

Avaya Solution & Interoperability Test Lab

Application Notes for configuring Avaya Aura® Communication Manager R6.2 and Avaya Aura® Session Manager R6.2 to Interoperate with Speakerbus iD808 iTurret – Issue 1.0

Abstract

These Application Notes describe the steps required to connect Speakerbus iD808 iTurret to a SIP infrastructure consisting of Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Also described is how Avaya Aura® Communication Manager features can be made available in addition to the standard features supported in the iD808 deskstations. In this configuration, the Off-PBX Station (OPS) feature set is extended from Avaya Aura® Communication Manager to the Speakerbus iD808 iTurret, providing the iD808 deskstations with enhanced calling features.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the steps required to connect Speakerbus iD808 iTurret to a SIP infrastructure consisting of Avaya Aura® Session Manager and Avaya Aura® Communication Manager. Also described is how Avaya Aura® Communication Manager features can be made available in addition to the standard features supported by iTurret. In this configuration, the Off-PBX Stations (OPS) feature set is extended from Avaya Aura® Communication Manager to the Speakerbus iD808 iTurret, providing the iTurret deskstation with enhanced calling features.

The following table provides a summary of the supported features available on iTurret with the Avaya SIP offer. Some features are supported locally in iTurret, while others are only available with Avaya Aura® Communication Manager and Avaya Aura® Session Manager with OPS. In addition to basic calling capabilities, the Internet Engineering Task Force (IETF) has defined a supplementary set of calling features, often referred to as the SIPPING-19 [6]. This provides a useful framework to describe product capabilities and compare features supported by various equipment vendors. Additional features beyond the SIPPING-19 can be extended to iTurret using OPS.

Some OPS features listed in the following table can be invoked by dialing a Feature Name Extension (FNE). A speed dial button on iTurret can also be programmed to a FNE. Other features, such as Exclusion/Privacy and Call Forwarding, are available by using the AST (Advanced SIP Telephony) FNU (Feature Name URI). Avaya Aura® Communication Manager automatically handles many other standard features via OPS, such as call coverage, trunk selection using Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS), Class Of Service (COS), Class Of Restriction (COR), and voice messaging. Details on operation and administration of OPS can be found in References [2] and [3]. The Avaya SIP solution requires all SIP telephones to be configured in Avaya Aura® Communication Manager as OPS. Items in the table below shown in **bold** were tested using an FNU or FNE.

FEATURE	SUPPORT	ΓED	COMMENTS			
	Locally at	With Avaya	1			
	the phone	SIP Offer				
SIPPING-19 Features						
Call Hold	YES	YES				
Consultation Hold	YES	YES				
Unattended Transfer	YES	YES				
Attended Transfer	YES	YES				
Call Forward All	YES	YES	Local menu option on iTurret and FNU			
Call Forward Busy/No answer	YES	YES	Local menu option on iTurret and FNU			
Call Forward Cancel	YES	YES	Local menu option on iTurret and FNU			
3-way conferencing (3 rd party added)	YES	YES				
3-way conferencing (3 rd party joins)	YES	YES				
Find me	NO	YES	Via OPS Coverage Paths			
Incoming call screening	NO	YES	Via OPS Class Of Restriction			
Outgoing call screening	NO	YES	Via OPS Class Of Restriction			
Call Park/Unpark	NO	YES	Via OPS FNE 6300/6301			
Call Pickup	NO	YES	Via OPS FNE 6312			
Automatic Redial	NO	YES	Via OPS FNE n/a			
OPS - Selected Additional Station-Si	de Features					
Conference on answer	NO	YES	Via OPS FNE 6302			
Directed call pickup	NO	YES	Via OPS FNE 6303			
Drop last added party	NO	YES	Via OPS FNE 6304			
Exclusion/Privacy	YES	YES	Local hard key on iTurret using FNU			
Last number dialed	YES	YES	Via OPS FNE 6306			
Priority Call	NO	YES	Via OPS FNE 6307, iTurret doesn`t			
			support distinctive ring indication			
Send All Calls	NO	YES	Via OPS FNE 6308			
Send All Calls Cancel	NO	YES	Via OPS FNE 6309			
Transfer to Voicemail	NO	YES	Via OPS FNE 6310			
Whisper Page	NO	YES	Via OPS FNE 6311			

Table 1

2. General Test Approach and Test Results

To verify interoperability of the iD808 iTurret with Communication Manager and Session Manager, calls were made between iD808 deskstations and Avaya SIP, H.323 and Digital stations using various codec settings and exercising common PBX features. The telephony features were activated and deactivated using buttons and menu options on iTurret, FNEs, and FNUs.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Successful registration of iTurret with Session Manager
- Calls between iTurret and Avaya SIP, H.323, and digital stations with correct calling/called name presentation
- Direct IP-IP Media (shuffling)
- Correct SIP signaling
- G.711, G.722-64k and G.729 codec support
- COR restricted calls
- Multi appearance call handling
- Hold/Retrieve operations
- Consultation calls
- Supervised and blind transfers
- Conferencing
- Bridged appearances
- Privacy
- PSTN calls
- Proper recognition of DTMF transmissions by navigating voicemail menus
- Proper operation of voicemail with message waiting indicators (MWI)
- Extended telephony features using Communication Manager Feature Name Extensions (FNEs) shown in bold in the table above
- Exclusion/Privacy using the Exclusion FNU
- Call forwarding (busy and no-answer) and Send All Calls using Call Forwarding and Send All Call FNU`s.
- Proper system recovery after an iTurret restart and loss of IP connection
- Proper failover to alternate Session Manager

2.2. Test Results

All tests were executed succesfully.

2.3. Support

For technical support of Speakerbus products contact the Speakerbus Service Desk:

Web: http://www.speakerbus.comEmail: info@speakerbus.com

• Telephone: (646) 289-4700 in North America

+44 (0) 870 240 7252 in Europe

+65 6222 4577 in Asia

3. Reference Configuration

An Avaya S8800 Server running Communication Manager R6.2 serving H323 endpoints with an Avaya G450 Media Gateway was configured along with Session Manager R6.2 hosted on an Avaya S8800 Server serving SIP endpoints. An additional Session Manager hosted as a virtual appliance was used to provide failover in an active-active configuration. System Manager hosted

on an Avaya S8800 Server is used to provision Communication Manager and Session Manager. Speakerbus iTurrets were connected to the LAN and managed by the iManager application running on a local Windows server. Simulated connection to the PSTN was provided by an E1 QSIG trunk connected to the Avaya G450 Media Gateway. Avaya Modular Messaging provided voicemail.

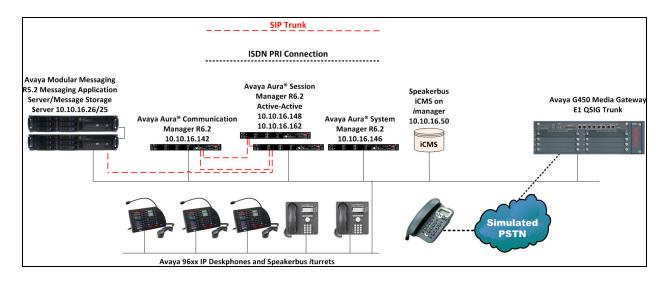


Figure 1: Avaya Aura® Communication Manager and Avaya Aura® Session Manager with Speakerbus Solution

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager	R6.2 SP5 build R016x.02.0.823.0-20396
running on Avaya S8800 Server	
Avaya Aura® Session Manager running on	R6.2 SP3
Avaya S8800 Server	
Avaya Aura® Session Manager running as	R6.2 SP3
a Virtual Appliance	
Avaya Aura® System Manager running on	R6.2 SP4
Avaya S8800 Server	
Avaya Modular Messaging running on	5.2 Patch 8
S3500 Servers	MAS - 9.2.150.13
Avaya 9630 IP Deskphone	• H323 S3.105S
	• SIP 2.6.8.4
Speakerbus iCMS with iManager	v2.100
Administration running on Windows	
Server	
Speakerbus iTurret	v2.1/ v1.40 SIP revision

5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring the iTurret as an Off-PBX Station (OPS), administering support for the OPS features indicated in **Error! Reference source not found.**, and configuring a SIP trunk between Communication Manager and Session Manager. Unless otherwise stated, administration of Communication manager is performed using the System Access Terminal (SAT).

5.1. Define System Features

Enter the command **change system-parameters features** to administer system wide features for SIP endpoints. Those related to features listed in **Error! Reference source not found.** are shown in bold. On **Page 18**, set the **Whisper Page Tone Given To** field to **all**

```
change system-parameters features
                                                              Page 18 of 19
                        FEATURE-RELATED SYSTEM PARAMETERS
INTERCEPT TREATMENT PARAMETERS
      Invalid Number Dialed Intercept Treatment: tone
                  Invalid Number Dialed Display:
   Restricted Number Dialed Intercept Treatment: tone
               Restricted Number Dialed Display:
   Intercept Treatment On Failed Trunk Transfers? n
WHISPER PAGE
  Whisper Page Tone Given To: all
6400/8400/2420J LINE APPEARANCE LED SETTINGS
                    Station Putting Call On Hold: green wink
                     Station When Call is Active: steady
        Other Stations When Call Is Put On Hold: green
              Other Stations When Call Is Active: green
                                         Ringing: green flash
                                            Idle: steady
                              Pickup On Transfer? y
```

On Page 18 make sure Directed Call Pickup is set to y.

```
Page 19 of 19
change system-parameters features
                          FEATURE-RELATED SYSTEM PARAMETERS
IP PARAMETERS
                     Direct IP-IP Audio Connections? y
                                IP Audio Hairpinning? n
                            Synchronization over IP? n
               SDP Capability Negotiation for SRTP? y
                      SIP Endpoint Managed Transfer? n
CALL PICKUP
  Maximum Number of Digits for Directed Group Call Pickup: 4
   Call Pickup on Intercom Calls? y Call Pickup Alerting? n Temporary Bridged Appearance on Call Pickup? y \bf Directed\ Call\ Pickup? y
                        Extended Group Call Pickup: none
                     Enhanced Call Pickup Alerting? n
                          Display Information With Bridged Call? n
  Keep Bridged Information on Multiline Displays During Calls? y
                   PIN Checking for Private Calls? n
```

