

# IP Office 7.0 and BCM 6.0 – SIP Interoperability Configuration Notes

Abstract: This document provides information on how to configure a network solution with IP Office 7.0 and BCM 6.0 using SIP trunks.

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# **Document Publication History**

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# **1.0 Overview**

# 1.1 Capabilities

This document provides a description of the solution where a network of BCMs is connected to a network of IP Offices. In this configuration single BCM is connected to a single IP Office using SIP trunks.

Basic Call:

Basic Call Completion Handling of busy called party DTMF and ring-back tone Hold and Retrieve on both ends Call Waiting presentation Called Number display Calling number and name display Abandoned call

• Call Redirection (at node level - **no network optimization**):

Call Forward Call Transfer (blind and consultative) Call Redirection, including call redirect to voicemail

• Conferencing

3-party and multi party conferencing

- FAX
- Tandem call
- Privacy-Name and Number Blocking

# 1.2 Exclusions

The following was not tested since it is not supported:

- BCM users using IP Office voicemail system
- IP Office users using BCM voicemail system

# 1.3 Known Issues

### 1.3.1 CFAC to remote endpoint

#### **Test Procedure**

- 1. Called party has CFAC to remote set on the other switch(originating switch)
- 2. Caller calls called party on remote switch
- 3. Verify ringing on the forwarded destination
- 4. Verify displays on both sets
- 5. Answer the call at the forwarded destination and verify 2-way audio
- 6. Hang up the call and verify successful call termination

25 - 30 seconds later, set C drops the call Set A still shows the call as live (timer is still going and icon is lit)

### 1.3.2 Blind Transfer to Originating switch

The target of the of the transfer will display the external line that the call is on and not the set on the other end and the transferee will display the set that did the transfer and not the target, even after the transfer is complete.

BCMIP OfficeSet ASet BSet C|-----Originate------>|||-----Transfer----->|||||||

Display of Set B is the line number that the call is on and the display of Set C is of the originator, Set A.

#### 1.3.3 Blind Transfer to remote switch

Display appears to not function correctly. If Set A on BCM calls Set B on IP Office and then IP Office transfers to Set C on BCM, then hangs up. When Set C answers call, both Set A and Set C have their display appearing as though they are both connected to Set B who has since exited the call.

BCM	IP Office
Revision 1.0	Page 5 of 28

Display of both Set A and Set B is that it is connected with Set C

Same display problem appears if both Set A and Set B are on IP Office and Set C is on BCM.

#### **1.3.4 Consult Transfer To Originating switch**

**Note:** same display note as in test case 1.3.2

#### 1.3.5 Consult Transfer To Remote switch

**Note:** Same display note as in test case 1.3.3

,



# 1.4 Network Diagram

# 2.0 Configuration Guide

# 2.1 IP Office Software Versions

o IP Office 7.0 release

# 2.2 BCM Software Versions

This Solution Configuration Guide is applicable to the following BCM Releases:

- BCM50 rls 6.0 with the latest SU
- BCM450 Rls 6.0 with the latest SU

# 2.3 Provisioning SIP trunks on IP Office

This is only a representative configuration. For more detailed explanations, please refer to the IP Office SIP trunking guide.

ID Officer	Licence	8-	SIB Truck Observate	
IP Offices           IBOOTP (3)           Operator (3)           ra_213_new           System (1)           -f7 Line (10)           Control Unit (3)           Extension (17)           User (19)           HuntGroup (2)           Short Code (66)           Service (0)           RAS (1)           Incoming Call Route (2)           WanPort (0)           Time Profile (0)           Firewall Profile (1)           IP Route (2)           Account Code (0)           License (3)           Tunnel (0)           Lers (8)           Auto Attendant (0)           XES (1)	License Type Avaya IP endpoints IP500 Voice Networking Cham SIP Trunk Channels	Licenses License Key License Type License Status Instances Expiry Date	yyNShfbmXGPnswP75kjW4EKZnDdvUlX SIP Trunk Channels Valid 6 Never	

