

#### Avaya Solution & Interoperability Test Lab

# Application Notes for Configuring H.323 Trunking between British Telecom Spain H.323 Trunking service and Avaya Aura<sup>TM</sup> Communication Manager – Issue 1.0

#### **Abstract**

These Application Notes describe the steps to configure H.323 trunking between the British Telecom (BT) Spain H.323 trunking service and an Avaya IP telephony environment consisting of Avaya Aura<sup>TM</sup> Communication Manager, Avaya Aura<sup>TM</sup> SIP Enablement Services, and various Avaya telephones.

Enterprise customers with Avaya Aura™ Communication Manger 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 H.323 trunking between the British Telecom Spain H.323 trunking service and an Avaya IP telephony environment consisting of Avaya Aura<sup>TM</sup> SIP Enablement Services (SES), Avaya Aura<sup>TM</sup> Communication Manager and Avaya IP telephones using H.323 and SIP protocol as endpoints.

Customers using this Avaya IP telephony solution with the British Telecom Spain H.323 trunking service are able to place and receive PSTN calls via a dedicated broadband Internet connection using the H.323 family of protocols. This converged network solution is an alternative to more traditional PSTN analog and digital trunks. It allows customers to possibly lower local and long distance costs, add and delete DID and toll-free numbers in minutes, as well as to benefit from capabilities such as having local numbers from numerous area codes easily terminated at a single location.

### 1.1. British Telecom Spain H.323 trunking service - IP Trunking Overview

The British Telecom Spain H.323 trunking 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.

#### 1.2. Support

BT Spain: : **Phone**: 901111109

Phone from outside of Spain: +34 912706009

Email: 901111109.callbt@bt.com

**Fax:** 912706243 **Web:** <a href="http://www.bt.es">http://www.bt.es</a>

#### 2. H.323 Trunking Configuration

**Figure 1** illustrates a sample Avaya Aura<sup>™</sup> Communication Manager solution connected to the British Telecom Spain H.323 trunking service. This is the configuration that was used during the DevConnect compliance testing.

The Avaya H.323 telephony solution used as a simulated customer location included:

- Two Avaya S8730 Media Server with an Avaya G650 Media Gateway. The Avaya S8730 Media Server served as the processor for Avaya Aura<sup>TM</sup> Communication Manager.
- Avaya Aura<sup>TM</sup> 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

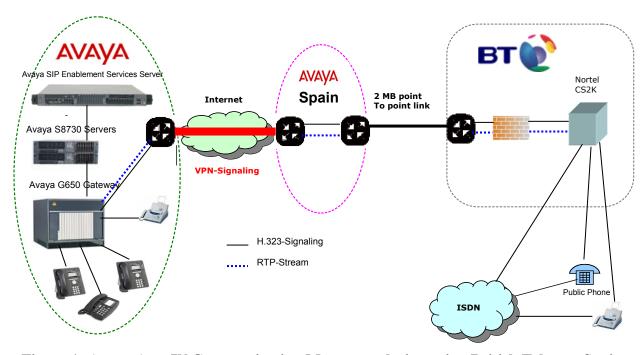


Figure 1: Avaya Aura<sup>TM</sup> Communication Manager solution using British Telecom Spain H.323 trunking service

#### 2.1. Specific Characteristic of H.323 Trunk Configuration

Communication Manager doesn't support "RAS Mode Gateway to Gatekeeper".

British Telecom Spain H.323 trunking service has to configure the H.323 trunk into "RASLess Mode Gateway to Gateway". For detecting "Loss of IP connectivity", the CS2000 uses as a keepalive, the opening of a TCP channel to port 1720 every 30 seconds.

NOTE: Avaya does not support this kind of keepalive method and uses ICMP instead. ICMP is denied within the BT Media PoP to avoid ICMP flood attacks. Communication Manager sends ICMP packets as a keepalive every 900 seconds. This keepalive only checks IP connectivity and it can't detect application problems to put H323 trunk out of service. Opening the ICMP protocol from the customer network to the CS2K network has security risks and this configuration would have to be validated by BT systems security engineering. So, whenever considering connecting to the BT H.323 trunking service, contact your BT representative earlier during the sales phase before attempting to connect to it.

