DevConnect Program

Application Notes for configuring Speakerbus iTurret with Avaya Aura® Communication Manager and Avaya Aura® Session Manager – Issue 1.0

Abstract

These Application Notes describe the steps required to connect Speakerbus iTurret v4.1 to Avaya Aura® Session Manager R10.1 and Avaya Aura® Communication Manager R10.1 as a SIP User. Avaya Aura® Communication Manager features can be made available in addition to the standard features supported on the iTurret.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

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 iTurret v4.1 to Avaya Aura® Session Manager R10.1 and Avaya Aura® Communication Manager R10.1 as a SIP user. Also described, is how Avaya Aura® Communication Manager features can be made available in addition to the standard features supported by iTurret. In this configuration, the Off-PBX Stations (OPS) feature set is extended from Avaya Aura® Communication Manager to the Speakerbus iTurret, providing the iTurret deskstation with enhanced calling features.

The table below provides a summary of the supported features available on iTurret with the Avaya SIP offer. Some features are supported locally on the iTurret, while others are only available with Communication Manager and Session Manager with OPS. In addition to basic calling capabilities, the Internet Engineering Task Force (IETF) has defined a supplementary set of calling features, often referred to as the SIPPING-19 **[5]**. 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 iTurret using OPS.

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

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	

FEATURE	SUPPORTED		COMMENTS
	Locally at the phone	*With Avaya SIP Offer*	
SIPPING-19 Features			
Call Hold	Yes	Yes	
Consultation Hold	Yes	Yes	
Unattended Transfer	Yes	Yes	
Attended Transfer	Yes	Yes	
Call Forward All	Yes	Yes	Local menu option on iTurret and FNU
Call Forward Busy/No answer	Yes	Yes	Local menu option on iTurret and FNU
Call Forward Cancel	Yes	Yes	Local menu option on iTurret and FNU
3-way conferencing (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			
Conference on answer	No	Yes	Via OPS FNE
Directed call pickup	**No**	**Yes**	**Via OPS FNE**
Drop last added party	No	Yes	Via OPS FNE
Exclusion/Privacy	**Yes**	**Yes**	**Local hard key on iD808 iTurret using FNU**
Last number dialed	**Yes**	**Yes**	**Via OPS FNE**
Priority Call	No	Yes	Via OPS FNE, iTurret doesn`t support distinctive ring indication
Send All Calls	**No**	**Yes**	**Via OPS FNE**
Send All Calls Cancel	**No**	**Yes**	**Via OPS FNE**
Transfer to Voicemail	No	Yes	Via OPS FNE
Whisper Page	**No**	**Yes**	**Via OPS FNE**

Table 1

2. General Test Approach and Test Results

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

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

Avaya's formal testing and Declaration of Conformity is provided only on the headsets/Smartphones that carry the Avaya brand or logo. Avaya may conduct testing of non-Avaya headset/handset to determine interoperability with Avaya phones. However, Avaya does not conduct the testing of non-Avaya headsets/Smartphones for: Acoustic Pressure, Safety, Hearing Aid Compliance, EMC regulations, or any other tests to ensure conformity with safety, audio quality, long-term reliability or any regulation requirements. As a result, Avaya makes no representations whether a particular non-Avaya headset will work with Avaya's telephones or with a different generation of the same Avaya telephone.

Since there is no industry standard for handset interfaces, different manufacturers utilize different handset/headset interfaces with their telephones. Therefore, any claim made by a headset vendor that its product is compatible with Avaya telephones does not equate to a guarantee that the headset will provide adequate safety protection or audio quality.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with these Application Notes, the interface between Avaya systems and the iTurret did not include use of any specific encryption features as requested by Speakerbus.

Note: Compliance testing was carried out using both UDP and TCP as the transport for SIP signaling.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Successful registration of the iTurret deskstation with Session Manager.
- Calls between the iTurret and Avaya SIP, H.323, and digital extensions.
- Hold/Retrieve operations.
- Supervised/blind transfers and Conference.
- Codec support (tested G.722, G.711A and G.729).
- COR restricted calls.
- Bridged appearances.
- Barge in and Privacy.
- PSTN calls.
- Voicemail and message waiting indicators (MWI).
- Extended telephony features using Communication Manager Feature Name Extensions (FNEs) shown in bold in **Table 1**.
- Call forwarding (busy and no-answer) and Send All Calls using Call Forwarding and Send All Call FNU`s.
- Serviceability testing after an iTurret restart and loss of IP connection.

2.2. Test Results

All the test cases passed successfully with the following observation.
In a particular scenario, where there are three iTurret deskstations each having the bridged appearances of the other two iTurret deskstations, there are issues observed with ‘Barge In’ and ‘Privacy’.

- If a call is made from User 1 to User 3 and then User 2 barges into the call (either to User 1 ext or to User 2 ext), hangs up and barges in a second time, upon hanging up for the second time in succession all calls are dropped. This behaviour is the same for Avaya SIP phones. Avaya are already aware of this issue.
- With the same call in place (User 1 to User 3) and User 1 presses the ‘Privacy Key’. When User 2 tries to barge into User 1’s call, User 2 is refused as expected but when User 2 tries again it results in all calls being dropped. This behavior is the same for Avaya SIP phones. Avaya are already aware of this issue.

2.3. Support

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

- Web: http://www.speakerbus.com
- Email: support@speakerbus.com
- Telephone: +1 (646) 289 4700 in North America
  +44 (0) 870 240 7252 in Europe
  +65 6590 9228 in Asia

3. Reference Configuration

Figure 1 illustrates the network topology used during compliance testing. The Avaya solution consists of a Communication Manager, Session Manager along with a Media Gateway and a Media Server. System Manager was used to provision Communication Manager and Session Manager. Speakerbus iTurrets were connected to the LAN and connect to Session Manager as a SIP user. SIP, Digital and H.323 telephones were used to place calls to and receive calls from the Speakerbus iTurrets. Avaya Messaging was used to provide and test voicemail and Message Waiting facilities.

