



**Application Notes for Configuring SIP Trunking (TCP)
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 (TCP) 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(TCP) between the British Telecom Spain SIP trunking standard service and an Avaya Aura™ SIP telephony solution consisting of Avaya Aura™ SIP Enablement Services, Avaya Aura™ Communication Manager and Avaya IP telephones using H.323 and SIP protocols.

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 TCP 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 (Point of Presence) 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
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 (TCP). This is the configuration that was used during the DevConnect compliance testing.

The simulated customer configuration included:

- Two Avaya S8730 Server with an Avaya G650 Media Gateway. The S8730 served as the processor for Avaya Aura™ Communication Manager. At the G650 are separate C-LAN and Media Processor (MEDPRO) for using British Telecom Spain SIP trunking service TCP.
- Avaya Aura™ SIP Enablement Services operating on a SES Combined Home/Edge server (S8500C).
- Avaya 9600 Series IP telephones using the H.323 / SIP software bundle.
- Avaya Analog phones
- Fax machines

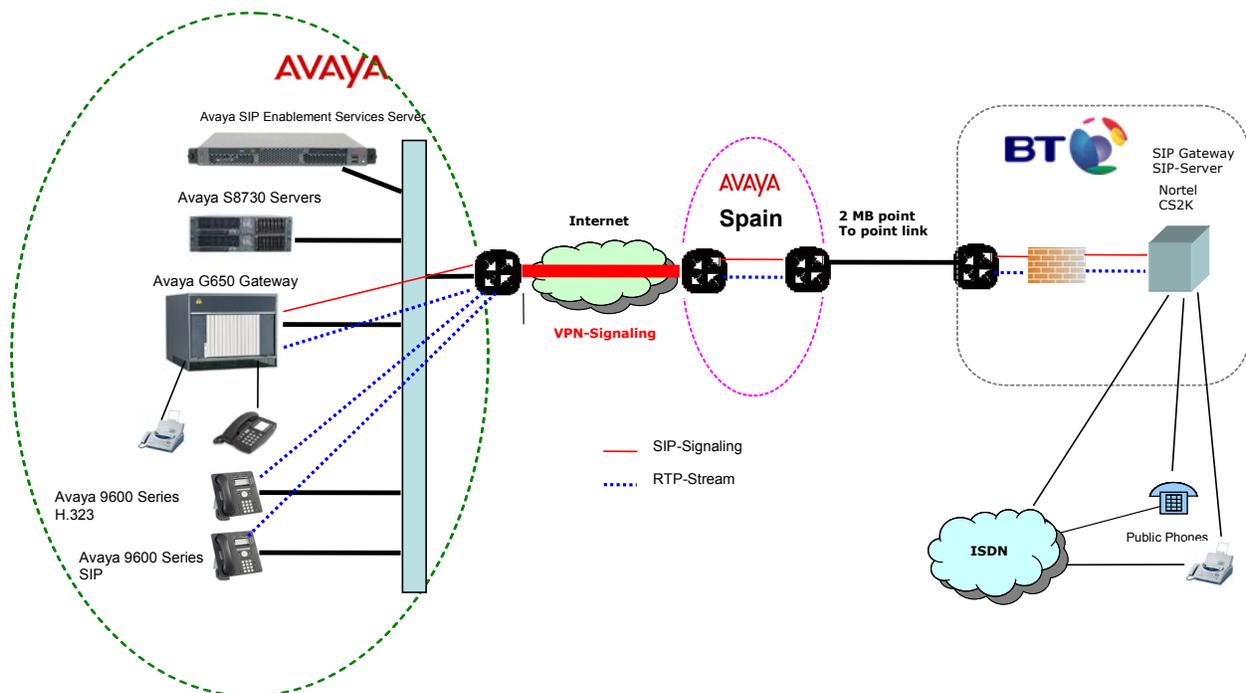


Figure 1: Avaya Aura™ SIP Telephony Solution using British Telecom Spain SIP Trunking Service

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 Communication Manager using SIP signaling messages sent over the managed access facility. Note that the assignment of the number and the address of the Communication Manager was previously established during the ordering and provisioning of the service.
3. Communication Manager terminates the call to an Avaya H.323/Analog/Digital telephone as shown in **Figure 2**.
- or -
3.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 3a arrow.)

