

Avaya Solution & Interoperability Test Lab

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

Abstract

These Application Notes describe the steps required to connect Speakerbus iD808 *i* turret to a SIP infrastructure consisting of Avaya SIP Enablement Services and Avaya Communication Manager. Also described is how Avaya 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 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 SIP Enablement Services (SES) server and Avaya S8730 Servers with G650 Media Gateway running Avaya Communication Manager. Also described is how Avaya 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 Communication Manager to the Speakerbus iD808 *i* turret¹, providing the iD808 desk station with enhanced calling features. The configuration steps described are also applicable to other Linux-based Avaya Servers and Media Gateways running Avaya 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 Avaya SIP Enablement Services and Avaya Communication Manager with OPS. In addition to basic calling capabilities, the Internet Engineering Task Force (IETF) has defined a supplementary set of calling features, often referred to as the SIPPING-19 [7]. 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. Avaya 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 Avaya Communication Manager as OPS.

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

	Supported		
FEATURE	Locally at the Phone	With Avaya SIP Offer	COMMENTS
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 admin or OPS FNE
Call Forward Busy/No Answer	Yes	Yes	Local admin or OPS FNE
Call Forward Cancel	Yes	Yes	Local admin or OPS FNE
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-Si	de 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	Via OPS FNE, Local hard key on i turret
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

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 SIP Enablement Services, a pair of Avaya S8730 Servers with a G650 Media Gateway running Avaya Communication Manager, and Avaya IP and digital endpoints. Avaya Modular Messaging provides voice mail service. The enterprise site also contains three Speakerbus iD808 *i* turret deskstations that register with Avaya SIP Enablement Services and are configured as OPS stations on Avaya Communication Manager. Avaya Communication Manager extends the telephony functionality that is supported by the SIP-based iD808 devices through the use of Feature Name Extensions (FNEs). The *i* cms server contains the *i* manager application for configuring the iD808 deskstations.

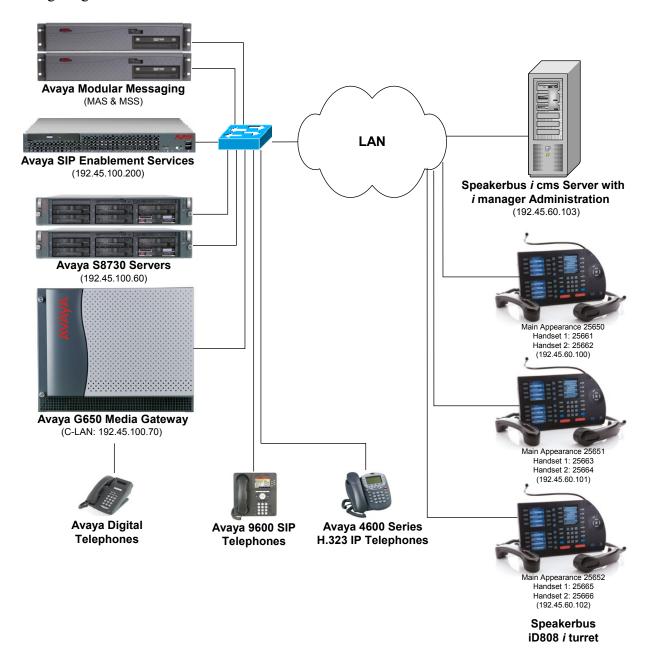


Figure 1: Speakerbus iD808 i turret with Avaya SIP Solution

2. Equipment and Software Validated

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

Hardware Component	Version
Avaya S8730 Servers	Communication Manager 5.1.1
	(R015x.01.1.415.1) with Service Pack 1 (Patch 16402)
Avaya G650 Media Gateway	
TN2302AP Media Processor	HW11 FW118
Avaya SIP Enablement Services	5.1 (SES-5.1.1.0-415.1)
Avaya Modular Messaging	4.0
Avaya 4600 Series IP Telephone	2.8 (H.323)
Avaya 9600 Series IP Telephones	2.0.5 (SIP)
Avaya Digital Telephones	
Speakerbus iD808 i turret	1.0.14.0
Speakerbus <i>i</i> cms Server with <i>i</i> manager Administration on Windows 2003 Server	1.100.11.0

3. Configure 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 Communication Manager and Avaya SIP Enablement Services. For complete SIP documentation, see References [2] [3]. Use the System Access Terminal (SAT) to configure Avaya Communication Manager. Log in with the appropriate credentials.

3.1. 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 746
                                 Maximum Stations: 36000 258
                             Maximum XMOBILE Stations: 0
                   Maximum Off-PBX Telephones - EC500: 0
                   Maximum Off-PBX Telephones - OPS: 50
                                                             33
                   Maximum Off-PBX Telephones - PBFMC: 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
                                                                       2 of 11
                                                                Page
                                OPTIONAL FEATURES
IP PORT CAPACITIES
                                                              USED
                    Maximum Administered H.323 Trunks: 5000
          Maximum Concurrently Registered IP Stations: 12000 6
            Maximum Administered Remote Office Trunks: 0
Maximum Concurrently Registered Remote Office Stations: 0
                                                              Ω
             Maximum Concurrently Registered IP eCons: 0
                                                              Ω
 Max Concur Registered Unauthenticated H.323 Stations: 10
                  Maximum Video Capable H.323 Stations: 0
                  Maximum Video Capable IP Softphones: 0
                      Maximum Administered SIP Trunks: 555
                                                              80
 Maximum Administered Ad-hoc Video Conferencing Ports: 0
                                                              Λ
  Maximum Number of DS1 Boards with Echo Cancellation: 0
                                                              Λ
                            Maximum TN2501 VAL Boards: 128
                    Maximum Media Gateway VAL Sources: 0
          Maximum TN2602 Boards with 80 VoIP Channels: 128
         Maximum TN2602 Boards with 320 VoIP Channels: 128
   Maximum Number of Expanded Meet-me Conference Ports: 0
        (NOTE: You must logoff & login to effect the permission changes.)
```

3.2. 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 Avaya Communication Manager features that are also available to OPS stations.

```
change system-parameters features
                                                                    Page
                                                                            4 of 17
                         FEATURE-RELATED SYSTEM PARAMETERS
          Reserved Slots for Attendant Priority Queue: 5
                            Time before Off-hook Alert: 10
               Emergency Access Redirection Extension:
Number of Emergency Calls Allowed in Attendant Queue: 5
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
 Deluxe Paging and Call Park Timeout to Originator? y
 Controlled Outward Restriction Intercept Treatment: tone
Controlled Termination Restriction (Do Not Disturb): tone
          Controlled Station to Station Restriction: tone
AUTHORIZATION CODE PARAMETERS
                                      Authorization Codes Enabled? n
               Controlled Toll Restriction Replaces: none
```

```
Page 17 of 17
change system-parameters features
                       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
          Display Information With Bridged Call? n
                            Pickup On Transfer? y
```

3.3. Define the Dial Plan

Use the **change dialplan analysis** command to define the dial plan formats 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 2 and 3, FNEs are five digits beginning with 2, and the FACs have formats as indicated with **Call Type** "fac".

change dialplar	analys	is					Page	1 of	12
			DIAL PLAN	ANALYSI	S TABLE				
			Loca	ation:	all	Per	cent Ful	1:	1
Dialed	Total	Call	Dialed	Total	Call	Dialed	Total	Call	
String	Length	Type	String	Length	Type	String	Length	Type	
1	3	dac	773	5	aar				
2	4	ext	8	1	fac				
2	5	ext	9	1	fac				
3	5	ext	*	3	fac				
4 5	5	ext	#	3	fac				
5	5	ext							
52	5	aar							
54	5	aar							
55	5	aar							
57	5	aar							
59	5	aar							
6	4	ext							
6	5	ext							
7	5	ext							
74	4	aar							

3.4. 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
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
        Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code: *85
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code: *20
                       Answer Back Access Code: *35
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 8
   Auto Route Selection (ARS) - Access Code 1: 9
                                                      Access Code 2:
                 Automatic Callback Activation: *21
                                                       Deactivation: *22
Call Forwarding Activation Busy/DA: *24
                                          All: *25
                                                       Deactivation: #25
   Call Forwarding Enhanced Status:
                                           Act:
                                                       Deactivation:
                         Call Park Access Code: *33
                       Call Pickup Access Code: *34
CAS Remote Hold/Answer Hold-Unhold Access Code:
                  CDR Account Code Access Code:
                        Change COR Access Code:
                   Change Coverage Access Code:
                   Contact Closure Open Code:
                                                         Close Code:
```

```
change feature-access-codes
                                                                       2 of
                                                                Page
                               FEATURE ACCESS CODE (FAC)
                   Contact Closure Pulse Code:
                  Data Origination Access Code:
                     Data Privacy Access Code:
              Directed Call Pickup Access Code: *42
      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
                                                       Deactivation:
       Extended Group Call Pickup Access Code:
               Facility Test Calls Access Code:
                             Flash Access Code:
                                                       Deactivation:
             Group Control Restrict Activation:
                    Hunt Group Busy Activation:
                                                       Deactivation:
                              ISDN Access Code:
               Last Number Dialed Access Code: *40
    Leave Word Calling Message Retrieval Lock:
  Leave Word Calling Message Retrieval Unlock:
```

change feature-access-codes 3 of Page FEATURE ACCESS CODE (FAC) Leave Word Calling Send A Message: *10 Leave Word Calling Cancel A Message: #10 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: *26 Per Call CPN Unblocking Code Access Code: #26 Priority Calling Access Code: 119 Program Access Code: Refresh Terminal Parameters Access Code: Remote Send All Calls Activation: Deactivation: Self Station Display Activation: Send All Calls Activation: 124 Deactivation: 125 Station Firmware Download Access Code:

change feature-access-codes Page 4 of 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: 167 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: *41