5.2. Define the Dial Plan

Use the **change dialplan analysis** command to define the dial plan used in the system. This includes all station extensions, OPS Feature Name Extensions (FNEs), and Feature Access Codes (FACs). To define the FNEs for the OPS features listed in **Error! Reference source not found.**, a Feature Access Code (FAC) must also be specified for the corresponding feature. In the sample configuration, telephone extensions are four digits long and begin with **6**, FNEs are also four digits beginning with **6**, and the FACs have formats as indicated with a **Call Type** of **fac**.

change dial	olan analysis	Page 1 of 12
		DIAL PLAN ANALYSIS TABLE Location: all Percent Full: 1
Dialed String 0 1 2 4 5 6 7 8 9 *	Total Call Length Type 1 attd 3 fac 10 udp 4 udp 4 ext 4 ext 3 dac 4 udp 1 fac 3 fac	Dialed Total Call Dialed Total Call String Length Type String Length Type

5.3. Define Feature Access Codes (FACs)

A FAC (feature access code) should be defined for each feature that will be used via the OPS FNEs. Use **change feature-access-codes** to define the required access codes. The FACs used in the sample configuration are shown in bold.

```
change feature-access-codes
                                                             Page
                                                                    1 of 10
                              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: *14
                      Answer Back Access Code: *15
     Auto Alternate Routing (AAR) Access Code: *00
   Auto Route Selection (ARS) - Access Code 1: 9
                                                    Access Code 2:
                Automatic Callback Activation:
                                                    Deactivation:
Call Forwarding Activation Busy/DA: All: *03
                                                    Deactivation: *04
   Call Forwarding Enhanced Status:
                                         Act:
                                                      Deactivation:
                        Call Park Access Code: *01
                      Call Pickup Access Code: *02
CAS Remote Hold/Answer Hold-Unhold Access Code:
                 CDR Account Code Access Code: *51
                       Change COR Access Code:
                  Change Coverage Access Code:
           Conditional Call Extend Activation:
                                                      Deactivation:
                  Contact Closure Open Code:
                                                        Close Code:
```

change feature-access-codes	Page 2 of 10
FEATURE ACCESS CO	ODE (FAC)
Contact Closure Pulse Code:	
Data Origination Access Code:	
Data Privacy Access Code:	
Directed Call Pickup Access Code:	*16
Directed Group Call Pickup Access Code:	
Emergency Access to Attendant Access Code:	
EC500 Self-Administration Access Codes:	
Enhanced EC500 Activation:	Deactivation:
Enterprise Mobility User Activation:	Deactivation:
Extended Call Fwd Activate Busy D/A All:	Deactivation:
Extended Group Call Pickup Access Code:	
Facility Test Calls Access Code:	
Flash Access Code:	
Group Control Restrict Activation:	Deactivation:
Hunt Group Busy Activation:	Deactivation:
ISDN Access Code:	
Last Number Dialed Access Code:	*17
Leave Word Calling Message Retrieval Lock:	
Leave Word Calling Message Retrieval Unlock:	

change feature-access-codes Page **3** of 10 FEATURE ACCESS CODE (FAC) Leave Word Calling Send A Message: *86 Leave Word Calling Cancel A Message: *87 Limit Number of Concurrent Calls Activation: Deactivation: Malicious Call Trace Activation: Deactivation: Meet-me Conference Access Code Change: Message Sequence Trace (MST) Disable: PASTE (Display PBX data on Phone) Access Code: Personal Station Access (PSA) Associate Code: Dissociate Code: Per Call CPN Blocking Code Access Code: *22 Per Call CPN Unblocking Code Access Code: *23 Posted Messages Activation: Deactivation: Priority Calling Access Code: *18 Program Access Code: Refresh Terminal Parameters Access Code: Remote Send All Calls Activation: Deactivation: Self Station Display Activation: Send All Calls Activation: *19 Deactivation: *20 Station Firmware Download Access Code:

change feature-access-codes	Page 4 of 10
FEATURE ACCESS CODE (FAC)	
Station Lock Activation:	Deactivation:
Station Security Code Change Access Code:	
Station User Admin of FBI Assign:	Remove:
Station User Button Ring Control Access Code:	
Terminal Dial-Up Test Access Code:	
Terminal Translation Initialization Merge Code:	Separation Code:
Transfer to Voice Mail Access Code:	
Trunk Answer Any Station Access Code:	
User Control Restrict Activation:	Deactivation:
Voice Coverage Message Retrieval Access Code:	
Voice Principal Message Retrieval Access Code:	
Whisper Page Activation Access Code: *21	
3PCC H323 Override SIP Station Activation:	Deactivation:
PIN Checking for Private Calls Access Code:	
PIN Checking for Private Calls Using ARS Access Code:	
PIN Checking for Private Calls Using AAR Access Code:	

5.4. Define Feature Name Extensions (FNEs)

The OPS FNEs can be defined using the **change off-pbx-telephone feature-name-extensions set 1** command. The following screens show in bold the FNEs defined for use with the sample configuration.

```
change off-pbx-telephone feature-name-extensions set 1
                                                               Page
                                                                      1 of
     EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME
                     Set Name:
       Active Appearance Select:
            Automatic Call Back:
      Automatic Call-Back Cancel:
               Call Forward All:
     Call Forward Busy/No Answer:
            Call Forward Cancel:
                       Call Park: 6300
           Call Park Answer Back: 6301
                    Call Pick-Up: 6312
            Calling Number Block:
          Calling Number Unblock:
 Conditional Call Extend Enable:
Conditional Call Extend Disable:
             Conference Complete:
            Conference on Answer:
           Directed Call Pick-Up: 6303
           Drop Last Added Party:
```

```
change off-pbx-telephone feature-name-extensions set 1
                                                               Page
                                                                       2 of
     EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME
       Exclusion (Toggle On/Off):
      Extended Group Call Pickup:
          Held Appearance Select:
          Idle Appearance Select:
              Last Number Dialed: 6306
            Malicious Call Trace:
     Malicious Call Trace Cancel:
             Off-Pbx Call Enable:
            Off-Pbx Call Disable:
                   Priority Call:
                          Recall:
                  Send All Calls: 6308
           Send All Calls Cancel: 6309
               Transfer Complete:
             Transfer On Hang-Up:
          Transfer to Voice Mail:
         Whisper Page Activation: 6311
```

5.5. Configure Class of Service (COS)

Use the **change cos 1** command to set the appropriate service permissions to support OPS features (shown in bold). For the sample configuration a COS of **1** was used.

change cos-group 1											E	Page)	1	of	2
CLASS OF SERVICE COS G	p:	1	СО	S N	ame	:										
	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
Auto Callback	n	У	У	n	У	n	У	n	У	n	У	n	У	n	У	n
Call Fwd-All Calls	n	У	У	У	У	n	n	У	У	n	n	У	У	n	n	У
Data Privacy	n	У	У	n	n	У	У	У	У	n	n	n	n	У	У	У
Priority Calling	n	У	n	n	n	n	n	n	n	У	У	У	У	У	У	У
Console Permissions	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	У
Off-hook Alert	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Client Room	n	n	У	n	n	n	n	n	n	n	n	n	n	n	n	n
Restrict Call Fwd-Off Net	У	n	n	У	У	У	У	У	У	У	У	У	У	У	У	У
Call Forwarding Busy/DA	n	У	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Personal Station Access (PSA)	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Extended Forwarding All	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Extended Forwarding B/DA	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Trk-to-Trk Transfer Override	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
QSIG Call Offer Originations	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Contact Closure Activation	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n

5.6. Configure Class of Restriction (COR)

Use the **change cor n** command where **n** is the number of the COR being configured, to enable applicable calling features. To use the Directed Call Pickup feature, the **Can Be Picked Up By Directed Call Pickup** and **Can Use Directed Call Pickup** fields must be set to **y**. In the sample configuration, the *i*Turrets were assigned to COR **1**.

```
change cor 1
                                                                                              1 of
                                                                                                      23
                                                                                    Page
                                         CLASS OF RESTRICTION
                    COR Number: 1
             COR Description:
                              FRL: 0
                                                                                 APLT? y
  Can Be Service Observed? y

Calling Party Restriction: none
an Be A Service Observer? y

Called Party Restriction: none
Time of Day Chart: 1

Priority Queuing? n

Calling Party Restriction: none
Called Party Restriction: none
Direct Agent Calling? n
Can Be A Service Observer? y
      Priority Queuing? n Direct Agent Calling? n
Restricted Call Lists n

Restricted Call Lists n
      Restricted Call List? n
                                                             Can Change Coverage? n
Access to MCT? y
Group II Category For MFC: 7
Send ANI for MFE? n
                                                    Fully Restricted Service? n
                                                    Hear VDN of Origin Annc.? n
                                                      Add/Remove Agent Skills? n
              MF ANI Prefix:
                                                    Automatic Charge Display? n
Hear System Music on Hold? y PASTE (Display PBX Data on Phone)? n
                                Can Be Picked Up By Directed Call Pickup? y
                                                 Can Use Directed Call Pickup? y
                                                 Group Controlled Restriction: inactive
```

5.7. Configure SIP Trunks to each Session Manager

Enter the command **change node-names ip** and enter the **IP Address** assigned for the **procr**, **sm1** and **sm2** where **sm1** and **sm2** are the SIP signaling interfaces for each Session Manager.

change node-names	ip	Page 1 of 2
	IP NODE NAMES	
Name	IP Address	
default	0.0.0.0	
gateway	10.10.16.1	
procr	10.10.16.142	
procr6	::	
sm2	10.10.16.162	
sm1	10.10.16.148	
(15 of 15 admin	nistered node-names were displayed)	
Use 'list node-nam	nes' command to see all the administe	ered node-names
Use 'change node-n	names ip xxx' to change a node-name	'xxx' or add a node-name

Enter the command **add signaling-group 1** where signaling group 1 is the first SIP signaling group. The following are configured:

- Group Type set to sip
- Transport Method configure as tls
- Near-end Node Name enter the procr node name
- Far-end Node Name enter the sm1 node name
- Near-end Listen Port and Far-end Listen Port by default for TLS this is 5061
- Far-end Network Region set to the relevant IP Network Region, in this case 1

```
add signaling-group 1
                                                        Page
                                                               1 of
                               SIGNALING GROUP
Group Number: 1
                             Group Type: sip
 IMS Enabled? n
                       Transport Method: tls
      Q-SIP? n
    IP Video? n
                                                Enforce SIPS URI for SRTP? n
 Peer Detection Enabled? y Peer Server: SM
  Near-end Node Name: procr
                                            Far-end Node Name: sm1
Near-end Listen Port: 5061
                                          Far-end Listen Port: 5061
                                       Far-end Network Region: 1
Far-end Domain:
                                          Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate
                                                  RFC 3389 Comfort Noise? 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? y
                                              Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n
                                              Alternate Route Timer(sec): 6
```

Configure accordingly for each signaling group required to each Session Manager.

```
display signaling-group 5
                                                              Page
                                                                    1 of
                                SIGNALING GROUP
Group Number: 5
                             Group Type: sip
 IMS Enabled? n
                       Transport Method: tls
       Q-SIP? n
    IP Video? n
                                                 Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y Peer Server: SM
                                            Far-end Node Name: sm2
  Near-end Node Name: procr
Near-end Listen Port: 5061
                                          Far-end Listen Port: 5061
                                       Far-end Network Region: 1
Far-end Domain:
                                          Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate
                                           RFC 3389 Comfort Noise? n
DTMF over IP: rtp-payload
Session Establishment Timer(min): 3
                                           Direct IP-IP Audio Connections? y
                                          IP Audio Hairpinning? n
                                               Initial IP-IP Direct Media? n
        Enable Layer 3 Test? y
H.323 Station Outgoing Direct Media? n
                                               Alternate Route Timer(sec): 6
```

Enter the command **add trunk-group 1** where trunk-group 1 is the first trunk group between Communication Manager and the first Session Manager. On **Page 1** configure the following:

- Group Type enter sip
- **Group Name** enter an identifying name
- TAC enter a TAC appropriate to the dialplan
- Service Type set to public-ntwrk
- Member Assignment Method set to auto
- **Signaling Group** configure with the signaling group to the first Session Manager
- Number of Members configure as required, in this case 30

```
add trunk-group 1
                                                            1 of 21
                                                      Page
                             TRUNK GROUP
Group Number: 1
                                Group Type: sip
                                                      CDR Reports: y
                                                   TN: 1 TAC: 701
                                     COR: 1
 Group Name: TRUNK TO sm61
  Direction: two-way Outgoing Display? y
Dial Access? n
                                             Night Service:
Queue Length: 0
Service Type: public-ntwrk Auth Code? n
                                          Member Assignment Method: auto
                                                  Signaling Group: 1
                                                Number of Members: 30
```

On Page 3 configure the Numbering Format as private.

add trunk-group 1

TRUNK FEATURES

ACA Assignment? n

Measured: both

Maintenance Tests? y

Numbering Format: private

UUI Treatment: service-provider

Replace Restricted Numbers? n
Replace Unavailable Numbers? n

Modify Tandem Calling Number: no

Show ANSWERED BY on Display? y

Repeat for the SIP trunk-group to the second Session Manager.

add trunk-group 5 Page **1** of 21 TRUNK GROUP Group Type: sip Group Number: 5 CDR Reports: y Group Name: trunk to sm2 COR: 1 TN: 1 TAC: 705 Direction: two-way Outgoing Display? n Dial Access? n Night Service: Queue Length: 0 Service Type: public-ntwrk Auth Code? n Member Assignment Method: auto Signaling Group: 5 Number of Members: 30

```
add trunk-group 5
TRUNK FEATURES
ACA Assignment? n Measured: both

Numbering Format: private

UUI Treatment: service-provider

Replace Restricted Numbers? n
Replace Unavailable Numbers? n
Modify Tandem Calling Number: no

Show ANSWERED BY on Display? y
```

5.8. Configure Route Pattern

Enter the command **change route-pattern 1** where route pattern 1 is used to route calls between Communication Manager and Session Manager. Enter an identifying **Pattern Name**. Ensure that both SIP trunk-groups are configured in the **Grp No** fields and enter an **FRL** as appropriate. In the instance where all the channels in trunk-group 1 are in use, or trunk-group 1 is out of service, traffic between Communication Manager and Session Manager will route over trunk-group 5.

cha	nge 1	route	-pa	atter	n 1										Ρċ	age	1	of	3
					Pat	tern 1	Numbe:	r: 1	I	Patte	rn	Name:	to	SMs	}				
							SCCA	N? n		Sec	ure	SIP?	n						
	${\tt Grp}$	FRL	NPA	A Pfx	и Нор	Toll	No.	Inse	erte	ed							DC	S/	IXC
	No			Mrk	Lmt	List	Del	Digi	its									QSI	G
							Dgts											Int	W
1:	1	0															n	u	ser
2:	5	0															n	u	ser
3:																	n	u	ser
4:																	n	u	ser
5:																	n	u	ser
6:																	n	u	ser
		VALU				SC	ITC :	BCIE	Sei	rvice	/Fe	ature	PAI					_	LAR
	0 1	2 M	4 V	V	Req	uest									_	s For	rmat		
														Sub	add:	ress			
1:	У У	У У	y r	n n			res											n	one
2:		У У	y r	n n			res											n	one
3:	У У	У У	y r	n n			res												one
4:		У У	_				res											n	one
5:		У У	y r	n n			res											n	one
6:	У У	УУ	y r	n n			res	t										n	one

5.9. Configure IP-Codec Set

Enter the command **change ip-codec-set 1** and enter the required codecs. For the purposes of the compliance test, IP-network-region 1 uses ip-codec-set 1.

```
change ip-codec-set 1
                                                                        Page
                                                                                1 of
                              IP Codec Set
    Codec Set: 1
    Audio Silence Frames Packet Codec Suppression Per Pkt Size(ms)
1: G.722-64K
2: G.711MU
3: G.711A
                                              20
                                    2
                                               20
                                   2
                                              20
                        n
 4: G.729
                                               20
                        n
 6:
 7:
     Media Encryption
 1: none
 2:
 3:
```

5.10. Configure Private Numbering

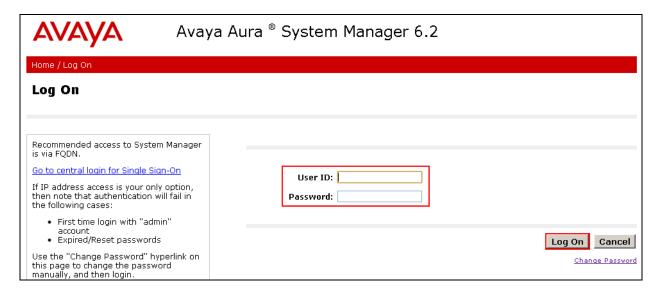
Enter the command **change private-numbering 0** and configure as follows: **Ext Len** – set to the extension length of the SIP extension number, in this case 4 **Ext Code** – set to the first digit of the SIP extension number, in this case 6 **Trk Grp** – enter the SIP trunk groups configured above, in this case 1 and 5 **Total Len** – enter the total length of the SIP extension number, in this case 4

char	nge private-numb	per 0			Page 1 o	f 2
			NUMBERING - PRIVATE	FORMA'	T	
Ext	Ext	Trk	Private	Total		
Len	Code	${\tt Grp}({\tt s})$	Prefix	Len		
4	6	1		4	Total Administered:	3
4	6	5		4	Maximum Entries:	540

6. Configure Avaya Aura® Session Manager

This section illustrates relevant aspects of the Session Manager configuration required for interoperating with Speakerbus.

Session Manager is managed via System Manager. Using a web browser, access https://<ip-addr of System Manager>/SMGR. In the Log On screen, enter appropriate User ID and Password and press the Log On button.



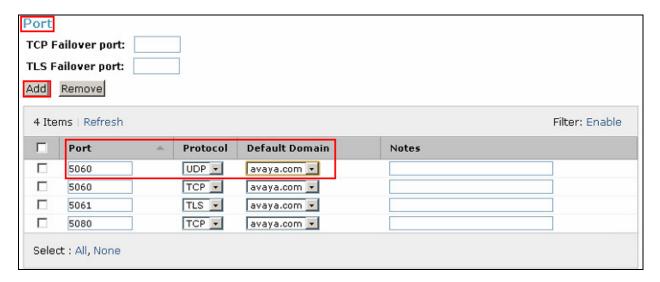
It is assumed that the Domains, Locations, SIP entities for each Session Manager, Communication Manager and Modular Messaging, Entity Links, Routing Policies, Dial Patterns and Application Sequences have been configured.

6.1. Configure UDP Port for Speakerbus Registration

Each Session Manager Entity must be configured so that the iTurret can register to it using UDP. From the web interface click **Routing** \rightarrow **SIP Entities** \rightarrow **sm1** where **sm1** is the first Session Manager entity. In the **Port** section click **Add** and enter the following:

- Port enter 5060 which is the UDP port the iTurret sends its SIP registration to
- **Protocol** select **UDP** from the drop down list
- **Default Domain** select the appropriate SIP domain from the drop down list

Click **Commit** (not shown) when done.

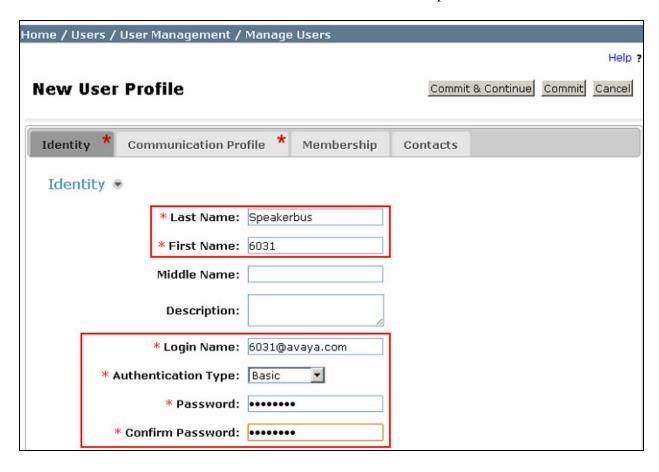


Repeat accordingly on the alternative Session Manager.