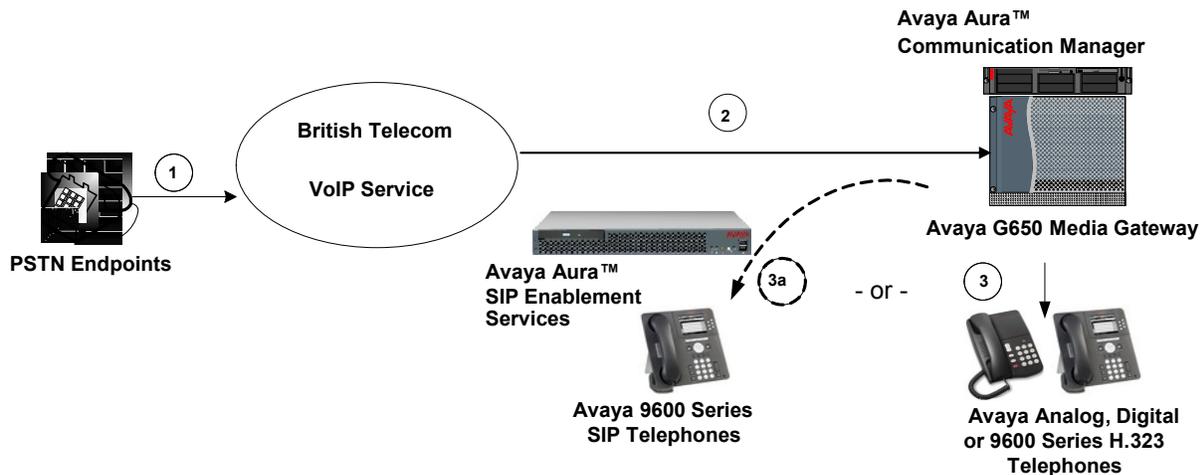


Figure 2: Incoming BT Spain network Calls to Avaya Aura™ Communication Manager

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 the British Telecom Spain SIP trunking service.
3. The British Telecom Spain SIP trunking service completes the call to the BT Spain network.

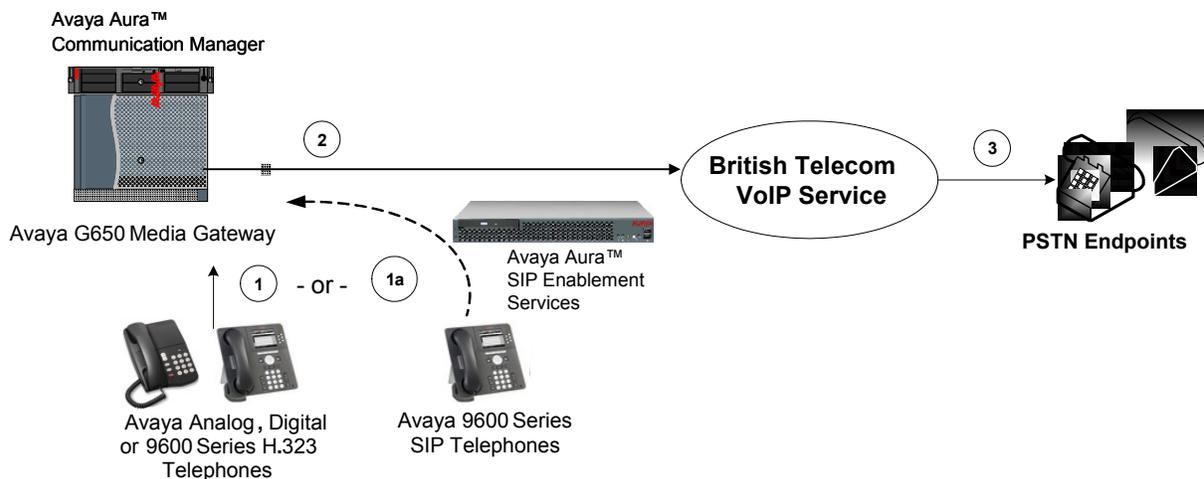


Figure 3: Outgoing Calls from Avaya Aura™ Communication Manager to the BT Spain Network

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 Server	Avaya Aura™ Communication Manager 5.2 R015x.02.0.947.3 Patch 02.0.947.3-17684
Avaya G650 Media Gateway TN2312BP IP Server Interface TN799DP Control-LAN TN2602AP IP Media Processor TN799DP Control LAN TN2602AP IP Media Processor TN793CP Analog Line	HW28 FW049 HW16 FW035 HW02 FW051 HW01 FW034 HW02 FW051 HW07 FW010
Avaya Aura™ SIP Enablement Services	SIP Enablement Services 5.2 SES05.2-02.0.947.3b Patch SES-02.0.947.3-SP3a
Avaya 9630 H.323 IP Telephones	<i>Application file:</i> ha96xxua3_10.bin <i>Boot file:</i> hb96xxua3_10.bin
Avaya 9630 SIP IP Telephones	<i>Application file:</i> SIP96xx_2_5_5_13.bin <i>Boot file:</i> hb96xxua3_00.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 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 (TCP) 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.

