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 Aura™ SIP Enablement Services and Avaya S8730 Servers with a G650 Media Gateway running Avaya Aura™ Communication Manager. Also described is how Avaya Aura™ 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 Aura™ Communication Manager to the Speakerbus iD808 *i* turret[1], 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 Aura™ 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* turret and FNU
Call Forward Busy/No Answer	Yes	Yes	Local menu option on *i* turret and FNU
Call Forward Cancel	Yes	Yes	Local menu option on *i* turret and FNU
3-way conferencing – 3rd party added	Yes	Yes	
3-way conferencing – 3rd 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* turret and FNU
Last Number Dialed	Yes	Yes	Via OPS FNE
Priority Call	Yes	Yes	Via OPS FNE, *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 Aura™ SIP Enablement Services, a pair of Avaya S8730 Servers with a G650 Media Gateway running Avaya Aura™ 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 Aura™ SIP Enablement Services and are configured as OPS stations on Avaya Aura™ 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.

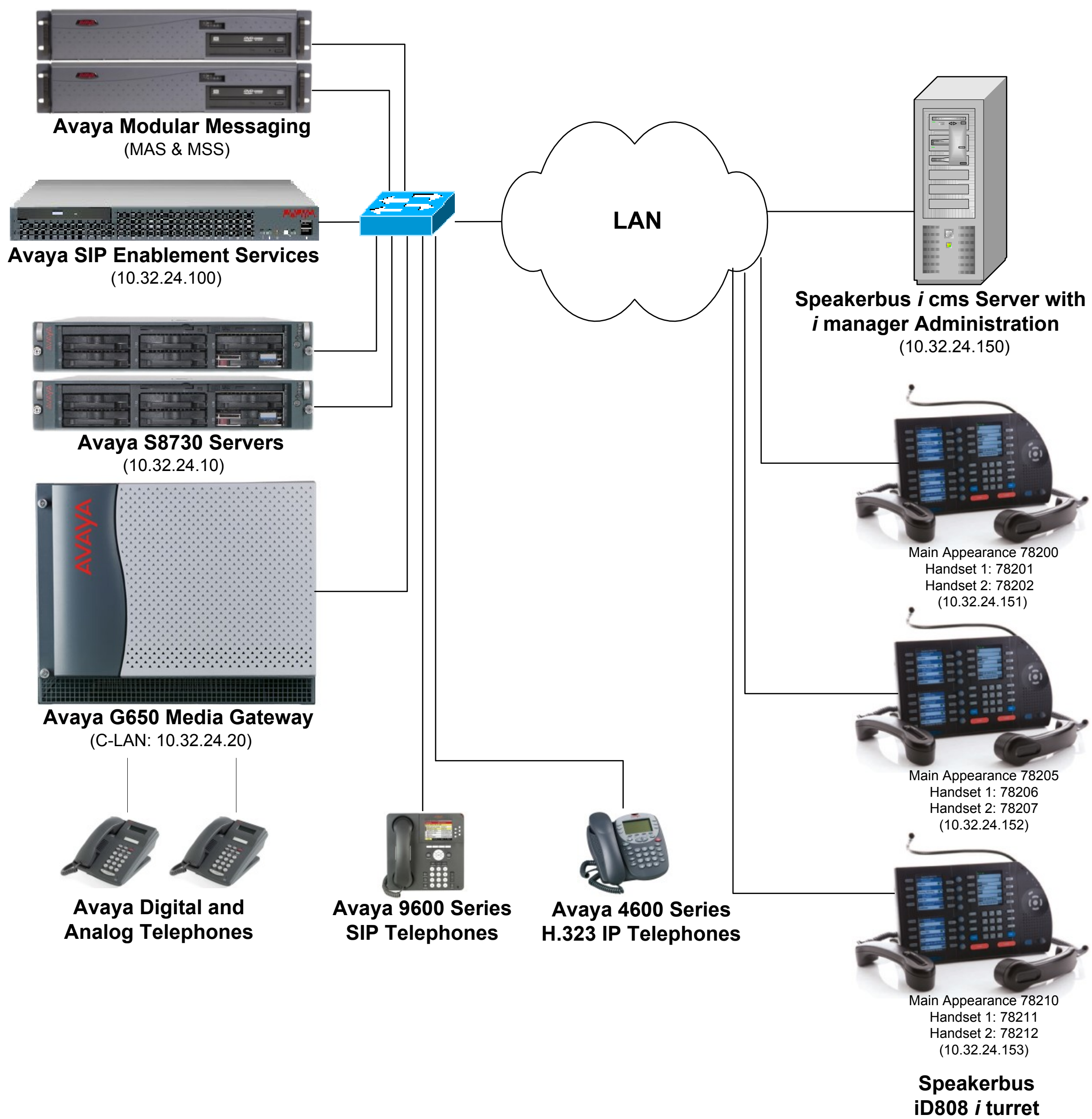


Figure 1: Speakerbus iD808 *i* turret with Avaya SIP Solution

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™ 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™ 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* turret	1.110.4.0
Speakerbus *i* cms Server with *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
```

```
change system-parameters features                              Page  18 of  18
                        FEATURE-RELATED SYSTEM PARAMETERS

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    Total  Call    Dialed   Total  Call     Dialed   Total  Call
       String    Length Type    String   Length Type     String   Length 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     Access Code 2:
                 Automatic Callback Activation: *25    Deactivation: #25
Call Forwarding Activation Busy/DA: *21    All: *20    Deactivation: #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  y  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
                                           Time of Day Lock Table:
              Loss Group: 19         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
                                           Time of Day Lock Table:
              Loss Group: 19         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 appearances[3] 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 configuration[4] 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
                                           Time of Day Lock Table:
              Loss Group: 19         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       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     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       Appl    Call        Mapping    Calls       Bridged       Location
 Extension     Name    Limit       Mode       Allowed     Calls
 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        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 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
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n

 Show ANSWERED BY on Display? y
```

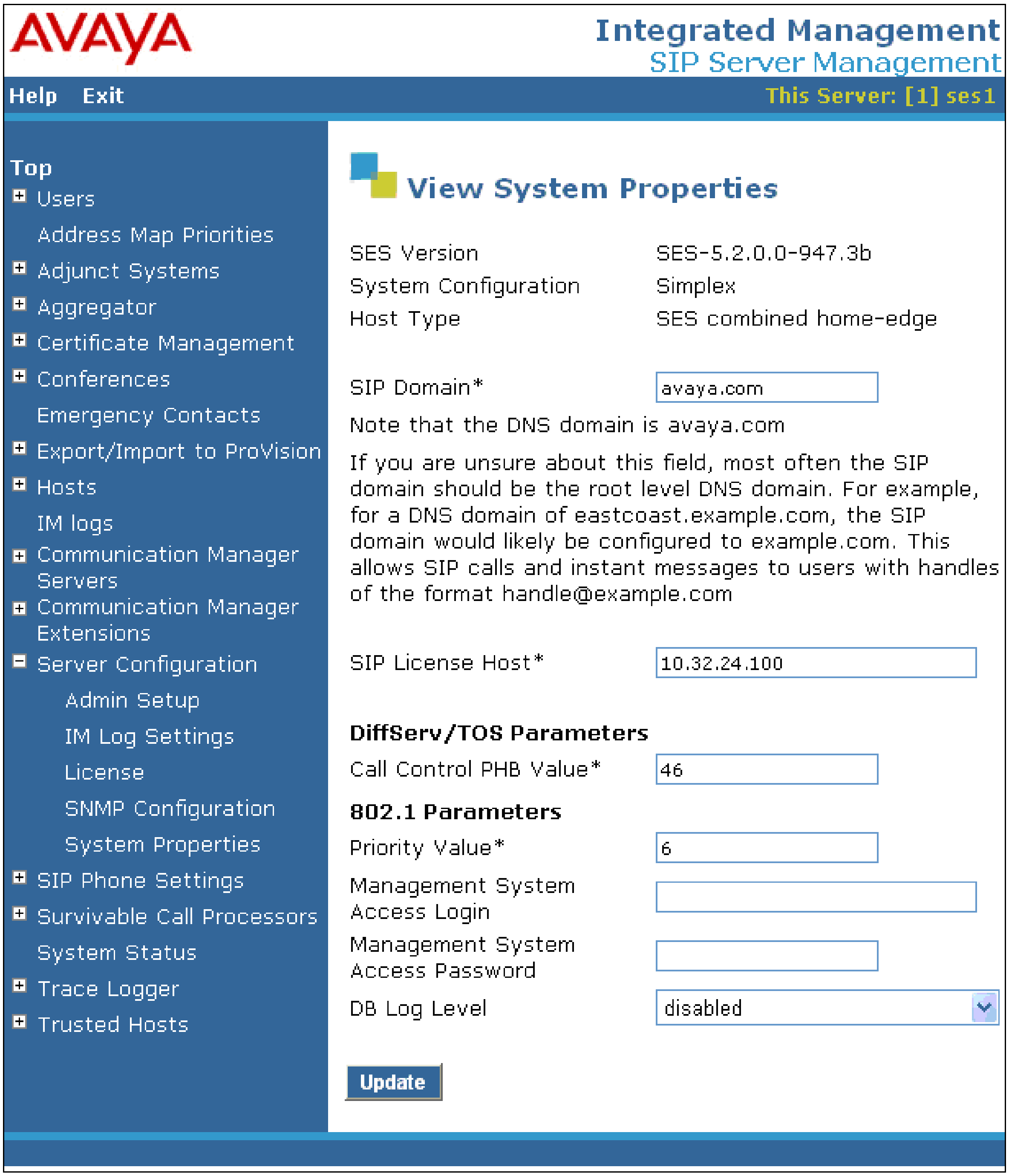
Configure the **Public/Unknown Numbering Format** form to send the calling party number to the far-end. Add an entry so that local stations with a 5-digit extension beginning with '7' and whose calls are routed over SIP trunk group '10' have the number sent to the far-end for display purposes.

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

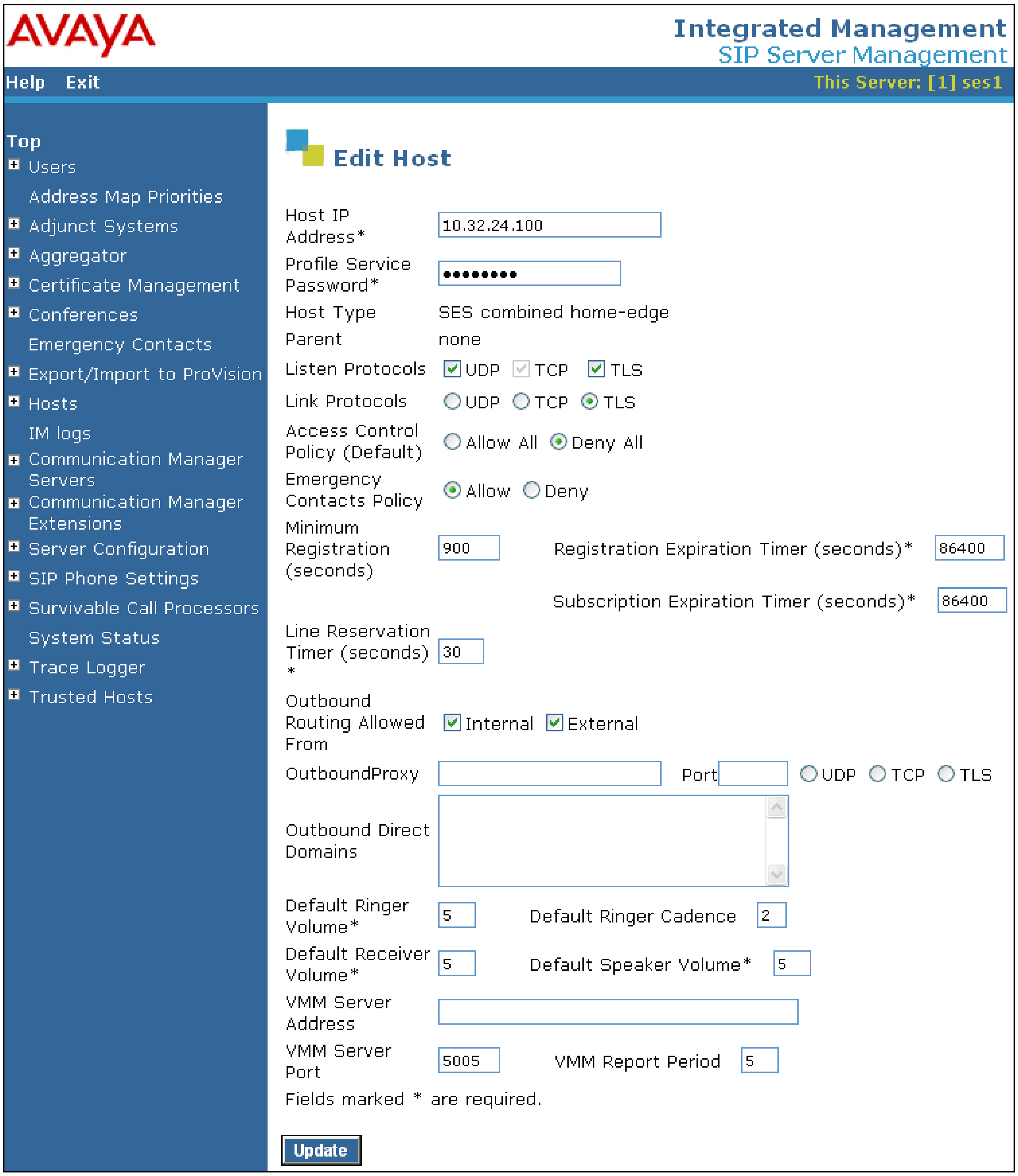
5. Configure Avaya Aura™ SIP Enablement Services

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.

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.