#### 3. Equipment and Software Validated

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

Avaya Aura™ Communication Manager Solution Components							
Component	Software Version						
Avaya Aura <sup>™</sup> Communication Manager on	5.2 (R015x.02.0.947.3)						
S8730 Media Server	Patch: 02.0.947.3-17250						
Avaya G650 Media Gateway							
TN2312BP IP Server Interface	HW28 FW045						
TN799DP Control-LAN	HW16 FW032						
TN2602AP IP Media Processor	HW02 FW048						
TN799DP Control LAN	HW01 FW032						
TN2302AP IP Media Processor	HW21 FW120						
TN793CP Analog Line	HW07 FW010						
Avaya Aura™ SIP Enablement Services on	5.2						
S8500C	SW release string:						
	SES-5.2.0.0-947.3b						
	SW load: SES05.2-02.0.947.3b						
Avaya 9630 H.323 IP Telephones	3.0						
	Application file: ha96xxua3_00.bin						
	Boot file: hb96xxua3_00.bin						
Avaya 9630 SIP IP Telephones	2.2						
	Application file: SIP96xx_2_2_0_0.bin						
	Boot file: hb96xxua1_50.bin						
Avaya 6221 Analog Telephones	n/a						
British Telecom Spain H.323 tru	nking service Components						
Component	Version						
Nortel CS2000 Call Agent and H323 GWC	ISN10						

Table 1: Equipment and Software used in the tested configuration

The specific configuration above was used for the British Telecom Spain H.323 trunking service interoperability testing. Note that this solution is extensible to other Avaya Media Server and Media Gateway platforms running similar versions of Avaya Aura<sup>TM</sup> Communication Manager and Avaya Aura<sup>TM</sup> SIP Enablement Services.

#### 4. Configure Avaya Aura™ Communication Manager

This section describes the steps for configuring a H.323 trunk on Communication Manager with the necessary signaling and media gateway characteristics for the H.323 trunk connection with the British Telecom Spain H.323 trunking service.

The H.323 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 SIP Enablement Services (SES) but have calling privileges and features managed by Communication Manager.

H.323 signaling messages for all incoming calls from British Telecom Spain H.323 trunking service are received by Communication Manager via the H.323 trunks. Incoming call treatment, such as performing incoming digit translations, may be performed as necessary. All outgoing calls to the PSTN via British Telecom Spain H.323 trunking service are routed through Communication Manager in order to use features such as automatic route selection and class of service restrictions. Communication Manager creates the outbound H.323 signaling that is routed to British Telecom Spain H.323 trunking.

Note that Communication Manager acts as a H.323 proxy through which all incoming and outgoing H.323 messages flow to British Telecom Spain H.323 trunking 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. H.323 Trunk Configuration

#### 4.1.1 Verify System Capacity

Using the SAT, verify that sufficient H.323 Trunk capacity exists by displaying the **System-Parameters Customer-Options** form shown in **Figure 2**. 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 2, verify that the number of available **H.323 Trunks** is sufficient for the combination of trunks to the British Telecom Spain H.323 trunking service and trunks for SIP endpoints.

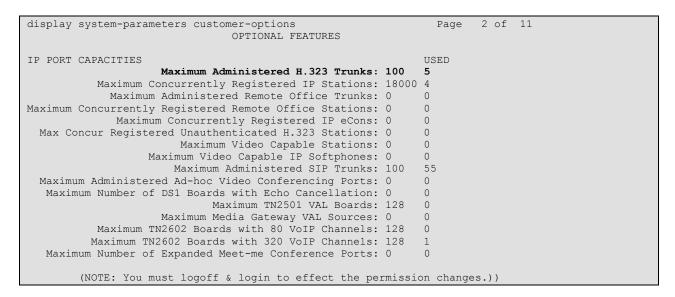


Figure 2: System-Parameters Customer-Options Form – Page 2

#### 4.1.2 Configure Special Applications

Using the SAT, verify that the necessary special applications are enabled by displaying the **System-Parameters Special-Applications** form shown in **Figure 3 and Figure 4**. 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 4 of the **System-Parameters Special-Applications** form, verify that the feature "H245 Support With Other Vendors (SA8507)" is enabled and on Page 8 the feature "Replace CPN when calls forwarded off-net (SA8983)" is enabled.

