



Avaya Solution & Interoperability Test Lab

Application Notes for Speakerbus iD808 *i* turret with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services - Issue 1.0

Abstract

These Application Notes describe the steps required to connect Speakerbus iD808 *i* turret to a SIP infrastructure consisting of Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services. Also described is how Avaya Aura™ Communication Manager features can be made available 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 *i* turret, 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 *i* turret to a SIP infrastructure consisting of Avaya AuraTM SIP Enablement Services and Avaya S8730 Servers with a G650 Media Gateway running Avaya AuraTM Communication Manager. Also described is how Avaya AuraTM Communication Manager features can be made available in addition to the standard features supported in the *i* turret. In this configuration, the Off-PBX Stations (OPS) feature set is extended from Avaya AuraTM Communication Manager to the Speakerbus iD808 *i* turret¹, providing the iD808 deskstation with enhanced calling features. The configuration steps described are also applicable to other Linux-based Avaya Servers and Media Gateways running Avaya AuraTM Communication Manager.

The following table provides a summary of the supported features available on the *i* turret with the Avaya SIP offer. Some features are supported locally in the *i* turret, while others are only available with Communication Manager and SIP Enablement Services 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 the *i* turret 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 the *i* turret can also be programmed to an FNE. Other features, such as Exclusion/Privacy and Call Forwarding, are available by using the Feature Name URI (FNU). 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/Class Of Restriction (COS/COR), and voice messaging. Details on operation and administration of OPS can be found in References [2] [3]. The Avaya SIP solution requires all SIP telephones to be configured in Communication Manager as OPS.

¹ In these Application Notes, *i turret* and *iD808 deskstation* will be used interchangeably to refer to the Speakerbus iD808 *i* turret.

FEATURE	Supported		COMMENTS
	Locally at the Phone	With Avaya SIP Offer	
Basic Calling Features			
Extension to Extension Call	Yes	Yes	
Basic Call to legacy phones	No	Yes	
Speed Dial Buttons	Yes	Yes	
Message Waiting Support	Yes	Yes	
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 <i>i</i> turret and FNU
Call Forward Busy/No Answer	Yes	Yes	Local menu option on <i>i</i> turret and FNU
Call Forward Cancel	Yes	Yes	Local menu option on <i>i</i> turret 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
Call Pickup	No	Yes	Via OPS FNE
Automatic Redial	No	Yes	Via OPS FNE
OPS– Selected Additional Station-Side Features			
Automatic Call Back	No	Yes	Via OPS FNE
Automatic Call-Back Cancel	No	Yes	Via OPS FNE
Conference on Answer	No	Yes	Via OPS FNE
Directed Call Pick-Up	No	Yes	Via OPS FNE
Drop Last Added Party	No	Yes	Via OPS FNE
Exclusion/Privacy	Yes	Yes	Local hard key on <i>i</i> turret and FNU
Last Number Dialed	Yes	Yes	Via OPS FNE
Priority Call	Yes	Yes	Via OPS FNE, <i>i</i> turret does not support distinctive ring indication; also local at phone
Send All Calls	No	Yes	Via OPS FNE
Send All Calls Cancel	No	Yes	Via OPS FNE
Transfer to Voice Mail	No	Yes	Via OPS FNE
Whisper Page	No	Yes	Via OPS FNE

Table 1: SIP Features Table

1.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Successful registration of the *i* turret with Avaya SIP Enablement Services.
- Calls between the *i* turret and Avaya SIP, H.323, digital, and analog stations.
- G.711u and G.729 codec support.
- Proper recognition of DTMF transmissions by navigating voicemail menus.
- Proper operation of voicemail with message waiting indicators (MWI).
- PBX features including Multiple Call Appearances, Hold, Transfer, and Conference.
- Extended telephony features using Avaya Communication Manager Feature Name Extensions (FNEs) such as Call Forwarding, Conference On Answer, Call Park, Call Pickup, Automatic Redial and Send All Calls. See **Table 1** for the complete list of features. For more information on FNEs, please refer to [2].
- Exclusion/Privacy using the AST Exclusion FNU.
- Proper system recovery after an *i* turret restart and loss of IP connection.

1.2. Support

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

Web: <http://www.speakerbus.com>

Email: info@speakerbus.com

Telephone: (646) 289-4700 in North America

+44 (0) 870 240 7252 in Europe

+65 6222 4577 in Asia

2. Reference Configuration

The configuration used as an example in these Application Notes is shown in **Figure 1**. The diagram illustrates an enterprise site with an Avaya SIP-based network, including Avaya AuraTM SIP Enablement Services, a pair of Avaya S8730 Servers with a G650 Media Gateway running Avaya AuraTM Communication Manager, and Avaya IP endpoints. Avaya Modular Messaging provides voice mail service. The enterprise site also contains three Speakerbus iD808 *i* turret deskstations that register with Avaya AuraTM SIP Enablement Services and are configured as OPS stations on Avaya AuraTM Communication Manager. Communication Manager extends the telephony functionality that is supported by the SIP-based iD808 devices through the use of Feature Name Extensions (FNEs) and FNUs. The *i* cms server contains the *i* manager application for configuring the iD808 deskstations.

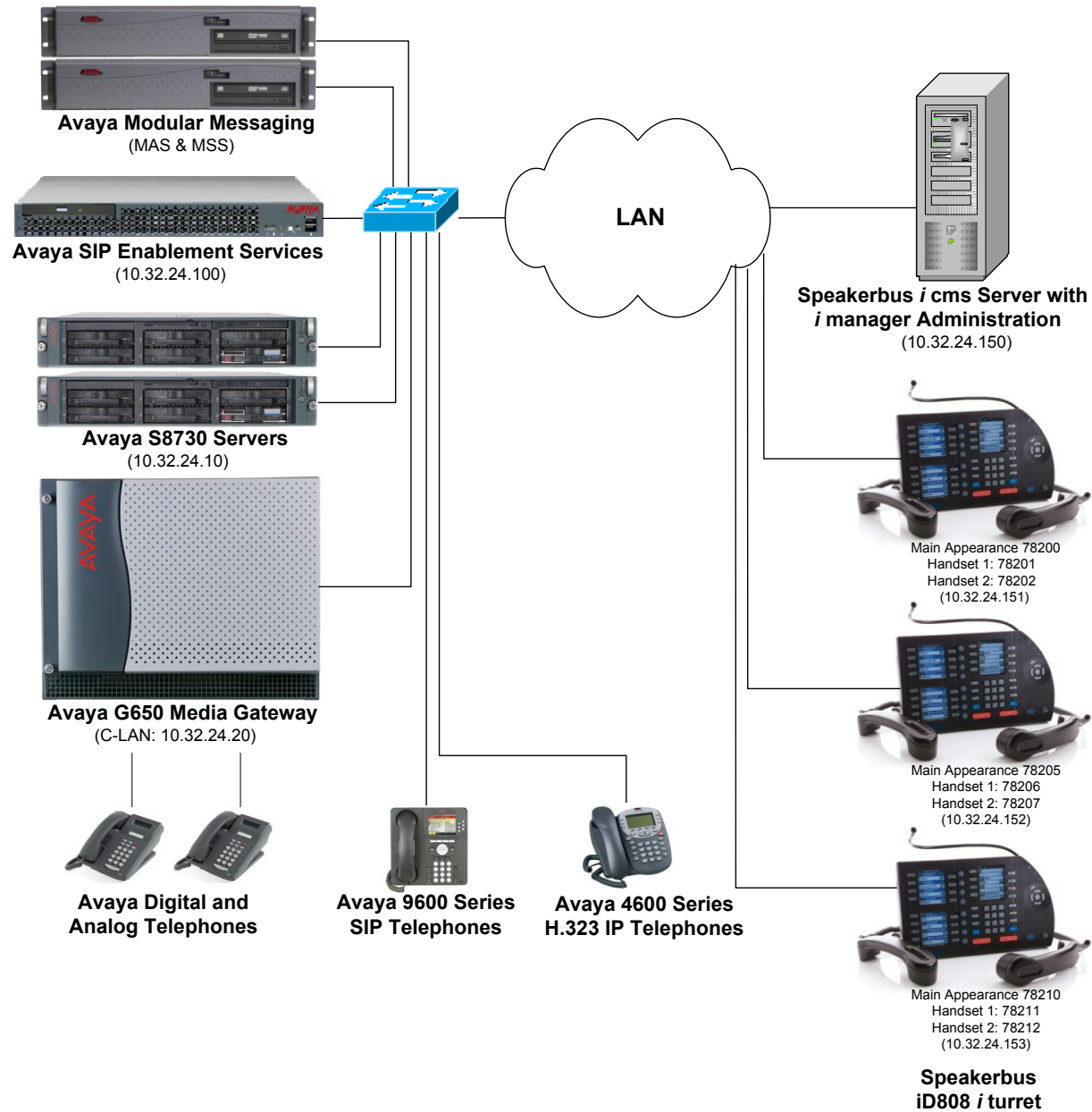


Figure 1: Speakerbus iD808 i turret with Avaya SIP Solution

3. Equipment and Software Validated

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

Hardware Component	Version
Avaya S8730 Servers	Avaya Aura TM Communication Manager 5.2 (R015x.02.0.947.3) with Service Pack 3 (Patch 17579)
Avaya G650 Media Gateway TN799DP C-LAN TN2302AP IP Media Processor	HW01 FW031 HW11 FW118
Avaya Aura TM SIP Enablement Services	5.2 (SES-5.2.0.0-947.3b) with Service Pack 2
Avaya Modular Messaging	5.0
Avaya 4600 Series IP Telephone	2.8 (H.323)
Avaya 9600 Series IP Telephones	2.0.5 (SIP)
Avaya Digital Telephones	--
Avaya Analog Telephones	--
Speakerbus iD808 <i>i</i> turret	1.110.4.0
Speakerbus <i>i</i> cms Server with <i>i</i> manager Administration on Windows 2003 Server	1.210.3.0

4. Configure Aura™ Avaya Communication Manager

This section describes the steps for configuring the Speakerbus iD808 *i* turret as an Off-PBX Station (OPS), administering support for the OPS features indicated in **Table 1**, and configuring a SIP trunk between Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services. For complete SIP documentation, see References [2] [3]. Use the System Access Terminal (SAT) to configure Avaya Aura™ Communication Manager. Log in with the appropriate credentials.

4.1. Verify OPS and SIP Trunk Capacity

Using the SAT, verify that the Off-PBX Telephones (OPS) and SIP Trunks features are enabled on the **system-parameters customer-options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative. On Page 1, verify that the number of OPS stations allowed in the system is sufficient. One OPS station is required per iD808 device.

display system-parameters customer-options		Page	1 of 11
OPTIONAL FEATURES			
G3 Version: V15	Software Package: Standard		
Location: 1	RFA System ID (SID): 1		
Platform: 6	RFA Module ID (MID): 1		
		USED	
Platform Maximum Ports: 48000		222	
Maximum Stations: 36000		149	
Maximum XMOBILE Stations: 0		0	
Maximum Off-PBX Telephones - EC500: 5		0	
Maximum Off-PBX Telephones - OPS: 100		10	
Maximum Off-PBX Telephones - PBFMC: 0		0	
Maximum Off-PBX Telephones - PVFMC: 0		0	
Maximum Off-PBX Telephones - SCCAN: 0		0	
(NOTE: You must logoff & login to effect the permission changes.)			

On Page 2 of the **system-parameters customer-options** form, verify that the number of SIP trunks supported by the system is sufficient.

display system-parameters customer-options		Page 2 of 11
OPTIONAL FEATURES		
IP PORT CAPACITIES		USED
Maximum Administered H.323 Trunks:	200	40
Maximum Concurrently Registered IP Stations:	18000	5
Maximum Administered Remote Office Trunks:	0	0
Maximum Concurrently Registered Remote Office Stations:	0	0
Maximum Concurrently Registered IP eCons:	0	0
Max Concur Registered Unauthenticated H.323 Stations:	0	0
Maximum Video Capable H.323 Stations:	0	0
Maximum Video Capable IP Softphones:	0	0
Maximum Administered SIP Trunks:	500	10
Maximum Administered Ad-hoc Video Conferencing Ports:	0	0
Maximum Number of DS1 Boards with Echo Cancellation:	0	0
Maximum TN2501 VAL Boards:	128	1
Maximum Media Gateway VAL Sources:	0	0
Maximum TN2602 Boards with 80 VoIP Channels:	128	0
Maximum TN2602 Boards with 320 VoIP Channels:	128	0
Maximum Number of Expanded Meet-me Conference Ports:	0	0
(NOTE: You must logoff & login to effect the permission changes.)		