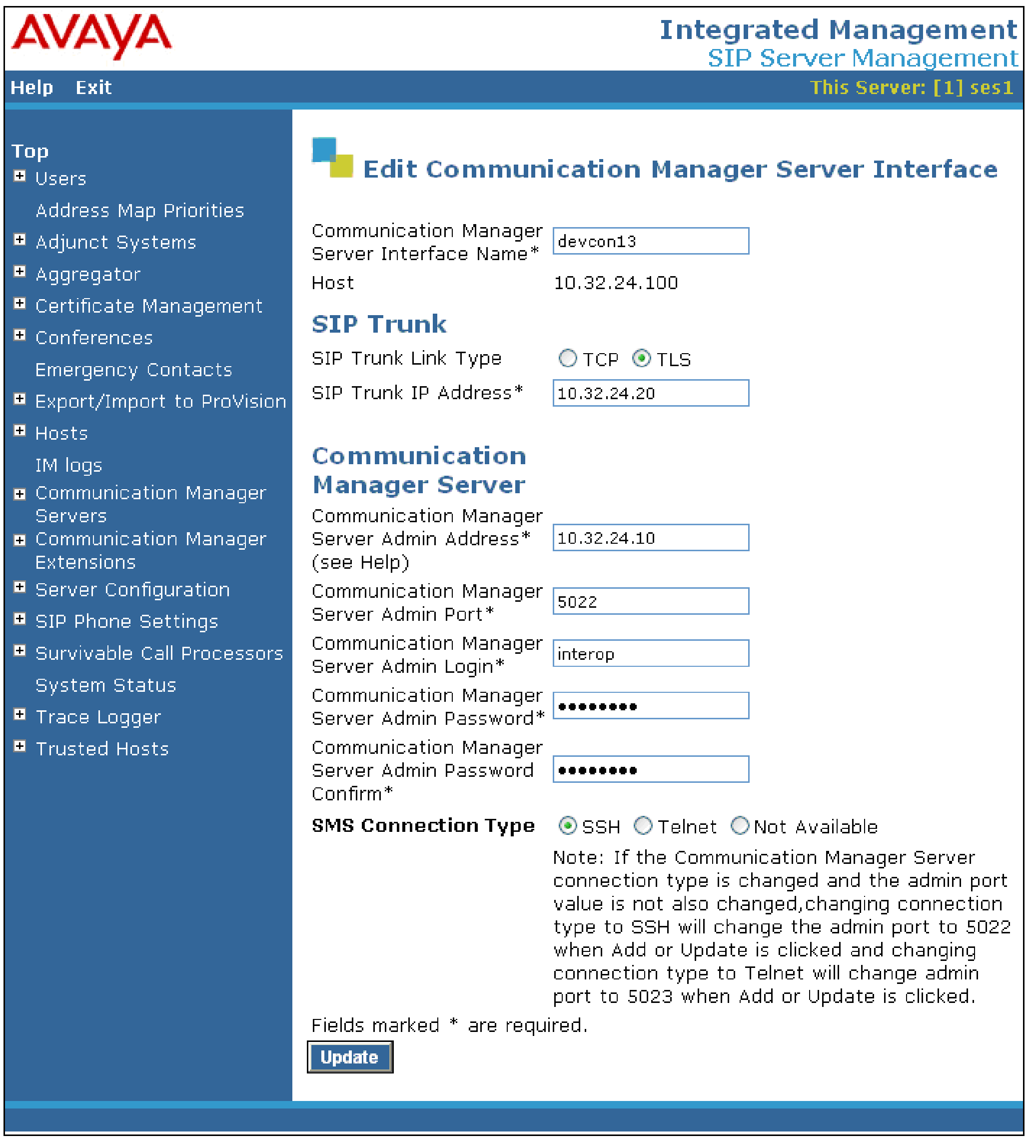
After setting up the domain in the **System Properties** screen, create a host entry for SIP Enablement Services. The following example shows the **Edit Host** screen since the host had already been configured. Enter the IP address of SIP Enablement Services in the **Host IP Address** field. The **Profile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



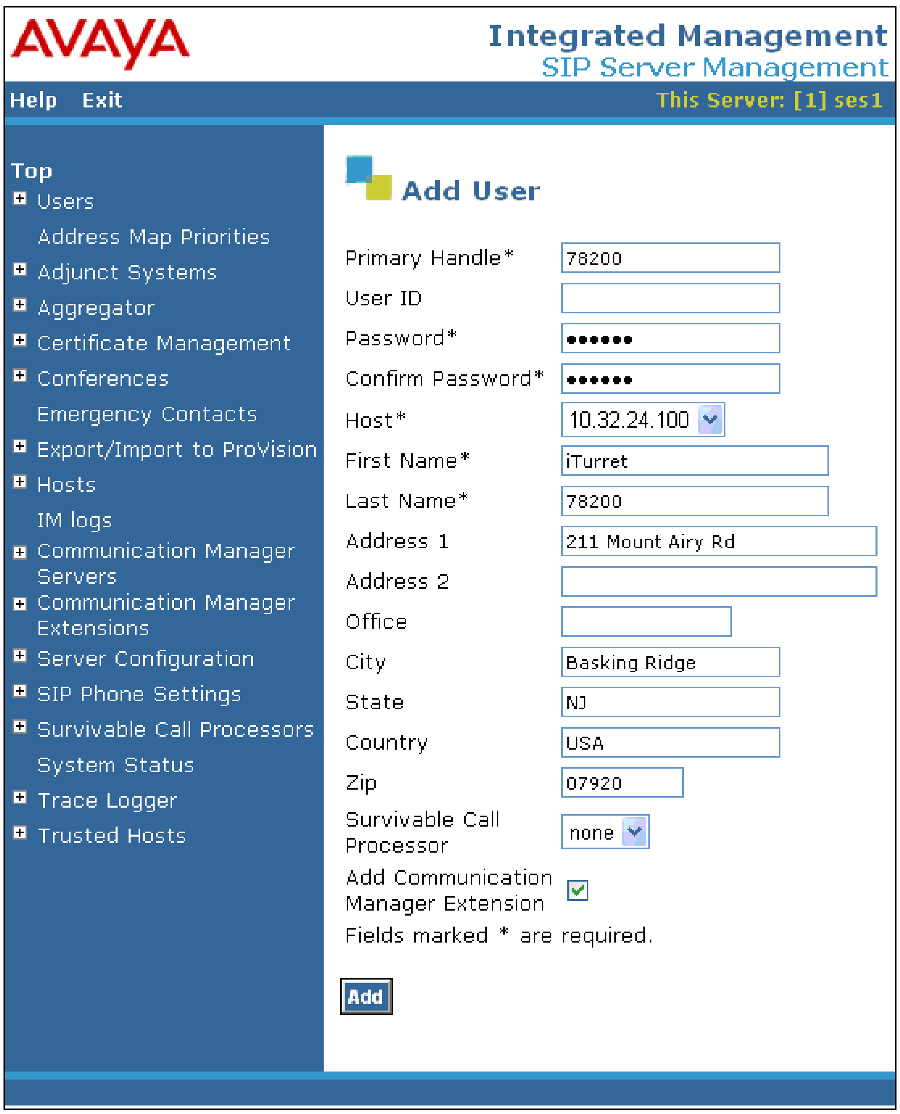
Under the **Communication Manager Servers** option in the Administration web interface, select **Add** to add the Avaya Media Server in the enterprise site since a SIP trunk is required between Communication Manager and SIP Enablement Services. The screen below shows the Edit Communication Manager Server 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.

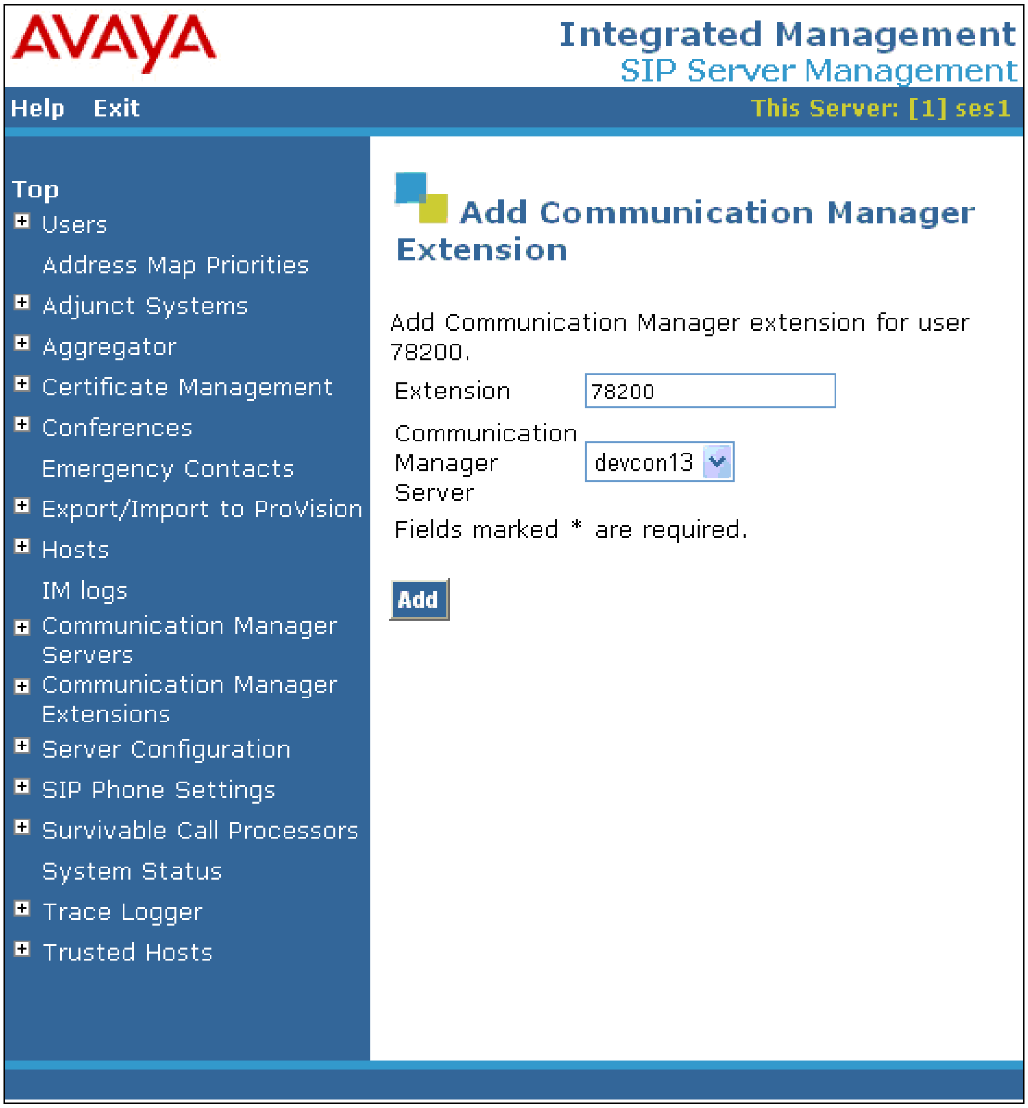
Refer to [3] for additional information on configuring the remaining fields.



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.



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.



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:

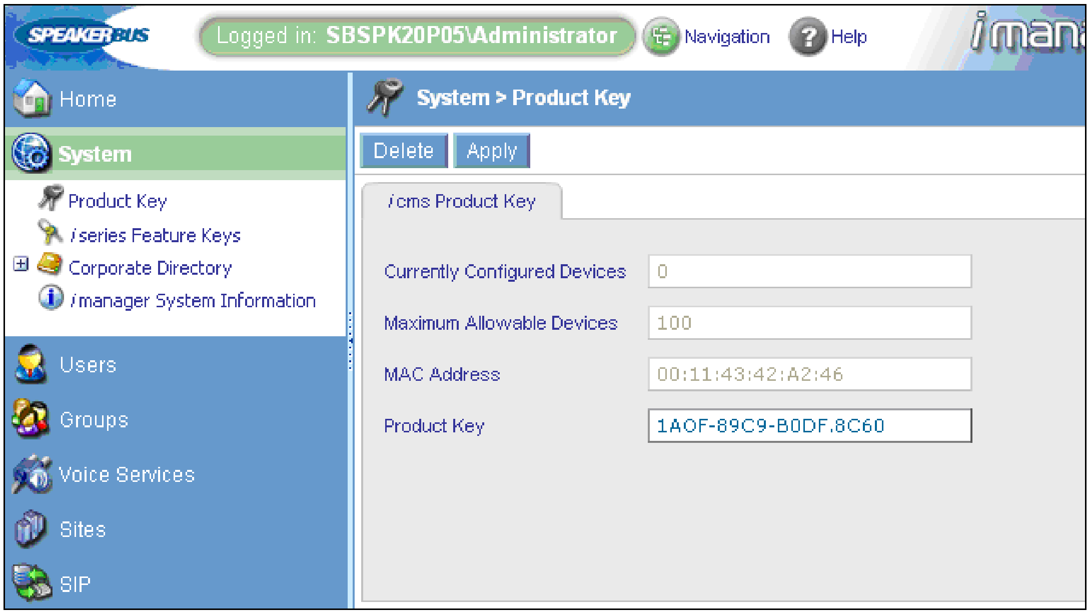
- Launch *i* manager
- Verify Product Key
- Create Site
  - Create Subnet
  - Create Deskstations
- Create SIP Server
  - Create PBX
  - Create Dial Plan
  - Create Appearances
- Create Users
- Create Groups
- Assign Ownership of Appearances to Users
- Assign Default Call Appearance for each User
- Program 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.