If a Communication Manager station is redirecting a call (CFU, CFB, CFNA or EC500) to an external destination and an external extension is calling this station, then Communication

Manager sends the calling party number in the outgoing SETUP message. The British Telecom Spain H.323 trunking service verifies if the calling number in the SETUP is valid for the H.323 trunk. The calling party number is not valid and the call dropped. By enabling the feature SA8972, the outgoing SETUP contains the number of the forwarding station as the calling party instead of the number of the calling party. The service provider sees this number as valid and the call will be established.

The display of the called party will indicate the number of the forwarding station instead the number of the calling station.

```
display system-parameters special-applications
                                                                       4 of
                                                                 Page
                             SPECIAL APPLICATIONS
     (SA8481) - Replace Calling Party Number with ASAI ANI? n
               (SA8500) - Expanded UUI Display Information? n
                 (SA8506) - Altura Interoperability (FIPN)? n
                (SA8507) - H245 Support With Other Vendors? y
                (SA8508) - Multiple Emergency Access Codes? n
 (SA8510) - NTT Mapping of ISDN Called-Party Subaddress IE? n
                      (SA8517) - Authorization Code By COR? n
          (SA8520) - Hoteling Application for IP Terminals? n
  (SA8558) - Increase Automatic MWI & VuStats (S8700 only)? n
                 (SA8567) - PHS X-Station Mobility over IP? n
       (SA8569) - No Service Observing Tone Heard by Agent? n
                 (SA8573) - Call xfer via ASAI on CAS Main? n
          (SA8582) - PSA Location and Display Enhancements? n
               (SA8587) - Networked PSA via QSIG Diversion? n
                         (SA8589) - Background BSR Polling? n
     (SA8608) - Increase Crisis Alert Buttons (S8700 only)? n
                       (SA8621) - SCH Feature Enhancements? n
```

Figure 3: System-parameters Special-applications Form – Page 4

```
display system-parameters special-applications
                                                                  Page
                                                                         8 of
                             SPECIAL APPLICATIONS
                     (SA8972) - Overwrite Calling Identity? n
       (SA8983) - Replace CPN when calls forwarded off-net? y
                (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? \ensuremath{\text{n}}
```