# 2.3.1 IP Office SIP Trunk Keycodes

# 2.3.2 IP Office SIP Line

🐮 Avaya IP Office R7 Manager ra	a_213_new	[7.0(11015)]	Administrator (Admin	istrator)]					
File Edit View Tools Help	Line	•	17	. 26-8		▲ ✓ □ ⇄ 🔞	1		
IP Offices		Line		E	SIP	Line - Line 17	,	📸 -   🗙	✔   <   >
<ul> <li>BOOTP (3)</li> <li>Operator (3)</li> <li>System (1)</li> <li>Control Unit (3)</li> <li>Extension (17)</li> <li>User (19)</li> <li>WhintGroup (2)</li> <li>Short Code (66)</li> <li>Service (0)</li> <li>RAS (1)</li> <li>Incoming Call Route (2)</li> <li>WanPort (0)</li> <li>Directory (0)</li> <li>Firewall Profile (1)</li> <li>Firewall Profile (1)</li> <li>Firewall Profile (1)</li> <li>Extense (3)</li> <li>Cucense (3)</li> <li>WanPort (0)</li> <li>User Rights (8)</li> <li>Auto Attendant (0)</li> <li>ARS (1)</li> <li>E911 System (1)</li> </ul>	Line Number ff1 ff2 ff3 ff4 ff5 ff6 ff7 ff8 17 18	Line Type Analogue Trunk Analogue Trunk Analogue Trunk Analogue Trunk Analogue Trunk Analogue Trunk Analogue Trunk SIP Line H323 Line	Line SubType	SIP Line Transport SI Line Number ITSP Domain Name Prefix National Prefix Country Code International Prefix Send Caller ID Association Method Incoming Outgoing	P URI VOIP 1 17 C bcmlab.com 0 0 None Source IP add	T38 Fax SIP Credentials	In Service Use Tel URI Check OOS Call Routing Method Originator number fo forwarded and twinn	r ing calls	est URI
	<	-101	>						
Pereived BOOTP request for 0026bb8dbb	6E 0 0 0 0 68	unable to process			Error Lis	st			< >

Some switches require a domain name match. BCM does not require it, but troubleshooting networks may be easier if this information is included.

ra_213_new	Line	-	17	- 2 - 1 -	] 🖬 🖌 🗸 🖉 🌆	1		
IP Offices		Line		E	SIP Line - Line 17		🗗 -   🗙	<ul><li>✓   &lt;</li></ul>
BOOTP (3)         Operator (3)         ra_213_new         System (1)         -?? Line (10)         Control Unit (3)         ->> Extension (17)         User (19)         HuntGroup (2)         +>> HontGroup (2)         +>>> Short Code (66)         ->>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>	Line Number 771 772 773 774 775 776 777 778 17 18	Line Type Analogue Trunk Analogue Trunk Analogue Trunk Analogue Trunk Analogue Trunk Analogue Trunk SIP Line H323 Line	Line SubType	SIP Line Transport SIP URI Vo ITSP Proxy Address 47.135. Network Configuration Layer 4 Protocol Use Network Topology Info Explicit DNS Server(s) Calls Route via Registrar Separate Registrar	IP T38 Fax SIP Credentials .151.56 UDP V None  0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0	Send Port 5 Listen Port 5	060	
	<		>			QK	Cancel	Help

## 2.3.3 IP Office SIP Line Transport

The ITSP Proxy Address would be, in this case, the IP address of the BCM that this route serves.

The BCM does not serve as a registrar, so there will be no registration credentials configured in the IP Office, and the Calls Route Via Registrar checkbox is irrelevant.