3.5. 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
                                                                       1 of
                                                                Page
    EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME
                    Set Name: SIP Endpoint FNEs
    Active Appearance Select: 25900
         Automatic Call Back: 25901
  Automatic Call-Back Cancel: 25902
            Call Forward All: 25903
 Call Forward Busy/No Answer: 25904
         Call Forward Cancel: 25905
                   Call Park: 25906
       Call Park Answer Back: 25907
               Call Pick-Up: 25908
        Calling Number Block: 25909
      Calling Number Unblock: 25910
        Conference on Answer: 25911
       Directed Call Pick-Up: 25912
       Drop Last Added Party: 25913
   Exclusion (Toggle On/Off): 25914
  Extended Group Call Pickup:
      Held Appearance Select: 25916
```

```
change off-pbx-telephone feature-name-extensions set 1
                                                                       2 of
                                                                              2
                                                                Page
    EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME
       Idle Appearance Select: 25917
          Last Number Dialed: 25918
         Malicious Call Trace:
 Malicious Call Trace Cancel:
         Off-Pbx Call Enable:
         Off-Pbx Call Disable:
               Priority Call: 25923
              Send All Calls: 25924
        Send All Calls Cancel: 25925
         Transfer On Hang-Up: 25926
       Transfer to Voice Mail: 25927
      Whisper Page Activation: 25928
```

3.6. 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.

3.7. 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

COR Number: 1
COR Description:

FRL: 7
Can Be Service Observed? y
Calling Party Restriction: none
Can Be A Service Observer? y
Partitioned Group Number: 1
Priority Queuing? n
Restriction Override: all
Restricted Call List? n
Access to MCT? y
Group II Category For MFC: 7
Send ANI for MFE? n
MF ANI Prefix:
Hear System Music on Hold? y

Can Be Picked Up By Directed Call Pickup? y
Group Controlled Restriction: inactive
```

3.8. Add Coverage Path

Configure the coverage path to one used for the voice messaging hunt group, which is group "h90" 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 33
                                                        Page 1 of
                              COVERAGE PATH
                Coverage Path Number: 33
    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
                        n
          Active?
                                       n
                            y
y
n
            Busy?
                                           У
      Don't Answer?
                                                    Number of Rings: 2
                                           У
             All?
                                           n
DND/SAC/Goto Cover?
                                           У
  Holiday Coverage?
COVERAGE POINTS
  Terminate to Coverage Pts. with Bridged Appearances? n
 Point1: h90 Rng: Point2: Point3: Point4:
                            Point4:
```

3.9. 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 defaults for the other fields on Page 1.

```
add station 25650
                                                               Page 1 of
                                      STATION
                                     Security Code:
Coverage Path 1: 33
Coverage Path 2:
Hunt-to State
Extension: 25650
                                                                        BCC: 0
                                                                         TN: 1
     Type: 9630
     Port: IP
                                                                        COR: 1
    Name: iturret 25650
                                                                         cos: 1
STATION OPTIONS
                                            Time of Day Lock Table:
              Loss Group: 19 Personalized Ringing Pattern: 1
       Speakerphone: 2-way
Display Language: english
able GK Node Name:
                                               Message Lamp Ext: 25650
                                             Mute Button Enabled? y
                                                   Button Modules: 0
 Survivable GK Node Name:
     Survivable COR: internal Media Complex Ext:
                                                      IP SoftPhone? n
   Survivable Trunk Dest? y
                                               Customizable Labels? v
```

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 25650
                                                                 2 of
                                                           Page
                                   STATION
FEATURE OPTIONS
                              Auto Select Any Idle Appearance? n
         LWC Reception: spe
        LWC Activation? y
                                               Coverage Msg Retrieval? y
 LWC Log External Calls? n
                                                            Auto Answer: none
           CDR Privacy? n
                                                       Data Restriction? n
                                          Idle Appearance Preference? n
Bridged Idle Line Preference? n
 Redirect Notification? v
Per Button Ring Control? n
  Bridged Call Alerting? y
                                               Restrict Last Appearance? y
 Active Station Ringing: single
                                                       EMU Login Allowed? n
                                  Per Station CPN - Send Calling Number?
       H.320 Conversion? n
      Service Link Mode: as-needed
        Multimedia Mode: enhanced
   MWI Served User Type: qsig-mwi
                                              Display Client Redirection? n
                                             Select Last Used Appearance? n
                                              Coverage After Forwarding? s
                                             Direct IP-IP Audio Connections? y
 Emergency Location Ext: 25650 Always Use? n IP Audio Hairpinning? n
```

On Page 4 under the heading **BUTTON ASSIGNMENTS**, 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 25650
                                                            Page 4 of
                                    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
                                        5: brdg-appr B:1 E:25661
1: call-appr
                                        6: brdg-appr B:1 E:25662
2: call-appr
3: call-appr
                                        7: auto-cback
 4: call-appr
                                        8: no-hld-cnf
   voice-mail Number:
```

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,	auto-cback
Automatic Callback Cancel	
Conference on Answer	no-hld-cnf

In the sample configuration, four call appearances were administered for extension 25650. 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 25661		Page	1 of 6	
		STATION		
Extension: 25661 Type: 9630 Port: IP Name: HS1 of 25650		Lock Messages? n Security Code: Coverage Path 1: Coverage Path 2: Hunt-to Station:	BCC: TN: COR: COS:	1 1
STATION OPTIONS				
Loss Group:	19	Time of Day Lock Table: Personalized Ringing Pattern: Message Lamp Ext:		
Speakerphone:	2-way	Mute Button Enabled?	У	
Display Language:	english	Button Modules:	0	
Survivable GK Node Name:				
Survivable COR:	internal	Media Complex Ext:		
Survivable Trunk Dest?	У	IP SoftPhone?	n	
		Customizable Labels?	У	

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

```
add station 25661
                                                            Page
                                                                   2 of
                                    STATION
FEATURE OPTIONS
          TIONS
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
Per Button Ring Control? n
                                              Idle Appearance Preference? n
                                           Bridged Idle Line Preference? n
  Bridged Call Alerting? y
                                                 Restrict Last Appearance? y
 Active Station Ringing: single
                                                        EMU Login Allowed? n
                                  Per Station CPN - Send Calling Number?
       H.320 Conversion? n
      Service Link Mode: as-needed
        Multimedia Mode: enhanced
   MWI Served User Type:
                                               Display Client Redirection? n
             AUDIX Name:
                                              Select Last Used Appearance? n
                                                Coverage After Forwarding? s
                                              Direct IP-IP Audio Connections? y
 Emergency Location Ext: 25661 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 appearances for each call appearance of the first station (default appearance) configured below.

```
add station 25661
                                                            Page
                                                                   4 of
                                    STATION
SITE DATA
                                                       Headset? n
      Room:
                                                       Speaker? n
      Jack:
     Cable:
                                                      Mounting: d
     Floor:
                                                   Cord Length: 0
  Building:
                                                     Set Color:
ABBREVIATED DIALING
                              List2:
    List1:
                                                        List3:
BUTTON ASSIGNMENTS
1: call-appr
                                        5: brdg-appr B:3 E:25650
                                        6: brdg-appr B:1 E:25662
2: exclusion
3: brdg-appr B:1 E:25650
                                        7:
 4: brdg-appr B:2 E:25650
                                        8:
   voice-mail Number:
```

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

```
add station 25662
                                                              Page 1 of 6
                                      STATION
                                         Lock Messages? n
Security Code:
Coverage Path 1:
Extension: 25662
                                                                        BCC: 0
    Type: 9630
                                                                          TN: 1
     Port: S00342
                                                                        COR: 1
                                       Coverage Path 1:
    Name: HS2 of 25650
                                       Coverage Path 2:
                                                                        cos: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19 Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 25662
       Speakerphone: 2-way
Display Language: english
able GK Node Name:
                                             Mute Button Enabled? y
                                                   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 25662		Page 2 of 6	
		STATION	
FEATURE OPTIONS			
LWC Reception:	spe	Auto Select Any Idle Appearance?	n
LWC Activation?	У	Coverage Msg Retrieval?	У
LWC Log External Calls?	n	Auto Answer:	none
CDR Privacy?	n	Data Restriction?	n
Redirect Notification?	У	Idle Appearance Preference?	n
Per Button Ring Control?	n	Bridged Idle Line Preference?	n
Bridged Call Alerting?	У	Restrict Last Appearance?	У
Active Station Ringing:	single		_
	-	EMU Login Allowed?	n
H.320 Conversion?	n	Per Station CPN - Send Calling Number?	
Service Link Mode:	as-needed		
Multimedia Mode:	enhanced		
MWI Served User Type:		Display Client Redirection?	n
AUDIX Name:		Select Last Used Appearance?	n
		Coverage After Forwarding?	S
		Direct IP-IP Audio Connection	ıs? v
Emergency Location Ext:	25662	Always Use? n IP Audio Hairpinning?	-

```
add station 25662
                                                         Page 4 of 6
                                   STATION
SITE DATA
     Room:
                                                    Headset? n
      Jack:
                                                    Speaker? n
                                                   Mounting: d
     Cable:
     Floor:
                                                 Cord Length: 0
  Building:
                                                  Set Color:
ABBREVIATED DIALING
    List1:
                            List2:
                                                     List3:
BUTTON ASSIGNMENTS
1: call-appr
                                      5: brdg-appr B:3 E:25650
2: exclusion
                                      6: brdg-appr B:1 E:25661
3: brdg-appr B:1 E:25650
                                      7:
 4: brdg-appr B:2 E:25650
                                      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 Avaya Communication Manager extensions (25650, 25661, and 25662) 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 Avaya Communication Manager and Avaya SIP Enablement Services. The SIP trunk group is configured in **Section 3.10**. The **Configuration Set** value can reference a set that has the default settings.

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-pk	Page 2 c	of 2					
Station Extension 25650 25661 25662	Call Limit 4 2 2	Mapping Mode both both both	Calls Allowed all all all	Bridged Calls none none none	Location	1	

3.10. 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 Avaya SIP Enablement Services at the enterprise site. The host names will be used throughout the other configuration screens of Avaya Communication Manager.

```
change node-names ip

IP NODE NAMES

Name
IP Address
clan2
192.45.100.70
default
0.0.0.0
medpro2
192.45.100.71
SIPserver1
192.45.100.200
( 4 of 4 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 Avaya SIP Enablement Services. In this configuration, the domain name is *example.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 Avaya 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.

