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* turrets to a SIP infrastructure consisting of Avaya Aura^{TM} SIP Enablement Services and Avaya Aura^{TM} Communication Manager. Also described is how Avaya Aura^{TM} 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^{TM} Communication Manager to the Speakerbus iD808 *i* turret, 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^{TM} Communication Manager.

The following table provides a summary of the supported features available on *i* turret with the Avaya SIP offer. Some features are supported locally in *i* turret, while others are only available with Avaya Aura^{TM} Communication Manager and Avaya Aura^{TM} 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 *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 *i* turret can also be programmed to a FNE. Other features, such as Exclusion/Privacy and Call Forwarding, are available by using the AST (Advanced SIP Telephony) FNU (Feature Name URI). Avaya Aura^{TM} 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]** and **[3]**. The Avaya SIP solution requires all SIP telephones to be configured in Avaya Aura^{TM} Communication Manager as OPS.

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	No	Yes	Via OPS FNE, *i* turret does not support distinctive ring indication
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 *i* turret with SIP Enablement Services
- Calls between *i* turret and Avaya SIP, H.323, and digital stations
- G.711, G729 and G.722 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 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
- Exclusion/Privacy using the Exclusion FNU
- Proper system recovery after an *i* turret restart and loss of IP connection
- Correct *i* turret behavior during SES failovers and simulated network failures

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 a pair of SIP Enablement Services servers, a pair of Avaya S8730 Servers with a G650 Media Gateway running 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 SIP Enablement Services and are configured as OPS stations on 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.

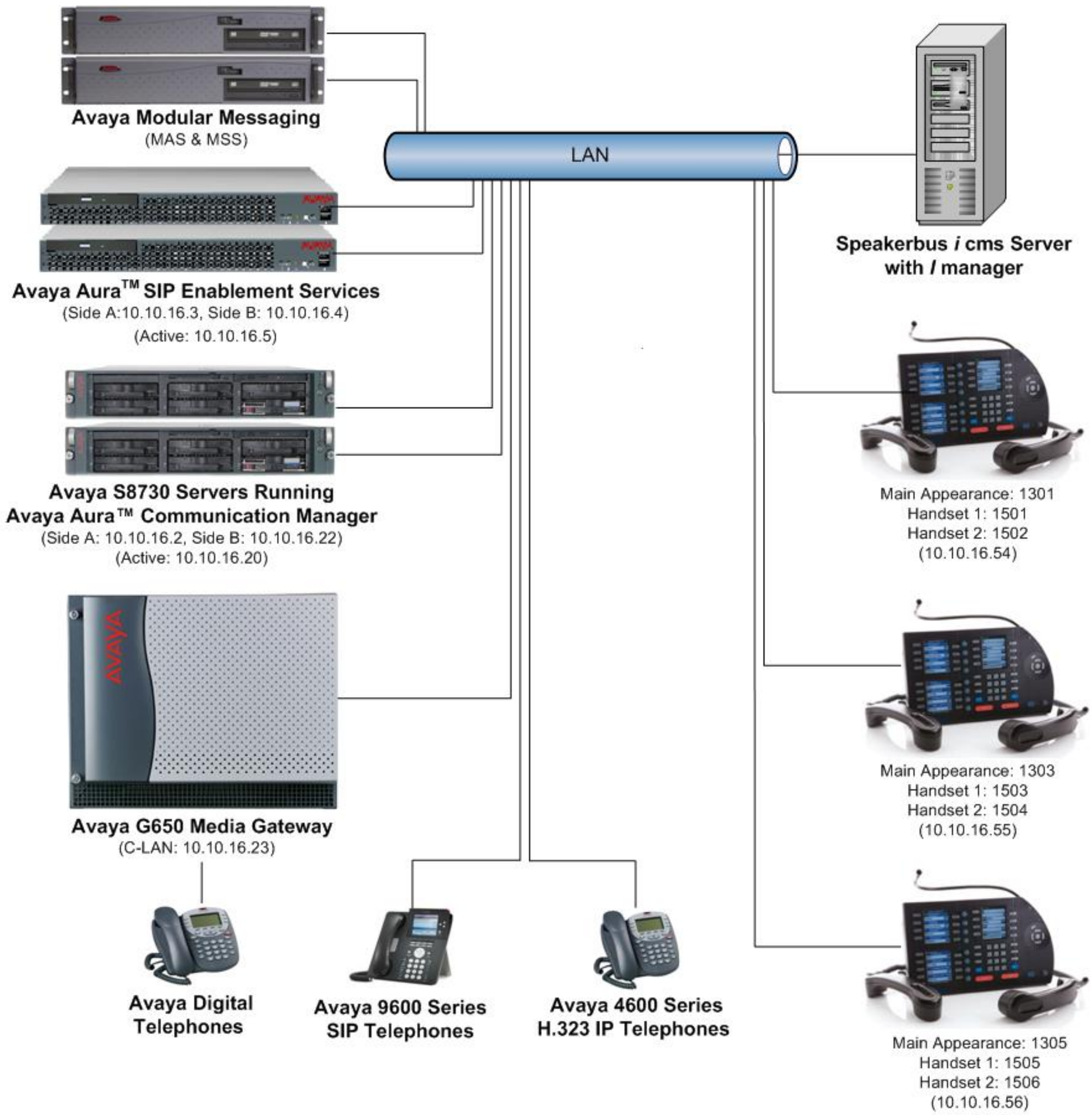


Figure 1: Speakerbus iD808 *i* turret with Avaya SIP Solution

3. Equipment and Software Validated

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

Hardware Component	Version
Avaya S8730 Servers (Redundant Pair)	Avaya Aura^{TM} Communication Manager 5.2.1 (R015x.02.1.016.4) with Service Pack 0 (Patch 17774)
Avaya G650 Media Gateway TN2302AP Media Processor	 HW32 FW120
Avaya S8500B Servers (Redundant Pair)	Avaya Aura^{TM} SIP Enablement Services 5.2.1(SES-5.2.1.0-016.4) with Service Pack 2
Avaya S3500 Servers Modular Messaging	Modular Messaging 5.2
Avaya 4600 Series IP Telephone	3.0 (H.323)
Avaya 9600 Series IP Telephones	2.5.0.0 (SIP)
Avaya Digital Telephones	--
Speakerbus iD808 *i* turret	1.120.2.0
Speakerbus *i* cms Server with *i* manager Administration on Windows 2003 Server	1.220.1.0

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 Communication Manager and SIP Enablement Services. Use the System Access Terminal (SAT) to configure Avaya Aura™ Communication Manager. Log in with the appropriate credentials.

4.1. Verify System Capacity

The license file installed on the system controls these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative. Use the **display system-parameters customer-options** command to determine these values. On **Page 1**, verify that the **Maximum Off-PBX Telephones** allowed in the system is sufficient. One OPS station is required per iD808 device.

```
display system-parameters customer-options                     Page   1 of  10
                                OPTIONAL FEATURES

     G3 Version: V15                             Software Package: Standard
       Location: 2                            RFA System ID (SID): 1
       Platform: 6                            RFA Module ID (MID): 1

                                                                USED
                                Platform Maximum Ports: 48000 282
                                      Maximum Stations: 36000 48
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 200   0
                    Maximum Off-PBX Telephones -   OPS: 200   18
                    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 **Maximum Administered SIP Trunks** supported by the system is sufficient.

```
display system-parameters customer-options                        Page   2 of  10
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 200     0
           Maximum Concurrently Registered IP Stations: 18000 1
             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 Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 300   138
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 100   0
                             Maximum TN2501 VAL Boards: 128   0
                     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. On **Page 17,** set **Whisper Page Tone Given To: all**

```
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
```

On **Page 18,** make sure **Directed Call Pickup** is set to **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? n
     Temporary Bridged Appearance on Call Pickup? y     Directed Call Pickup? y
                      Extended Group Call Pickup: none
                   Enhanced Call Pickup Alerting? n


                        Display Information With Bridged Call? n
  Keep Bridged Information on Multiline Displays During Calls? y
                  PIN Checking for Private Calls? n
```

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 four digits long and begin with **1**, FNEs are also four digits beginning with **1**, and the FACs have formats as indicated with a **Call Type** of **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    ext    7            4    ext
     1             4    ext    88           4    ext
     2             4    udp    89           4    ext
     3005          8    udp    9            1    fac
     3015          9    udp    *            3    fac
     31            4    udp    #            3    fac
     33            4    udp
     37            4    udp
     38            5    aar
     4             1    fac
     5             3    dac
     6             3    fac
     61            4    ext
     66            4    ext
     663           4    ext
```

4.4. Define Feature Access Codes (FACs)

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

```
change feature-access-codes                                     Page   1 of   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:
                      Answer Back Access Code: *24
                        Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 4
    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:
          Message Sequence Trace (MST) Disable:

 PASTE (Display PBX data on Phone) Access Code:
  Personal Station Access (PSA) Associate Code:        Dissociate Code:
        Per Call CPN Blocking Code Access Code: *34
      Per Call CPN Unblocking Code Access Code: *35
                    Posted Messages Activation:        Deactivation:
                  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 OPS FNEs can be defined using the **change off-pbx-telephone feature-name-extensions** command. The following screens show in bold the FNEs defined for use with the sample configuration.

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

        Active Appearance Select: 1700
             Automatic Call Back: 1701
      Automatic Call-Back Cancel: 1702
                Call Forward All: 1703
     Call Forward Busy/No Answer: 1704
             Call Forward Cancel: 1705
                       Call Park: 1706
           Call Park Answer Back: 1707
                    Call Pick-Up: 1708
            Calling Number Block: 1709
          Calling Number Unblock: 1710
  Conditional Call Extend Enable: 1711
 Conditional Call Extend Disable: 1712
             Conference Complete: 1713
            Conference on Answer: 1714
           Directed Call Pick-Up: 1715
           Drop Last Added Party: 1716
```

```
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): 1717
      Extended Group Call Pickup:
          Held Appearance Select: 1718
          Idle Appearance Select: 1719
              Last Number Dialed: 1720
            Malicious Call Trace:
     Malicious Call Trace Cancel:
             Off-Pbx Call Enable:
            Off-Pbx Call Disable:
                   Priority Call: 1725
                          Recall: 1726
                  Send All Calls: 1727
           Send All Calls Cancel: 1728
               Transfer Complete: 1729
             Transfer On Hang-Up: 1730
          Transfer to Voice Mail: 1731
         Whisper Page Activation: 1732
```

4.6. Configure Class of Service (COS)

Use the **change cos** command to set the appropriate service permissions to support OPS features (shown in bold). For the sample configuration, 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 turret, not covered in testing.

```
change cos                                                      Page   1 of   2

                                0  1  2  3  4  5  6  7  8  9 10 11 12 13 14 15
 Auto Callback                  n  y  y  n  y  n  y  n  y  n  y  y  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            n  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Off-hook Alert                 n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Client Room                    n  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  n  y  y  y  y
 Call Forwarding Busy/DA        n  y  n  n  n  n  n  n  n  n  n  y  n  n  n  n
 Personal Station Access (PSA)  n  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Extended Forwarding All        n  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Extended Forwarding B/DA       n  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Trk-to-Trk Transfer Override   n  y  n  n  n  n  n  n  n  n  n  y  n  n  n  n
 QSIG Call Offer Originations   n  n  n  n  n  n  n  n  n  n  n  y  n  n  n  n
 Contact Closure Activation     n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
```

4.7. Configure Class of Restriction (COR)

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

```
change cor 1                                                    Page   1 of  23
                              CLASS OF RESTRICTION

               COR Number: 1
          COR Description: Default

                      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? n
     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
         Send ANI for MFE? n
            MF ANI Prefix:              Automatic Charge Display? n
Hear System Music on Hold? y   PASTE (Display PBX Data on Phone)? y
                        Can Be Picked Up By Directed Call Pickup? y
                                    Can Use Directed Call Pickup? y
                                    Group Controlled Restriction: inactive
```

4.8. Add Coverage Path

Use the **add coverage path n** command where **n** is the number of the coverage path to be added. Configure **Point 1** in the coverage path to one used to the voice messaging hunt group, which is group **h89** 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 89                                         Page   1 of   1
                                 COVERAGE PATH

                   Coverage Path Number: 89
     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: h89            Rng:    Point2:
  Point3:                        Point4:
  Point5:                        Point6:
```

4.9. Add Stations

The Speakerbus iD808 *i* turret requires up to three stations for each device. The first station is referred to as the main appearance. The second and third stations are referred to as the privacy handsets. The privacy handsets are needed when privacy is required. If the privacy feature is not needed, then only the first station is required.

4.9.1. Main Appearance Station

Use the **add station** command to add a station for each *i* turret to be supported. To configure the main appearance, on **Page 1** use **9630** for the station **Type** and include the **Coverage Path** for voice messaging, if applicable. Use the **COS** and **COR** values administered in **Sections 4.6** and **4.7**. Enter a descriptive name in the **Name** field. Use the default values for the all other fields.

```
add station 1301                                                 Page   1 of   5
                                     STATION

Extension: 1301                          Lock Messages? n               BCC: 0
     Type: 9630                          Security Code: 123456           TN: 1
     Port: S00010                      Coverage Path 1: 89              COR: 1
     Name: iTurret 1                   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: 1301
            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
```

On **Page 2**, 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. Set the **MWI Served User Type** field to the appropriate value to allow message waiting indication to be sent to the *i* turret. Use the default values for the all other fields.

Note: By default, the **Restrict Last Appearance** field to is set to **y** to reserve the last call appearance for outgoing calls from the *i* turret, this should not be altered.

```
add station 1301                                                 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? y
       Service Link Mode: as-needed                      EC500 State: enabled
         Multimedia Mode: enhanced
    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: 1301         Always Use? n IP Audio Hairpinning? n
     Precedence Call Waiting? y
```

On **Page 4** under the heading **BUTTON ASSIGNMENTS**, fill in the number of call appearances that are to be supported for *i* turret. In this example, the first station for *i* turret was configured with four call appearances. Locally, *i* turret will actually be configured with 3 call appearances since the last appearance is restricted as configured on **Page 2**. Multiple bridged line appearances are configured for this example station. Button assignments **5** and **6** relate to the second and third stations corresponding to two stations that will be used as the privacy handsets at *i* turret.
Note: These stations are configured in **Section 4.9.2** and these bridged appearance buttons cannot be configured until those stations have been added. If privacy is not needed for *i* turret, then these bridged appearances are not required.

```
add station 1301                                                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:1501
 2: call-appr                          6: brdg-appr  B:1  E:1502
 3: call-appr                          7: brdg-appr  B:1  E:1303
 4: call-appr                          8: brdg-appr  B:2  E:1303

    voice-mail Number:
```

Continue on **Page 5** under the **BUTTON ASSIGNMENTS** section, enter the function button names (shown in bold) for OPS FNEs that will be used at *i* turret. Configure function buttons **call-fwd, cfwd-bsyda** and if required, **auto-cback** and **no-hld-cnf.**

```
add station 1301                                                Page   5 of   5
                                     STATION

BUTTON ASSIGNMENTS

 9: brdg-appr  B:3  E:1303
10: brdg-appr  B:1  E:1305
11: brdg-appr  B:2  E:1305
12: brdg-appr  B:3  E:1305
13: auto-cback
14: no-hld-cnf
15: cfwd-bsyda Ext:
16: call-fwd   Ext:
17:
18:
19:
20:
```

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

4.9.2. Privacy Handset Stations

Use the **add station** command to add a station for each privacy handset. On **Page 1,** use **9630** for the station **Type**. A coverage path is not required for this station. Use the **COS** and **COR** values administered in **Sections 4.6** and **4.7**. Enter a descriptive name in the **Name** field. Use the default values for the all other fields.

```
add station 1501                                                  Page   1 of   5
                                    STATION

Extension: 1501                         Lock Messages? n               BCC: 0
     Type: 9630                         Security Code:                  TN: 1
     Port: S00013                     Coverage Path 1:                 COR: 1
     Name: HS1 of 1301                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: 1501
            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

                                                         IP Video? n


                                             Customizable Labels? y

                                             Customizable Labels? y
```

On **Page 2,** the **Bridged Call Alerting** field should be set to **y**.

```
add station 1501                                                   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
     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: 1501          Always Use? n IP Audio Hairpinning? n
     Precedence Call Waiting? y
```

On **Page 4** of the first Privacy Handset station, one call appearance should be configured along with a feature button for the **exclusion** feature (required for privacy), and bridged appearances for each call appearance of the first station (main appearance) all shown in bold below.

```
add station 1501                                                   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:1301
 2: exclusion                           6: brdg-appr  B:1  E:1303
 3: brdg-appr  B:1  E:1301              7: brdg-appr  B:2  E:1303
 4: brdg-appr  B:2  E:1301              8: brdg-appr  B:3  E:1303

    voice-mail Number:
```

Below is the configuration of the third station for handset 2. Use the **add station** command to add a station for each privacy handset. On **Page 1** use **9630** for the station **Type**. A coverage path is not required for this station. Use the **COS** and **COR** values administered in **Sections 4.6** and **4.7**. Enter a descriptive name in the **Name** field. Use the default values for the all other fields.

```
add station 1502                                             Page   1 of   5
                                     STATION

Extension: 1502                         Lock Messages? n               BCC: 0
     Type: 9630                         Security Code:                  TN: 1
     Port: S00014                     Coverage Path 1:                 COR: 1
     Name: HS2 of 1301                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: 1502
            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

                                                         IP Video? n


                                              Customizable Labels? y
```

On **Page 2,** the **Bridged Call Alerting** field should be set to **y**.

```
Add station 1502                                             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
     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: 1502         Always Use? n IP Audio Hairpinning? n
     Precedence Call Waiting? y
```

On **Page 4** of the second privacy handset station, one call appearance should be configured along with a feature button for the **exclusion** feature (required for privacy), and bridged appearances for each call appearance of the first station (main appearance) all shown in bold below.

```
add station 1502                                           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:1301
 2: exclusion                           6: brdg-appr  B:1  E:1303
 3: brdg-appr  B:1  E:1301              7: brdg-appr  B:2  E:1303
 4: brdg-appr  B:2  E:1301              8: brdg-appr  B:3  E:1303

    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.

4.10. Administer Off PBX Station Mapping

Use the **change off-pbx-telephone station-mapping** command to map the Communication Manager extensions (1301, 1501, and 1502) to the extension defined on SIP Enablement Services for the corresponding SIP user. (See **Section 5.5**). 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 4.11**. The **Configuration Set** value can reference a set that has the default settings.

```
change off-pbx-telephone station-mapping 1301                   Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                  Prefix                     Selection   Set     Mode
 1301              OPS          -      1301             6          1
 1501              OPS          -      1501             6          1
 1502              OPS          -      1502             6          1
```

On **Page 2**, change the **Call Limit** to match the number of call appearances on 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 1301                   Page   2 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Appl    Call        Mapping    Calls       Bridged       Location
 Extension      Name    Limit       Mode       Allowed     Calls
 1301           OPS      10        both         all         none
 1501           OPS      10        both         all         none
 1502           OPS      10        both         all         none
```

4.11. 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 active SIP Enablement Services IP address. The host names will be used throughout the other configuration screens of Communication Manager.