The Use Network Topology Info may cause problems if set to LAN1.

e Edit View Tools Help	Line		17	. 26-		√ ⊴ ≉ ′∎			
IP Offices		Line		E	SIP Line	- Line 17		🖻 -   🗙	🗸   <
BOOTP (3)           Operator (3)           ra_213_new           System (1)           Time (10)           Control Unit (3)           Extension (17)           User (19)           HuntGroup (2)           Short Code (66)           Service (0)           RAS (1)           Time Profile (0)           Directory (0)           Time Profile (0)           Firewall Profile (1)           Firewall Profile (1)           Firewall Profile (3)           User Rights (8)           Account Code (0)           User Rights (8)           Auto Attendant (0)           ARS (1)           E911 System (1)	Line Number 771 772 773 774 774 776 778 17 18	Line Type Analogue Trunk Analogue Trunk Analogue Trunk Analogue Trunk Analogue Trunk Analogue Trunk Analogue Trunk Analogue Trunk SIP Line H323 Line	Line SubType	SIP Line Transport SIP UP Channel Groups 1 17 17 Edit Channel Via Local URI Contact Display Name PAI Registration Incoming Group Outgoing Group Max Calls per Channel	I VoIP T38 Fax SI Via Local URI < Via Local URI < Use Internal Data Use Internal Data Use Internal Data Use Internal Data 0: <none> 17 17 10 \$</none>	Contact Display Name Contact	PAI Credential 0: <non< td=""><td>Max Calls 10</td><td>Add Remove FAIL OK Cancel</td></non<>	Max Calls 10	Add Remove FAIL OK Cancel
	<	1111	>				QK	Cancel	Help

## 2.3.4 IP Office SIP Line SIP URI

The Incoming and Outgoing Groups are referenced in the Incoming Call Route section, and the Short Code for dialing.

	a Dee	- 17		1
IP Offices	Line		SIP Line - Line 17	)
BOOTP (3)     Operator (3)     Ta_213_new     System (1)     F1 Line (10)     Control Unit (3)     Extension (17)     User (19)     HuntGroup (2)     Xhort Code (66)     Service (0)     RAS (1)     Directory (0)     Directory (0)     Directory (0)     Time Profile (1)     IP Route (2)     Account Code (0)     License (3)     Tunnel (0)     User Rights (8)     Auto Attendant (0)     ARS (1)     XAS (1)     XAS (1)     Xes (1)	Line Number     Line Type       fil     Analogue Tru       fil     SIP Line       fil     H323 Line	nk nk nk nk nk nk nk nk Compression Mode Advanced Fax Transport Support nk Call Initiation Timeout (s) DTMF Support	8 Fax SIP Credentials	<ul> <li>VoIP Silence Suppression</li> <li>Re-invite Supported</li> <li>Use Offerer's Preferred Codec</li> <li>Codec Lockdown</li> </ul>
	1			OK Cancel Help

## 2.3.5 IP Office SIP Line VoIP

Coordinate the VoIP Silence Suppression setting with the BCM's Voice Activity Detection setting. Both endpoints must have this enabled for it to be used by whichever codecs support it, e.g. G.723 and G.729.

The codec preference order for IP Office is under the Compression Mode Advanced button.

Re-Invite Supported, as BCM will re-invite to put calls on hold, and transfer.

Use Offerer's Preferred Codec should be checked so as to more closely follow the RFC3264 OFFER/ANSWER specification.

Codec Lockdown is not a requirement for BCM as it always chooses a single codec from the offered list, but this setting does no harm.

