calls destined for a SIP Telephone at the enterprise are routed to Communication Manager. Communication Manager, acting as a SIP back-to-back user agent, then transmits the appropriate SIP signaling via SES to the telephone (as shown by the 4a arrow.)



**Figure 2: Incoming BT Spain network Calls to Avaya Aura™ Communication Manager**

The second call scenario illustrated in **Figure 3** is an outgoing call from an Avaya telephone at the enterprise site to the BT Spain network via the SIP trunk to the British Telecom Spain SIP trunking service.

1. An Avaya H.323, analog or DCP telephone served by Communication Manager originates a call to a user on the BT Spain network.  
- or -
- 1a. An Avaya SIP telephone originates a call that is routed via SES (as shown by the 1a arrow) to Communication Manager.
2. The call request is handled by Communication Manager where origination treatment such as Class of Restrictions (COR) and Automatic Route Selection (ARS) is performed. Communication Manager selects the SIP trunk and sends the SIP signaling messages to SES.
3. SES routes the call to the British Telecom Spain SIP trunking service.
4. The British Telecom Spain SIP trunking service completes the call to the BT Spain network.



**Figure 3: Outgoing Calls from Avaya Aura™ Communication Manager to the BT Spain network**

Appendix A illustrates examples of the SIP INVITE messages sent on the SIP trunk to begin each call.

### 3. Equipment and Software Validated

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

Avaya SIP Telephony Solution Components	
Component	Software Version
Avaya Aura™ Communication Manager on S8730 Media Server	Avaya Aura™ Communication Manager 5.2 R015x.02.0.947.3 Patch 02.0.947.3-17250
Avaya G650 Media Gateway TN2312BP IP Server Interface TN799DP Control-LAN TN2602AP IP Media Processor TN799DP Control LAN TN2302AP IP Media Processor TN793CP Analog Line	HW28 FW044 HW16 FW026 HW02 FW041 HW01 FW026 HW21 FW117 HW07 FW010
Avaya SIP Enablement Services on S8500C	<i>SW release string:</i> SES-5.1.0.0-414.3f <i>SW load:</i> SES05.1-01-0.414.3f
Avaya 9630 H.323 IP Telephones	<i>Application file:</i> ha96xxua2_00.bin <i>Boot file:</i> hb96xxua2_00.bin
Avaya 9630 SIP IP Telephones	<i>Application file:</i> SIP96xx_2_2_0_0.bin <i>Boot file:</i> hb96xxua1_50.bin
Avaya 6221 Analog Telephones	n / a
British Telecom Spain SIP trunking service Components	
Component	Version
Nortel CS2000 Call Agent	ISN10
Nortel CS2000 SST	NGSS10_11Bld_MA4

**Table 1: Equipment and Software used in the tested configuration**

The specific configuration above was used for the British Telecom SIP Spain trunking service interoperability testing. Note that this solution is extensible to other Avaya Media Server and Media Gateway platforms running similar versions of Communication Manager and SIP Enablement Services.

## 4. Configure Avaya Aura™ Communication Manager

This section describes the steps for configuring a SIP trunk on Communication Manager with the necessary signaling and media gateway characteristics for the SIP trunk connection with the British Telecom Spain SIP trunking standard service (UDP).

In the Avaya SIP trunk architecture, the SES acts as a SIP proxy through which all incoming and outgoing SIP messages flow to the British Telecom Spain SIP trunking service. There is no direct SIP signaling path between the British Telecom Spain SIP trunking service and Communication Manager or Avaya SIP endpoints.

The SIP trunk configuration also allows a shared trunking solution that supports SIP endpoint devices (such as Avaya 9600 Series SIP telephones) using Communication Manager in the recommended Avaya Off-PBX Station (OPS) configuration. In this configuration, the SIP endpoints are registered with SES but have calling privileges and features managed by Communication Manager. Communication Manager acts as a Back-to-Back SIP user agent for the SIP endpoint, when a call involves a SIP trunk to the British Telecom Spain SIP Trunk Service.

For incoming calls, SES uses media server address maps to direct the incoming SIP messages to the appropriate Communication Manager. Once the message arrives at the Communication Manager, further incoming call treatment such as incoming digit translations, class of service restrictions, etc., may be performed.

All outgoing calls to the BT Spain network are processed within Communication Manager and are subject to outbound features such as automatic route selection (ARS), digit manipulation and class of service restrictions (COS). Once Communication Manager selects a SIP trunk, the SIP signaling is routed to SES. Within SES, host address maps direct the outbound SIP messages to the British Telecom SIP trunking SIP Application Server. Note that SES acts as a SIP proxy through which all incoming and outgoing SIP messages flow to the British Telecom Spain SIP trunk service.

The Communication Manager configuration was performed using the System Access Terminal (SAT). The initial installation and configuration of the S8730 Media Server, G650 Media Gateway and associated media modules are presumed to have been previously completed and are not discussed within these Application Notes.



## 4.1. SIP Trunk Configuration

### 4.1.1 Verify System Capacity

Using the SAT, verify that sufficient SIP Trunk capacity exists by displaying the **System-Parameters Customer-Options** form shown in **Figure 4** and **Figure 5**. The Communication Manager license file controls the customer options. Contact an authorized Avaya sales representative for assistance if insufficient capacity exists or a required feature is not enabled.

On Page 1 of the **System-Parameters Customer-Options** form, verify that the number of OPS stations available is sufficient for the number of SIP telephones to be used.

display system-parameters customer-options		Page 1 of 11
OPTIONAL FEATURES		
G3 Version: V15	Software Package: Standard	
Location: 2	RFA System ID (SID): 1	
Platform: 6	RFA Module ID (MID): 1	
		USED
	Platform Maximum Ports: 48000	74
	Maximum Stations: 36000	14
	Maximum XMOBILE Stations: 0	0
	Maximum Off-PBX Telephones - EC500: 500	1
	<b>Maximum Off-PBX Telephones - OPS: 500</b>	<b>6</b>
	Maximum Off-PBX Telephones - PBFMC: 0	0
	Maximum Off-PBX Telephones - PVFMC: 0	0
	Maximum Off-PBX Telephones - SCCAN: 0	0
(NOTE: You must logoff & login to effect the permission changes.)		

**Figure 4: System-Parameters Customer-Options Form – Page 1**

On Page 2, verify that the number of available **SIP Trunks** is sufficient for the combination of trunks to the British Telecom Spain SIP trunking service, trunks for SIP endpoints and any other SIP trunking applications.

Note that each call from a non-SIP endpoint to the British Telecom Spain SIP trunking service uses one SIP trunk for the duration of the call. However, due to the SIP OPS configuration, each SIP endpoint on a call with the British Telecom Spain SIP trunking service requires two SIP trunks for the duration of the call.

display system-parameters customer-options		Page 2 of 11
OPTIONAL FEATURES		
IP PORT CAPACITIES		USED
Maximum Administered H.323 Trunks:	100	5
Maximum Concurrently Registered IP Stations:	18000	4
Maximum Administered Remote Office Trunks:	0	0
Maximum Concurrently Registered Remote Office Stations:	0	0
Maximum Concurrently Registered IP eCons:	0	0
Max Concur Registered Unauthenticated H.323 Stations:	0	0
Maximum Video Capable Stations:	0	0
Maximum Video Capable IP Softphones:	0	0
<b>Maximum Administered SIP Trunks:</b>	<b>100</b>	<b>55</b>
Maximum Administered Ad-hoc Video Conferencing Ports:	0	0
Maximum Number of DS1 Boards with Echo Cancellation:	0	0
Maximum TN2501 VAL Boards:	128	0
Maximum Media Gateway VAL Sources:	0	0
Maximum TN2602 Boards with 80 VoIP Channels:	128	0
Maximum TN2602 Boards with 320 VoIP Channels:	128	1
Maximum Number of Expanded Meet-me Conference Ports:	0	0
(NOTE: You must logoff & login to effect the permission changes.))		

**Figure 5: System-Parameters Customer-Options Form – Page 2**

## 4.2. Configure Special Applications

Using the SAT, verify that sufficient SIP Trunk capacity exists by displaying the **System-parameters Special-applications** form shown in **Figure 6**. The Communication Manager license file controls the Special-applications. Contact an authorized Avaya sales representative for assistance if insufficient capacity exists or a required feature is not enabled.

On Page 8 of the **System-parameters Special-applications** form, verify that the feature Overwrite Calling Identity (SA8972) is enabled. When the SA8972 feature is enabled, the outgoing INVITE for a forwarded call (CFU, CFB, CFNA) contains the number of the forwarding station in the FROM header and P-Asserted-Identity. This number is considered valid by the BT Spain Service and the call will be established. Otherwise the forwarded call will have the original called party number, and when the BT service checks the number, it considers it invalid and it drops the call.

The reason why BT does this is because some attributes are extracted from Calling Line Identification to perform several features and in order to enable other applications, like emergency numbering translations, blocking calls to mobile phones, filtering calls, etc. This way of working sometimes disables the usage of advanced Communications Manager features, like Extension to Mobile, some Unified Communications features, Dial Plan Transparency, etc. If the end-customer needs these features, BT considers this a non-standard configuration, and can be handled as a special configuration. One reason for this consideration as a non-standard configuration by BT is because it needs to disable CLI validation, which has implications and legal constraints in different countries, like identification of emergency calls, etc. Contact your BT Spain representative if you encounter this situation.

The following should be noted:

- **On the display of the called party is indicated the number of the forwarding station instead the number of the calling station.**
- **The SA8972 feature doesn't work with EC500. Forwarding calls through EC500 will fail.**

```

display system-parameters special-applications
SPECIAL APPLICATIONS

(SA8972) - Overwrite Calling Identity? y
(SA8983) - Replace CPN when calls forwarded off-net? n
(SA8986) - Simultaneous Ring of Attendants? n
(SA8993) - Increase Tenant Partitions? n
(SA9004) - Multi-Location Call Routing for IP-DECT? n
(SA9009) - Duplicated Processor Ethernet for Sip? n
(SA9014) - PIN Release of TOD Lock? n
(SA9016) - G.726 Support on TN2302? n
(SA9018) - Support 127 Bridge Appearances per Call-appr? n
(SA9020) - IGAR triggered by IP network performance? n
(SA9025) - ISDN Subaddress Treatment? n
(SA9028) - Station TBCT for Heavily Trunked Systems? n
(SA9035) - Increased Intercom Groups? n
(SA9050) - Increased TOD Routing Tables/Partition Grp Num? n
(SA9062) - Permissive Use CMS and EDP? n
(SA9065) - Crisis Alert to Stations by Location? n
(SA9070) - Increased Logged-In ACD Agents for XL? n
(SA9073) - Use Called Party Location For LWC Time/Date? n

```

**Figure 6: System-parameters Special-applications Form – Page 8**

#### 4.2.1 Determine Node Names

In the **IP Node Names** form, view (or assign) the node names to be used in this configuration using the **change node-names ip** command shown in **Figure 7**.

**SES** and **192.168.193.110** are used below as the **Name** and **IP Address** of the SES server at the enterprise site.

```

display node-names ip
IP NODE NAMES

Name          IP Address
Gateway001    192.168.193.1
H323_BT_SPAIN 62.7.40.88
SIP_BT_SPAIN  190.169.40.100
clan-cab-a-2   192.168.193.106
clan-cabinet-a 192.168.193.104
default       0.0.0.0
medpro-cabineta 192.168.193.105
medpro2caba   192.168.193.107
procr         0.0.0.0
ses          192.168.193.110

( 10 of 10   administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name

```

**Figure 7: IP Nodes Names Form**

## 4.2.2 Define IP Codec Sets

British Telecom Spain SIP trunking service supports the codecs G711A, G729 and T38 (Fax).

- Define a codec set (**Codec Set 1** below) to support voice and Fax calls with the British Telecom Spain SIP trunking service. G.729 and G.711A are the codecs to be used.
- The only accepted packet size for BT Spain SIP Trunking service is 20 ms and 2 Frames per packet.

Use the **change ip-codec-set** command (**Figure 8 and 8a**) to define the codec set. On Page 2 of **ip-codec-set** form, the **Fax**, must be **t.38-standard**, **Modem** and **TTD/TTY** modes must be **off**.

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

IP Codec Set

Codec Set: 1

Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size (ms)
1: G.711A	n	2	20
2: G.729	n	2	20
3: G.729A	n	2	20
4: G.729B	n	2	20
5: G.729AB	n	2	20
6:			
7:			

Media Encryption

1: none  
2:  
3:

**Figure 8: IP Codec Set Forms for Voice**

```
display 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	off	0
Clear-channel	n	0

**Figure 8a: IP Codec Set Forms for Fax**

### 4.2.3 Verify Near End IP Network Region

The **IP Network Region** form defines the parameters associated with the SIP trunk group serving the SES proxy (used to reach British Telecom and any optional SIP endpoints). In the **IP Network Region** form shown in **Figure 9**:

- The **Authoritative Domain** field is configured to match the domain name configured on the SES server. In this configuration, the domain name is "*sipdomain.olymp.org*". This field is required for endpoints to call the public network.
- By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between SIP endpoints without using media resources in the Avaya G650 Media Gateway.
- The **Codec Set** is set to the number of the ip-codec-set ("1" for voice and fax) to be used for calls within IP network region 1.<sup>1</sup>

display ip-network-region 1		Page 1 of 19
IP NETWORK REGION		
Region: 1		
Location: 1	Authoritative Domain: sipdomain.olymp.org	
Name: CLAN1		
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	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		

**Figure 9: IP Network Region Form**

There were some issues uncovered during the testing to have in mind when considering shuffling (IP-to-IP direct audio). The CPE (Customer Premises Equipment) router that is delivered as standard by the service provider supports NAT, PAT and ALG features, sorting the problem of NAT traversal (in general), and having the advantage of a customer to not need a Session Border Controller (SBC) equipment to sort this out. On other hand, there are circumstances where there is a need for an SBC that are out of the scope of this document, as those cases are not part of the BT Standard SIP Trunking Service. But to have the customer solution work well in the standard

<sup>1</sup> In this case, the SIP trunk is assigned to the same ip-network-region as the G650 media gateway. If multiple network regions are used, page 3 of each ip-network-region form must be used to specify the codec set for inter-region communications.

service, there are some recommendations BT suggests to be implemented in when designing customer network configuration at the very beginning of the project:

- Every device that uses RTP (i.e., Telephone, IP Media Resources, etc.), to use a fixed IP port or a non overlapping port. For example, assume 2 IP Telephones, everyone with different DHCP IP Addresses but first phone with port number X and the latest with port X+2 (X+1 is normally used for RTCP). Tests showed that with this, there is a low possibility for the trunking to use the same port for different calls, a situation that produces calls with no audio.
- Use a wide range of UDP ports to be assigned to the Network Region. This causes again a low probability for repeating the same port at the same time as mentioned before (i.e., a solution with 20 IP Phones in the same network region, do not use a UDP range of 2048-2060, but 2048-2200 for example).
- BT can provide a set of Public IP Addresses that can be used for static NATting (NOTE that IP Media Resources can be used either in Hairpinning mode for the whole duration of the call or just in the call setup in shuffling mode).
- In case there are audio problems due to port overlapping, Hairpinning for IP Media Resources is recommended to sort this out.
- If you experience other problems, contact your BT Spain representative.

#### 4.2.4 Define SIP Trunk Groups

Configure the **Signaling Group** form shown as follows, shown in **Figure 10**:

- Set the **Group Type** field to *sip*.
- The **Transport Method** field will default to *tls* (Transport Layer Security). TLS is the only link protocol that is supported on a SIP trunk between Communication Manager and SES.
- Specify the Avaya S8730 Media Server CLAN board (node name “clan-cabinet-a”) and the SES Server (node name “ses”) as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** fields, respectively. These field values are taken from the **IP Node Names** form shown in Figure 7<sup>2</sup>.
- Ensure that the recommended TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- Enter the IP Network Region value assigned in the ip-network-region form (**Figure 9**). Note that if the **Far-end Network Region** field is different from the near-end network region, the preferred codec will be selected from the IP codec set assigned for the inter-region connectivity for the pair of network regions. In this case, the same IP network region (Network Region 1) was used for local and BT Spain network calls; however, different network regions can be used in the field.
- Leave the **Far-end-Domain** field blank. When an incoming SIP call is placed to the SES, then SES and Communication Manager will look for a trunk to route the call between them. This is done by looking at the domain/IP address of the SIP URI in the “FROM”-header and then by looking for a match on one of the SIP signaling groups configured on Communication Manager. When the Far-end-Domain field of a trunk is left blank, all calls in with no valid match for the Domain part of the SIP URI, will go over this trunk. (Note: This domain is specified in the Uniform Resource Identifier (URI) of the “SIP To Address” in the INVITE message. Misconfiguring this field may prevent calls from being successfully established to other SIP endpoints or to the PSTN.)
- If calls to/from SIP endpoints are to be shuffled, then the **Direct IP-IP Audio Connections** field must be set to ‘y’.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*. This value is used to have Communication Manager send DTMF transmissions using RFC 2833.
- The default values for the other fields may be used.
- The **Alternate Route Timer** field is set to *10* for establishing calls to international and mobile extensions.

---

<sup>2</sup> For other media server platforms, the near (local) end of the SIP signaling group may be a media server instead of a C-LAN board.