4.2. Define System Features

Use the **change system-parameters features** command to administer system wide features for SIP endpoints. Those related to features listed in **Table 1** are shown in bold. These are all standard Communication Manager features that are also available to OPS stations.

change system-parameters features	Page 17 of 18
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 wink
Other Stations When Call Is Active:	green
Ringing:	green flash
Idle:	steady
Pickup On Transfer?	y

IP PARAMETERS

Direct IP-IP Audio Connections? y

IP Audio Hairpinning? y

SDP Capability Negotiation for SRTP? n

CALL PICKUP

Maximum Number of Digits for Directed Group Call Pickup: 4

Call Pickup on Intercom Calls? y Call Pickup Alerting? y

Temporary Bridged Appearance on Call Pickup? y **Directed Call Pickup? y**

Extended Group Call Pickup: none

Enhanced Call Pickup Alerting? y

Enhanced Call Pickup Delay Timer (sec.) Display: 5 Audible Notification: 5

Display Information With Bridged Call? n

Keep Bridged Information on Multiline Displays During Calls? n

PIN Checking for Private Calls? n

4.3. Define the Dial Plan

Use the **change dialplan analysis** command to define the dial plan used in the system. This includes all telephone extensions, OPS Feature Name Extensions (FNEs), and Feature Access Codes (FACs). To define the FNEs for the OPS features listed in **Table 1**, a Feature Access Code (FAC) must also be specified for the corresponding feature. In the sample configuration, telephone extensions are five digits long and begin with ‘7’, FNEs are also five digits beginning with 7, and the FACs have formats as indicated with **Call Type fac**.

change dialplan analysis

Page 1 of 12

DIAL PLAN ANALYSIS TABLE

Location: all Percent Full: 1

	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type
0		1	attd						
1		4	dac						
2		5	ext						
3		5	ext						
4		5	aar						
5		5	ext						
7		5	ext						
8		1	fac						
9		1	fac						
*		3	fac						
#		3	fac						

4.4. Define Feature Access Codes (FACs)

Use **change feature-access-codes** to define the access codes for the OPS FNEs, shown in bold.

change feature-access-codes	Page 1 of 9
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:	*11
Answer Back Access Code:	*24
Auto Alternate Routing (AAR) Access Code:	8
Auto Route Selection (ARS) - Access Code 1:	9
Automatic Callback Activation:	*25
Call Forwarding Activation Busy/DA:	*21 All: *20
Call Forwarding Enhanced Status:	Act: Deactivation:
Call Park Access Code:	*26
Call Pickup Access Code:	*27
CAS Remote Hold/Answer Hold-Unhold Access Code:	
CDR Account Code Access Code:	
Change COR Access Code:	
Change Coverage Access Code:	
Conditional Call Extend Activation:	Deactivation:
Contact Closure Open Code:	Close Code:

change feature-access-codes	Page 2 of 9
FEATURE ACCESS CODE (FAC)	
Contact Closure Pulse Code:	
Data Origination Access Code:	
Data Privacy Access Code:	
Directed Call Pickup Access Code:	*28
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:	*29
Leave Word Calling Message Retrieval Lock:	
Leave Word Calling Message Retrieval Unlock:	

change feature-access-codes	Page 3 of 9
FEATURE ACCESS CODE (FAC)	
Leave Word Calling Send A Message:	
Leave Word Calling Cancel A Message:	
Limit Number of Concurrent Calls Activation:	Deactivation:
Malicious Call Trace Activation:	Deactivation:
Meet-me Conference Access Code Change:	
PASTE (Display PBX data on Phone) Access Code:	
Personal Station Access (PSA) Associate Code:	Dissociate Code:
Per Call CPN Blocking Code Access Code: *34	
Per Call CPN Unblocking Code Access Code: *35	
Priority Calling Access Code: *30	
Program Access Code:	
Refresh Terminal Parameters Access Code:	
Remote Send All Calls Activation:	Deactivation:
Self Station Display Activation:	
Send All Calls Activation: *31	Deactivation: #31
Station Firmware Download Access Code:	

change feature-access-codes	Page 4 of 9
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: *32	
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: *33	
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:	

4.5. Define Feature Name Extensions (FNEs)

The FNEs can be defined using the **change off-pbx-telephone feature-name-extensions** command.

```
change off-pbx-telephone feature-name-extensions set 1          Page 1 of 2
EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME
Set Name:

Active Appearance Select: 78100
Automatic Call Back: 78101
Automatic Call-Back Cancel: 78102
Call Forward All: 78103
Call Forward Busy/No Answer: 78104
Call Forward Cancel: 78105
Call Park: 78106
Call Park Answer Back: 78107
Call Pick-Up: 78108
Calling Number Block: 78109
Calling Number Unblock: 78110
Conditional Call Extend Enable: 78111
Conditional Call Extend Disable: 78112
Conference Complete: 78113
Conference on Answer: 78114
Directed Call Pick-Up: 78115
Drop Last Added Party: 78116
```

```
change off-pbx-telephone feature-name-extensions set 1          Page 2 of 2
EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME

Exclusion (Toggle On/Off): 78117
Extended Group Call Pickup:
Held Appearance Select: 78118
Idle Appearance Select: 78119
Last Number Dialed: 78120
Malicious Call Trace:
Malicious Call Trace Cancel:
Off-Pbx Call Enable:
Off-Pbx Call Disable:
Priority Call: 78125
Recall: 78126
Send All Calls: 78127
Send All Calls Cancel: 78128
Transfer Complete: 78129
Transfer On Hang-Up: 78130
Transfer to Voice Mail: 78131
Whisper Page Activation: 78132
```

4.6. Specify Class of Service (COS)

Use the **change cos** command to set the appropriate service permissions to support OPS features (shown in bold). For the example, a COS of 1 was used. Priority call indication (e.g., distinctive ring) is not supported on the *i* turret when using the Priority FNE. However, the iD808 does support a distinctive-ring/alerting mechanism locally on the phone, not covered in testing.

change cos	Page 1 of 2															
CLASS OF SERVICE	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
Auto Callback	y	y	y	n	y	n	y	n	y	n	y	n	y	n	y	n
Call Fwd-All Calls	n	y	n	y	y	n	n	y	y	n	n	y	y	n	n	y
Data Privacy	n	n	n	n	n	y	y	y	y	n	n	n	n	y	y	y
Priority Calling	n	y	n	n	n	n	n	n	n	y	y	y	y	y	y	y
Console Permissions	y	n	y	y	y	y	y	y	y	y	n	n	n	n	y	y
Off-hook Alert	y	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Client Room	y	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Restrict Call Fwd-Off Net	y	n	y	y	y	y	y	y	y	y	y	y	y	y	y	y
Call Forwarding Busy/DA	y	y	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	y	y	y	n	n	n

4.7. Specify Class of Restriction (COR)

Use the **change cor** command 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” for the affected stations. In the sample configuration, the *i* turrets were assigned to COR 1. Note that Page 3 can be used to implement a form of centralized call screening for groups of stations and trunks.

change cor 1	Page 1 of 23	
CLASS OF RESTRICTION		
COR Number: 1		
COR Description:		
FRL: 0	APLT? y	
Can Be Service Observed? y	Calling Party Restriction: none	
Can Be A Service Observer? y	Called Party Restriction: none	
Partitioned Group Number: 1	Forced Entry of Account Codes? n	
Priority Queuing? n	Direct Agent Calling? y	
Restriction Override: all	Facility Access Trunk Test? n	
Restricted Call List? n	Can Change Coverage? n	
Access to MCT? y	Fully Restricted Service? n	
Group II Category For MFC: 7	Hear VDN of Origin Annc.? y	
Send ANI for MFE? 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		

4.8. Add Coverage Path

Configure the coverage path to one used for the voice messaging hunt group, which is group *h30* in the sample configuration. The default values shown for **Busy**, **Don't Answer**, and **DND/SAC/Goto Cover** can be used for the **Coverage Criteria**.

add coverage path 20		Page 1 of 1	
COVERAGE PATH			
Coverage Path Number: 20			
Cvg Enabled for VDN Route-To Party? n		Hunt after Coverage? n	
Next Path Number:		Linkage	
COVERAGE CRITERIA			
Station/Group Status	Inside Call	Outside Call	
Active?	n	n	
Busy?	y	y	
Don't Answer?	y	y	Number of Rings: 2
All?	n	n	
DND/SAC/Goto Cover?	y	y	
Holiday Coverage?	n	n	
COVERAGE POINTS			
Terminate to Coverage Pts. with Bridged Appearances? n			
Point1: h30	Rng:	Point2:	
Point3:		Point4:	
Point5:		Point6:	

4.9. Add Stations

Use the **add station** command to add a station for each *i* turret to be supported. The Speakerbus iD808 *i* turret requires up to three stations for each device. The first station is referred to as the default appearance. The second and third stations are needed when privacy is required on the handsets/headsets. If the privacy feature is not needed, then only the first station is required. To configure the default appearance, use *9630* for the **Station Type** and include the **Coverage Path** for voice messaging, if applicable. Use the **COS** and **COR** values administered in the previous sections. The **Name** field is optional and is shown on the display of Avaya non-SIP telephones when receiving calls from this station. Use the default values for the other fields on Page 1.

add station 78200		Page 1 of 5
STATION		
Extension: 78200	Lock Messages? n	BCC: 0
Type: 9630	Security Code:	TN: 1
Port: IP	Coverage Path 1: 20	COR: 1
Name: iturret 78200	Coverage Path 2:	COS: 1
	Hunt-to Station:	
STATION OPTIONS		
Loss Group: 19	Time of Day Lock Table:	
	Personalized Ringing Pattern: 1	
	Message Lamp Ext: 78200	
Speakerphone: 2-way	Mute Button Enabled? y	
Display Language: english	Button Modules: 0	
Survivable GK Node Name:		
Survivable COR: internal	Media Complex Ext:	
Survivable Trunk Dest? y	IP SoftPhone? n	
	Customizable Labels? y	

On Page 2, note the following:

- If this *i* turret will have a bridged appearance for another telephone (see Page 4 for this station), then **Bridged Call Alerting** should be set to *y*, so that this *i* turret will ring when the other telephone is called.
- By default, the last call appearance is reserved for outgoing calls from the *i* turret. Set the **Restrict Last Appearance** field to *y*.
- Set the **MWI Served User Type** field to the appropriate value to allow message waiting indication to be sent to the *i* turret.

add station 78200		Page 2 of 5	
		STATION	
FEATURE OPTIONS			
LWC Reception:	spe	Auto Select Any Idle Appearance?	n
LWC Activation?	y	Coverage Msg Retrieval?	y
LWC Log External Calls?	n	Auto Answer:	none
CDR Privacy?	n	Data Restriction?	n
Redirect Notification?	y	Idle Appearance Preference?	n
Per Button Ring Control?	n	Bridged Idle Line Preference?	n
Bridged Call Alerting?	y	Restrict Last Appearance?	y
Active Station Ringing:	single		
		EMU Login Allowed?	n
H.320 Conversion?	n	Per Station CPN - Send Calling Number?	
Service Link Mode:	as-needed	EC500 State:	enabled
Multimedia Mode:	enhanced	Audible Message Waiting?	n
MWI Served User Type:	qsig-mwi	Display Client Redirection?	n
		Select Last Used Appearance?	n
		Coverage After Forwarding?	s
		Multimedia Early Answer?	n
		Direct IP-IP Audio Connections?	y
Emergency Location Ext:	78200	Always Use? n IP Audio Hairpinning?	n

On Page 4 under the heading **BUTTON ASSIGNMENTS**², fill in the number of call appearances that are to be supported for the *i* turret. In this example, the first station for the *i* turret was configured with four call appearances. Locally, the *i* turret will actually be configured with 3 call appearances since the last appearance is restricted as configured on Page 2.

```

add station 78200                                     Page 4 of 5

                                STATION

SITE DATA
  Room:                                     Headset? n
  Jack:                                    Speaker? n
  Cable:                                  Mounting: d
  Floor:                                Cord Length: 0
  Building:                             Set Color:

ABBREVIATED DIALING
  List1:                                List2:                                List3:

BUTTON ASSIGNMENTS
  1: call-appr                               5: brdg-appr  B:1  E:78201
  2: call-appr                               6: brdg-appr  B:1  E:78202
  3: call-appr                               7: auto-cback
  4: call-appr                               8: no-hld-cnf

voice-mail Number:

```

Under the **BUTTON ASSIGNMENTS** section, configure Call Forwarding feature buttons.

```

add station 78200                                     Page 5 of 5

                                STATION

BUTTON ASSIGNMENTS

  9: call-fwd  Ext:
 10: cfwd-bsyda Ext:
 11:
 12:

```

Under the same heading, enter the function button names, if required, for OPS FNEs that will be used at the *i* turret. Only the FNEs shown in the table below require the station to have a corresponding function button.

FNE Name	Function Button
Automatic Callback, Automatic Callback Cancel	auto-cback
Call Forward All	call-fwd
Call Forward Busy/No Answer	cfwd-bsyda
Conference on Answer	no-hld-cnf

² The bridged appearances for the handsets (i.e., button assignments 5-6) and the automatic call back and conference on answer feature buttons (i.e., button assignments 7-8) may not be required as described in [5].

In the sample configuration, four call appearances were administered for extension 78200. Two bridged appearances were for second and third stations corresponding to two handsets. Note that these stations are configured below and these bridged appearance buttons cannot be configured until those stations have been added. If privacy is not needed for this *i* turret, then these bridged appearances are not required.