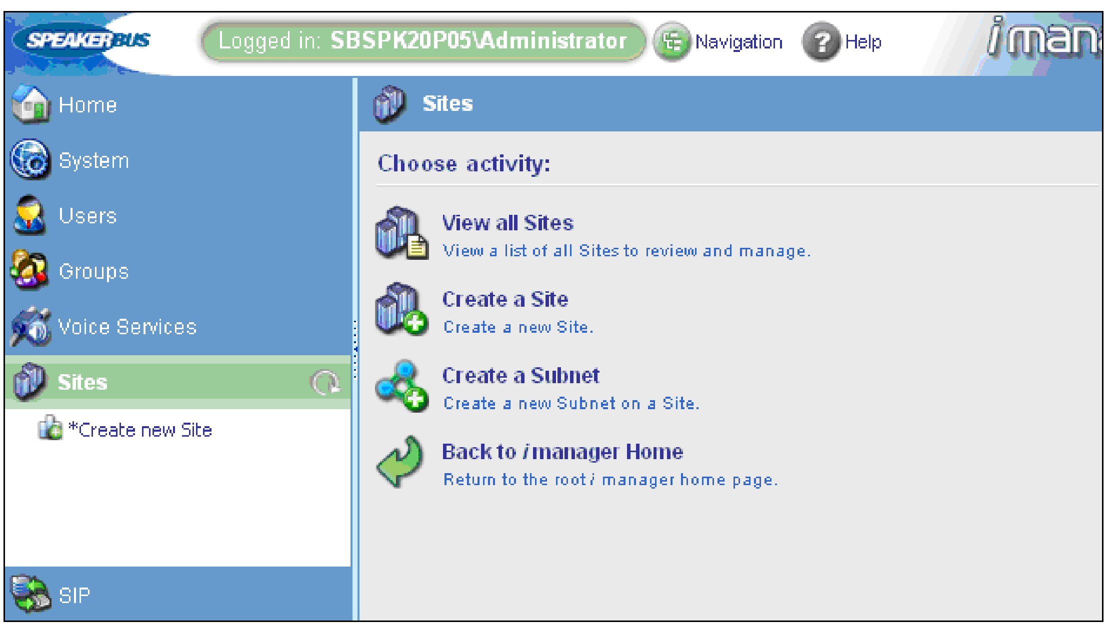
6.2. Verify Product Key

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

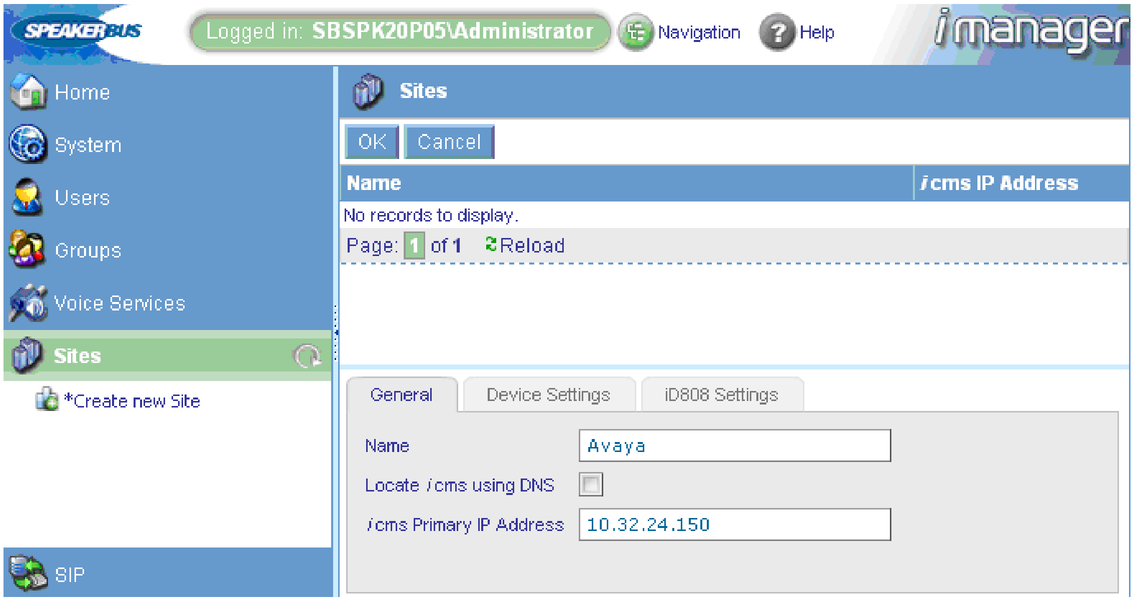


6.3. Create Site

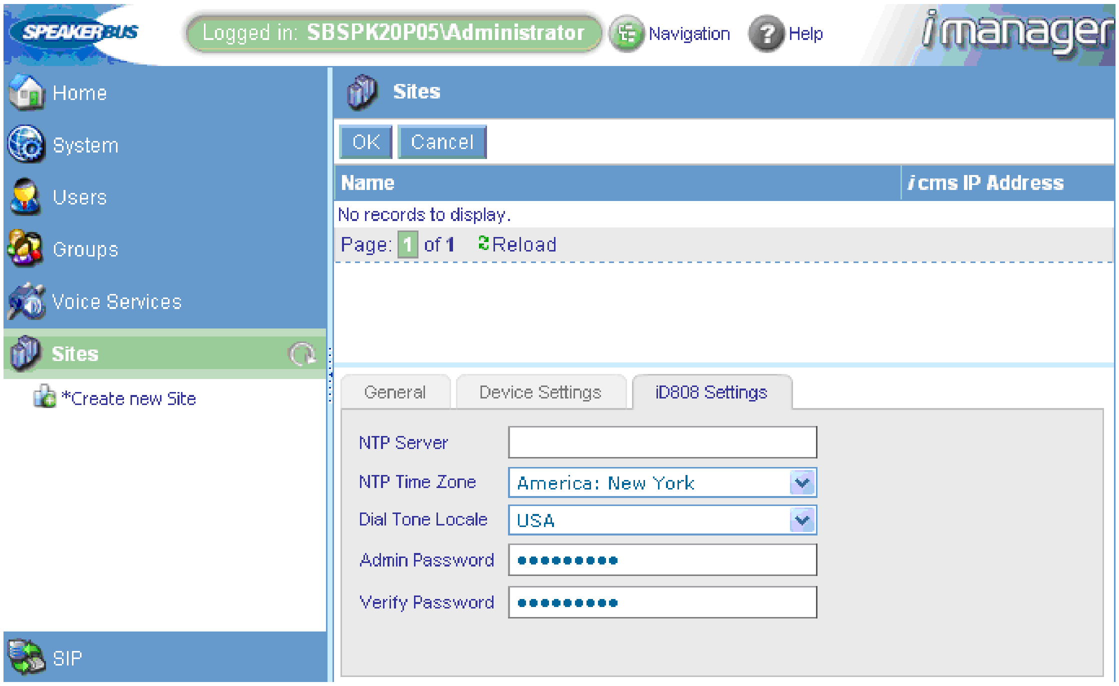
Configure a site representing the location where the Speakerbus iD808 devices are installed. Under **Sites** in the left pane, click on **Create new Site**. The **Sites** page is displayed.



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.



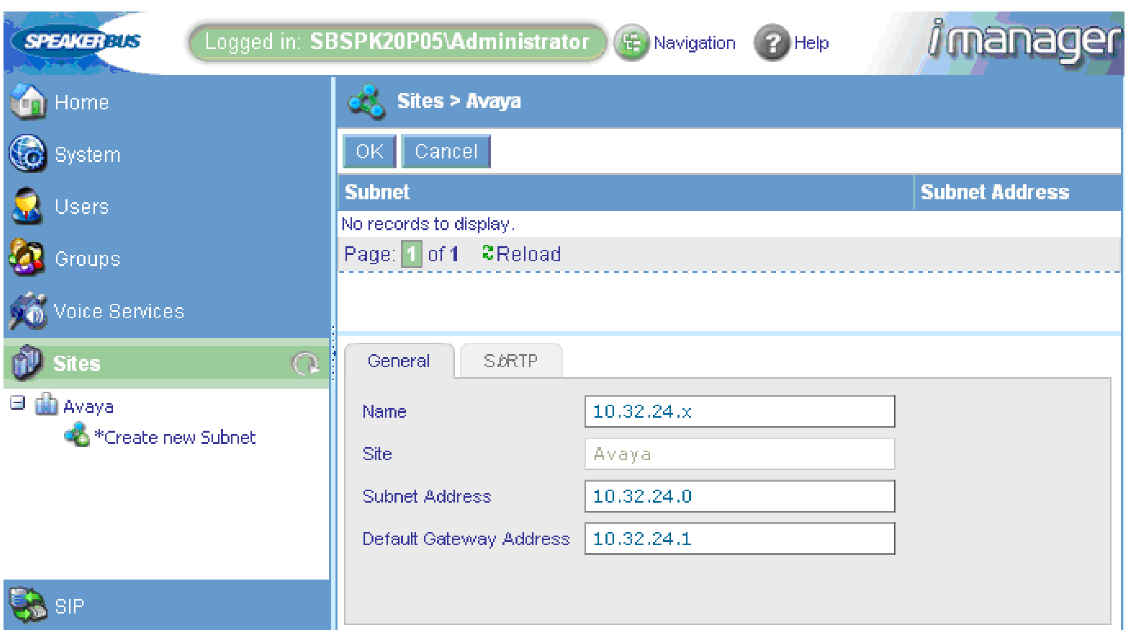
In the **iD808 Settings** tab, set the network time protocol time zone and configure the password for logging into the iD808 deskstation. The **NTP Server** field may be set to the IP address of the NTP server if one is used. Click **OK**. The “Avaya” site will be now listed under **Sites**.



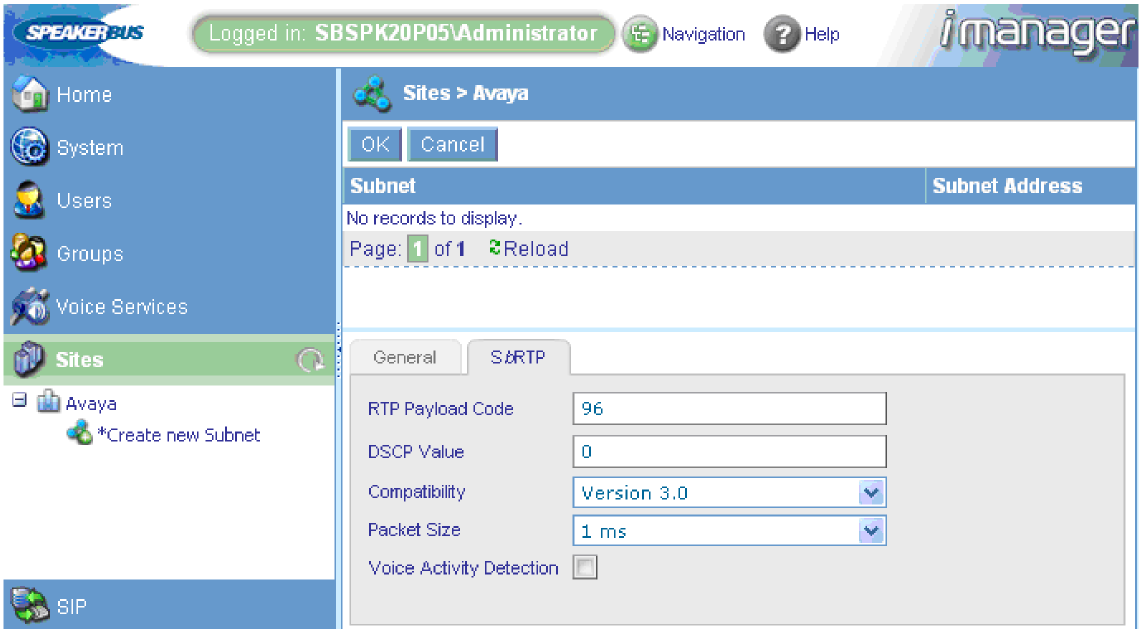
6.4. Create Subnet

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.

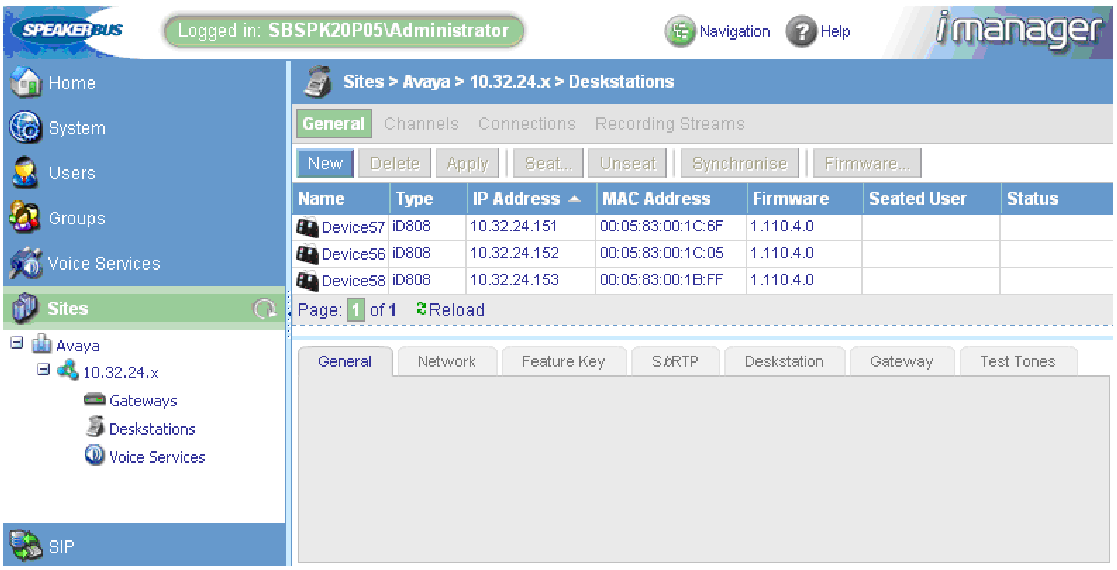


In the **SbRTP** tab, set the **Compatibility** field to *Version 3.0*. Click **OK**.