Normally, in the Avaya SIP trunk architecture, SIP Enablement Services acts as a SIP proxy through which all incoming and outgoing SIP messages flow to a SIP trunking service provider (UDP).

If the British Telecom Spain SIP trunk using TCP as transport protocol for the SIP messages instead of UDP, the Communication Manager acts as a SIP proxy.

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, the SIP messages are sent directly to Communication Manager. Once the message arrives at 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 (COS). Communication Manager selects the SIP trunk to the British Telecom SIP trunking to route the call to the British Telecom Spain Application Server.

The Communication Manager configuration was performed using the System Access Terminal (SAT). The initial installation and configuration of the S8730 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
Maximum Off-PBX Telephones - OPS: 500 6
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.

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
Maximum Administered SIP Trunks:	100	55
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

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.

```
display system-parameters special-applications                               Page 8 of 9
                                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 IP 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 for the SIP endpoints.

clan-cab-a-2 and **212.163.105.50** are used below as the **Name** and **IP Address** of the C-LAN for incoming and outgoing SIP messages from and to the British Telecom Spain SIP trunking service.

def_gatew_sp_sp and **212.163.105.1** are used below as the **Name** and **IP Address** for the Default gateway in the BT network.

medpro2caba and **212.163.105.52** are used below as the **Name** and **IP Address** of the MEDPRO for incoming and outgoing RTP packages from and to the British Telecom Spain SIP trunking service (not for direct media connections).

SIP-BT-SP-TCP and **62.7.40.48** are used below as the **Name** and **IP Address** of the SIP proxy at British Telecom Spain SIP trunking service.

```
change node-names ip                                     Page 1 of 2
                                                    IP NODE NAMES
  Name          IP Address
Gateway001     192.168.193.1
SIP-BT-SP-TCP 62.7.40.48
clan-cab-a-2  212.163.105.50
clan-cabinet-a 192.168.193.104
def_gatew_bt_sp 212.163.105.1
default        0.0.0.0
medpro-cabineta 192.168.193.105
medpro2caba   212.163.105.52
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 Determine CLAN for SIP Trunk TCP British Telecom Spain SIP Trunking Standard Service

In the **IP-Interfaces** form assigns the parameters for the C-LAN, which serves the British Telecom Spain SIP trunking standard service using the **change ip-interface 01a04** command shown in **Figure 8**.

Network Region 13 is the region used for serving the British Telecom Spain SIP trunking standard service.

Node Name clan-cab-a-2 is the IP address for the C-LAN board, which is used for serving the British Telecom Spain SIP trunking standard service (see **Figure 7**).

Gateway Node Name def_gatew_sp_sp is the ip address of the BT Network (see **Figure 7**).

```
change ip-interface 01a04                                     Page 1 of 3
                                                           IP INTERFACES

      Type: C-LAN
      Slot: 01A04      Target socket load and Warning level: 400
      Code/Suffix: TN799 D      Receive Buffer TCP Window Size: 8320
      Enable Interface? y      Allow H.323 Endpoints? n
      VLAN: n      Allow H.248 Gateways? y
      Network Region: 13

                                                           IPV4 PARAMETERS
      Node Name: clan-cab-a-2
      Subnet Mask: /24
      Gateway Node Name: def_gatew_bt_sp

      Ethernet Link: 2
      Network uses 1's for Broadcast Addresses? Y
```

Figure 8: IP Nodes Names Form

4.2.3 Determine MEDPRO for SIP Trunk TCP British Telecom Spain SIP trunking standard service

In the **IP-Interfaces** form assigns the parameters for the MEDPRO, which serves the British Telecom Spain SIP trunking standard service using the **change ip-interface 01a05** command shown in **Figure 9**.

Network Region 13 is the region used for serving the British Telecom Spain SIP trunking standard service.

Node Name medpro2caba is the ip address for the MEDPRO board, which is used for serving the British Telecom Spain SIP trunking standard service (see **Figure 7**).