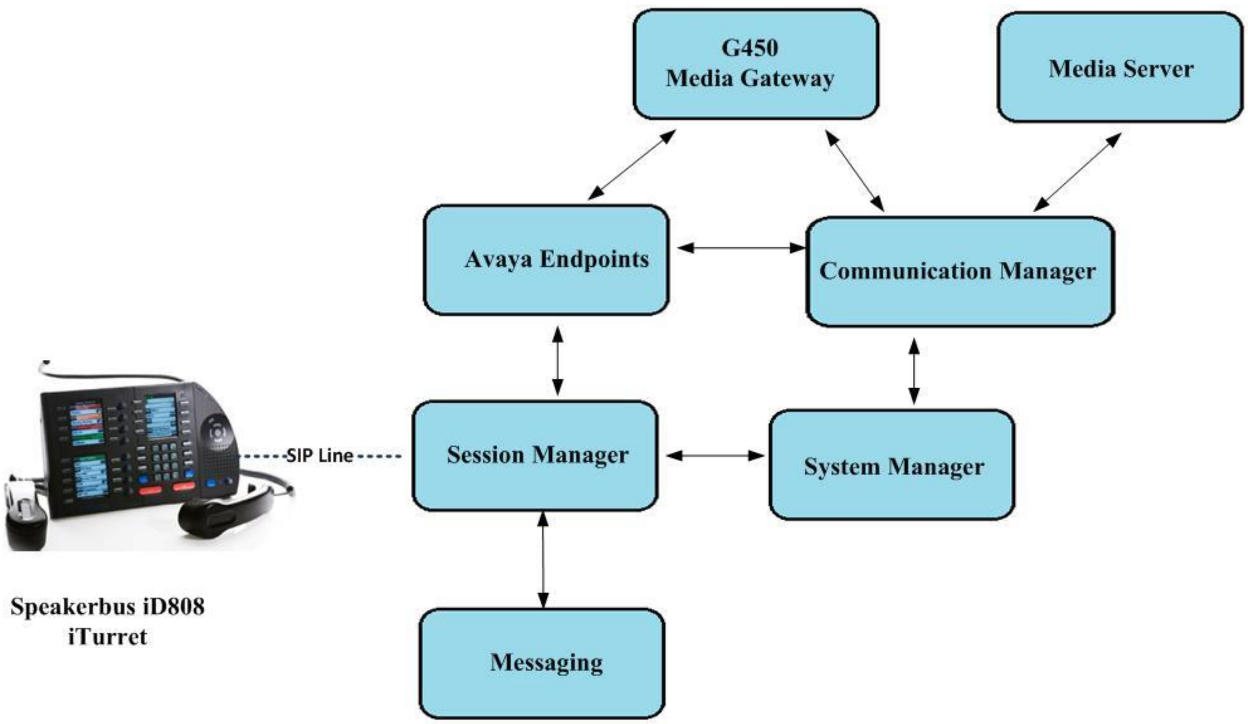


Figure 1: Avaya Aura® Communication Manager and Avaya Aura® Session Manager with Speakerbus solution

4. Equipment and Software Validated

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

Avaya Equipment/Software	Release/Version
Avaya Aura® System Manager	10.1.2.0 Build no. 10.1.0.0.537353 Software update 10.1.2.0.0715476
Avaya Aura® Session Manager	10.1 Build No. – 10.1.2.0.1012016
Avaya Aura® Communication Manager	10.1.2.0 – FP2 Update 01.0.974.0-27783
Avaya Messaging	11.0 SP2 Build 11.0.0.324
Avaya Aura® Media Server	10.1.0.101
Avaya Media Gateway G450	42.7.0 /2
Avaya J100 Series (H323) Deskphone	6.8.5.3.2
Avaya J100 Series (SIP) Deskphone	4.0.14.0.7
Avaya 9404 Digital Deskphone	17.0
Speakerbus Equipment/Software	**Release/Version**
Speakerbus iCMS with iManager	V4.001.1.0
Speakerbus iTurret (SIP interface version)	V2.20
Speakerbus iTurret (Main code version)	V4.100.5.0

5. Configure Avaya Aura® Communication Manager

No specific changes were made on Communication Manager to facilitate the connection of the iTurret with Session Manager. The iTurret utilizes some of the features provided by Communication Manager. These features along with the dial plan, SIP trunk and coverage path are displayed in this section to provide the reader with some helpful information on how Communication Manager was setup for compliance testing.

Every site will have a unique setup, the information contained in the System Parameters Features or the System Parameters Customer Options will be suited to that particular site. The information provided in this section serves to show how this system was setup during compliance testing and is <u>not</u> an instruction guide to setup the Communication Manager for the iTurret to work. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 10**.

Configuration and verification operations on Communication Manager illustrated in this section were all performed using Communication Manager System Administration Terminal (SAT). Communication Manager information displayed in this section can be summarized as follows:

- System Parameters and Features.
- SIP Trunk.
- Call Routing for iTurret.
- Feature Access Codes (FACs).
- Feature Name Extensions (FNEs).
- Class of Service (COS).
- Class of Restriction (COR).
- Coverage Path.

Note: Any settings not in **Bold** in the following screen shots may be left as default.

5.1. Verify System Parameters and Features

Each Communication Manager system will have its own setup with different System Parameters and Features configured depending on the requirement of the customer. Here is a snapshot of some of these values that were configured on the DevConnect lab for compliance testing.

5.1.1. Verify System Parameters Customer Options

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 iTurret device.

```
display system-parameters customer-options                    Page   1 of  12
                                OPTIONAL FEATURES

     G3 Version: V18                                 Software Package: Enterprise
       Location: 2                                    System ID (SID): 1
       Platform: 28                                   Module ID (MID): 1

                                                           USED
                                Platform Maximum Ports:   6400     82
                                      Maximum Stations:   2400     22
                              Maximum XMOBILE Stations:   2400      0
                    Maximum Off-PBX Telephones - EC500:   9600      0
                    Maximum Off-PBX Telephones -   OPS:   9600     18
                    Maximum Off-PBX Telephones - PBFMC:   9600      0
                    Maximum Off-PBX Telephones - PVFMC:   9600      0
                    Maximum Off-PBX Telephones - SCCAN:      0      0
                         Maximum Survivable Processors:  313        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  12
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                         USED
                 Maximum Administered H.323 Trunks:  4000      0
       Maximum Concurrently Registered IP Stations:  1000      2
         Maximum Administered Remote Office Trunks:  4000      0
Max Concurrently Registered Remote Office Stations:  1000      0
          Maximum Concurrently Registered IP eCons:   68       0
     Max Concur Reg Unauthenticated H.323 Stations:   100      0
                    Maximum Video Capable Stations:   2400     0
               Maximum Video Capable IP Softphones:   1000     1
                   Maximum Administered SIP Trunks:  4000     50
  Max Administered Ad-hoc Video Conferencing Ports:  4000      0
   Max Number of DS1 Boards with Echo Cancellation: 80     0






        (NOTE: You must logoff & login to effect the permission changes.)
```

5.1.2. Define System Features

Use the **change system-parameters features** command to administer system wide features for SIP endpoints. Those related to features listed in Error! Reference source not found. are shown in bold. These are all standard Communication Manager features that are also available to OPS stations. On **Page 18**, set the **Whisper Page Tone Given To** field to **all**.

```
display system-parameters features                             Page  18 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS

INTERCEPT TREATMENT PARAMETERS
       Invalid Number Dialed Intercept Treatment: tone
                   Invalid Number Dialed Display:
    Restricted Number Dialed Intercept Treatment: tone
                Restricted Number Dialed Display:
   Intercept Treatment On Failed Trunk Transfers? n

WHISPER PAGE
   Whisper Page Tone Given To: all

6400/8400/2420J LINE APPEARANCE LED SETTINGS
                    Station Putting Call On Hold: green  wink
                     Station When Call is Active: steady
         Other Stations When Call Is Put On Hold: green  wink
              Other Stations When Call Is Active: green
                                         Ringing: green  flash
                                            Idle: steady

                              Pickup On Transfer? y
```

On **Page 19** make sure **Directed Call Pickup** is set to **y**.

```
display system-parameters features                             Page  19 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS
IP PARAMETERS
            Direct IP-IP Audio Connections? y         IP Audio Hairpinning? n
                   Synchronization over IP? n Allow SIP-H323 Video in SDES? n
    Initial INVITE with SDP for secure calls? y
               SIP Endpoint Managed Transfer? n

  Expand ISDN Numbers to International for 1XCES? 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: simple
                 Enhanced Call Pickup Alerting? n

   Call Pickup for Call to Coverage Answer Group? y
                        Display Information With Bridged Call? n
  Keep Bridged Information on Multiline Displays During Calls? y
                  PIN Checking for Private Calls? n
```

5.2. Configure SIP Trunk

In the **Node Names IP** form, note the IP Address of the **procr** and the Session Manager (**sm101x**). The host names will be displayed throughout the other configuration screens of Communication Manager and Session Manager. Type **display node-names ip** to show all the necessary node names.

```
display node-names ip
                                  IP NODE NAMES
    Name              IP Address
IPOffice            10.10.40.25
aes101x             10.10.40.16
ams101x             10.10.40.17
default             0.0.0.0
g430                10.10.40.15
procr               10.10.40.13
procr6              ::
sm101x              10.10.40.12
```

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager in **Section 6.1.1**. In this configuration, the domain name is **greaneyp.sil6.avaya.com**. 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 Session Manager as **ip-network region 1** is specified in the SIP signaling group.

```
display ip-network-region 1                                   Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: greaneyp.sil6.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? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        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 by the iTurret. 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. Note the **Media Encryption** includes a setting of **none** to allow for unencrypted media.