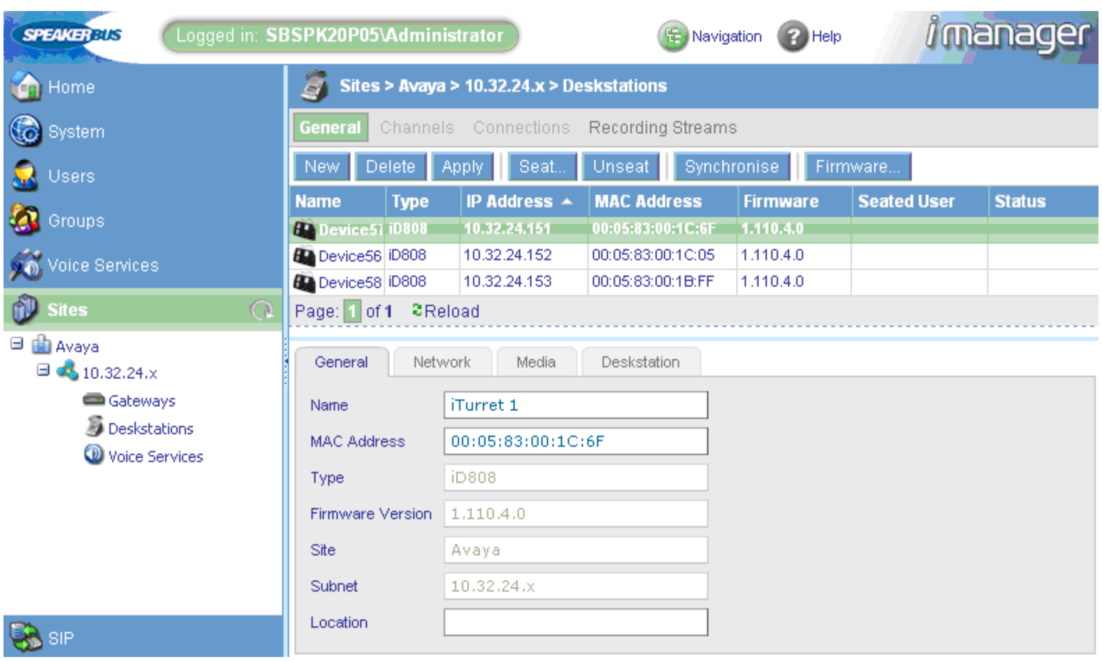


6.5. Create Deskstations

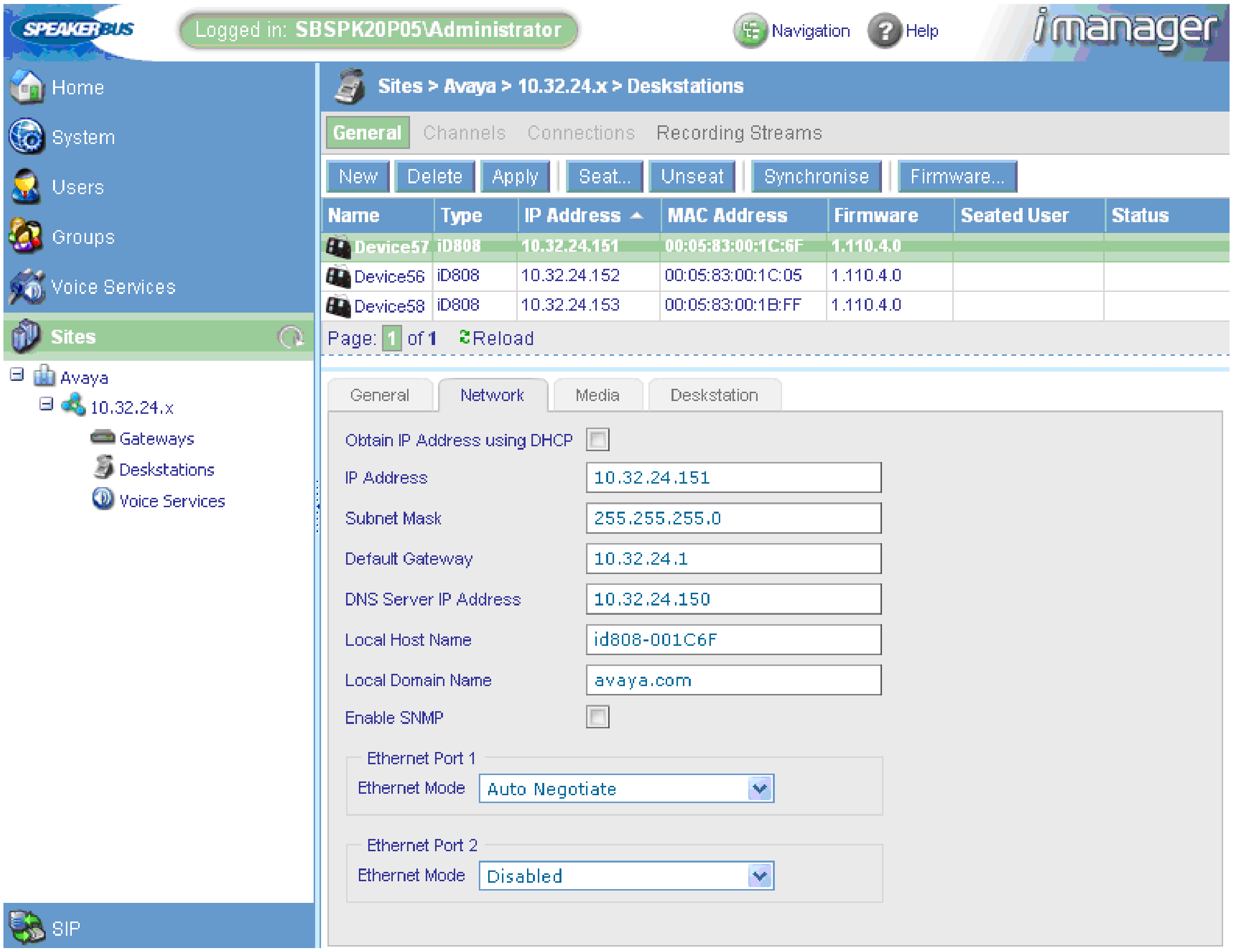
Once the site and subnet exist, the iD808 devices can be created. The network was set up using DNS without DHCP so IP configuration was manually performed on the iD808 devices. The deskstations then registered to *i* manager through DNS. After the iD808 devices are synchronized with *i* cms, they should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



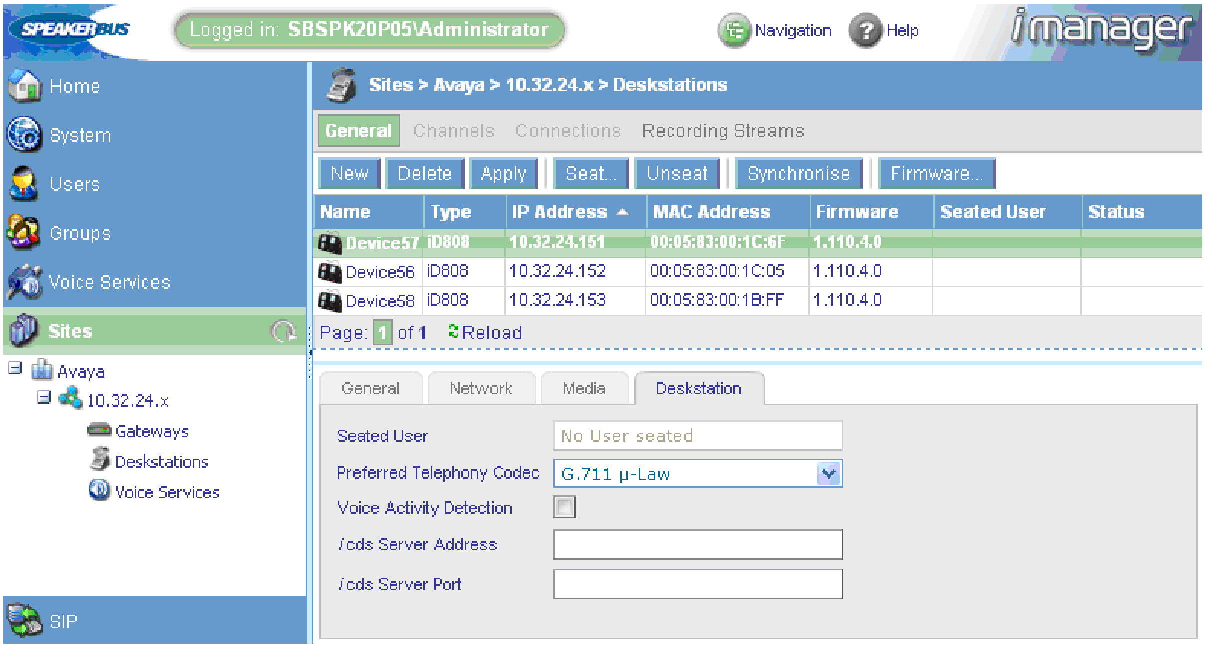
Select a device and change the name to a more descriptive one in the **General** tab.



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



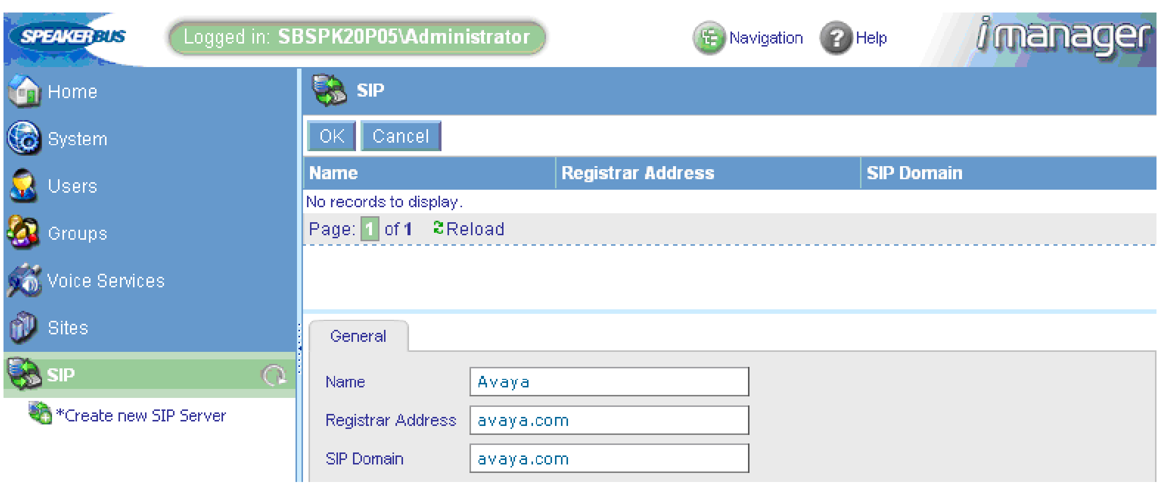
In the **Deskstation** tab, select preferred codec. In this configuration, G.711mu-law is the preferred codec. Click **Apply**. Repeat these steps for all deskstations.



6.6. Create SIP Server

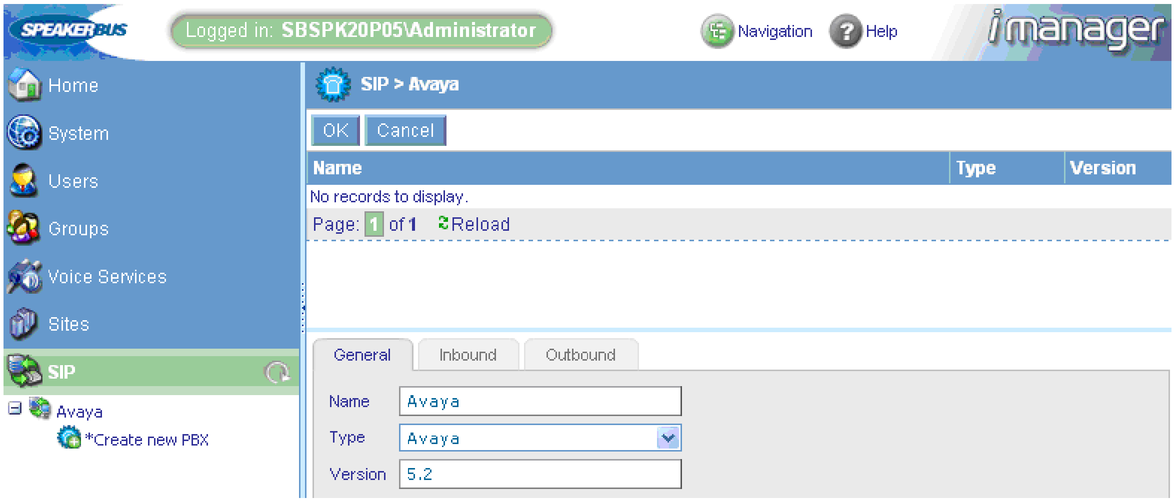
To create a SIP Server, click **Create a new SIP Server** under the **SIP** directory in the left pane. Provide a descriptive name for the SIP server and set the **Registrar Address** and **SIP Domain** fields to *avaya.com*, which in this configuration DNS resolves the domain name to 10.32.24.100, the 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 locater record (SRV) for registrar address and SIP domain on DNS. Refer to [5] for more details.



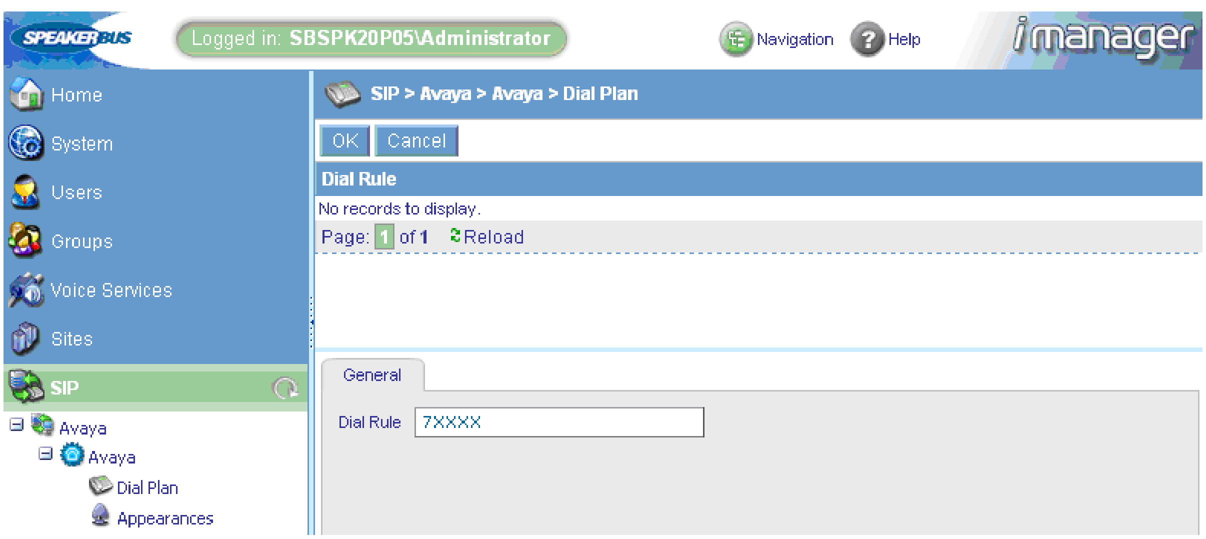
6.7. Create PBX

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**.



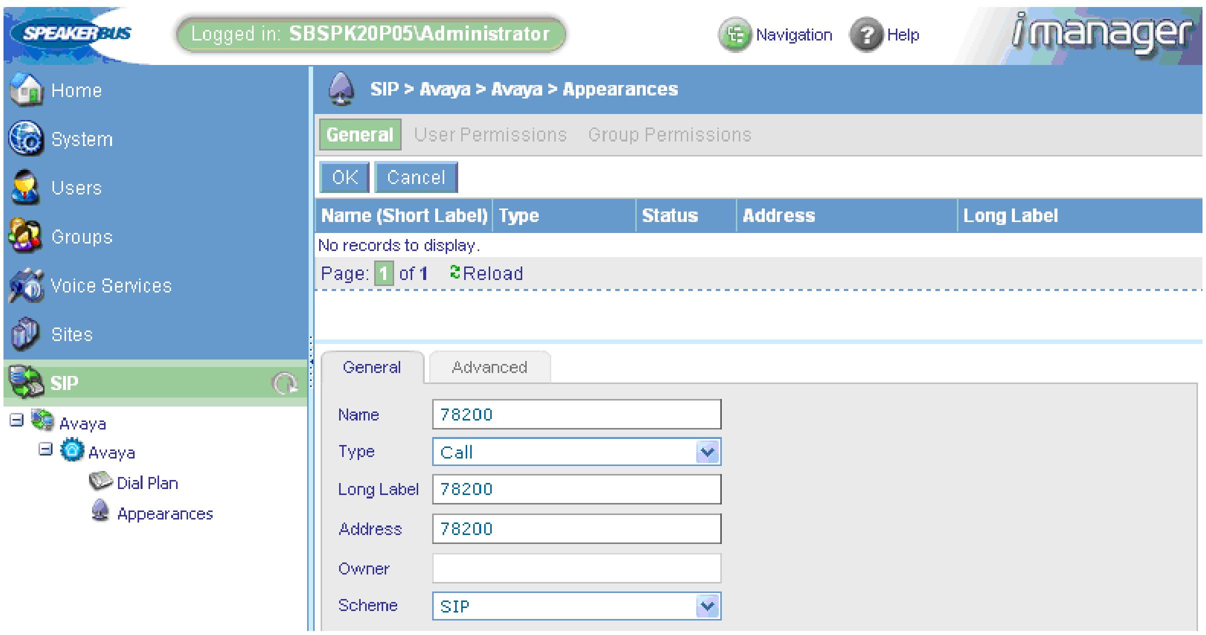
6.8. Create Dial Plan

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.

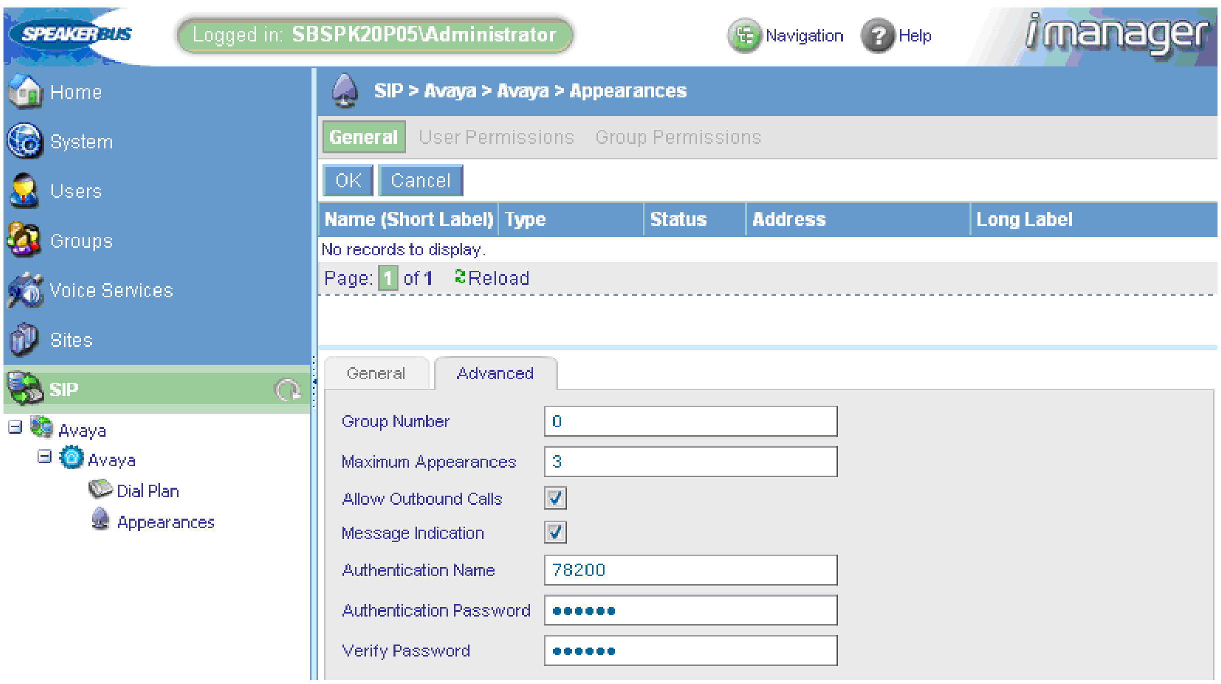


6.9. Create Call and Handset Appearances

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*.