Page 1 of the second station for handset 1 is configured as follows. A coverage path is not required for this station.

add station 78201		Page 1 of 5
STATION		
Extension: 78201	Lock Messages? n	BCC: 0
Type: 9630	Security Code:	TN: 1
Port: IP	Coverage Path 1:	COR: 1
Name: HS1 of 78200	Coverage Path 2:	COS: 1
	Hunt-to Station:	
STATION OPTIONS		
Loss Group: 19	Time of Day Lock Table:	
	Personalized Ringing Pattern: 1	
	Message Lamp Ext: 78201	
Speakerphone: 2-way	Mute Button Enabled? y	
Display Language: english	Button Modules: 0	
Survivable GK Node Name:		
Survivable COR: internal	Media Complex Ext:	
Survivable Trunk Dest? y	IP SoftPhone? n	
	Customizable Labels? y	

On Page 2, the **Bridged Call Alerting** and **Restrict Last Appearance** fields should be set to y.

add station 78201		Page 2 of 5
STATION		
FEATURE OPTIONS		
LWC Reception: spe	Auto Select Any Idle Appearance? n	
LWC Activation? y	Coverage Msg Retrieval? y	
LWC Log External Calls? n	Auto Answer: none	
CDR Privacy? n	Data Restriction? n	
Redirect Notification? y	Idle Appearance Preference? n	
Per Button Ring Control? n	Bridged Idle Line Preference? n	
Bridged Call Alerting? y	Restrict Last Appearance? y	
Active Station Ringing: single		
	EMU Login Allowed? n	
H.320 Conversion? n	Per Station CPN - Send Calling Number?	
Service Link Mode: as-needed	EC500 State: enabled	
Multimedia Mode: enhanced	Audible Message Waiting? n	
MWI Served User Type:	Display Client Redirection? n	
AUDIX Name:	Select Last Used Appearance? n	
	Coverage After Forwarding? s	
	Multimedia Early Answer? n	
	Direct IP-IP Audio Connections? y	
Emergency Location Ext: 78201	Always Use? n IP Audio Hairpinning? n	

On Page 4 of the second station for handset 1, one call appearance should be configured, one feature button for the Exclusion feature (required for privacy), and bridged appearances³ for each call appearance of the first station (default appearance) configured below.

add station 78201		Page 4 of 5	
STATION			
SITE DATA			
Room:		Headset?	n
Jack:		Speaker?	n
Cable:		Mounting:	d
Floor:		Cord Length:	0
Building:		Set Color:	
ABBREVIATED DIALING			
List1:	List2:	List3:	
BUTTON ASSIGNMENTS			
1: call-appr	5: brdg-appr	B:3	E:78200
2: exclusion	6: brdg-appr	B:1	E:78202
3: brdg-appr B:1 E:78200	7:		
4: brdg-appr B:2 E:78200	8:		
voice-mail Number:			

³ The bridged appearance for the second handset (i.e., button assignment 6) on Page 4 may not be required as described in [5].

Below is the configuration⁴ of the third station for handset 2.

add station 78202		Page 1 of 5
STATION		
Extension: 78202	Lock Messages? n	BCC: 0
Type: 9630	Security Code:	TN: 1
Port: IP	Coverage Path 1:	COR: 1
Name: HS2 of 78200	Coverage Path 2:	COS: 1
	Hunt-to Station:	
STATION OPTIONS		
Loss Group: 19	Time of Day Lock Table:	
	Personalized Ringing Pattern: 1	
	Message Lamp Ext: 78202	
Speakerphone: 2-way	Mute Button Enabled? y	
Display Language: english	Button Modules: 0	
Survivable GK Node Name:		
Survivable COR: internal	Media Complex Ext:	
Survivable Trunk Dest? y	IP SoftPhone? n	
	Customizable Labels? y	

add station 78202		Page 2 of 5
STATION		
FEATURE OPTIONS		
LWC Reception: spe	Auto Select Any Idle Appearance? n	
LWC Activation? y	Coverage Msg Retrieval? y	
LWC Log External Calls? n	Auto Answer: none	
CDR Privacy? n	Data Restriction? n	
Redirect Notification? y	Idle Appearance Preference? n	
Per Button Ring Control? n	Bridged Idle Line Preference? n	
Bridged Call Alerting? y	Restrict Last Appearance? y	
Active Station Ringing: single		
	EMU Login Allowed? n	
H.320 Conversion? n	Per Station CPN - Send Calling Number?	
Service Link Mode: as-needed	EC500 State: enabled	
Multimedia Mode: enhanced	Audible Message Waiting? n	
MWI Served User Type:	Display Client Redirection? n	
AUDIX Name:	Select Last Used Appearance? n	
	Coverage After Forwarding? s	
	Multimedia Early Answer? n	
	Direct IP-IP Audio Connections? y	
Emergency Location Ext: 78202	Always Use? n IP Audio Hairpinning? n	

⁴ The bridged appearance for the first handset (i.e., button assignment 6) on Page 4 may not be required as described in [5].

add station 78202		Page 4 of 5
STATION		
SITE DATA		
Room:		Headset? n
Jack:		Speaker? n
Cable:		Mounting: d
Floor:		Cord Length: 0
Building:		Set Color:
ABBREVIATED DIALING		
List1:	List2:	List3:
BUTTON ASSIGNMENTS		
1: call-appr	5: brdg-appr	B:3 E:78200
2: exclusion	6: brdg-appr	B:1 E:78201
3: brdg-appr B:1 E:78200	7:	
4: brdg-appr B:2 E:78200	8:	
voice-mail Number:		

Note: If a bridged appearance is required for another *i* turret or telephone, a bridged appearance button must be added to all three stations corresponding to the *i* turret device.

Use the **change off-pbx-telephone station-mapping** command to map the Communication Manager extensions (78200, 78201, and 78202) to the same SIP Enablement Services Communication Manager extension. Enter the field values shown. For the sample configuration, the **Trunk Selection** value indicates the SIP trunk group between Communication Manager and SIP Enablement Services. The SIP trunk group is configured in Section 4.10. The **Configuration Set** value can reference a set that has the default settings.

change off-pbx-telephone station-mapping 78200							Page 1 of 3
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION							
Station Extension	Application	Dial Prefix	CC	Phone Number	Trunk Selection	Config Set	Dual Mode
78200	OPS	-		78200	10	1	
78201	OPS	-		78201	10	1	
78202	OPS	-		78202	10	1	

On Page 2, change the **Call Limit** to match the number of *call-appr* entries in the station form. Also, verify that **Mapping Mode** is set to *both* (the default value for a newly added station). It is recommended that 10 be used for the primary stations call limit as this is the Avaya maximum and would not have to be subsequently changed if bridged appearances are added to the user.

change off-pbx-telephone station-mapping 78200							Page 2 of 3
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION							
Station Extension	Appl Name	Call Limit	Mapping Mode	Calls Allowed	Bridged Calls	Location	
78200	OPS	4	both	all	none		
78201	OPS	2	both	all	none		
78202	OPS	2	both	all	none		

4.10. Configure SIP Trunk

In the **IP Node Names** form, assign an IP address and host name for the C-LAN board in the Avaya G650 Media Gateway and for the SIP Enablement Services server at the enterprise site. The host names will be used throughout the other configuration screens of Communication Manager.

change node-names ip		Page 1 of 2
IP NODE NAMES		
Name	IP Address	
Gateway001	10.32.24.1	
ModMsg	192.50.10.45	
clancrm	10.32.24.20	
default	0.0.0.0	
medprocrm	10.32.24.21	
procr	0.0.0.0	
ses	10.32.24.100	

(7 of 7 administered node-names were displayed)
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on SIP Enablement Services. In this configuration, the domain name is *avaya.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G650 Media Gateway. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to SIP Enablement Services. This codec set is used when its corresponding network region (i.e., IP Network Region '1') is specified in the SIP signaling group.

change ip-network-region 1		Page 1 of 19
IP NETWORK REGION		
Region: 1		
Location:	Authoritative Domain: avaya.com	
Name:		
MEDIA PARAMETERS		Intra-region IP-IP Direct Audio: yes
Codec Set: 1		Inter-region IP-IP Direct Audio: yes
UDP Port Min: 2048	IP Audio Hairpinning? y	
UDP Port Max: 3029		
DIFFSERV/TOS PARAMETERS		RTCP Reporting Enabled? y
Call Control PHB Value: 34	RTCP MONITOR SERVER PARAMETERS	
Audio PHB Value: 46	Use Default Server Parameters? y	
Video PHB Value: 26		
802.1P/Q PARAMETERS		
Call Control 802.1p Priority: 7		
Audio 802.1p Priority: 6		
Video 802.1p Priority: 5	AUDIO RESOURCE RESERVATION PARAMETERS	
H.323 IP ENDPOINTS		RSVP Enabled? n
H.323 Link Bounce Recovery? y		
Idle Traffic Interval (sec): 20		
Keep-Alive Interval (sec): 5		
Keep-Alive Count: 5		

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to the *i* turret deskstations. The form is accessed via the **change ip-codec-set 1** command. Note that IP codec set '1' was specified in IP Network Region '1' shown above. The default settings of the **IP Codec Set** form are shown below. However, the **IP Codec Set** form may specify multiple codecs, including G.711 and G.729, which are supported by the iD808 deskstations.

Note: G.729 calls between an *i* turret and an Avaya SIP telephone are not shuffled. However, G.729 calls between two *i* turret deskstations and an *i* turret and Avaya H.323 telephone are shuffled.

change ip-codec-set 1

Page 1 of 2

IP Codec Set

Codec Set: 1

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

Prior to configuring a SIP trunk group for communication with SIP Enablement Services, a SIP signaling group must be configured. Configure the Signaling Group form shown as follows:

- Set the **Group Type** field to *sip*.
- The **Transport Method** field will default to *tls* (Transport Layer Security).
- Specify the C-LAN board in the G650 Media Gateway and the SIP Enablement Services Server as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form shown above.
- Ensure that the recommended TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of SIP Enablement Services in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*. This domain is specified in the Uniform Resource Identifier (URI) of the “SIP To Address” in the INVITE message. Mis-configuring this field may prevent calls from being successfully established to other SIP endpoints or to the PSTN.
- If calls to/from SIP endpoints are to be shuffled, then the **Direct IP-IP Audio Connections** field must be set to *y*
- The **DTMF over IP** field should be set to the default value of *rtp-payload*. Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

add signaling-group 10		Page 1 of 1
SIGNALING GROUP		
Group Number: 10	Group Type: sip	
	Transport Method: tls	
IMS Enabled? n		
Near-end Node Name: clancrm	Far-end Node Name: ses	
Near-end Listen Port: 5061	Far-end Listen Port: 5061	
	Far-end Network Region: 1	
Far-end Domain: avaya.com		
Bypass If IP Threshold Exceeded? n		
DTMF over IP: rtp-payload	Direct IP-IP Audio Connections? y	
Session Establishment Timer(min): 3	IP Audio Hairpinning? n	
Enable Layer 3 Test? n	Direct IP-IP Early Media? n	
H.323 Station Outgoing Direct Media? n	Alternate Route Timer(sec): 6	

Configure the **Trunk Group** form as shown below. This trunk group is used for calls to the *i* turret deskstations. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Configure the other fields in bold and accept the default values for the remaining fields.

add trunk-group 10		Page 1 of 21	
TRUNK GROUP			
Group Number: 10	Group Type: sip	CDR Reports: y	
Group Name: To SES	COR: 1	TN: 1	TAC: 1010
Direction: two-way	Outgoing Display? n		
Dial Access? n	Night Service:		
Queue Length: 0			
Service Type: tie	Auth Code? n		
		Signaling Group: 10	
		Number of Members: 10	

On Page 3 of the trunk group form, set the **Numbering Format** field to *public*. This field specifies the format of the calling party number sent to the far-end.

add trunk-group 10		Page 3 of 21	
TRUNK FEATURES			
ACA Assignment? n	Measured: none	Maintenance Tests? y	
Numbering Format: public		UII Treatment: service-provider	
		Replace Restricted Numbers? n	
		Replace Unavailable Numbers? n	
Show ANSWERED BY on Display? y			

Configure the **Public/Unknown Numbering Format** form to send the calling party number to the far-end. Add an entry so that local stations with a 5-digit extension beginning with '7' and whose calls are routed over SIP trunk group '10' have the number sent to the far-end for display purposes.

change public-unknown-numbering 0					Page 1 of 2
NUMBERING - PUBLIC/UNKNOWN FORMAT					
Ext	Ext	Trk	CPN	Total	
Len	Code	Grp(s)	Prefix	CPN	
				Len	
5	7	10		5	Total Administered: 1
					Maximum Entries: 9999

5. Configure Avaya Aura™ SIP Enablement Services

This section covers the administration of Avaya Aura™ SIP Enablement Services. SIP Enablement Services is configured via an Internet browser using the Administration web interface. To access the Administration web interface, enter *http://<ip-addr>/admin* as the URL in an Internet browser, where *<ip-addr>* is the IP address of SIP Enablement Services. Log in with the appropriate credentials and then select the Administration→SIP Enablement Services from the next screen. The main screen is displayed as shown below.




Integrated Management
SIP Server Management

Help ExitThis Server: [1] ses1

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Manage Event Aggregators	Add/Delete Event Aggregators.
Certificate Management	Manage Certificates.
Manage Conferencing	Add and delete Conference Extensions.
Manage Emergency Contacts	Add and delete Emergency Contacts.
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Manage Hosts	Add and delete Hosts.
IM logs	Download IM Logs.
Manage Communication Manager Servers	Add and delete Communication Manager Servers.
Manage Communication Manager Extensions	Add and delete Communication Manager Extensions.
Server Configuration	View Properties of the system.
Manage SIP Phone Settings	Add/Delete Phone Settings
Manage Survivable Call Processors	Add and delete Survivable Call Processors.
System Status	View System Status.
Trace Logger	Manage SIP Trace Logs.
Manage Trusted Hosts	Add and delete Trusted Hosts.

From the left pane of the Administration web interface, expand the **Server Configuration** option and select **System Properties**. In the **System Properties** screen, enter the domain name assigned to the Avaya SIP-based network and the SIP License Host. For the **SIP License Host** field, enter the fully qualified domain name or the IP address of the SES server that is running the WebLM application and has the associated license file installed. This entry should always correspond to the localhost unless the WebLM server is not co-resident with this server. After configuring the **System Properties** screen, click the **Update** button.