```

display signaling-group 1
                                SIGNALING GROUP

Group Number: 1                Group Type: sip
                                Transport Method: tls
IMS Enabled? n
IP Video? n

Near-end Node Name: clan-cabinet-a    Far-end Node Name: ses
Near-end Listen Port: 5061            Far-end Listen Port: 5061
Far-end Network Region: 1
Far-end Domain:

                                Bypass If IP Threshold Exceeded? 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): 10

```

**Figure 10: Signaling Group Form**

Configure the **Trunk Group** form as shown in **Figure 11** through using the **add trunk-group** command. **Trunk Group** number 12 is defined for using British Telecom SIP trunking service in this configuration.

On Page 1 of the **Trunk Group** form:

- Set the **Group Type** field to **sip**.
- Choose a descriptive **Group Name** such as ***BT Spain UDP***.
- Specify an available trunk access code (TAC) such as ***\*05***.
- Set the **Service Type** field to ***public-ntwrk***.
- Enter **1** as the **Signaling Group** number. This value was previously determined during the **Signaling Group** configuration specified in **Figure 10**.
- Specify the **Number of Members** supported by this SIP trunk group.  
Note that one trunk member is required for each call between a non-SIP endpoint and the British Telecom SIP trunking service. Calls involving a SIP endpoint and the British Telecom SIP trunking service will use two trunk members for the duration of the call.

```

display trunk-group 12                                     Page 1 of 21
                                TRUNK GROUP

Group Number: 12                      Group Type: sip      CDR Reports: y
  Group Name: BT Spain UDP              COR: 1            TN: 1      TAC: *05
    Direction: two-way                Outgoing Display? n
    Dial Access? n                      Night Service:
    Queue Length: 0
  Service Type: public-ntwrk          Auth Code? n

                                Signaling Group: 1
                                Number of Members: 30

```

**Figure 11: Trunk Group Form – Page 1**

On Page 3 of the **Trunk Group** form (**Figure 12**):

- Set the **Numbering Format** field to **public**. This field specifies the format of the calling party number sent to the far-end.

```

display trunk-group 12                                     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

Show ANSWERED BY on Display? y

```

**Figure 12: Trunk Group Form – Page 3**

On Page 4 of the **Trunk Group** form (**Figure 13**):

- Set the **Send Diversion Header** field to **n**. This field specifies the Diversion header at the SIP INVITE. It contains the number of the number of the call forwarding station. BT Spain doesn't support the Diversion header
- Set the **Support Request History** field to **n**. This field specifies the call forwarding history. BT SIP trunking service does not support the SIP Header history-info in SIP messages and has to be disabled.
- Set the **Telephone Event Payload Type** field to **96**. This field specifies the payload for DTMF signaling (RFC2833).

display trunk-group 12	Page 4 of 21
<p style="text-align: center;">PROTOCOL VARIATIONS</p> <p> Mark Users as Phone? n  Prepend '+' to Calling Number? n  Send Transferring Party Information? n  Network Call Redirection? n  <b>Send Diversion Header? n</b>  <b>Support Request History? n</b>  <b>Telephone Event Payload Type: 96</b>    Overwrite Calling Identity? y </p>	

**Figure 13: Trunk Group Form – Page 4**

#### 4.2.5 Configure Calling Party Number Information for outgoing calls

Configure the **public-unknown-numbering** form to send the correct calling party number to the British Telecom Spain SIP trunking service for calls originating from Communication Manager. The calling party number (sent in the SIP “From” header) must match the value expected by the British Telecom Spain SIP trunking service. As mentioned before, if outgoing calls do not go out with the correct number assigned by the BT service, calls will be dropped as they are supposed to be a security violation and an illegal call. BT needs to check that calls made by end-customer have the correct numbers and that they match the range assigned to this customer. Again, if a customer needs advanced Communication Manager features, BT can supply the solution per special request as a bespoke implementation of the SIP Trunking service. In that case a deep solution design analysis will be needed where implementation tests with the customer equipment will be made and additional Spanish legal regulations contracts will have to be signed by the customer and BT to support this way of work.

In these Application Notes, the extensions on Communication Manager use a 3 digit dialing plan with extensions between 200 and 249 that match the DID numbers assigned by the British Telecom Spain SIP trunking service. Note this requirement may vary with other numbering plans provided by British Telecom Spain.

**Figure 14** shows the use of the **change public-unknown-numbering** command to send the correct calling party number. The entries below indicate that 3-digit extension numbers beginning with 2 will send the corresponding digits via trunk group 12.

display public-unknown-numbering 0					Page 1 of 2
NUMBERING - PUBLIC/UNKNOWN FORMAT					
Total			Total		
Ext Len	Ext Code	Trk Grp(s)	CPN Prefix	CPN Len	
					Total Administered: 7
					Maximum Entries: 9999
3	200	12	911779080	9	
3	201	12	911779081	9	
3	203	12	911779086	9	
3	210	12	911779082	9	
3	211	12	911779083	9	
3	220	12	911779084	9	
3	230	12	911779085	9	

**Figure 14: Public/Unknown Numbering Form**

## 4.2.6 Configure Call Routing

### 4.2.6.1 Outbound Calls

In these Application Notes, the Automatic Route Selection feature is used to route calls via the SIP trunk to the British Telecom Spain SIP trunking service, which in turn completes the calls to the BT Spain network destinations.

Use the **change dialplan analysis** command to add **0** as a feature access code (**fac**).

display dialplan analysis							Page 1 of 12		
DIAL PLAN ANALYSIS TABLE									
							Percent Full: 0		
Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	
0	1	fac							
1	1	fac							
2	3	ext							
3	4	ext							
4	3	ext							
61	3	ext							
7	3	ext							
*	3	dac							
*6	3	fac							
#	3	dac							
#6	3	fac							

**Figure 15: Dialplan Analysis Form**

Use the **change feature-access-codes** command to specify **0** as the access code for external dialing (ARS ACCESS CODE).

```
display feature-access-codes                                     Page 1 of 8

                        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:
Attendant Access Code:
Auto Alternate Routing (AAR) Access Code:
Auto Route Selection (ARS) - Access Code 1: 0      Access Code 2: 1
Automatic Callback Activation:                      Deactivation:
Call Forwarding Activation Busy/DA:                All: *10      Deactivation: #10
Call Forwarding Enhanced Status:                   Act:          Deactivation:
Call Park Access Code:
Call Pickup Access Code:
CAS Remote Hold/Answer Hold-Unhold Access Code:
CDR Account Code Access Code:
Change COR Access Code:
Change Coverage Access Code:
Conditional Call Extend Activation:                 Deactivation:
Contact Closure Open Code:                         Close Code:
```

**Figure 16: Dialplan Analysis Form**

Now use the **change ars analysis** command to configure the route pattern selection rule based upon the number dialed following the dialed digit “0”. Note that further administration of ARS is beyond the scope of these Application Notes but is discussed in References [1] and [2].

```
display ars analysis

                        ARS DIGIT ANALYSIS REPORT

                        Location: all          Percent Full: 0

Dialed      Total   Route   Call   Node   ANI
String      Min   Max   Pattern   Type   Num   Req'd