```
display ip-codec-set 1                                          Page   1 of   2
                          IP MEDIA PARAMETERS
    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.729A            n          2        20
 4: G.722-64k         n          2        20

     Media Encryption                   Encrypted SRTCP: enforce-unenc-srtcp
 1: 1-srtp-aescm128-hmac80
 2: none
 3:
```

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. The configuration of the Signaling group used to send calls from Communication Manager to Session Manager for SIP users is as follows.

- Set the **Group Type** field to **sip**.
- Set the **Transport Method** to the appropriate setting, in this case it was set to **tls**.
- The **Peer Detection Enabled** field should be set to **y** allowing the Communication Manager to automatically detect if the peer server is a Session Manager.
- Specify the node names for the procr and the Session Manager 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.
- Set the **Near-end Node Name** to **procr**. This value is taken from the **IP Node Names** form shown above.
- Set the **Far-end Node Name** to the node name defined for the Session Manager (node name **sm101x**).
- Ensure that the recommended TLS port value of **5061** is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- In the **Far-end Network Region** field, enter the IP Network Region configured above**.** This field logically establishes the **far-end** for calls using this signaling group as network region 1.
- **Far-end Domain** was set to the domain used during compliance testing.
- The **DTMF over IP** field should remain set to the default value of **rtp-payload**. This value enables Communication Manager to send DTMF transmissions using RFC 2833.
- The **Direct IP-IP Audio Connections** field is set to **y**.
- **Initial IP-IP Direct Media** is set to **n**.
- The default values for the other fields may be used.

```
change signaling-group 11                                       Page   1 of   2
                                SIGNALING GROUP

 Group Number: 11             Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                 Enforce SIPS URI for SRTP? n
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                 Far-end Node Name: sm101x
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1

Far-end Domain: greaneyp.sil6.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? n
         Enable Layer 3 Test? Y                    Initial IP-IP Direct Media? n
                                                  Alternate Route Timer(sec): 6
```

The Trunk Groups used to send calls between Communication Manager and Session Manager was setup as follows. 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.

```
change trunk-group 11                                           Page   1 of   5
                                TRUNK GROUP

Group Number: 1                   Group Type: sip          CDR Reports: y
  Group Name: SIP Phones                 COR: 1       TN: 1        TAC: *811
   Direction: two-way       Outgoing Display? y
 Dial Access? n                                 Night Service:
Queue Length: 0
Service Type: tie                 Auth Code? n
                                             Member Assignment Method: auto
                                                      Signaling Group: 11
                                                    Number of Members: 10
```

On **Page 2** of the trunk-group form the **Preferred Minimum Session Refresh Interval (sec)** field was set to a value of **1200** to prevent unnecessary SIP messages during call setup. Session refresh is used throughout the duration of the call, to check the other side has not gone away. This may be changed if required by Speakerbus.

```
change trunk-group 11                                          Page   2 of  5
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 5000

            SCCAN? n                               Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 1200

 Disconnect Supervision - In? y  Out? y


           XOIP Treatment: auto    Delay Call Setup When Accessed Via IGAR? n
```

Settings on **Page 3** can be left as default. However, the **Numbering Format** in the example below is set to **private**.

```
change trunk-group 11                                          Page   3 of   5

TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                        Maintenance Tests? y



   Suppress # Outpulsing? n  Numbering Format: private
                                            UUI Treatment: shared
                                           Maximum Size of UUI Contents: 128
                                              Replace Restricted Numbers? n
                                             Replace Unavailable Numbers? n


                                Modify Tandem Calling Number: no
               Send UCID? y



 Show ANSWERED BY on Display? y

 DSN Term? n
```

Settings on **Page 4** are as follows.

```
change trunk-group 11                                           Page   4 of   5
                           SHARED UUI FEATURE PRIORITIES

                             ASAI: 1

         Universal Call ID (UCID): 2

MULTI SITE ROUTING (MSR)

                      In-VDN Time: 3
                         VDN Name: 4
                 Collected Digits: 5
            Other LAI Information: 6
                   Held Call UCID: 7
                          ECD UUI: 8
```

Settings on **Page 5** are as follows.

```
change trunk-group 11                                            Page   5 of  5
                              PROTOCOL VARIATIONS

                                       Mark Users as Phone? y
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                       Send Transferring Party Information? y
                                  Network Call Redirection? y
          Build Refer-To URI of REFER From Contact For NCR? n
                                     Send Diversion Header? n
                                   Support Request History? y
                              Telephone Event Payload Type: 101

                      Convert 180 to 183 for Early Media? n
                Always Use re-INVITE for Display Updates? n
                      Identity for Calling Party Display: From
          Block Sending Calling Party Location in INVITE? n
               Accept Redirect to Blank User Destination? n
                                            Enable Q-SIP? n

        Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
                              Request URI Contents: may-have-extra-digits
```

5.3. Configure Call Routing for SIP phones

For compliance testing all calls beginning with 31 with a total length of 4 digits were to be sent across the SIP trunk to Session Manager as all SIP phones begin with 31. Automatic Alternate Routing (aar) was used to route the calls.

5.3.1. Administer 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 **Section 5.5**, 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 **3**, FNEs are also four digits beginning with **1**, and the FACs have formats as indicated with a **Call Type** of **fac**, these begin with either a ***** or a **#** as shown in **Section 5.4**.

```
change dialplan analysis                                      Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all          Percent Full: 5

    Dialed   Total  Call       Dialed   Total  Call       Dialed   Total  Call
    String   Length Type       String   Length Type       String   Length Type
   1           4   udp
   2           4   udp
   3           4   ext
   5           4   udp
   6           4   ext
   8           1   fac
   9           1   fac
   *8          4   dac
   *           3   fac
   #           3   fac
```

5.3.2. Administer Route Selection for SIP Phones

Use the **change aar analysis** x command to further configure the routing of the dialed digits. Calls to SIP phones begin with **31** and are matched with the AAR entry shown below. Calls are sent to **Route Pattern 11**, which contains the outbound SIP Trunk Group.

```
change aar analysis 3                                        Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all          Percent Full: 1

          Dialed            Total     Route     Call   Node   ANI
          String           Min  Max  Pattern    Type   Num    Reqd
    31                     4    4     11        lev0          n
    5                      7    7     999       aar           n
    666                    4    4     66        aar           n
    7                      7    7     999       aar           n
    8                      7    7     999       aar           n
    9                      7    7     999       aar           n
                                                              n
                                                              n
```

Use the **change route-pattern *n*** command to add the SIP trunk group to the route pattern that AAR selects. In this configuration, **Route Pattern Number 11** is used to route calls to trunk group (**Grp No**) **11**. This is the SIP Trunk configured in **Section 5.2**.