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View System Properties

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

Note that the DNS domain is avaya.com

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

SIP License Host*	<input type="text" value="10.32.24.100"/>
-------------------	---

DiffServ/TOS Parameters

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

802.1 Parameters

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

Update

After setting up the domain in the **System Properties** screen, create a host entry for SIP Enablement Services. The following example shows the **Edit Host** screen since the host had already been configured. Enter the IP address of SIP Enablement Services in the **Host IP Address** field. The **Profile Service Password** was specified during the system installation. Next, verify the **Host Type** field. In this example, the host server was configured as an *SES combined home/edge* during the initial setup. The default values for the other fields may be used as shown below.

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Edit Host

Host IP Address* 10.32.24.100

Profile Service Password*

Host Type SES combined home-edge

Parent none

Listen Protocols ☒ UDP ☒ TCP ☒ TLS

Link Protocols ☐ UDP ☐ TCP ☒ TLS

Access Control Policy (Default) ☐ Allow All ☒ Deny All

Emergency Contacts Policy ☒ Allow ☐ Deny

Minimum Registration (seconds) 900 Registration Expiration Timer (seconds)* 86400

Subscription Expiration Timer (seconds)* 86400

Line Reservation Timer (seconds) * 30

Outbound Routing Allowed ☒ Internal ☒ External

OutboundProxy Port ☐ UDP ☐ TCP ☐ TLS

Outbound Direct Domains

Default Ringer Volume* 5 Default Ringer Cadence 2

Default Receiver Volume* 5 Default Speaker Volume* 5

VMM Server Address

VMM Server Port 5005 VMM Report Period 5

Fields marked * are required.

Update

Under the **Communication Manager Servers** option in the Administration web interface, select **Add** to add the Avaya Media Server in the enterprise site since a SIP trunk is required between Communication Manager and SIP Enablement Services. The screen below shows the Edit Communication Manager Server Interface screen since the server has already been added. In this screen, enter the following information:

- A descriptive name in the **Communication Manager Server Interface Name** field (e.g., devcon13).
- Select the home server in the **Host** field.
- Select *TLS* (Transport Link Security) for the **SIP Trunk Link Type**.
- Enter the IP address of the C-LAN board in the Avaya G650 Media gateway in the **SIP Trunk IP Address** field.

Refer to [3] for additional information on configuring the remaining fields.

The screenshot displays the Avaya Integrated Management SIP Server Management web interface. The top header includes the Avaya logo, the title 'Integrated Management SIP Server Management', and a session indicator 'This Server: [1] ses1'. A navigation menu on the left lists various system components, with 'Communication Manager Servers' selected. The main content area is titled 'Edit Communication Manager Server Interface' and contains several configuration sections:

- Communication Manager Server Interface Name***: devcon13
- Host**: 10.32.24.100
- SIP Trunk** section:
 - SIP Trunk Link Type**: Radio buttons for TCP and TLS, with TLS selected.
 - SIP Trunk IP Address***: 10.32.24.20
- Communication Manager Server** section:
 - Communication Manager Server Admin Address***: 10.32.24.10 (with a note to see Help)
 - Communication Manager Server Admin Port***: 5022
 - Communication Manager Server Admin Login***: interop
 - Communication Manager Server Admin Password***: masked with dots
 - Communication Manager Server Admin Password Confirm***: masked with dots
- SMS Connection Type**: Radio buttons for SSH, Telnet, and Not Available, with SSH selected.

A note at the bottom right states: 'Note: If the Communication Manager Server connection type is changed and the admin port value is not also changed, changing connection type to SSH will change the admin port to 5022 when Add or Update is clicked and changing connection type to Telnet will change admin port to 5023 when Add or Update is clicked.' Below this note, a text label indicates 'Fields marked * are required.' and an 'Update' button is present.

Add three users for each Speakerbus iD808 *i* turret registering with SIP Enablement Services. Three users are required, one for the main appearance and two for the handset appearances. The handset appearances are required to support privacy with Communication Manager. If fewer than two handsets are used, or Privacy is not enabled on the iD808, then it is not necessary to enable three users via SIP Enablement Services. In the **Add User** screen, enter the extension of the SIP endpoint in the **Primary Handle** field. Enter a user password in the **Password** and **Confirm Password** fields. In the **Host** field, select the SIP Enablement Services server hosting the domain (*avaya.com*) for this user. Enter the **First Name** and **Last Name** of the user. To associate the extension for this user with a Communication Manager extension, select the **Add Communication Manager Extension** checkbox. Calls from this user will always be routed through Avaya Communication Manager over the SIP trunk. The **Add Communication Manager Extension** screen is displayed next after adding this user profile by clicking on the **Add** button.

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Add User

Primary Handle* 78200

User ID

Password*

Confirm Password*

Host* 10.32.24.100

First Name* iTurret

Last Name* 78200

Address 1 211 Mount Airy Rd

Address 2

Office

City Basking Ridge

State NJ

Country USA

Zip 07920

Survivable Call Processor none

Add Communication Manager Extension ☒

Fields marked * are required.

Add

In the **Add Communication Manager Extension** screen, enter the **Extension** configured in Avaya Communication Manager for the previously added user. Usually, the Communication Manager extension and the user extension are the same (recommended). Click the **Add** button.

The screenshot displays the Avaya Integrated Management SIP Server Management web interface. The top header features the Avaya logo and the title 'Integrated Management SIP Server Management'. Below the header, a navigation bar includes 'Help' and 'Exit' links, and a status indicator 'This Server: [1] ses1'. A left-hand navigation menu lists various system management options, with 'Communication Manager Extensions' highlighted. The main content area is titled 'Add Communication Manager Extension' and contains the following text: 'Add Communication Manager extension for user 78200.' Below this, there are two input fields: 'Extension' with the value '78200' and 'Communication Manager Server' with a dropdown menu showing 'devcon13'. A note states 'Fields marked * are required.' At the bottom of the form is a blue 'Add' button.

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Add Communication Manager Extension

Add Communication Manager extension for user 78200.

Extension

Communication Manager Server

Fields marked * are required.

Add

6. Speakerbus iD808 *i* turret Configuration

This section provides the procedure for configuring the Speakerbus iD808 *i* turret using *i* manager Administration. *i* manager allows users to manage the iD808 *i* turret devices from a single workstation through a point-and-click interface using a web browser. The procedures for configuring the *i* turret fall into the following areas:

- Launch *i* manager
- Verify Product Key
- Create Site
 - Create Subnet
 - Create Deskstations
- Create SIP Server
 - Create PBX
 - Create Dial Plan
 - Create Appearances
- Create Users
- Create Groups
- Assign Ownership of Appearances to Users
- Assign Default Call Appearance for each User
- Program Feature Name Extensions (FNEs)
- Synchronize Deskstations

6.1. Launch *i* manager

To access the *i* manager software interface, open a web browser and type the *i* manager web address, for example, <http://10.32.24.150/imanager>. Press the **Enter** key. At the *i* manager logon page enter the appropriate credentials. The *i* manager home page is displayed as shown below.



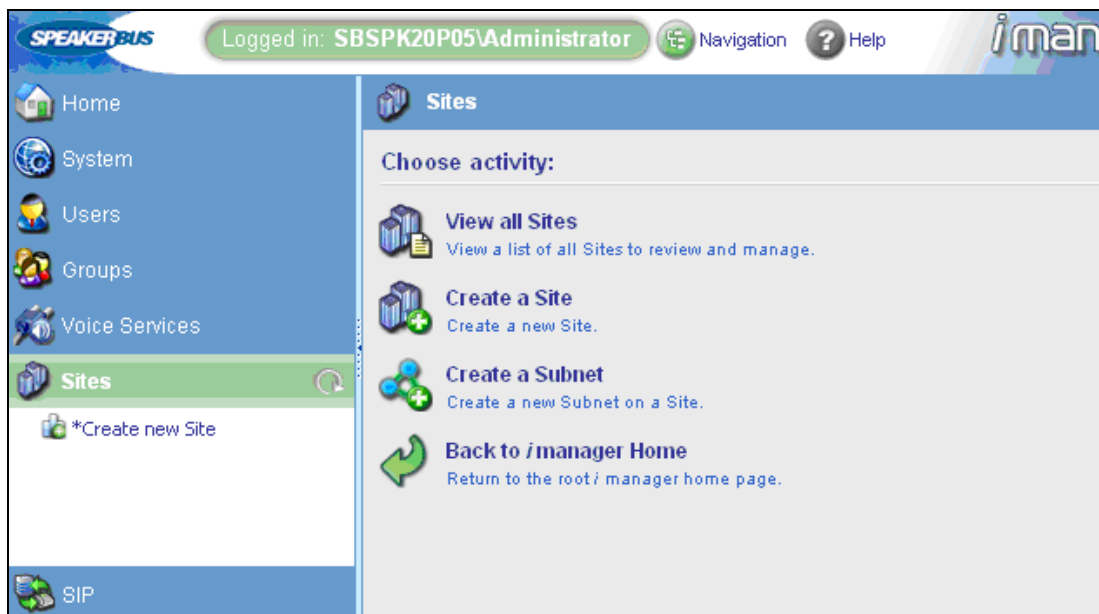
6.2. Verify Product Key

Navigate to System→Product Key to verify that a valid key is installed and sufficient devices are allowed.



6.3. Create Site

Configure a site representing the location where the Speakerbus iD808 devices are installed. Under **Sites** in the left pane, click on **Create new Site**. The **Sites** page is displayed.



In the **General** tab of the **Sites** page, set the **Name** field to a descriptive name. If the *i cms* is located via DNS, then select the **Locate *i cms* using DNS** checkbox. When this option is selected, the *i turret* will use the DNS server to locate the *i cms* server IP address. Refer to [5] for correct configuration of DNS.

The screenshot shows the iManager web interface. At the top, it says 'Logged in: SBSPK20P05\Administrator'. The left sidebar has a menu with 'Home', 'System', 'Users', 'Groups', 'Voice Services', 'Sites' (selected), and 'SIP'. The main content area is titled 'Sites' and has 'OK' and 'Cancel' buttons. Below this is a table with columns 'Name' and 'i cms IP Address'. The table is empty, showing 'No records to display.' and 'Page: 1 of 1' with a 'Reload' button. Below the table are three tabs: 'General' (selected), 'Device Settings', and 'iD808 Settings'. The 'General' tab contains three fields: 'Name' with the value 'Avaya', 'Locate i cms using DNS' with an unchecked checkbox, and 'i cms Primary IP Address' with the value '10.32.24.150'.

In the **iD808 Settings** tab, set the network time protocol time zone and configure the password for logging into the iD808 deskstation. The **NTP Server** field may be set to the IP address of the NTP server if one is used. Click **OK**. The “Avaya” site will be now listed under **Sites**.

6.4. Create Subnet

To create a subnet, click on **Create new Subnet** under the newly configured “Avaya” site. In the **General** tab, provide a descriptive name for the subnet and configure the **Subnet Address** and **Default Gateway Address**.

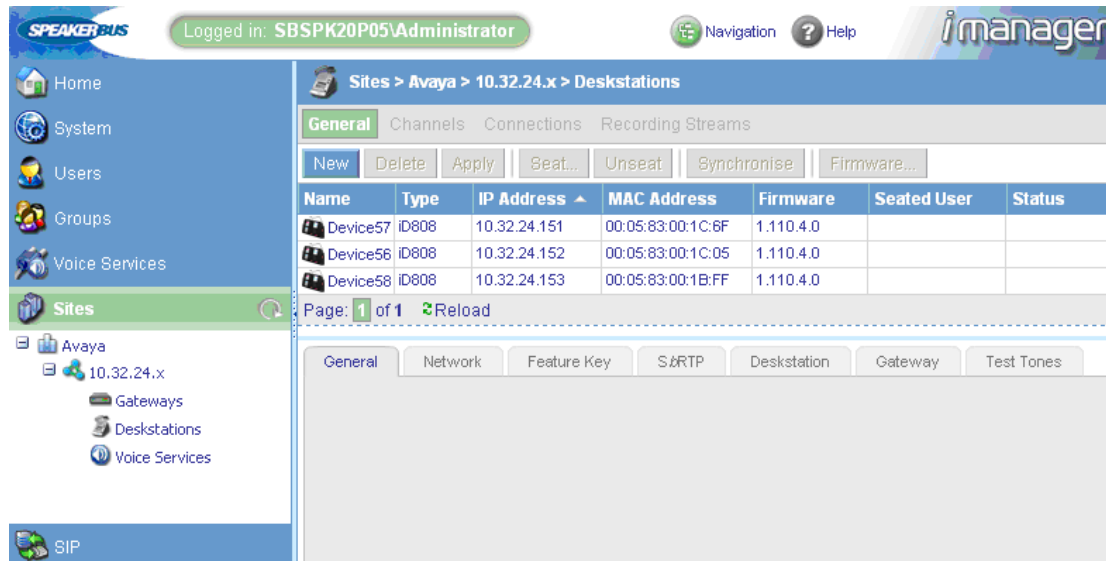
Note: A Service Locator Record (SRV) needs to be added to the DNS server in order to allow the iD808 to locate and register to *i cms*. Refer to [5] for more details.

In the **SbRTP** tab, set the **Compatibility** field to *Version 3.0*. Click **OK**.