```
Page 1 of 19
change ip-network-region 1
                                        IP NETWORK REGION
  Region: 1
Location: 1
                     Authoritative Domain: example.com
    Name: Avaya region
MEDIA PARAMETERS
                                        Intra-region IP-IP Direct Audio: yes
       Codec Set: 1
                                       Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048
                                                       IP Audio Hairpinning? n
   UDP Port Max: 65531
DIFFSERV/TOS PARAMETERS
                                                     RTCP Reporting Enabled? y
 Call Control PHB Value: 34 RTCP Reporting Enabled
                                        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.729B calls between an *i* turret and an Avaya SIP telephone are not shuffled. However, G.729B calls between two *i* turret deskstations and an *i* turret and Avaya H.323 telephone are shuffled.

```
Change ip-codec-set 1

IP Codec Set

Codec Set: 1

Audio Silence Frames Packet
Codec Suppression Per Pkt Size(ms)

1: G.711MU n 2 20

2:
3:
4:
5:
6:
7:
```

Prior to configuring a SIP trunk group for communication with Avaya 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 Avaya 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 Avaya SIP Enablement Services in the **Far-end Domain** field. In this configuration, the domain name is *example.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*. Avaya Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

```
Page 1 of 1
add signaling-group 702
                               SIGNALING GROUP
Group Number: 702 Group Transport Method: tls
  Near-end Node Name: clan2
                                           Far-end Node Name: SIPserver1
Near-end Listen Port: 5061
                                         Far-end Listen Port: 5061
                                      Far-end Network Region: 1
      Far-end Domain: example.com
                                            Bypass If IP Threshold Exceeded? n
        DTMF over IP: rtp-payload
                                            Direct IP-IP Audio Connections? y
                                                     IP Audio Hairpinning? y
        Enable Layer 3 Test? n
Session Establishment Timer(min): 3
                                                Alternate Route Timer(sec): 6
```

Configure the **Trunk Group** form as shown below. This trunk group is used for calls to the *i* turret desk stations. 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 702

TRUNK GROUP

Group Number: 702

Group Type: sip

Group Name: To SIPserver1

Direction: two-way

Dial Access? n

Queue Length: 0

Service Type: tie

Page 1 of 21

TRUNK GROUP

COR: 1 TN: 1 TAC: 176

Night Service:

Night Service:

Signaling Group: 702

Number of Members: 20
```

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 702
TRUNK FEATURES

ACA Assignment? n

Measured: none

Maintenance Tests? y

Numbering Format: public

UUI Treatment: service-provider

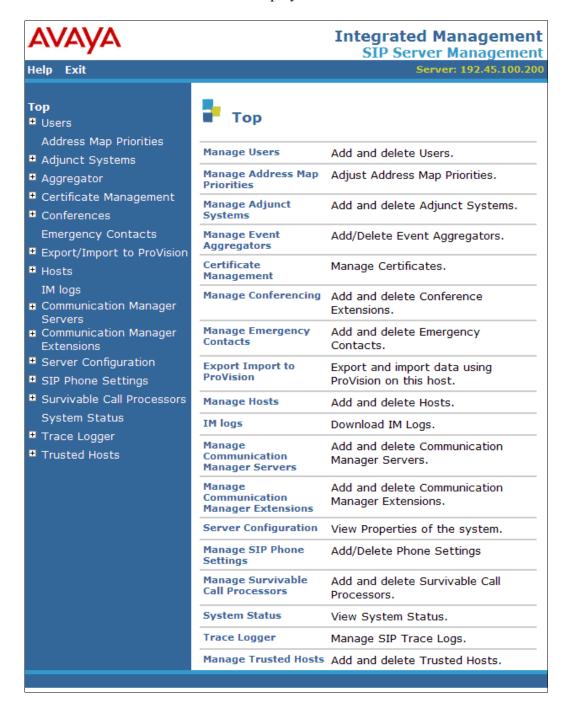
Replace Restricted Numbers? n
Replace Unavailable Numbers? n
Show ANSWERED BY on Display? y
```

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 '2' and whose calls are routed over SIP trunk group '702' have the number sent to the far-end for display purposes.

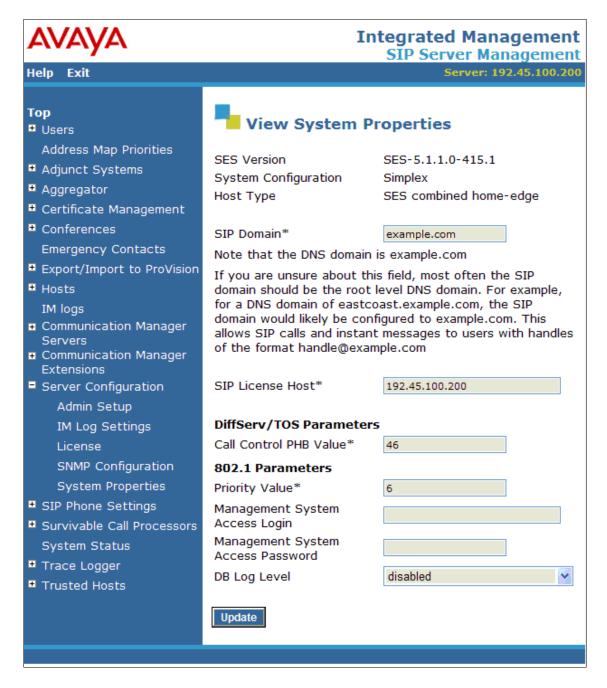
char	nge public-unk	nown-numbe	ring 0		Page	1 of	2
NUMBERING - PUBLIC/UNKNOWN				KNOWN FOR	MAT		
				Total			
Ext	Ext	Trk	CPN	CPN			
Len	Code	Grp(s)	Prefix	Len			
					Total Administered	: 1	
5	2	702		5	Maximum Entries	: 9999	9

4. Configure Avaya SIP Enablement Services

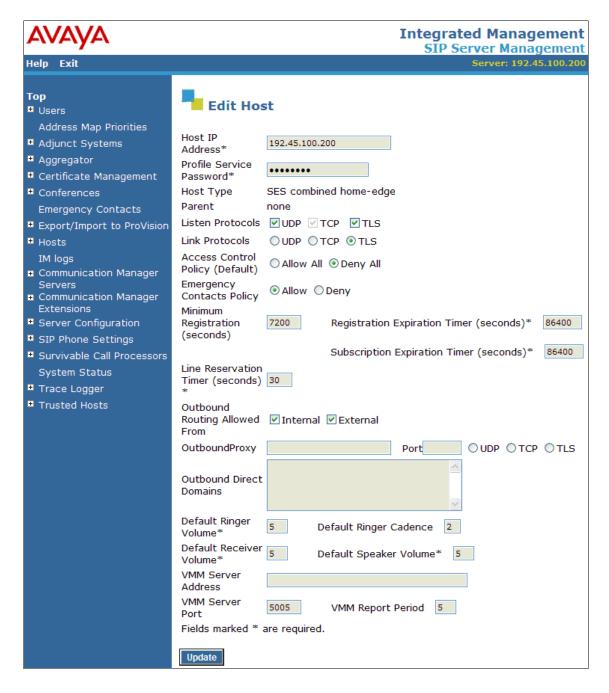
This section covers the administration of Avaya SIP Enablement Services (SES). Avaya SIP Enablement Services is configured via an Internet browser using the Administration web interface. To access the Administration web interface, enter <a href="http://<ip-addr>/admin as the URL in an Internet browser, where /admin is the IP address of Avaya SIP Enablement Services. Log in with the appropriate credentials and then select the Launch SES Administration Interface link in the next screen. The main screen is displayed as shown below.



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.



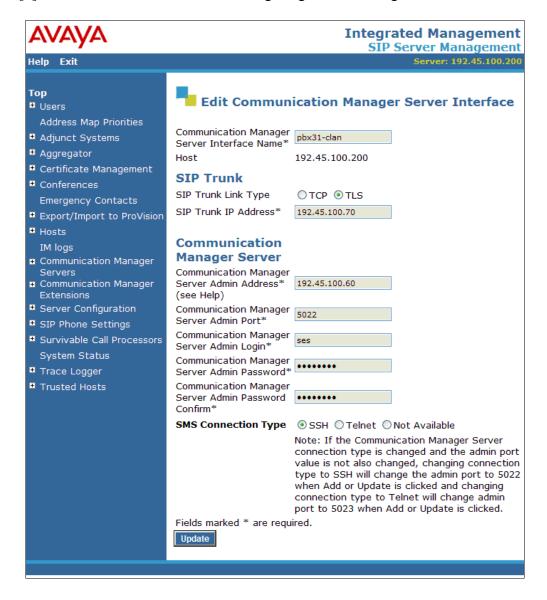
After setting up the domain in the **System Properties** screen, create a host entry for Avaya SIP Enablement Services. The following example shows the **Edit Host** screen since the host had already been configured. Enter the IP address of Avaya SIP Enablement Services in the **Host IP Address** field. 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.



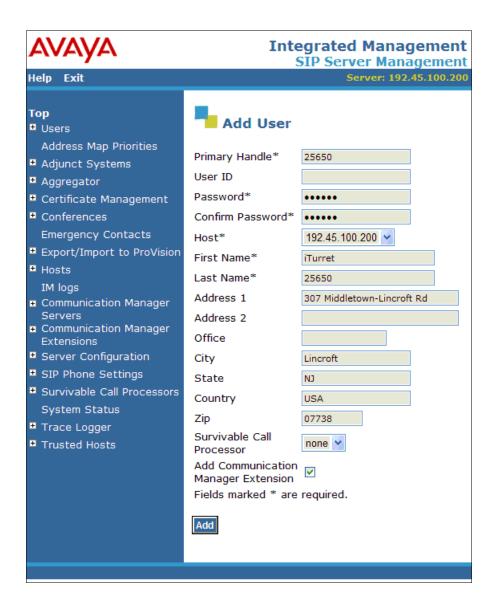
Under the **Communication Manager Servers** option in the Administration web interface, select **Add** to add the Avaya Media Server in the enterprise site since a SIP trunk is required between Avaya Communication Manager and Avaya SIP Enablement Services. The screen below shows the Edit Communication 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., pbx31-clan).
- Select the home server in the Host field.
- Select *TLS* (Transport Link Security) for the **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.