```
change node-names ip
                                  IP NODE NAMES
    Name              IP Address
CLAN1               10.10.16.23
Gateway             10.10.16.1
MedPro1             10.10.16.24
SM100               10.10.16.11
default             0.0.0.0
procr               0.0.0.0
sesactive           10.10.16.5
```

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 **sip.avaya.com**. By default, **Intra-region IP-IP Direct Audio** and **Inter-region 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. This codec set will be used for calls routed over the SIP trunk to SIP Enablement Services as **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: 1       Authoritative Domain: sip.avaya.com
    Name: Default 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? y
   UDP Port Max: 8001
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      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: 6
        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 codecs supported for calls routed over the SIP trunk to *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. Multiple codecs may be specified in the **IP Codec Set** form in order of preference; the example below includes **G.711A** (a-law), **G.711MU** (mu-law) and **G.729**, which are supported by the iD808 deskstations.

```
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.711A            n         2        20
 2: G.711MU           n         2        20
 3: G.729             n         2        20
 4:
 5:
 6:
 7:

     Media Encryption
 1: none
 2:
 3:
```

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.**
- Set the **Transport Method** to the desired transport method; **tcp** (transport control protocol) or tls (Transport Layer Security). **Note:** for transparency tcp was used during this compliance test but the recommended method is tls.
- Specify the node names for the C-LAN board in the G650 Media Gateway and the active SIP Enablement Services node name as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These values are taken from the **IP Node Names** form shown above.
- Ensure that the recommended port value of **5060** for tcp is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields **Note**: If tls is used then the recommended port value is 5061.
- 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 **sip.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 6                                        Page   1 of   1
                                SIGNALING GROUP

 Group Number: 6              Group Type: sip
                        Transport Method: tcp
  IMS Enabled? n
     IP Video? n




   Near-end Node Name: CLAN1                Far-end Node Name: sesactive
 Near-end Listen Port: 5060               Far-end Listen Port: 5060
                                       Far-end Network Region: 1
Far-end Domain: sip.avaya.com

                                          Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate               RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload          Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                   IP Audio Hairpinning? y
         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 *i* turret deskstations. Enter a descriptive name in the **Group Name** field. Set the **Group Type** field to **sip**. Enter a **TAC** code compatible with the Communication Manager dial plan. 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. Accept the default values for the remaining fields.

```
add trunk-group 6                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 6                   Group Type: sip           CDR Reports: y
  Group Name: SES OPS                   COR: 1        TN: 1        TAC: 506
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n