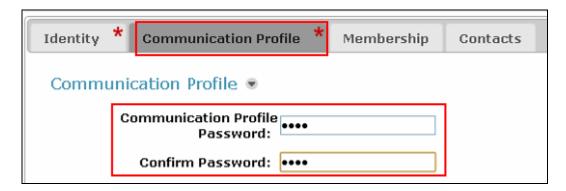
6.2. Add Primary iTurret User

A user must be added for each iTurret. Click **User Management** → **Manage Users** → **New** and configure as follows:

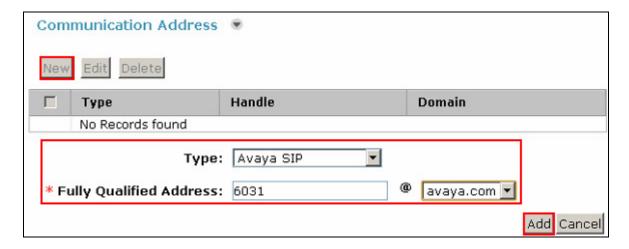
- First Name and Last Name enter an identifying name
- Login Name enter the extension number followed by the domain, in this case 6031@avaya.com
- **Authentication Type** select **Basic** from the drop down list
- Password and Confirm Password enter and confirm a password



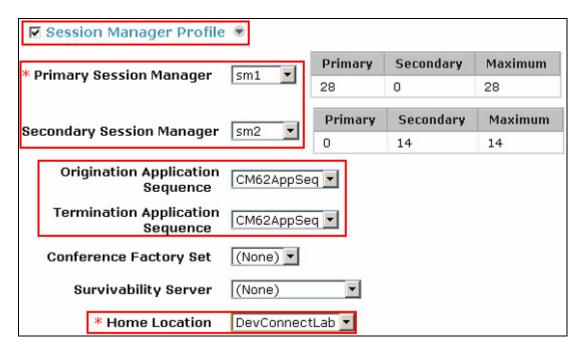
Click the **Communication Profile** tab and in the **Communication Profile Password** and **Confirm Password** fields, enter a numeric password. This will be used to register the iTurret during login.



In the Communication Address section click New. In the Fully Qualified Address field enter the extension number as required, and select the appropriate Domain from the drop down list. Click Add when done.



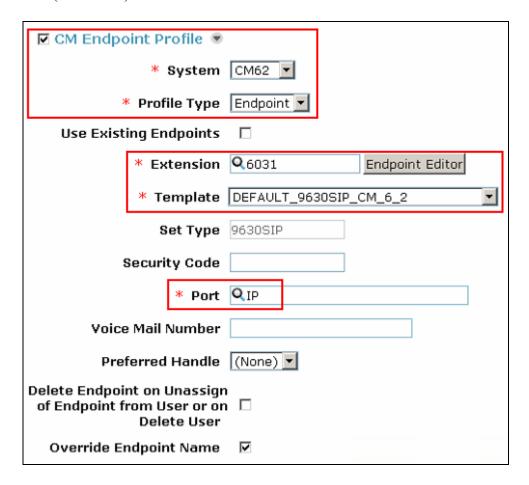
Place a tick in the Session Manager Profile check box and configure the Primary Session Manager, Secondary Session Manager, Origination Application Sequence, Termination Application Sequence and Home Location, from the respective drop down lists. The Primary and Secondary Session Manager are sm1 and sm2 respectively.



Place a tick in the **CM Endpoint Profile** check box and configure as follows:

- System select the relevant Communication Manager SIP Entity from the drop down list
- **Profile Type** select **Endpoint** from the drop down list
- Extension enter the required extension number, in this case 6031
- Template select DEFAULT_9630SIP_CM_6_2 from the drop down list
- Port enter IP

Click **Commit** (not shown) when done.

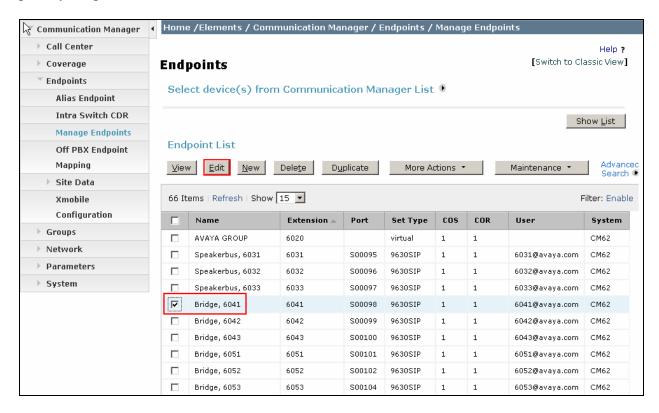


6.3. Configure Privacy Users

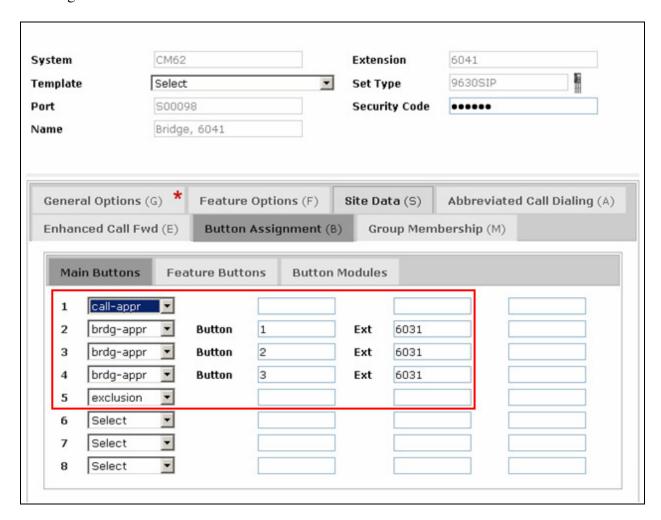
Privacy users are configured on System Manager as bridged appearances to the Primary User. Add a Privacy User in the same way as the Primary User is configured in **Section 6.2**. In this case the Privacy Users created for Extension 6031 are extensions 6041 and 6051.

6.4. Configure Privacy Endpoint

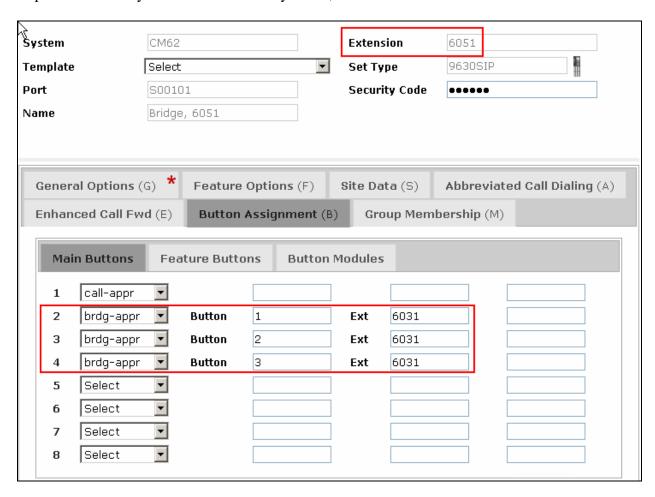
Click Communication Manager → Endpoints → Manage Endpoints and select the relevant privacy endpoint and click Edit, in this case Extension 6041.



Click the **Button Assignments** tab and configure as shown below, where buttons **2**, **3** and **4** are configured as **brdg-appr** for **Button 1**, **2** and **3** of Primary User **Extn 6031**. Ensure that button **5** is configured as **exclusion**.

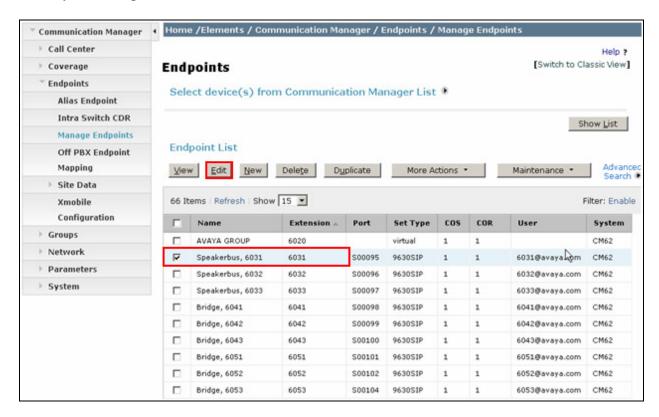


Repeat as necessary for additional Privacy Users, as shown below for Extension 6051.



6.5. Configure iTurret Endpoint

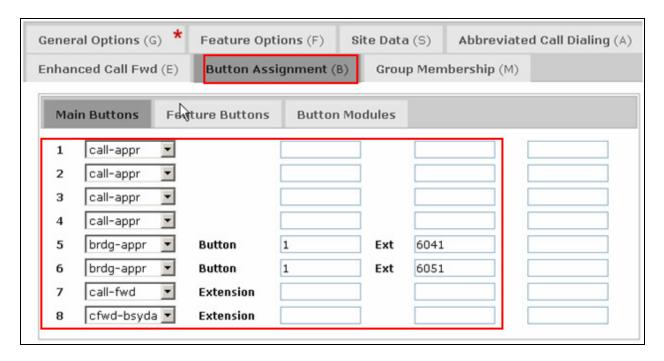
Click Communication Manager → Endpoints → Manage Endpoints and select the relevant Primary User endpoint and click Edit, in this case Extension 6031.



Click the Feature Options tab and place a check in the Bridged Call Alerting box.