Add three users for each Speakerbus iD808 *i* turret registering with Avaya 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 Avaya Communication Manager. If fewer than 2 handsets are used, or Privacy is not enabled on the iD808, then it is not necessary to enable three users via Avaya 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 Avaya SIP Enablement Services server hosting the domain (*example.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.



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.



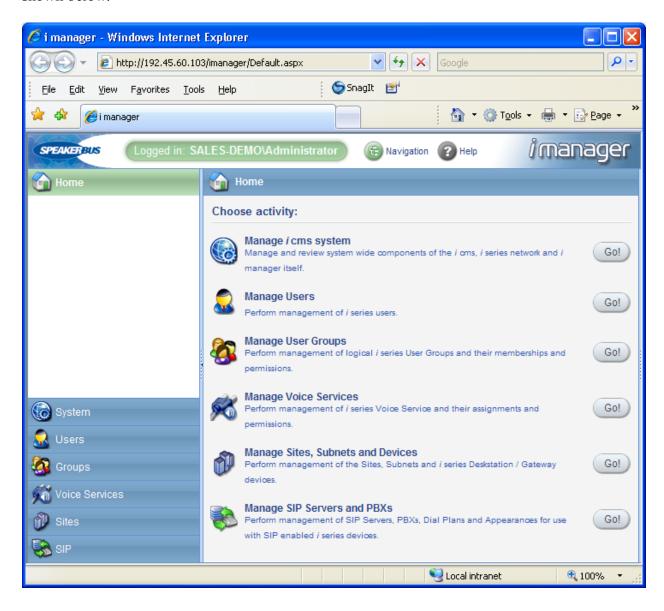
5. Speakerbus iD808 i turret Configuration

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
 - o Create PBX
 - o 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)
- Program iD808 Buttons
 - o Create Home Page
 - o Create FNEs Page
 - o Create Page Links Page
- Synchronize Deskstations

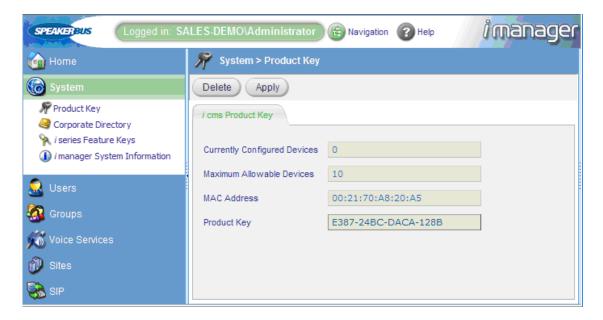
5.1. Launch i manager

To access the *i* manager software interface, open a web browser and type the *i* manager web address, for example, $\frac{\text{http:}//192.45.60.103/\text{imanager}}{\text{imanager}}$. Press the **Enter** key. At the *i* manager logon page enter the appropriate credentials. The *i* manager home page is displayed as shown below.



5.2. Verify Product Key

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

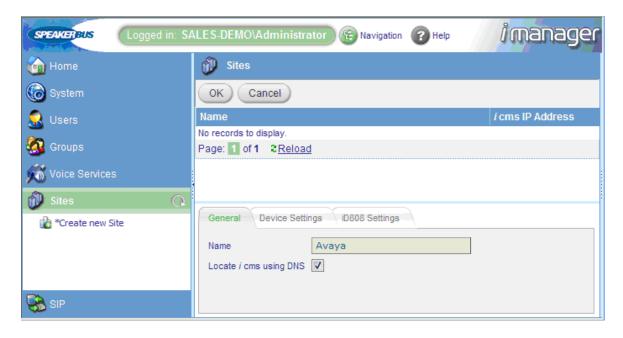


5.3. Create Site

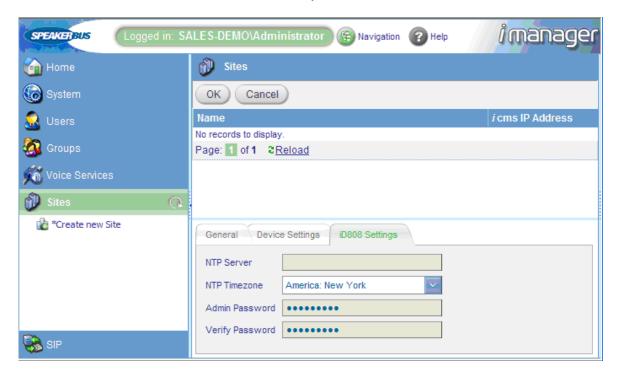
Configure a site representing the location where the Speakerbus iD808 devices are installed. Under **Sites** in the left pane, click on **Create new Site**. The **Sites** page is displayed.



In the **General** tab of the **Sites** page, set the **Name** field to a descriptive name and 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 [6] for correct configuration of DNS.



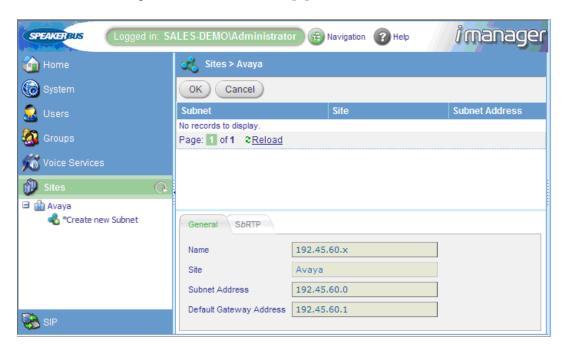
In the **iD808 Settings** tab, set the network time protocol timezone and configure the password for logging into the iD808 desk station. The **NTP Server** field may be set to the IP address of the NTP server if one is used. Click **OK**. The "Avaya" site will be now listed under **Sites**.



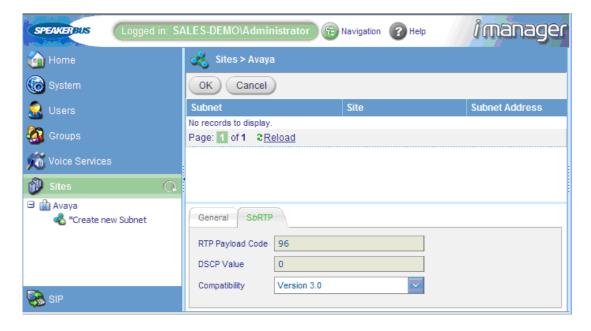
5.4. Create Subnet

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.

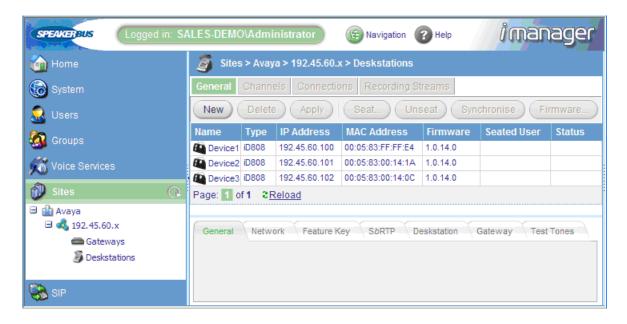


In the **SbRTP** tab, set the **Compatibility** field to "Version 3.0". Click **OK**.

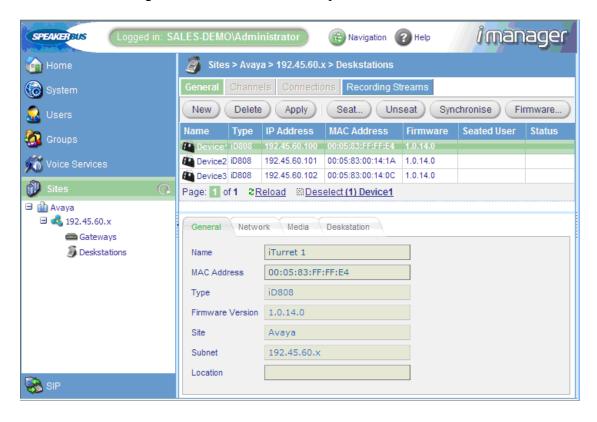


5.5. Create Deskstations

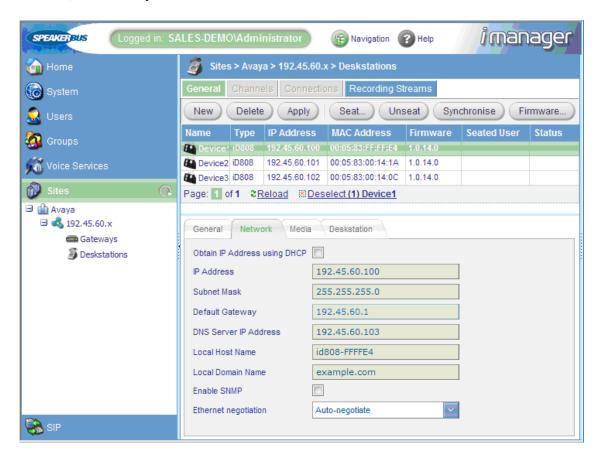
Once the site and subnet exist, the iD808 devices can be created. The network was set up using DNS without DHCP so IP configuration was manually performed on the iD808 devices. The deskstations then registered to *i* manager through DNS. After the iD808 devices are synchronized with *i* cms, they should 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.