                                                      Signaling Group: 6
                                                    Number of Members: 30
```

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 6                                           Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                       Maintenance Tests? y

                     Numbering Format: public
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? y
                                                Replace Unavailable Numbers? y

 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 4-digit extension beginning with **1** and whose calls are routed over SIP trunk group **6** have the number sent to the far-end for display purposes.

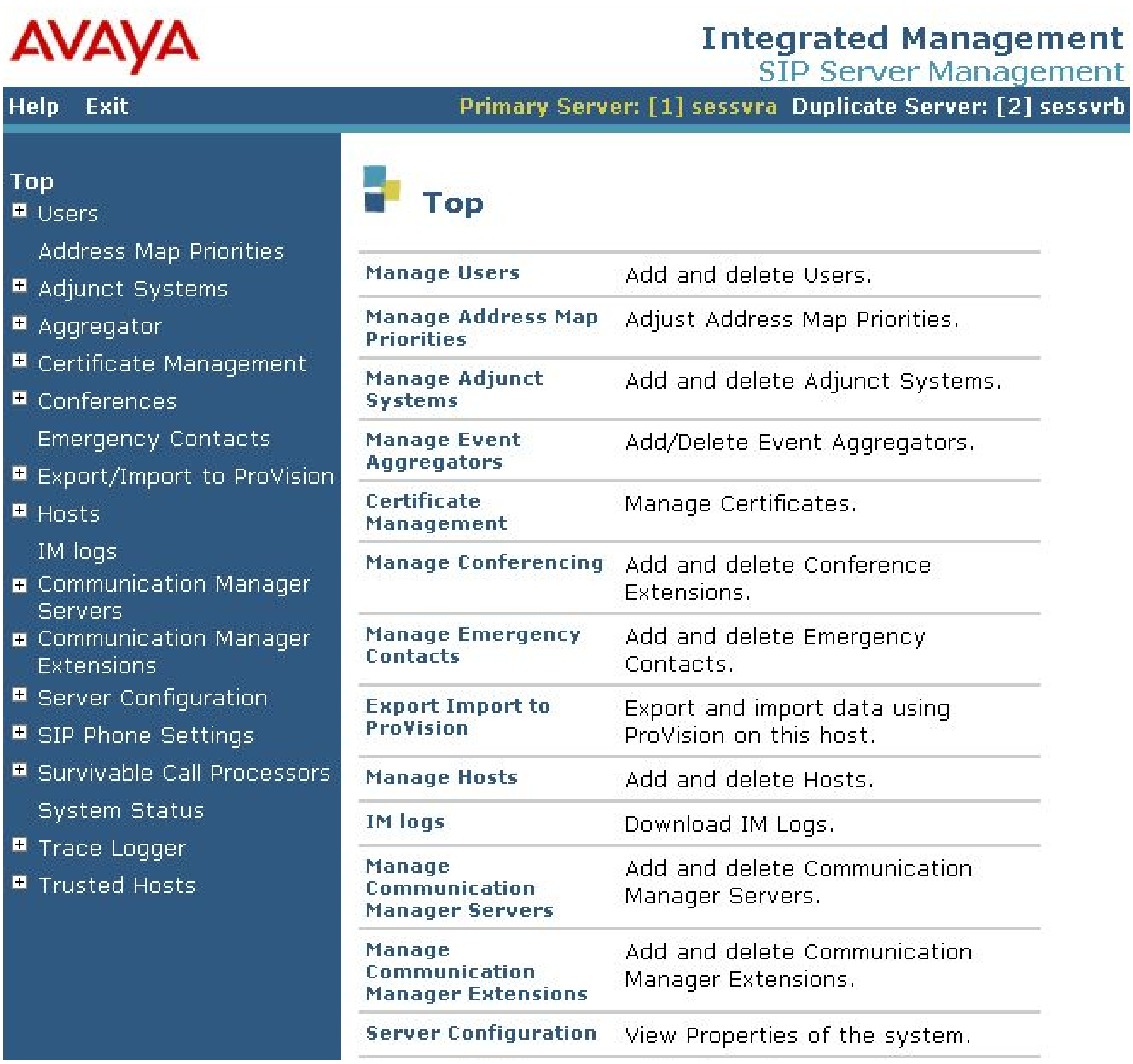
```
change public-unknown-numbering 0                              Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext           Trk       CPN             CPN
Len Code          Grp(s)    Prefix          Len
                                                     Total Administered: 1
 4  1              6                          4         Maximum Entries: 9999
```

5. Configure Avaya Aura™ SIP Enablement Services

This section covers the administration of SIP Enablement Services. SIP Enablement Services is configured via an Internet browser using the Administration web interface. It is assumed that SIP Enablement Services software and the license file have already been installed. For additional information on installation tasks refer to **[4]**.

5.1. Logging in to Avaya Aura™ SIP Enablement Services

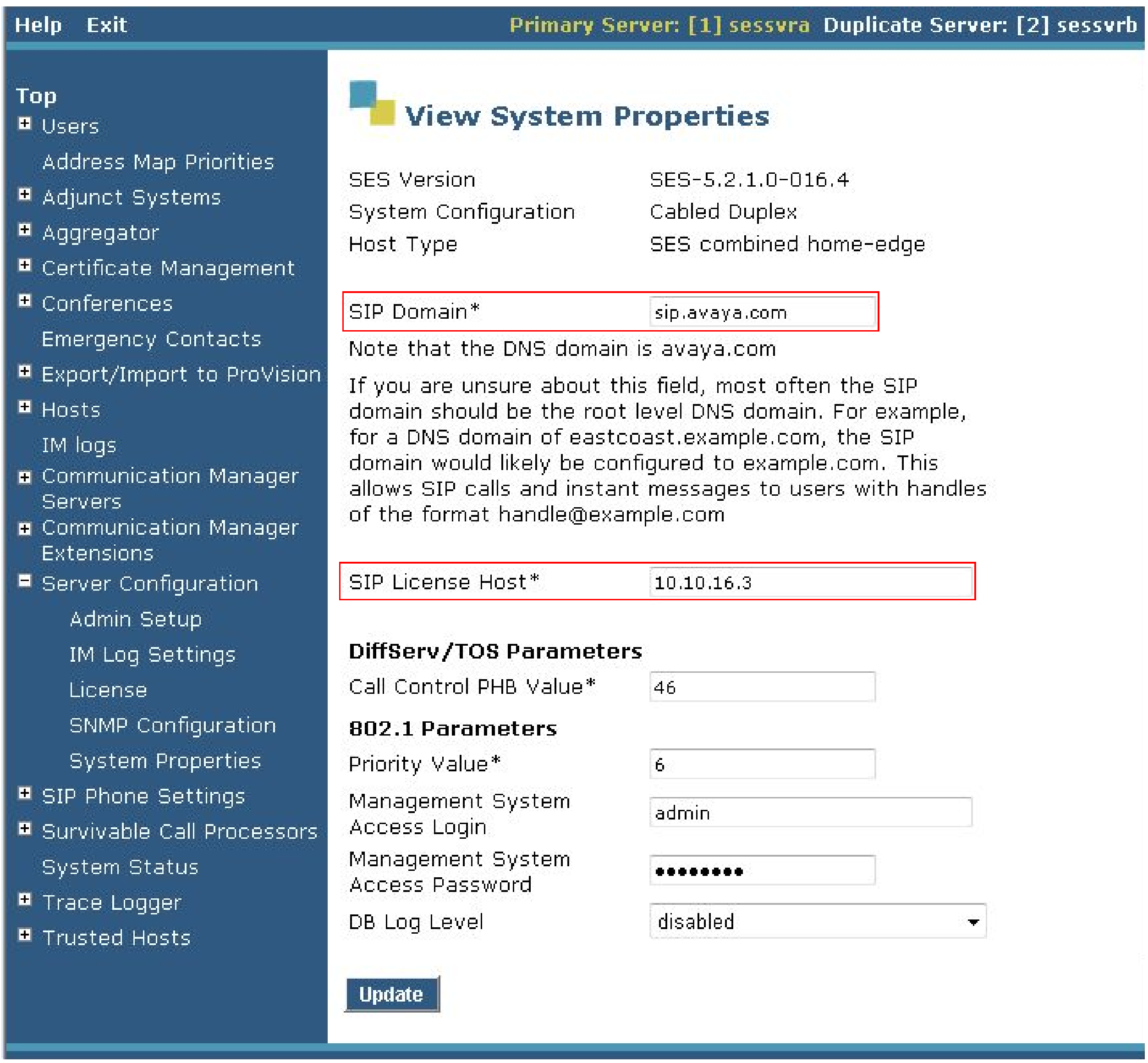
To access the administration web interface, enter **http://<ip-addr>/admin** as the URL in an Internet browser, where <ip-addr> is the active IP address of SIP Enablement Services. Log in with the appropriate credentials and then select the **Administration → SIP Enablement Services** (not shown). The main screen is displayed, as shown below.



5.2. Verify System Properties

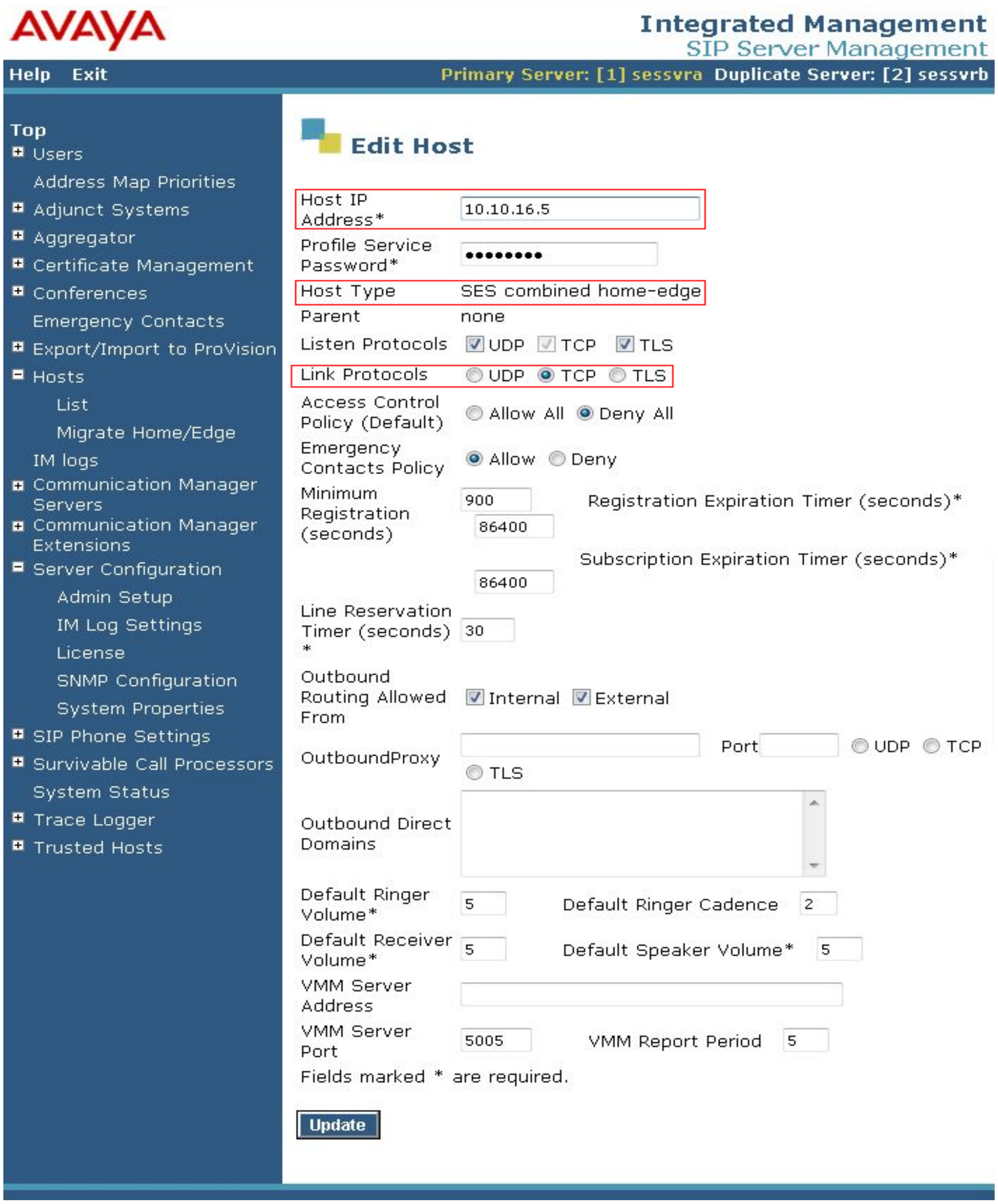
From the left pane of the Administration web interface, expand the **Server Configuration** option and select **System Properties**. In the **System Properties** screen, enter the **SIP Domain** name assigned to the Avaya SIP-based network. For the **SIP License Host** field, enter the fully qualified domain name or the IP address of the local host unless the WebLM server is not co-resident with this server. In the example screen below the IP address for SES server side A is displayed in the **SIP License Host** field.

Note: Separate licenses are needed for each SIP Enablement Services server. After configuring the **System Properties** screen, click the **Update** button.



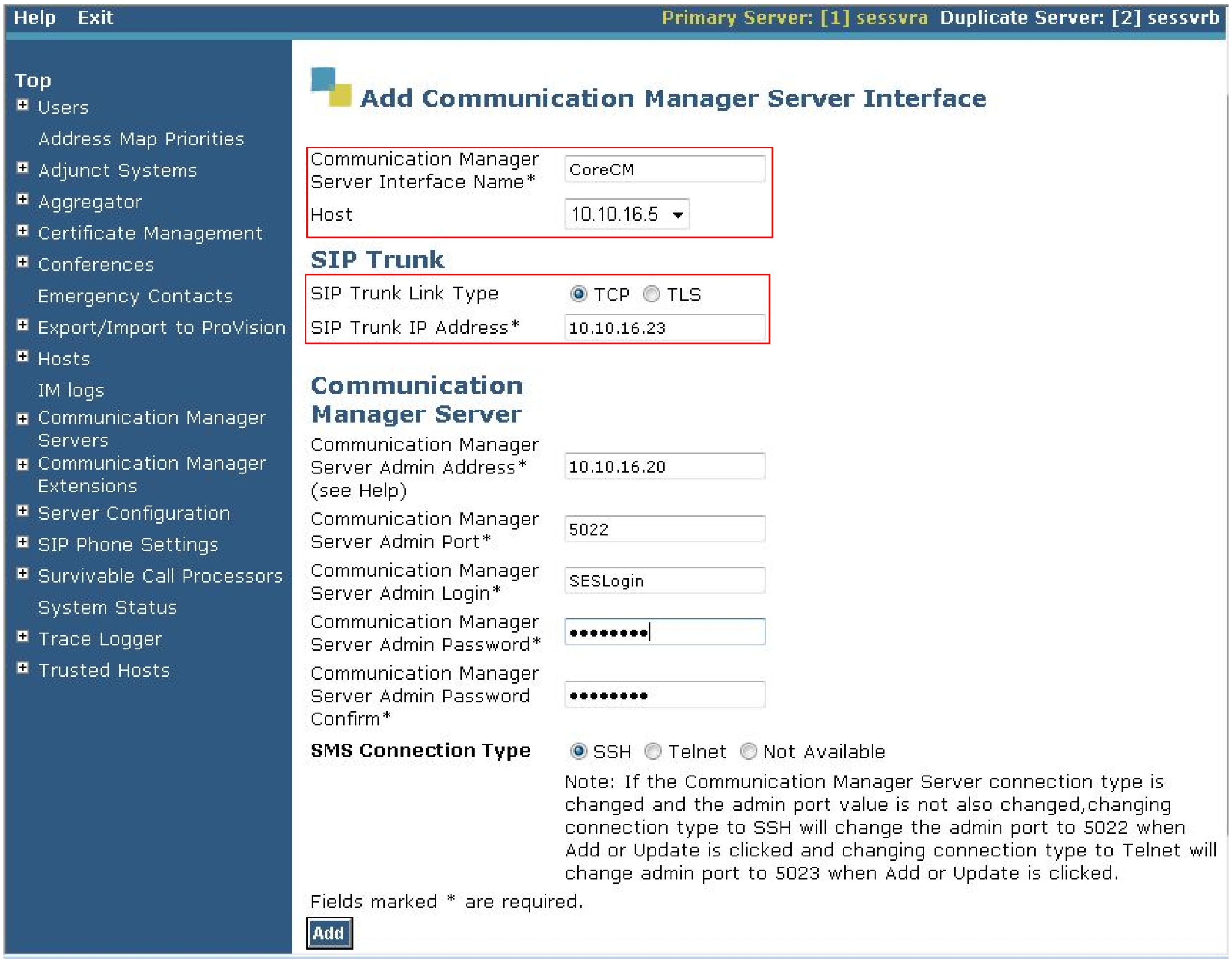
5.3. Create a Host

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 active 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, both servers in the redundant pair were configured as an **SES combined home/edge** during the initial setup. The **Link Protocols** selected defaults to TLS but in this example **TCP** was used. The default values for the other fields may be used as shown below.



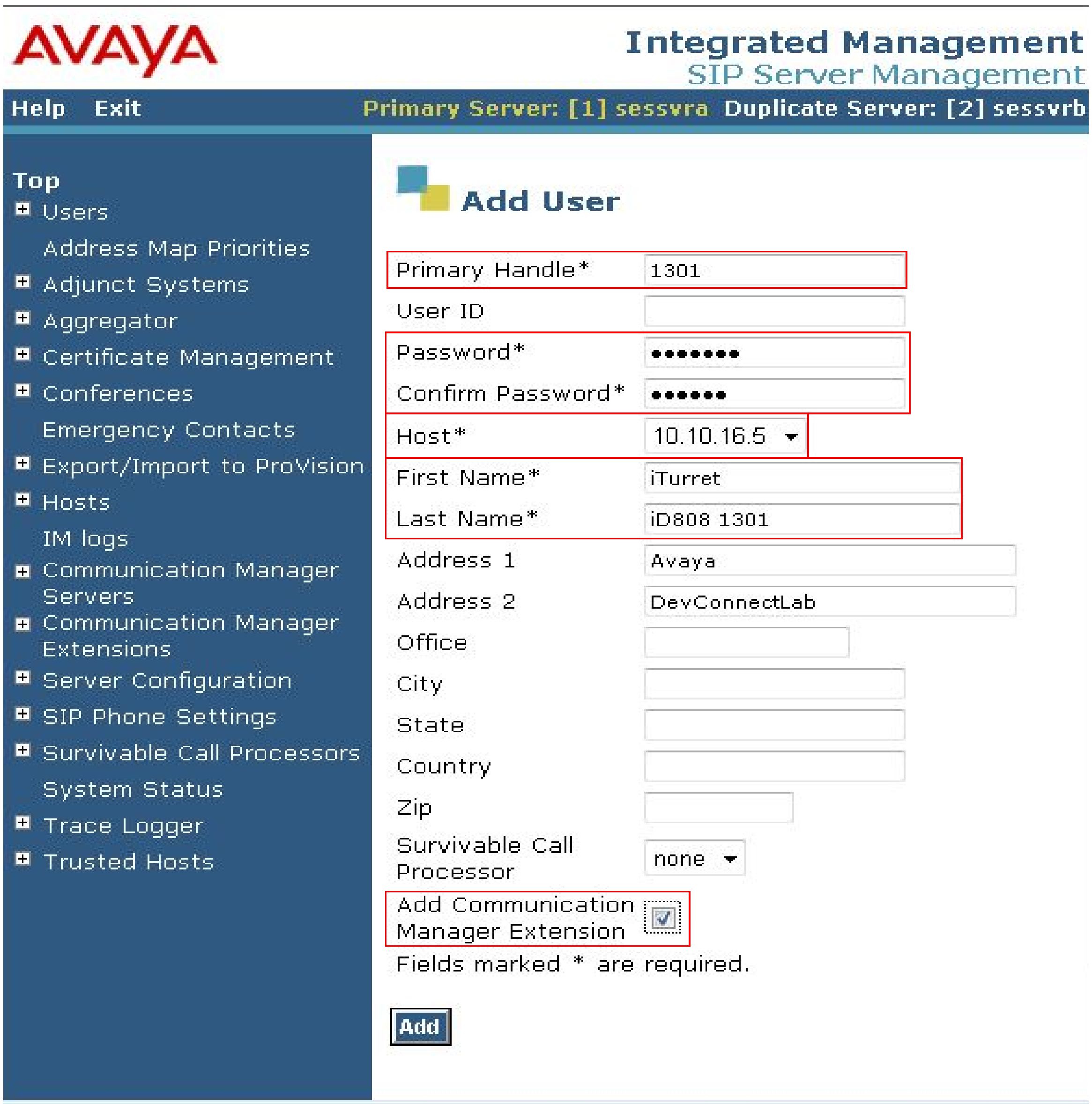
5.4. Add Avaya Aura™ Communication Manager Interface

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. In this screen, enter a descriptive name in the **Communication Manager Server Interface Name** field and select the home server from the drop down menu in the **Host** field. Select TCP for the **Link Type** and enter the IP address of the C-LAN board in the Avaya G650 Media gateway in the **SIP Trunk IP Address** field. Refer to **[4]** for additional information on configuring the remaining fields.