# 2.3.6 IP Office LAN1 VoIP

🐮 Avaya IP Office R7 Manag	er ra_213_new [7.0(11	015)] [Administrator(Administrator)]	
File Edit View Tools H	elp		
: ra_213_new	System	ra_213_new Image: I	
IP Offices	System	a_213_new	×   ×   <   >
<ul> <li>BOOTP (3)</li> <li>Operator (3)</li> <li>⊂ Operator (3)</li> <li>⊂ System (1)</li> <li>← { Line (10)</li> <li>⊂ Control Unit (3)</li> <li>← Extension (17)</li> <li>↓ User (19)</li> <li>← HuntGroup (2)</li> <li>← Short Code (66)</li> <li>← Service (0)</li> <li>← RAS (1)</li> <li>← Directory (0)</li> <li>← Time Profile (0)</li> <li>← Firewall Profile (1)</li> <li>← Firewall Profile (1)</li> <li>← License (3)</li> <li>← Tunnel (0)</li> <li>← User Rights (8)</li> <li>← Auto Attendant (0)</li> <li>← ARS (1)</li> <li>← E911 System (1)</li> </ul>	Name ≪ra_213_new	System       LAN1       LAN2       DNS       Voicemail       Telephony       Directory Services       System Events       SMTP         LAN Settings       VOIP       Network Topology       SIP Registrar         Image: H323 Gatekeeper Enable       SIP Trunks Enable       SIP Registrar Enable         Image: SIP Registrar Enable       SIP Registrar Enable         Image: H323 Auto-create Extn       Port Range (Minimum)       49152         Image: H323 Auto-create User       Port Range (Maximum)       53246         Image: Enable RTCP Monitoring       On Port 5005       Port Range (Maximum)       53246         Image: Enable RTCP Monitoring       DSCP(Hex)       FC       DSCP Mask (Hex)       88       SIG DSCP (Hex)         Image: B8       DSCP(Hex)       FC       DSCP Mask (Hex)       88       SIG DSCP (Hex)         Image: Differ V Settings       B8       DSCP (G3       DSCP Mask       34       SIG DSCP         DHCP Settings       Primary Site Specific Option Number (SSON)       176       Trimary Site Specific Option Number (SSON)       176	SMDR Twinning VCM CCR
<		Error List	< >
Received BOOTP request for 70f395	abda8d, 192.168.1.113:68, u	nable to process	

Ensure that SIP Trunks are enabled.

### 2.3.7 IP Office Short Code

🜃 Avaya IP Office R7 Manag	er ra_213_new [7.0(11015	)] [Administrator(Ad	lministrator)]	
<u>File E</u> dit <u>V</u> iew <u>T</u> ools <u>H</u> i	elp			
ra_213_new	Short Code	• 33N)	🛓 🖄 🖙 🖬 🖪 🔛 🖬 🗸 🛹 🏹	
IP Offices	Short Code		33N;: Dial	<b>☆</b> •   ×   <   >
<ul> <li>★ BOOTP (3)</li> <li>✓ Operator (3)</li> <li>✓ Operator (3)</li> <li>✓ System (1)</li> <li>← 7 (Line (10)</li> <li>✓ Control Unit (3)</li> <li>✓ Extension (17)</li> <li>✓ User (19)</li> <li>✓ HuntGroup (2)</li> <li>✓ Short Code (66)</li> <li>✓ Service (0)</li> <li>✓ RAS (1)</li> <li>← Time Profile (0)</li> <li>← Time Profile (0)</li> <li>← Time Profile (0)</li> <li>← Time Profile (1)</li> <li>✓ IP Route (2)</li> <li>✓ Account Code (0)</li> <li>✓ License (3)</li> <li>✓ Tunnel (0)</li> <li>✓ User Rights (8)</li> <li>✓ Auto Attendant (0)</li> <li>✓ ARS (1)</li> <li>✓ Fill System (1)</li> </ul>	Code         Telephone Num           9x*47         9x*47           9x*48         9x*50           9x*51         9x*52           9x*53*N#         N           9x*77*N#         N           9x*77*N#         N           9x*71*N#         N           9x*71*N#         N           9x*71*N#         N           9x*71*N#         N           9x*71*N#         N           9x*86         "2,1,1600"           9x*87         "1,5,12345678"           9x*88         "1,11,Get Lost!!           9x*90.0°         "MAINTENANCE           9x*90.0°         "IO151/ERR - "           9x*30.0°         30N           9x4AM         AnMaintas 20.11	Short Code Code Feature Telephone Number Line Group Id Locale Force Account Code	33N; Dial 33N 17	QK Cancel Help
< >			Error List	< >
Ready				),;;

The short code with the dial action matches any dial string that begins with '33', waits for the rest of the dial string 'N' until the inter-digit timeout ';', the sends the '33' plus the rest of the dialed number 'N' to Line Group Id '17' which was the Outgoing Group Id specified on the SIP Line // SIP URI tab.