```
change route-pattern 11                                         Page   1 of   4
               Pattern Number: 1   Pattern Name: SIP Phones
           SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No. Inserted                               DCS/ IXC
    No          Mrk Lmt List Del  Digits                                QSIG
                             Dgts                                       Intw
 1: 11   0                                                               n   user
 2:                                                                      n   user
 3:                                                                      n   user
 4:                                                                      n   user
 5:                                                                      n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM   No. Numbering  LAR
    0 1 2 M 4 W     Request                                  Dgts  Format
 1: y y y y y n  n            unre                                 lev0-pvt  none
 2: y y y y y n  n            rest                                           none
 3: y y y y y n  n            rest                                           none
 4: y y y y y n  n            rest                                           none
 5: y y y y y n  n            rest                                           none
 6: y y y y y n  n            rest                                           none
```

5.4. Define Feature Access Codes (FACs)

A FAC (feature access code) should be defined for each feature that will be used via the OPS FNEs. These are the FAC's that were used during compliance testing, these will be configured differently for every site. The FACs used in the sample configuration are shown in bold.

```
change feature-access-codes                                     Page   1 of  12
                            FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code: *11
         Abbreviated Dialing List2 Access Code: *12
         Abbreviated Dialing List3 Access Code: *13
Abbreviated Dial - Prgm Group List Access Code: *10
                      Announcement Access Code: *27
                       Answer Back Access Code: #02
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 8
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation: *05    Deactivation: #05
Call Forwarding Activation Busy/DA: *03    All: *04    Deactivation: #04
   Call Forwarding Enhanced Status: *73    Act: *74    Deactivation: #74
                         Call Park Access Code: *02
                       Call Pickup Access Code: *09
CAS Remote Hold/Answer Hold-Unhold Access Code:
                  CDR Account Code Access Code: *14
                        Change COR Access Code:
                   Change Coverage Access Code:
            Conditional Call Extend Activation:        Deactivation:
                   Contact Closure   Open Code:          Close Code:
```

Some other Feature Access Codes used.

```
display feature-access-codes                                 Page   2 of  12
                               FEATURE ACCESS CODE (FAC)
                   Contact Closure  Pulse Code:

                  Data Origination Access Code:
                      Data Privacy Access Code:
              Directed Call Pickup Access Code: *29
       Directed Group Call Pickup  Access Code:
     Emergency Access to Attendant Access Code:
        EC500 Self-Administration Access Codes: *61     *62     *63     *64
                     Enhanced EC500 Activation: *60    Deactivation: #60
            Enterprise Mobility User Activation:       Deactivation:
  Extended Call Fwd Activate Busy D/A      All: *06    Deactivation: #06
        Extended Group Call Pickup Access Code:
               Facility Test Calls Access Code:
                             Flash Access Code:
              Group Control Restrict Activation:       Deactivation:
                     Hunt Group Busy Activation: *30    Deactivation: #30
                              ISDN Access Code:
                Last Number Dialed Access Code: *08
     Leave Word Calling Message Retrieval Lock: *15
   Leave Word Calling Message Retrieval Unlock: #15:
```

```
display feature-access-codes                                 Page   3 of  12
                               FEATURE ACCESS CODE (FAC)
             Leave Word Calling Send A Message: *16
           Leave Word Calling Cancel A Message: #16
   Limit Number of Concurrent Calls Activation: *18    Deactivation: #18
               Malicious Call Trace Activation: *17    Deactivation: #17
         Meet-me Conference Access Code Change:
          Message Sequence Trace (MST) Disable:

 PASTE (Display PBX data on Phone) Access Code: *28
  Personal Station Access (PSA) Associate Code: *20    Dissociate Code: #20
        Per Call CPN Blocking Code Access Code: *24
      Per Call CPN Unblocking Code Access Code: #24
                    Posted Messages Activation:        Deactivation:
                  Priority Calling Access Code: *07
                           Program Access Code: *00

       Refresh Terminal Parameters Access Code: #28
              Remote Send All Calls Activation: #11    Deactivation:
               Self Station Display Activation:
                   Send All Calls Activation: *01    Deactivation: #01
         Station Firmware Download Access Code:
```

5.5. Define Feature Name Extensions (FNEs)

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

```
display off-pbx-telephone feature-name-extensions set 1      Page   1 of   3

     EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME
                     Set Name: PG

        Active Appearance Select:
             Automatic Call Back: 1301
      Automatic Call-Back Cancel: 1302
                Call Forward All:
     Call Forward Busy/No Answer:
             Call Forward Cancel:
                       Call Park: 1303
           Call Park Answer Back: 1304
                    Call Pick-Up: 1309
            Calling Number Block:
          Calling Number Unblock:
  Conditional Call Extend Enable:
 Conditional Call Extend Disable:
             Conference Complete:
            Conference on Answer:
           Directed Call Pick-Up: 1310
           Drop Last Added Party:
```

```
display off-pbx-telephone feature-name-extensions set 1      Page   2 of   3

     EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME

       Exclusion (Toggle On/Off):
      Extended Group Call Pickup:
          Held Appearance Select:
          Idle Appearance Select:
              Last Number Dialed: 1305
            Malicious Call Trace:
     Malicious Call Trace Cancel:
             Off-Pbx Call Enable:
            Off-Pbx Call Disable:
                   Priority Call:
                          Recall:
                  Send All Calls: 1306
           Send All Calls Cancel: 1307
               Transfer Complete:
             Transfer On Hang-Up:
           Transfer to Voice Mail:
         Whisper Page Activation: 1311
```

5.6. Configure Class of Service (COS)

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

```
display cos-group 1                                           Page   1 of   2
CLASS OF SERVICE         COS Group: 1   COS Name: PG Default

                               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  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  y  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  y  y  y  y  y  y  y  y  y  y  y  y  y  y  y
Call Forwarding Busy/DA        n  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  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  n  n  n  n  n
QSIG Call Offer Originations   n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
Contact Closure Activation     n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
```

```
display cos-group 1                                           Page   2 of   2
                                CLASS OF SERVICE

                               0  1  2  3  4  5  6  7  8  9 10 11 12 13 14 15
VIP Caller                     n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n

Masking CPN/Name Override      n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
Call Forwarding Enhanced       y  y  y  y  y  y  y  y  y  y  y  y  y  y  y  y
Priority Ip Video              n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
Ad-hoc Video Conferencing      n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
MOC Control:                   n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
Match BCA Display To Principal n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
DCC Activation/Deactivation    n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
Bridging Exclusion Override    n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
```

5.7. Configure Class of Restriction (COR)

The COR that was used during compliance testing is shown below. 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**.

```
display cor 1                                               Page   1 of  43
                              CLASS OF RESTRICTION

               COR Number: 1
          COR Description: PG Default

                      FRL: 0                                APLT? y
  Can Be Service Observed? y           Calling Party Restriction: none
Can Be A Service Observer? y            Called Party Restriction: none
        Time of Day Chart: 1       Forced Entry of Account Codes? n
         Priority Queuing? n                Direct Agent Calling? n
     Restriction Override: none       Facility Access Trunk Test? y
     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.? n
         Send ANI for MFE? n             Add/Remove Agent Skills? y
            MF ANI Prefix:              Automatic Charge Display? n
Hear System Music on Hold? y   PASTE (Display PBX Data on Phone)? n
                        Can Be Picked Up By Directed Call Pickup? y
                                    Can Use Directed Call Pickup? y
                                    Group Controlled Restriction: inactive
```

5.8. Configure Coverage Path

The coverage path configuration is shown below. The default values shown for **Busy**, **Don't Answer**, and **DND/SAC/Goto Cover** can be used for the **Coverage Criteria**.

The coverage path setup used for compliance testing is illustrated below. Note the following:

Don't Answer is set to **y:**	The coverage path will be used in the event the phone set is not answered.
Number of Rings is set to **3:**	The coverage path will be used after 3 rings.
Point 1 is set to **h66**	Hunt Group 66 is utilised by this coverage path.