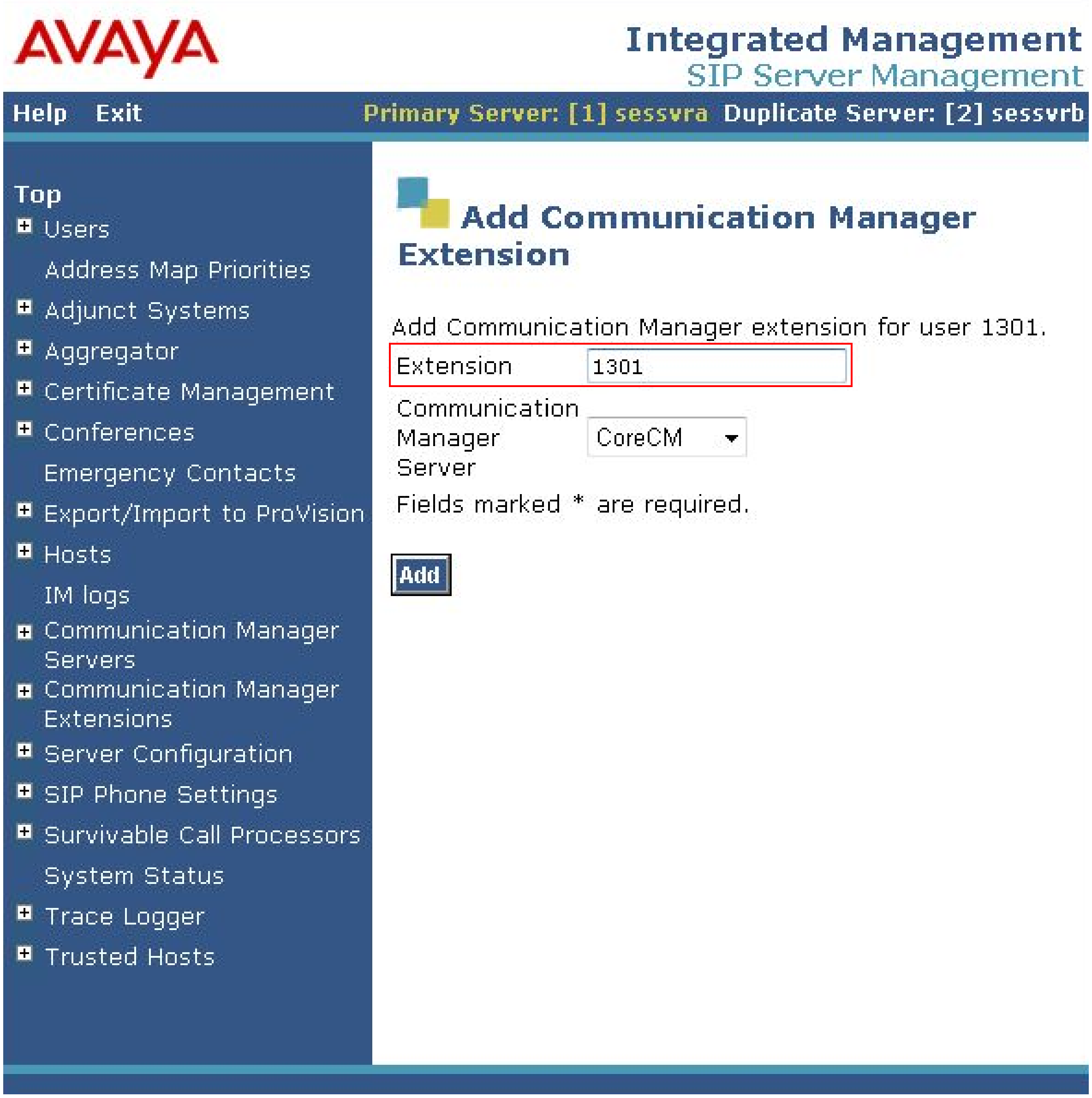


5.5. Add User

Three users are required for each Speakerbus iD808 *i* turret registering with SIP Enablement Services, one for the main appearance and two for the handset appearances. The handset appearances are required to support privacy with Communication Manager. The procedure to add all three users is the same. 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 (*sip.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 Communication Manager over the SIP trunk. Click the **Add** button to commit entries.



The **Add Communication Manager Extension** screen is displayed. In the **Add Communication Manager Extension** screen, enter the **Extension** configured in 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. The *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 an *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 User Permissions
- Assign Ownership (of Appearances to Users)
- Assign Default Call Appearances
- Programming iD808 Deskstations
- Assign Appearances to Deskstation Keys
- Assign Bridged Call Appearances to Deskstations
- Synchronise Deskstations
- Feature Name Extensions (FNEs)

6.1. Launch *i* manager

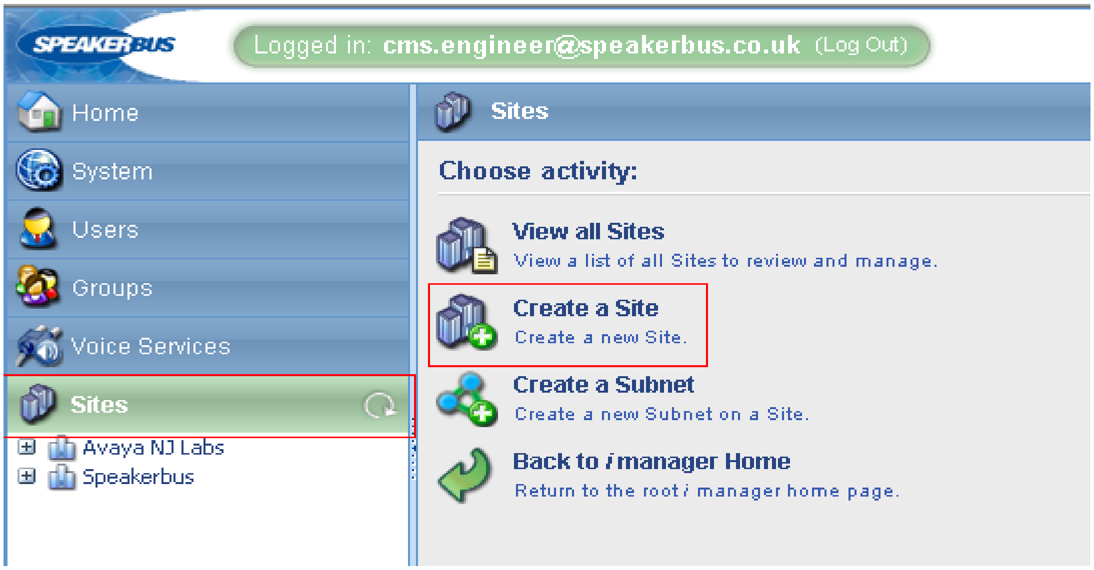
To access the *i* manager software interface, open a web browser and type the *i* manager web address, for example, http://10.10.16.50/imanager. Press the **Enter** key. At *i* manager logon page enter the appropriate credentials. The *i* manager home page is displayed as shown below.

6.2. Verify Product Key

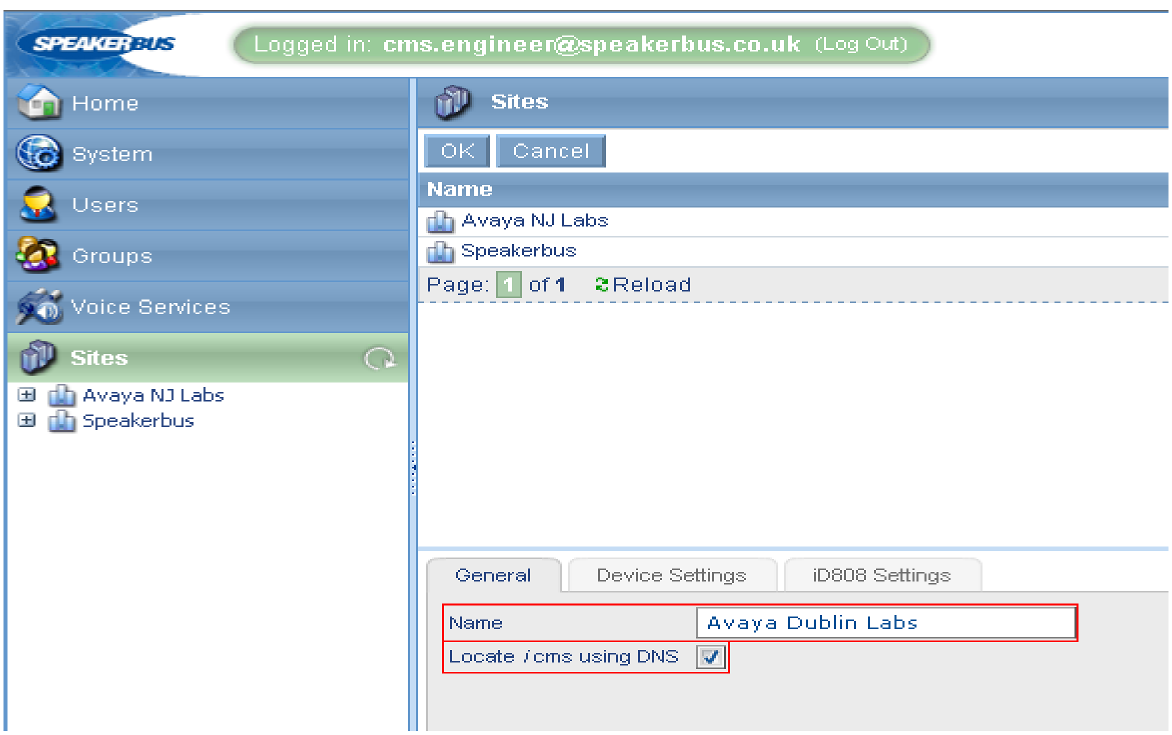
In the left pane, navigate to **System → Product Key** to verify that a valid key is installed and sufficient devices are allowed.

6.3. Create Site

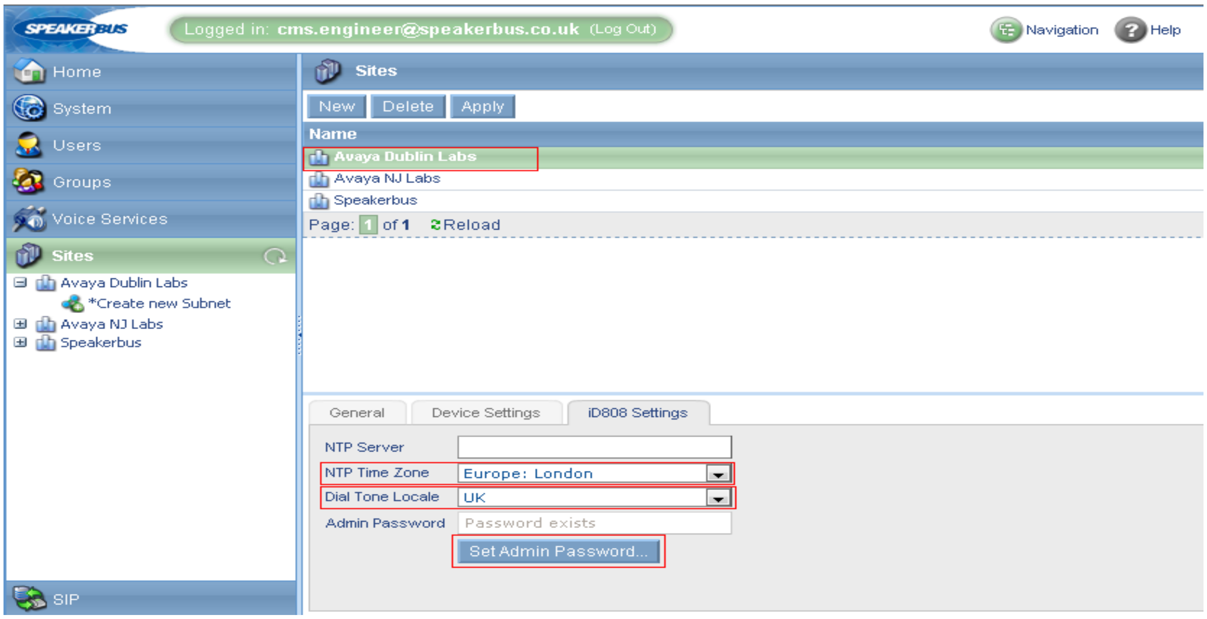
Configure a site representing the location where the Speakerbus iD808 devices are installed. Click **Sites** in the left pane, click on **Create a Site** in the right pane. The **Sites** page is displayed.



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, *i* turret will use the DNS server to locate *i* cms server IP address. Refer to **[5]** for correct configuration of DNS.

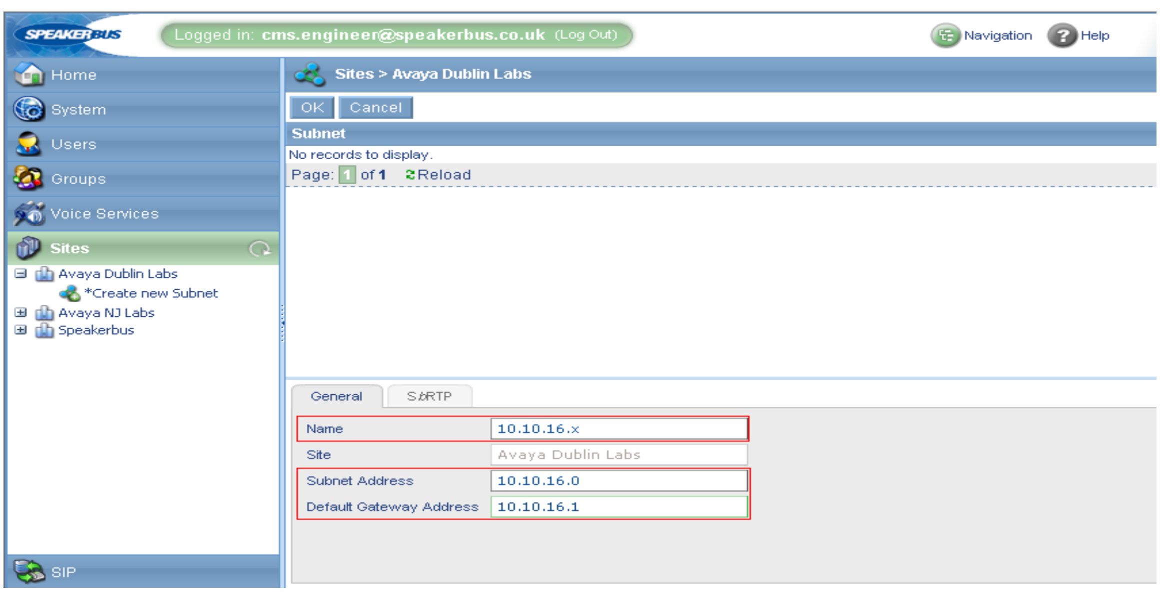


In the **iD808 Settings** tab, set the **NTP Time Zone** (network time protocol time zone) and configure the password for logging into the iD808 deskstation by clicking the **Set Admin Password** button. The NTP Server field may be set to the IP address of the NTP server if one is used. Click **Apply.** The site will be now listed under **Sites**.

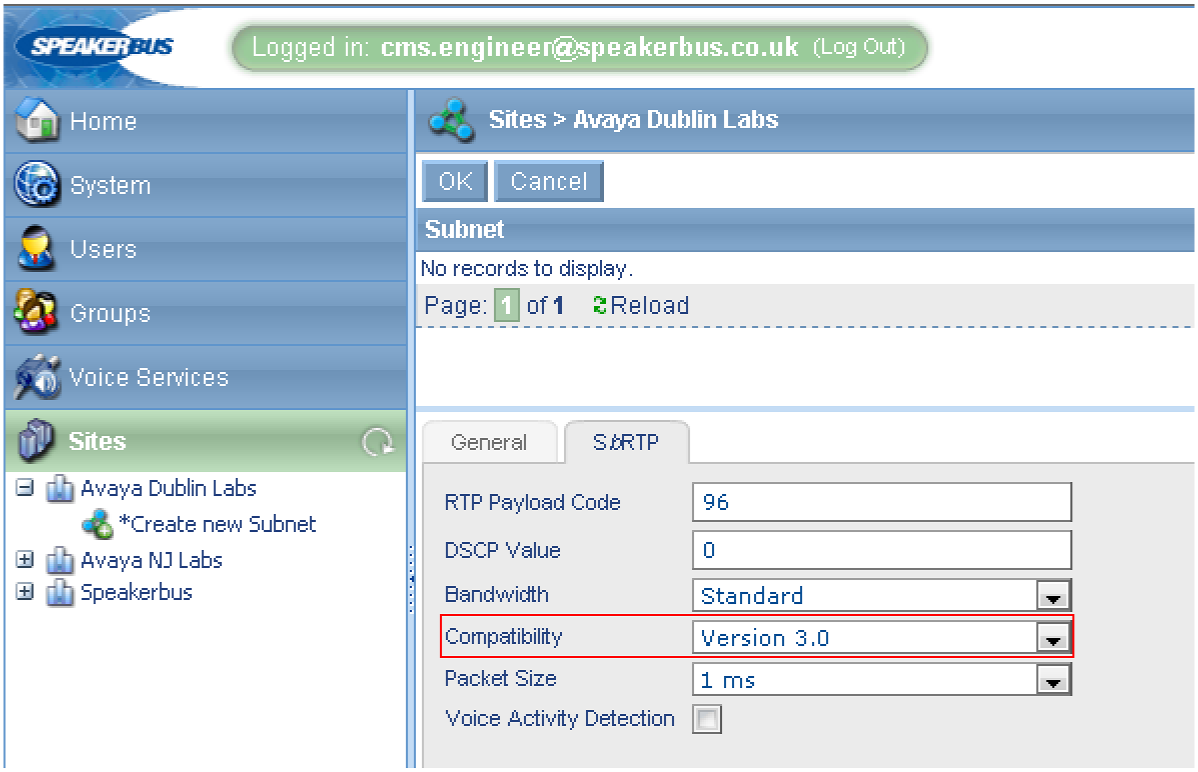


6.4. Create Subnet

To create a subnet, click on **Create new Subnet** under the newly configured **Avaya Dublin Labs** 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.

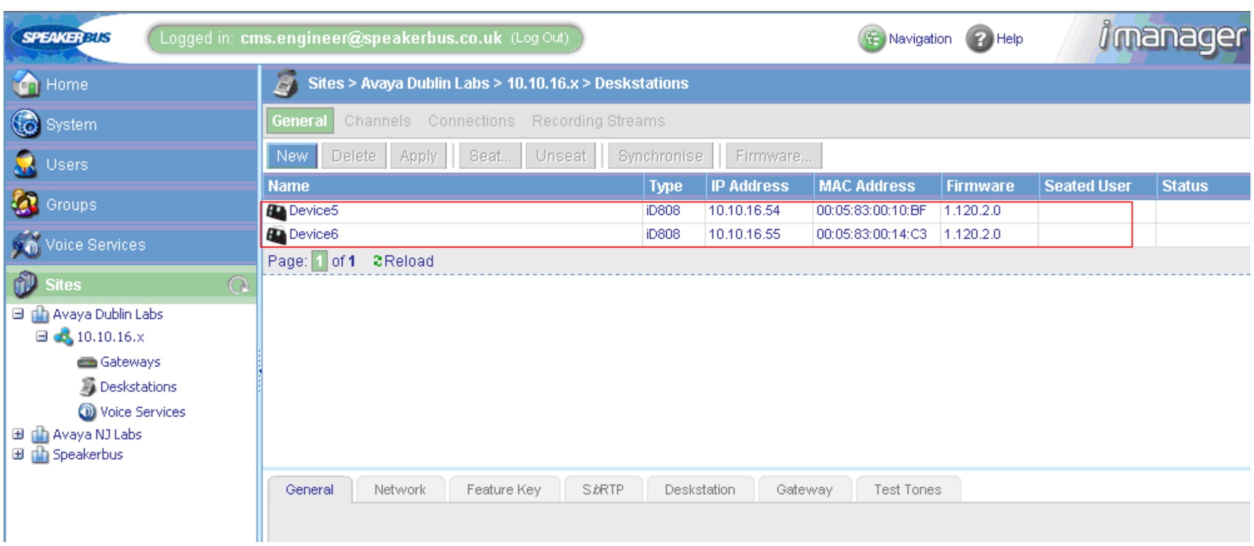


In the **SbRTP** tab, set the **Compatibility** field to **Version 3.0**. Click **OK**.

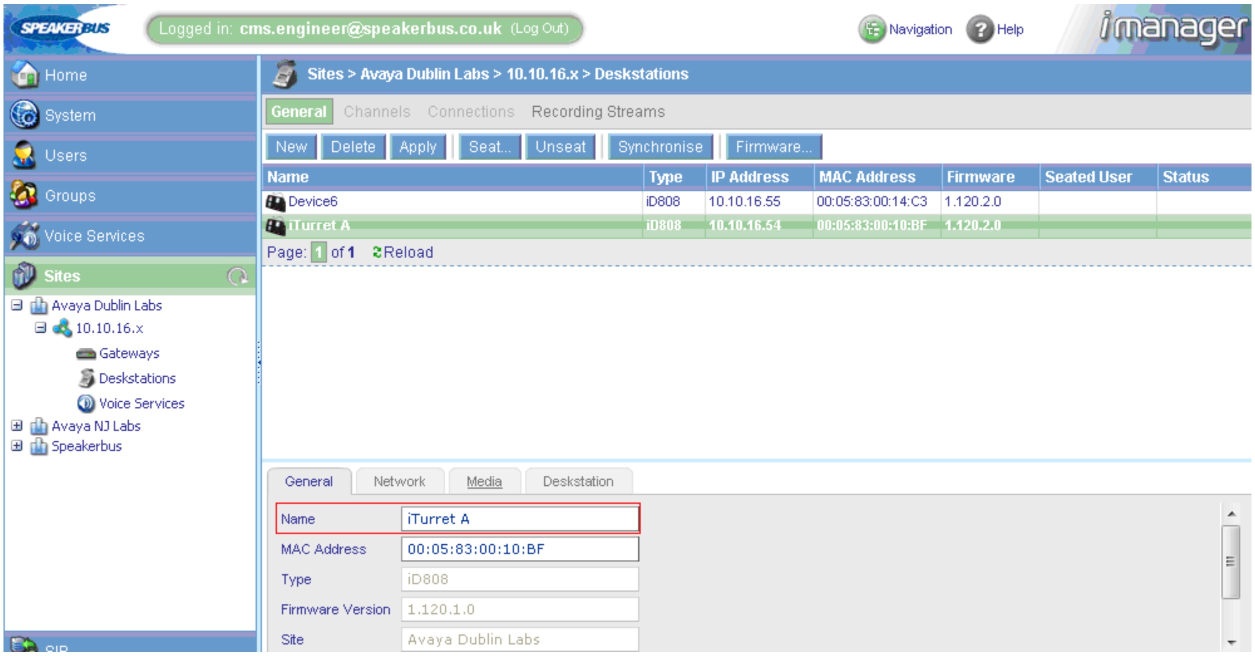


6.5. Create Deskstations

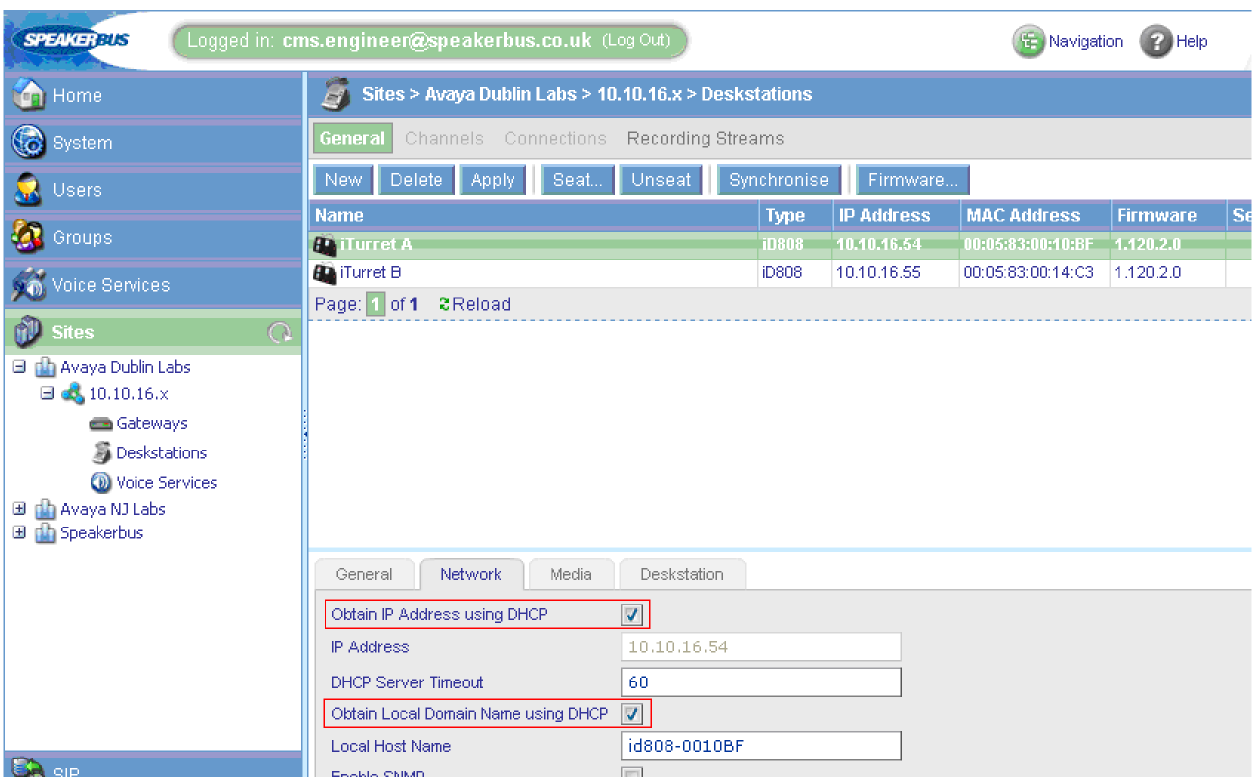
iD808 deskstations will automatically register to this subnet within *i* cms as the appropriate DHCP and DNS records were created prior to ID808 deskstations being connected to the IP network. The newly registered deskstations are automatically displayed in the list.



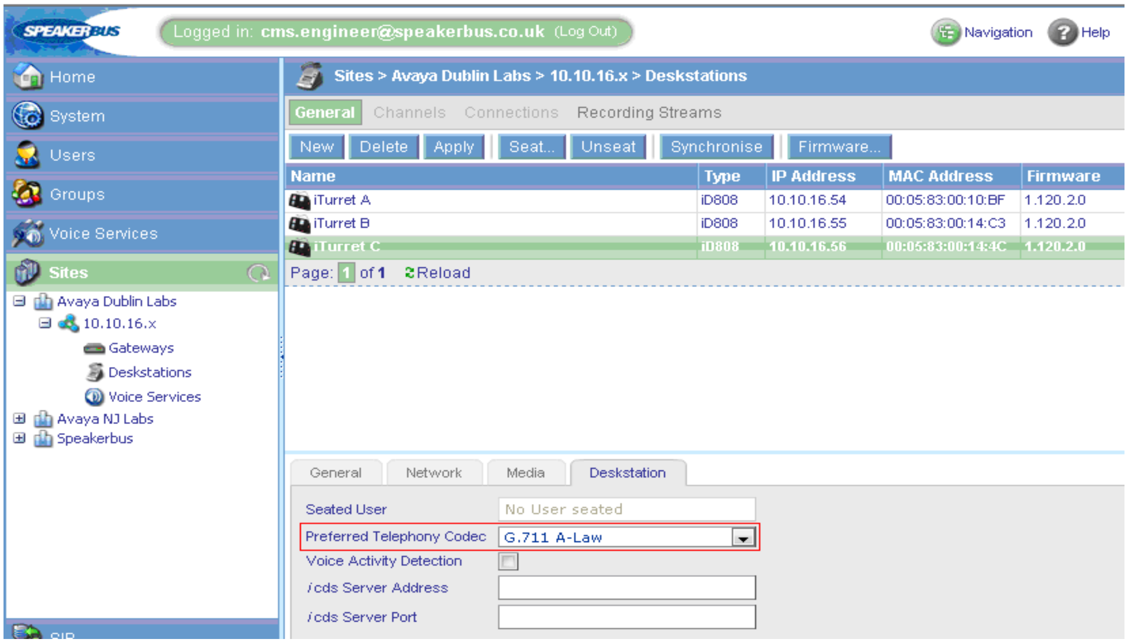
Select a device and change the name to a more descriptive one in the **General** tab.