Figure 4: System-parameters Special-applications Form – Page 8

#### 4.1.3 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 5**.

**H323\_BT\_SPAIN** and **62.7.40.88** are used below as the **Name** and **IP Address** for the British Telecom Spain H.323 trunking service gateway.

```
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 192.168.193.105
medpro-cabineta 192.168.193.105
medpro2caba 192.168.193.107
proor 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 5: IP Nodes Names Form** 

#### 4.1.4 Define IP Codec Sets

British Telecom Spain H.323 trunking service supports the codecs G.729, G.711A and supports T38 for fax.

• Define a codec set (**Codec Set 1** below) to support voice and Fax calls with the British Telecom Spain H.323 trunking service. G.729 and G.711A must be included in the set.

Use the **change ip-codec-set** command (**Figure 6 and 7**) to define the codec set. On Page 2 of **ip-codec-set** form, the **Fax** mode, must be **t.38-standard**, and the **Modem** and **TTD/TTY** modes must be **off**. Default values may be used for all other fields.

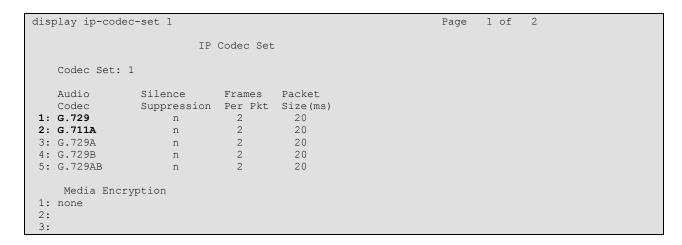


Figure 6: IP Codec Set Forms for Voice

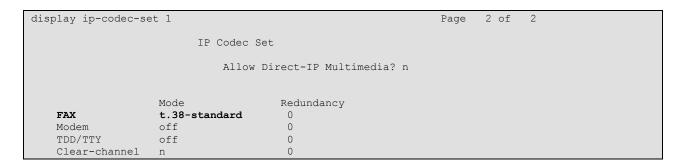


Figure 7: IP Codec Set Forms for Fax

#### 4.1.5 Configure the IP Network Region Form

The **IP Network Region** form defines the parameters associated with the IP region. In the **IP Network Region** form shown in **Figure 8**:

- By default, IP-IP Direct Audio (shuffling) is enabled to allow audio traffic to be sent directly between H323 endpoints without using media resources in the Avaya G650 Media Gateway. All H.323 endpoints are in this Network Region.
- 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
                 Authoritative Domain:
Location: 1
    Name: CLAN1
                              Intra-region IP-IP Direct Audio: yes
Inter-region IP-IP Direct Audio: no
MEDIA PARAMETERS
                                    Intra-region IP-IP Direct Audio: yes
       Codec Set: 1
UDP Port Min: 2048
UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
Call Control PHB Value: 46
Audio PHB Value: 46
RTCP MONITOR SERVER PARAMETERS
Use Default Server Parameters? y
   UDP Port Min: 2048
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
         Audio 802.1p Priority: 6
                                        AUDIO RESOURCE RESERVATION PARAMETERS
         Video 802.1p Priority: 5
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 8: IP Network Region Used for H323 phones and H.323 trunk

#### 4.1.6 Define H.323 Trunk Groups

Configure the **Signaling Group** form as shown in **Figure 9**:

- **Group Type** field set to *h.323*.
- Near End Node Name: This setting depends upon which Avaya Media Server and Avaya Media Gateway is being used. Since the Avaya S8730 Media Server is being used with the G650 Media Gateway, enter clan-cab-a-2 as the near-end node of the signalling group<sup>2</sup>.

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<sup>&</sup>lt;sup>1</sup> In this case, the near-end and far-end of the H.323 trunk are assigned to the same ip-network-region. If multiple network regions are used, Page 3 of each ip-network-region form must be used to specify the codec set for interregion communications.

<sup>&</sup>lt;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.

- Far End Node Name: Far-end is set to H323\_BT\_SPAIN (the IP address of BT Spain IP gateway).
- Near End Listen Port: Enter 1720 as the near-end listen port.
- Far End Listen Port: Enter 1720 as the far-end listen port.
- Far-end Network Region: field is set to 1, this should match the IP Network Region form created in Figure 8.
- If calls to/from H323 endpoints are to be shuffled, then the **Direct IP-IP Audio Connections** field must be set to 'y'.
- **DTMF over IP**: Set this field to **out-of-band**, the recommended setting of BT Spain.
- Trunk Group for Channel Selection: This field associates a trunk group with this signalling group. Since the trunk group has not been created yet, this field will be left blank. Once the trunk group is created as described in the next section, return to this form and add the number of the trunk group to this field.
- Default values may be used for all other fields.