🚹 Avaya IP Office R7 Manag	er ra_213_new [7.0(11015)] [Administra	tor(Administrator)]			
File Edit View Tools H ra_213_new	elp Incoming Call Route 17	. 26	- 🖬 💽 🔜 🗸 🖂	≉ 1	
IP Offices	Incoming Call Route		17		<b>☆ -   ×   &lt;  </b> >
<ul> <li>BOOTP (3)</li> <li>Operator (3)</li> <li>ra_213_new</li> <li>System (1)</li> <li>Control Unit (3)</li> <li>Extension (17)</li> <li>User (19)</li> <li>HuntGroup (2)</li> <li>Short Code (65)</li> <li>Service (0)</li> <li>RA5 (1)</li> <li>Directory (0)</li> <li>Time Profile (0)</li> <li>Firewall Profile (1)</li> <li>IP Route (2)</li> <li>Account Code (0)</li> <li>License (3)</li> <li>Tunnel (0)</li> <li>User Rights (8)</li> <li>Auto Attendant (0)</li> <li>AR5 (1)</li> <li>E911 System (1)</li> </ul>	Line Group Id Incoming Number Destination	Standard Voice Recordin Bearer Capability Line Group Id Incoming Number Incoming Sub Address Incoming CLI Locale Priority Tag Hold Music Source	g Destinations Any Voice 17 17 1 1 1 1 1 1 1 1 5 ystem Source		Çancel Help
< >			Error List		< >

#### 2.3.8 IP Office Incoming Call Route

Any number coming in from Group Id 17 (the Incoming Group Id from the SIP Line // SIP URI tab) will get routed to itself '.'

This destination is configured on the Destinations tab.

🖬 Avaya IP Office R7 Manage	er ra_213_new [7.0(110	015)] [Administra	ator (Adn	ninistrator)]			
File Edit View Tools He	elp Tincoming Call Route	<b>•</b> 17		• 2 8 - 9 1	• 🖳 🔝 🗸 🗸 🌾	3	
IP Offices	Incoming Ca	ll Route	E		17		×
<ul> <li>BOOTP (3)</li> <li>Operator (3)</li> <li>Ta_213_new</li> <li>System (1)</li> <li>T{ Line (10)</li> <li>Control Unit (3)</li> <li>Extension (17)</li> <li>User (19)</li> <li>HuntGroup (2)</li> <li>Short Code (65)</li> <li>Service (0)</li> <li>RAS (1)</li> <li>Time Profile (0)</li> <li>Firewall Profile (1)</li> <li>IP Route (2)</li> <li>Account Code (0)</li> <li>License (3)</li> <li>Tunnel (0)</li> <li>User Rights (8)</li> <li>Auto Attendant (0)</li> <li>K E911 System (1)</li> </ul>	Line Group Id Incoming Nu	umber Destination	Stan	dard Voice Recording Dest	nations Destination	Pallback Extension	Help
				Ē	rror List		<   3

## 2.3.9 IP Office Incoming Call Route Destination

The addition of the "." will look for an incoming digit match, User or Hunt Group. The drop down list can be used to associate an incoming call route to a specific user. The "." must be manually input, it is not part of the drop down list.

The Line Group ID relates to the 'Incoming Group' on the SIP Line SIP URI tab.

ra_213_new	<ul> <li>System</li> </ul>	🔹 ra_213_new	🖪 🖬 🗸 🧹 🖉	
IP Offices	System	E ra_2	13_new	lik -   ×   ✔   <
BOOTP (3)           Operator (3)           ra_213_new           System (1)           -f1 Line (10)           Control Unit (3)           Extension (17)           User (19)           HuntGroup (2)           M Short Code (65)           Service (0)           RAS (1)           Time Profile (0)           Directory (0)           Firewall Profile (1)           IP Route (2)           Account Code (0)           Licerse (3)           User Rights (8)           Aux Attendant (0)           XAS (1)	Name	System LAN1       LAN2       DNS       Voicemail       Telephony       Dire         Telephony       Tones & Music       Call Log       Analogue Extensions         Default Outside Call Sequence       Ring Type 1         Default Ring Back Sequence       Ring Type 2         Restrict Analogue Extension Ringer Voltage       Image: Sequence         Dial Delay Time (secs)       4         Default No Answer Time (secs)       15         Hold Timeout (secs)       120         Park Timeout (secs)       5         Call Priority Promotion Time (secs)       Disabled         Default Currency       USD         Automatic Codec Preference       G.729(a) 8K CS-ACELP	ctory Services System Events SMTP SMDR Twinning	VCM CCR
	<		QK	Cancel Hel