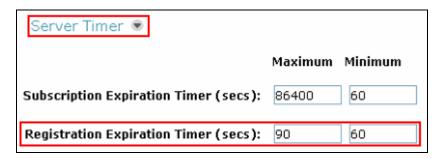
General Options (G) * Feature Options (F)	Site Data (S) Abbreviated Call Dialing (A)
Enhanced Call Fwd (E) Button Assignment	t (B) Group Membership (M)
Active Station single	Auto Answer none 🔽
MWI Served User Type Select	Coverage After system system
Per Station CPN - Send Calling Number	Display Language english
IP Phone Group ID	Hunt-to Station
Remote Soft Phone Emergency Calls	Loss Group 19
LWC Reception spe	Survivable COR internal
AUDIX Name	Time of Day Lock Table
Speakerphone Select •	
Short/Prefixed Registration Allowed	Voice Mail Number
EC500 State enabled	
_Features	
☐ Always Use	☐ Idle Appearance Preference
☐ IP Audio Hairpinning	☐ IP SoftPhone
✓ Bridged Call Alerting	✓ LWC Activation
☐ Bridged Idle Line Preference	CDR Privacy
✓ Coverage Message Retrieval	7 · ·
☐ Data Restriction	✓ Direct IP-IP Audio Connections
✓ Survivable Trunk Dest	☐ H.320 Conversion
☐ Bridged Appearance Origination Restriction	☐ IP Video
Restrict Last Appearance	Per Button Ring Control

Click the **Button Assignments** tab and configure as shown below where buttons **1-4** are configured as **call-appr**, buttons **5** and **6** are configured as **brdg-appr** of **Button 1** of the corresponding Privacy Users. Configure button **7** as **call-fwd** and button **8** as **cfwd-bsyda** in order that the FNU for these features can be activated from the *i*Turret.



6.6. Configure Registration Expiration Timer

The Registration Expiration Timer must be configured in order that SIP endpoints recover from failure of Session Manager with the least amount of downtime. Click Session Manager → Device and Location Configuration → Device Settings Groups → Default Group. In the Server Timer section configure the Registration Expiration Timer (secs) with Maximum and Minimum values. Click Save (not shown) when done. This will cause the endpoints to attempt re-registration at regular intervals. In the event that an endpoint is unable to register to its Primary Session Manager, the endpoint will attempt to register to the alternate Session Manager.



7. Speakerbus iTurret Configuration

This section provides the procedure for configuring the Speakerbus iTurret via the iManager Centralised Management System (iCMS). The iCMS comprises of three components, the iManager web portal application, the iCMS communication service and the iCMS database. The iManager web portal application consists of a series of configuration web pages that allow administrators to manage the iTurret devices. The procedure for configuring an iTurret falls into the following areas

- Launch iManager Web Portal
- Verify Product Key
- Create Site
- Create Call Region
- Create/Verify User Policies
- Create/Verify Device Policies
- Create Network Services
- Confirm Defaults
- Create iTurrets Deskstations
- Create PBX
- Create Dial Plan
- Create Call and Handset Appearances
- Create Users
- Assign User Permissions
- Assign Ownership (of Appearances to Users)
- Assign Default Call Appearances
- Program iTurrets Deskstations (iTurrets Layout)
- Program Appearances to Deskstation Keys (iTurrets Layout)
- Assign Bridged Call Appearances to Deskstations Keys (iTurrets Layout)
- Synchronize Deskstations

Note: This section displays some the configuration screens that may have already been configured.

7.1. Launch iManager Web Portal

To access the *i*Manager software interface, open a web browser and type the *i*Manager web address, for example, http://10.10.16.50/imanager. Press the **Enter** key (not show). In the iManager Web Portal logon page (not shown), enter the appropriate credentials. The iManager Web Portal home page is displayed as shown below.



7.2. Verify Product Key

Select **System** → **Product Key** in the left pane to verify that a valid key is installed and sufficient devices are allowed.



7.3. Create a Site

Configure a site representing the location where the Speakerbus iTurret devices are installed. Select **Network** → **Sites** in the left pane (not shown), click on **NEW** as shown below.



Enter an identifying **Name** for the new site, then press **OK**. As shown below.



The created site will be visible in the list view as shown below.



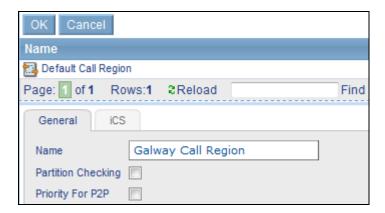
Note: A default site is available and can be used for a single site setup. Refer to the *Speakerbus iManager Administrator's Guide* for further configuration information.

7.4. Create a Call Region

Call regions represent part of an organisation's network. Select **Network** → **Call Regions** in the left pane (not shown), click on **NEW** as shown below.



Enter an indentifying **Name** for the new call region, leave the **Partition Checking** and **Priority for P2P** boxes unchecked, and press **OK**. As shown below.



The created call region will be visible in the list view as shown below.



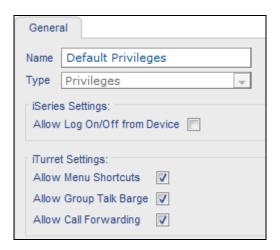
Note: A default call region is available and can be used for a single site setup. Refer to the *Speakerbus iManager Administrator`s Guide* for further configuration information.

7.5. Creating/Verifying User policies

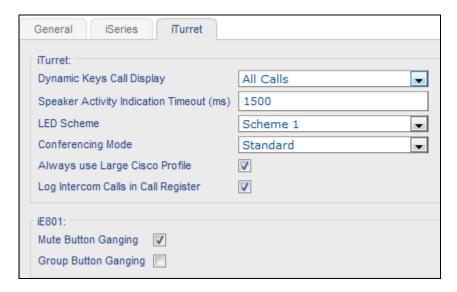
Select Users \rightarrow Policies in the left pane (not shown) and click on NEW (not shown). Enter an identifying Name, in the Type dropdown box select Voicemail, and enter a valid address for the voicemail server, in this case a pre-configured hunt group number for voicemail access is used. Click **OK** once completed, as seen below.



Select Users \rightarrow Policies in the left pane (not shown). Select and view the **Default Privileges** policy. (no changes should be needed to this, however it is referred to later in this document)

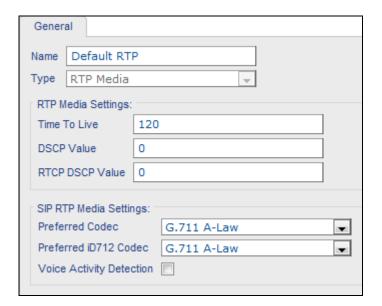


Select **Users** \rightarrow **Policies** in the left pane (not shown) Select the **Default Preferences** policy, click the **iTurret** tab and review the default settings. (no changes should be needed to this, but it's referred to later in this document)

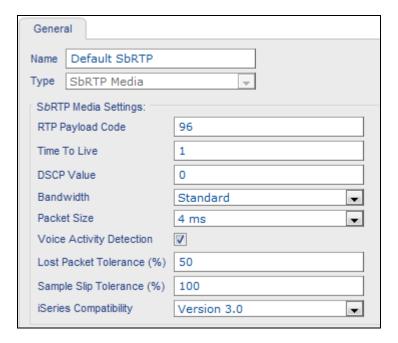


7.6. Creating/Verifying Device Policies

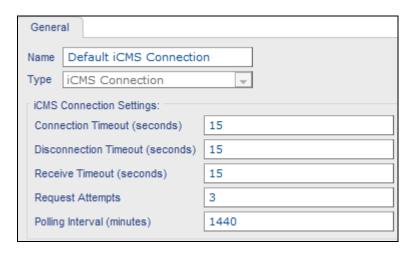
Select **Devices** \rightarrow **Policies** in the left pane (not shown). Select and view the **Default RTP** policy. (no changes should be needed to this, however it is referred to later in this document)



Select **Devices** \rightarrow **Policies** in the left pane. Select and view the **Default SbRTP** policy. (no changes should be needed to this, however it is referred to later in this document)

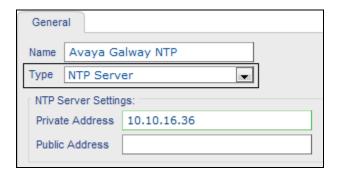


Select **Devices** → **Policies** in the left pane (not shown). Select and view the **Default iCMS** Connection policy. (no changes should be needed to this, however it is referred to later in this document)

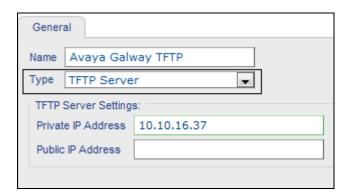


7.7. Create Network Services

Create records for the NTP and TFTP servers from the Network Services. Select **Network** → **Network Services** in the left pane (not shown), click on **NEW**, enter a descriptive **Name**, in the **Type** dropdown list select **NTP Server** and entera valid address for an NTP server if available. Press **OK** once completed, as shown below.

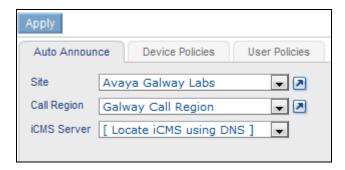


Select **Network** → **Network Services** in the left pane, click on **NEW**, enter a descriptive **Name**, in the **Type** dropdown list select **TFTP Server**, and enter a valid address for a TFTP server if available. Press **OK** once completed, as shown below.



7.8. Confirm Defaults

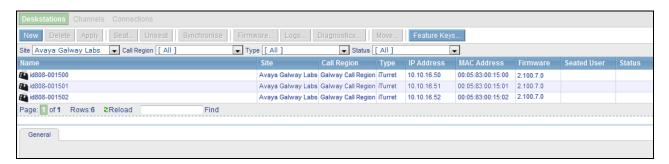
Select System → Defaults in the left pane (not shown), under the Auto Announce tab select the Site and Call Region created above and confirm that iCMS Server is set to [Locate iCMS using DNS]. Click Apply when completed as shown below:



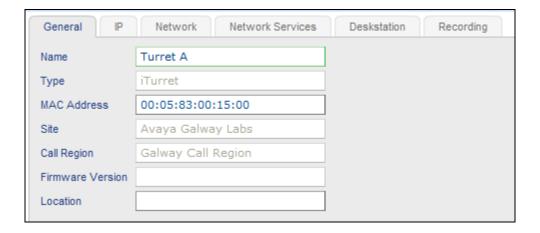
Note: DNS and DHCP must be set up in accordance with the Speakerbus administrator's guide. Refer to the *Speakerbus iManager Administrator's Guide* for further configuration information

7.9. Create iTurret Deskstations

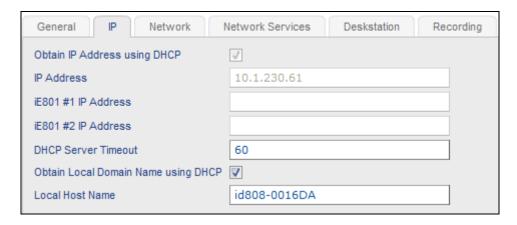
The iTurret deskstations will automatically register to the iCMS server if appropriate **DHCP** and **DNS** records were created prior to the iTurret deskstations being connected to the IP network. To view the newly registered deskstations, select **Devices** →**Deskstations** in the left pane (not shown), confirm they are seen as follows:



Select the iTurret deskstation and under the **General** tab enter an identifying **Name**.