```
display signaling-group 20
                                                                    1 of
                                                               Page
                               SIGNALING GROUP
Group Number: 20
                             Group Type: h.323
                          Remote Office? n
                                                  Max number of NCA TSC: 0
                                   SBS? n
                                                    Max number of CA TSC: 0
    IP Video? n
                                                Trunk Group for NCA TSC:
      Trunk Group for Channel Selection: 20
     TSC Supplementary Service Protocol: a
                                                   Network Call Transfer? n
                        T303 Timer(sec): 10
  H.245 DTMF Signal Tone Duration (msec):
  Near-end Node Name: clan-cab-a-2
                                            Far-end Node Name: H323 BT SPAIN
Near-end Listen Port: 1720
                                         Far-end Listen Port: 1720
                                     Far-end Network Region: 1
        LRQ Required? n
                                      Calls Share IP Signaling Connection? n
        RRQ Required? n
                                              H245 Control Addr On FACility? n
    Media Encryption? n
                                            Bypass If IP Threshold Exceeded? n
                                                    H.235 Annex H Required? n
        DTMF over IP: out-of-band
                                            Direct IP-IP Audio Connections? y
                                                      IP Audio Hairpinning? n
 Link Loss Delay Timer(sec): 90
        Enable Layer 3 Test? n
                                                Interworking Message: PROGress
                                        DCP/Analog Bearer Capability: 3.1kHz
```

Figure 9: Signaling Group Form

Configure the **Trunk Group** form as shown in **Figure 10** through using the **add trunk-group** command. **Trunk Group** number 20 is defined for using British Telecom H.323 trunking service in these Application Notes.

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

- Set the Group Type field to isdn.
- Choose a descriptive **Group Name** such as **H323 BT Spain**.
- Carrier Medium: Enter H.323.
- Specify an available trunk access code (TAC) such as **\*04**.
- Set the Service Type field to public-ntwrk.
- Enter 20 as the **Signaling Group** number. This value was previously determined during the **Signaling Group** configuration specified in **Figure 9**.
- Specify the **Number of Members** supported by this H.323 trunk group. Note that one trunk member is required for each call between an endpoint and the British Telecom H.323 trunking service.

```
add trunk-group 20

TRUNK GROUP

Group Number: 20
Group Name: H323 BT Spain
Direction: two-way
Dial Access? n
Queue Length: 0
Service Type: public-ntwrk

Auth Code? n

Member Assignment Method: auto
Signaling Group: 20
Number of Members: 5
```

Figure 10: Trunk Group Form – Page 1

#### On Page 2 of the **Trunk Group** form (**Figure 11**):

• Set the **CPN to Send for Redirected Calls** field to **redirecting** (appears after enabling special application SA8983). This field specifies the calling party number sent to British Telecom Spain H.323 trunking service, if a Communication Manager station has a call redirected (CFU, CFB, CFNA or EC500) to a external destination and a external extension is calling this station

```
add trunk-group 20
Group Type: isdn

TRUNK PARAMETERS
Codeset to Send Display: 0 Codeset to Send National IEs: 6
Charge Advice: none
Supplementary Service Protocol: a Digit Handling (in/out): enbloc/enbloc

Digital Loss Group: 18
Incoming Calling Number - Delete: Insert: Format:

Disconnect Supervision - In? y Out? n
Answer Supervision Timeout: 0

CONNECT Reliable When Call Leaves ISDN? n

CPN to Send for Redirected Calls: redirecting
```

Figure 11: Trunk Group Form - Page 2

#### 4.1.7 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 H.323 trunking service for calls originating from Communication Manager. The calling party number must match the value expected by the British Telecom Spain H.323 trunking service.

In these Application Notes, the extensions on Communication Manager use a 3 digit dialing plan with extensions between 200 and 249 that match to the DID numbers assigned by the British Telecom Spain H.323 trunking service. Note this requirement may vary with other numbering plans provided by British Telecom Spain.

**Figure 12** 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 20.

char	nge public-unk		ring 0 RING - PUBLIC/UNE	KNOMN EODI	Page 1 of 2
		NOPIDEI	Total	MOWN FOIN	Total
Ext	Ext	Trk	CPN	CPN	
Len	Code	Grp(s)	Prefix	Len	
					Total Administered: 7
3	200	20	911779080	9	Maximum Entries: 9999
3	201	20	911779081	9	
3	203	20	911779086	9	
3	210	20	911779082	9	
3	220	20	911779083	9	

Figure 12: Public/Unknown Numbering Form

#### 4.1.8 Configure Call Routing

#### 4.1.8.1 Outbound Calls

In these Application Notes, the Automatic Route Selection feature is used to route calls via the H.323 trunk to the British Telecom Spain H.323 trunking service, which in turn completes the calls to the BT Spain network destinations.

Use the **change dialplan analysis** command (**Figure 13**) to add **0** as a feature access code (**fac**).

change dialpl	an analy	ysis					Page	1 of	12
				DIAL PLAN ANALYSIS TABLE					
						Per	cent Fu	11:	0
Dialed	Total	Call	Dialed	Total	Call	Dialed	Total	Call	
String	Length	Type	String	Length	Type	String	Length	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 13: Dialplan Analysis Form

Use the **change feature-access-codes** command (**Figure 14**) to specify **0** as the access code for external dialing (ARS ACCESS CODE).