# 2.3.10 IP Office System Telephony

Choose the appropriate Companding Law depending on region, and the preferred Automatic Codec Preference.

# 2.4 Provisioning SIP trunks on BCM

	2.4.1	BCM	SIP	Trunk	Keycodes
--	-------	-----	-----	-------	----------

File Edit View Network	Session Tools Help y 💼 Paste 💳 Web Page 🖣	🖊 Validate Device 🧏 Discor	nnect 🛛 🔗 Refre	sh 🍘 Auto-refresh		
Element Navigation Panel	Task Navigation Panel Configuration Administration Welcome Group Josephine J	Keycodes System ID: Key Type: Region: Manufacturing SW version: Feature licenses	001598FE5792 3 Global 50.06	Sequence Date St SW Ver	e #: 13 amp: 2010-06-11 sion: Avaya BCM50 Relea	ise 6
	Administrator Access Resources Application Resource Port Ranges Telephony Resource Dial Up Interfaces Telephony Data Services Applications	Status         Name           ACTIVE         VM seat           ACTIVE         Fax Mess           ACTIVE         MODN           ACTIVE         VOIP GW           ACTIVE         IP Client           ACTIVE         NCM BCh           ACTIVE         NCM BCh           ACTIVE         Int Analo           ACTIVE         Int Analo           ACTIVE         Int Analo           ACTIVE         Int Digita           Load Keycode File         Load Keycode File	Data           4           saging           1           Trunks           seat           4           186           150a           g Tr           g Sets           4           I Sets	Expiry Date		
Done,					C:	0 🔜 M:2 🔜 m:11 🔄 W:6 💟 Include ACKed alarms

Keycodes are required in order to activate IP trunks on BCM. The 'VoIP GW Trunks' license enables H.323 and SIP trunks. There is, alternatively, a separate keycode to enable only SIP trunks.

#### 2.4.2 BCM Business Name

File Edit View Network Session Tools Help	
🐗 Exit 🛛 💥 Cut 🌇 Copy 💼 Paste 🔭 Wi	) Page 🛹 Validate Device 🧏 Disconnect 🛛 🛃 Refresh 👹 Auto-refresh
Element Navigation Panel Task Navigation Pa	el 🛛
E 🔗 Network Elements Configuration Admir	stration Feature Settings
47.135.151.55	Business Names
🔁 System	1: MyCompany 2: 3:
Administrator Acce	s 4: 5:
Telephony	
🖨 🦳 Global Settings	Feature Settings
Feature Se     Advanced B	1995 Background music: On hold: Tones 🗸 Answer keys: Basic 🗸
IP Terminal	eatures Page tone: V Held line reminder: Off V Receiver volume: Use sys volume V
• DMC Featu	List Message reply enhancement: Delayed ring transfer: After 4 rings 🔍 Directed pickup: 🔽
System Spe	d Dial Force auto/spd dial over ic/conf: Park mode: Lowest V Set relocation:
⊕ 🔂 Sets	Maximum CLI per line: 30 Alarm set: 33000
🗈 🦳 Lines	
Coops	Timers
😥 🦳 Dialing Plan	Camp timeout (sec.): 45 V Transfer callback timeout: After 4 rings V Host delay (ms.): 1000 V
Ring Groups	Park timeout (sec.): 45 🔍 Link time (ms.): 600 💌
	Page timeout (sec.): 180 👽
Done,	C:0 M:2 m:11 W:6 ☑ Include ACKed alarms

Before BCM will send CLID, a business name must be configured for the system.