```
display coverage path 3
                                 COVERAGE PATH

                   Coverage Path Number: 3
     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: 3
                All?             n               n
 DND/SAC/Goto Cover?             y               y
   Holiday Coverage?             n               n


COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h66            Rng: 3  Point2:
  Point3:                        Point4:
  Point5:                        Point6:
```

The hunt group used for compliance testing is shown below. Note that on **Page 1** the **Group Extension** is **6666**, which is used to dial for messaging and **Group Type** is set to **ucd-mia**.

```
display hunt-group 66                                           Page   1 of  60
                                  HUNT GROUP

            Group Number: 66                                ACD? n
              Group Name: Messaging                       Queue? n
         Group Extension: 6666                           Vector? n
              Group Type: ucd-mia                  Coverage Path: 1
                      TN: 1            Night Service Destination:
                     COR: 1                    MM Early Answer? n
           Security Code:                 Local Agent Preference? n
 ISDN/SIP Caller Display:



SIP URI::
```

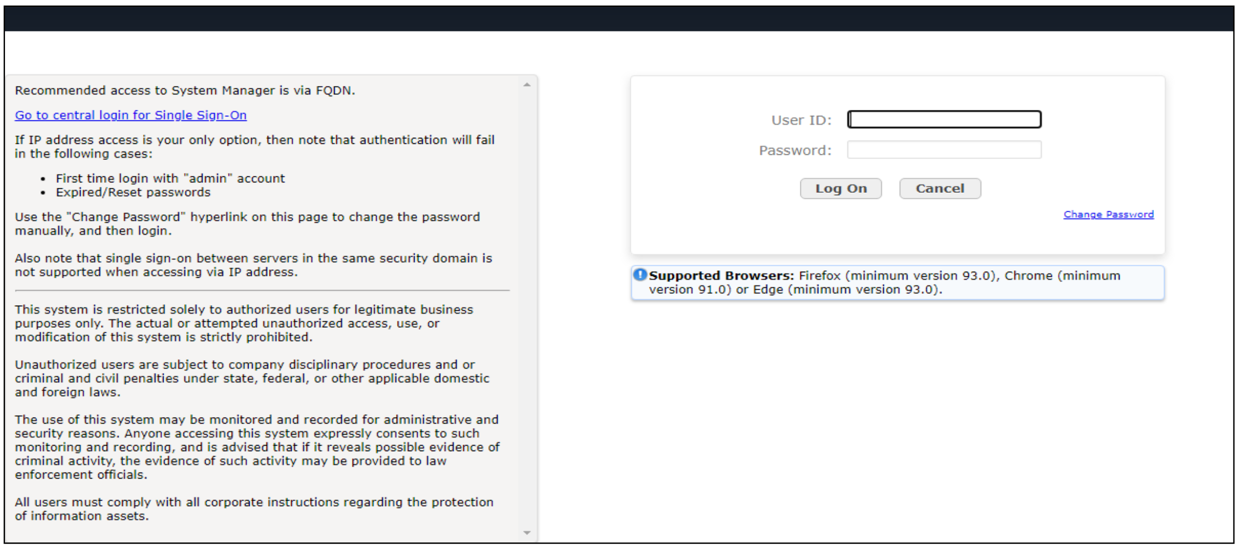
On **Page 2 Message Center** is set to **sip-adjunct**.

```
display hunt-group 66                                           Page   2 of  60
                                  HUNT GROUP