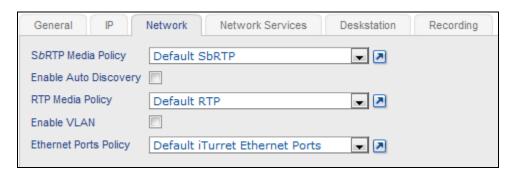


Click the **IP** tab, verify that the **Obtain IP Address using DHCP** and the **Obtain Local Domain Name using DHCP** tick boxes are checked



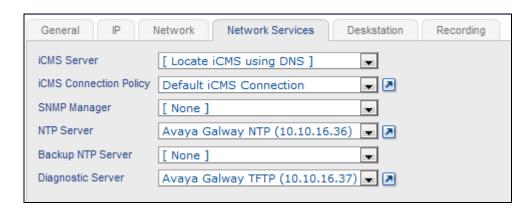
In the **Network** tab, verify the following are configured as mentioned above.:

- SbRTP Media Policy is set to Default SbRTP
- **RTP Media Policy** is set to **Default RTP** (use the link to go to the policy to change the audio codec used, default is G.711 A-law)
- Ethernet Ports Policy is set to Default iTurret Ethernet Ports



In the **Network Services** tab, verify or configure the following:

- iCMS Server shows [Locate iCMS using DNS]
- iCMS Connection Policy shows Default iCMS Connection
- NTP Server, select the newly created NTP server network service configured above
- **Diagnostics Server**, select the newly created TFTP server network service configured above



In the **Deskstation** tab ensure that **Enable Live Update** is checked and **Time Zone** and **Dial Tone Locale** are changed to the required setting. Click on **Set Administration Password.**



Enter a valid password and press **OK**.



The Deskstation tab contents now displays the following:



7.10. Create PBX (SIP Server)

To create a PBX, select **Call Servers** \rightarrow **PBXs**, click **NEW** (not shown) and complete the following fields:

- Provide a descriptive **Name** for the SIP/PBX server
- Select **Avaya** from the **Type** dropdown box
- In the **Registrar Address** and **SIP Domain** fields set to a FQDN address and domain respectively if using DNS to resolve the Session Manager active IP address
- After the PBX is created, the **Port** field will be displayed on this page with the default value of 5060



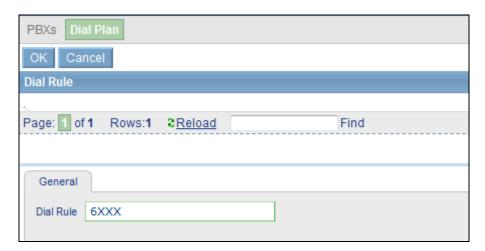
Note 1: A server locater record (SRV) for the registrar address and SIP domain must be created on DNS. Refer to the *Speakerbus iManager Administrator`s Guide* for the correct configuration of DNS

Note 2: If using failover, then a second PBX will be created and added to the **Secondary PBX** dropdown box.

The **Outbound** and **Inbound** tabs are left with their default values, Click **OK**.

7.11. Create Dial Plan

To create a PBX specific dial plan, select **Call Servers** → **PBXs** (not shown), select the **Dial Plan** tab, click **NEW** and then fill in the **Dial Rule**. Press **OK** when completed.



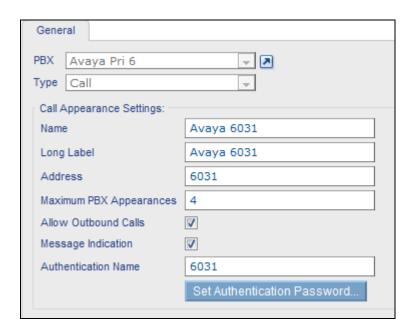
Repeat this for all valid extension formats.

7.12. Create Call and Handset Appearances

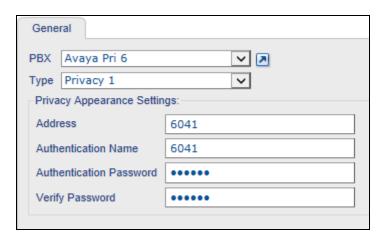
Three call appearances must be created for each iTurret device. One is for the main appearance, and one for each of the privacy appearances (handset 1 and handset 2). As previously explained, three extensions are configured in System Manager for this purpose.

To create the main appearance, click Call Servers → PBX Appearances in the left pane, click on NEW, select the PBX created in Section 7.10 (in this case Avaya Pri 6), then select the Type of appearance to be created (Call in this case) (not shown) and configure as follows under the General tab:

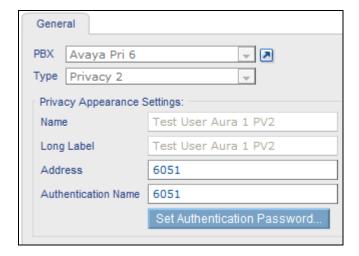
- Provide a descriptive name for the appearance in the **Name** field, such as the extension or user's name.
- Set the **Long Label** field to the label that will be displayed for the call appearance button on the iTurret deskstation. The **Address** field should also be set to the appearance extension.
- Set the Maximum Appearance field to the number of call appearances configured on the station in System Manager (the number of call appearance buttons dictates the number of calls on the system the user can have directed to them). When all of the call appearances are not idle the user is considered busy and no further calls can be routed to them. Up to a maximum of 10 call appearances may be configured on Communication Manager for each iTurret deskstation.
- Check the Message Indication checkbox for voice mail purposes and the Allow Outbound Calls.
- The **Authentication Name** and **Authentication Password** fields should be set to the extension and password configured on System Manager in **Section 6.2** These are the credentials that the iTurret deskstation will use to authenticate and register with Session Manager. Use the default values for the other fields. Click **OK**.



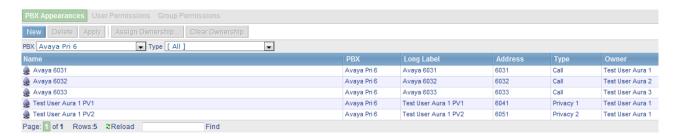
Repeat the procedure for the two corresponding privacy appearances. Click the **New** button to add another appearance. In the **General** tab select the **PBX** created in **Section 7.10**, set the **Type** field to **Privacy 1** and complete the **Address**, **Authentication Name** and **Authentication Password** fields. The last two fields should be identical to the setup in System Manager for registration to occur. Press **OK** to commit the created appearance.



Repeat the above procedure to add the Privacy 2 appearance.

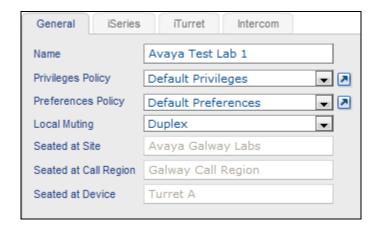


Repeat the above procedures for adding the Main and Privacy appearances for each iTurret.

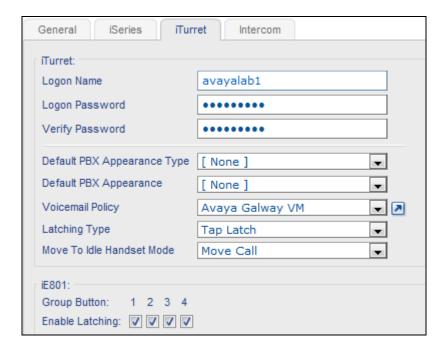


7.13. Create Users

Select Users → Users in the left pane, click on NEW, within the General tab fill in a descriptive name for the user, leave the **privilege** and **preference policies** at the defaults along with **local muting**:



Within the **iTurret** tab, provide the **logon** credentials for the user to log into their iTurret deskstation and assign the **Voicemail Policy** set up in **Section 7.5**.

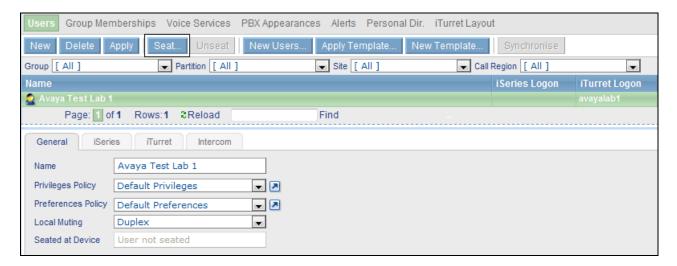


Click **APPLY** (not shown) once completed (although, this page will be revisited later to configure the default call appearance for this user).

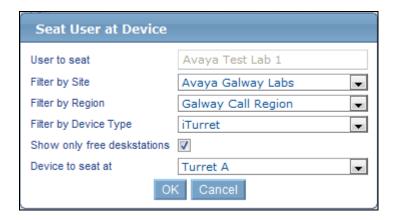
Repeat the previous steps to add more users.



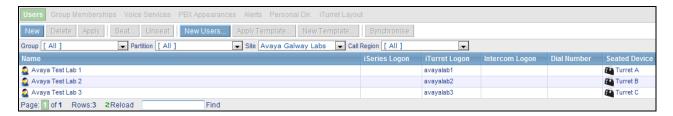
After a user has been created, that user can then be seated on an iTurret deskstation. Select the user to be seated and click **Seat** from the bar as shown below.