#### 2.4.3 BCM Trunk Routes



There IP Office does not make use of the Phone-context in SIP messages to determine NPI/TON, so the DN Type field will be set to Public(Unknown), and will be routed using a public SIP Trunk profile.

## 2.4.4 BCM Destination Codes

🔺 Avaya Business Eleme	nt Manager - 47.135.151.!	5				
File Edit View Network :	Session Tools Help					
📲 Exit 🛛 🗶 Cut 🐚 Copy	🖷 Paste 🔚 💳 Web Page 🧹	🖊 Validate Device 🧏 Disconn	ect 🛛 🎯 Refresh 🏼 Auto-re	fresh		
Element Navigation Panel	Task Navigation Panel					~
🖃 🤣 Network Elements 🏾	Configuration Administration	Dialing Plan - Routing				
47.134.206.140	Telephony	Routes Destination Codes	Second Dial Tone			
47.135.151.55	🕀 🛅 Global Settings	Destination Codes				
- 🚔 135.20.248.146	H Collines	Destination Code 🔺	Normal Route	Absorbed Length	Wild Card: 0	1
	Loops	26	010	0		
	Scheduled Service     Dialing Plan     General     DNs     Public Networ     Private Networ     Dire Pools     Ring Groups     Call Security     Hospitality     Call Detail Record     Call Recording     Call Recording	Add Delete Alternate Routes for Destina Alternate Routes Schedule A F Evening	ation Code: 26 First Route Absort All	red Length Seco	nd Route Absorb All	ed Length
Done.				C:0M	:2m:11W:6 🕑 Include A	ACKed alarms

The leading digits are not stripped off the dialed string as they will be used by the IP Office to route the call to the destination Set or Hunt Group.

There will also be a corresponding entry for this ('26') in the SIP trunk routing table.

TExit 🐰 Cut 🗞 Co	py 뼼 Paste 🔚 Web Page 🗸	Validate Device 🧏 Disconnect 🛛 🔗 Refresh 🏉 Auto-r	efresh	
Element Navigation Panel	Task Navigation Panel         Configuration       Administration         Welcome       System         Administrator Access       Global Settings         Global Settings       Sets         Configuration       Global Settings         Sets       Lines         Dialing Plan       General         Phylic Network       Public Network         Private Network       Line Pools         Ring Groups       Call Security         Hospitality       Hunt Groups         Call Detail Recording       Call Recording	Dialing Plan - Public Network         Public Network Settings         Public Received number length:         10         Public Auto DN:         Public Network DN Lengths         DN Prefix         DN Prefix         DN Prefix         DN Length         0         11         00         12         01         11         03         11         04         11         05         12         01         11         11         11         12         13         14         15         16         17         11         13         14         15         16         17         18         1911         31         11         31         16         17         18         191         191         191         191         191	Public network dialing plan: Public (Unknown)  Public network code:  Carrier Codes  Code Prefix  ID Length 10 3 101 4  Add Delete	
Done,			C:0M:2m:11	W:6 🖌 Include ACKed alarms

# 2.4.5 BCM Public Networking

The received number length has been set to 10, assuming a full NPA/NXX/Extn North American dialing plan will be used.

On the outgoing side, the public network dialing plan type has been set to Public (Unknown) as private trunking is not supported.

General Configuration Administration Welcome System Administrator Access	Element Navigation Panel		~
	<ul> <li>Network Elements</li> <li>17.135.151.55</li> <li>Configuration Administration</li> <li>Welcome</li> <li>System</li> <li>Administrator Access</li> <li>Resources</li> <li>Application Resource</li> <li>Media Gateways</li> <li>Port Ranges</li> <li>Telephony Resource</li> <li>SIP Trunking</li> <li>Haza Trunking</li> <li>Diat Up Interfaces</li> <li>Applications</li> </ul>	General         Call Routing Summary       IP Trunk Settings         Telephony Settings         Forward redirected OLI:       Send name display:         Remote capability MWI:       Ignore in-band DTMF in RTP:	

2.4.6 BCM SIP Trunks General Settings

Set the 'Forward redirected OLI' to Last or First Redirect depending on the expected treatment from voicemail.