                       Message Center: sip-adjunct

     Voice Mail Number         Voice Mail Handle          Routing Digits
                                                  (e.g., AAR/ARS Access Code)
     6666                      6666                       8
```

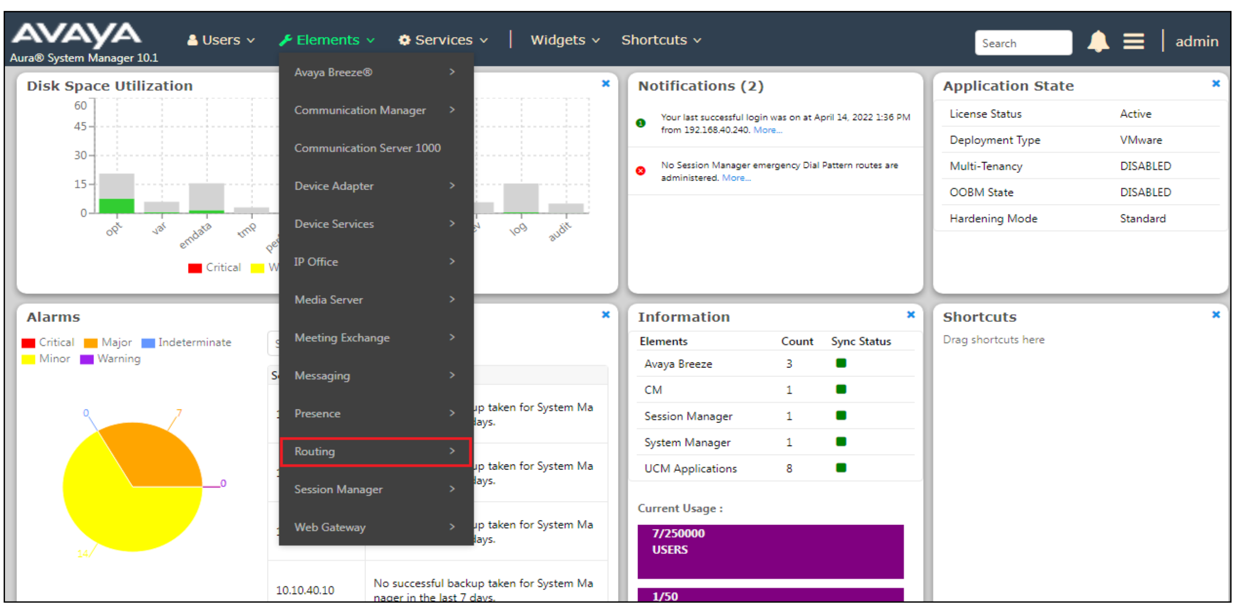
6. Configure Avaya Aura® Session Manager

This section describes aspects of the Session Manager configuration required for interoperating with Speakerbus. It is assumed that the Domains, Locations, SIP entities for each Session Manager, Communication Manager and Aura Messaging, Entity Links, Routing Policies, Dial Patterns and Application Sequences have been configured.

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



Once logged in navigate to **Elements** and click on **Routing** highlighted below.

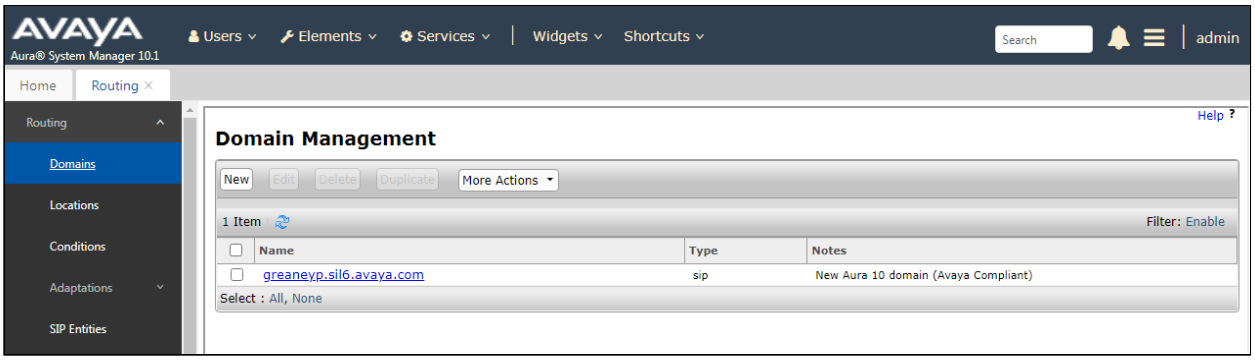


6.1. Domains and Locations

Note: It is assumed that a domain and a location have already been configured, therefore a quick overview of the domain and location that was used in compliance testing is provided here.

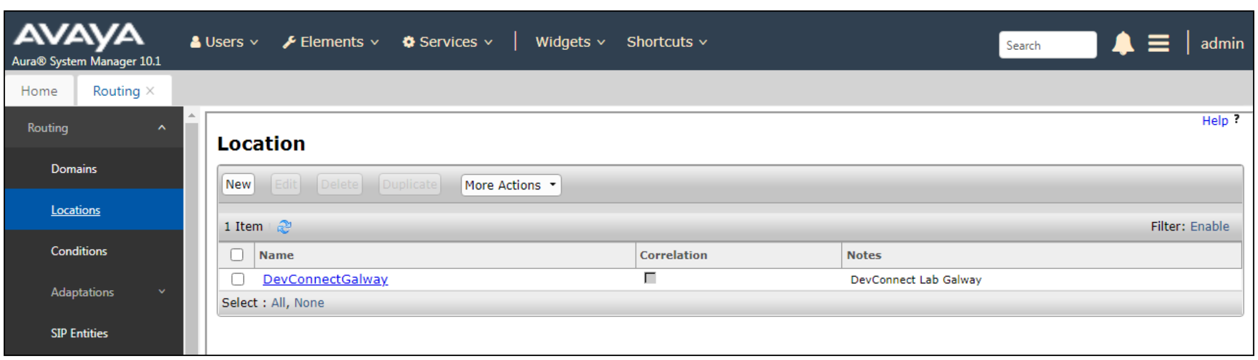
6.1.1. Display the Domain

Select **Domains** from the left window. This will display the domain configured on Session Manager. For compliance testing this domain was **greaneyp.sil6.avaya.com** as shown below. If a domain is not already in place, click on **New**. This will open a new window (not shown) where the domain can be added.



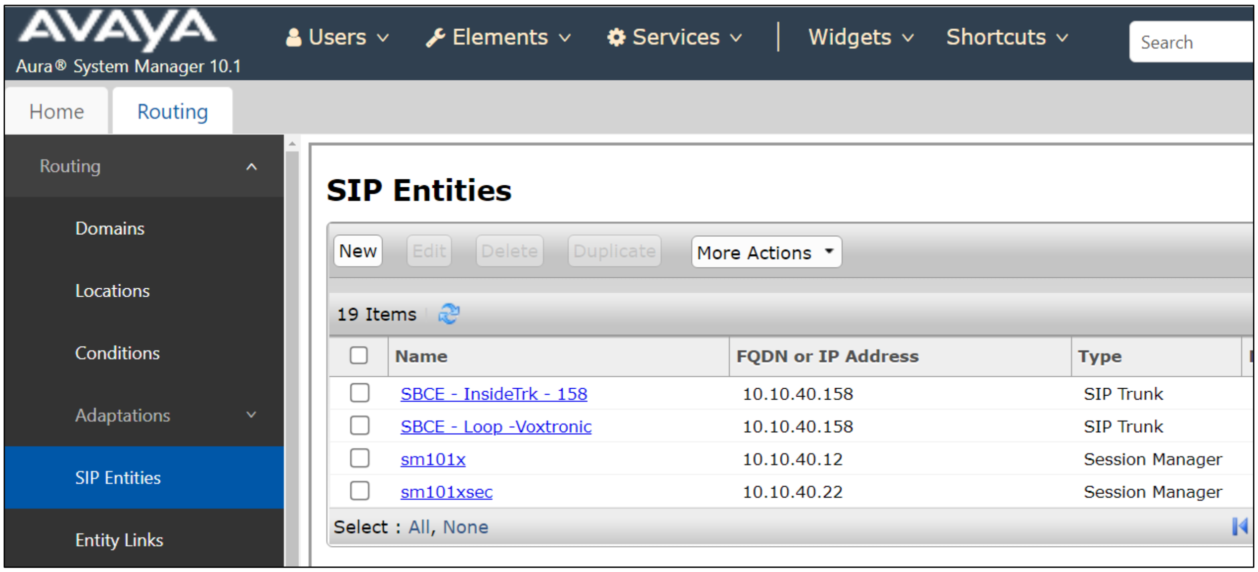
6.1.2. Display the Location