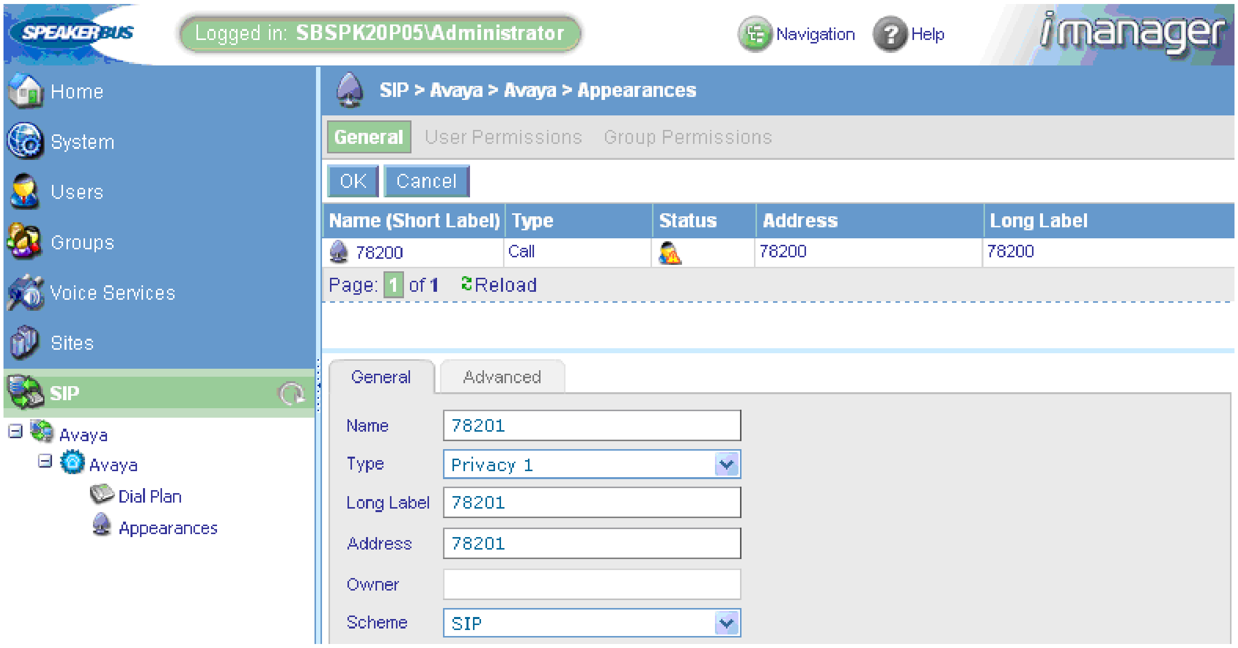


In the **Advanced** tab, set the **Maximum Appearances** field to "3". This field should be set to the number of call appearances configured on the station in Communication Manager minus one since the last call appearance is restricted. See the button assignment section of the station form and the second 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**.

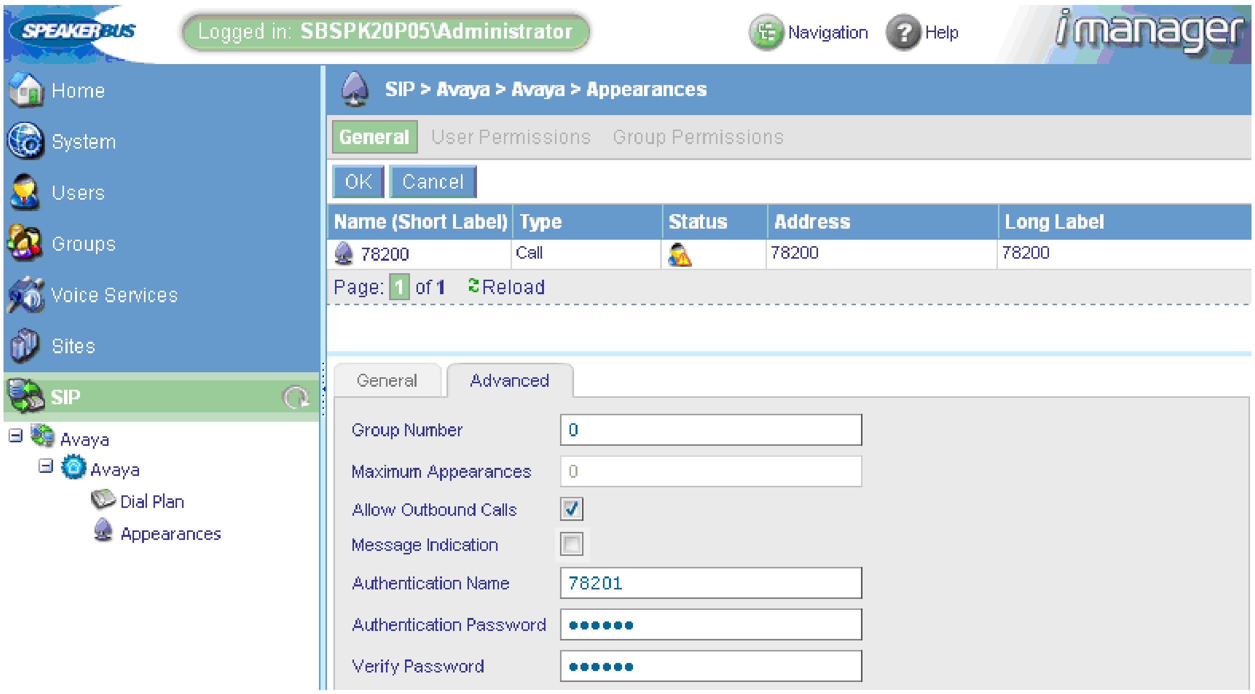


Next, this procedure will be repeated for the two handset appearances.

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*.

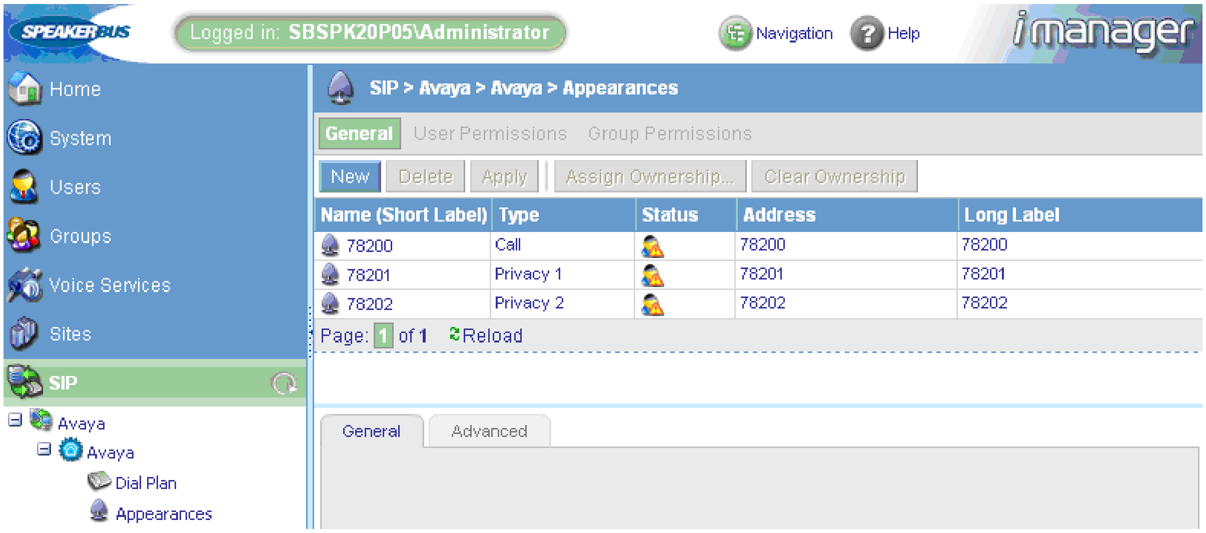


In the **Advanced** tab, configure the **Authentication Name** and **Authentication Password** fields with the credentials for registering with SIP Enablement Services. For the handset appearances, the **Maximum Appearances** field should be set to "0" since no calls will be made to the handset appearances 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**.



Repeat the above procedure to add the handset 2 appearance.

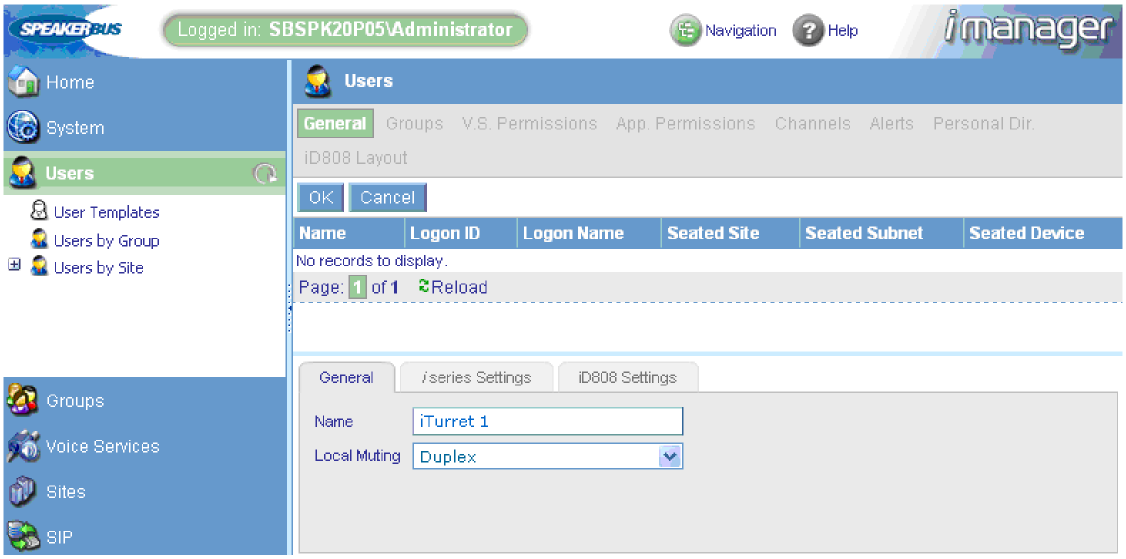
The three call appearances for the previously configured iD808 deskstation are listed below.



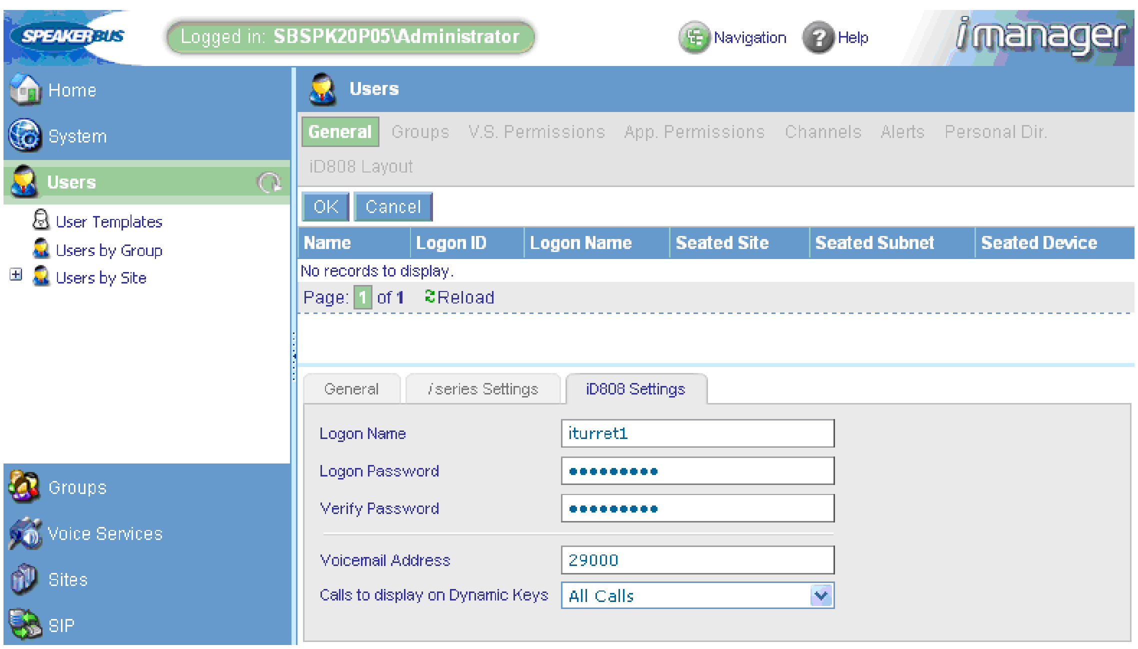
Repeat the above procedure for adding the main and handset appearances for each iD808 deskstation.

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.

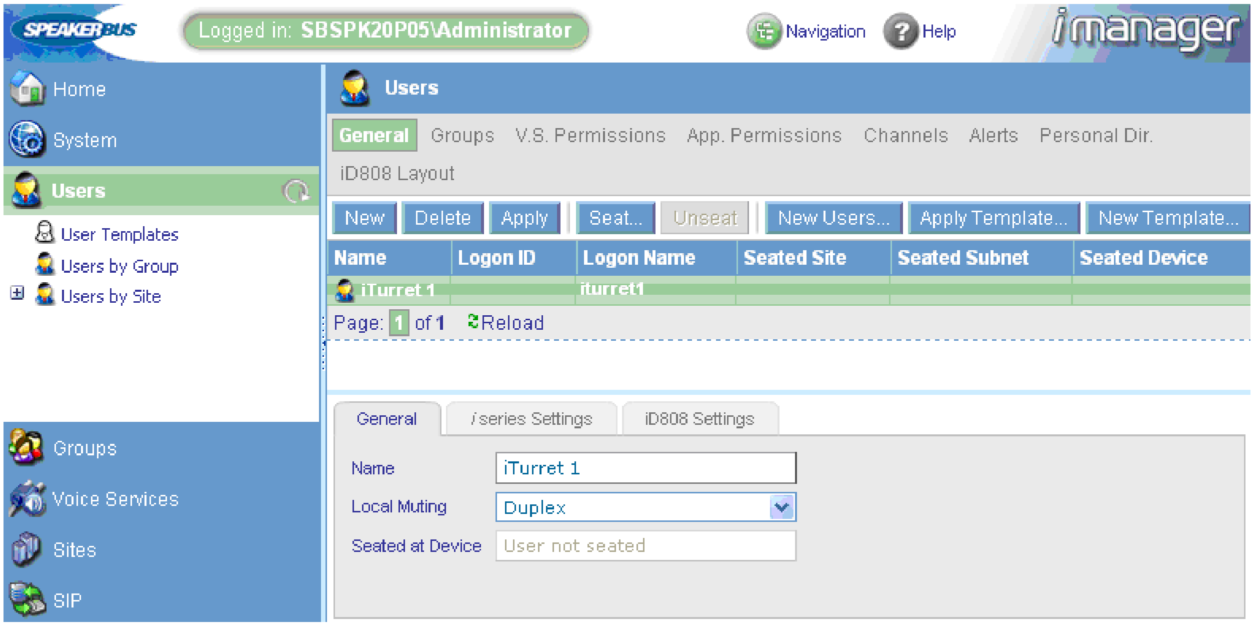


In the **iD808 Settings** tab, provide the logon credentials for the user to log into their iD808 deskstation. This page will be revisited later in **Section 6.14** to configure the default call appearance for this deskstation. Click **OK**.