00          7     14     2       pubu       n
8           3     9      2       pubu       n
9           3     9      2       pubu       n
```

**Figure 17: ARS Analysis Form**

Use the **change route-pattern** command to define the SIP trunk groups included in the route pattern that ARS selects.

display route-pattern 2														Page 1 of 3				
Pattern Number: 2														Pattern Name: SIP BT Spain				
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:	12	1											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	n	n	n	n	n	n							rest	none			
3:	y	n	n	n	n	n	n							rest	none			
4:	y	n	n	n	n	n	n							rest	none			
5:	y	y	y	y	y	n	n							rest	none			
6:	y	y	y	y	y	n	n							rest	none			

**Figure 18: Route Pattern Form**

#### 4.2.6.2 Incoming Calls

This step configures the mapping of incoming DID numbers to the proper extensions.

In this configuration, the British Telecom Spain SIP trunking service sends exactly 9 digits in the incoming INVITE message. The incoming digits received are matched to the intended extension. Therefore the incoming call handling treatment is used to modify the number in order to reach the desired extension. The example below illustrates the technique to assign the incoming DID numbers beginning with 911779080 to the corresponding 200.

Use the **change inc-call-handling-trmt trunk-group** command to administer this assignment. For this example shown in **Figure 19**:

- Enter **9** into the **Number Len** field to match the length of the incoming digits.
- Enter **911779080** into the **Number Digits** field as the digit pattern to be matched.
- Enter **9** into the **Del** field as the number of digits that should be deleted from the end of the incoming digits.
- Enter **200** into the **Insert** field as the digits that should be appended to the end of the adjusted incoming digit string (following the deletion step above).

display inc-call-handling-trmt trunk-group 1					Page 1 of 30
INCOMING CALL HANDLING TREATMENT					
Service/ Feature	Number Len	Number Digits	Del	Insert	
public-ntwrk	9	911779080	9	200	
public-ntwrk	9	911779081	9	201	
public-ntwrk	9	911779082	9	210	
public-ntwrk	9	911779083	9	211	
public-ntwrk	9	911779084	9	220	
public-ntwrk	9	911779085	9	230	
public-ntwrk	9	911779086	9	200	
public-ntwrk	9	911779087	9	210	
public-ntwrk	9	911779088	9	210	
public-ntwrk	9	911779089	9	240	

**Figure 19: Incoming Call Handling Treatment – Simple Extension Mapping**

#### 4.2.7 Save Avaya Aura™ Communication Manager Changes

Enter the **save translation** command to make the changes permanent.

### 4.3. SIP Endpoint Configuration

This section describes the administration of optional SIP endpoints and requires the preceding SIP Trunk configuration been completed beforehand. SIP endpoints are optional, but not required, to use the British Telecom Spain SIP Trunk Service.

#### Step 1:

The first step in adding an off-PBX station (OPS) for Avaya SIP telephones registered with SES is to assign a station as shown in **Figure 20, 21 and 22**.

Using the **add station** command from SAT using the station's desired extension as parameter:

- Set the station **Type** at the default **9630** value.
- The **Port** field will default to a system specified value.
- Enter a **Name** for the station that will be displayed.
- As **Security Code** was "123456" entered but normally it could be left blank for SIP OPS extensions.

The remaining fields are configured per normal station administration that is beyond the scope of these Application Notes. Note that the Class of Restrictions (**COR**) and Class of Service (**COS**) will govern the features and call restrictions that apply to this station.

add station 210		Page 1 of 5
STATION		
Extension: 210	Lock Messages? n	BCC: 0
Type: 9630	Security Code: 123456	TN: 1
Port: S00005	Coverage Path 1:	COR: 1
Name: 210	Coverage Path 2:	COS: 1
	Hunt-to Station:	
STATION OPTIONS		
Loss Group: 19	Time of Day Lock Table:	
	Personalized Ringing Pattern: 1	
	Message Lamp Ext: 200	
Speakerphone: 2-way	Mute Button Enabled? y	
Display Language: english	Button Modules: 0	
Survivable GK Node Name:		
Survivable COR: internal	Media Complex Ext:	
Survivable Trunk Dest? y	IP SoftPhone? n	
	IP Video? n	
	Customizable Labels? y	

**Figure 20: Add Station – Page 1**



On Page 2 of the station form,

- Verify that the **Per Station CPN – Send Calling Number** field is set to **y** or blank to allow calling party number information to be sent to the far-end when placing outgoing calls from this station. The default value for this field is blank. The **Direct IP-IP Audio Connections?** field should be set to **y**. If there are shuffling problems set the **Direct IP-IP Audio Connections?** to **n**.

add station 210		Page 2 of 5
STATION		
FEATURE OPTIONS		
LWC Reception: spe	Auto Select Any Idle Appearance? n	
LWC Activation? y	Coverage Msg Retrieval? y	
LWC Log External Calls? n	Auto Answer: none	
CDR Privacy? n	Data Restriction? n	
Redirect Notification? y	Idle Appearance Preference? n	
Per Button Ring Control? n	Bridged Idle Line Preference? n	
Bridged Call Alerting? n	Restrict Last Appearance? y	
Active Station Ringing: single		
	EMU Login Allowed? n	
H.320 Conversion? n	<b>Per Station CPN - Send Calling Number? y</b>	
Service Link Mode: as-needed	EC500 State: disabled	
Multimedia Mode: enhanced	Audible Message Waiting? n	
MWI Served User Type:	Display Client Redirection? n	
AUDIX Name:	Select Last Used Appearance? n	
	Coverage After Forwarding? s	
	<b>Direct IP-IP Audio Connections? y</b>	
Emergency Location Ext: 200	Always Use? n	IP Audio Hairpinning? n

**Figure 21: Add Station – Page 2**

On Page 4 of the station form, configure the appropriate number of call appearances for the SIP telephone.

display station 210		Page 4 of 5	
STATION			
SITE DATA			
Room:		Headset?	n
Jack:		Speaker?	n
Cable:		Mounting:	d
Floor:		Cord Length:	0
Building:		Set Color:	
ABBREVIATED DIALING			
List1:	List2:	List3:	
BUTTON ASSIGNMENTS			
1: call-appr	5: cpn-blk		
2: call-appr	6: ec500	Timer?	n
3: call-appr	7:		
4: cfwd-enh Ext:	8:		
voice-mail Number:			

**Figure 22: Add Station– Page 3**

## Step 2:

The second step of configuring an off-PBX station is to configure the **Off-PBX Telephone** form so that calls destined for a SIP telephone at the enterprise site are routed to SES, which will then forward the call to the intended SIP telephone.

On the **Off-PBX-Telephone Station-Mapping** form shown in **Figure 23**:

- Specify the extension of the SIP endpoint, as shown in **Figure 20**.
- Set the **Application** field to **OPS**.
- Set the **Phone Number** field to the digits to be sent over the SIP trunk. In this case, the SIP telephone extensions configured for SES also match the extensions of the corresponding stations on Avaya Aura™ Communication Manager
- Set the **Trunk Selection** field to **1**. Note that a separate SIP trunk (Trunk Group 1) was configured to support SIP endpoints. Figure 11: Trunk Group Form – Page 1

display off-pbx-telephone station-mapping 210					Page	1 of 2
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION						