In the **Network** tab, verify that the **Obtain IP Address using DHCP** and the **Obtain local Domain Name using DHCP** check boxes are selected.



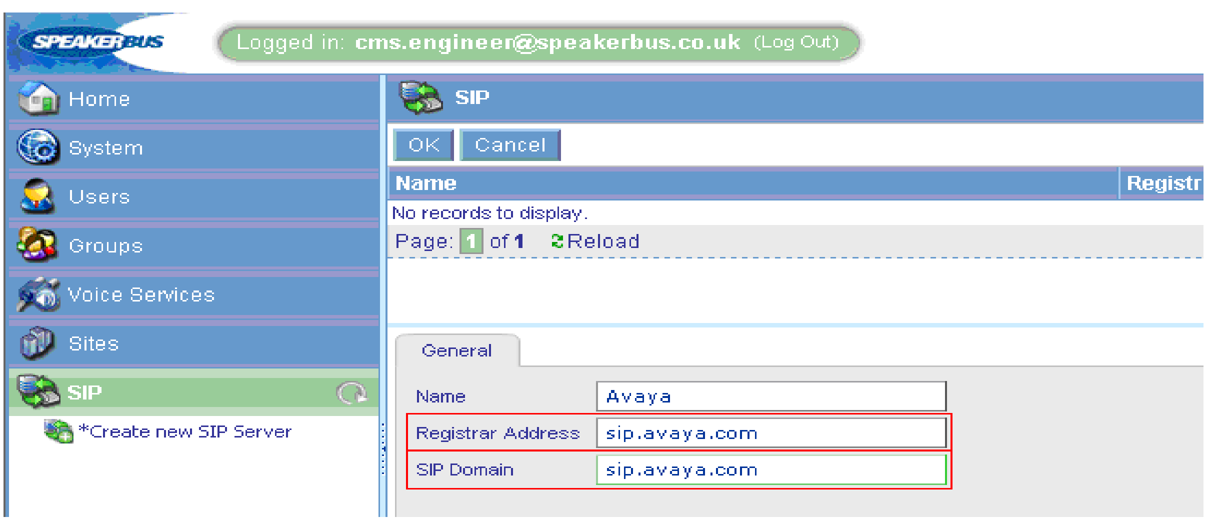
In the **Deskstation** tab, select a preferred codec. In this configuration, **G.711a-law** is the preferred codec. Click **Apply**. Repeat these steps for all deskstations.



6.6. Create SIP Server

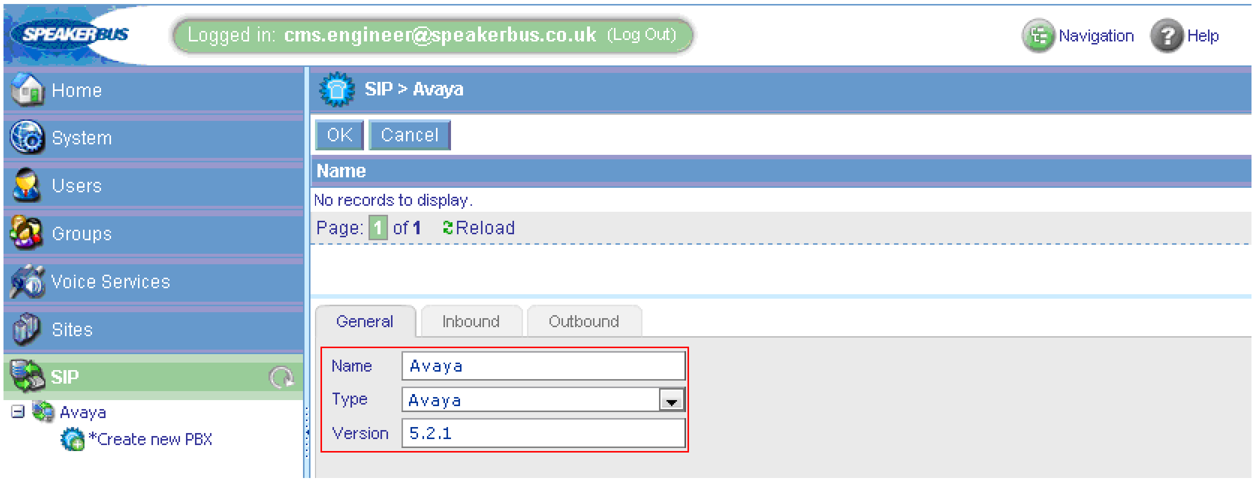
To create a SIP Server, click **Create a new SIP Server** under the **SIP** directory in the left pane. Provide a descriptive name for the SIP server and set the **Registrar Address** and **SIP Domain** fields to **sip.avaya.com**. In this configuration DNS resolves the domain name to 10.10.16.5, the SIP Enablement Services active 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 (Not shown).

Note: A server locater record (SRV) for the registrar address and SIP domain must be created on DNS. Refer to **[5]** for more details.



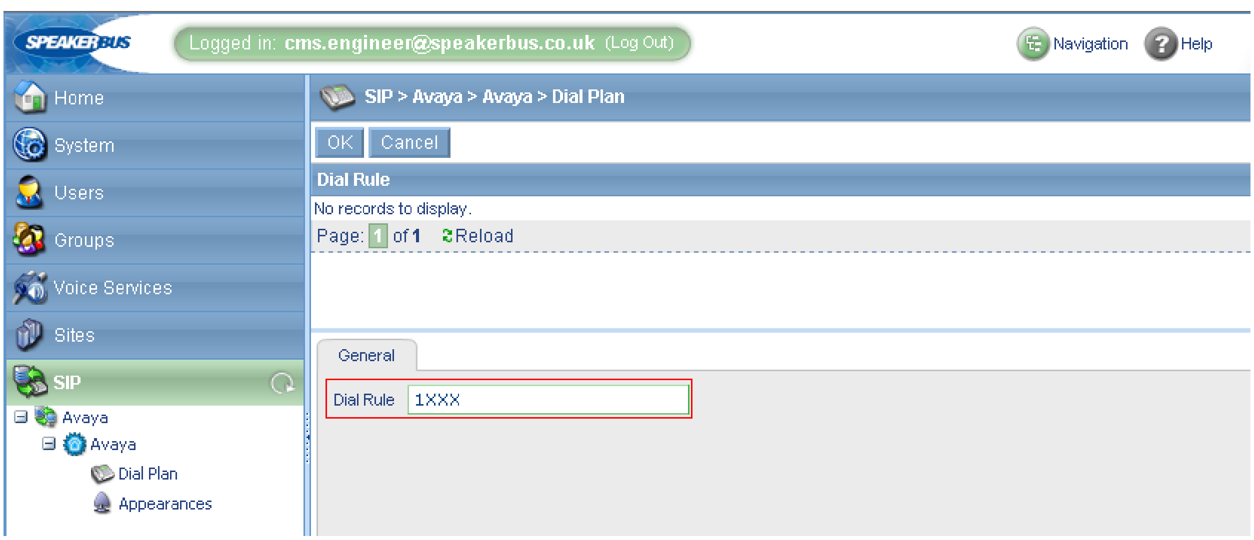
6.7. Create PBX

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**.



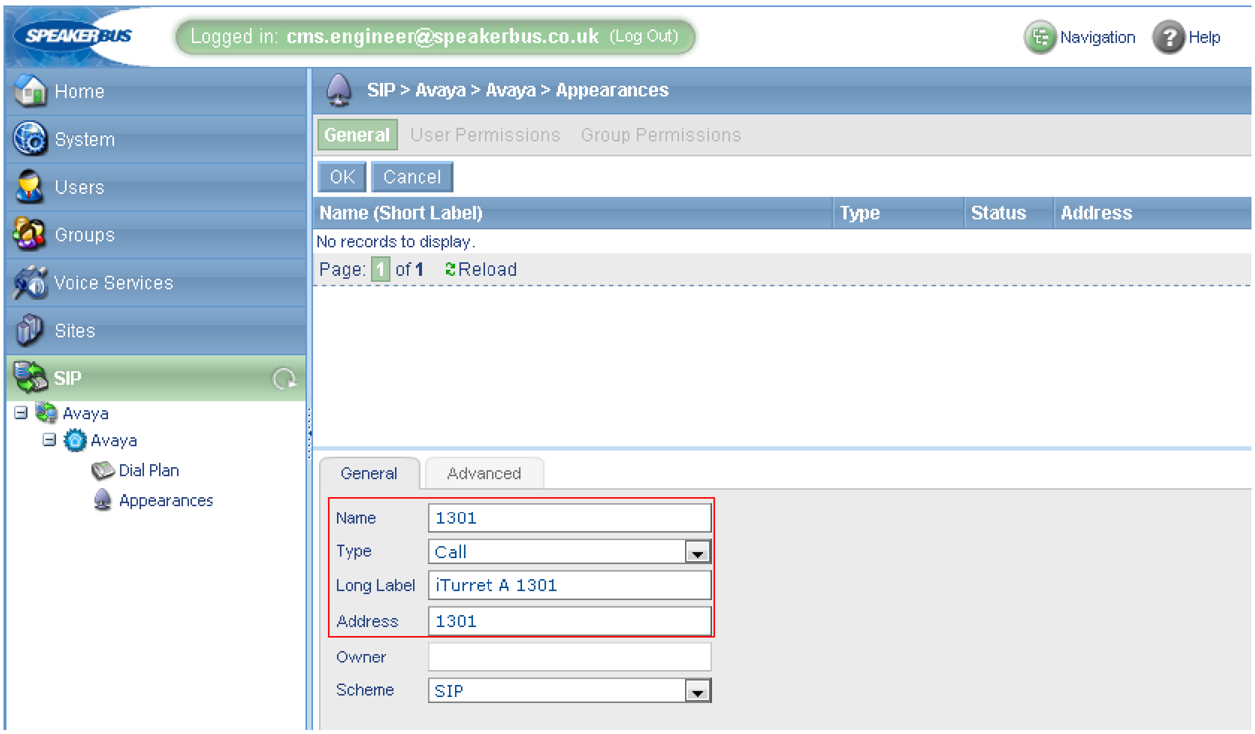
6.8. Create Dial Plan

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, 4-digit extensions beginning with **1** 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 4-digit extension beginning with **8**. The example below corresponds to 4-digit extensions beginning with **1**. 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.

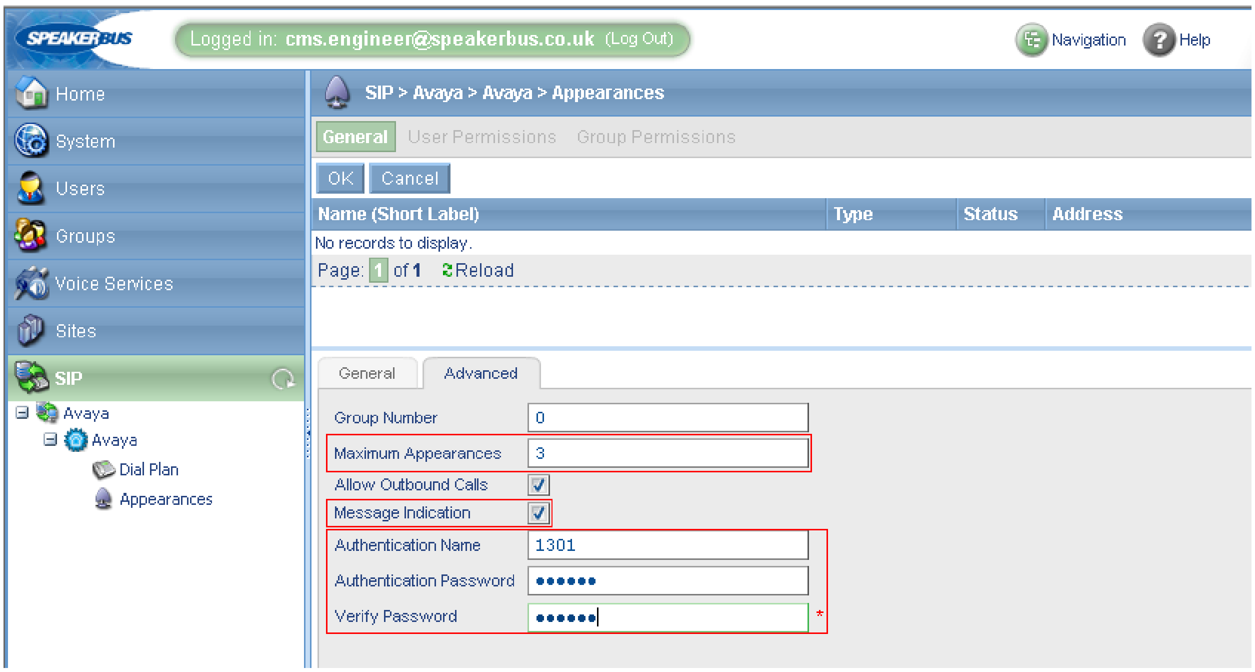


6.9. Create Call and Handset Appearances

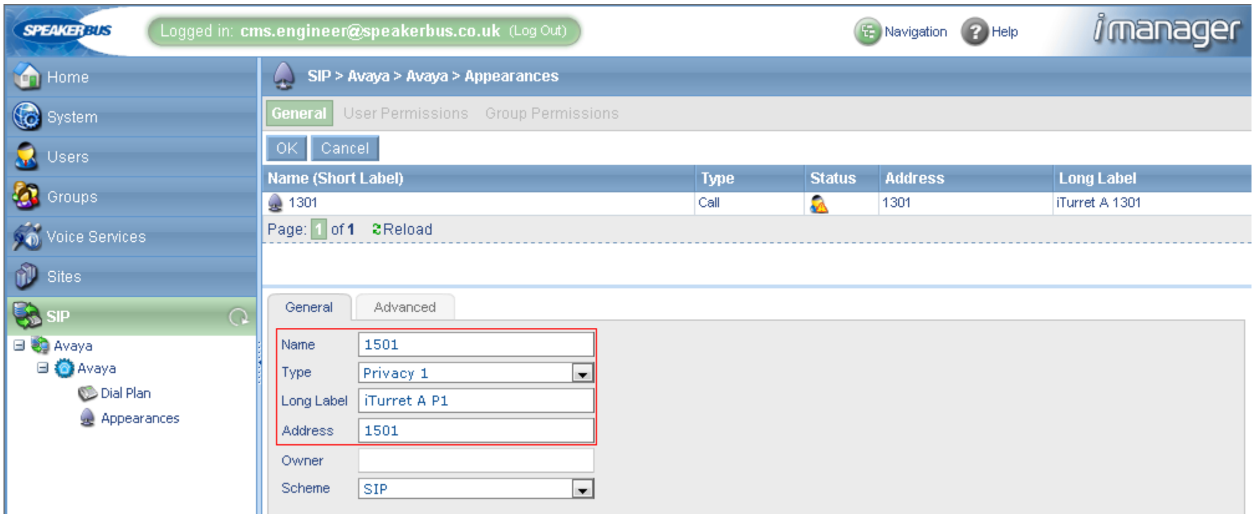
Three call appearances need to be created for each iD808 device: 1 for its main appearance, 1 call appearance is required for privacy handset 1 and another for privacy handset 2. As previously mentioned, three extensions are also required on Communication Manager and SIP Enablement Services. To create the main appearance, click **Appearances** under the **Avaya** PBX which is 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 iTurret A followed by the extension number 1301. The **Address** field should also be set to the appearance extension. Set the **Type** field to **Call**.



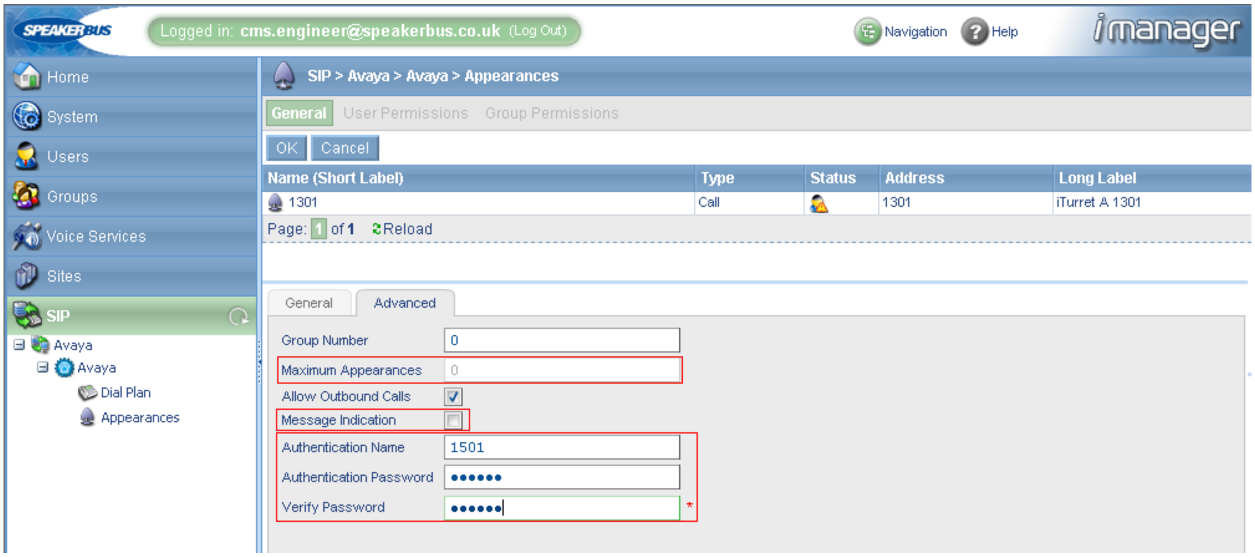
In the **Advanced** tab, set the **Maximum Appearances** field to the number of call appearances configured on the station in Communication Manager minus one since the last call appearance is restricted. See the button assignment section of the station form and the second page of the Off-PBX-Telephone Station-Mapping form in **Section 4.10** 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**.



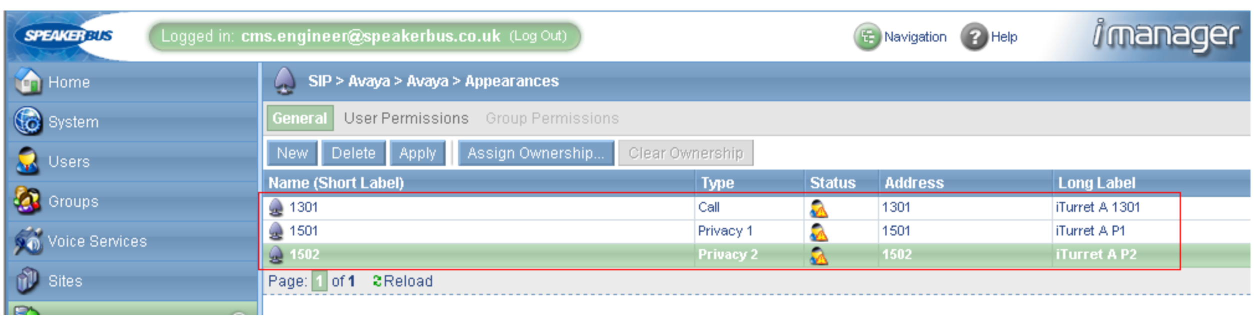
Next, this procedure will be repeated for the two privacy appearances. Click the **New** button to add another appearance. In the **General** tab, set the **Name** and **Address** fields to the extension of handset 1 and the **Long Label** field to the name of the handset. In this example, the extension is **1501**. 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 Privacy appearances, the **Maximum Appearances** field should be set to **0** since no calls will be made to the Privacy appearances directly. The **Message Indication** checkbox does not need to be enabled since the handset appearances are not voice mail subscribers. Privacy 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**.



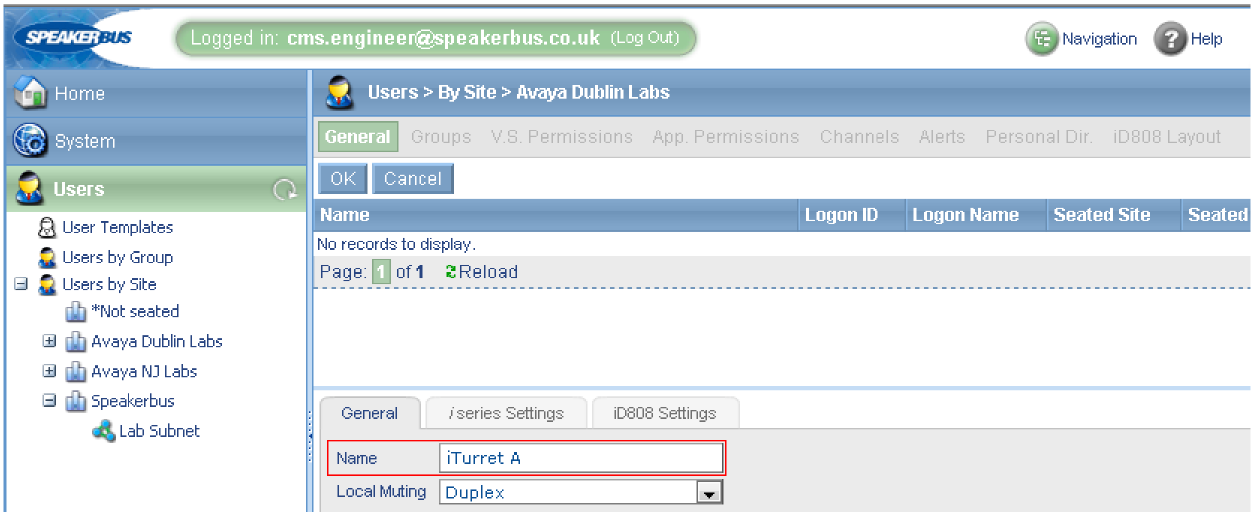
Repeat the above procedure to add the Privacy 2 appearance. The three call appearances for the previously configured iD808 deskstation are listed below.