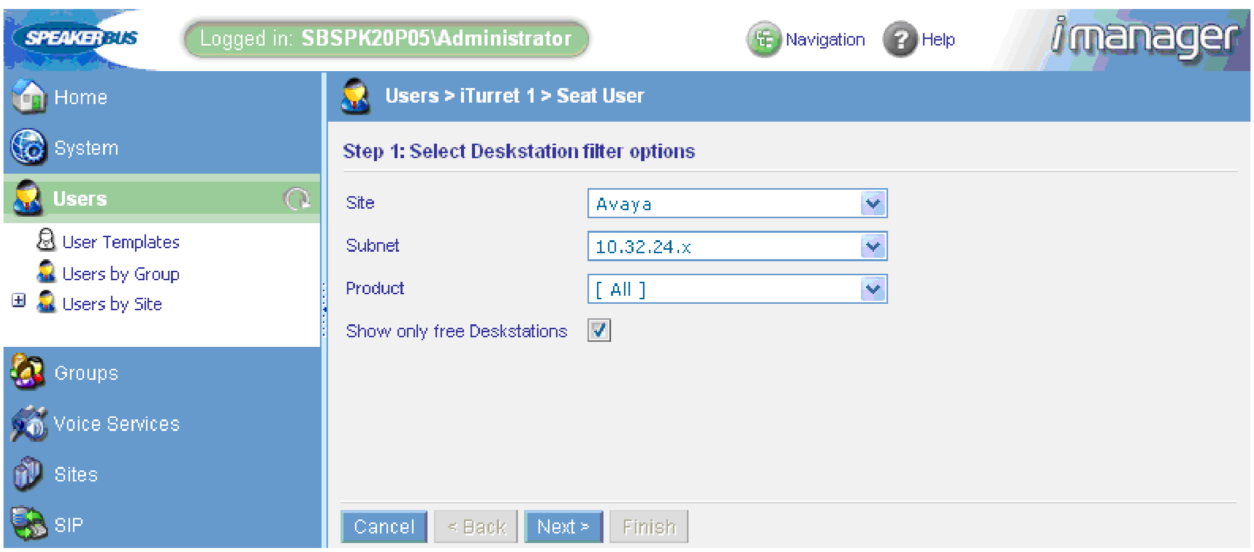


Repeat the previous procedure to add more users.

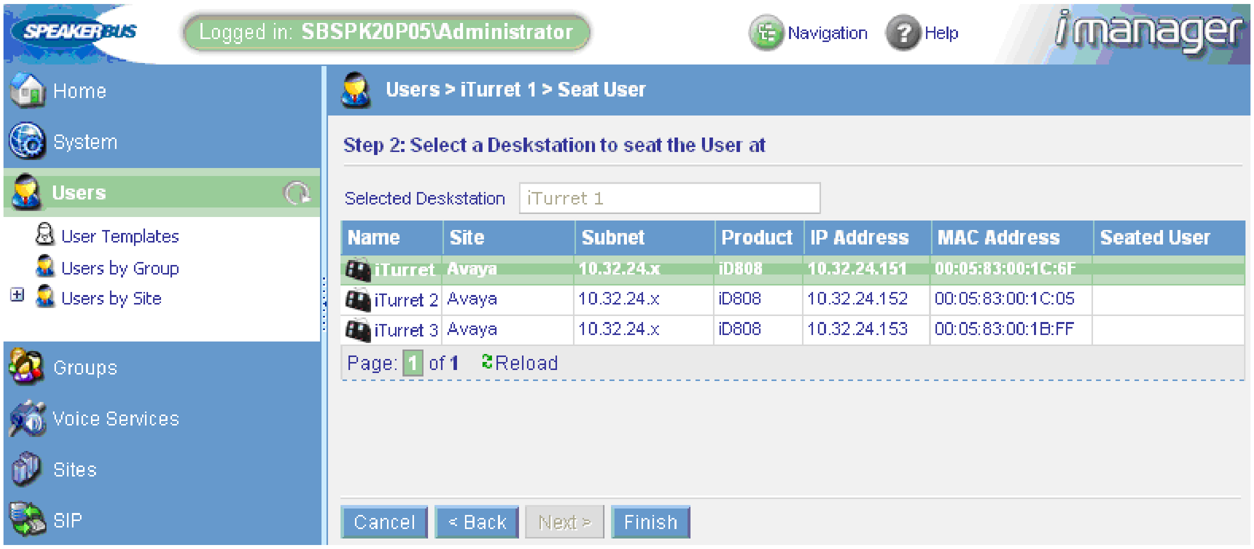
After a user has been created, the user needs to be "seated" on an iD808 deskstation. In the **Users** directory tree, click on the "Avaya" link under **Users by Site** to display the list of users. Select the user previously configured (i.e., iTurret 1) and click on the **Seat…** button.



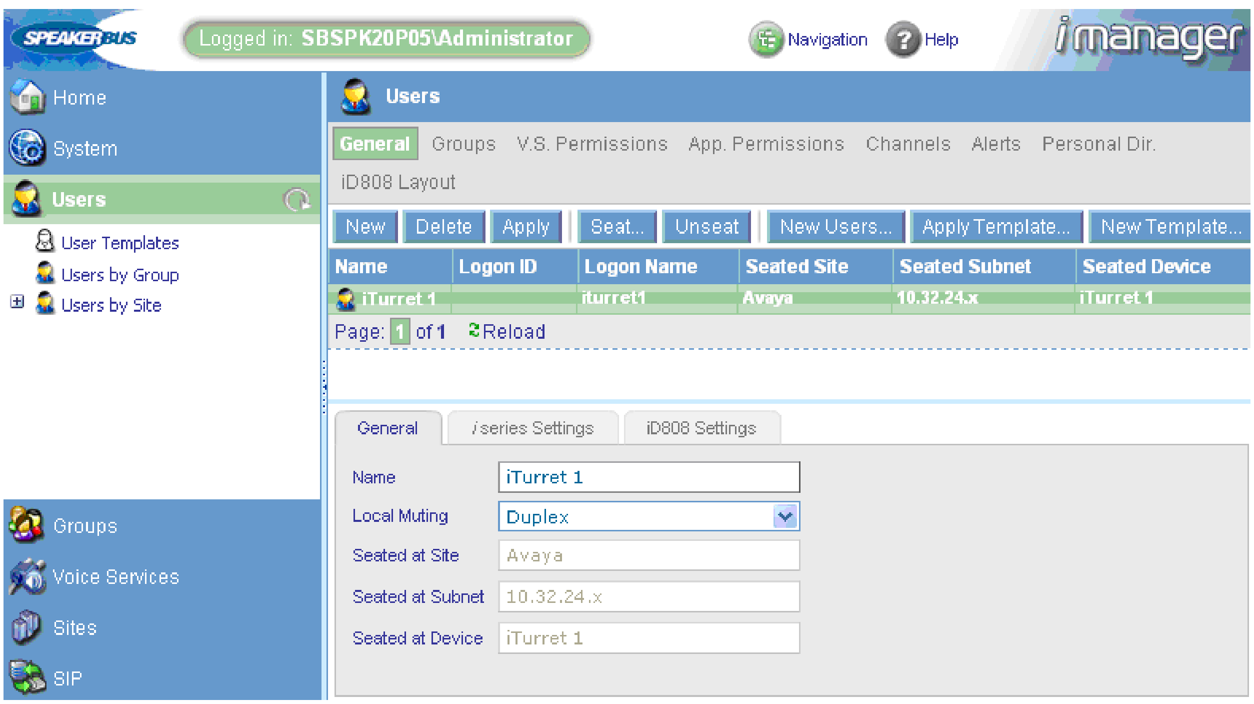
On the next page, filter options are presented. Filter deskstations in the "Avaya" site and in the "10.32.24.x" subnet as shown below. The user will be seated on an iD808 deskstation with these properties. Click **Next**.



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**.

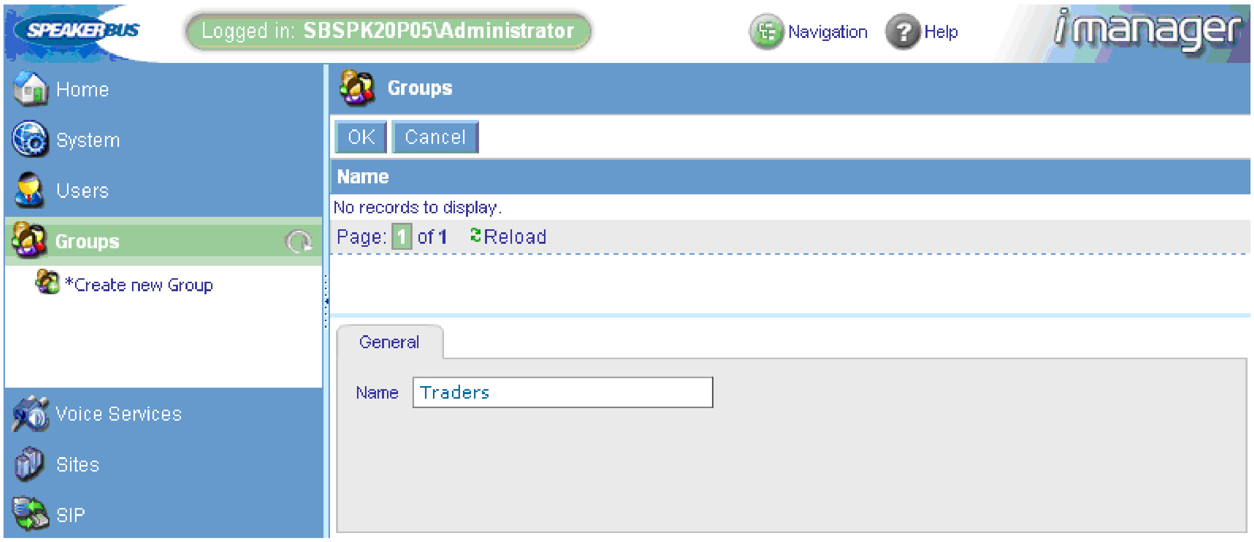


The user has been successfully seated as indicated by the deskstation displayed in the **Seated Device** column on the following page. Repeat this procedure for seating other users.

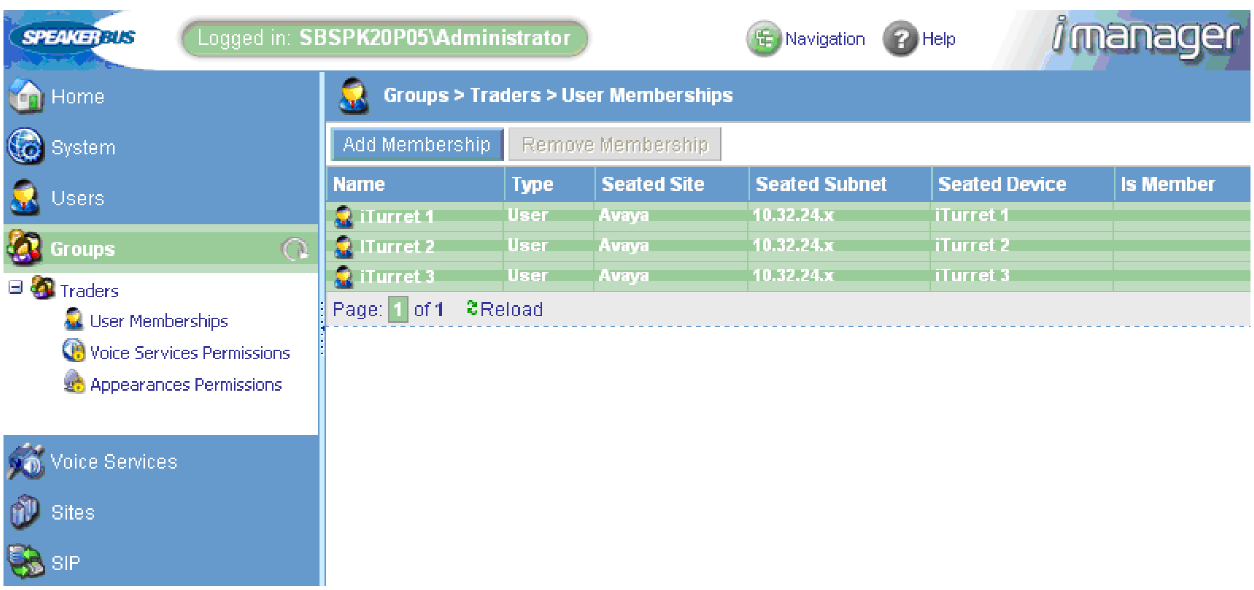


6.11. Create Group

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.