The screenshot shows the Speakerbus iManager web interface. The top navigation bar includes the Speakerbus logo, a login status bar showing 'Logged in: SBSPK20P05\Administrator', and links for Navigation and Help. A left sidebar contains a menu with 'Home', 'System', 'Users', 'Groups', 'Voice Services', 'Sites' (highlighted), and 'SIP'. Under 'Sites', there is a tree view showing 'Avaya' and a link '*Create new Subnet'. The main content area is titled 'Sites > Avaya' and contains a table with columns 'Subnet' and 'Subnet Address'. The table is empty, with a message 'No records to display.' and pagination 'Page: 1 of 1' with a 'Reload' link. Below the table, there are two tabs: 'General' and 'SbRTP'. The 'SbRTP' tab is active, showing configuration fields: 'RTP Payload Code' (text box with '96'), 'DSCP Value' (text box with '0'), 'Compatibility' (dropdown menu with 'Version 3.0' selected), 'Packet Size' (dropdown menu with '1 ms' selected), and 'Voice Activity Detection' (checkbox, unchecked).

6.5. Create Deskstations

Once the site and subnet exist, the iD808 devices can be created. The network was set up using DNS without DHCP so IP configuration was manually performed on the iD808 devices. The deskstations then registered to *i* manager through DNS. After the iD808 devices are synchronized with *i* cms, they should automatically be created in *i* manager. To create a device, select the **Sites** directory tree and expand the sites directory. Click on **Deskstations** to display the device list. The newly created devices are automatically displayed in the list. Verify that the deskstations are automatically created as shown below.



The screenshot shows the iManager web interface. The left sidebar contains a navigation tree with 'Sites' selected. The main content area displays the 'Deskstations' list for the site 'Avaya' and subnet '10.32.24.x'. The list has columns for Name, Type, IP Address, MAC Address, Firmware, Seated User, and Status. Three devices are listed: Device57, Device56, and Device58, all of type iD808.

Name	Type	IP Address	MAC Address	Firmware	Seated User	Status
Device57	iD808	10.32.24.151	00:05:83:00:1C:6F	1.110.4.0		
Device56	iD808	10.32.24.152	00:05:83:00:1C:05	1.110.4.0		
Device58	iD808	10.32.24.153	00:05:83:00:1B:FF	1.110.4.0		

Select a device and change the name to a more descriptive one in the **General** tab.



The screenshot shows the iManager web interface with the 'Deskstations' list. The 'Device57' row is highlighted. The 'General' tab is selected, showing the configuration for the selected device. The fields are: Name (iTurret 1), MAC Address (00:05:83:00:1C:6F), Type (iD808), Firmware Version (1.110.4.0), Site (Avaya), Subnet (10.32.24.x), and Location (empty).

Name	Type	IP Address	MAC Address	Firmware	Seated User	Status
Device57	iD808	10.32.24.151	00:05:83:00:1C:6F	1.110.4.0		
Device56	iD808	10.32.24.152	00:05:83:00:1C:05	1.110.4.0		
Device58	iD808	10.32.24.153	00:05:83:00:1B:FF	1.110.4.0		

In the **Network** tab, verify the IP settings. Set the **DNS Server IP Address** and **Local Domain Name** fields, if necessary.

The screenshot shows the iManager web interface. The top navigation bar includes the 'SPEAKERBUS' logo, a login status 'Logged in: SBSPK20P05\Administrator', and links for 'Navigation' and 'Help'. The left sidebar contains a tree view with 'Home', 'System', 'Users', 'Groups', 'Voice Services', and 'Sites'. Under 'Sites', 'Avaya' is expanded, showing '10.32.24.x', 'Gateways', 'Deskstations', and 'Voice Services'. The main content area is titled 'Sites > Avaya > 10.32.24.x > Deskstations'. It features a 'General' tab (selected), 'Channels', 'Connections', and 'Recording Streams' sub-tabs. Below these are buttons for 'New', 'Delete', 'Apply', 'Seat...', 'Unseat', 'Synchronise', and 'Firmware...'. A table lists three devices: Device57, Device56, and Device58, all of type ID808. The table columns are Name, Type, IP Address, MAC Address, Firmware, Seated User, and Status. Below the table, it shows 'Page: 1 of 1' and a 'Reload' button. The 'Network' tab is selected, showing fields for 'Obtain IP Address using DHCP' (unchecked), 'IP Address' (10.32.24.151), 'Subnet Mask' (255.255.255.0), 'Default Gateway' (10.32.24.1), 'DNS Server IP Address' (10.32.24.150), 'Local Host Name' (id808-001C6F), 'Local Domain Name' (avaya.com), and 'Enable SNMP' (unchecked). There are also sections for 'Ethernet Port 1' and 'Ethernet Port 2', each with a mode dropdown set to 'Auto Negotiate' and 'Disabled' respectively.

Name	Type	IP Address	MAC Address	Firmware	Seated User	Status
Device57	ID808	10.32.24.151	00:05:83:00:1C:6F	1.110.4.0		
Device56	ID808	10.32.24.152	00:05:83:00:1C:05	1.110.4.0		
Device58	ID808	10.32.24.153	00:05:83:00:1B:FF	1.110.4.0		

In the **Deskstation** tab, select preferred codec. In this configuration, G.711mu-law is the preferred codec. Click **Apply**. Repeat these steps for all deskstations.

Logged in: SBSPK20P05\Administrator

Navigation ? Help

Home System Users Groups Voice Services Sites SIP

Sites > Avaya > 10.32.24.x > Deskstations

General Channels Connections Recording Streams

New Delete Apply Seat... Unseat Synchronise Firmware...

Name	Type	IP Address	MAC Address	Firmware	Seated User	Status
Device57	ID808	10.32.24.151	00:05:83:00:1C:6F	1.110.4.0		
Device56	ID808	10.32.24.152	00:05:83:00:1C:05	1.110.4.0		
Device58	ID808	10.32.24.153	00:05:83:00:1B:FF	1.110.4.0		

Page: 1 of 1 Reload

General Network Media Deskstation

Seated User: No User seated

Preferred Telephony Codec: G.711 μ-Law

Voice Activity Detection: ☐

i cds Server Address:

i cds Server Port:

6.6. Create SIP Server

To create a SIP Server, click **Create a new SIP Server** under the **SIP** directory in the left pane. Provide a descriptive name for the SIP server and set the **Registrar Address** and **SIP Domain** fields to *avaya.com*, which in this configuration DNS resolves the domain name to 10.32.24.100, the SIP Enablement Services IP address. Click **OK**. After the SIP server is created, the **Port** field will be displayed on this page with the default value of 5060. The default value was used in this configuration.

Note: Create a server locator record (SRV) for registrar address and SIP domain on DNS. Refer to [5] for more details.

Logged in: SBSPK20P05\Administrator

Navigation ? Help

Home System Users Groups Voice Services Sites SIP

SIP

OK Cancel

Name	Registrar Address	SIP Domain
No records to display.		

Page: 1 of 1 Reload

General

Name: Avaya

Registrar Address: avaya.com

SIP Domain: avaya.com

*Create new SIP Server

6.7. Create PBX

Select the **SIP** directory and click **Create new PBX**. Provide a descriptive name/text for the PBX in the **Name** and **Version** fields and set the **Type** field to “Avaya”. The **Outbound** and **Inbound** tabs are left with their default values. Click **OK**.

Logged in: SBSPK20P05\Administrator

Navigation ? Help

SIP > Avaya

OK Cancel

Name	Type	Version
No records to display.		
Page: 1 of 1 Reload		

General Inbound Outbound

Name: Avaya

Type: Avaya

Version: 5.2

6.8. Create Dial Plan

Under the **SIP** directory, click **Dial Plan** and then the **New** button to add a dial rule. Dial rules specify the valid digit formats that the iD808 devices are allowed to dial, otherwise the user will have to press OK after entering the dial string on the iD808 device. In this configuration, 5-digit extensions beginning with ‘7’ were used to dial other iD808 devices and Avaya telephones. A dial rule is also required for the voice mail pilot number which was a 5-digit extension beginning with “29”. The example below corresponds to 5-digit extensions beginning with ‘7’. The X’s in the dial rule match any digit. Note that the X must be a capital letter. Click **OK**. Repeat this for all valid extension formats.

Logged in: SBSPK20P05\Administrator

Navigation ? Help

SIP > Avaya > Avaya > Dial Plan

OK Cancel

Dial Rule
No records to display.
Page: 1 of 1 Reload

General

Dial Rule: 7XXXX

6.9. Create Call and Handset Appearances

Three call appearances need to be created for each iD808 device for the main appearance, handset 1 appearance, and handset 2 appearance. As previously mentioned, three extensions are also required on Communication Manager and SIP Enablement Services. To create the main appearance, click **Appearances** under the “Avaya” PBX under the **SIP** directory. Click the **New** button on the next page to add a new appearance. In 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 iD808 deskstation. In this example, the label was set to the main appearance extension 78200. The **Address** field should also be set to the appearance extension. Set the **Type** field to *Call*.

Logged in: SBSPK20P05\Administrator

SIP > Avaya > Avaya > Appearances

General User Permissions Group Permissions

OK Cancel

Name (Short Label)	Type	Status	Address	Long Label
No records to display.				

Page: 1 of 1 Reload

General Advanced

Name: 78200

Type: Call

Long Label: 78200

Address: 78200

Owner:

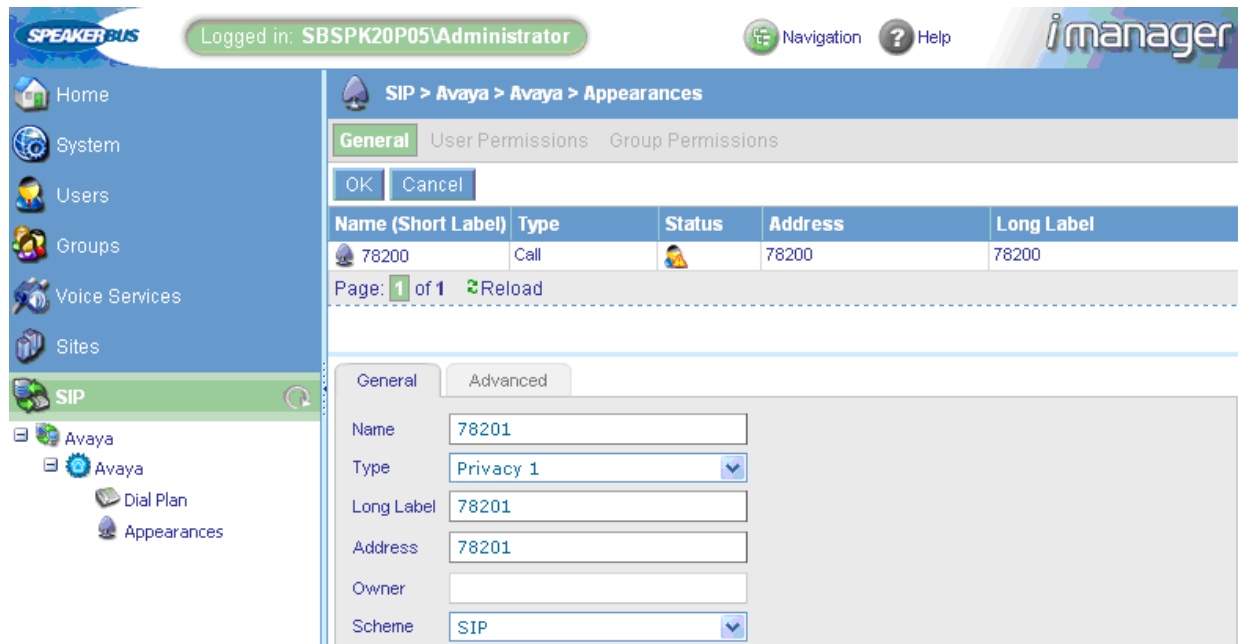
Scheme: SIP

In the **Advanced** tab, set the **Maximum Appearances** field to “3”. This field should be set to the number of call appearances configured on the station in Communication Manager minus one since the last call appearance is restricted. See the button assignment section of the station form and the second page of the Off-PBX-Telephone Station-Mapping form in Section 4.9 as an example. The number of call appearance buttons dictates the number of calls on the system the user can have directed to them. When all of a user’s call appearances are in-use (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 iD808 deskstation. Select the **Message Indication** checkbox for voice mail purposes. The **Authentication Name** and **Authentication Password** fields should be set to the extension and password, respectively, configured on SIP Enablement Services. These are the credentials that the iD808 deskstation will use to authenticate and register with SIP Enablement Services. Use the default values for the other fields as shown below. Click **OK**.

The screenshot shows the iManager web interface. The top navigation bar includes 'SPEAKERBUS', a login status 'Logged in: SBSPK20P05\Administrator', and links for 'Navigation' and 'Help'. The left sidebar contains a tree view with 'SIP' selected, showing sub-items like 'Avaya', 'Avaya', 'Dial Plan', and 'Appearances'. The main content area is titled 'SIP > Avaya > Avaya > Appearances'. It has tabs for 'General', 'User Permissions', and 'Group Permissions', with 'General' being active. Below the tabs are 'OK' and 'Cancel' buttons. A table with columns 'Name (Short Label)', 'Type', 'Status', 'Address', and 'Long Label' is shown, with a message 'No records to display.' and 'Page: 1 of 1' with a 'Reload' link. At the bottom, there are two tabs: 'General' and 'Advanced'. The 'Advanced' tab is selected, showing fields for 'Group Number' (0), 'Maximum Appearances' (3), 'Allow Outbound Calls' (checked), 'Message Indication' (checked), 'Authentication Name' (78200), 'Authentication Password' (masked with dots), and 'Verify Password' (masked with dots).