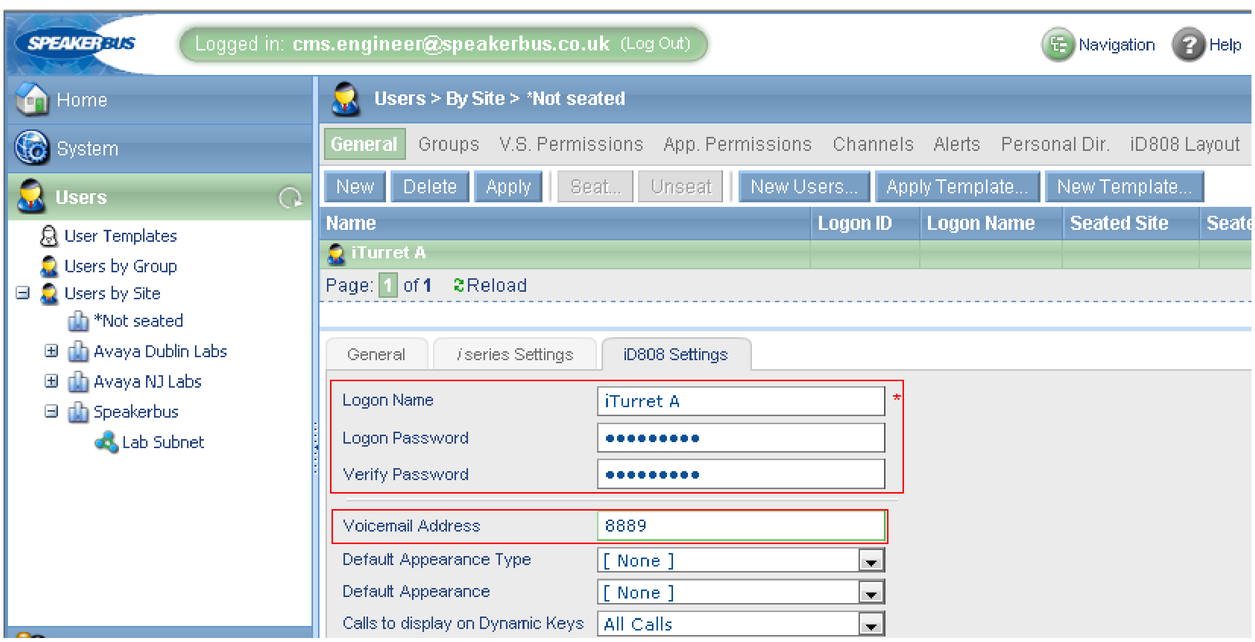
Repeat the above procedures for adding the Main and Privacy appearances for each iD808 deskstation.

6.10. Create Users

In this section, the users are created. In the left panel click on **Users** and in the directory tree expand **User by Site**, click on **Avaya Dublin Labs** followed by **New**. In the **General** tab, provide a descriptive name in the **Name** field.

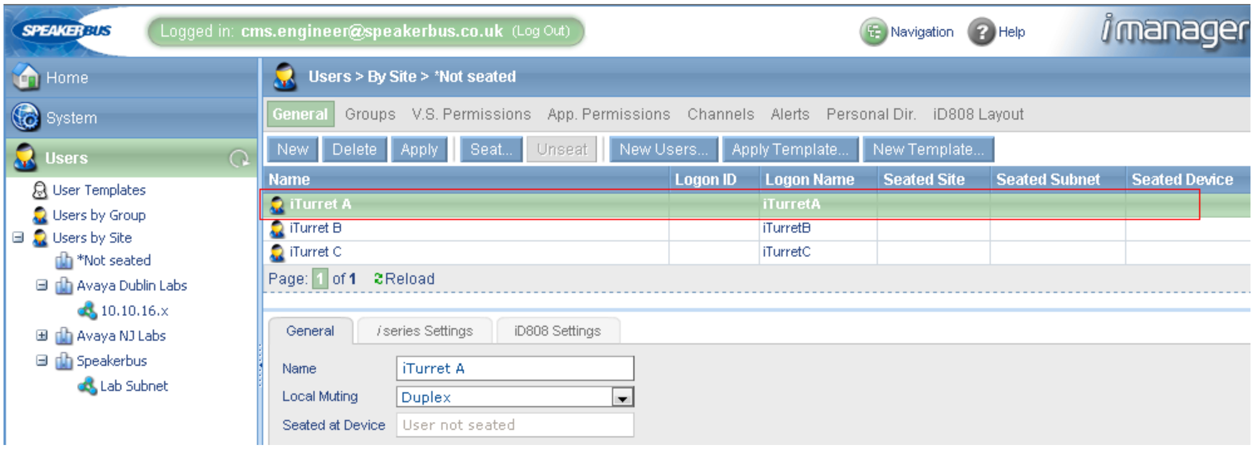


In the **iD808 Settings** tab, provide the logon credentials for the user to log into their iD808 deskstation and enter the pilot number for Voicemail in the **Voicemail Address** field. This page will be revisited later in **Section 6.14** to configure the default call appearance for this deskstation.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

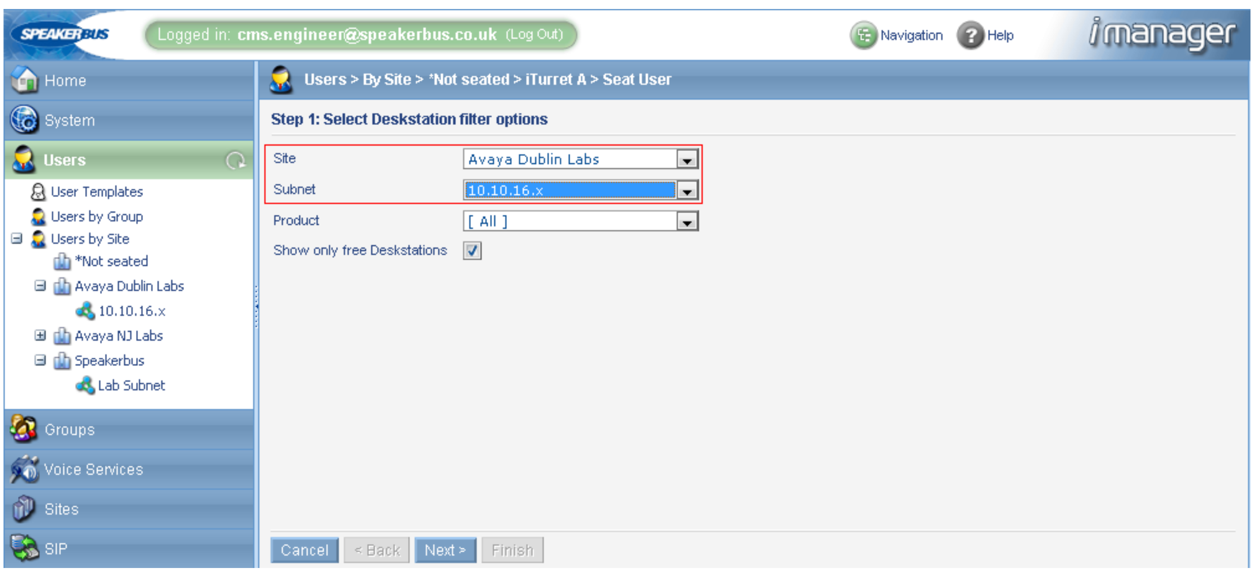


Repeat the previous procedure to add more users.

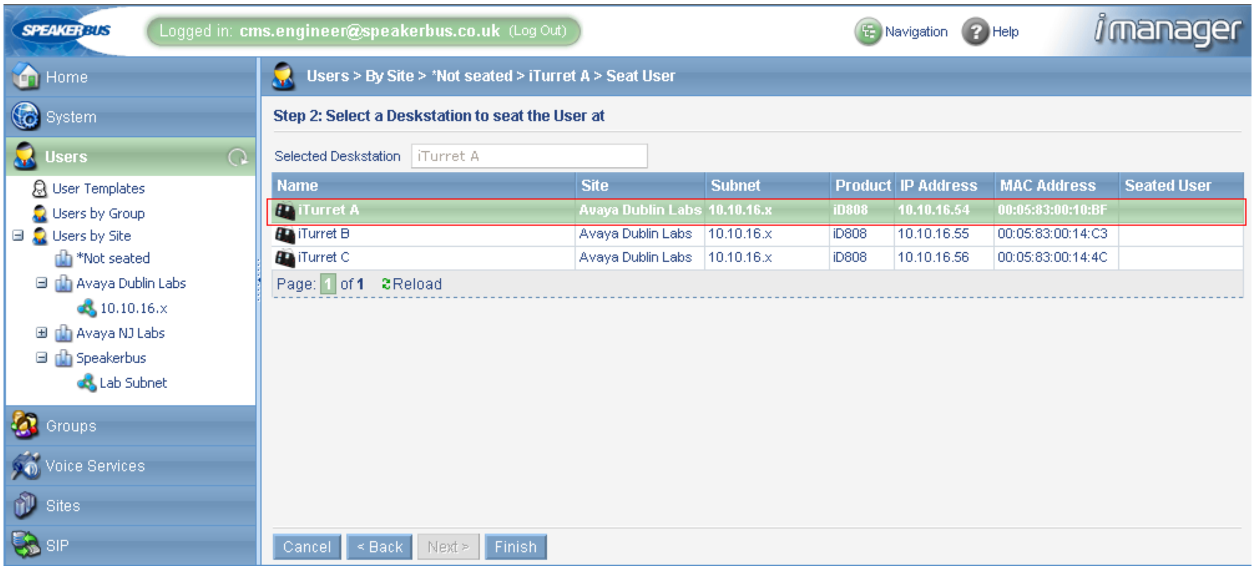
After a user has been created, the user needs to be **seated** on an iD808 deskstation. In the left panel under the **Users** directory tree, click the ***Not seated** link under **Users by Site** to display the list of users. Select the user previously configured (i.e., iTurret A) and click on the **Seat…** button.



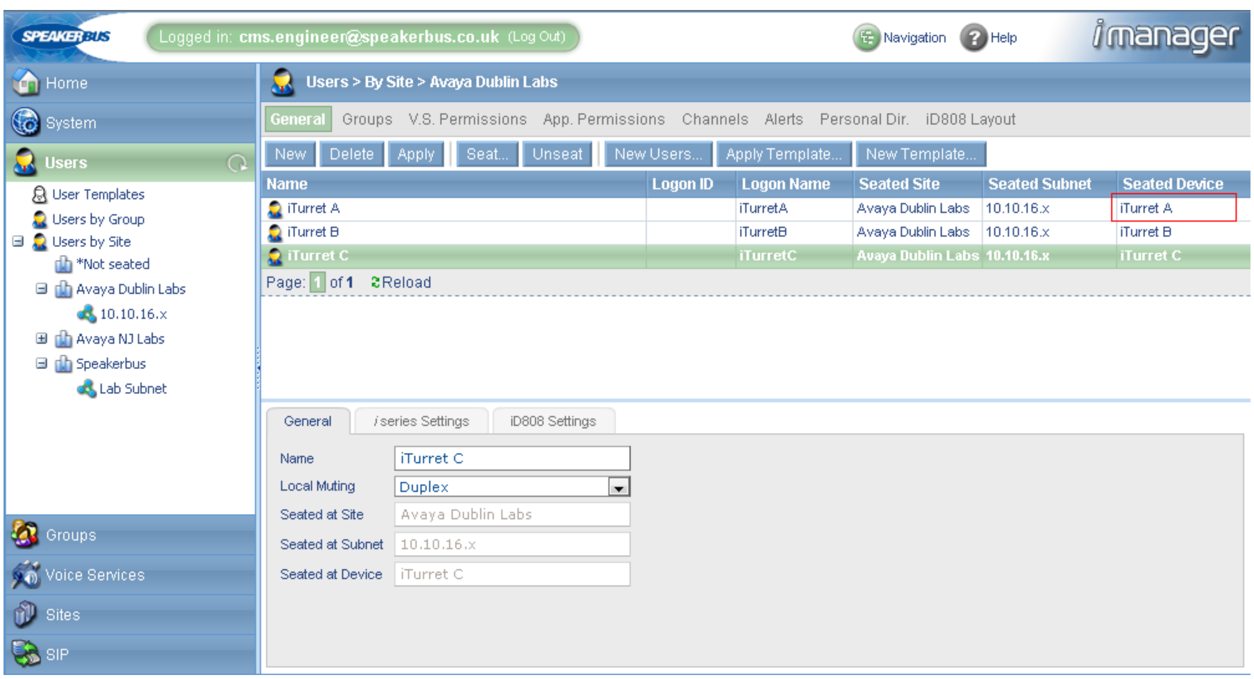
On the next page, filter options are presented. Filter deskstations in the **Avaya Dublin Labs** site and in the **10.10.16.x** subnet as shown below. The user will be seated on an iD808 deskstation with these properties. Click **Next**.



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



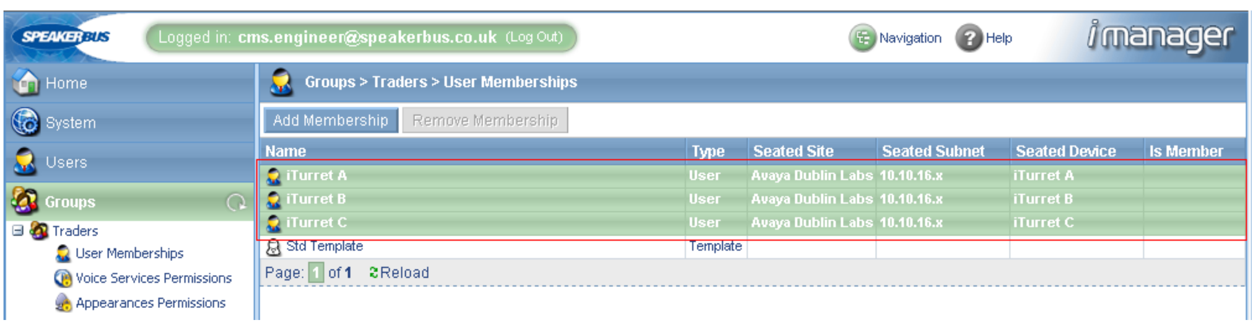
The user has been successfully seated as indicated by the deskstation displayed in the **Seated Device** column on the following page. Repeat this procedure for seating other users.



6.11. Create Group

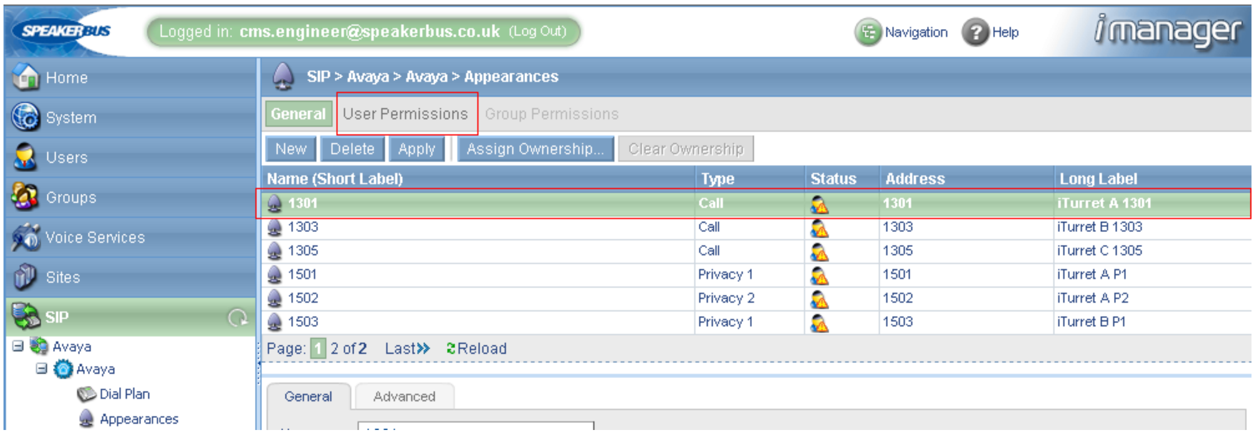
To create a group, click on **Create new Group** in the left panel under the **Groups** directory tree. In the **General** tab, provide a descriptive name in the **Name** field, such as **Traders**. Click **OK**. The **Traders** group has been successfully added.

The user is now 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 (not shown).

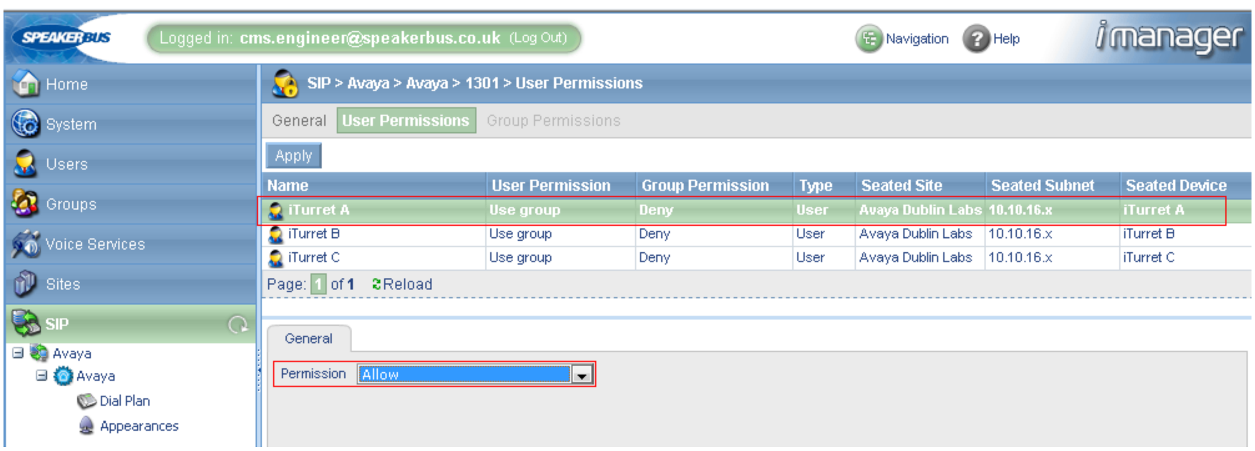


6.12. Assign User Permissions

The next step will be to assign appearances permissions to users. In the left panel under the **SIP** directory tree, expand **Avaya → Avaya** and click on **Appearances**. The list of appearances is displayed. Select the main call appearance for **iTurret A** (i.e., **1301**) and click **User Permissions**.

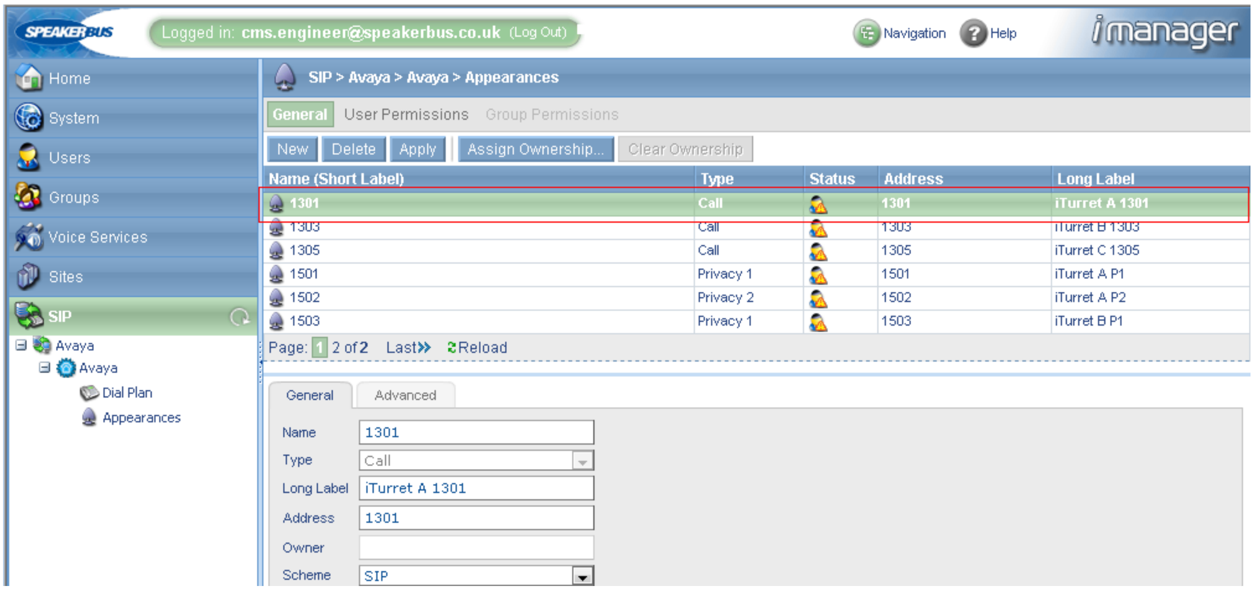


On the resulting page, select the user to which the appearance will be assigned. Set the **Permission** field to **Allow** as shown below. Click **Apply**. Assign the relevant Privacy 1 and Privacy 2 permissions to this user by repeating this procedure.

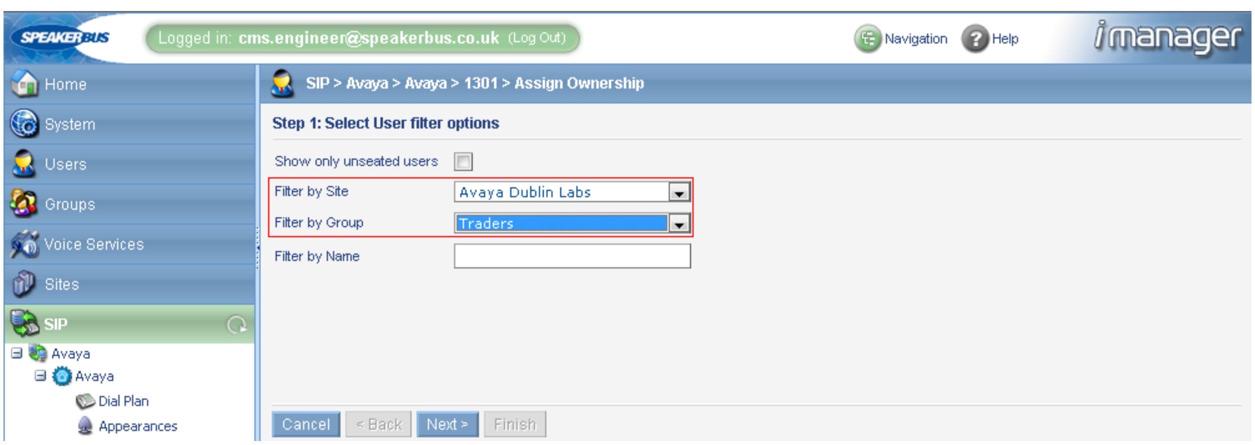


6.13. Assign Ownership

To assign ownership of the appearances to a user, in the left panel under the **SIP** directory tree, expand **Avaya → Avaya** and 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 filter options are presented. Filter users in the **Avaya Dublin Labs** site and in the **Traders** group as shown below. Click **Next**.

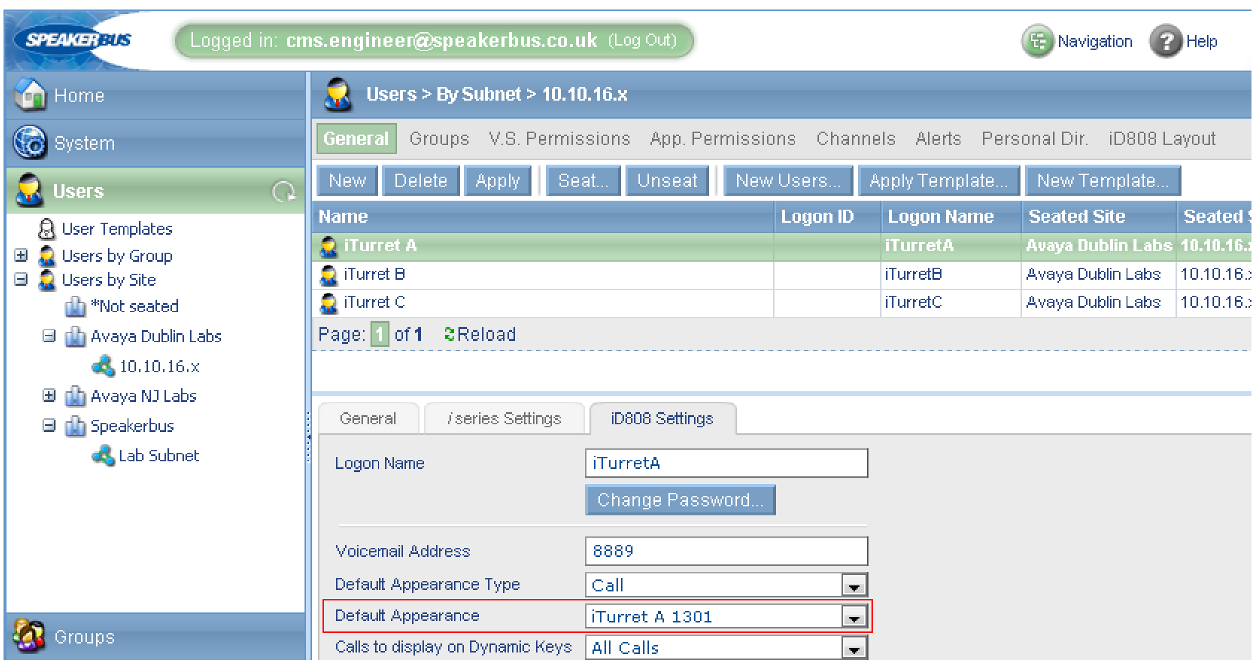


On the next page, select the user to which ownership will be assigned to the main call appearance. In this example, the main call appearance **1301** will be assigned to **iTurret A**. Click **Finish**.