Gateway Node Name def_gatew_sp_sp is the ip address of the BT Network (see **Figure 7**).

```
change ip-interface 01a05                                     Page 1 of 4
                                                           IP INTERFACES
                                                           Critical Reliable Bearer? n
Type: MEDPRO
Slot: 01A05
Code/Suffix: TN2602
Enable Interface? y
VLAN: n
Network Region: 13
VOIP Channels: 320

                                                           IPV4 PARAMETERS
Node Name: medpro2caba
Subnet Mask: /24
Gateway Node Name: def_gatew_bt_sp
```

Figure 9: IP Nodes Names Form

4.2.4 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 10 and 10a**) to define the codec set. On Page 2 of **ip-codec-set** form, the **Fax Mode** must be **t.38-standard**, **Modem** and **TTD/TTY** modes must be **off**.

```
change 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 10: IP Codec Set Forms for Voice

```
change ip-codec-set 1                                     Page 2 of 2
                                                         IP Codec Set
Allow Direct-IP Multimedia? n
Mode              Redundancy
FAX            t.38-standard    0
Modem             off              0
TDD/TTY          off              0
Clear-channel     n                0
```

Figure 10a: IP Codec Set Forms for Fax

4.2.5 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). In the **IP Network Region** form shown in **Figure 11** and **Figure 11a**:

- The **Authoritative Domain** field is configured to the domain name is "localdomain.com". 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 1.¹

```
change ip-network-region 13                                     Page 1 of 19
                                                                IP NETWORK REGION
Region: 13
Location: 1           Authoritative Domain: localdomain.com
Name: BT-Spain-TCP
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 11: IP Network Region Form - Page 1

¹ 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.

All other IP devices (9600 phones, CLAN, MEDPRO) are in Network Region 1. For connecting the BT SIP trunk with the enterprise phones the connecting between Network Region 1 and 13 must be allowed

- For **dst rgn 1**, the codec set is set to "I", direct WAN is set to "y", IGAR is set to "n" and AGL is set to "all".

change ip-network-region 13										Page	3 of	19
Source Region: 13 Inter Network Region Connection Management										I		M
dst	codec	direct	WAN-BW-limits	Video	Intervening	Dyn	A	G	a	G	A	e
rgn	set	WAN	Units	Total Norm	Prio Shr	Regions	CAC	R	L	s		
1	1	y	NoLimit					n	all			
2												
3												
4												
5												
6												
7												
8												
9												
10												
11												
12												
13	1										all	
14												
15												

Figure 11a: IP Network Region Form - Page 3

There were some issues uncovered during the testing to keep 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 outside 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 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.

4.2.6 Define SIP Trunk Groups

Configure the **Signaling Group** form for the British Telecom SIP trunking service shown as follows, shown in **Figure 12a**:

- Set the **Group Type** field to *sip*.
- The **Transport Method** field will default to *tcp* (Transmission Control Protocol).
- Specify the Avaya S8730 Server C-LAN board (node name “clan-cab-a-2”) and the SIP proxy at British Telecom Spain SIP trunking service (node name “SIP-BT-SP-TCP”) 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 7².
- Ensure that the recommended TCP port value of *5060* 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 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.

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

```

change signaling-group 13                                     Page 1 of 1
                                SIGNALING GROUP

Group Number: 1                      Group Type: sip
                                Transport Method: tcp

IMS Enabled? n
IP Video? n

Near-end Node Name: clan-cab-a-2      Far-end Node Name: SIP-BT-SP-TCP
Near-end Listen Port: 5060            Far-end Listen Port: 5060
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
                                Alternate Route Timer(sec): 20

```

**Figure 12a: Signaling Group Form
British Telecom SIP trunking service**

```

change signaling-group 1                                     Page 1 of 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
                                Alternate Route Timer(sec): 20

```

**Figure 12b: Signaling Group Form
SES Trunk**

Configure the **Trunk Group** form as shown in **Figure 13** through using the **add trunk-group** command. **Trunk Group** number 13 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 SpainTCP**.
- Specify an available trunk access code (**TAC**) such as ***06**.
- Set the **Service Type** field to **public-ntwrk**.
- Enter **13** 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.

```
add trunk-group 13                                     Page 1 of 21
                                     TRUNK GROUP
Group Number: 13                                     Group Type: sip           CDR Reports: y
  Group Name: BT Spain TCP                          COR: 1                   TN: 1           TAC: *06
  Direction: two-way                               Outgoing Display? n
  Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: public-ntwrk                         Auth Code? n
                                                    Signaling Group: 13
                                                    Number of Members: 30
```

**Figure 13: Trunk Group Form – Page 1
British Telecom SIP trunking service**

```
add trunk-group 1                                     Page 1 of 21
                                     TRUNK GROUP
Group Number: 13                                     Group Type: sip           CDR Reports: y
  Group Name: SES Trunk                             COR: 1                   TN: 1           TAC: *01
  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 13a: Trunk Group Form – Page 1
SES Trunk**

On Page 3 of the **Trunk Group** form (**Figure 14**):

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

```
add trunk-group 13                                     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 14: Trunk Group Form – Page 3