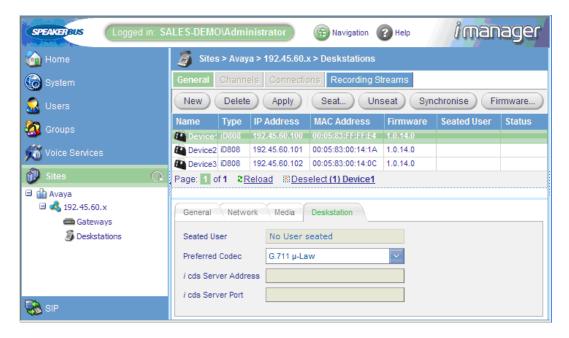
Select a device and change the name to a more descriptive one in the **General** tab.



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



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.



5.6. 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 "example.com", which in this configuration DNS resolves the domain name to 192.45.100.200, the Avaya SES 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 locater record (SRV) for registrar address and SIP domain on DNS. Refer to [5] for more details.



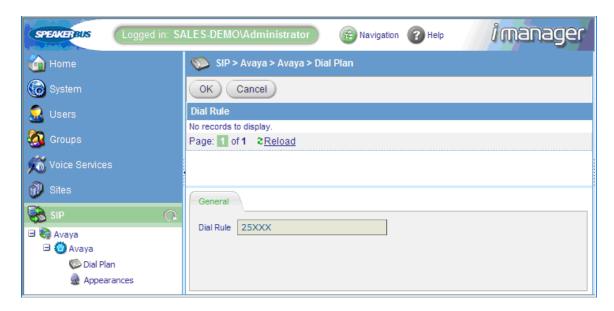
5.7. Create PBX

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**.



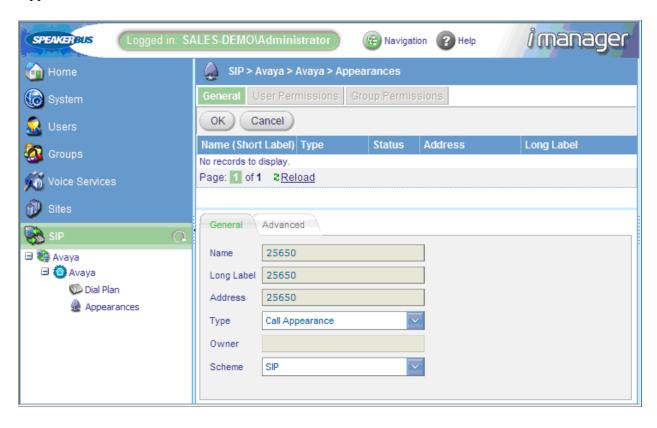
5.8. Create Dial Plan

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 "25" 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 "33". The example below corresponds to 5-digit extensions beginning with "25". 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.

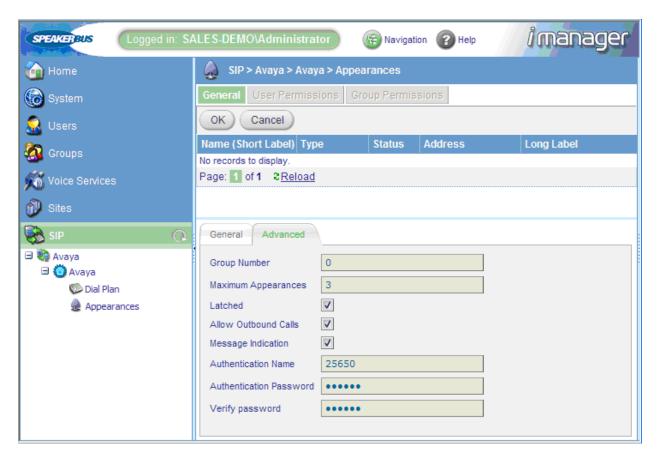


5.9. Create Call and Handset Appearances

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 Avaya Communication Manager and Avaya 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 desk station. In this example, the label was set to the main appearance extension 25650. The **Address** field should also be set to the appearance extension. Set the **Type** field to "Call Appearance".

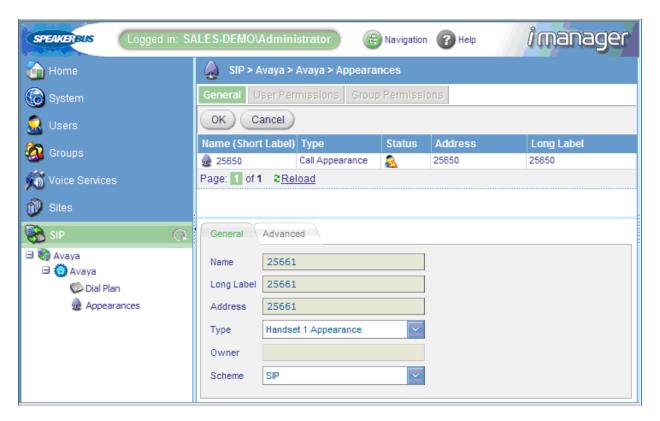


In the **Advanced** tab, set the **Maximum Appearances** field to "3". This field should be set to the number of call appearances configured on the station in Avaya Communication Manager minus one since the last call appearance is restricted. See the button assignment section of the station form and the second page of the Off-PBX-Telephone Station-Mapping form in Section 3.9 as an example. The number of call appearance button 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 Avaya 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 Avaya SIP Enablement Services. These are the credentials that the iD808 deskstation will use to authenticate and register with Avaya SIP Enablement Services. Use the default values for the other fields as shown below. Click **OK**.

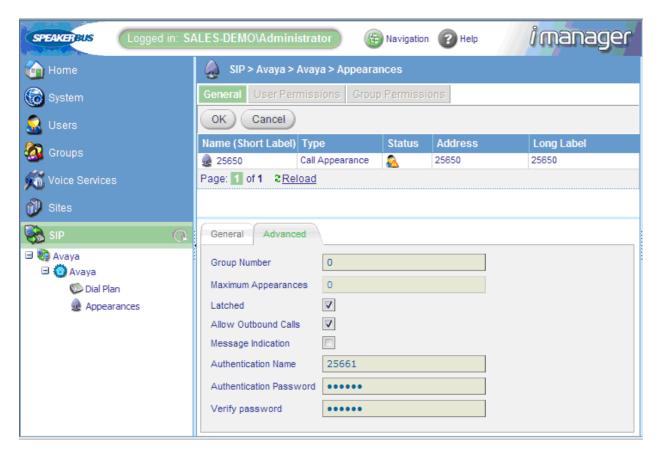


Next, this procedure will be repeated for the two handset appearances.

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 25661. Review the previous section for a description of these fields. Set the **Type** field to "Handset 1 Appearance".



In the **Advanced** tab, configure the **Authentication Name** and **Authentication Password** fields with the credentials for registering with Avaya 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 Avaya Communication Manager. Click **OK**.



Repeat the above procedure to add the handset 2 appearance.

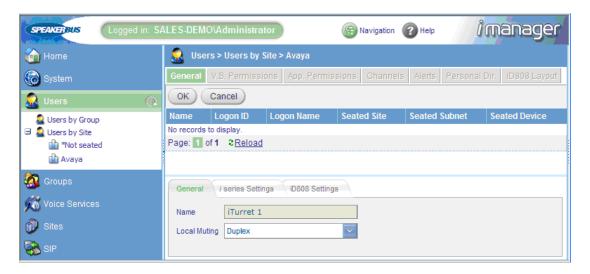
The three call appearances for the previously configured iD808 deskstation are listed below.



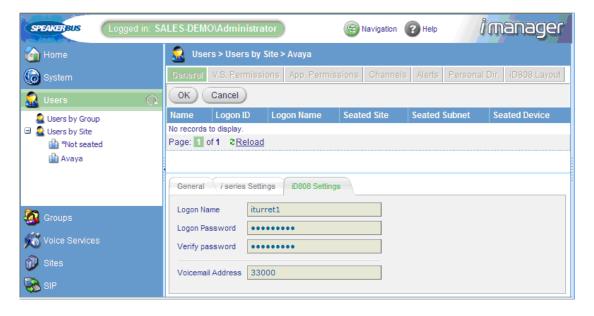
Repeat the above procedure for adding the main and handset appearances for each iD808 deskstation.

5.10. Create Users

In this section, the users are created. In the **Users** directory tree, click on the "Avaya" link, followed by the **New** button on the next page to add a new user. In the **General** tab, provide a descriptive name in the **Name** field.

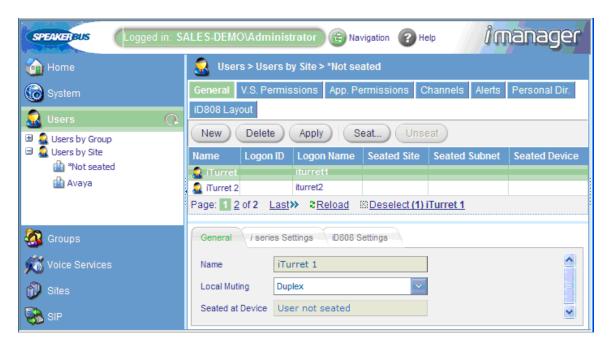


In the **iD808 Settings** tab, provide the logon credentials for the user to log into their iD808 deskstation. This page will be revisited later in **Section 5.13** to configure the default call appearance for this deskstation. Click **OK**.



Repeat the previous procedure to add more users.

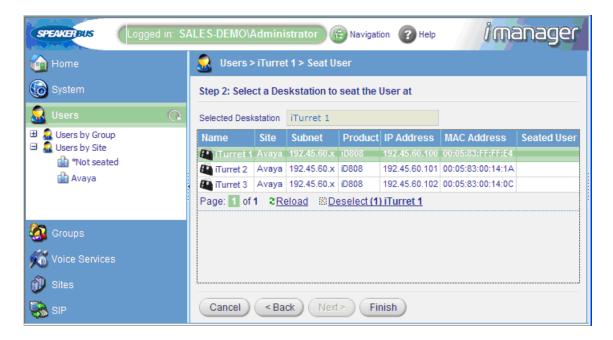
After a user has been created, the user needs to be "seated" on an iD808 deskstation. In the **Users** directory tree, click on the "Avaya" link to display the list of users. Select the user previously configured (i.e., *i* turret 1) and click on the **Seat...** button.