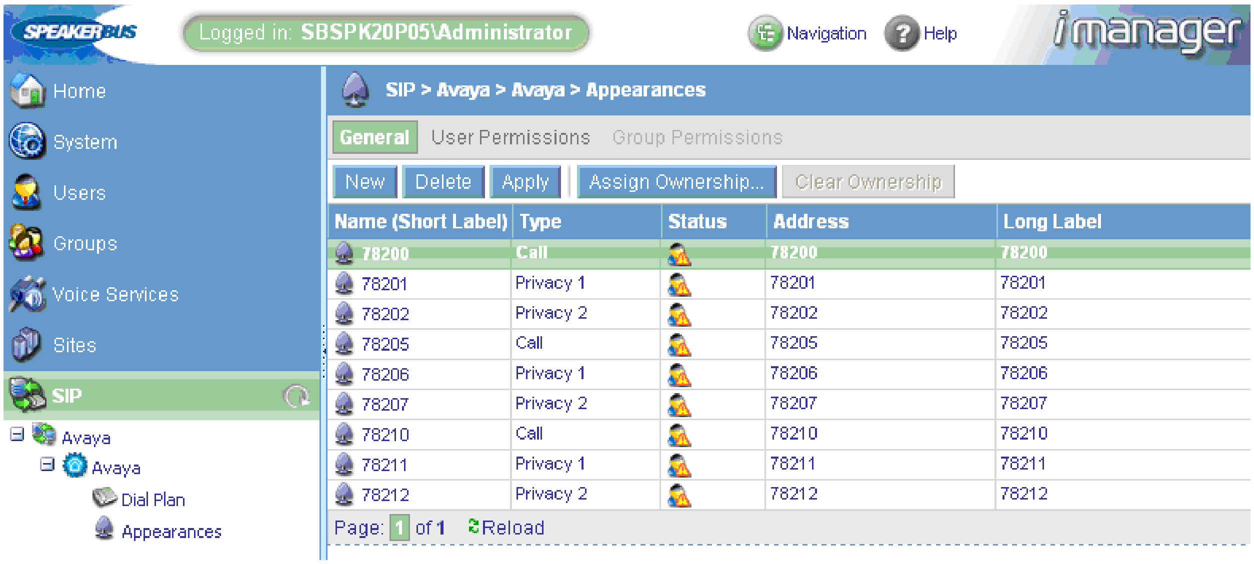


In the **Groups** directory tree, expand "Traders" and click on **User Memberships** in the left pane. A list of users is displayed. Select all the users to be added to the "Traders" group as shown below and then click **Add Membership**. The **Is Member** column will then indicate that the selected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

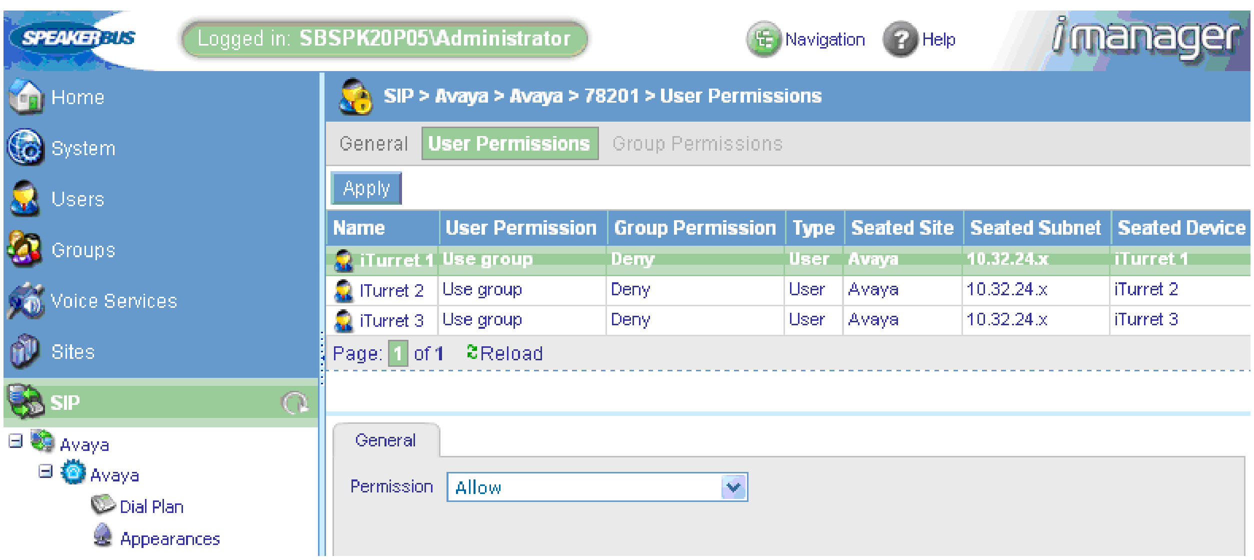


6.12. Assign User Permissions

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**.

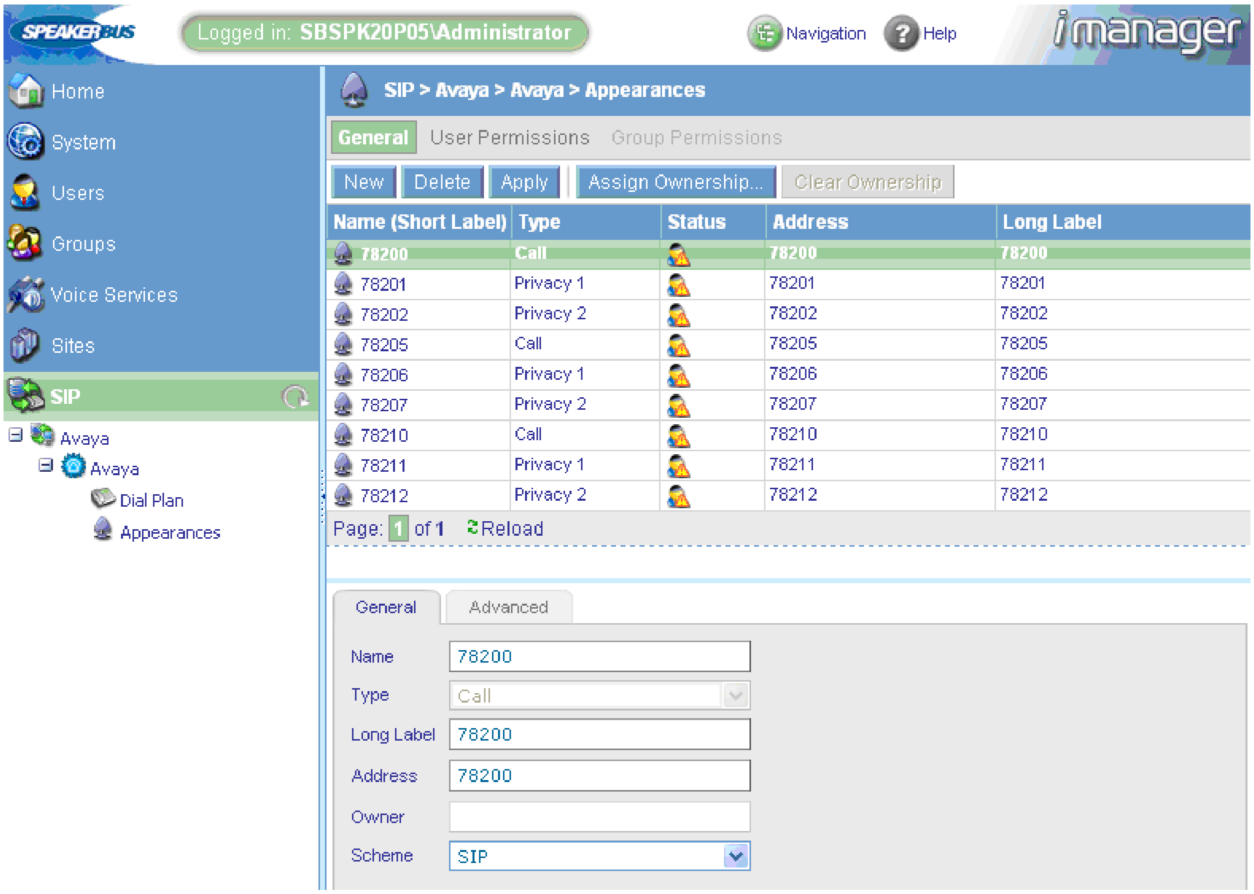


Select the user on the next page to which the appearance will be assigned. Set the **Permission** field to *Allow* as shown below. Click **Apply**. Assign "Privacy 1" and "Privacy 2" permissions to this user by repeating this procedure.

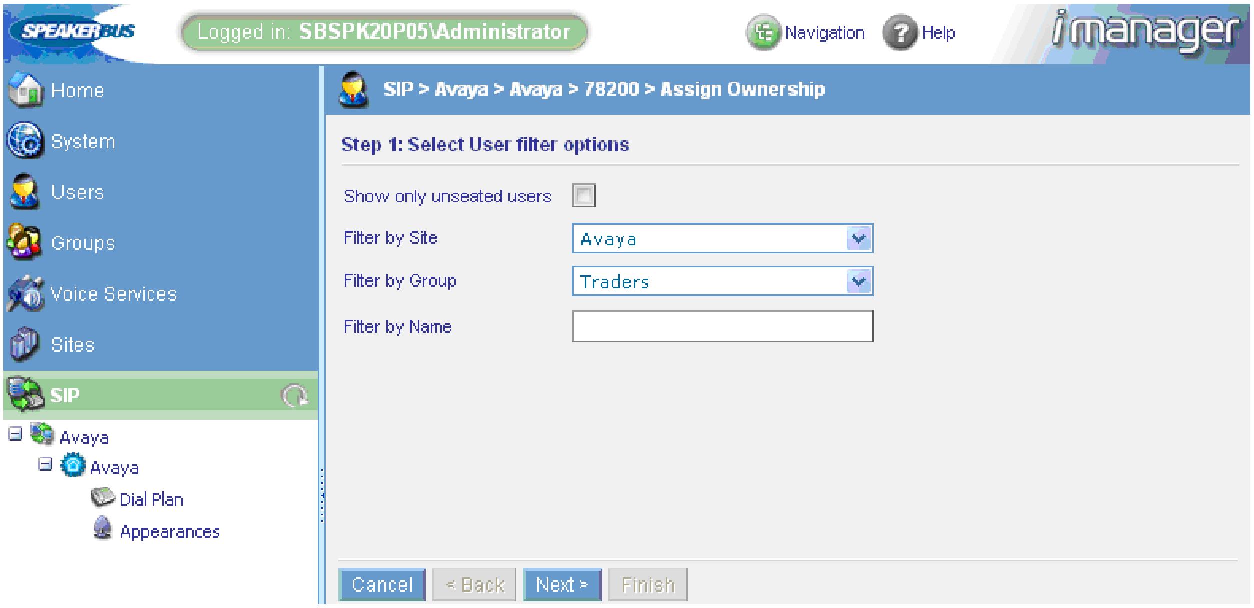


6.13. Assign Ownership

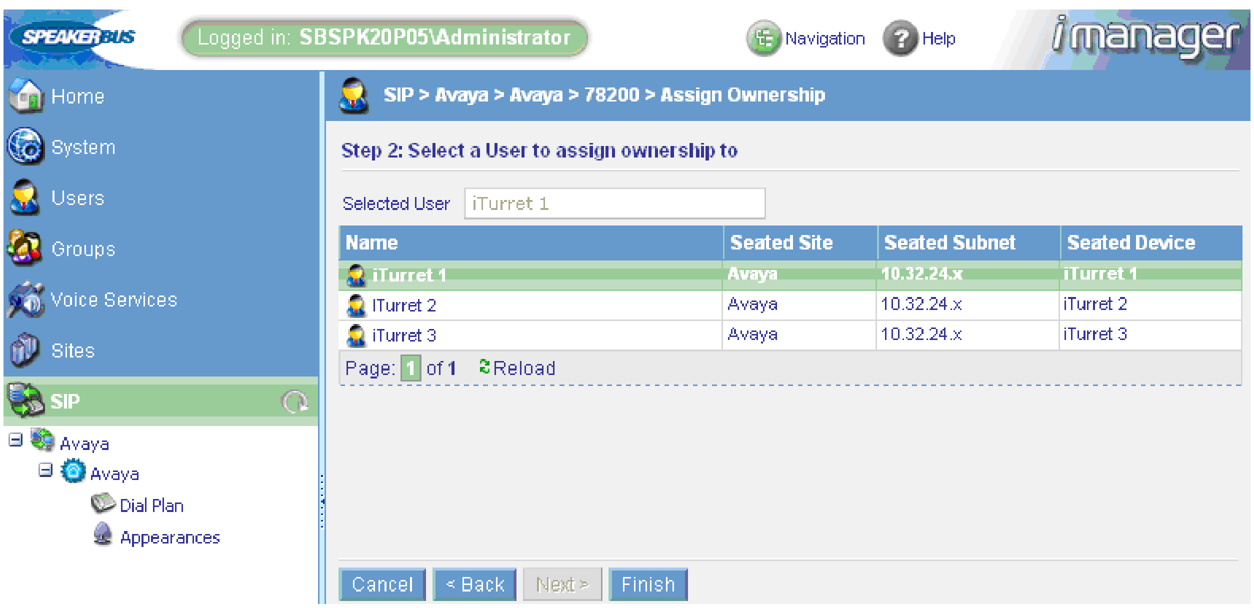
Assign ownership of the appearances to a user. In the **SIP** directory tree, click on **Appearances** to display the appearances list as shown below. In the **General** tab, select the main call appearance and click on the **Assign Ownership** button.



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



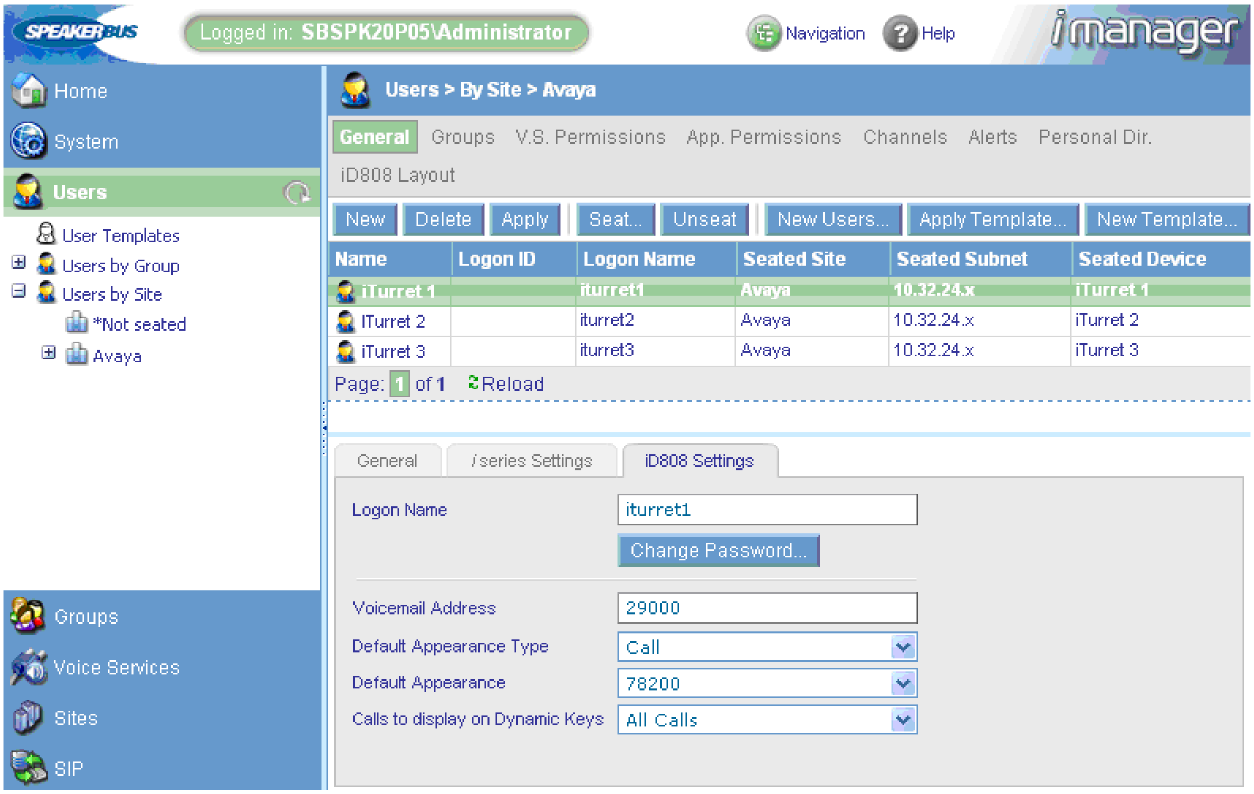
On the next page, select the user to which ownership will be assigned to the main call appearance. In this example, the main call appearance 78200 will be assigned to **“**iTurret 1”. Click **Finish**.