On Page 4 of the **Trunk Group** form (**Figure 15**):

- 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).
- Set the **Overwrite Calling Identity** field to **y**. This field appears, if the special application SA8972 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.

```

add trunk-group 13
                                Page 4 of 21
                                PROTOCOL VARIATIONS
                                Mark Users as Phone? n
                                Prepend '+' to Calling Number? n
                                Send Transferring Party Information? n
                                Network Call Redirection? n
                                Send Diversion Header? n
                                Support Request History? n
                                Telephone Event Payload Type: 96
                                Overwrite Calling Identity? y

```

Figure 15: Trunk Group Form – Page 4

4.2.7 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 unauthorized 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 dial 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 16 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 13.

```

change public-unknown-numbering 0
                                Page 1 of 2
                                NUMBERING - PUBLIC/UNKNOWN FORMAT
                                Total
                                Total
Ext  Ext      Trk      CPN      CPN
Len  Code      Grp(s)   Prefix   Len
                                Total Administered: 7
                                Maximum Entries: 9999
 3  200        13      911779080  9
 3  201        13      911779081  9
 3  203        13      911779086  9
 3  210        13      911779082  9
 3  211        13      911779083  9
 3  220        13      911779084  9
 3  230        13      911779085  9

```

Figure 16: Public/Unknown Numbering Form

4.2.8 Configure Call Routing

4.2.8.1 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**).

```
change 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 17: Dial Plan Analysis Form

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

```
change 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 18: Feature Access 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].

```
change ars analysis 0
```

Page 1 of 2

ARS DIGIT ANALYSIS REPORT

Location: all Percent Full: 0

Dialed String	Total		Route Pattern	Call Type	Node Num	ANI Reqd
	Min	Max				
00	7	14	2	pubu		n
8	3	9	2	pubu		n
9	3	9	2	pubu		n

Figure 19: ARS Analysis Form

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

```
change route-pattern 2
```

Page 1 of 3

Pattern Number: 2 Pattern Name: SIP BT Spain
 SCCAN? n Secure SIP? n

Grp No	FRL	NPA	Pfx	Hop	Toll	No. Del	Inserted Digits	DCS/ IXC	IXC
								QSIG	Intw
1:	13	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. Dgts	Numbering Format	LAR
0	1	2	M	4	W	Request				
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 20: Route Pattern Form

4.2.8.2 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 21**:

- 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).

```
change inc-call-handling-trmt trunk-group 13                               Page 1 of 30
                                INCOMING CALL HANDLING TREATMENT
Service/      Number  Number      Del Insert
Feature       Len      Digits
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 21: Incoming Call Handling Treatment – Simple Extension Mapping

4.2.9 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 22, 23 and 24**.

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 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: 9630SIP                                     Security Code: *****   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 22: 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                           Per Station CPN - Send Calling Number? y
  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
                                                    Direct IP-IP Audio Connections? y
  Emergency Location Ext: 200                     Always Use? n IP Audio Hairpinning? n
```

Figure 23: 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 24: 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 25**:

- Specify the extension of the SIP endpoint, as shown in **Figure 22**.
- 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**, the SIP trunk group number assigned in **Figure 13**:
Trunk Group Form – Page 1
British Telecom SIP trunking service

```
change off-pbx-telephone station-mapping 210 Page 1 of 2
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION
```

Station Extension	Application	Dial Prefix	CC	Phone Number	Trunk Selection	Configuration Set
210	OPS	-		210	1	1

Figure 25: 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 Extension	Appl Name	Call Limit	Mapping Mode	Calls Allowed	Bridged Calls	Location
210	OPS	3	both	all	none	

Figure 26: 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 Aura™ SIP Enablement Services

This section covers the administration of SIP Enablement Services. 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].

SIP endpoints are not necessary to use the British Telecom Spain SIP trunking service.

5.1. 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.

5.1.1 Log in to Avaya Aura™ 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 27**.

Help Log Off Installation Administration Upgrade

Server (Maintenance)
SIP Enablement Services

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Communication Manager System Management Interface

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Figure 27: Avaya Aura™ SES Main Page

The SES administration home page shown in **Figure 28** is displayed.