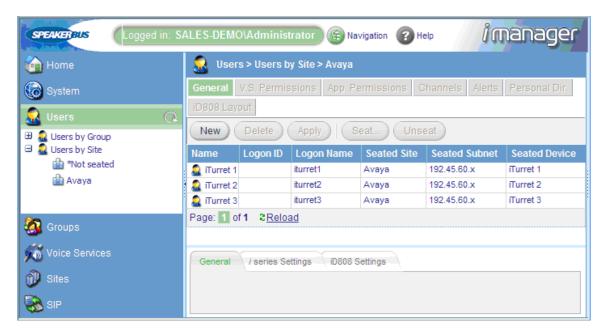
On the next page, filter options are presented. Filter deskstations in the "Avaya" site and in the "192.45.60.x" subnet as shown below. The user will be seated on an iD808 deskstation with these properties. Click **Next**.



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 "*i* turret 1" deskstation. Select "*i* turret 1" in the list and click **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.



5.11. 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.



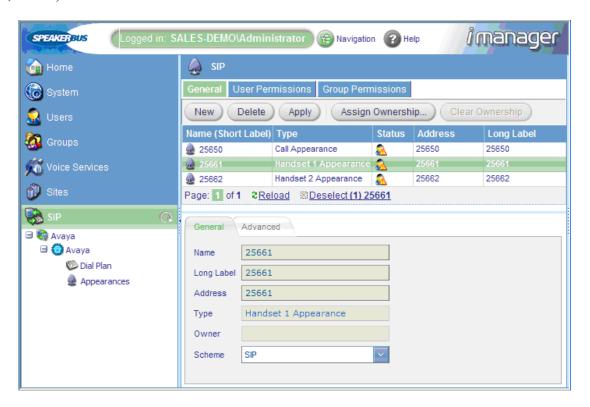
In the **Groups** directory tree, expand "Traders" and click on **Non-Member Users**. A list of non-member users is displayed. Select all the users to be added to the "Traders" group as shown below and then click **Add Membership**.



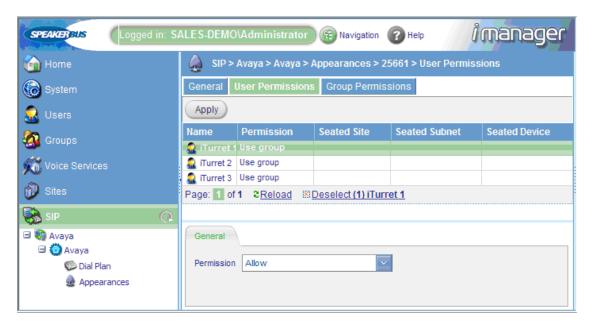
Next, select the main call appearance to assign permissions for it to the group as shown below. For example, 25650 is the main call appearance for *i* turret 1. Do not assign the handset appearances. Set the **Permission** field to "Allow" and click **Apply**. The **Permission** column for 25650 will be changed to "Allow". This procedure should be performed for all main call appearances. It is recommended that groups are not given permission to the call-appearances since call-appearances are analogous to an extension and is not shared by a group. Therefore, it is suggested that only the intended owner of the call-appearances and selected others who bridge this call would have their permissions set to "allow".



The next step will be to assign handset appearances permissions to users. In the **SIP** directory tree, click on **Appearances** under "Avaya". Select the "Handset 1 Appearance" for *i* turret 1 (i.e., 25661) and click on **User Permissions**.

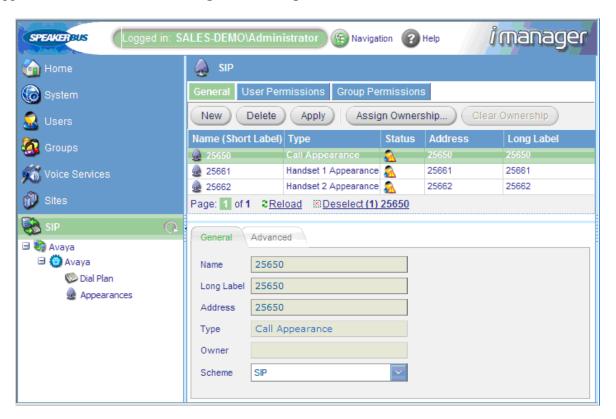


Select the user on the next page to which handset appearance permissions will be assigned. Set the **Permission** field to "Allow" as shown below. Click **Apply**. Assign "Handset 2 Appearance" permissions to this user by following the previous procedure.

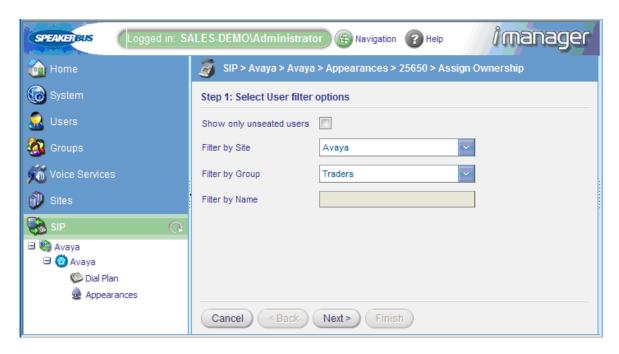


5.12. Assign Ownership

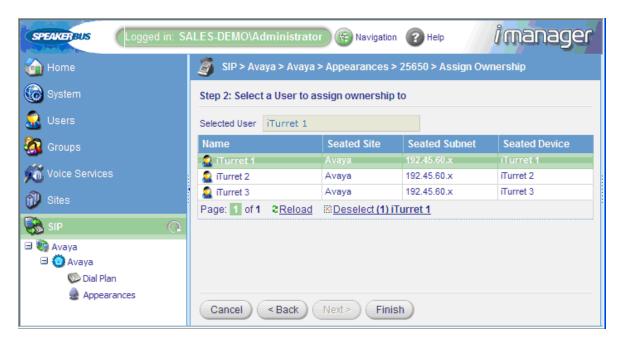
Assign ownership of the appearances to a user. In the **SIP** directory tree, click on **Appearances** to display the appearances list as shown below. In the **General** tab, select the main call appearance and click on the **Assign Ownership** button.



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



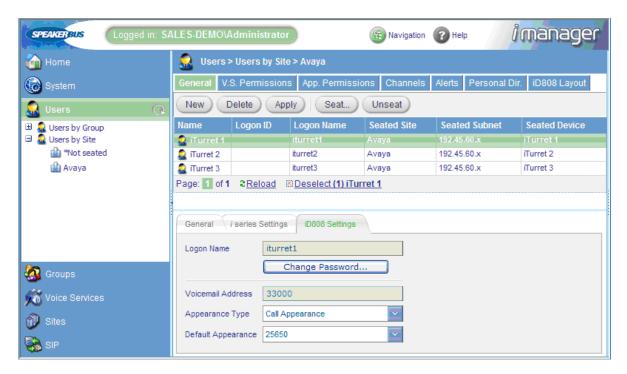
On the next page, select the user to which ownership will be assigned to the main call appearance. In this example, the main call appearance 25650 will be assigned to *i* turret 1. Click Finish.



Repeat this procedure to assign "Handset 1 Appearance" and "Handset 2 Appearance" to *i* turret 1.

5.13. Assign Default Call Appearance

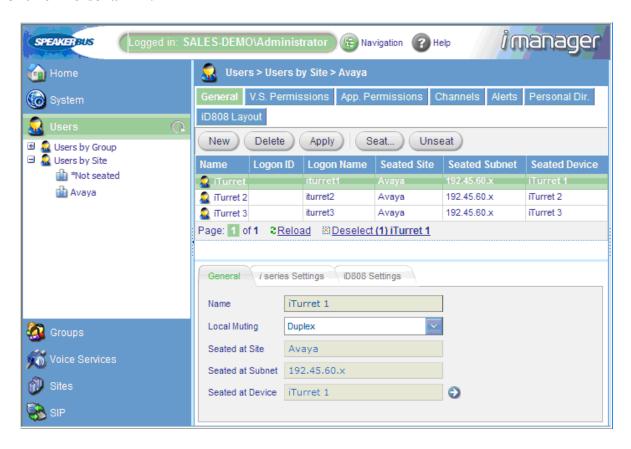
In the **Users** directory tree, click on the "Avaya" link to display the users list. Set the **Default Appearance** field to the main call appearance (e.g., 25650). Click **Apply**.



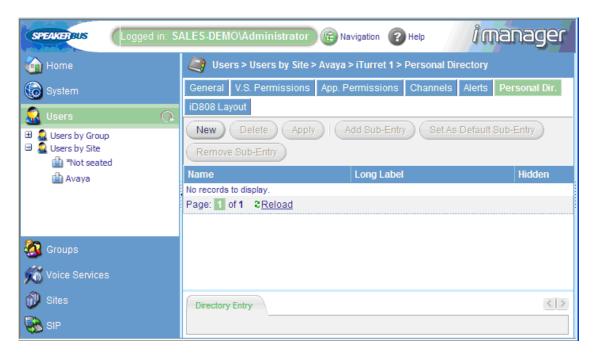
5.14. Program Feature Name Extensions (FNEs)

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.