Next, this procedure will be repeated for the two handset appearances.

Click the **New** button to add another appearance. In the **General** tab, set the **Name**, **Long Label**, and **Address** fields to the extension of handset 1. In this example, the extension is 78201. Review the previous section for a description of these fields. Set the **Type** field to *Privacy 1*.



Logged in: SBSPK20P05\Administrator

Navigation ? Help

SIP > Avaya > Avaya > Appearances

General User Permissions Group Permissions

OK Cancel

Name (Short Label)	Type	Status	Address	Long Label
78200	Call		78200	78200

Page: 1 of 1 Reload

General Advanced

Name: 78201

Type: Privacy 1

Long Label: 78201

Address: 78201

Owner:

Scheme: SIP

In the **Advanced** tab, configure the **Authentication Name** and **Authentication Password** fields with the credentials for registering with SIP Enablement Services. For the handset appearances, the **Maximum Appearances** field should be set to “0” since no calls will be made to the handset appearances directly. The **Message Indication** checkbox does not need to be enabled since the handset appearances are not voice mail subscribers. Handset appearances are hidden on the iD808 deskstation and need to be defined in order for privacy to work on the iD808 with Communication Manager. Click **OK**.

The screenshot shows the iManager web interface. The top navigation bar includes the 'SPEAKERBUS' logo, a login status 'Logged in: SBSPK20P05\Administrator', and links for 'Navigation' and 'Help'. The left sidebar contains a tree view with 'SIP' selected, showing sub-items: 'Avaya', 'Avaya', 'Dial Plan', and 'Appearances'. The main content area is titled 'SIP > Avaya > Avaya > Appearances'. It features three tabs: 'General' (active), 'User Permissions', and 'Group Permissions'. Below the tabs are 'OK' and 'Cancel' buttons. A table lists the configuration for a single appearance:

Name (Short Label)	Type	Status	Address	Long Label
78200	Call		78200	78200

Below the table, it says 'Page: 1 of 1' and 'Reload'. The 'Advanced' tab is also visible, showing the following configuration fields:

- Group Number: 0
- Maximum Appearances: 0
- Allow Outbound Calls: ☒
- Message Indication: ☐
- Authentication Name: 78201
- Authentication Password: (masked with dots)
- Verify Password: (masked with dots)

Repeat the above procedure to add the handset 2 appearance.

The three call appearances for the previously configured iD808 deskstation are listed below.

The screenshot shows the iManager web interface. The top navigation bar includes the 'SPEAKERBUS' logo, a login status 'Logged in: SBSPK20P05\Administrator', and links for 'Navigation' and 'Help'. The left sidebar contains a tree view with categories: Home, System, Users, Groups, Voice Services, Sites, and SIP. Under the SIP category, there are sub-items: Avaya, Avaya, Dial Plan, and Appearances. The main content area is titled 'SIP > Avaya > Avaya > Appearances'. It features tabs for 'General', 'User Permissions', and 'Group Permissions'. Below the tabs are buttons: 'New', 'Delete', 'Apply', 'Assign Ownership...', and 'Clear Ownership'. A table lists three call appearances:

Name (Short Label)	Type	Status	Address	Long Label
78200	Call		78200	78200
78201	Privacy 1		78201	78201
78202	Privacy 2		78202	78202

Below the table, it says 'Page: 1 of 1' and 'Reload'. At the bottom of the main area, there are tabs for 'General' and 'Advanced'.

Repeat the above procedure for adding the main and handset appearances for each iD808 deskstation.

6.10. Create Users

In this section, the users are created. Click on **Users** in the directory tree and then click on **Create a User** link on the next page to add a new user. In the **General** tab, provide a descriptive name in the **Name** field.

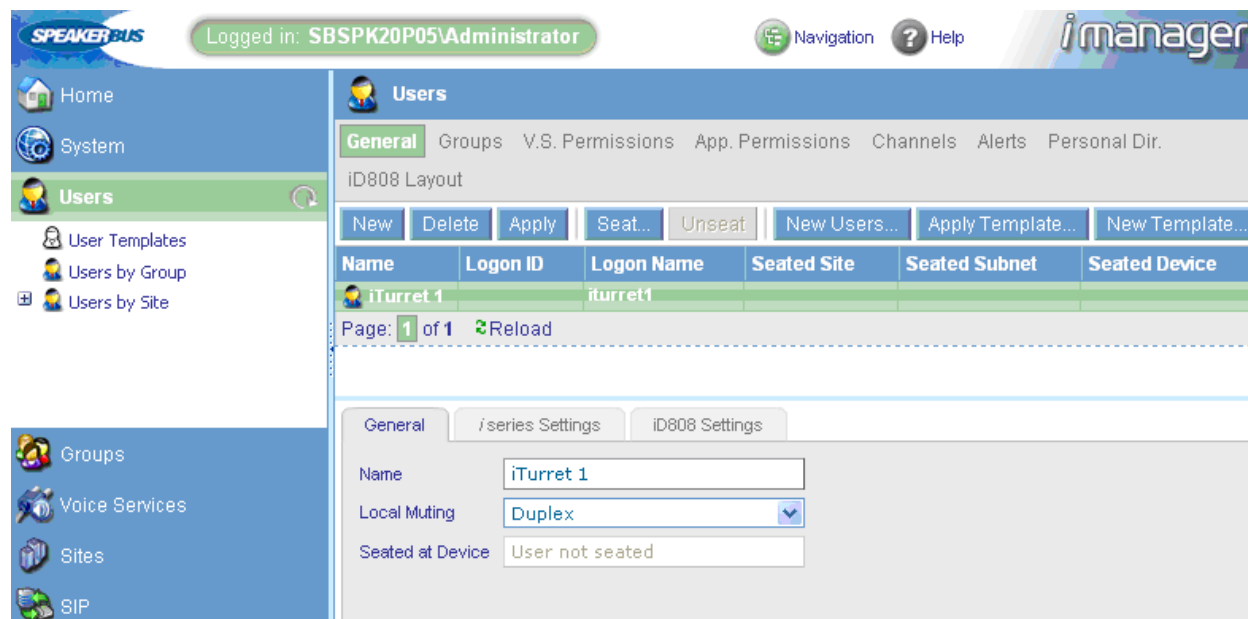
The screenshot shows the iManager web interface. The top navigation bar includes 'SPEAKERBUS', a login status 'Logged in: SBSPK20P05\Administrator', and links for 'Navigation' and 'Help'. The left sidebar contains a directory tree with 'Home', 'System', 'Users' (selected), 'User Templates', 'Users by Group', and 'Users by Site'. Below this are 'Groups', 'Voice Services', 'Sites', and 'SIP'. The main content area is titled 'Users' and has tabs for 'General', 'Groups', 'V.S. Permissions', 'App. Permissions', 'Channels', 'Alerts', and 'Personal Dir.'. The 'General' tab is active, showing an 'ID808 Layout' section with 'OK' and 'Cancel' buttons. Below this is a table with columns: 'Name', 'Logon ID', 'Logon Name', 'Seated Site', 'Seated Subnet', and 'Seated Device'. The table is empty, displaying 'No records to display.' and 'Page: 1 of 1' with a 'Reload' link. At the bottom, there are three tabs: 'General', '/series Settings', and 'ID808 Settings'. The 'General' tab is selected, showing fields for 'Name' (containing 'iTurret 1') and 'Local Muting' (set to 'Duplex').

In the **ID808 Settings** tab, provide the logon credentials for the user to log into their iD808 deskstation. This page will be revisited later in **Section 6.14** to configure the default call appearance for this deskstation. Click **OK**.

This screenshot shows the same iManager interface as the previous one, but with the 'ID808 Settings' tab selected. The 'General' tab is still visible at the top. The 'ID808 Settings' tab contains several input fields: 'Logon Name' (filled with 'iturret1'), 'Logon Password' (masked with dots), 'Verify Password' (masked with dots), 'Voicemail Address' (filled with '29000'), and 'Calls to display on Dynamic Keys' (set to 'All Calls'). The 'OK' and 'Cancel' buttons are still present above the table.

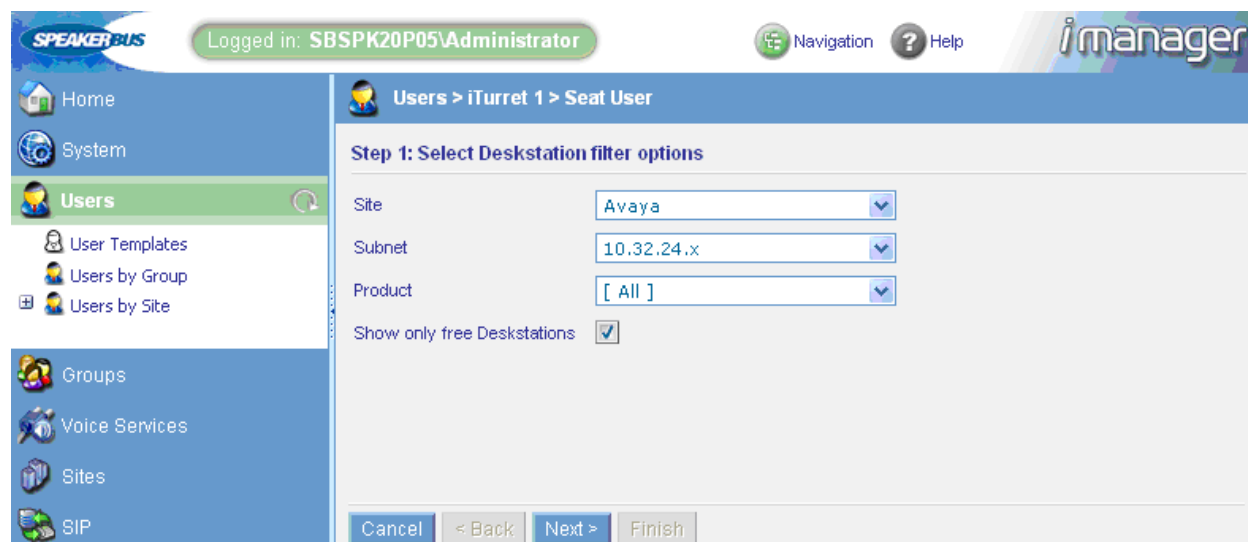
Repeat the previous procedure to add more users.

After a user has been created, the user needs to be “seated” on an iD808 deskstation. In the **Users** directory tree, click on the “Avaya” link under **Users by Site** to display the list of users. Select the user previously configured (i.e., iTurret 1) and click on the **Seat...** button.



The screenshot shows the iManager web interface. The top navigation bar includes 'SPEAKERBUS', 'Logged in: SBSPK20P05\Administrator', 'Navigation', 'Help', and the 'iManager' logo. The left sidebar contains a tree view with 'Home', 'System', 'Users' (selected), 'User Templates', 'Users by Group', 'Users by Site', 'Groups', 'Voice Services', 'Sites', and 'SIP'. The main content area is titled 'Users' and has tabs for 'General', 'Groups', 'V.S. Permissions', 'App. Permissions', 'Channels', 'Alerts', and 'Personal Dir.'. Under the 'General' tab, there's a section for 'iD808 Layout' with buttons: 'New', 'Delete', 'Apply', 'Seat...', 'Unseat', 'New Users...', 'Apply Template...', and 'New Template...'. Below these is a table with columns: 'Name', 'Logon ID', 'Logon Name', 'Seated Site', 'Seated Subnet', and 'Seated Device'. The table contains one entry: 'iTurret 1' with 'iturret1' in the Logon Name column. Below the table, it says 'Page: 1 of 1' and 'Reload'. At the bottom, there are tabs for 'General', 'iSeries Settings', and 'iD808 Settings'. The 'General' tab is active, showing fields for 'Name' (iTurret 1), 'Local Muting' (Duplex), and 'Seated at Device' (User not seated).

On the next page, filter options are presented. Filter deskstations in the “Avaya” site and in the “10.32.24.x” subnet as shown below. The user will be seated on an iD808 deskstation with these properties. Click **Next**.



The screenshot shows the 'Users > iTurret 1 > Seat User' page in iManager. The top navigation bar is the same as the previous screenshot. The left sidebar is also the same. The main content area is titled 'Users > iTurret 1 > Seat User' and has a section 'Step 1: Select Deskstation filter options'. This section contains three dropdown menus: 'Site' (set to 'Avaya'), 'Subnet' (set to '10.32.24.x'), and 'Product' (set to '[All]'). There is also a checkbox labeled 'Show only free Deskstations' which is checked. At the bottom, there are buttons: 'Cancel', '< Back', 'Next >', and 'Finish'.

In the following deskstation list, select the iD808 deskstation where the selected user will be seated. In this example, the user will be seated on the “iTurret 1” deskstation. Select “iTurret 1” in the list and click **Finish**.

Logged in: SBSPK20P05\Administrator

Navigation ? Help

Users > iTurret 1 > Seat User

Step 2: Select a Deskstation to seat the User at

Selected Deskstation: iTurret 1

Name	Site	Subnet	Product	IP Address	MAC Address	Seated User
iTurret 1	Avaya	10.32.24.x	iD808	10.32.24.151	00:05:83:00:1C:6F	
iTurret 2	Avaya	10.32.24.x	iD808	10.32.24.152	00:05:83:00:1C:05	
iTurret 3	Avaya	10.32.24.x	iD808	10.32.24.153	00:05:83:00:1B:FF	

Page: 1 of 1 Reload

Cancel < Back Next > Finish

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

Logged in: SBSPK20P05\Administrator

Navigation ? Help

Users

General Groups V.S. Permissions App. Permissions Channels Alerts Personal Dir.

iD808 Layout

New Delete Apply Seat... Unseat New Users... Apply Template... New Template...