Station	Application	Dial	CC	Phone Number	Trunk	Configuration
Extension		Prefix			Selection	Set
210	OPS	-		210	1	1

**Figure 23: Stations with Off-PBX Telephone Integration – Page 1**

On Page 2, set the **Call Limit** field to the maximum number of calls that may be active simultaneously at the station. In this example, the call limit is set to **3**, which corresponds to the number of call appearances configured on the station form. Accept the default values for the other fields.

display off-pbx-telephone station-mapping 210					Page	2 of 2
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION						
Station	Appl	Call	Mapping	Calls	Bridged	Location
Extension	Name	Limit	Mode	Allowed	Calls	
210	OPS	<b>3</b>	both	all	none	

**Figure 24: Stations with Off-PBX Telephone Integration – Page 2**

**Step 3:**

Repeat steps 1 and 2 for each SIP Telephone to be added.

**Step 4:**

Perform a **save translation** to make the changes permanent.

## 5. Configure Avaya SIP Enablement Services

This section covers the administration of SIP Enablement Services (SES). SES is configured via an Internet browser using **SIP Server Management** screens. It is assumed that SES software and the license file have already been installed. During the software installation, the install script is run on the Linux shell of the server to specify the IP network properties of the server including DNS server address(es). For additional information on these installation tasks, refer to Reference [4].

This section is divided into two parts: section 5.1 provides the steps necessary to configure SIP trunking to the British Telecom Spain SIP trunking service. Section 5.2 provides the steps necessary to complete the administration for optional SIP endpoints in this Avaya SIP telephony solution. SIP endpoints are not necessary to use the British Telecom Spain SIP trunking service.

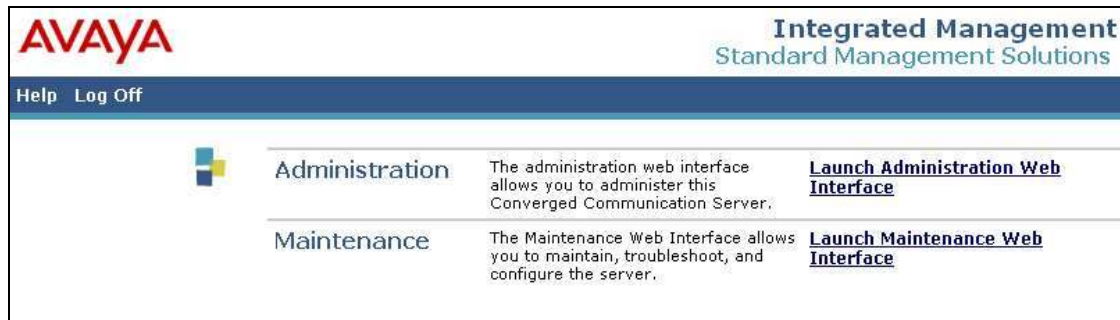
### 5.1. SIP Trunking to the British Telecom Spain SIP trunking service

The “standard” BT Spain SIP Trunking service uses UDP as the transport protocol. Other transport protocols can be used and requested in a bespoke manner. Contact your BT representative for more information.

### 5.1.1 Log in to Avaya SIP Enablement Services

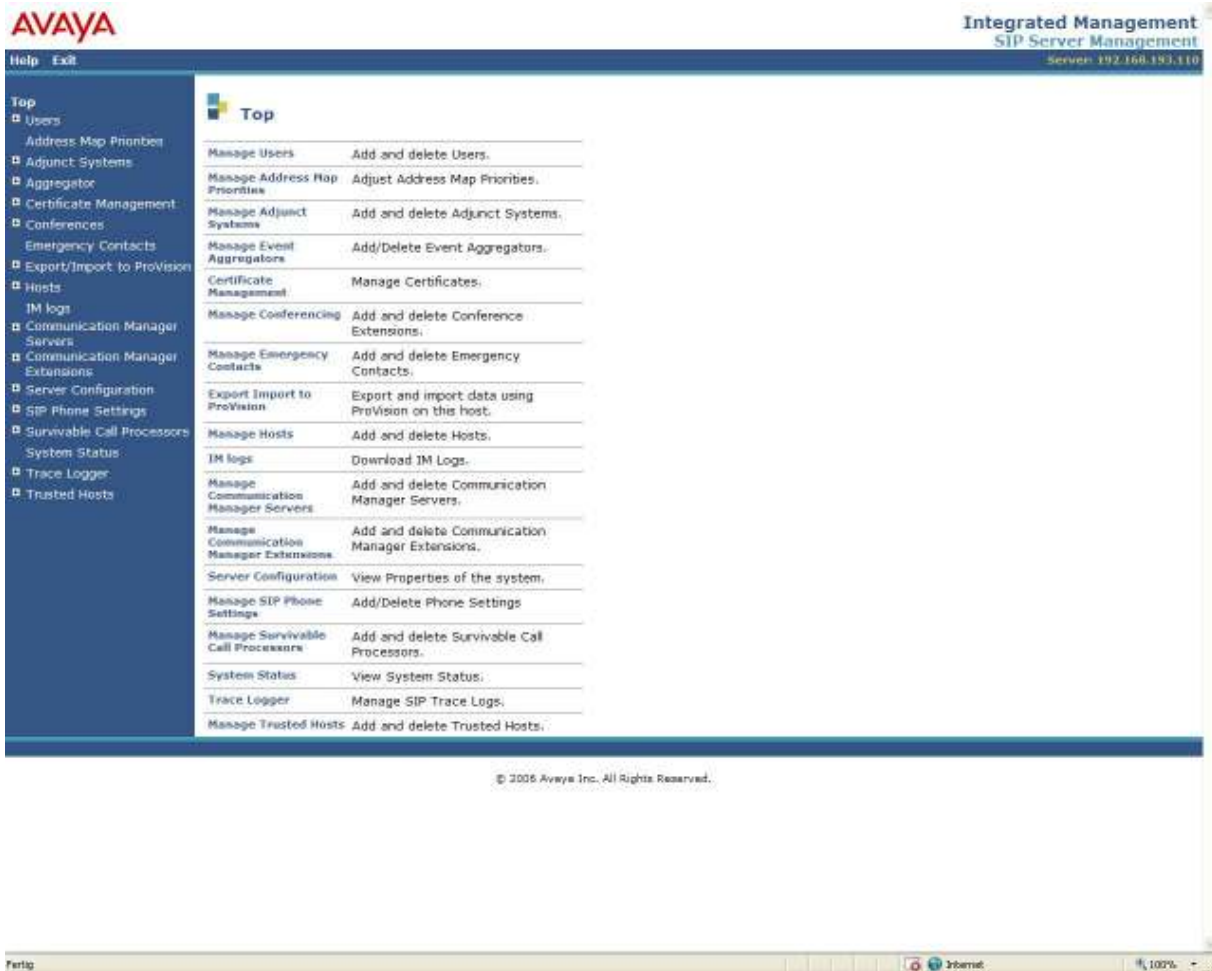
Access the SES SIP Server Management pages by entering **https://<ip-addr>/admin** as the URL in an Internet browser, where <ip-addr> is the IP address of the SES server.

Log in with the appropriate credentials and then select the **Launch Administration Web Interface** link from the main page as shown in **Figure 25**.



**Figure 25: Avaya Aura™ SES Main Page**

The SES administration home page shown in **Figure 26** is displayed.



**Figure 26: SES Administration Home Page**

## 5.1.2 Verify System Properties

From the left pane of any **SIP Server Management** page, expand the **Server Configuration** option and select **System Properties**. This page displays the **SES Version** and the **Network Properties** entered via the install script during the installation process.

In the **Edit System Properties** page note the **SIP Domain** entered. The **SIP Domain** **sipdomain.olymp.org** is used in these Application Notes (see **Figure 9**).

**AVAYA** Integrated Management  
SIP Server Management  
Server: 192.168.193.110

Help Exit

**Top**

- Users
- Address Map Priorities
- Adjunct Systems
- Aggregator
- Certificate Management
- Conferences
- Emergency Contacts
- Export/Import to ProVision
- Hosts
- IM logs
- Communication Manager Servers
- Communication Manager Extensions
- Server Configuration**
  - Admin Setup
  - IM Log Settings
  - License
  - SNMP Configuration
  - System Properties**
  - SIP Phone Settings
  - Survivable Call Processors
  - System Status
  - Trace Logger
  - Trusted Hosts

**View System Properties**

SES Version: SES-5.1.0.0-414.3f  
System Configuration: Simplex  
Host Type: SES, combined home-edge

SIP Domain\*:

Note that the DNS domain is olymp.org

If you are unsure about this field, most often the SIP domain should be the root level DNS domain. For example, for a DNS domain of eastcoast.example.com, the SIP domain would likely be configured to example.com. This allows SIP calls and instant messages to users with handles of the format handle@example.com

SIP License Host\*:

**DiffServ/TOS Parameters**

Cell Control PHB Value\*:

**802.1 Parameters**

Priority Value\*:

Management System Access Login:

Management System Access Password:

DB Log Level:

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**Figure 27: System Properties**

### 5.1.3 Verify the Avaya SES Host Information

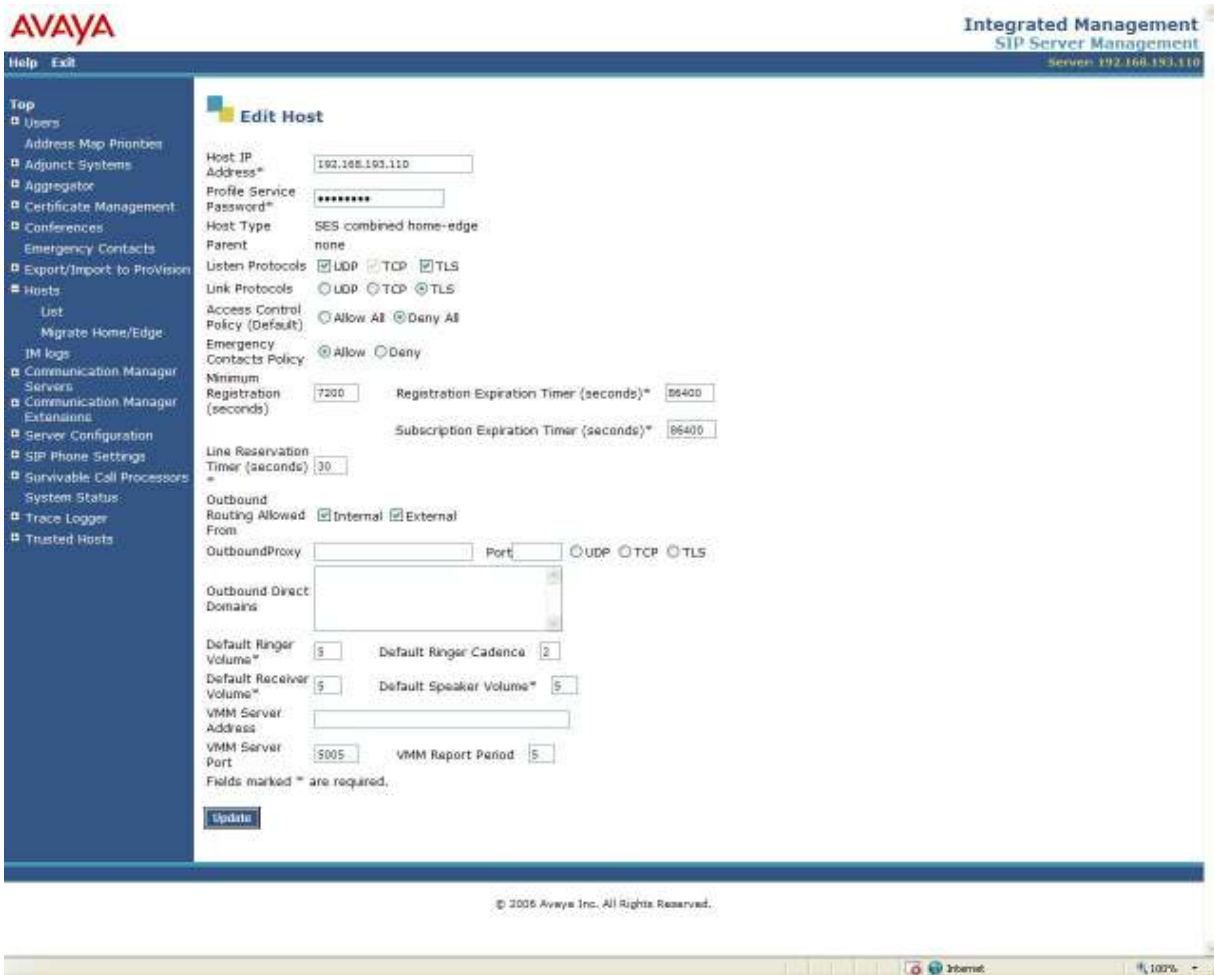
Verify the SES Host information using the **Edit Host** page. In these Application Notes the SES **Host Type** is a combined **home/edge**. This means that the both the British Telecom Spain SIP trunking service and Communication Manager are contacting the same SES.

If the SES **Host Type** is a separate home and edge SES, the British Telecom Spain SIP trunking service and Communication Manager are contacting the home SES.

Display the **Edit Host** page (**Figure 28**) 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.

On the **Edit Host** screen:

- Verify that the IP address of this combined SES Home/Edge server is in the **Host IP Address** field.
- Do not modify the **DB Password** or **Profile Service Password** fields. If these fields are changed, exit the form without using the **Update** button. These values must match the values entered during the SES installation; incorrect changes may disable the SES.
- 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 British Telecom SIP trunking service SIP Application Server (e.g., **BT-sip.net**) is not in the **Outbound Proxy** or **Outbound Direct Domains** fields.
- Default values for the remaining fields may be used.
- After completing the **Edit Host** screen, click on the **Update** button.



**Figure 28: Edit Host**



### 5.1.4 Add Communication Manager Media Server Interfaces

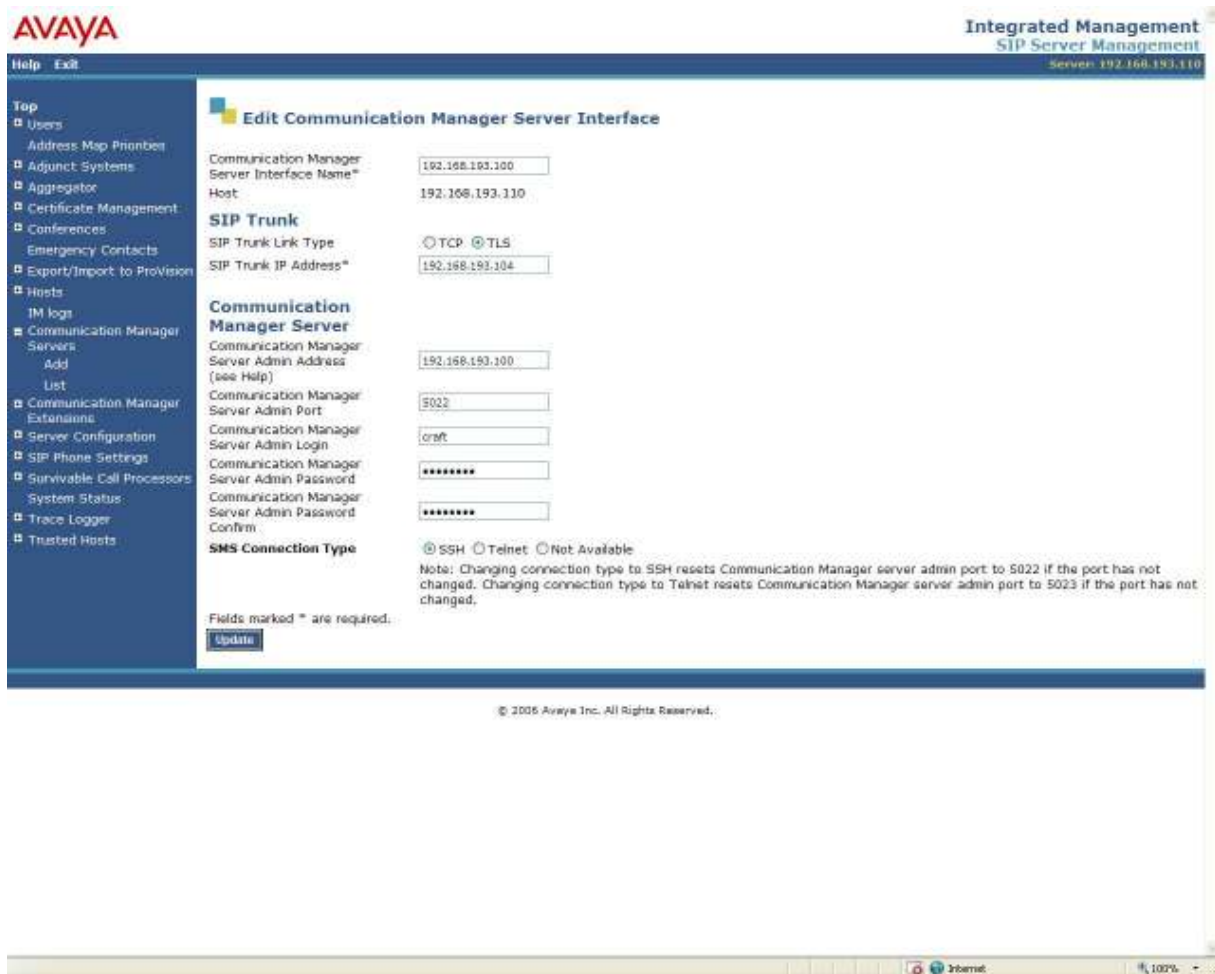
Under the **Communication Manager Servers** option in the Administration web interface, select **Add** to add the Avaya Media Server in the enterprise site. This will create the SES side of the SIP trunk previously created in Communication Manager.

In the **Add Communication Manager Server Interface** screen, enter the following information:

- A descriptive name in the **Media Server Interface** field (e.g., *procr*).
- Select the SES home/edge IP address in the **Host** field.
- Select **TLS** (Transport Link Security) for the **Link Type**. TLS provides encryption at the transport layer. TLS is the only link protocol that is supported for SIP trunking between SES and Communication Manager.
- Enter the IP address of the C-LAN board in the **SIP Trunk IP Address** field.<sup>3</sup>
- After completing the **Add Media Server** screen, click on the **Update** button.

---

<sup>3</sup> Depending on the platform of the media server and gateway, this field may be set to the IP address of a C-LAN board or a media server.