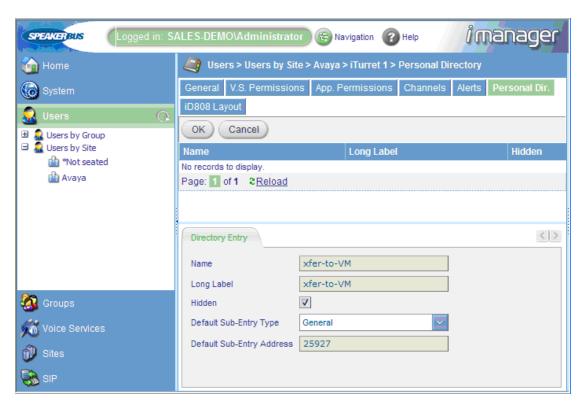
In the **Users** directory tree, click on the "Avaya" link and then select a user (e.g., *i* turret 1). Click on **Personal Dir.**



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



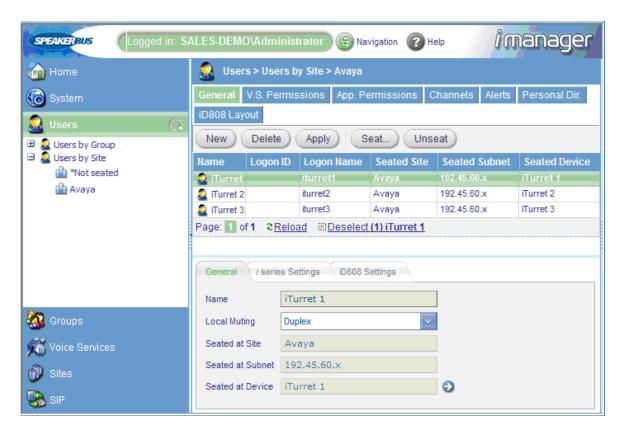
In the **Directory Entry** tab, provide the name of the FNE in the **Name** and **Long Label** fields and set the **Default Sub-Entry Address** to the FNE extension configured in **Section 3.5**. In this example, the transfer-to-voicemail FNE is added. Click **OK**. Repeat for other FNEs of interest.



5.15. Program iD808 Deskstation Keys

This section describes 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. In this configuration, each user will be 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.

In the **Users** directory tree, click on the "Avaya" link to display a list of users in the "Avaya" site. Select a user (e.g., *i* turret 1) and click **iD808 Layout** to display the iD808 key layout for this user.

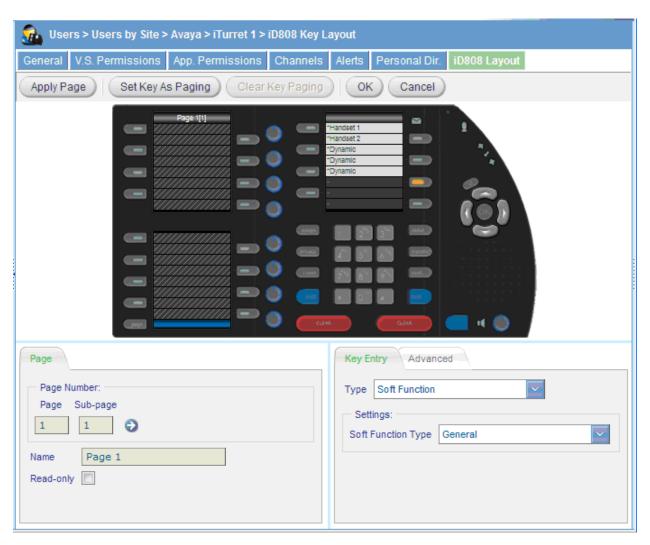


In the iD808 key layout, click on the key highlighted below under Handset 2. In the **Key Entry** tab, set the **Type** field to "Dynamic". Click **OK**.



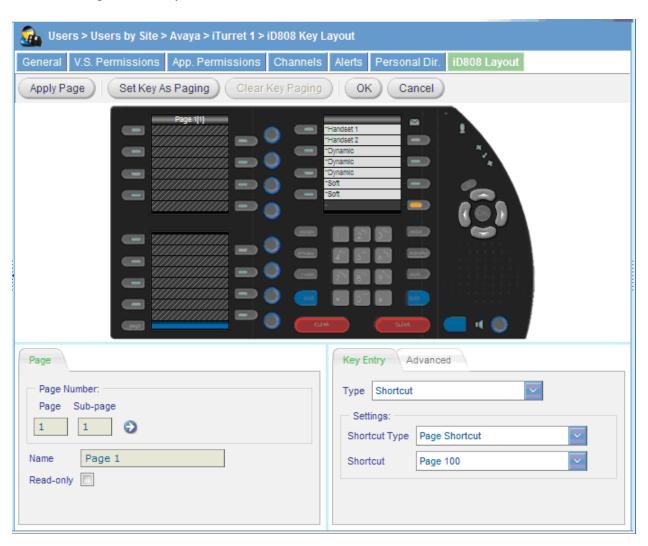
Three Dynamic keys will be added so repeat this step for the next two keys. The iD808 layout will appear as shown below once the three dynamic keys have been added.

Next, configure two Soft Function keys. Select the next available key under the last Dynamic key. In the **Key Entry** tab, set the Type field to "Soft Function" and click **OK**. Repeat this step for the second Soft Function key.

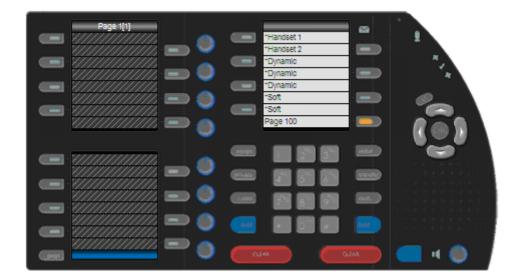


Once the two Soft Function keys have been created, the iD808 layout will be appear as shown below.

Finally, add a Shortcut key under the last Soft Function key. In the **Key Entry** tab, set the **Type** field to "Shortcut". Set the **Shortcut** field to "Page 100". This deskstation page will contain shortcuts to the "Home Page", which will contain the call appearances, and "FNEs" page, which will contain speed dial keys for FNEs. Click **OK**.

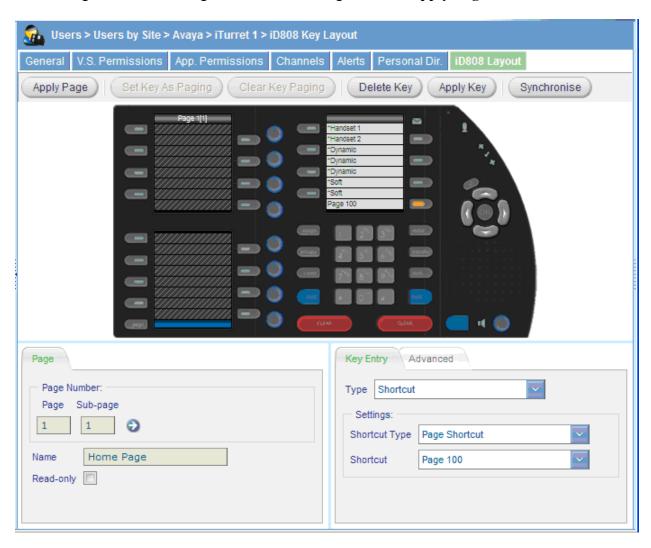


After all of the iD808 keys have been created on the deskstation, the iD808 layout will appear as shown below.



5.15.1. Create Home Page

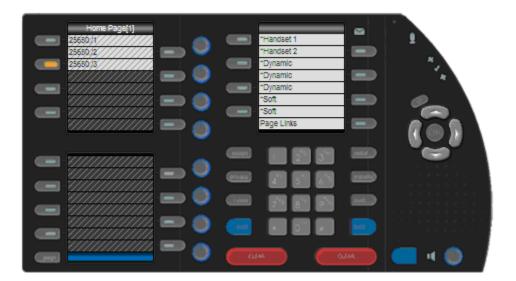
In the iD808 key layout page, go to Page 1 of the deskstation by setting the **Page** field to "1" in the **Page** tab and clicking the arrow key to the right. Set the **Name** field to "Home Page" to rename Page 1. Note that Page 1 is the "Home Page". Click **Apply Page**.



In the "Home Page", select the next available key as highlighted by the red box below. The next three keys on this page will be assigned to call appearances. In the **Key Entry** tab, set the **Type** field to "Appearance". In the **Settings** section, set the **Appearance Type** field to "Call Appearance" and the **Appearance** field to the main call appearance extension (e.g., 25650). Select the **Alert Enabled** checkbox so the deskstation rings when a call is received on this call appearance. The **Alert Profile** field is set to a particular ring type specified in "Profile 1". Click **OK**. Repeat this procedure to add the next two call appearances to the "Home Page".

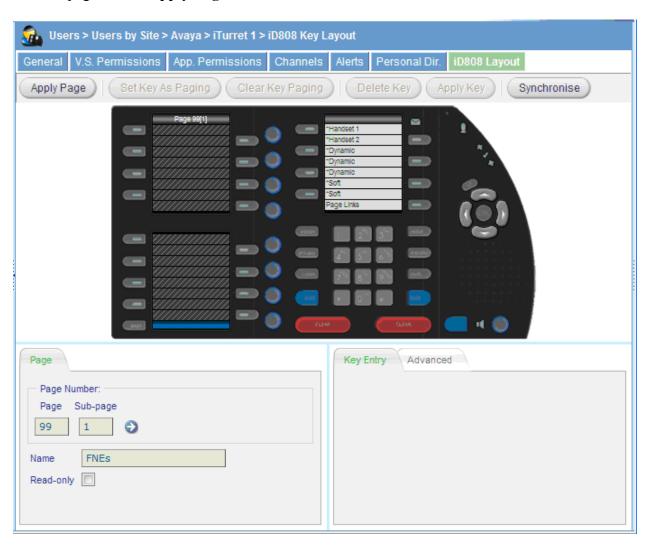


Once the three call appearances have been added to the "Home Page", the iD808 layout will appear as follows.