AVAYA

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

Top	
Manage Users	Add and delete Users.
Manage Address Map Priorities	Adjust Address Map Priorities.
Manage Adjunct Systems	Add and delete Adjunct Systems.
Manage Event Aggregators	Add/Delete Event Aggregators.
Certificate Management	Manage Certificates.
Manage Conferencing	Add and delete Conference Extensions.
Manage Emergency Contacts	Add and delete Emergency Contacts.
Export Import to ProVision	Export and import data using ProVision on this host.
Manage Hosts	Add and delete Hosts.
IM logs	Download IM Logs.
Manage Communication Manager Servers	Add and delete Communication Manager Servers.
Manage Communication Manager Extensions	Add and delete Communication Manager Extensions.
Server Configuration	View Properties of the system.
Manage SIP Phone Settings	Add/Delete Phone Settings
Manage Survivable Call Processors	Add and delete Survivable Call Processors.
System Status	View System Status.
Trace Logger	Manage SIP Trace Logs.
Manage Trusted Hosts	Add and delete Trusted Hosts.

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Figure 28: 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 localdomain.com** is used in these Application Notes (see **Figure 11**).

The screenshot shows the Avaya SIP Server Management interface. The top left features the Avaya logo and navigation links for 'Help' and 'Exit'. A left-hand navigation pane lists various system management options, with 'Server Configuration' expanded to show 'System Properties'. The main content area is titled 'View System Properties' and displays the following configuration details:

SES Version	SES-5.2.0.0-947.3b
System Configuration	Simplex
Host Type	SES combined home-edge
SIP Domain*	<input type="text" value="localdomain.com"/>

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*	<input type="text" value="192.168.193.110"/>
-------------------	--

DiffServ/TOS Parameters

Call Control PHB Value*	<input type="text" value="46"/>
-------------------------	---------------------------------

802.1 Parameters

Priority Value*	<input type="text" value="6"/>
Management System Access Login	<input type="text" value="craft"/>
Management System Access Password	<input type="password" value="....."/>
DB Log Level	<input type="text" value="disabled"/>

Figure 29: System Properties

5.1.3 Verify the Avaya Aura™ SIP Enablement Services 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 30**) 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.

Help Exit

Top

- ▣ Users
 - Address Map Priorities
- ▣ Adjunct Systems
- ▣ Aggregator
- ▣ Certificate Management
- ▣ Conferences
 - Emergency Contacts
- ▣ Export/Import to ProVision
- ▣ Hosts
 - List
 - Migrate Home/Edge
 - IM logs
- ▣ Communication Manager Servers
 - Communication Manager Extensions
- ▣ Server Configuration
- ▣ SIP Phone Settings
- ▣ Survivable Call Processors
 - System Status
- ▣ Trace Logger
- ▣ Trusted Hosts

Edit Host

Host IP Address*

Profile Service Password*

Host Type SES combined home-edge

Parent none

Listen Protocols UDP TCP TLS

Link Protocols UDP TCP TLS

Access Control Policy (Default) Allow All Deny All

Emergency Contacts Policy Allow Deny

Minimum Registration Registration Expiration Timer (seconds)*

Subscription Expiration Timer (seconds)*

Line Reservation Timer (seconds)*

Outbound Routing Allowed Internal External

From Port UDP TCP TLS

Outbound Proxy

Outbound Direct Domains

Default Ringer Volume* Default Ringer Cadence

Default Receiver Volume* Default Speaker Volume*

VMM Server Address

VMM Server Port VMM Report Period

Fields marked * are required.

Figure 30: Edit Host

5.1.4 Add Communication Manager Media Server Interface

Under the **Communication Manager Servers** option in the Administration web interface, select **Add** to add the Avaya 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 (Figure 31), enter the following information:

- A descriptive name in the **Communication Manager 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.³
- After completing the **Add Communication Manager Server Interface** screen, click on the **Update** button.

The screenshot shows the Avaya Integrated Management SIP Server Manager web interface. The top navigation bar includes the Avaya logo, 'Help', 'Exit', and 'Integrated Man: SIP Server Ma' with a 'This Server' link. A left sidebar contains a navigation menu with categories like 'Top', 'Users', 'Address Map Priorities', 'Adjunct Systems', 'Aggregator', 'Certificate Management', 'Conferences', 'Emergency Contacts', 'Export/Import to ProVision', 'Hosts', 'IM logs', 'Communication Manager Servers' (with sub-items 'Add' and 'List'), 'Communication Manager Extensions', 'Server Configuration', 'SIP Phone Settings', 'Survivable Call Processors', 'System Status', 'Trace Logger', and 'Trusted Hosts'. The main content area is titled 'Edit Communication Manager Server Interface' and contains the following fields and options:

- Communication Manager Server Interface Name***: Input field with value '192.168.193.100'
- Host**: Input field with value '192.168.193.110'
- SIP Trunk**
 - SIP Trunk Link Type**: Radio buttons for TCP and TLS (selected)
 - SIP Trunk IP Address***: Input field with value '192.168.193.104'
- Communication Manager Server**
 - Communication Manager Server Admin Address* (see Help)**: Input field with value '192.168.193.100'
 - Communication Manager Server Admin Port***: Input field with value '5022'
 - Communication Manager Server Admin Login***: Input field with value 'init'
 - Communication Manager Server Admin Password***: Password field with masked characters
 - Communication Manager Server Admin Password Confirm***: Password field with masked characters
- SMS Connection Type**: Radio buttons for SSH (selected), Telnet, and Not Available

Below the fields, there is a note: 'Note: If the Communication Manager Server connection type is changed and the admin port value is not also changed, changing connection type to SSH will change the admin port to 5022 when Add or Update is clicked and connection type to Telnet will change admin port to 5023 when Add or Update is clicked.' At the bottom left of the form area, it says 'Fields marked * are required.' and there is an 'Update' button.

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Figure 31: Add Media Server Interface

³ 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.

5.1.5 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 32**) 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 Communication Manager 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.

- Top
- Users
 - Add
 - Default Profile
 - Delete
 - Edit
 - List
 - Password
 - Search
 - Manage All Registered Users
 - Search Registered Devices
 - Search Registered 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
- 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.

Figure 32: Add User

5.1.6 Specify Corresponding Communication Manager Extension

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

If the **Add Communication Manager Extension** checkbox was not selected, complete this step:

- In the **Add Communication Manager Extension** page, enter the extension number (e.g., **210**) corresponding to the extension previously configured on Communication Manager **Figure 22** in section 4.3. The **Communication Manager Media Server Extension** and the SIP Primary Handle are usually the same, but are not required to be.
- Select the **Communication Manager Server** interface assigned to this extension.
- Click the **Add** button.