Select **Locations** from the left window and this will display the location setup. The example below shows the location **DevConnectGalway** which was used for compliance testing. If a location is not already in place, then one must be added to include the IP address range of the Avaya solution. Click on **New** to add a new location.

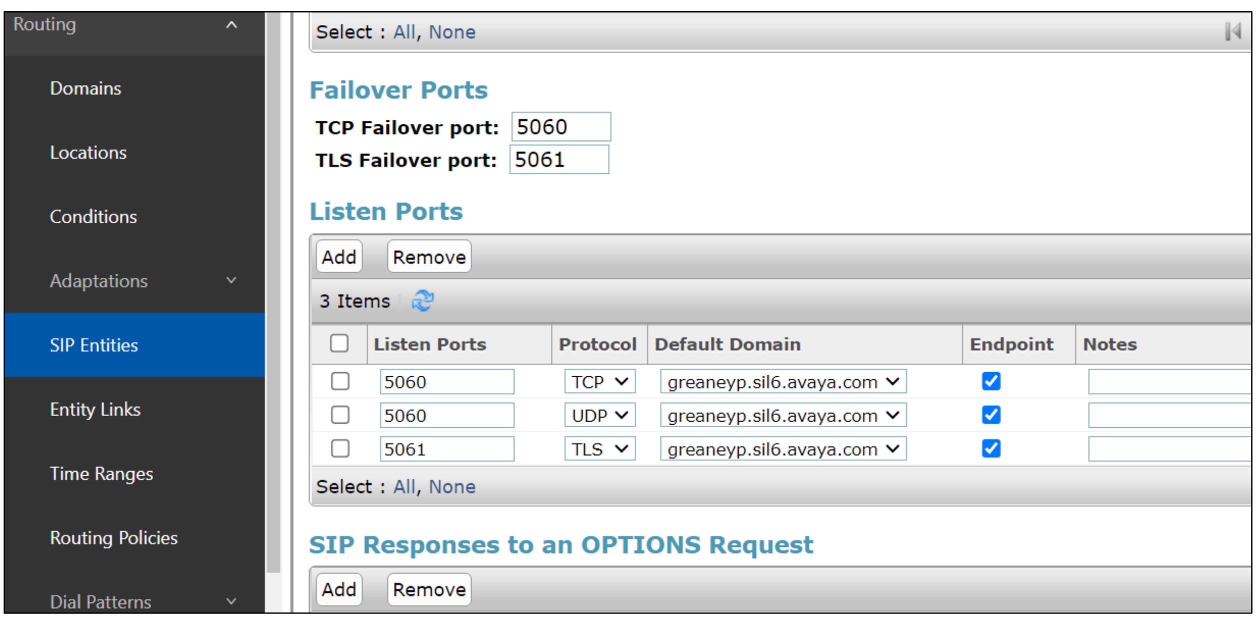


6.2. Configure Ports for Speakerbus Registration

Each Session Manager Entity must be configured so that the Speakerbus iTurret can register to it using either TCP or UDP. From the web interface click **Routing → SIP Entities → <Session Manager>** (**sm101x** in the example below).

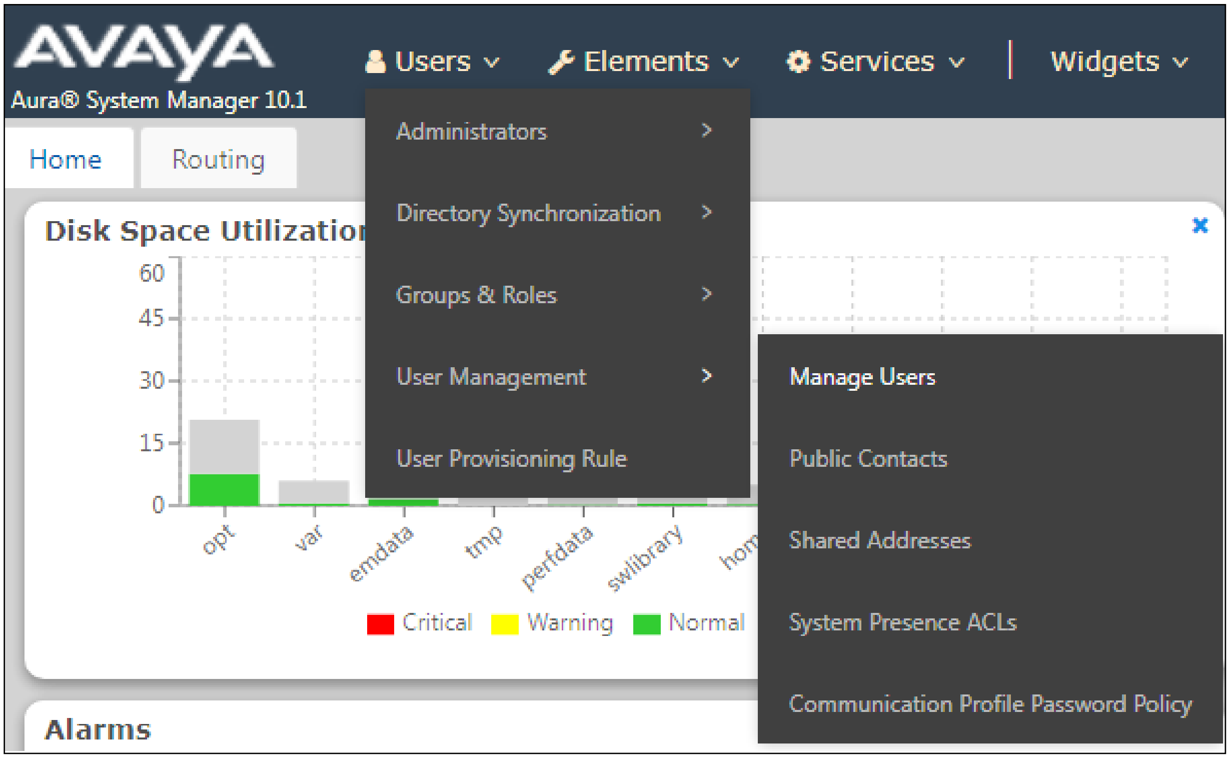


In the **Port** section, ensure that port **5060** of type **UDP** and **TCP** are added as shown below. This is the port the Speakerbus iTurret sends its SIP registration to. Select the appropriate SIP domain from the drop-down list and **Endpoint** is also ticked. Click **Commit** when done (not shown). Note that Avaya phones use **TLS** port **5061** which was also configured.



6.3. Add Primary iTurret User

A user must be added for each iTurret. Click **User Management → Manage Users**. Click on **New**, (not shown).

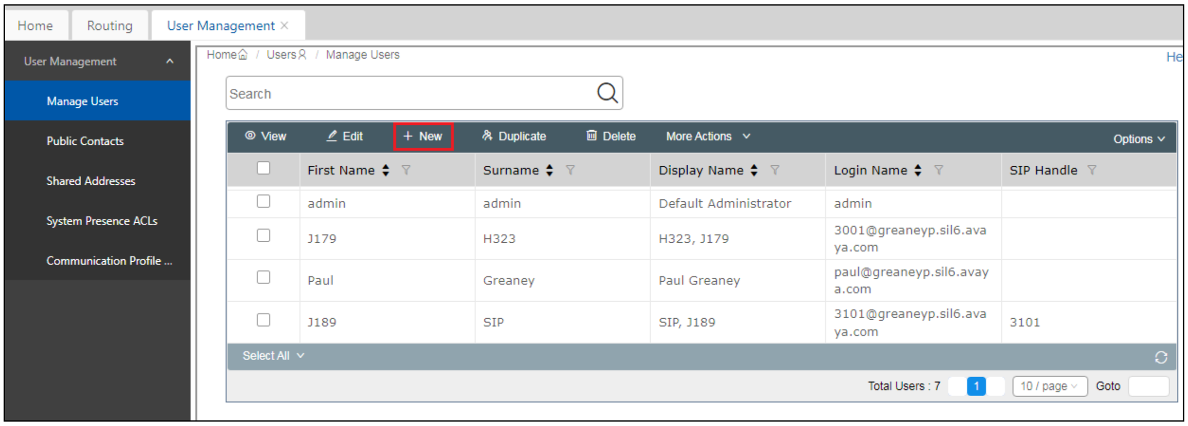


The iTurret uses 'bridged appearance' to enable calls to be presented and picked up at different iTurret endpoints. A site may have a group of say five iTurrets all with each other's extensions represented as bridged appearances so as each of them will display and can answer each