On the next page, filter options are presented. Filter for **iTurret** deskstations in the site configured in **Section 7.3** and the region configured in **Section 7.4** and place a tick in the **Show only free deskstations** check box. Select the appropriate iTurret device from the **Device to seat at** drop down box and click **OK**.

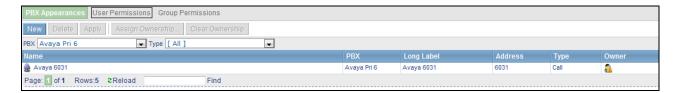


The user has been successfully seated as indicated by the iTurret deskstation in the **Seated Device** column on the following page. Repeat this process for seating all other users.

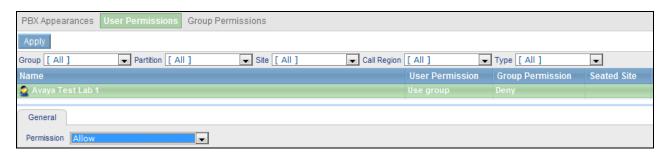


7.14. Assign User Permissions

Appearance permissions must be assigned to the created users. Select **Call Servers** →**PBX Appearances** in the left pane (not shown), select the **Call Appearance** from the list, and select the **User Permissions** tab at the top of the page:

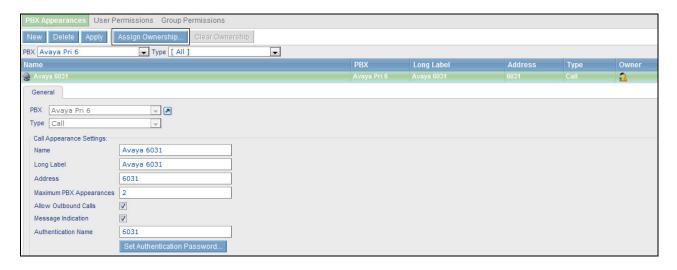


Select the user to give permissions to and select **Allow** from the **Permissions** dropdown box.



7.15. Assign Ownership

Appearance ownership must be assigned to a user as it enables the iTurret to distinguish between the owner of the call or appearance as opposed to someone who is bridged on to that appearance. Select Call Servers >PBX Appearances in the left pane, select the Call Appearance from the list, and select the Assign Ownership button:



The following screen will appear allowing filtering of users. Filter accordingly and select the user from the **User to assign ownership to** dropdown box. Click **OK**.

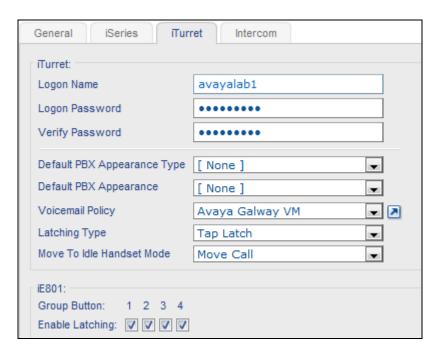


Repeat the process in both **Section 7.14** and this Section and to assign Privacy 1 and Privacy 2 call appearances to the user.

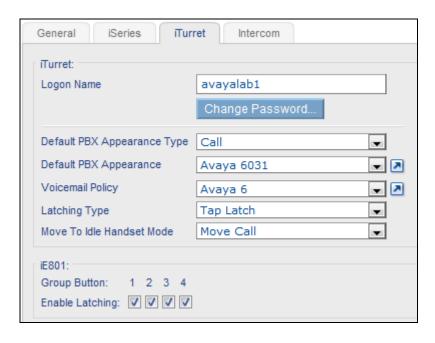


7.16. Assign Default Call Appearance

Select Users \rightarrow Users in the left pane (not shown), select the user to add a default call appearance to (not shown), and select the **iTurret** tab (as seen below):



Set the **Default PBX Appearance Type** field to **Call** and then set the **Default PBX Appearance** field to the main call appearance (e.g. 6031), Click **Apply**.



7.17. Program iTurret Deskstations

This section describes how to create iTurret deskstation keys. In this configuration, each user will be configured with two Dynamic keys, two Soft Function keys, one function (DND) key and one Shortcut key. Select **Users** → **Users** in the left pane (not shown), select the user to be updated (not shown), then select the **iTurret Layout** tab as shown below:

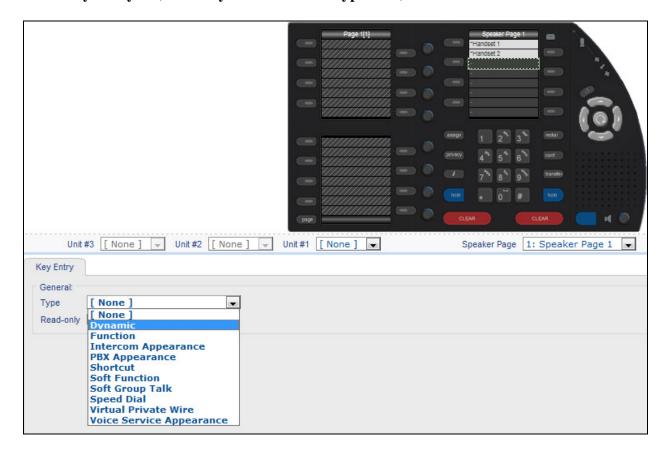


Create two dynamic keys (one after the other) under Handset 2.

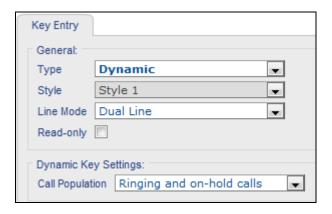
To add the first dynamic key, select the next available fixed key below Handset 2 as seen below:



In the **Key Entry** tab, select **Dynamic** from the **Type** field, as seen below:



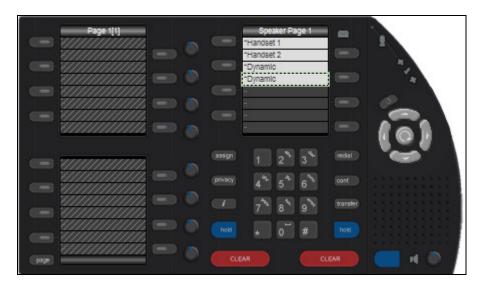
Leave the Call Population field at the default Ringing and on-hold calls. Click OK.



The iTurret layout looks as follows with the first dynamic key assigned:



Now add the second dynamic key under the first by following the steps above, once completed the iTurret layout will look as follows:



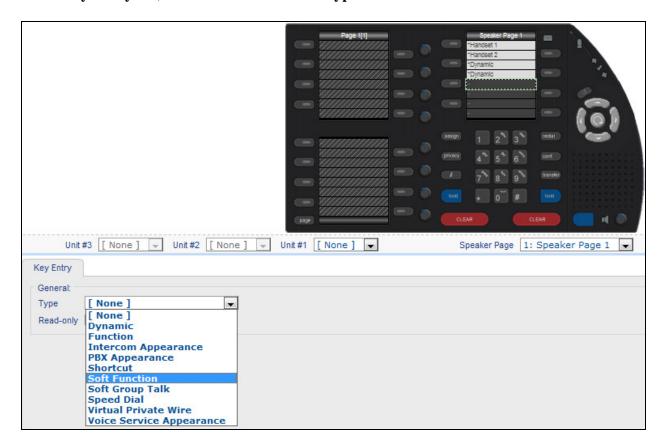
NOTE: The **Call Population** field can be set up for different key assignments, which are "Ringing and on-hold calls", "Ringing calls only", "On-hold calls only", "Busy-elsewhere calls only" and "Busy-elsewhere and on-hold calls". The default is "Ringing and on-hold calls", but any combination of these can be used depending on the user requirement.

Create two soft function keys under the second dynamic key.

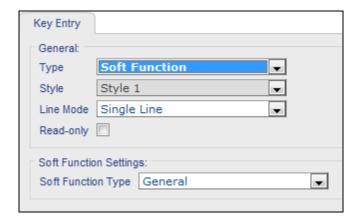
To add the first soft function key, select the next available fixed key as seen below:



In the Key Entry tab, set Soft Function in the Type field.



Leave the **Soft Function Type** field at the default **General**. Click **OK** (not shown)



The iTurret layout looks as follows with the first soft function key assigned:



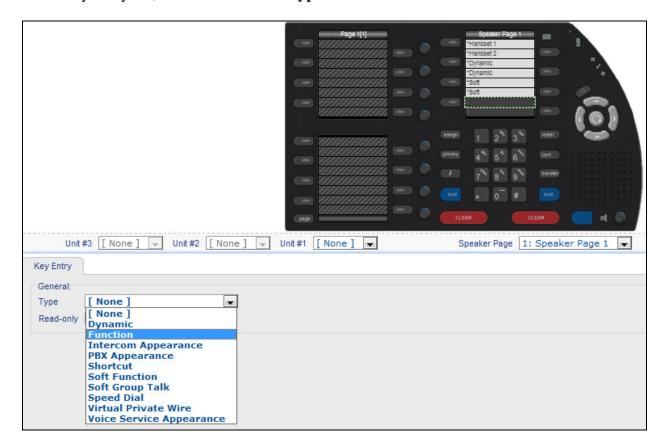
Now add the second soft function key under the first by following the steps above, once completed the iTurret layout will look as follows:



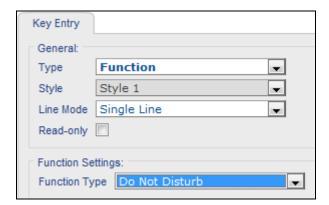
To create a function (Do Not Disturb (DND)) key, select the next available fixed key under the last soft function key, as seen below:



In the Key Entry tab, set **Function** in the **Type** field.



Select Do Not Disturb from the Function Type dropdown box. Click OK.