Name	Logon ID	Logon Name	Seated Site	Seated Subnet	Seated Device
iTurret 1	iturret1	iturret1	Avaya	10.32.24.x	iTurret 1

Page: 1 of 1 Reload

General /series Settings iD808 Settings

Name: iTurret 1

Local Muting: Duplex

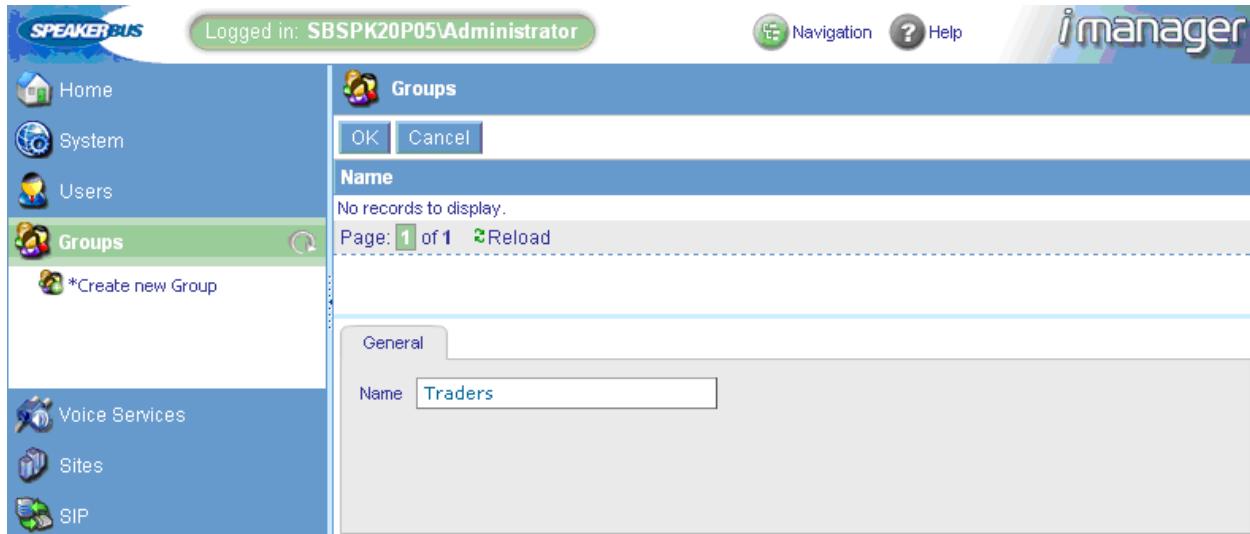
Seated at Site: Avaya

Seated at Subnet: 10.32.24.x

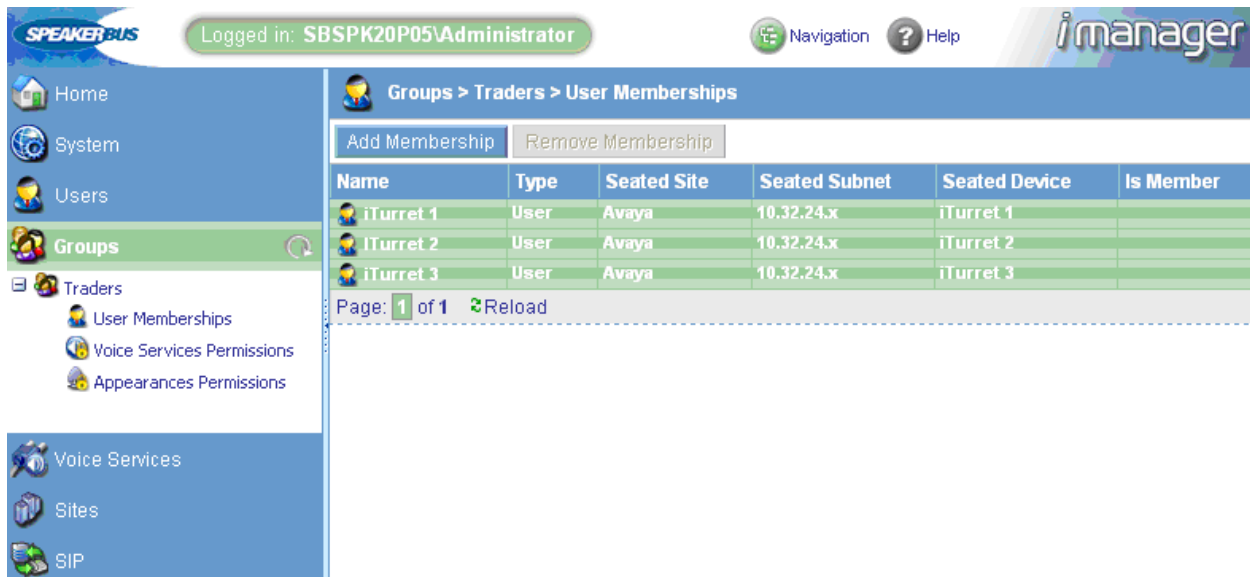
Seated at Device: iTurret 1

6.11. Create Group

To create a group, click on **Create new Group** in the **Groups** directory tree. In the **General** tab, provide a descriptive name in the **Name** field, such as “Traders”. Click **OK**. The “Traders” group has been successfully added. Next, the user will be added to this new group.



In the **Groups** directory tree, expand “Traders” and click on **User Memberships** in the left pane. A list of users is displayed. Select all the users to be added to the “Traders” group as shown below and then click **Add Membership**. The **Is Member** column will then indicate that the selected users are members of the “Traders” group.



6.12. Assign User Permissions

The next step will be to assign appearances permissions to users. In the SIP directory tree, click on **Appearances** under “Avaya”. The list of appearances is displayed. Select the main call appearance for “iTurret 1” (i.e., 78200) and click **User Permissions**.

The screenshot shows the iManager interface with the 'SIP > Avaya > Avaya > Appearances' path selected. The 'General' tab is active, displaying a table of appearances. The table has columns: Name (Short Label), Type, Status, Address, and Long Label. The first row, 78200, is highlighted. Below the table, it says 'Page: 1 of 1' and 'Reload'.

Name (Short Label)	Type	Status	Address	Long Label
78200	Call		78200	78200
78201	Privacy 1		78201	78201
78202	Privacy 2		78202	78202
78205	Call		78205	78205
78206	Privacy 1		78206	78206
78207	Privacy 2		78207	78207
78210	Call		78210	78210
78211	Privacy 1		78211	78211
78212	Privacy 2		78212	78212

Select the user on the next page to which the appearance will be assigned. Set the **Permission** field to *Allow* as shown below. Click **Apply**. Assign “Privacy 1” and “Privacy 2” permissions to this user by repeating this procedure.

The screenshot shows the iManager interface with the 'SIP > Avaya > Avaya > 78201 > User Permissions' path selected. The 'User Permissions' tab is active, displaying a table of permissions. The table has columns: Name, User Permission, Group Permission, Type, Seated Site, Seated Subnet, and Seated Device. The first row, iTurret 1, is highlighted. Below the table, it says 'Page: 1 of 1' and 'Reload'. Below the table, there is a 'Permission' dropdown menu set to 'Allow'.

Name	User Permission	Group Permission	Type	Seated Site	Seated Subnet	Seated Device
iTurret 1	Use group	Deny	User	Avaya	10.32.24.x	iTurret 1
iTurret 2	Use group	Deny	User	Avaya	10.32.24.x	iTurret 2
iTurret 3	Use group	Deny	User	Avaya	10.32.24.x	iTurret 3

6.13. Assign Ownership

Assign ownership of the appearances to a user. In the **SIP** directory tree, click on **Appearances** to display the appearances list as shown below. In the **General** tab, select the main call appearance and click on the **Assign Ownership** button.

The screenshot shows the iManager web interface for managing SIP appearances. The top navigation bar includes the SPEAKERBUS logo, a login status 'Logged in: SBSPK20P05\Administrator', and links for Navigation and Help. The left sidebar contains a tree view with categories like Home, System, Users, Groups, Voice Services, Sites, and SIP. Under the SIP category, there are sub-items for Avaya, Avaya, Dial Plan, and Appearances. The main content area is titled 'SIP > Avaya > Avaya > Appearances' and features tabs for General, User Permissions, and Group Permissions. The General tab is active, displaying a table of appearances and a form for editing a selected entry.

Logged in: SBSPK20P05\Administrator

Navigation ? Help

SIP > Avaya > Avaya > Appearances

General User Permissions Group Permissions

New Delete Apply Assign Ownership... Clear Ownership

Name (Short Label)	Type	Status	Address	Long Label
78200	Call		78200	78200
78201	Privacy 1		78201	78201
78202	Privacy 2		78202	78202
78205	Call		78205	78205
78206	Privacy 1		78206	78206
78207	Privacy 2		78207	78207
78210	Call		78210	78210
78211	Privacy 1		78211	78211
78212	Privacy 2		78212	78212

Page: 1 of 1 Reload

General Advanced

Name: 78200

Type: Call

Long Label: 78200

Address: 78200

Owner:

Scheme: SIP

The next page displays filter options. Filter users by selecting the “Avaya” site and the “Traders” group and click **Next**.

The screenshot shows the iManager interface with the left sidebar containing navigation links: Home, System, Users, Groups, Voice Services, Sites, SIP, Avaya, Avaya, Dial Plan, and Appearances. The main content area is titled "SIP > Avaya > Avaya > 78200 > Assign Ownership". Below the title bar, it says "Step 1: Select User filter options". There are three filter options: "Show only unseated users" with a checkbox, "Filter by Site" with a dropdown menu set to "Avaya", and "Filter by Group" with a dropdown menu set to "Traders". There is also a "Filter by Name" text input field. At the bottom, there are navigation buttons: "Cancel", "< Back", "Next >", and "Finish".

On the next page, select the user to which ownership will be assigned to the main call appearance. In this example, the main call appearance 78200 will be assigned to “iTurret 1”. Click **Finish**.

The screenshot shows the iManager interface with the left sidebar containing navigation links: Home, System, Users, Groups, Voice Services, Sites, SIP, Avaya, Avaya, Dial Plan, and Appearances. The main content area is titled "SIP > Avaya > Avaya > 78200 > Assign Ownership". Below the title bar, it says "Step 2: Select a User to assign ownership to". There is a "Selected User" text input field containing "iTurret 1". Below this is a table with four columns: "Name", "Seated Site", "Seated Subnet", and "Seated Device". The table contains three rows of data. At the bottom of the table, it says "Page: 1 of 1" and "Reload". At the bottom of the page, there are navigation buttons: "Cancel", "< Back", "Next >", and "Finish".

Name	Seated Site	Seated Subnet	Seated Device
iTurret 1	Avaya	10.32.24.x	iTurret 1
iTurret 2	Avaya	10.32.24.x	iTurret 2
iTurret 3	Avaya	10.32.24.x	iTurret 3

Repeat this procedure to assign “Privacy 1” and “Privacy 2” to “iTurret 1”.

6.14. Assign Default Call Appearance

In the **Users** directory tree, click on the “Avaya” link to display the users list. Set the **Default Appearance** field to the main call appearance (e.g., 78200). Click **Apply**.

The screenshot shows the iManager interface. The top navigation bar includes the SPEAKERbus logo, a login status bar showing 'Logged in: SBSPK20P05\Administrator', and links for Navigation and Help. The left sidebar contains a directory tree with 'Users' selected, showing sub-items like 'User Templates', 'Users by Group', 'Users by Site', '*Not seated', and 'Avaya'. The main content area displays the 'Users > By Site > Avaya' view. It includes a 'General' tab and a table of users. Below the table is a 'Page: 1 of 1' indicator and a 'Reload' button. The bottom section shows the 'ID808 Settings' for a user, with fields for 'Logon Name', 'Voicemail Address', 'Default Appearance Type', 'Default Appearance', and 'Calls to display on Dynamic Keys'.

Logged in: SBSPK20P05\Administrator

Navigation ? Help

Home System Users

User Templates Users by Group Users by Site *Not seated Avaya

Users > By Site > Avaya

General Groups V.S. Permissions App. Permissions Channels Alerts Personal Dir.

ID808 Layout

Name	Logon ID	Logon Name	Seated Site	Seated Subnet	Seated Device
iTurret 1		iturret1	Avaya	10.32.24.x	iTurret 1
iTurret 2		iturret2	Avaya	10.32.24.x	iTurret 2
iTurret 3		iturret3	Avaya	10.32.24.x	iTurret 3

Page: 1 of 1 Reload

General /series Settings ID808 Settings

Logon Name: iturret1
Change Password...

Voicemail Address: 29000

Default Appearance Type: Call

Default Appearance: 78200

Calls to display on Dynamic Keys: All Calls

6.15. Program Feature Name Extensions (FNEs)

In this section, the Feature Name Extensions (FNEs) are configured. FNEs can be created in the Corporate directory or Personal directory. In this example, the FNEs are created under the Personal directory of an iD808 deskstation. This makes the FNE available to this deskstation only as opposed to the whole corporation.