Repeat this procedure to assign Privacy 1 and Privacy 2 call appearances to iTurret A.

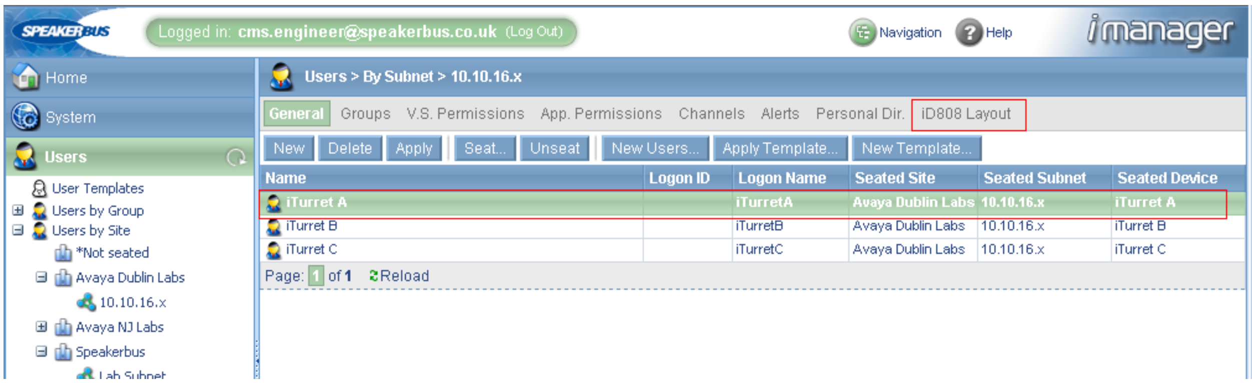
6.14. Assign Default Call Appearance

In the **Users** directory tree, navigate to **Users by Site → AvayaDublin Labs** and click **10.10.16.x** link to display the users list. Set the **Default Appearance** field to the main call appearance (e.g., **1301**). Click **Apply**.

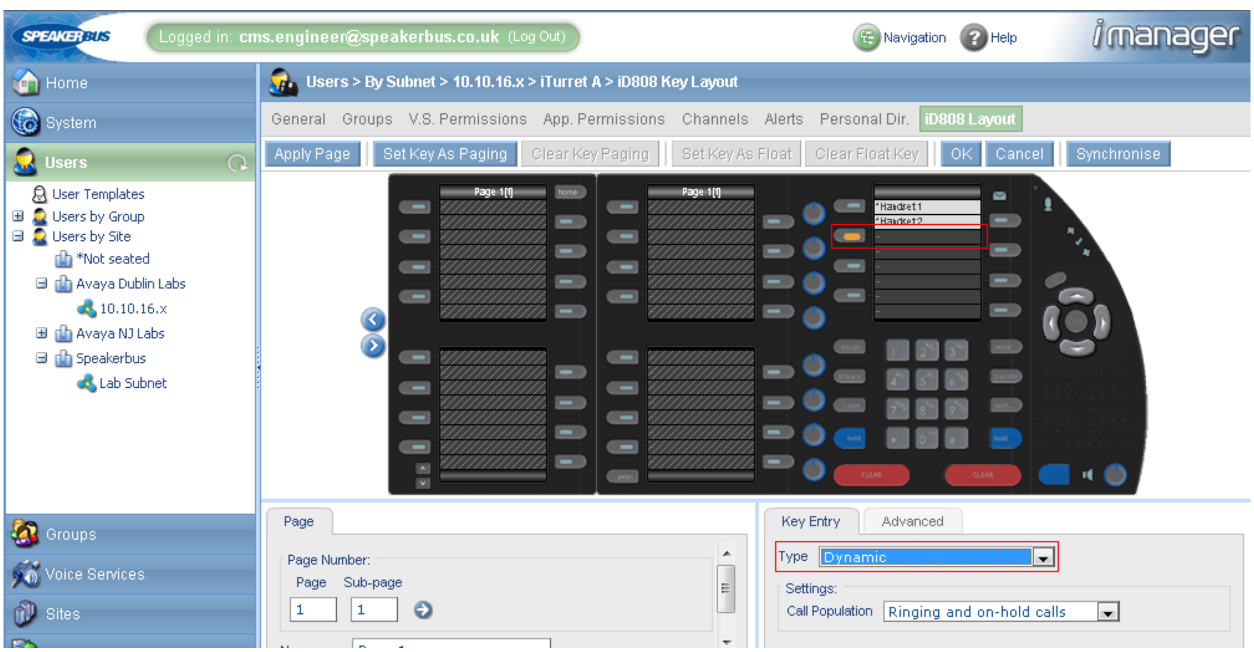


6.15. Programming iD808 Deskstations

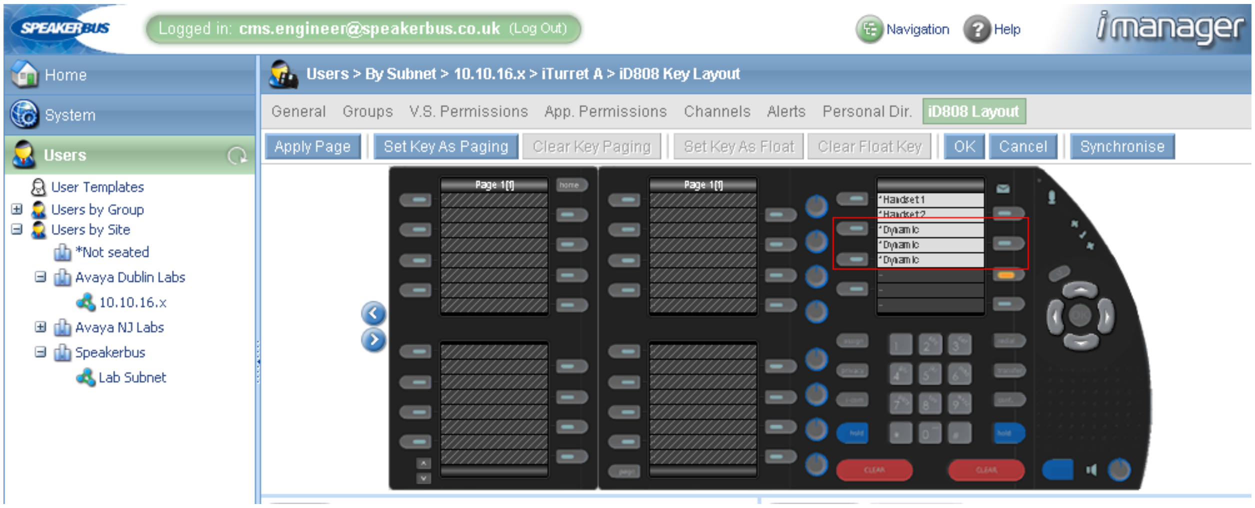
This section describes how to create iD808 deskstation keys. The following keys can be created using the iD808 layout page: Dynamic, Appearance, Shortcut, Soft Function, and Speed Dial amongst others. In this configuration, each user will be configured with three Dynamic keys, two Soft Function keys, and one Shortcut key. Although the configuration may vary, this configuration is suitable for most users. In left panel under the **Users** directory tree, expand **Users by Site → Avaya Dublin Labs** and click on **10.10.16.x** link to display a list of users. Select a user (e.g., **iTurret A**) and click **iD808 Layout** to display the iD808 key layout for this user.



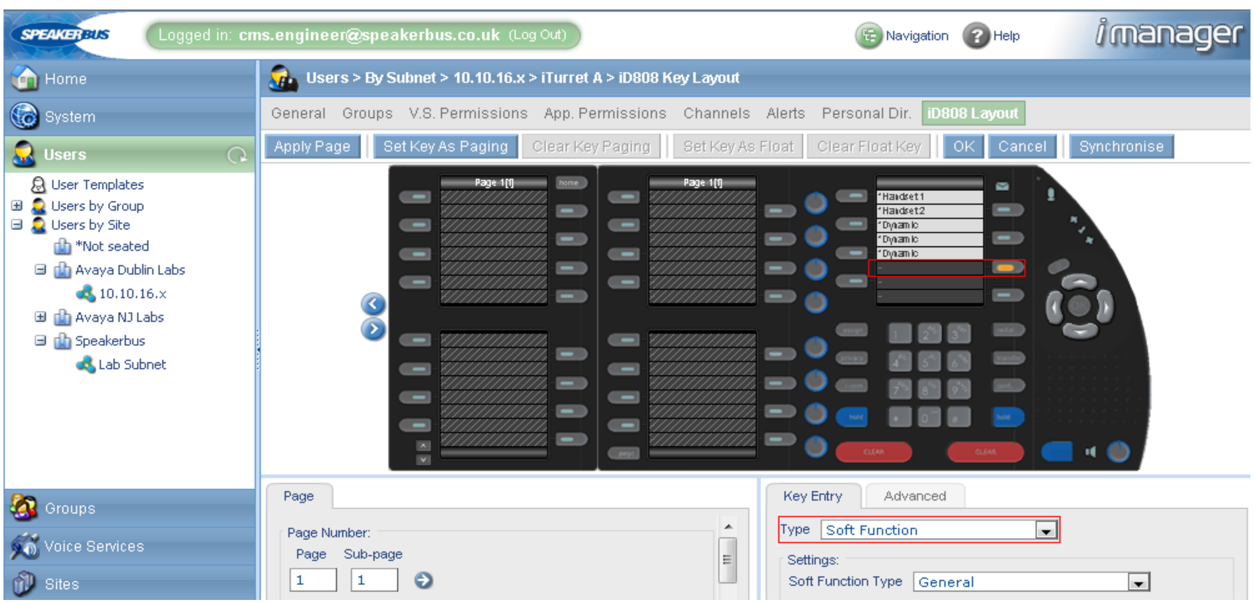
In the iD808 key layout, click on the key highlighted below Handset 2. In the Key Entry tab, set the **Type** field to **Dynamic**. Click **OK**.



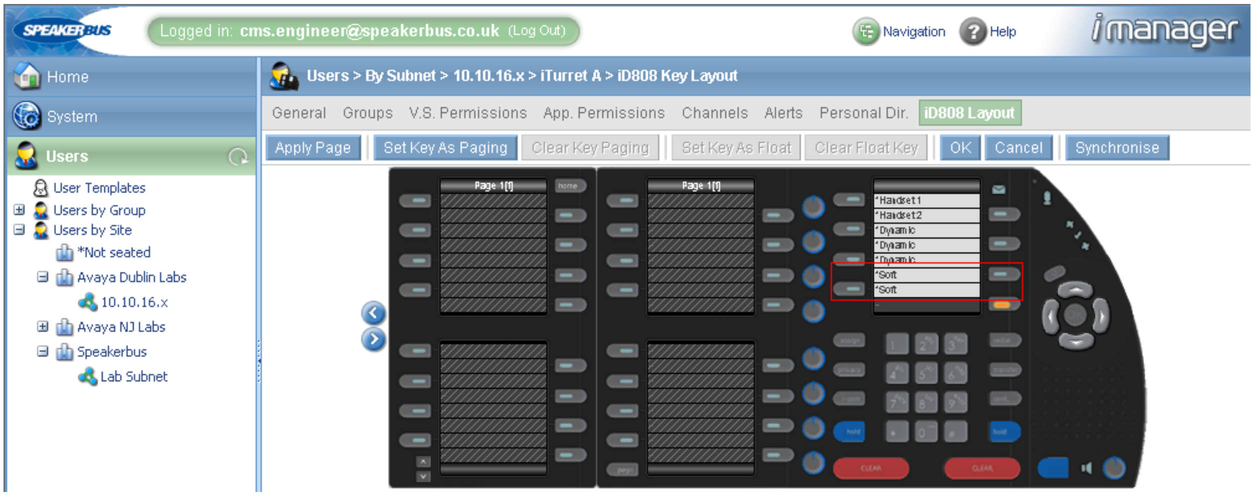
Three Dynamic keys will be added so repeat this step for the next two keys. The iD808 layout will appear as shown below once the three dynamic keys have been added.



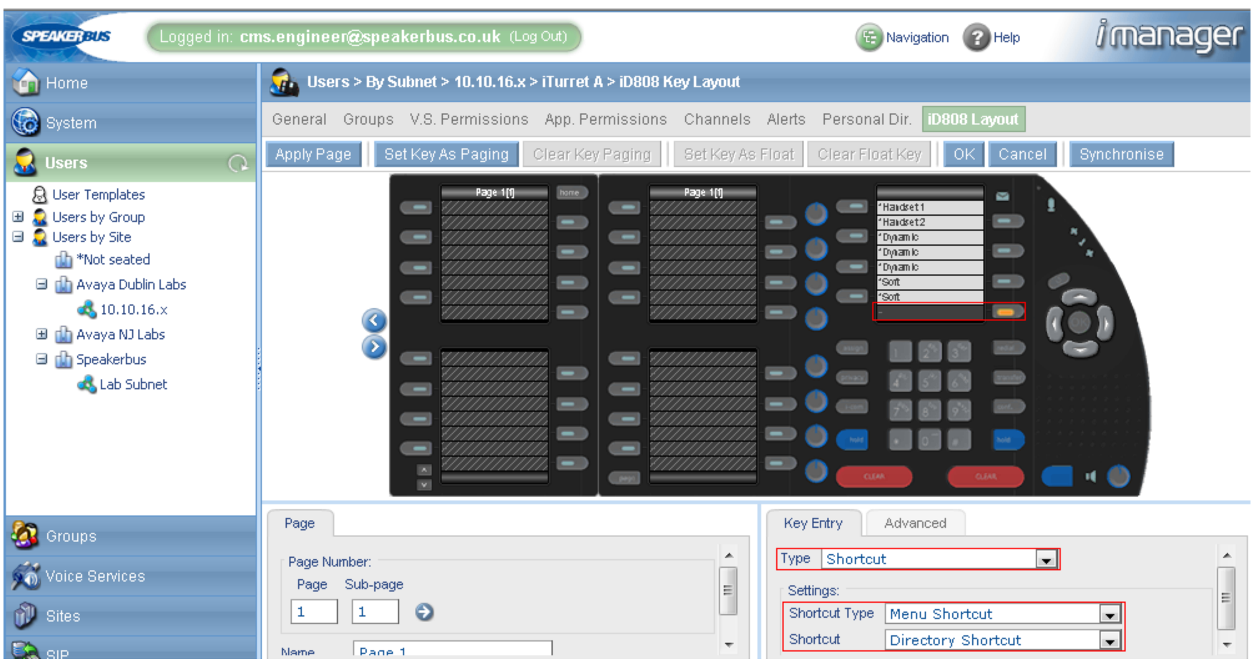
Next, configure two Soft Function keys. Select the next available key under the last Dynamic key. In the **Key Entry** tab, set the Type field to **Soft Function** and click **OK**. Repeat this step for the second Soft Function key.



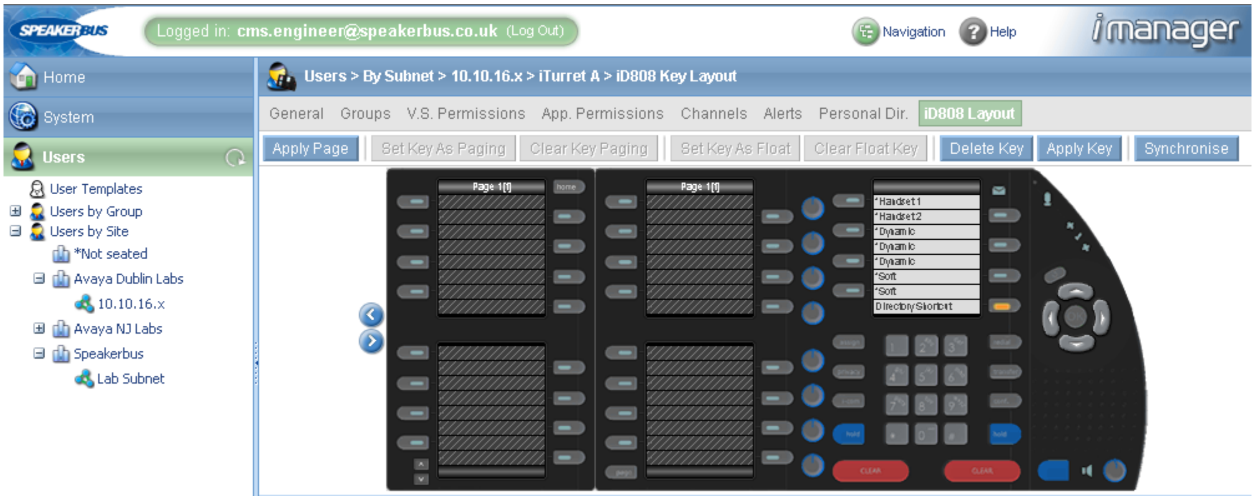
Once the two Soft Function keys have been created, the iD808 layout will be appear as shown below.



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 type** field to **Menu Shortcut**. Set the **Shortcut** field to **Directory Shortcut**. Click **OK**.

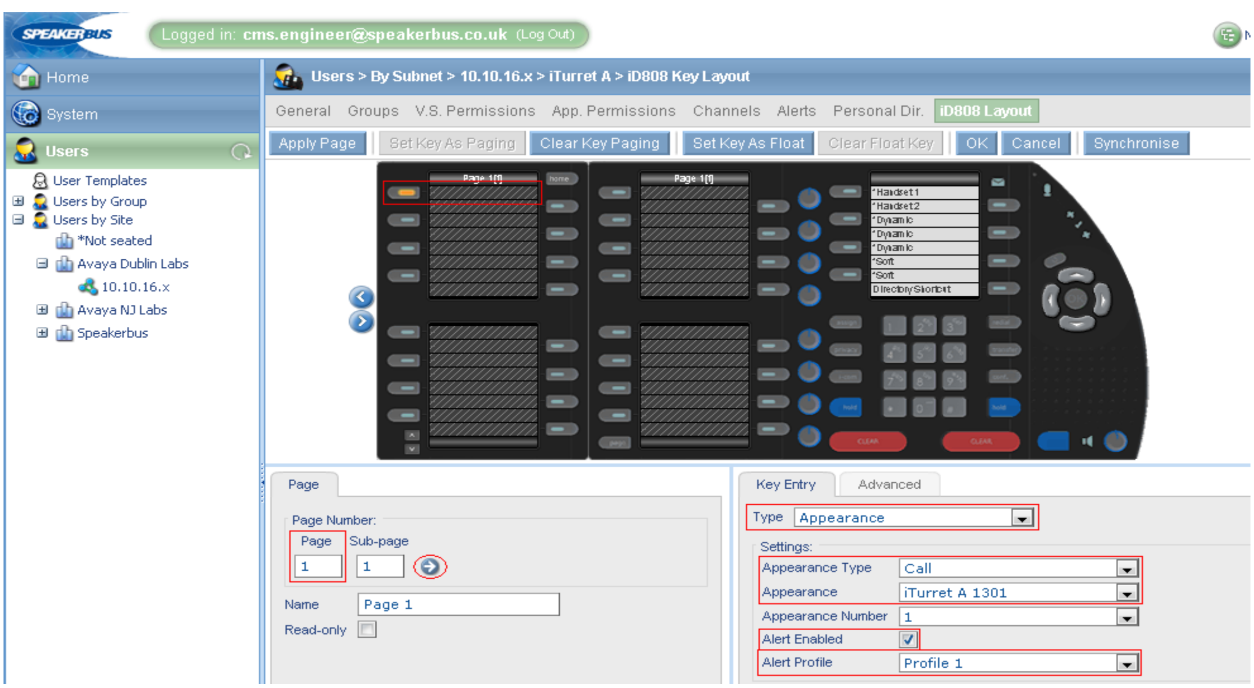


After all of the iD808 keys have been created on the deskstation, the iD808 layout will appear as shown below.

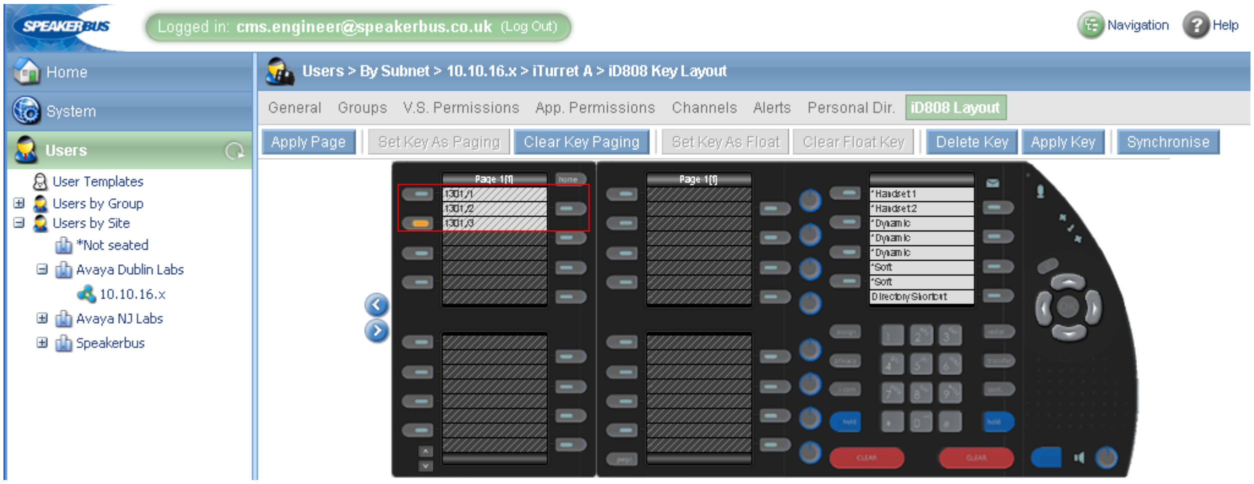


6.16. Assigning Appearances to Deskstation Keys

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. 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**. Under the **Settings** section, set the **Appearance Type** field to **Call** and the **Appearance** field to the main call appearance (e.g., iTurret1301). 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.

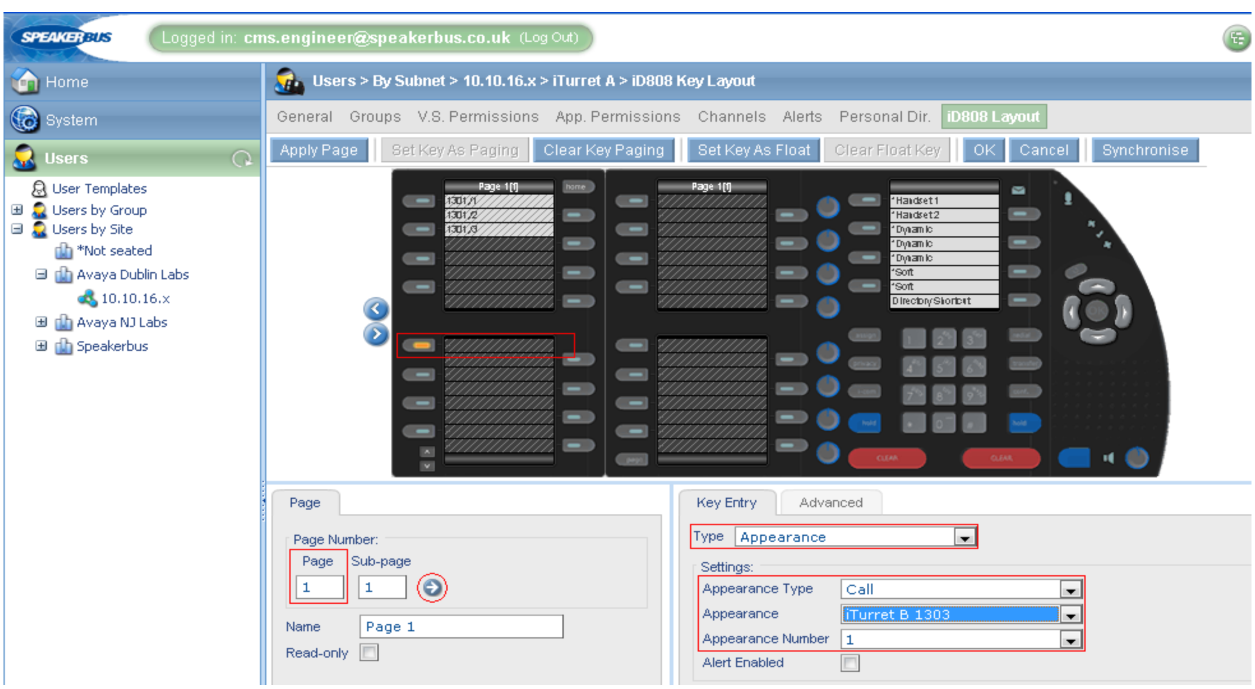


Once the three call appearances have been added, the iD808 layout will appear as follows.

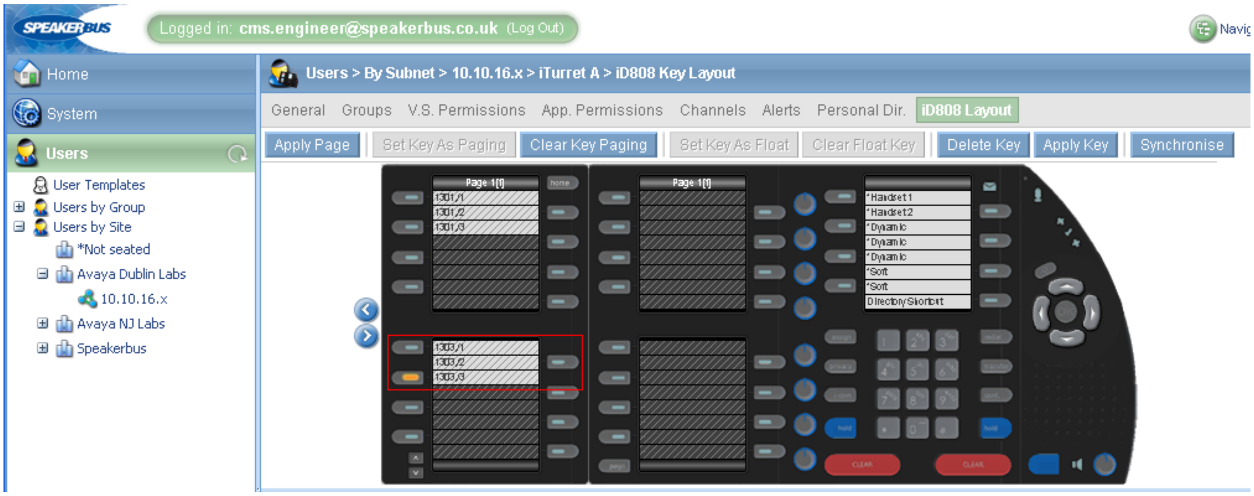


6.17. Assign a Bridge Call Appearance to Deskstation

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. Select the next available key in the lower section of page one. The next three keys on this page will be assigned to bridge call appearances. In the **Key Entry** tab, set the **Type** field to **Appearance**. Under the **Settings** section, set the **Appearance Type** field to **Call** and the **Appearance** field to the main call appearance of **iTurret B 1303**. Set the **Appearance Number** field to the number of line appearance that will be bridged (1 for 1st line appearance, 2 for the 2nd line appearance, etc.). Repeat this procedure to add the next two bridge call appearances.