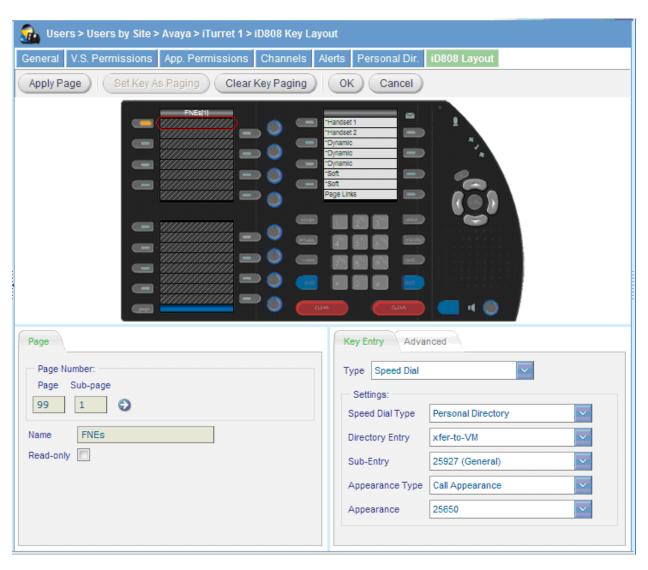


5.15.2. Create FNEs Page

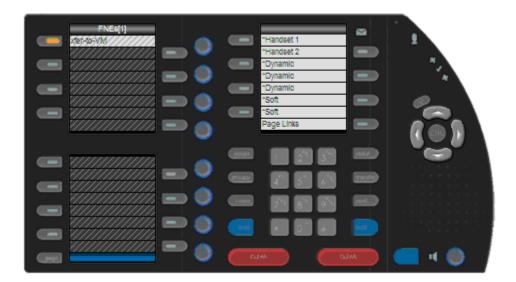
In the iD808 key layout page, go to Page 99 of the deskstation by setting the **Page** field to "99" in the **Page** tab and clicking the arrow key to the right. This page will contain the speed dial keys for FNEs. Set the **Name** field to "FNEs" to rename Page 99. Note that Page 99 is the "FNEs" page. Click **Apply Page**.



Select the next available key in the "FNEs" page. In the **Key Entry** tab, set the **Type** field to "Speed Dial" and the **Speed Dial Type** field to "Personal Directory". The **Directory Entry** field may be set to the transfer-to-voicemail FNE created in **Section 5.14**. The **Sub-Entry** field is set to the FNE extension. The **Appearance Type** field is set to "Call Appearance" and the **Appearance** field is set to the main call appearance for this deskstation (e.g., 25650). Click **OK**. Repeat this procedure to create speed dial buttons for other FNEs.

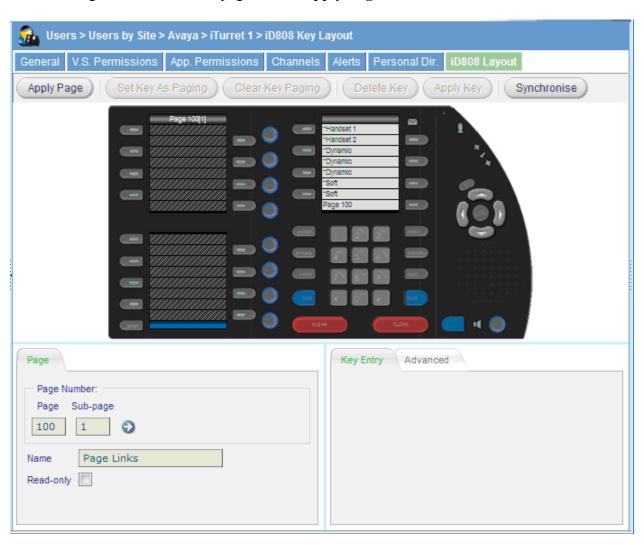


After the FNE is added to the "FNEs" page, the iD808 layout will appear as follows.

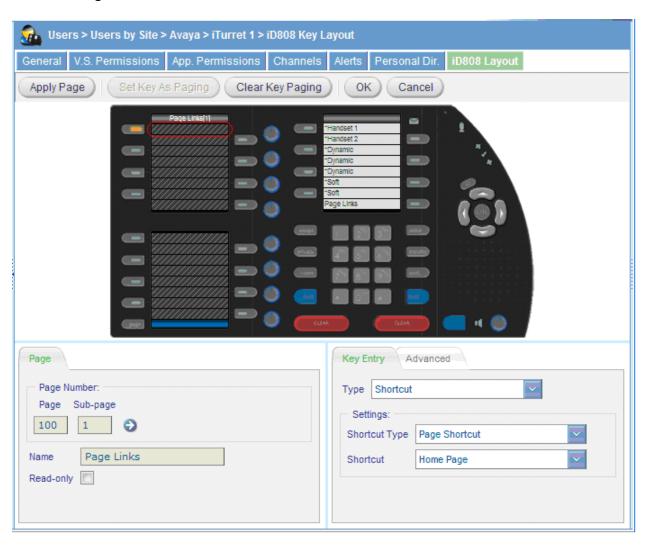


5.15.3. Create a "Page Links" Page

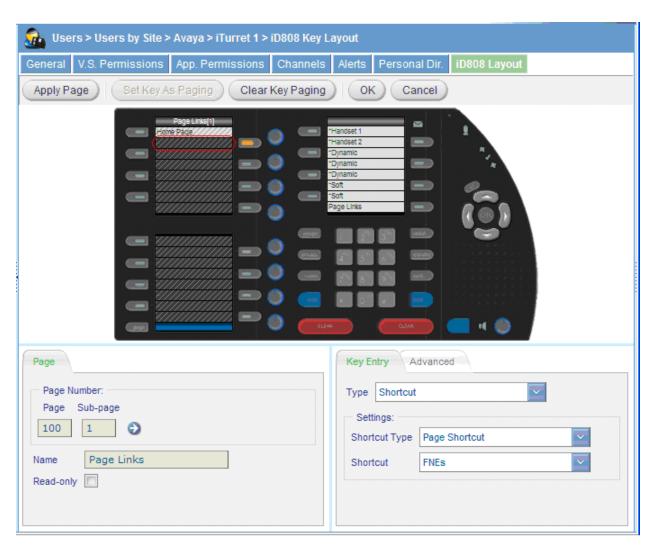
In the iD808 key layout page, go to Page 100 of the desk station by setting the **Page** field to "100" in the **Page** tab and clicking the arrow key to the right. This page will contain shortcuts to the "Home Page" and "FNEs" page. Set the **Name** field to "Page Links" to rename Page 100. Note that Page 99 *is* the "FNEs" page. Click **Apply Page**.



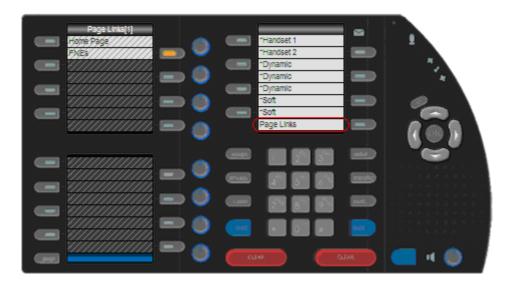
In "Page Links" page, select an available key to create a shortcut to the "Home Page" containing call appearances. In the **Key Entry** tab, set the **Type** field to "Shortcut" and the **Shortcut** field to "Home Page". Click **OK**.



In "Page Links" page, select the next available key to create a shortcut to the "FNEs" page containing speed dial keys for FNEs. In the **Key Entry** tab, set the **Type** field to "Shortcut" and set the **Shortcut** field to "FNEs". Click **OK**.



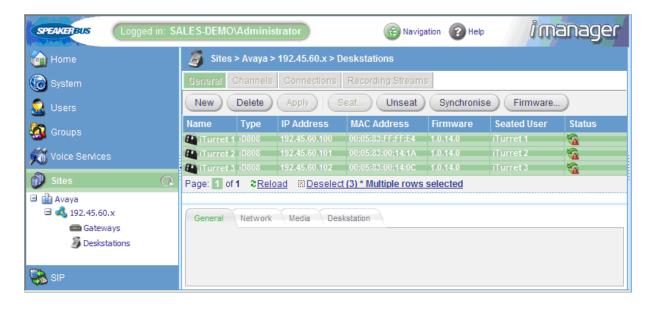
Once the "Home Page" and "FNEs" page shortcuts have been added, the "Page Links" page will appear as follows. The "Page Links" page is now accessible by clicking on the **Page links** shortcut key highlighted below.



5.16. Synchronize Deskstations

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

Note: Active calls will be dropped.



6. Interoperability Compliance Testing

This section describes the compliance testing used to verify the interoperability of Speakerbus iD808 *i* turret with Avaya SIP Enablement Services and Avaya Communication Manager. This section covers the general test approach and the test results.

6.1. General Test Approach

The general test approach was to make calls between iD808 deskstations and Avaya SIP, H.323, and digital stations using various codec settings and exercising common PBX features.

6.2. Test Results

Speakerbus iD808 *i* turret passed compliance testing. The following features and functionality were verified.

- Successful registration of the *i* turret with Avaya SIP Enablement Services.
- Calls between the *i* turret and Avaya SIP, H.323, and digital stations.
- G.711u and G.729B 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.

Observation(s): G.729B calls between an *i* turret and an Avaya SIP telephone are not shuffled. However, G.729B calls between two *i* turrets or between an *i* turret and an Avaya H.323 telephone are shuffled.

7. 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 [6] for more details.
- 2. Verify that the iD808 deskstations have successfully registered with Avaya SIP Enablement Services. Each iD808 deskstation contains three extensions on Avaya 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 Avaya 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.

8. 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

9. Conclusion

These Application Notes have described the administration steps required to use Speakerbus iD808 *i* turret with Avaya SIP Enablement Services and Avaya Communication Manager. 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] Administrator Guide for Avaya Communication Manager, January 2008, Issue 4, Document Number 03-300509.
- [2] Avaya Extension to Cellular Guide, January 2008, Issue 11, Document Number 210-100-700.
- [3] SIP Support in Avaya Communication Manager Running on the Avaya S8xxx Servers, January 2008, Issue 8, Document Number 555-245-206.
- [4] *Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services*, January 2008, Issue 5.0, Document Number 03-600768.
- [5] Speakerbus i manager Administrator's Guide, V1.1, Revision 3, July 2008.
- [6] Speakerbus iD808 User Guide, April 2008.
- [7] 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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