**Figure 29: Add Media Server Interface**

### 5.1.5 Configure Call Routing

SES functions as a SIP proxy server for the SIP trunking with the British Telecom Spain SIP trunking service. In this role, for outbound calls the SES must direct SIP messages originating from Communication Manager to the British Telecom Spain SIP Application Server. In a similar manner for incoming DID calls, the SES must route messages received from the British Telecom Spain SIP Application Server to the proper signaling interface on Communication Manager.

In these Application Notes, the SIP message routing will be done for both outbound and inbound calls using Address Maps that examine some or all of the *called number* (using a Pattern) and route to a specific predetermined destination (called a Contact). The outbound proxy and direct domain routing feature is not used due to interactions with the trusted host capabilities.

The *called number* is contained within the *user* part of the Uniform Resource Identifier (URI) of an incoming SIP INVITE message. The URI usually takes the form of *sip:user@domain*, where

*domain* can be a fully qualified domain name or an IP address. The *user* part for SIP trunking in these Application Notes will only contain digits<sup>4</sup>.

The Address Map Patterns are specified using Linux regular expression syntax. Patterns are generally designed to match a collection of *called numbers* that require identical SIP message routing. However, each Pattern must also be specific enough to direct each unique *called number* to the proper signaling Contact. The Address Map Patterns must also be mutually exclusive (non-overlapping) from all other Address Map Patterns used in the SES to ensure proper operation.

**Appendix B** provides a detailed description of the Linux regular expression syntax used within the address map patterns.

#### 5.1.5.1 Outbound PSTN Calls

SIP signaling for outbound calls is directed to SES based upon rules configured for Automatic Route Selection in Communication Manager. The choice of trunk group (made within ARS) determines the codec that will be used for voice (or fax or modem) calls. For outbound calls, no further differentiation of voice vs. fax / modem is required within the SES address maps.

Outbound calls require SIP signaling messages to be routed to the British Telecom Spain SIP trunking SIP Application Server using Host Address Maps within SES. Calls matching Host Address Map Patterns will be directed to the British Telecom Spain SIP trunking SIP Application Server contained within the corresponding Contact information.

More specific pattern matches would be used if necessary to selectively route SIP traffic to different destinations (such as multiple service providers serving different geographic regions). Also note that a user dialed access code (such as “9” or “0” to place a PSTN or ISDN call) has been previously deleted (by ARS) prior to seizing the outbound SIP trunk.

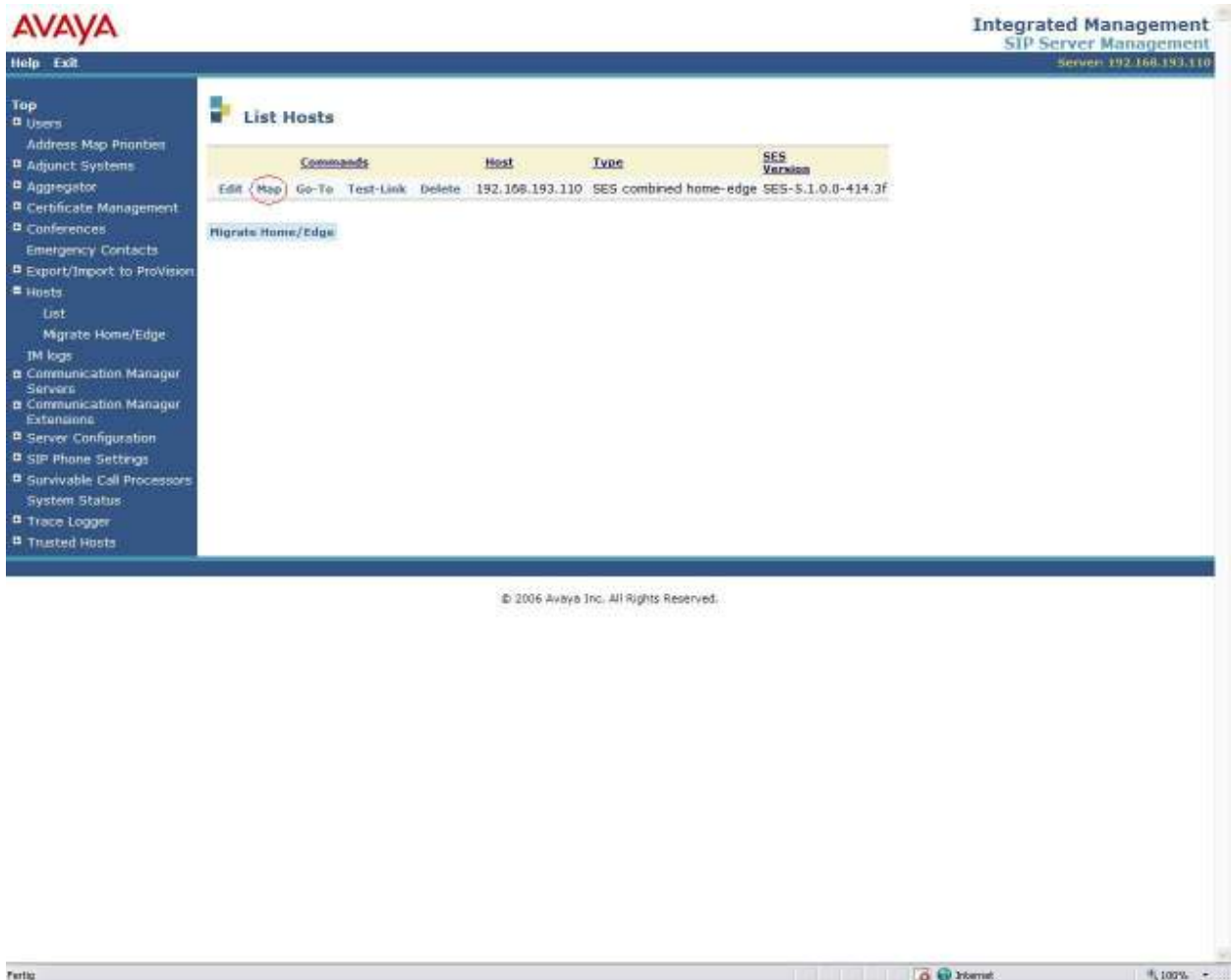
##### 5.1.5.1.1 Outbound Routing - Host Maps

Begin the outbound routing configuration by navigating to the **Add Host Address Map** pages.

- From any **SIP Server Management** page, expand the **Hosts** link and choose the **List** link causing the **List Host** page to appear.
- Select the **Map** link on the **List Host** page causing the **List Host Address Map** page to appear (**Figure 30** ).

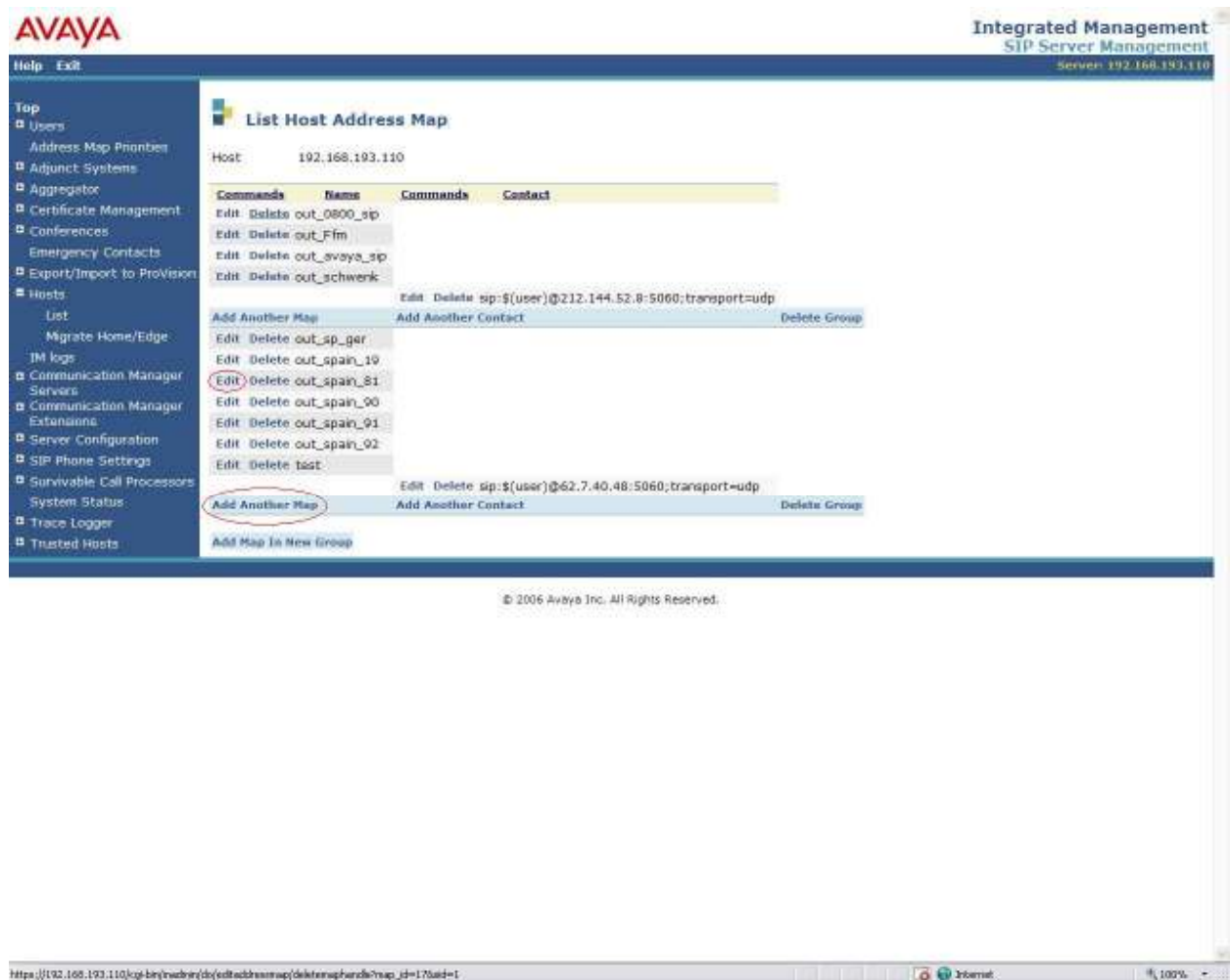
---

<sup>4</sup> SIP does permit mnemonic addressing such as “sip:john.doe@customer.com”. However, this convention is not used in these Application Notes for SIP Trunking. Further discussion of this topic is beyond the scope of this document.



**Figure 30: Accessing Host Address Maps from List Hosts Screen**

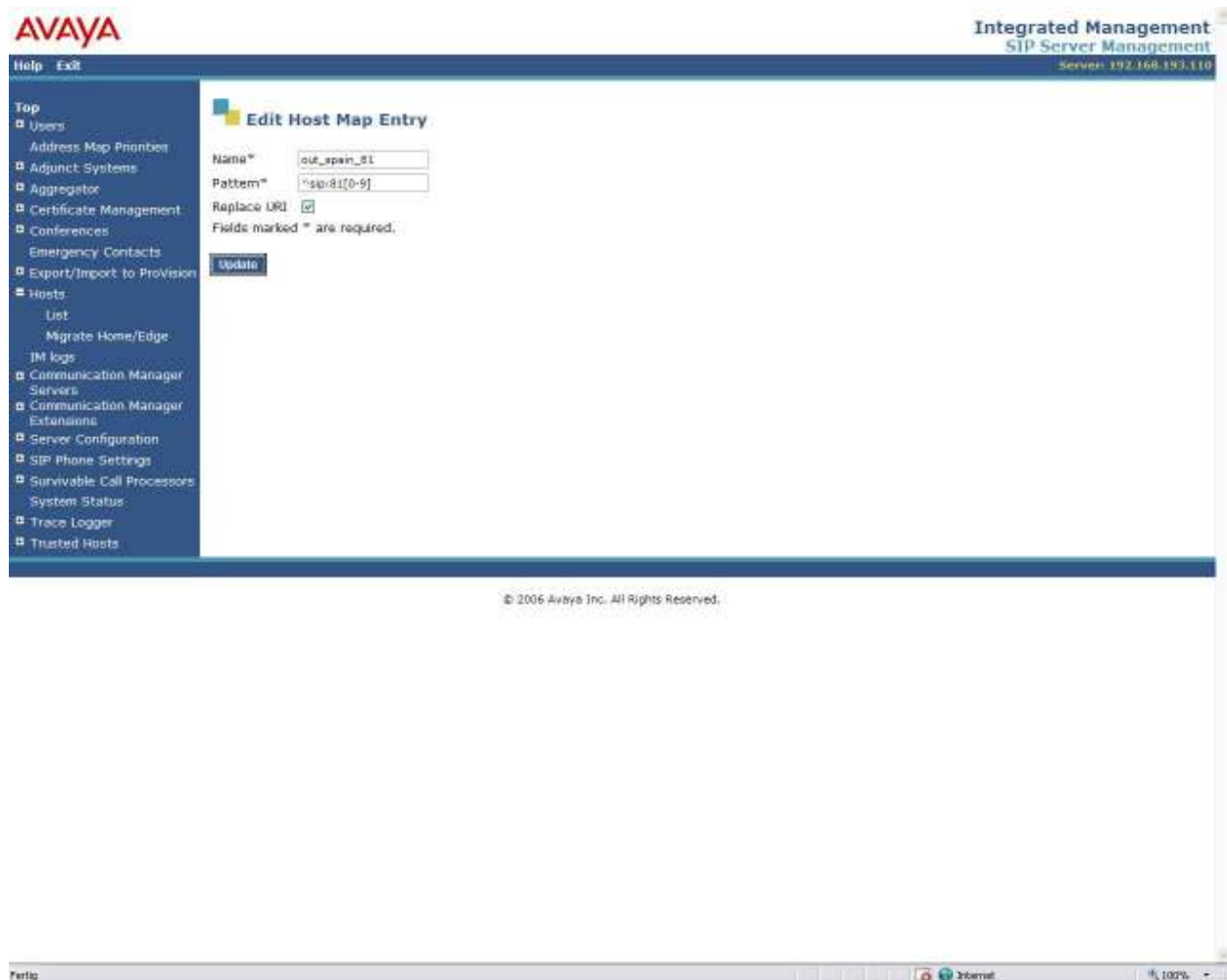
- Select the **Add Map In New Group** link or **Edit** link on the **List Host Address Map** page (**Figure 30**). This will display the **Add Host Address Map** screen (**Figure 31**).



**Figure 31: Adding a Host Address Map Group**

The configuration of the **Host Address Map** for a call to an extension in BT Spain network is shown in **Figure 32**.

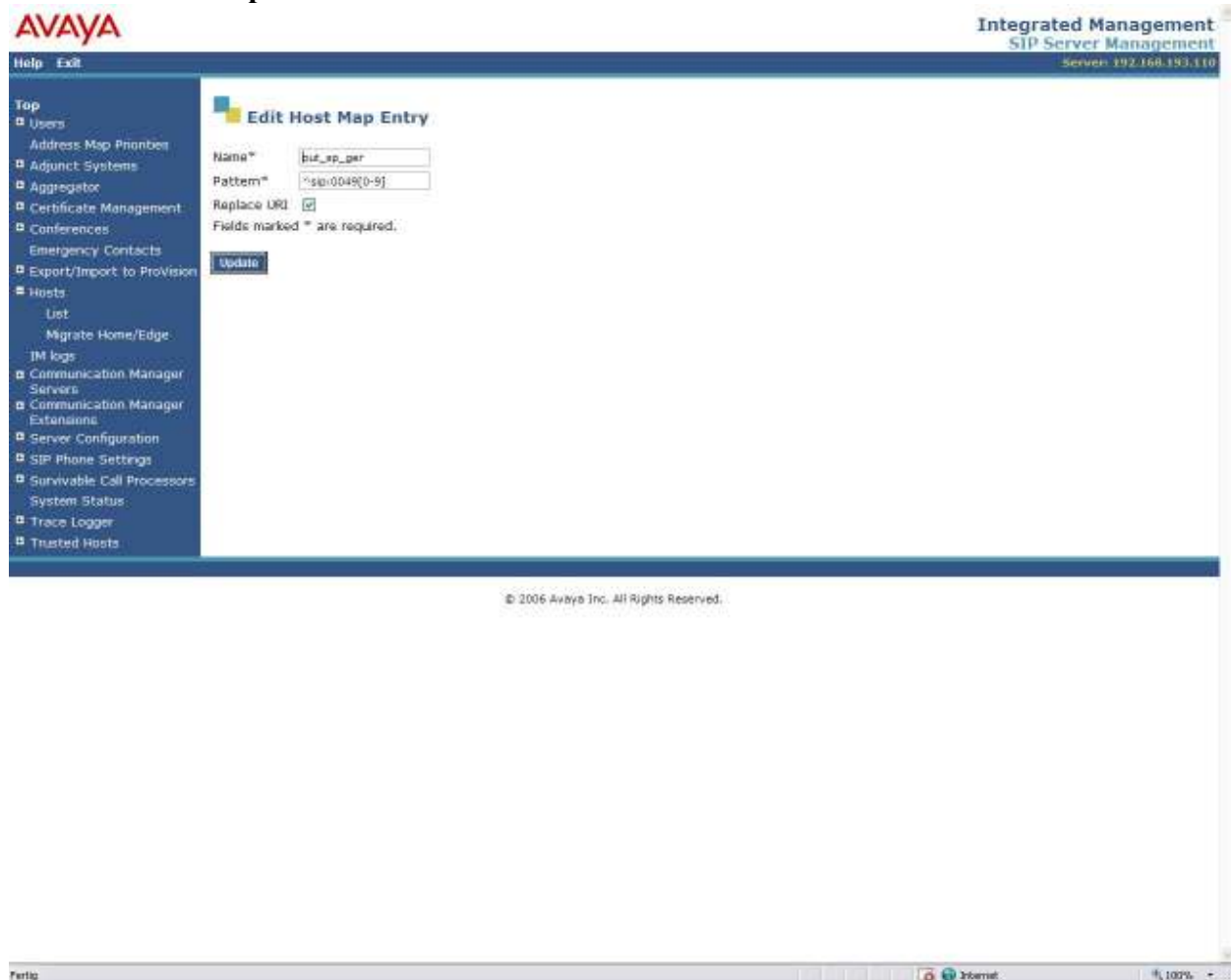
- Enter a descriptive **Name** for the map, such as **out\_spain\_81**.
- Enter the appropriate pattern for the call type. In this example, the pattern is used for calls to an extension in BT Spain network: **^sip:81[0-9]**.
- Select the **Replace URI** checkbox.
- Click the **Update** button.



**Figure 32: Address Map for an extension in BT network**

The configuration of the **Host Address Map** for calls to extensions in Germany is shown in **Figure 33**.

- Enter a descriptive **Name** for the map, such as **out\_sp\_ger**. (“*outgoing calls to Germany via BT trunk*”).
- Enter the appropriate pattern for the call type. In this example, the pattern is used for extensions of Avaya in Frankfurt: **^sip:0049[0-9]**.  
Select the **Replace URI** checkbox.
- Click the **Update** button.



**Figure 33: Address Map for extensions in Germany**

Note that in these Application Notes some other Maps were used, that are not explained here in detail.

#### 5.1.5.1.2 Outbound Routing–BT SIP trunking service Gateway

The next step is to enter the **Contact** information for the British Telecom Spain SIP trunking SIP Application Server. For these Application Notes, the test environment IP address 62.7.40.48 was used. The correct IP Address will be provided by your BT representative to ensure the solution have correct routing and redundancy options.

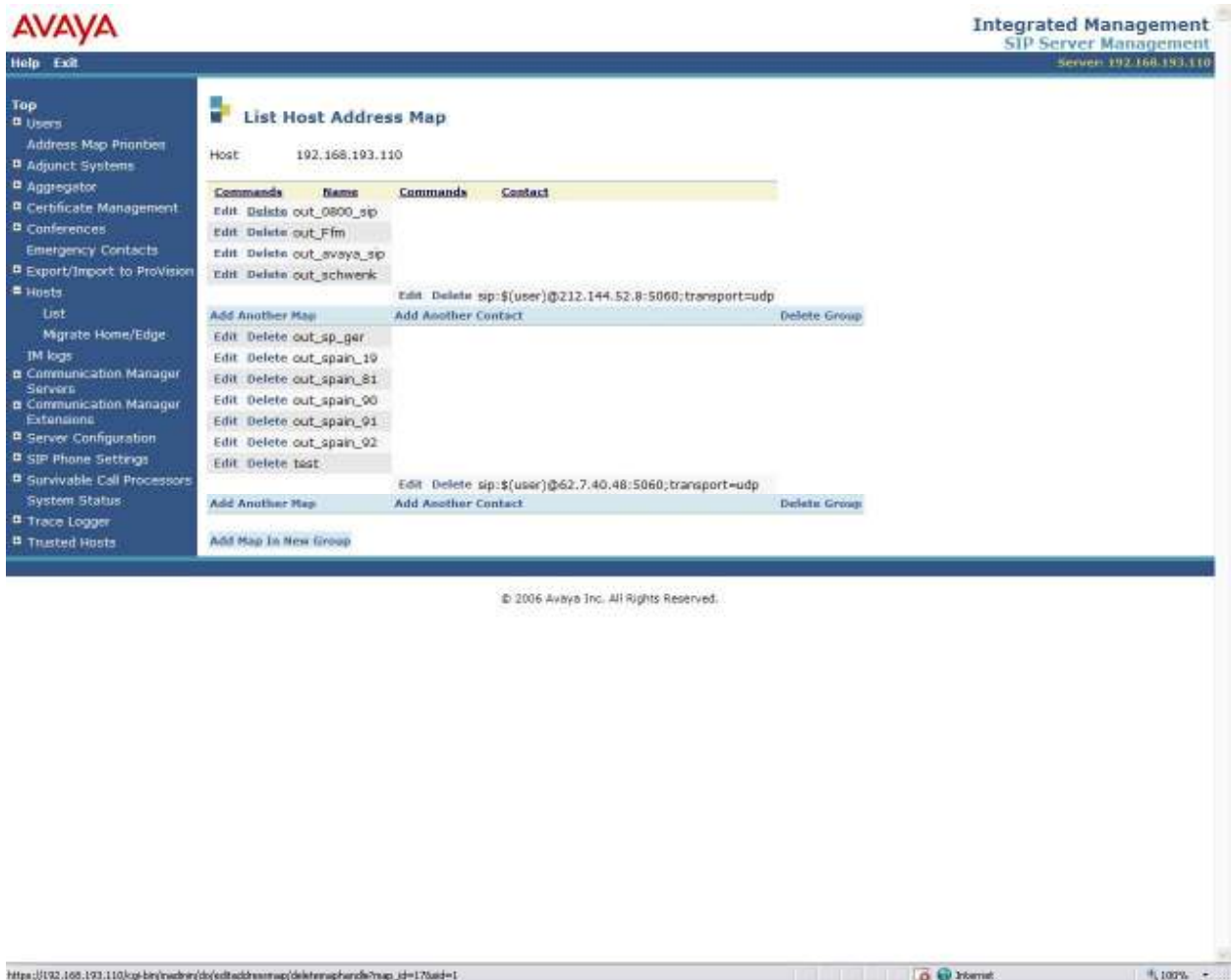
To enter the British Telecom Spain SIP trunking SIP Application Server information:

- Access the **Host Address Map** page by expanding the **Hosts** link in the left pane of any **SIP Server Management** page, selecting **List** and then clicking on the **Map** link associated with the appropriate **Host**.

The **List Host Address Map** page is displayed as shown in **Figure 34**. Notice that the **Contact** information is blank when first creating this entry.

Note: Should an entry already exist due to prior administration, the entry should be edited or deleted instead of using **Add Another Contact**.





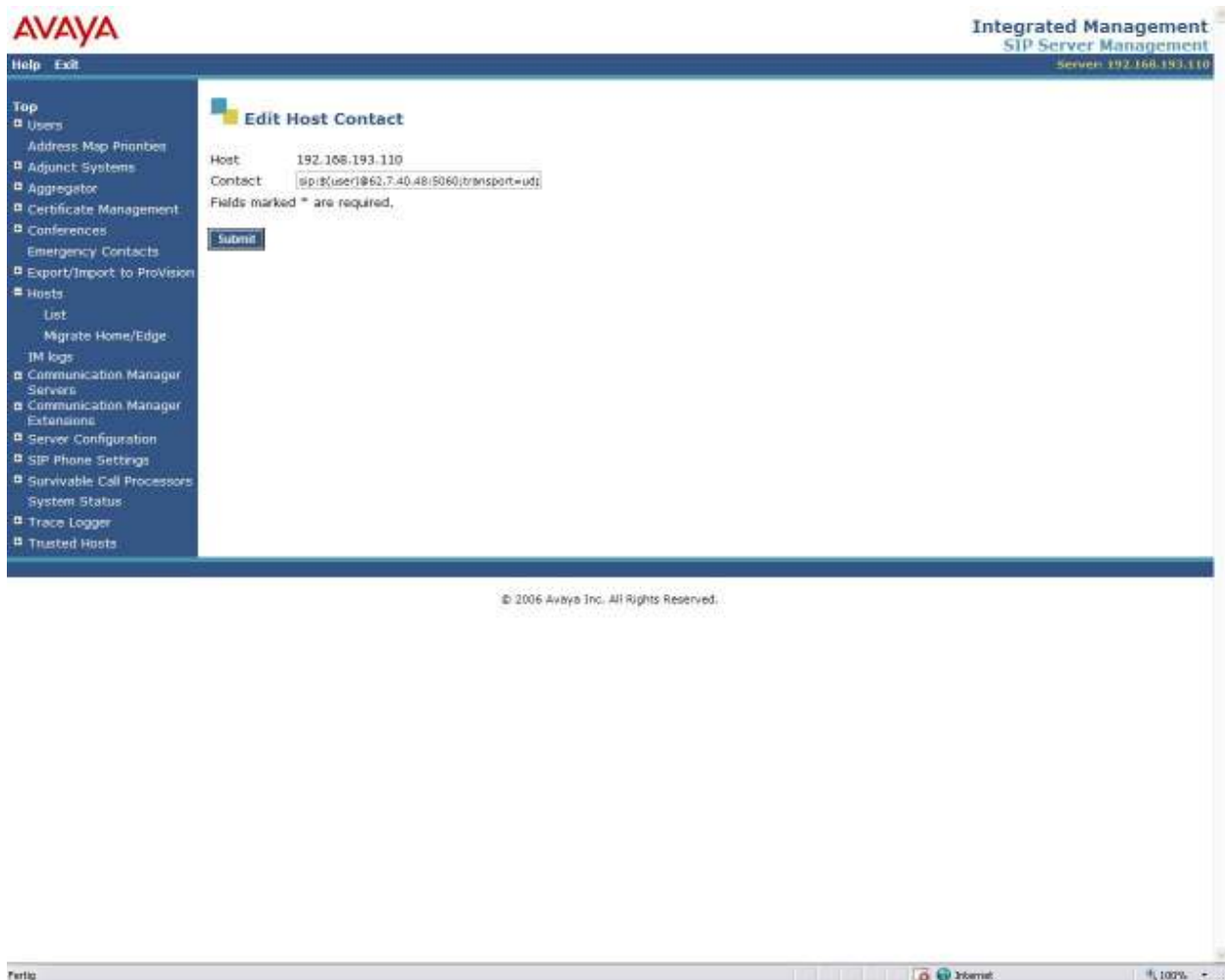
**Figure 34: List Host Address Map**

- Click on the **Add Another Contact / Edit Host Contact** link to open the **Add Host Contact / Edit Host Contact** page (Figure 35).
- In the **Add/Edit Host Contact** page, the **Contact** field specifies the destination for the call. In these Application Notes the Contact field is entered as:

**sip:\$(user)@62.7.40.48:5060;transport=udp**

Note: the IP address is delivered by British Telecom Spain

- Click the **Submit** button when completed.



**Figure 35: Edit (/Add) Host Contact Entry**

### 5.1.5.2 Inbound Direct Inward Dialed Calls

SIP messages for incoming calls from the British Telecom Spain SIP trunking service are sent to the SES server. SES then routes these messages to the appropriate Communication Manager **Signaling Group** using an **SES Media Server Address Map**.