In the **Users** directory tree, click on the “Avaya” link and then select a user (e.g., iTurret 1). Click on **Personal Dir.**

The screenshot shows the iManager web interface. The top navigation bar includes the SPEAKERBUS logo, a login status bar showing 'Logged in: SBSPK20P05\Administrator', and links for Navigation and Help. The left sidebar contains a directory tree with 'Users' selected, showing sub-items like 'User Templates', 'Users by Group', 'Users by Site', '*Not seated', and 'Avaya'. The main content area displays the 'Users > By Site > Avaya' view. It includes tabs for 'General', 'Groups', 'V.S. Permissions', 'App. Permissions', 'Channels', 'Alerts', and 'Personal Dir.'. Below these is a table of users with columns: Name, Logon ID, Logon Name, Seated Site, Seated Subnet, and Seated Device. The table lists three users: iTurret 1, iTurret 2, and iTurret 3. Below the table is a 'Page: 1 of 1' indicator and a 'Reload' button. The bottom section shows the configuration for the selected user, iTurret 1, with tabs for 'General', '/series Settings', and 'iD808 Settings'. The 'General' tab is active, showing fields for Name (iTurret 1), Local Muting (Duplex), Seated at Site (Avaya), Seated at Subnet (10.32.24.x), and Seated at Device (iTurret 1).

Name	Logon ID	Logon Name	Seated Site	Seated Subnet	Seated Device
iTurret 1		iturret1	Avaya	10.32.24.x	iTurret 1
iTurret 2		iturret2	Avaya	10.32.24.x	iTurret 2
iTurret 3		iturret3	Avaya	10.32.24.x	iTurret 3

Page: 1 of 1 Reload

General /series Settings iD808 Settings

Name: iTurret 1

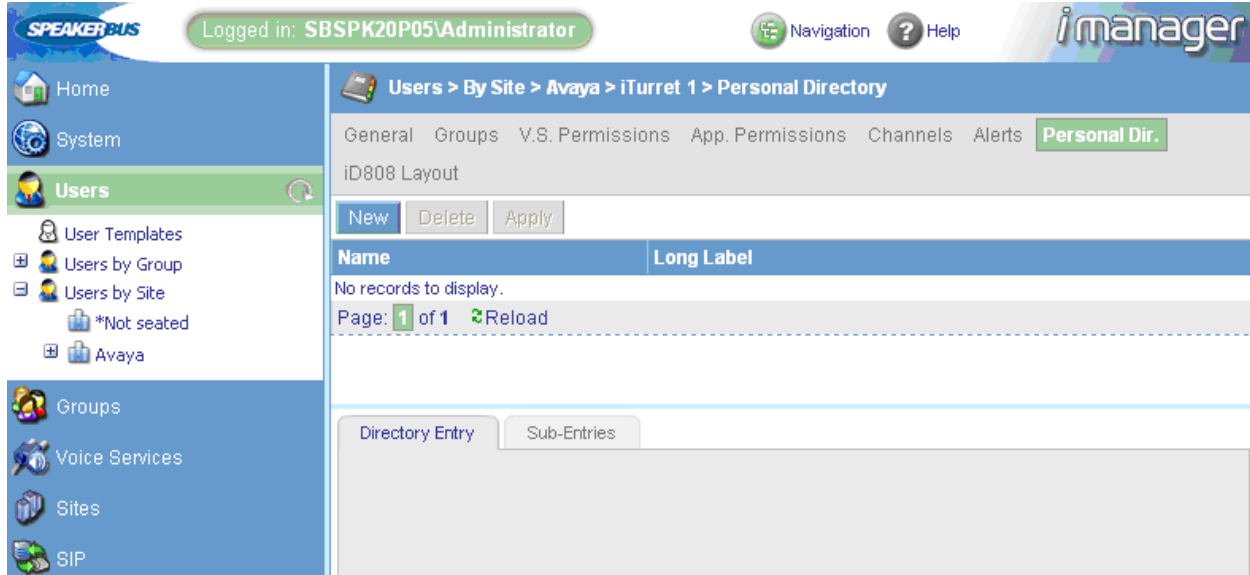
Local Muting: Duplex

Seated at Site: Avaya

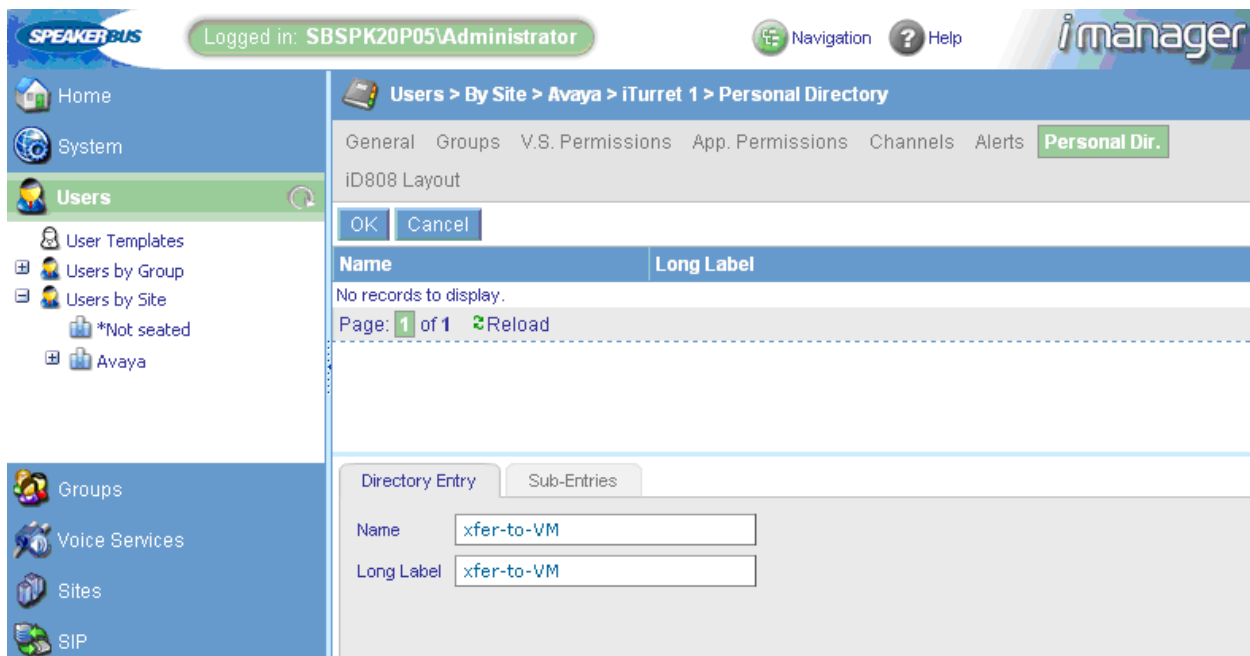
Seated at Subnet: 10.32.24.x

Seated at Device: iTurret 1

On the next page, click the **New** button to add an FNE in the personal directory.



In the **Directory Entry** tab, provide the name of the FNE in the **Name** and **Long Label** fields and set the **Default Sub-Entry Address** to the FNE extension configured in **Section 4.5**. In this example, the transfer-to-voicemail FNE is added. Click **OK**. Repeat for other FNEs of interest.



SPEAKERbus Logged in: SBSPK20P05\Administrator Navigation ? Help **iManager**

Home System **Users** User Templates Users by Group Users by Site *Not seated Avaya

Users > By Site > Avaya > iTurret 1 > Personal Directory

General Groups V.S. Permissions App. Permissions Channels Alerts **Personal Dir.**

iD808 Layout

New Delete Apply

Name	Long Label
xfer-to-VM	xfer-to-VM

Page: 1 of 1 Reload

Directory Entry Sub-Entries

Type	Address	Default
General	78131	<input checked="" type="radio"/>
		<input type="radio"/>
		<input type="radio"/>
		<input type="radio"/>

Refer to [5] for instructions on how to create iD808 deskstation keys. The following keys can be created using the iD808 layout page: Dynamic, Appearance, Shortcut, Soft Function, and Speed Dial amongst others. For the compliance test, each user was configured with three Dynamic keys, two Soft Function keys, and one Shortcut key. Although the configuration may vary, this configuration is suitable for most users.

6.16. Synchronize Deskstations

To send the new configuration to the iD808 deskstations, the deskstations need to be synchronized with *i cms*. In the **Sites** directory tree, click on **Deskstations** to display the deskstation list. Select the deskstations as shown below and click the **Synchronise** button. The iD808 deskstation will indicate that they are being synchronized on their displays. After the deskstations have been synchronized, the status icons on the iD808 deskstations corresponding to the network, *i cms*, and SIP registrar status should be green.

Note: Active calls will be dropped.

The screenshot shows the iManager web interface. The top navigation bar includes the 'SPEAKERBUS' logo, a login status 'Logged in: SBSPK20P05\Administrator', and links for 'Navigation' and 'Help'. The left sidebar contains a tree view with 'Home', 'System', 'Users', 'Groups', 'Voice Services', 'Sites', and 'SIP'. The 'Sites' section is expanded, showing 'Avaya' and its sub-items: '10.32.24.x', 'Gateways', 'Deskstations', and 'Voice Services'. The main content area is titled 'Sites > Avaya > 10.32.24.x > Deskstations'. It features a tabbed interface with 'General' selected, and buttons for 'New', 'Delete', 'Apply', 'Seat...', 'Unseat', 'Synchronise', and 'Firmware...'. Below these is a table of deskstations:

Name	Type	IP Address	MAC Address	Firmware	Seated User	Status
iTurret 1	iD808	10.32.24.151	00:05:83:00:1C:6F	1.110.4.0	iTurret 1	
iTurret 2	iD808	10.32.24.152	00:05:83:00:1C:05	1.110.4.0	iTurret 2	
iTurret 3	iD808	10.32.24.153	00:05:83:00:1B:FF	1.110.4.0	iTurret 3	

Below the table, it says 'Page: 1 of 1' and 'Reload'. At the bottom, there is a configuration panel for 'iTurret 1' with tabs for 'General', 'Network', 'Media', and 'Deskstation'. The 'Deskstation' tab is active, showing fields for 'Name' (iTurret 1), 'MAC Address' (00:05:83:00:1C:6F), and 'Type' (iD808).

7. General Test Approach and Test Results

To verify interoperability of Speakerbus iD808 *i* turret with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services, calls were made between iD808 deskstations and Avaya SIP, H.323, digital, and analog stations using various codec settings and exercising common PBX features. The telephony features were activated and deactivated using buttons and menu options on the *i* Turret, FNEs, and FNUUs. The PBX features listed in Section 1 were covered. Speakerbus iD808 *i* turret passed compliance testing with the following observation.

Observation(s): Calls between an *i* turret and an Avaya SIP telephone are not shuffled. However, calls between two *i* turrets or between an *i* turret and an Avaya H.323 telephone are shuffled.

8. Verification Steps

All features shown in **Table 1** were tested using the sample configuration. The following steps can be used to verify and/or troubleshoot installations in the field.

1. On the Speakerbus iD808 *i* turret, verify that the status icons are green. These status icons indicate whether the *i* turret is connected to the network, *i* cms server, and SIP registrar (i.e., Avaya SIP Enablement Services). Refer to [5] for more details.
2. Verify that the iD808 deskstations have successfully registered with SIP Enablement Services. Each iD808 deskstation contains three extensions on SIP Enablement Services for one main appearance and two handset appearances.
3. Verify basic feature set administration by making calls from one *i* turret to another *i* turret and phones. Test supported features according to **Table 1** and feature deployment plans at the site.
4. Verify that speed dials defined locally at the *i* turret are correct. If any are missing or are inoperative, use *i* manager administration to re-check the configuration.
5. Verify extended OPS features by selecting the speed dial button for the feature or dialing the FNE. If busy or intercept tone is heard, check Communication Manager administration for the correct FNE, proper permissions under COS/COR, and the proper station button assignment to support the feature.
6. Call an *i* turret that currently has no voice messages, and leave a message. Verify that the message waiting indicator illuminates on the called *i* turret. Press the **Message** soft key on that *i* turret and verify that the voice messaging system is called. Use the voice messaging menus to retrieve and delete the voice message, verifying that DTMF is interpreted correctly by the system, and that the message waiting indicator extinguishes.

9. Conclusion

These Application Notes have described the administration steps required to use Speakerbus iD808 *i* turret with Avaya AuraTM Communication Manager and Avaya AuraTM SIP Enablement Services. Both basic and extended feature sets were covered as shown in **Table 1**. The extended set includes features not yet available to SIP telephones via the current IETF standards.

10. 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 AuraTM Communication Manager*, Release 5.2, May 2009, Issue 5.0, Document Number 03-300509.
- [2] *Avaya Extension to Cellular User Guide*, April 2009, Issue 12, Document Number 210-100-700.
- [3] *SIP Support in Avaya AuraTM Communication Manager Running on the Avaya S8xxx Servers*, May 2009, Issue 9, Document Number 555-245-206.
- [4] *Installing, Administering, Maintaining, and Troubleshooting Avaya AuraTM SIP Enablement Services*, May 2009, Issue 7.0, Document Number 03-600768.
- [5] *Speakerbus i manager Administrator's Guide*, V1.210, Revision 5, November 2009.
- [6] *Session Initiation Protocol Service Examples - draft-ietf-sipping-service-examples-14*, SIPPING Working Group, Internet-Draft, 7/16/2007, available at <http://tools.ietf.org/wg/sipping/draft-ietf-sipping-service-examples/draft-ietf-sipping-service-examples-14.txt>.

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