The 'Remote capability MWI' indicates that Message Waiting indications will be sent across the SIP trunk if there is a message for a set on the remote switch, though centralized voicemail is not supported between IP Office and BCM.

### 2.4.7 BCM SIP Trunks Public Account Basic

Add Account	
Name:	BCMtoIPO
Description:	Public Trunk - No Template
Template:	None
SIP domain:	135.20.245.56
Registration required:	
SIP username:	
Password:	
	OK Cancel

Create a public account for this network.

Enter the IP Address of the IP Office in the SIP Domain field.

nt Navigation Panel	Task Navigation Panel	-	
Network Elements	Configuration Administration	ublic Private Global Settings Media Parama	ters
	Welcome     System	Routing Table Settings Accounts ITSP Te	nplates
	Administrator Access	Name 🔺	Description
		pub1	publ
	Application Resource     Media Gateways		
	Port Ranges	Add Delete	
	Telephony Resource     Telephony Resource	Basic Advanced User Accounts	
	General		NAT Pinhole Maintenance
	SIP Trunking		
	H323 Trunking     Dial Up Interfaces		Signaling interval
	🗄 🧰 Telephony		
	Data Services		
	Applications	Support 100rel:	Session timer
		Allow UPDATE: 🗹	Session refresh method: Disable ⊻
		Use Null IP to hold:	
		Use user=phone:	
		Force E164 International dialing:	
		Enable SDP OPTIONS query:	
		Allow REFER:	Active call limit: 0
		Support Replaces: 🗹	
		Enable Connected Identity: 📃	
		Standard SIP Caps Exchange: 📃	
			ITSP association method: From header domain match 🛛 🚽 🥃
		1	

#### 2.4.8 BCM SIP Trunks Public Account Advanced

The default settings on this page, assuming the Account was not created based on a template, should be correct for successful interoperability with IP Office.

### 2.4.9 BCM SIP Trunks Public Routing Table

Add Route		
Name:	IPO_Route	
Destination Digits:	26	
ITSP Account:	BCMtoIPO	~
ОК	Cancel	

Add a route matching the '26' dialed digits from the Telephony // Dialing Plan // Routes // Destination Digits tab, and specifying the account created above.



#### 2.4.10 BCM SIP Trunks Media Parameters

Choose the codecs and put them in the preferred order.

Select whether Voice Activity Detection (Silence Suppression) will be available for those codecs which support it.

Choose whether T.38 will be preferred for fax transport.

Choose whether in-band ringback should always be provided by the BCM on incoming SIP calls.



### 2.4.11 BCM Set Line Assignments

Calls into BCM are routed to Target lines. If the BCM had been set up with the DID configuration, it would have had a target line assigned to each set, and populated the sets' intercom numbers in the Received # fields. A BCM in 'PBX'' profile will have to have these assigned and configured manually.

Calls with public, or without numbering plan information will be matched to the digits in the Public Received #.

Configure the Public OLI for each set to be used as its CLID.



2.4.12 BCM Set Line Pool Access

By default, SIP lines/trunks are put into pool BlocA on BCM. Sets do not automatically have access to this pool, so it must be added on a set by set basis.

# 3.0 References

IP Office 7.0 Installation Manual, Document Number 15-601042 https://support.avaya.com/css/P8/documents/100119958

IP Office Release 7.0 Manager 9.0, Document Number 15-601011 https://support.avaya.com/css/P8/documents/100119917

IP Office Release 6.0 System Status Application, Issue 05a, February 12, 2010 Document Number 15-601758 http://support.avaya.com/css/P8/documents/100073300

IP Office System Monitor, Document Number 15-601019 http://support.avaya.com/css/P8/documents/100073350