In these Application Notes, the incoming ISDN calls from the British Telecom SIP trunking service use **Media Server Address Map Patterns** matching the 9-digit *called number* in the *user* part of the SIP URI.

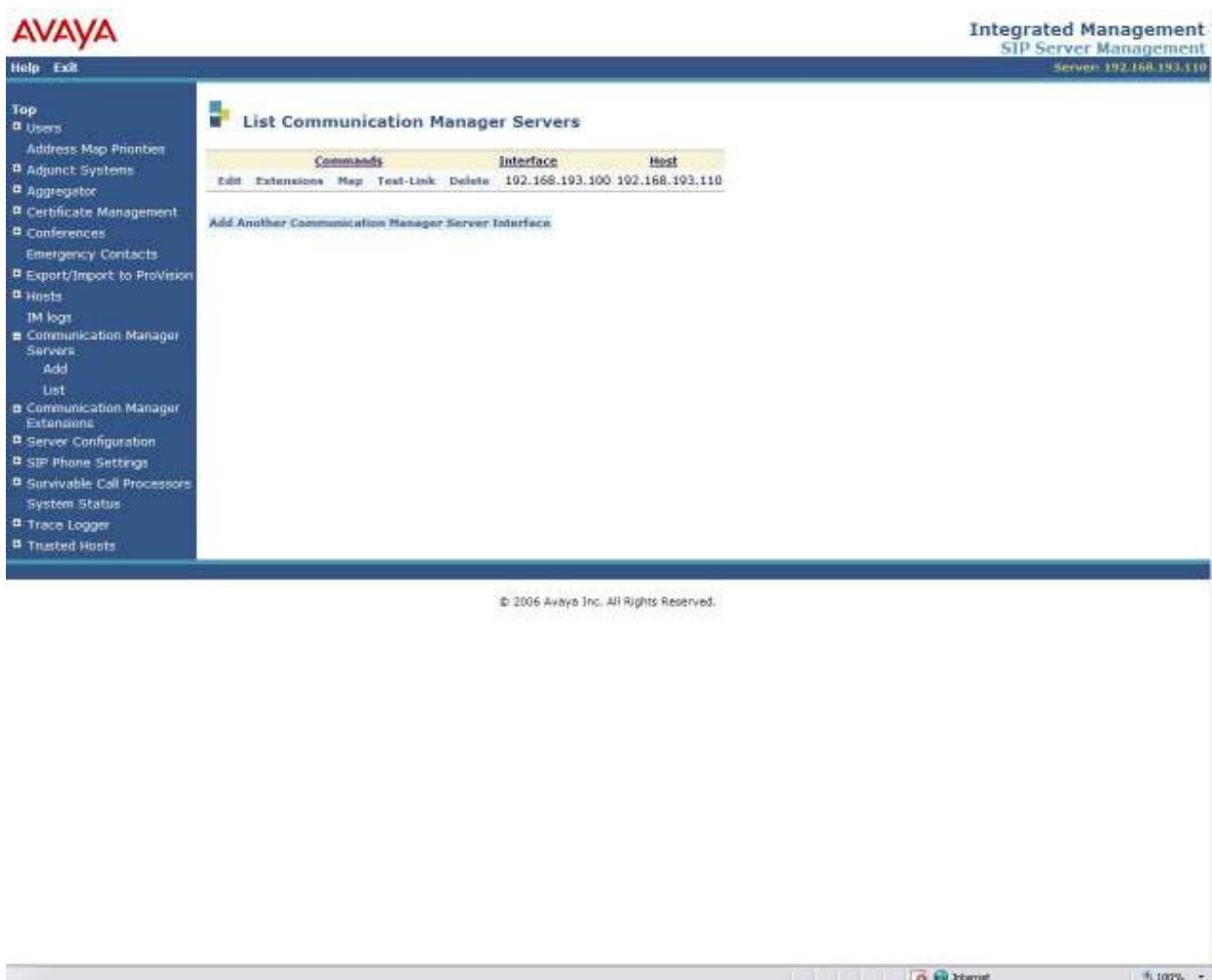
An example of a SIP URI in an INVITE message received from the British Telecom Spain SIP trunking service for the number **911779080** is:

***sip:911779080@192.168.193.110***

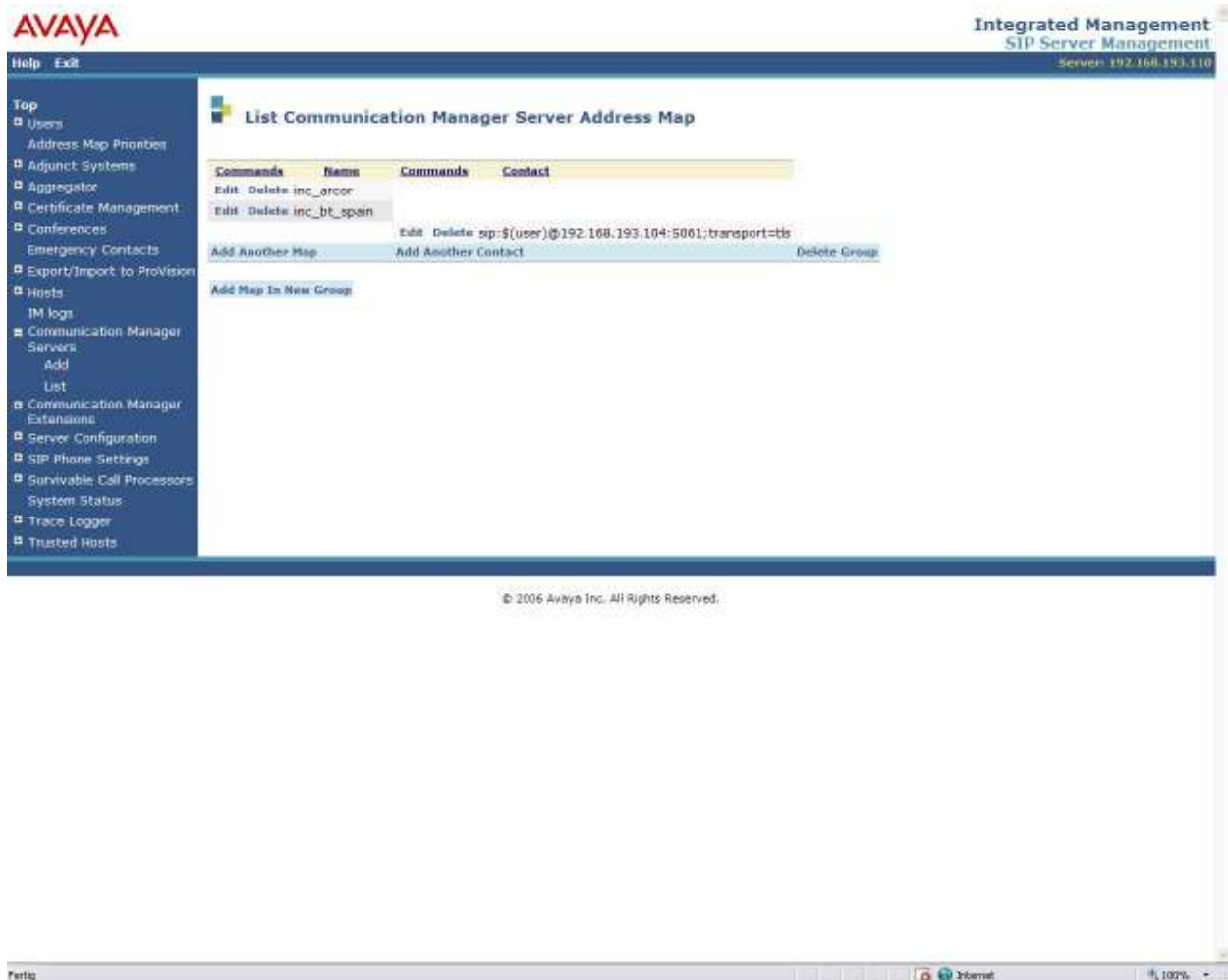
The *user* part in this case is the 9-digit number “**911779080**”.

To configure the **Media Server Address Map** for voice calls:

- Expand the **Media Servers** link in the left navigation menu of any **SIP Server Management** page. Select **List** to display the **List Media Servers** page as shown in **Figure 36**.
- Click on the **Map** link of the **procr** Interface to display the **List Media Server Address Map** (shown in **Figure 37**) screen associated with this signaling interface.



**Figure 36: List Communication Manager Servers**

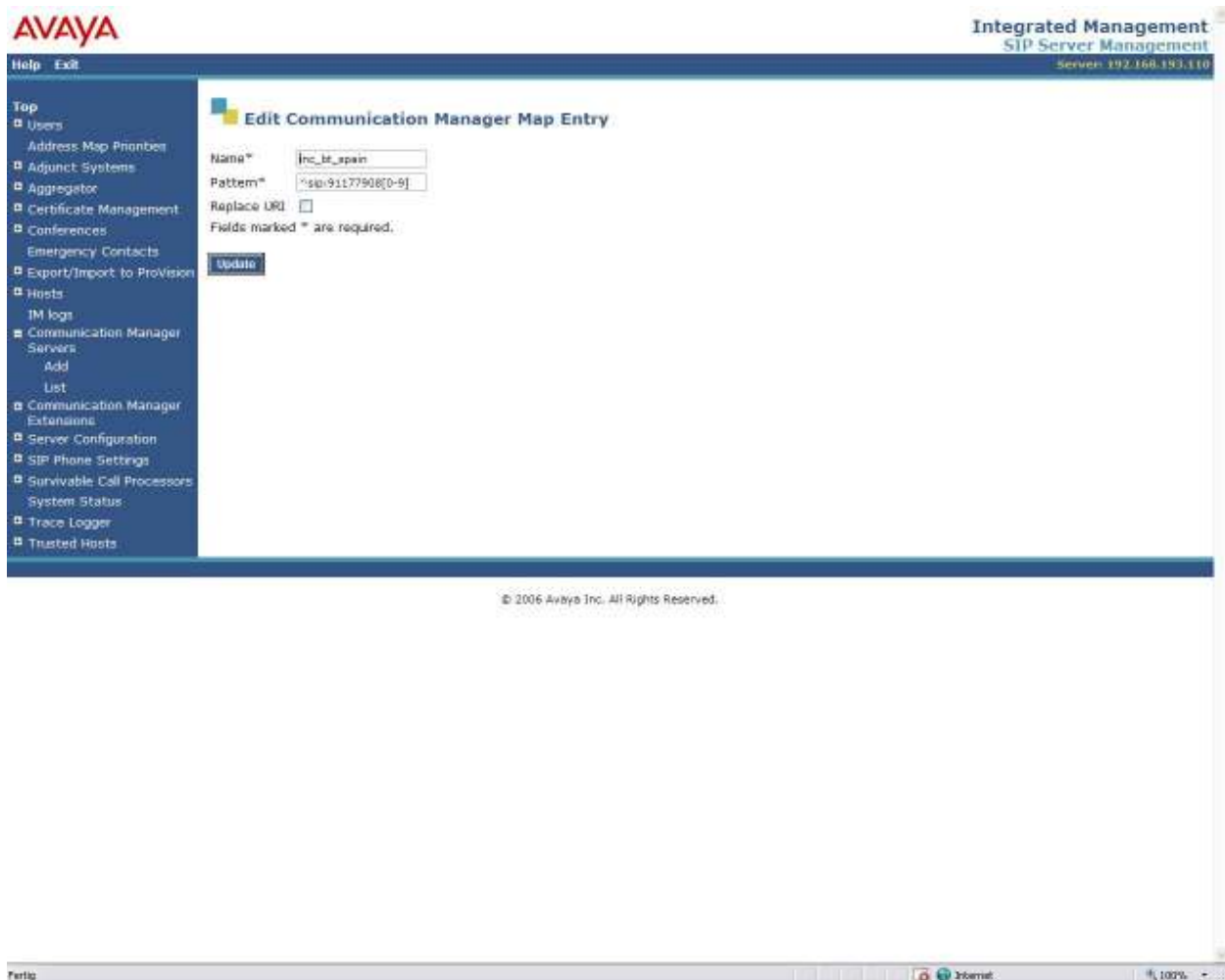


**Figure 37: List Communication Manager Server Address Map**

- Click on the **Add Map In New Group** (/Edit in the Command column) link. The page shown in **Figure 38** is displayed. The **Host** field displays the name of the media server to which this map applies.
- Enter a descriptive name in the **Name** field, such as **inc\_bt\_spain**.
- Enter the **Address Map Pattern** for incoming calls into the **Pattern** field.  
In this case, the numbers provided by British Telecom SIP trunking for voice extensions are 911779080 thru 911779089. The pattern specification for voice DID numbers is:  
**^sip:91177908[0-9]**

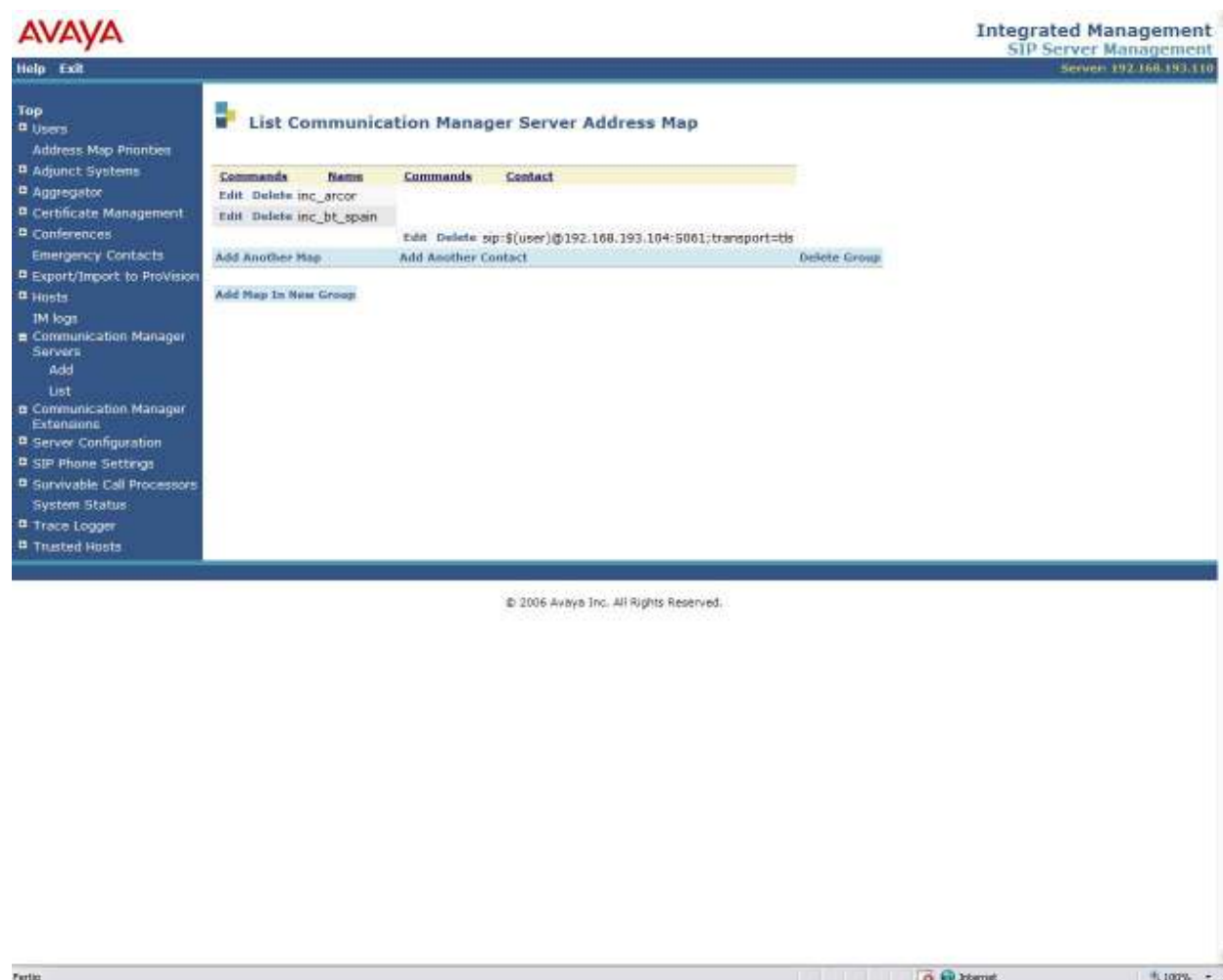
This means that URIs beginning with “sip:91177908” followed the single digit 0, 1, 2, 3, 4, 5, 6, 7, 8 or 9 will match the pattern and be routed to the S8730 media server.

- Click the **Update** button once the form is completed.



**Figure 38: Incoming Calls - Communication Manager Server Address Map**

After configuring the media server address map, the **List Media Server Address Map** page appears as shown in **Figure 39**.



**Figure 39: List Communication Manager Server Address Map**

Note that after the first **Communication Manager Server Address Map** is created, a corresponding media server **Contact** entry is created automatically.

**sip:\$(user)@192.168.193.104:5061;transport=tls**

This **Contact** entry contains the IP address of a CLAN to contact the Avaya S8730 Media Server, the port (5061) and the transport protocol (TLS) to be used to send SIP signaling messages.

The incoming digits sent in the *user* part of the original request URI will replace the \$(user) string when the message is sent to the **Contact**.

### 5.1.6 Specify the British Telecom Spain SIP trunking SIP Application Server as a Trusted Host

The final step to complete the SIP trunk administration on SES is to designate the IP address of the British Telecom Spain SIP Trunk Service as a trusted host. As a trusted host, SES will not issue SIP authentication challenges for incoming requests from the designated IP address.<sup>5</sup>

If multiple IP addresses are used to access the British Telecom Spain SIP Trunk Service, each IP Address must be added as a trusted host.

To configure a trusted host:

- Expand the **Trusted Hosts** link in the left navigation menu of any **SIP Server Management** page. Select **List** to display the **List Trusted Hosts** page as shown in **Figure 40**.
- Click on the **Add Another Trusted Host** link. The page shown in **Figure 41** is displayed. The **Host** field displays the ip address of the SIP server
- Enter the IP address of the UPC SIP Trunk Service in the **IP Address** field, such as **income\_upc** (**Figure 41**).
- Enter a descriptive name in the **Comment** field such as **SIP BT-Spain**.
- Click the **Update** button once the form is completed.

---

<sup>5</sup> Note, if the trusted host step is not done, authentication challenges to incoming SIP messages (such as INVITEs and BYEs) will be issued but not responded to. This may cause call setup may fail, active calls be disconnected after timeout periods, and/or SIP protocol errors.



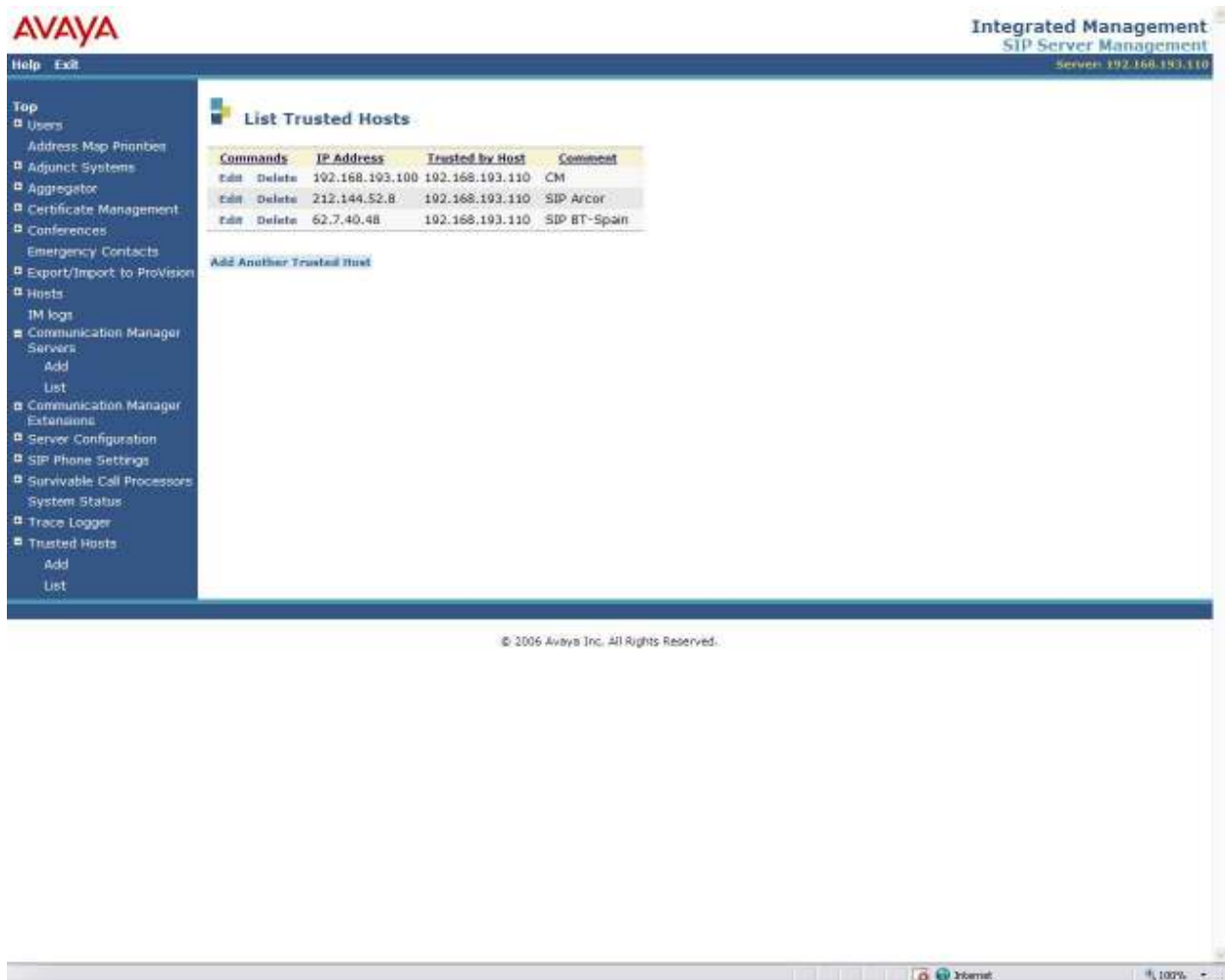
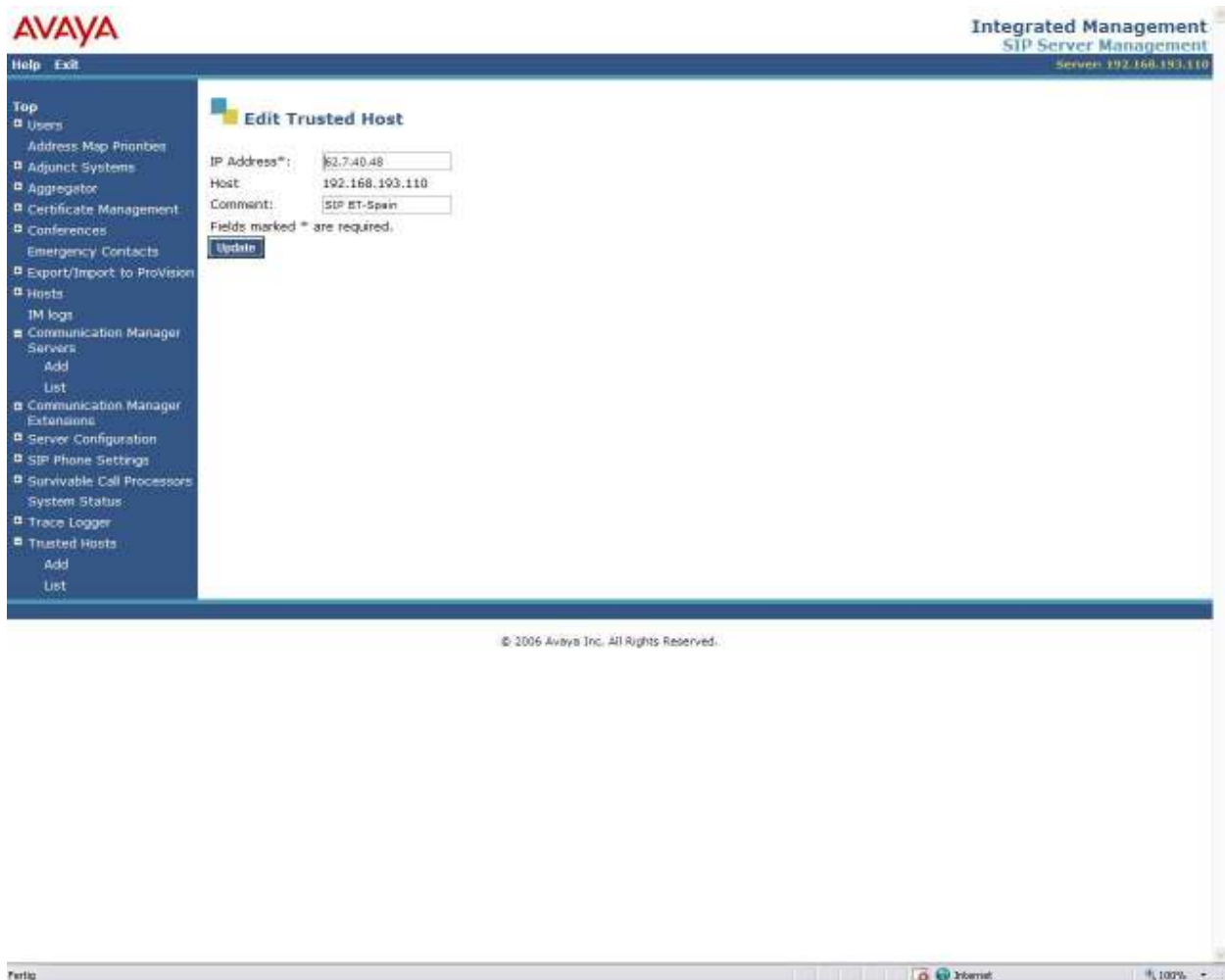


Figure 40: List Trusted Hosts Map



**Figure 41: Edit Trusted Hosts Map**

## 5.2. Configuration for SIP Telephones

This section provides basic instructions for completing the administration to use Avaya 9600 Series SIP telephones with the described configuration. Additional features, such as the use of mnemonic addressing, are beyond the scope of these Application Notes.

As noted previously, SIP telephones are optional; it is not necessary to have SIP telephones to use SIP trunking to the British Telecom Spain SIP trunking service.

The steps below are repeated for each SIP telephone provisioned.

### 5.2.1 Add a SIP User

Add a SIP User to the SES as follows:

- In **SES SIP Server Management**, expand the **Users** link in the left side blue navigation bar and click on the **Add** link.
- In the **Add User** page (**Figure 42**) enter the extension number for the SIP telephone in the **Primary Handle** field.
- Enter a user password in the **Password** and **Confirm Password** fields. This password will be used when logging into the user's SIP telephone.
- In the **Host** field, select the SES server hosting the domain for this user.
- Enter the **First Name** and **Last Name** of the user.
- Select the **Add Media Server Extension** checkbox. This associates a Communication Manager extension with this SIP User. Calls from this user will be provided features and routing via Communication Manager.
- Click the **Add** button. This will cause a confirmation screen to appear.
- Click **Continue** on the confirmation screen.

**AVAYA**

**Integrated Management**  
**SIP Server Management**  
Servers: 192.168.193.110

[Help](#)
[Exit](#)

**Top**  

- Users
  - Add
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- SIP Phone Settings
- Survivable Call Processors
- System Status
- Trace Logger
- Trusted Hosts

**Add User**

Primary Handle\*   
 User ID   
 Password\*   
 Confirm Password\*   
 Host\*    
 First Name\*   
 Last Name\*   
 Address 1   
 Address 2   
 Office   
 City   
 State   
 Country   
 Zip   
 Survivable Call Processor    
 Add Communication Manager Extension ☒  
 Fields marked \* are required.

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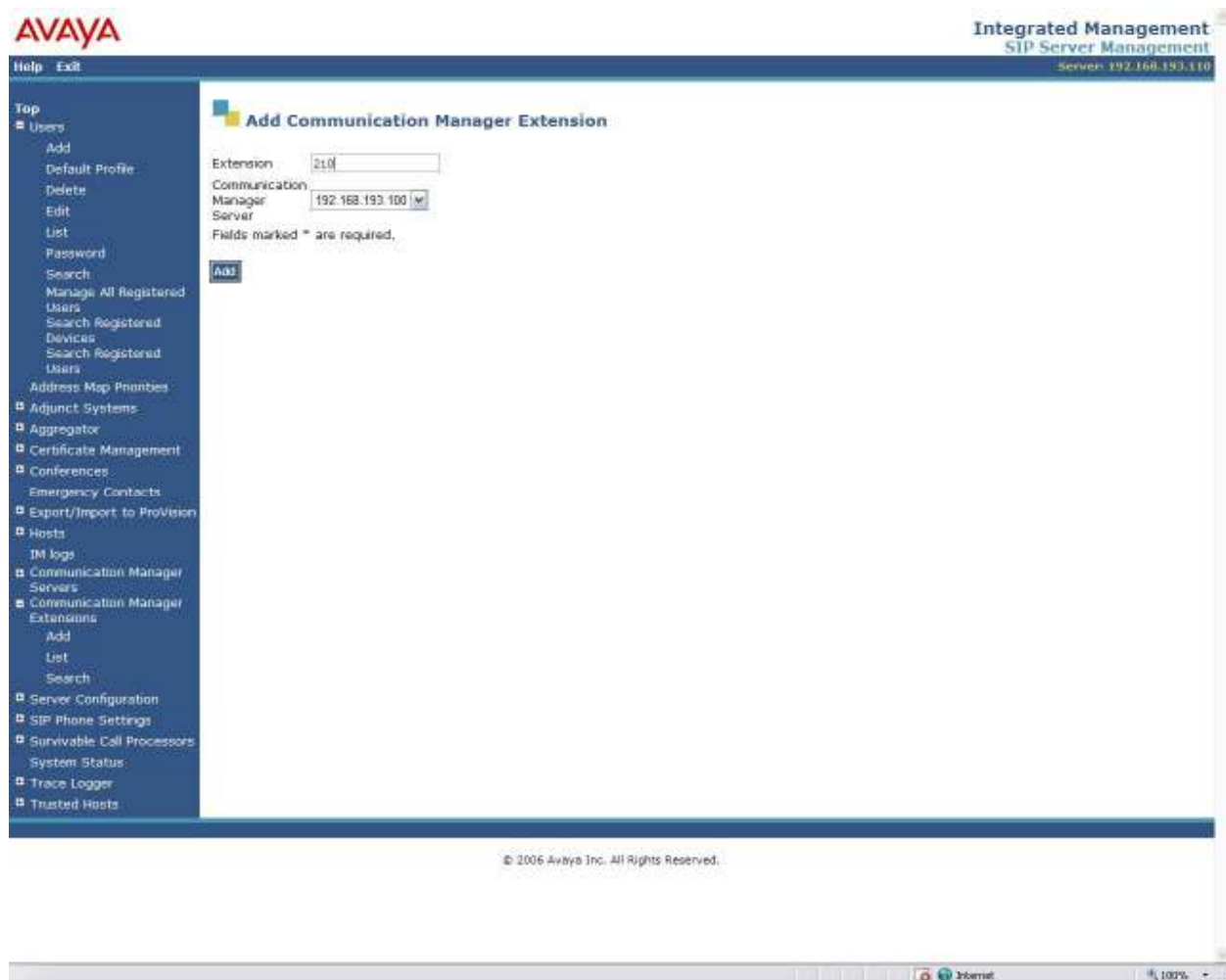
**Figure 42: Add User**

## 5.2.2 Specify Corresponding Communication Manager Extension

The SIP phone handle must now be associated with the corresponding extension on Communication Manager.

If the **Add Media Server Extension** checkbox was not selected, complete this step:

- In the **Add Media Server Extension** page, enter the extension number (e.g., **210**) corresponding to the extension previously configured on Communication Manager **Figure 43** in section 4.3. The Media Server **Extension** and the SIP Primary Handle are usually the same, but are not required to be.