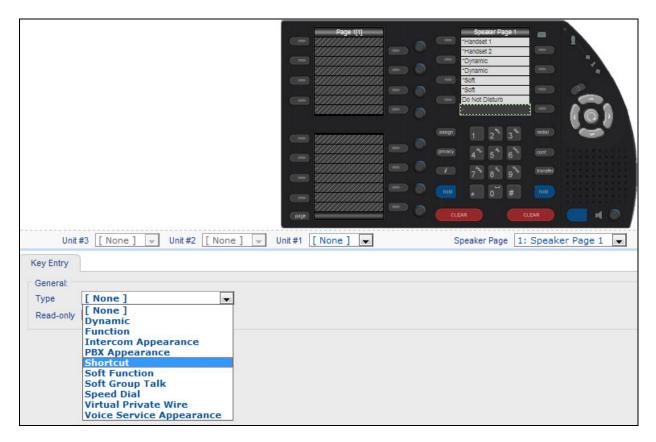
The layout looks as follows:



To create a menu shortcut key, select the next available fixed key under the last function key, as seen below:



In the Key Entry tab, set **Shortcut** in the **Type** field, as seen below:



Select $Menu\ Shortcut\ from\ the\ Shortcut\ Type\ dropdown\ box\ and\ then\ Directory\ Shortcut\ from\ the\ Shortcut\ dropdown\ box\ .$ Click OK



The iTurret layout will appear as shown below when completed.



7.18. Program Appearances to iTurret Deskstation Keys

This section describes how to create appearance keys for the iTurret deskstation.

Select Users → Users in the left pane (not shown), select the user to be configured, click the iTurret Layout tab and ensure the default page 1:Page 1 is selected from the Page dropdown box, as shown below:

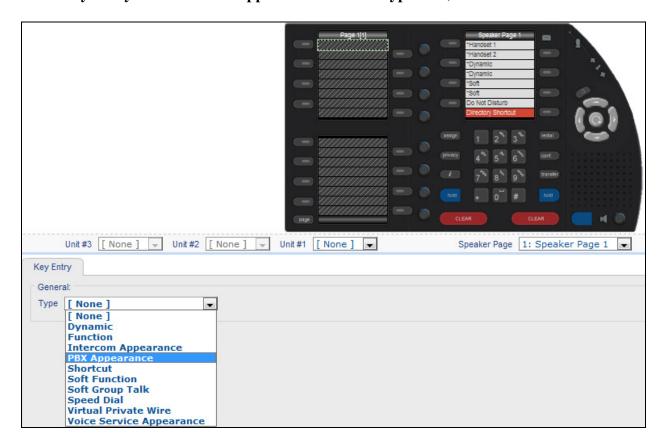


Select the first key on the top left display, as highlighted by a white box, as show below:



The next three keys on this page will be assigned to call appearances.

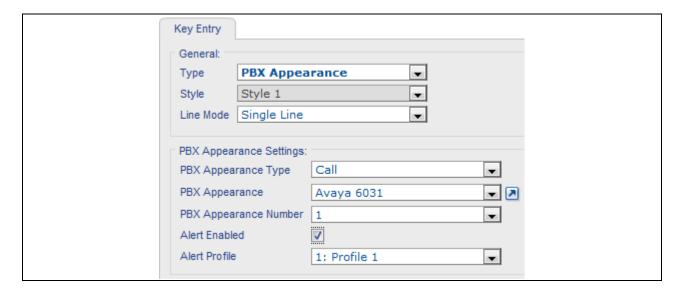
In the **Key Entry** tab select **PBX Appearance** from the **Type** field, as seen below:



Configure the following (as seen on the next screenshot):

- Under the PBX Appearance Settings, select the PBX Appearance Type (Call in this case)
- PBX Appearance select a configured PBX Appearance, in this case Avaya 6031
- **PBX Appearance Number** (1 in this case)
- Check Alert Enabled
- Leave 1: Profile 1 as default for Alert Profile

Once completed, Click **OK**.



NOTE: The **PBX Appearance Number** setting will allow the user to be configured with multiple instances of the same extension number thus allowing the user to make and receive multiple calls to and from the same extension number. The **PBX Appearance Number** relates to the **Maximum PBX Appearance** setting configured in **Section 7.12**, which governs how many instances of the extension number are allowed. The **Maximum PBX Appearance** is related to the number of call-appr keys added as feature buttons on the endpoint in System Manager configured in **Section 6.5**. In this example, 4 call-appr keys are administered in System Manager on endpoint 6031, thus the **Maximum PBX Appearance** value in iManager is configured to 4, which allows up to four instances of 6031 to be added on the iTurret layout and have up to four calls to and/or from the iTurret.

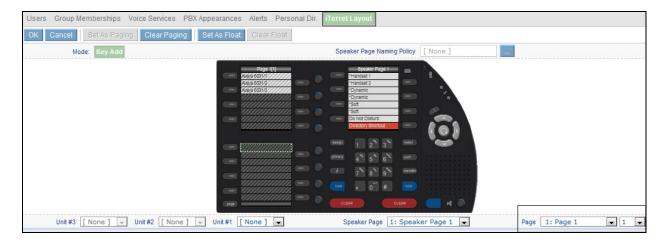
Repeat this procedure to add the next two call appearances and the layout looks as follows:



7.19. Assign Bridge Call Appearances to iTurret Deskstation Keys

This section describes how to create bridged appearance keys for the iTurret deskstation. Make sure permissions have been set for the call appearance being bridged to this user.

Select Users → Users in the left pane (not shown), select the user to be configured, click the iTurret Layout tab and ensure the default page 1:Page1 is selected from the Page dropdown box, as shown below:

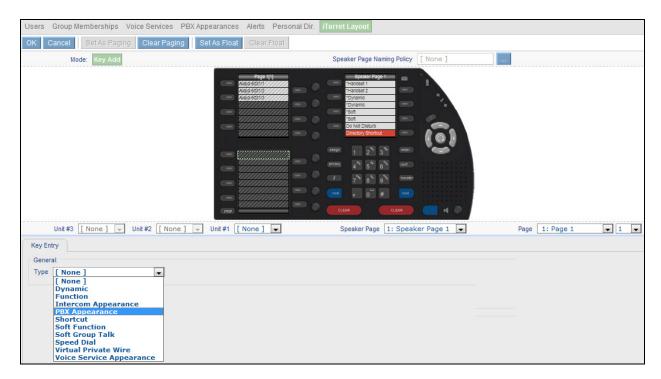


Select the next available key as highlighted by the white box below in the screenshot below.



The next three keys on this page will be assigned to bridged call appearances.

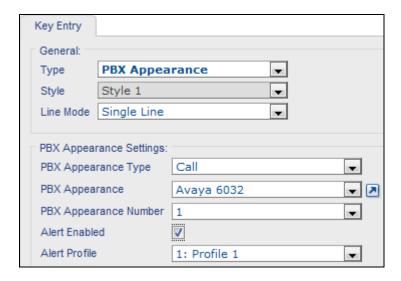
In the **Key Entry** tab configure **PBX Appearance** in the **Type** field as shown in the screenshot below.



Configure the following (as seen on the next screenshot):

- PBX Appearance Type select a configured PBX Appearance, in this case Avaya 6032
- **PBX Appearance Number** (1 in this case)
- Check **Alert Enabled** and leave **1: Profile 1** as default for **Alert Profile**.

Once completed click **OK**.



NOTE: The **PBX Appearance Number** setting will allow the user to be configured with multiple instances of the same extension number thus allowing the user to make and receive multiple calls to and from the same extension number. The PBX Appearance Number relates to the **Maximum PBX Appearance** setting configured in **Section 7.12**, which governs how many instances of the extension number are allowed. The **Maximum PBX Appearance** is related to the number of call-appr keys added as feature buttons on the endpoint in System Manager configured in **Section 6.5**. In this example, 4 call-appr keys are administered in System Manager on endpoint 6032 then the **Maximum PBX Appearance** value in iManager is configured to 4, which allows up to four instances of 6032 to be added on the iTurret layout and have up to four calls to and/or from the iTurret.

Repeat this procedure to add all the bridged call appearances and the layout appears as follows:



7.20. Synchronise Deskstations

With Live updates enabled in **Section 7.9** synchronise an iTurret device to push the new configuration to the iTurret without disruption to the user. Select **Devices → Deskstations** (not shown) and select the desired deskstations and click the **Synchronise** button. The iTurret deskstations will indicate that they are being synchronized on their displays. After the deskstations have been synchronized, the status icons on the iTurret deskstations corresponding to the network, iCMS, and SIP registrar status will be green.



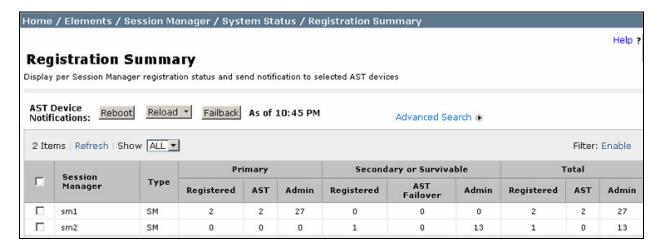
Note: Any changes you make to the profile within iManager will be updated on the iTurret device after **OK** or **Apply** is pressed. However, some changes will require a synchronization. Refer to the *Speakerbus iManager Administrator's Guide* for more details.

8. Verification Steps

All features shown in **Error! Reference source not found.** were tested using the sample configuration. The following steps can be used to verify the solution.

On the iTurret, verify that the status icons are green. These status icons indicate whether iTurret is connected to the network, iCMS server, and SIP registrar (i.e., Session Manager). Refer to [5] for more details.

To verify that the iTurret have successfully registered with Session Manager, from the System Manager web interface click **Session Manager** →**System Status** → **Registration Summary**. This will display a summary of registered stations on each Session Manager as shown below.



9. Conclusion

These Application Notes describe the compliance tested configuration of the Speakerbus iTurret Solution with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. All tests passed with observations noted in **Section 2.2**.

10. Additional References

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

- [1] Administering Avaya Aura® Communication Manager, May 2013, Document Number 03-300509.
- [2] Avaya Extension to Cellular User Guide Avaya Aura® Communication Manager, June 2010
- [3] *Implementing Avaya Aura*[®] *Session Manager*, Release 6.2, March 2013, Document Number 03-603473.
- [4] Speakerbus iManager Administrator's Guide, V1.220, Revision 6, March 2010.
- [5] Session Initiation Protocol Service Examples draft-ietf-sipping-service-examples-15, Internet-Draft, 11th July 2008, available at http://tools.ietf.org/html/draft-ietf-sipping-service-examples-15

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