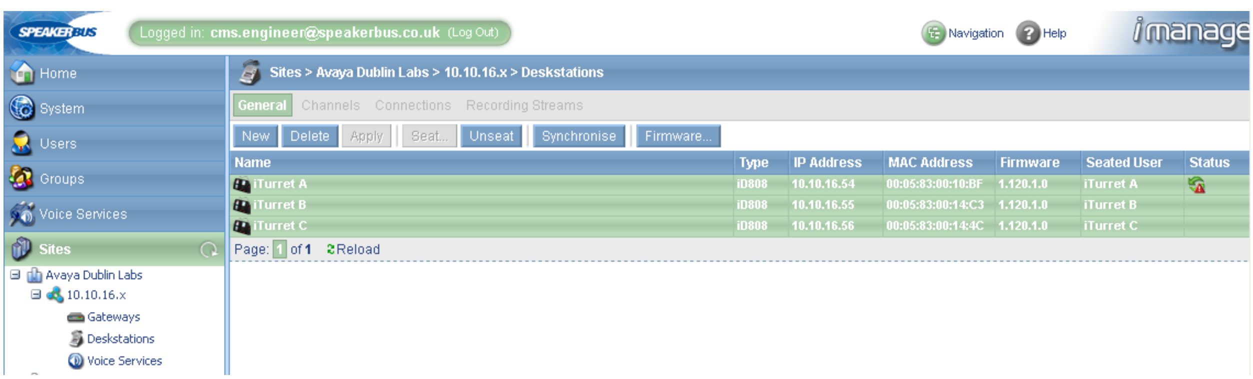
Once the three bridge call appearances have been added, the iD808 layout will appear as follows.



6.18. Synchronise Deskstations

To send the new configuration to the iD808 deskstations, the deskstations need to be synchronised with *i* cms. Under the **Sites** directory tree, expand **Avaya Dublin Labs** → **10.10.16.x** and click on **Deskstations** to display the deskstation list. Select the desired deskstations and click the **Synchronise** button. The iD808 deskstation will indicate that they are being synchronised on their displays. After the deskstations have been synchronised, the status icons on the iD808 deskstations corresponding to the network, *i* cms, and SIP registrar status should be green.

Note: Executing a synchronisation will cause active calls on the deskstation being synchronised to drop.



6.19. Feature Name Extensions (FNEs)

FNEs can be accessed by dialing the appropriate number via the dial pad. It is also possible to create FNEs as speed dials by defining the FNE in the corporate or personal directory within *i* cms. Please refer to Speakerbus documentation [**5**] for further details.

7. General Test Approach and Test Results

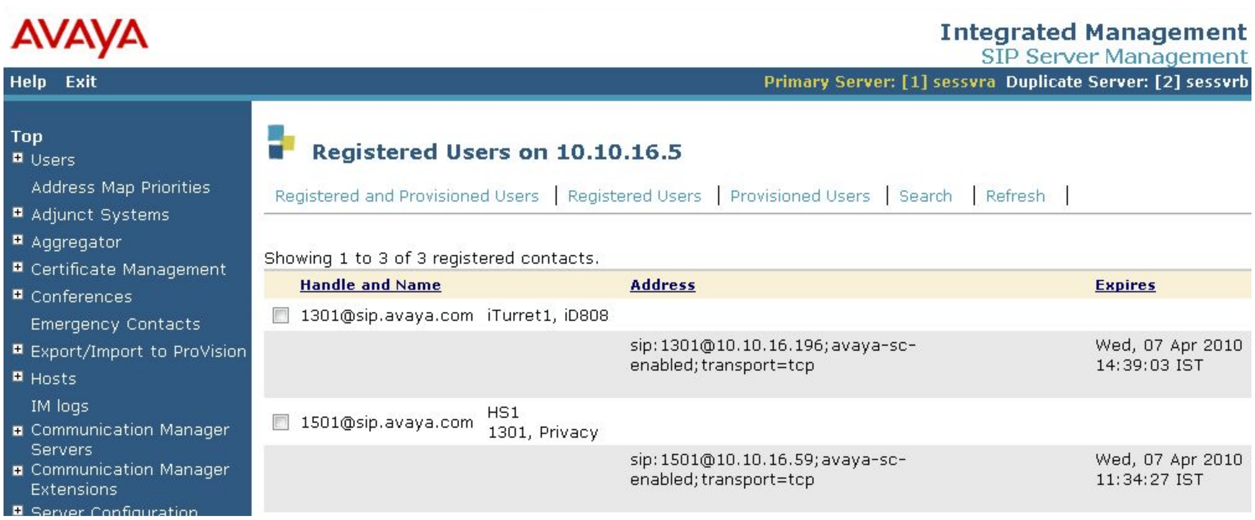
To verify interoperability of Speakerbus iD808 *i* turret with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services, calls were made between iD808 deskstations and Avaya SIP, H.323 and Digital stations using various codec settings and exercising common PBX features. The telephony features were activated and deactivated using buttons and menu options on *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): The *i* turrets do not support telephony events meaning 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 *i* turret is connected to the network, *i* cms server, and SIP registrar (i.e., SIP Enablement Services). Refer to [5] for more details.
2. Verify that the iD808 deskstations have successfully registered with SIP Enablement Services. From the administration web page navigate to **Users → Search Registered Users** and click the **Search** button (not shown). This will display a list of registered user's on SIP Enablement Services as shown below.



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 extended OPS features by dialing the Feature Name Extensions and listening for the confirmation tones.
5. 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. Call the voice messaging system from *i* turret and use the voice messaging menus to retrieve and delete the voice message. Verify that DTMF is interpreted correctly by the system and that the message waiting indicator extinguishes.

9. Conclusion

These Application Notes describe the administration steps required to use Speakerbus iD808 *i* turret with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services. Both basic and extended feature sets were covered as shown in **Table 1**.

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 Avaya Aura™ Communication Manager*, Nov 2009

[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*, Nov 2009, Issue 8.0, Document Number 03-600768.

[5] *Speakerbus i manager Administrator's Guide*, V1.220, Revision 6, March 2010.

[6] *Session Initiation Protocol Service Examples draft-ietf-sipping-service-examples-15*, Internet-Draft, 11th July 2008, available at http://tools.ietf.org/html/draft-ietf-sipping-service-examples-15

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.