- Select the **Media Server** interface assigned to this extension.
- Click the **Add** button.



**Figure 43: Add Media Server Extension**

## 6. British Telecom Spain SIP trunking service Configuration

In order to use British Telecom Spain SIP trunking service, a customer must order service from British Telecom Spain SIP trunking using their sales processes. The process can be started by contacting a British Telecom Spain SIP trunking sales representative.

British Telecom Spain SIP trunking provided the information contained in **Table 2** to complete the configuration in these Application Notes. This information was necessary to complete the Avaya Communication Manager and Avaya SIP Enablement Services administration discussed in the previous sections.

Required Information	Values Used In These Application Notes
British Telecom Spain SIP trunking service SIP Application Server	<b>62 . 7 . 40 . 48</b>
Assigned DID Numbers	<b>911779080 to 911779089</b>
Codecs Supported	<b>G.729, G.711A, T38</b>

**Table 2: British Telecom Spain SIP trunking service Configuration Information**

## 7. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between the British Telecom Spain SIP trunking standard service and an Avaya SIP telephony solution using SIP trunking. This section covers the general test approach and the test results.

### 7.1. General Test Approach

A simulated enterprise site consisting of an Avaya SIP telephony solution supporting SIP trunking was connected to a laboratory version of the British Telecom Spain SIP trunking service via simulated managed access facilities. The enterprise site was configured as if using the generally available service provided by British Telecom Spain.

The following features and functionality were covered during the SIP trunking interoperability compliance test. All calls involved various Avaya telephones, the British Telecom Spain SIP trunking service and various ISDN telephones.

- Outgoing calls to ISDN telephones
- Incoming calls to British Telecom Spain SIP trunking provided DID numbers from PSTN telephones
- Calls using Avaya 9600 Series IP Telephones with the H.323 firmware configurations
- Calls using Avaya 9600 Series IP Telephones with the SIP firmware configurations
- G.711A and G.29 codecs for voice calls
- G.711A, G.729 and T38 codecs for fax calling
- DTMF tone transmission using RFC 2833 with successful Voice Mail navigation
- Telephone features such as hold, transfer and conference
- Direct IP-to-IP media (also known as “shuffling”)

### 7.2. Test Results

Interoperability testing of the sample configuration was completed with partial unsuccessful results.

The following issues described in **Table 3** were observed.

Item	Issue Observed	Discussion / Workaround
Fax/T38 between Communication Manager and BT Voicemail-Fax platform	BT doesn't increment the version in the o-header in the INVITE for switchover to T38	Problem is investigated by BT voicemail support

**Table 3: Summary of observations identified during Interoperability Testing**

## 8. Verification Steps

This section provides the verification steps that may be performed to verify operation of the Avaya SIP telephony solution with the British Telecom Spain SIP trunking service.

- Verify outbound voice calls can be placed to ISDN phones and that the calls remain connected for 3 minutes.
- Verify incoming voice calls can be received from ISDN phones and that the calls remain connected for 3 minutes.
- Verify that calls are properly disconnected when either end disconnects first.
- Using the **list trace station** command, verify that voice calls are using the expected codec and SIP trunk group (for both outbound and incoming calls).
- Verify outbound and inbound fax calls can be placed to and from ISDN fax machines.
- Using the **list trace station** command verify that fax calls are using the expected codec and trunk group (for inbound and outbound calls).
- If Avaya 9600 Series SIP telephones are used with G.711 codecs and Direct IP-to-IP is enabled, verify that the RTP media path is reconfigured once a stable call is established with an ISDN telephone. Verify by using the **list trace station** command to observe that the IP address of the SIP phone and British Telecom SIP Application Server are the media endpoints after the call is established.

## 9. Conclusion

These Application Notes describe the configuration steps required to connect customers using an Avaya SIP telephony solution to the British Telecom Spain SIP trunking standard service using SIP trunking. The British Telecom Spain SIP trunking service is a Voice over IP solution for customers ranging from small businesses to large enterprises. SIP trunking uses the Session Initiation Protocol to connect private company networks to the public telephone network via converged IP access. It provides businesses a flexible, cost-saving alternative to current TDM-based telephony trunk lines.



## 10. References

This section references documentation relevant to these Application Notes.

The Avaya product documentation is available at <http://support.avaya.com>.

- [1] *Administering Avaya Aura Communication Manager*, May 2009, Issue 5.0, Release 5.2, Document Number 03-300509.
- [2] *Avaya Aura Communication Manager Feature Description and Implementation*, May 2009, Issue 7, Release 5.2, Document Number 555-245-205
- [3] *Avaya Extension to Cellular and Off-PBX Station (OPS) Installation and Administration Guide Release 3.0*, June 2005, Issue 9, Version 6.0, Document Number 210-100-500.
- [4] *Installing, Administering, Maintaining and Troubleshooting SIP Enablement Services R5.1*, June 2008, Issue 6.0, Document Number 03-600768
- [5] *SIP Support in Avaya Aura Communication Manager Running on S8xxx Servers*, May 2009, Issue 9, Document Number 555-245-206.
- [6] *Avaya one-X Deskphone Edition for 9600 Series IP Telephones Administrator Guide Release 3.0*, May 2009, Issue 7, Document Number 16-300698
- [7] *Avaya one-X Deskphone Edition for 9600 Series SIP IP Telephones Administrator Guide Release 2.0*, December 2007, Issue 2, Document Number 16-601944

Several Internet Engineering Task Force (IETF) standards track RFC documents were referenced within these Application Notes. The RFC documents may be obtained at: <http://www.rfc-editor.org/rfcsearch.html>.

- [8] RFC 3261 - *SIP (Session Initiation Protocol)*, June 2002, Proposed Standard
- [9] RFC 2833 - *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, May 2000, Proposed Standard
- [10] RFC 2782 - *A DNS RR for Specifying the Location of Services (DNS SRV)*, February 2000, Proposed Standard

## APPENDIX A: Sample SIP INVITE Messages

This section displays the format of the SIP INVITE messages sent by British Telecom Spain SIP trunking and the Avaya SIP network at the enterprise site. These INVITE messages may be used for comparison and troubleshooting purposes. Differences in these messages may indicate that different configuration options were selected.

### Sample SIP INVITE Message from British Telecom Spain SIP trunking to Avaya SIP Enablement Services:

```
sip:911779081@192.168.193.110:5060;transport=UDP;user=phone SIP/2.0
From: <sip:912700131@62.7.40.48:5060;user=phone>;tag=c4-45026-2d7f51-27ff71a5-2d7f51
To: <sip:911779081@192.168.193.110:5060;user=phone>
Call-ID: a58b7a983028073e13c42d7f51b1b955603bdd609cc047f0-0026-5067
CSeq: 1 INVITE
User-agent: CS2000_NGSS/8.0
Remote-Party-ID: <sip:912700131@62.7.40.48;user=phone>; party=calling; privacy=off; screen=yes
Allow: ACK,BYE,CANCEL,INVITE,OPTIONS,INFO,SUBSCRIBE,REFER,NOTIFY,PRACK
Via: SIP/2.0/UDP 62.7.40.48:5060;maddr=62.7.40.48;branch=z9hG4bK-2d7f51-b1b95560-68c80fca
P-Asserted-Identity: <sip:912700131@62.7.40.48;user=phone>
Max-Forwards: 70
Contact: <sip:62.7.40.48:5060;transport=UDP>
Supported: 100rel
Content-Type: application/sdp
Content-Length: 208

v=0
o=MGCP 3514817881 2583700740 IN IP4 62.7.41.200
s=MGCP Call
c=IN IP4 62.7.41.200
t=0 0
m=audio 40340 RTP/AVP 18 96
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=fmtp:18 annexb=no
a=ptime:20
```

## Sample SIP INVITE Message from Avaya SIP Enablement Services to British Telecom Spain SIP trunking:

sip:912700131@62.7.40.48:5060;lr SIP/2.0  
Call-ID: 0f0afecbbedd19fb49325ab200  
CSeq: 1 INVITE  
From: "201" <sip:911779081@sipdomain.olymp.org:5061>;tag=0f0afecbbedd19eb49325ab200  
Record-Route: <sip:192.168.193.110:5060;lr>,<sip:192.168.193.104:5061;lr;transport=tls>  
To: "912700131" <sip:912700131@192.168.193.110>  
Via: SIP/2.0/UDP 192.168.193.110:5060;branch=z9hG4bK030303666666030303538f.0,SIP/2.0/TLS  
192.168.193.104;psrrposn=2;received=192.168.193.104;branch=z9hG4bK0f0afecbbedd1a0b49325ab200  
Content-Length: 217  
Content-Type: application/sdp  
Contact: "201" <sip:911779081@192.168.193.104:5061;transport=tls>  
Max-Forwards: 61  
User-Agent: Avaya CM/R015x.02.0.939.0  
Allow: INVITE,CANCEL,BYE,ACK,PRACK,SUBSCRIBE,NOTIFY,REFER,OPTIONS,INFO,PUBLISH  
Supported: timer,replaces,join,histinfo,100rel  
Alert-Info: <cid:internal@invalid.unknown.domain>;avaya-cm-alert-type=internal  
Min-SE: 1200  
Session-Expires: 1200;refresher=uac  
P-Asserted-Identity: "201" <sip:911779081@sipdomain.olymp.org:5061>  
P-Charging-Vector: icid-value="AAS:192.168.193.110-615-caff0001ddbccc32490b9db25a"  
  
v=0  
o=- 1 1 IN IP4 192.168.193.104  
s=-  
c=IN IP4 192.168.193.105  
b=AS:64  
t=0 0  
m=audio 2472 RTP/AVP 8 18 127  
a=rtpmap:8 PCMA/8000  
a=rtpmap:18 G729/8000  
a=fmtp:18 annexb=yes  
a=rtpmap:127 telephone-event/8000

## APPENDIX B: Specifying Pattern Strings in Address Maps

The syntax for the pattern matching used within the Avaya SES is a Linux regular expression used to match against the URI string found in the SIP INVITE message.

Regular expressions are a way to describe text through pattern matching. The regular expression is a string containing a combination of normal text characters, which match themselves, and special *metacharacters*, which may represent items like quantity, location or types of character(s).

In the pattern matching string used in the Avaya Aura™ SES:

- Normal text characters and numbers match themselves.
- Common metacharacters used are:
  - A period `.` matches any character once (and only once).
  - An asterisk `*` matches zero or more of the preceding characters.
  - Square brackets enclose a list of any character to be matched. Ranges are designated by using a hyphen. Thus the expression `[12345]` or `[1-5]` both describe a pattern that will match any single digit between 1 and 5.
  - Curly brackets containing an integer 'n' indicate that the preceding character must be matched exactly 'n' times. Thus `5{3}` matches '555' and `[0-9]{10}` indicates any 10 digit number.
  - The circumflex character `^` as the first character in the pattern indicates that the string must begin with the character following the circumflex.

Putting these constructs together as used in this document, the pattern to match the SIP INVITE string for any valid 1+ 10 digit number in the North American dial plan would be:

**`^sip:1[0-9]{10}`**

This reads as: "Strings that begin with exactly **sip:1** and having any 10 digits following will match.

A typical INVITE request below uses the shaded portion to illustrate the matching pattern.

```
INVITE sip:17325551638@20.1.1.54:5060;transport=udp SIP/2.0
```

---

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