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

6.14. Assign Default Call Appearance

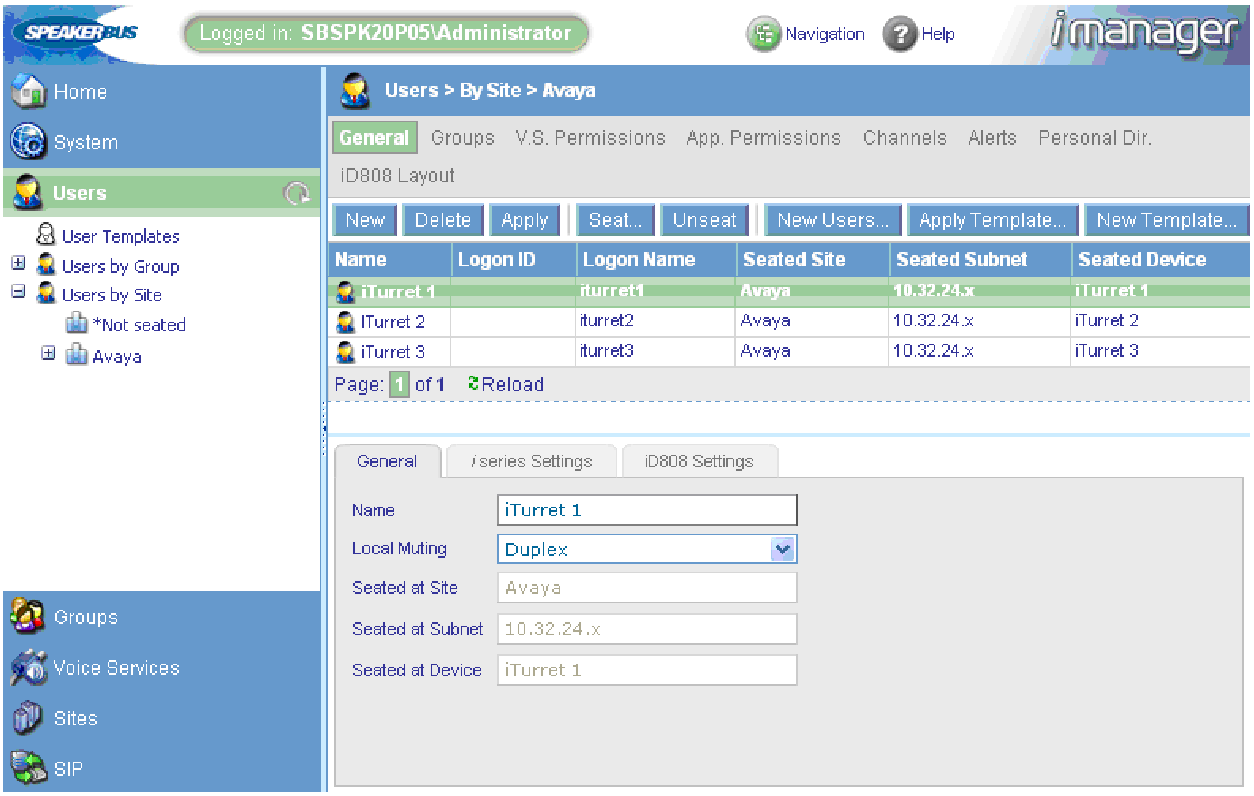
In the **Users** directory tree, click on the "Avaya" link to display the users list. Set the **Default Appearance** field to the main call appearance (e.g., 78200). Click **Apply**.



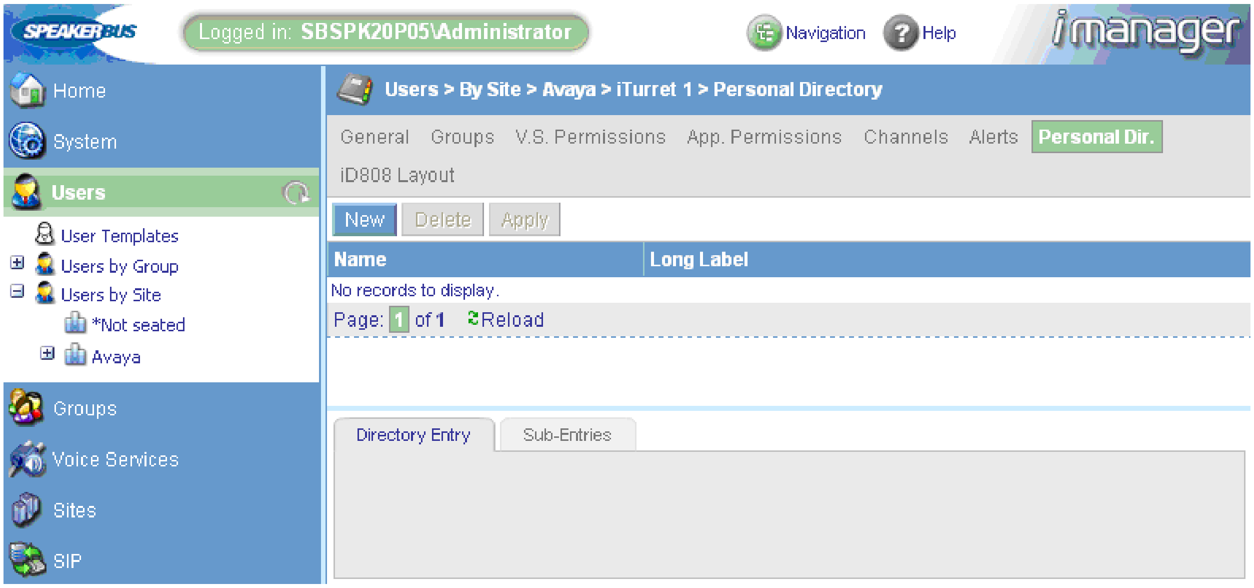
6.15. Program Feature Name Extensions (FNEs)

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.

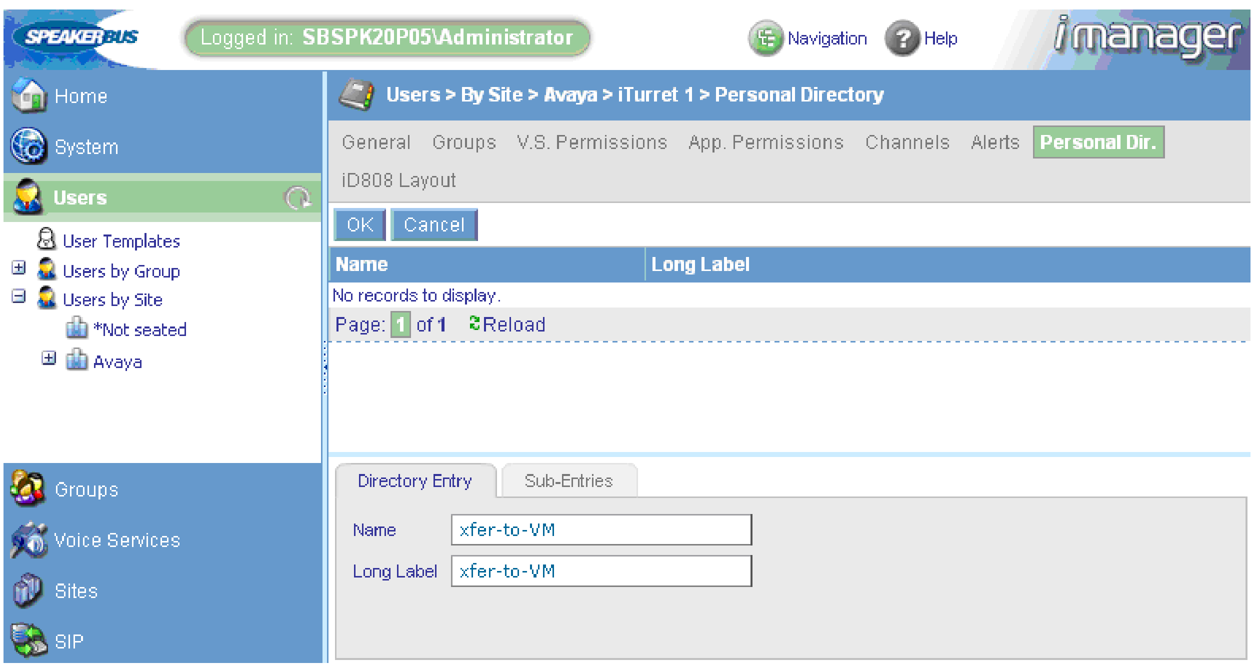
In the **Users** directory tree, click on the "Avaya" link and then select a user (e.g., iTurret 1). Click on **Personal Dir.**

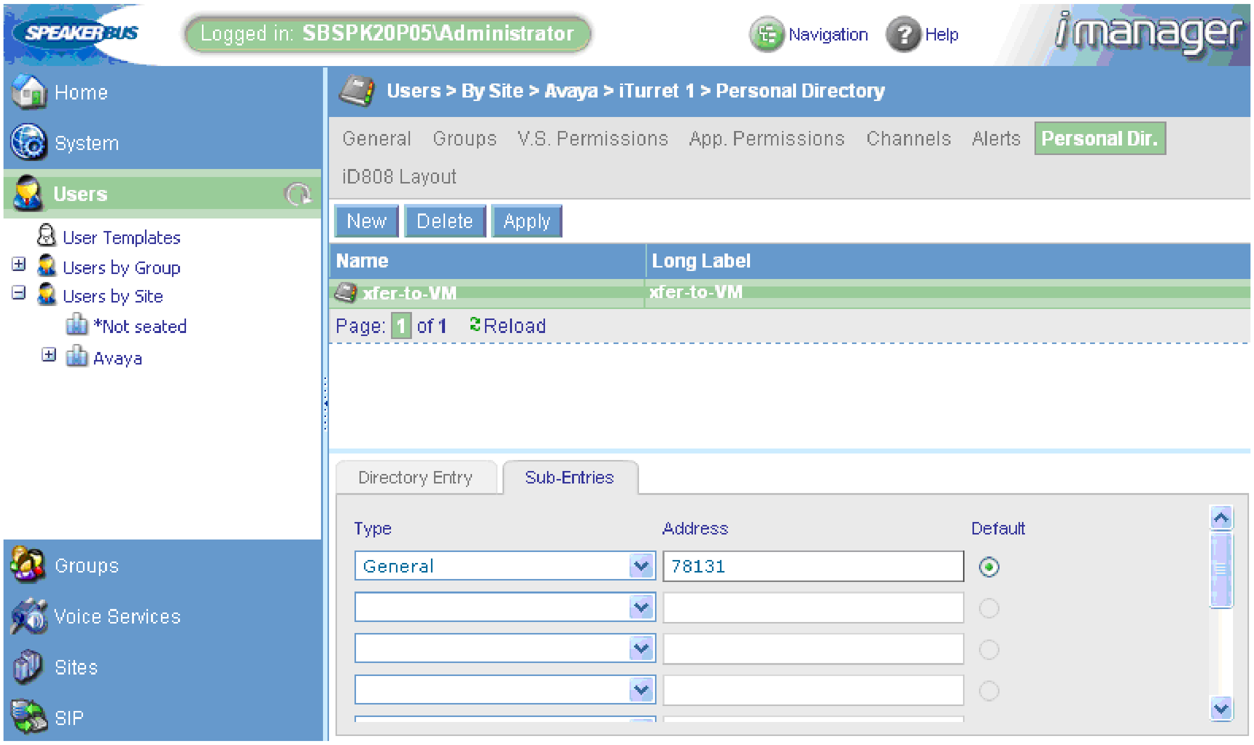


On the next page, click the **New** button to add an FNE in the personal directory.



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.



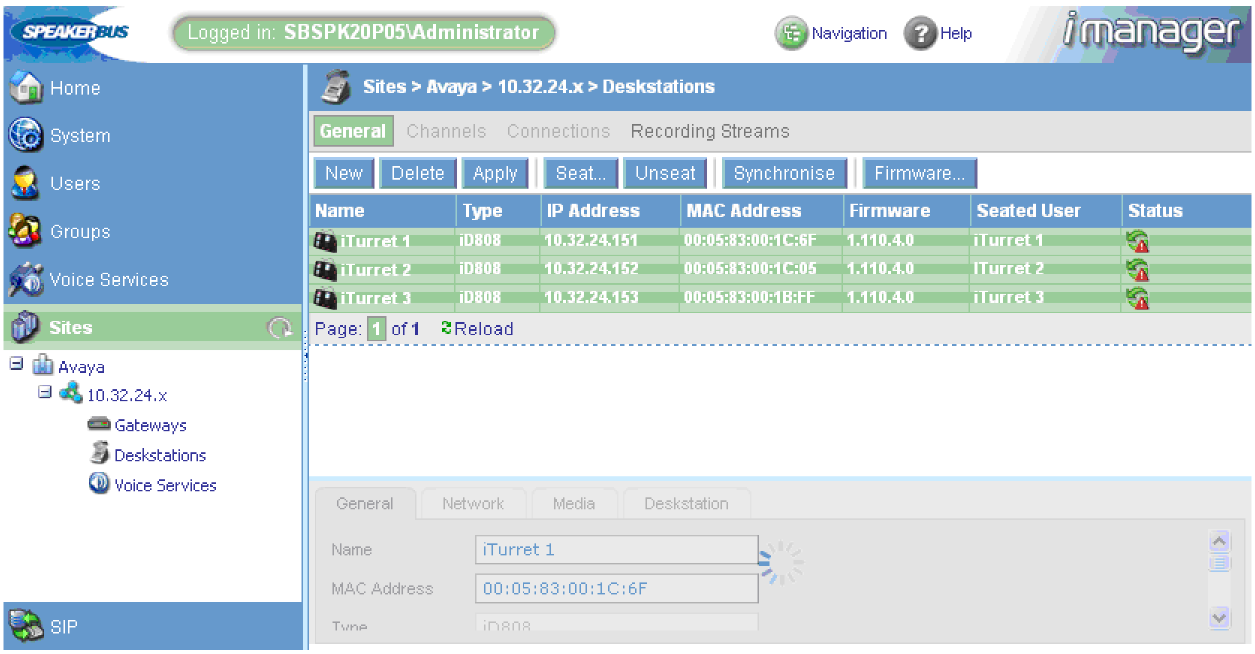


Refer to [5] for instructions on how to create iD808 deskstation keys. The following keys can be created using the iD808 layout page: Dynamic, Appearance, Shortcut, Soft Function, and Speed Dial amongst others. For the compliance test, each user was configured with three Dynamic keys, two Soft Function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



7. General Test Approach and Test Results

To verify interoperability of Speakerbus iD808 *i* turret with Avaya Aura^{TM} Communication Manager and Avaya Aura^{TM} 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 FNUs. 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 Aura^{TM} Communication Manager and Avaya Aura^{TM} 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 Aura™ 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 Aura™ Communication Manager Running on the Avaya S8xxx Servers*, May 2009, Issue 9, Document Number 555-245-206.

[4] *Installing, Administering, Maintaining, and Troubleshooting Avaya Aura™ 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.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.