```
change feature-access-codes
                                                                    1 of
                                                             Page
                             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:
                                                 Access Code 2: 1
   Auto Route Selection (ARS) - Access Code 1: 0
               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:
                                                  Deactivation:
           Conditional Call Extend Activation:
                  Contact Closure Open Code:
                                                      Close Code:
```

Figure 14: Feature Access Codes Form - Page 1

Now use the **change ars analysis** command (**Figure 15**) 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].

chnage ars analysis					
	ARS DIGIT A	ANALYSIS I	REPORT		
	Loc	cation:	all	Percent Full:	0
Dialed String			Call Node Type Num	ANI Reqd	
00 8 9	3 9 2	2 j	pubu pubu pubu	n n n	

Figure 15: ARS Analysis Form

Use the **change route-pattern** command (**Figure 16**) to define the H.323 trunk groups included in the route pattern that ARS selects.

char	nae r	0111.6	e-pai	tterr	1 2						1	Page 1	of	3
					Pattern 1	Number	: 2	Patter	n Name:	SIP BT		_		
						SCCAN			re SIP?		-			
	Grp	FRL	NPA	Pfx	Hop Toll	No.	Inser	rted				DC	s/ I	IXC
	No			Mrk	Lmt List	Del	Digit	s				QS	IG	
						Dgts						In	tw	
-	20	1										n	υ	ıser
2:												n	υ	ıser
3:												n		ıser
4:												n		ıser
5:												n		ıser
6:												n	U	ıser
	BCC	' \/AT	JIE	TSC	CA-TSC	TTC	BCTE	Service	/Feature	PARM	No	Numberin	or T.Z	AR
	0 1				Request	110	DOIL	DCIVICC	, reacure			Format	9 ===	
	0 1		- "		requese						addre			
1:	уу	УУ	y n	n		rest							nc	one
	y n		_			rest							nc	one
	уn					rest							nc	one
	y n					rest							nc	one
5:	у у	у у	y n	n		rest							nc	one
6:	У У	УУ	y n	n		rest							nc	one

Figure 16: Route Pattern Form

#### 4.1.8.2 Incoming Calls

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

In these Application Notes, the British Telecom Spain H.323 trunking service sends exactly 9 digits in the incoming SETUP message. The incoming digits received are mapped 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 extension.

Use the **change inc-call-handling-trmt trunk-group** command to administer this assignment. For this example shown in **Figure 17**:

- 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-cal	l-handl	Page	1 of	30			
,		,					
Service/	Number	Number	Del	Insert			
Feature	Len	Digits					
public-ntwrk	9 9	11779080	9	200			
public-ntwrk	9 9	11779081	9	201			
public-ntwrk	9 9	11779082	9	210			
public-ntwrk	9 9	11779083	9	220			

Figure 17: Incoming Call Handling Treatment – Simple Extension Mapping

# **4.2. Save Communication Manager Changes**Enter the **save translation** command to make the changes permanent.

## 5. British Telecom Spain H.323 trunking service Configuration

In order to use British Telecom Spain H.323 trunking service, a customer must order service from British Telecom Spain H.323 trunking using their sales processes. The process can be started by contacting a British Telecom Spain H.323 trunking sales representative.

British Telecom Spain H.323 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 administration discussed in the previous sections.

Required Information	Values Used In These Application Notes			
British Telecom Spain H.323 trunking service	62.7.40.88			
H.323 Application Server				
Assigned DID Numbers	911779080 to 911779084			
Codecs Supported	G.729, G.711A			
Fax	T.38			

Table 2: British Telecom Spain H.323 trunking service Configuration Information