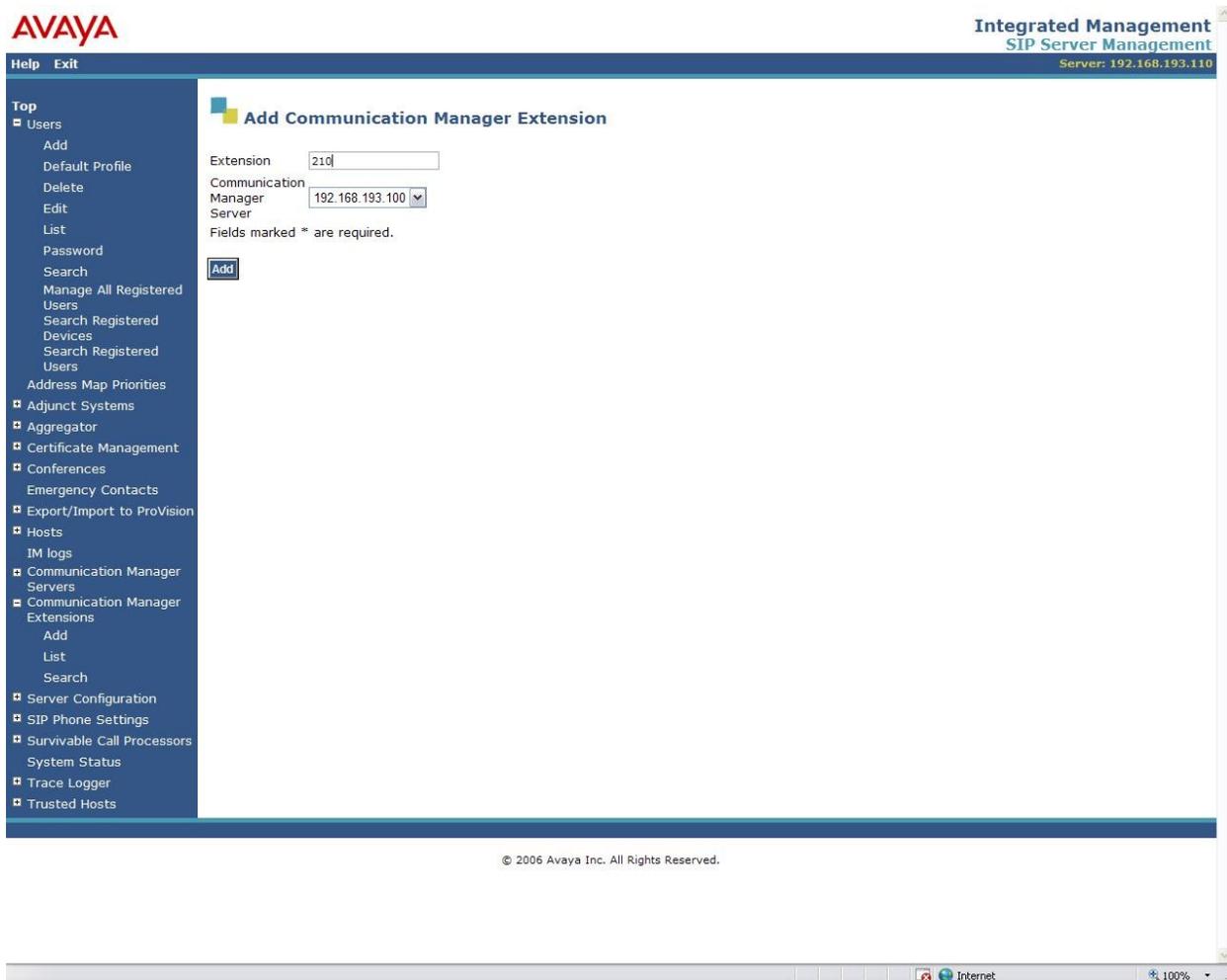


Figure 33: Add Media Server Extension

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 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 Communication Manager and 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	IP address provided by British Telecom.
Assigned DID Numbers	<i>911779080 to 911779089</i>
Codecs Supported	<i>G.729, G.711A, T38</i>

Table 2: British Telecom Spain SIP trunking service Configuration Information

7. General Test Approach and Test Results

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.

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 H.323 firmware
- Calls using Avaya 9600 Series IP Telephones with SIP firmware
- G.711A and G.729 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”)
Calling Party Number and Called Party Number presentation and restriction for incoming and outgoing calls
- Simultaneous Calls
- Long Calls

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
External Call forwarding	<ul style="list-style-type: none">- The display of the called party indicates the number of the forwarding station instead of the number of the calling station.- The SA8972 feature doesn't work with EC500. Forwarding calls through EC500 will fail.	

Item	Issue Observed	Discussion / Workaround
SIP INFO DTMF-RELAY	Wrong DTMF behaviour when DTMF tones need to be exchanged with other H323 IP PBX connected to CS2K without RFC2833 support	Avaya Communication Manager doesn't support SIP INFO messages with DTMF-RELAY information

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 a 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=TCP;user=phone SIP/2.0
From: <sip:912700131@xxx.xxx.xxx.xxx:5060;user=phone>;tag=c4-45026-2d7f51-27ff71a5-2d7f51
To: <sip:911779081@192.168.193.110:5060;user=phone>
Call-ID: a58b7a983028073e13c42d7f51b1b9556033bdd609cc047f0-0026-5067
CSeq: 1 INVITE
User-agent: CS2000_NGSS/8.0
Remote-Party-ID: <sip:912700131@xxx.xxx.xxx.xxx;user=phone>; party=calling; privacy=off; screen=yes
Allow: ACK,BYE,CANCEL,INVITE,OPTIONS,INFO,SUBSCRIBE,REFER,NOTIFY,PRACK
Via: SIP/2.0/UDP xxx.xxx.xxx.xxx:5060;maddr=xxx.xxx.xxx.xxx;branch=z9hG4bK-2d7f51-b1b95560-68c80fca
P-Asserted-Identity: <sip:912700131@xxx.xxx.xxx.xxx;user=phone>
Max-Forwards: 70
Contact: <sip:xxx.xxx.xxx.xxx:5060;transport=TCP>
Supported: 100rel
Content-Type: application/sdp
Content-Length: 208

v=0
o=MGCP 3514817881 2583700740 IN IP4 xxx.xxx.xxx.xxx
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 Communication Manager to British Telecom Spain SIP trunking:

sip:912700131@xxx.xxx.xxx.xxx:5060;lr SIP/2.0
Call-ID: 0f0afecbbedd19fb49325ab200
CSeq: 1 INVITE
From: "201" <sip:911779081@localdomain.com:5060>;tag=0f0afecbbedd19eb49325ab200
Record-Route: <sip:192.168.193.110:5060;lr>,<sip:192.168.193.104:5060;lr;transport=tcp>
To: "912700131" <sip:912700131@192.168.193.110>
Via: SIP/2.0/TCP 192.168.193.110:5060;branch=z9hG4bK030303666666030303538f.0,SIP/2.0/TLS
192.168.193.104;psrposn=2;received=192.168.193.104;branch=z9hG4bK0f0afecbbedd1a0b49325ab200
Content-Length: 217
Content-Type: application/sdp
Contact: "201" <sip:911779081@192.168.193.104:5060;transport=tcp>
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@localdomain.com:5060>
P-Charging-Vector: icid-value="AAS:192.168.193.110-615-eaff0001ddbcecc32490b9db25a"

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

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