#### 6. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify H.323 trunking interoperability between the British Telecom Spain H.323 trunking service and a Communication Manager solution using H.323 trunking. This section covers the general test approach and the test results

#### 6.1. General Test Approach

A simulated enterprise site consisting of a Communication Manager solution supporting H.323 trunking was connected to a laboratory version of the British Telecom Spain H.323 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 H.323 trunking interoperability compliance test. All calls involved various Avaya telephones, the British Telecom Spain H.323 trunking service and various ISDN telephones.

- Outgoing calls to ISDN telephones
- Incoming calls to British Telecom Spain H.323 trunking provided DID numbers from PSTN telephones
- Calls using Avaya 9600 Series IP Telephones with the H.323 firmware
- Calls using Avaya 9600 Series IP Telephones with the SIP firmware
- G.711A and G.729 codecs for voice calls
- T38 for fax calling
- DTMF tone transmission using out-of-band with successful Voice Mail navigation
- Telephone features such as hold, transfer and conference
- Direct IP-to-IP media (also known as "shuffling") only for H323 phones. The H323 trunk needs "out of band" DTMF signaling (see 4.1.6 and Figure 9). If Direct IP-to-IP media is enabled for SIP phones, the phone signals DTMF according to RFC2833. With a SIP phone signaling "out of band", Direct IP-to-IP must disabled for SIP phones.

#### 6.2. 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
Consultative Transfer to external stations	External call via H.323 trunk to an H323/SIP phone, consultative transfer to external station, transfer can't be completed.	Problem is investigated by Avaya
Codec negotiation	Codec list: 1. G.711A, 2. G.729 Incoming call from BT with G.729 only Call failed with release complete, cause=Resources unavailable	Problem is investigated by Avaya
Codec negotiation	A H323 phone and a SIP phone are in different network regions and codec lists, shuffling is disabled. Codec list for SIP phone: G.729 and G.711A Codec list for H323 phone: G.711A and G.729 or G.729 and G.711A The H323 phone is calling the SIP phone, SIP phone goes off hook. There is no talkpath, because the SIP phone is sending RTP G.729 to MEDPRO and MEDPRO is sending RTP G.711A to SIP phone.	Problem is investigated by Avaya
Basic call with CS2K SIP Trunk interface	Voice path problem on calls Avaya> CS2K> BT Lab SIP Trunk. CS2K send wrong RTP IP Address.	Need to open a case through BT Spain to Nortel.
Codec negotiation	In calls MGCP> CS2K> Avaya, CS2K is sending G711A codec in TCS message when origin only support G729	Need to open a case through BT Spain to Nortel

Table 3: Summary of problems identified during Interoperability Testing

#### 7. Verification Steps

This section provides the verification steps that may be performed to verify operation of the Communication Manager solution with the British Telecom Spain H.323 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 H.323 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 H323 telephones are used with G.729 and 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 H323 phone and British Telecom H.323 Application Server are the media endpoints after the call is established.

#### 8. Conclusion

These Application Notes describe the configuration steps required to connect customers using an Avaya Aura<sup>TM</sup> Communication Manager solution to the British Telecom Spain H.323 trunking service using H.323 trunking. The British Telecom Spain H.323 trunking service is a Voice over IP solution for customers ranging from small businesses to large enterprises. H.323 trunking uses the H.323 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.

#### References

This section references documentation relevant to these Application Notes.

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

- [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] Avaya one-X Deskphone Edition for 9600 Series IP Telephones Administrator Guide Release 3.0, May 2009, Issue 7, Document Number 16-300698
- [5] Avaya one-X Deskphone Edition for 9600 Series SIP IP Telephones Administrator Guide Release 2.0, December 2007, Issue 2, Document Number 16-601944

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