other's calls. This may be different on every site and in some cases perhaps only two out of the five may have bridged appearances there is no set rule on how the buttons should or would be configured. What is shown in the next section is one iTurret which has its own call appearance and bridged appearances of extensions 3181 and 3182. It also has bridged appearances of 3191 and 3192 which are 'Privacy' extensions used specifically for making active calls private.

A user of a multi-appearance telephone can activate Privacy, a Manual Exclusion to keep the participants with appearance of the same extension from bridging on to an existing call. To use manual exclusion, the user presses the privacy button, either before the user places the call, or when the user is active on the call. If the user presses the privacy button while others are bridged onto the call, the iTurret rejects the privacy request with a message but keeps the call active. To turn off manual exclusion, the user presses the privacy button.

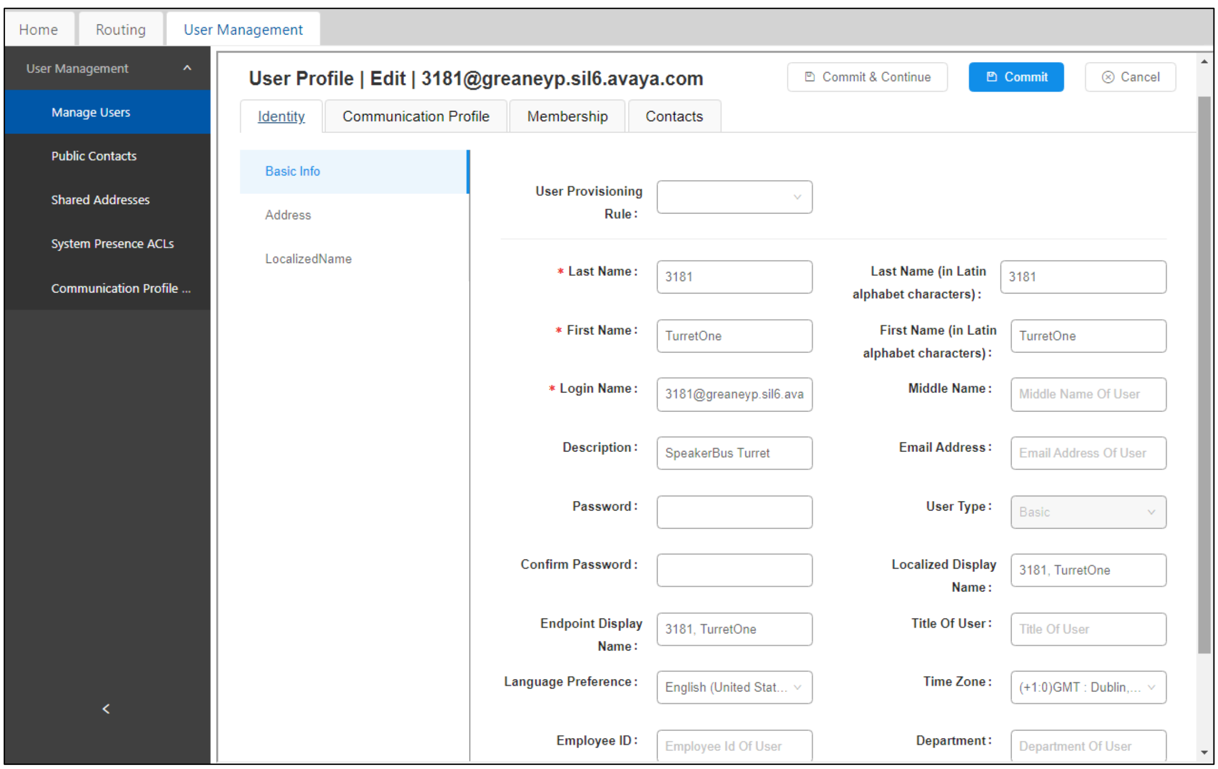
Note: The following screens will display an existing user 3181, the screens will show an edited user instead of a new user but the information that is displayed is the very same as that required to add a new user.

From **Manager Users** section, click on **New** to add a new SIP user.

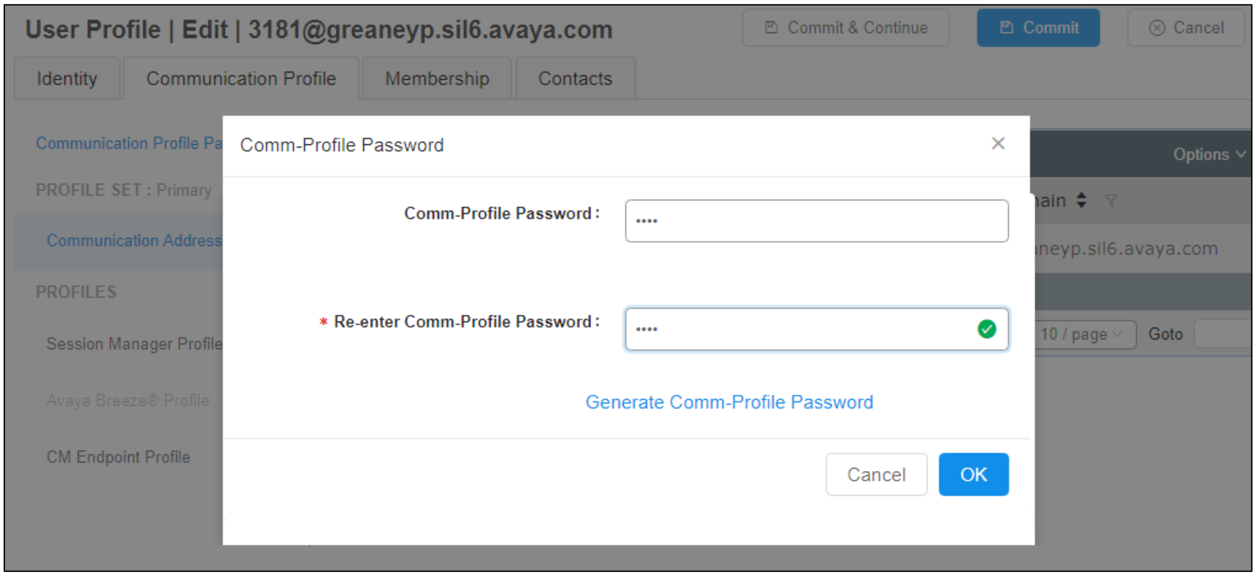


Configure as following in the **Identity** tab.

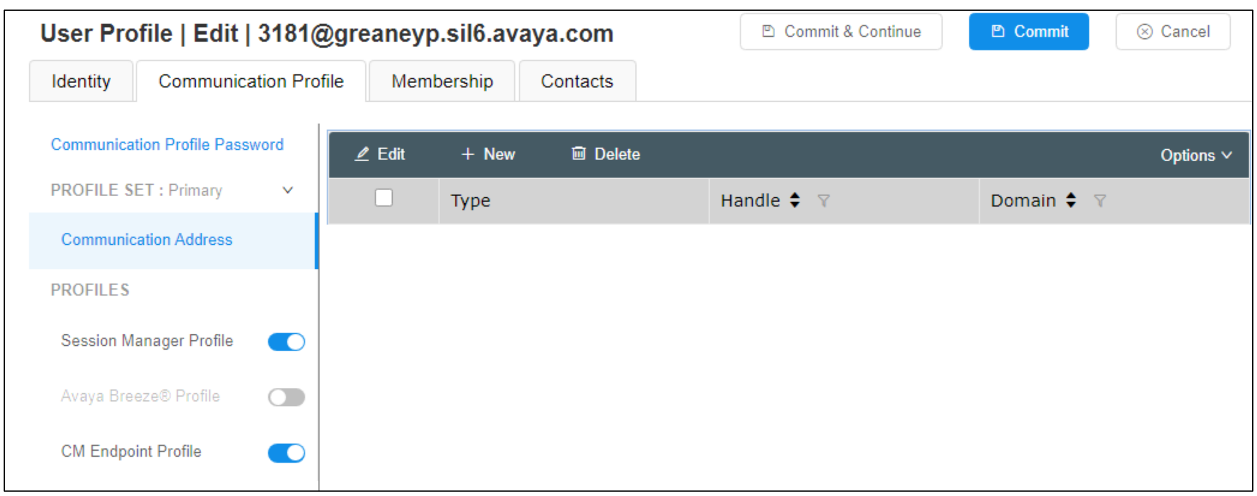
- **First Name** and **Last Name** — Enter an identifying name.
- **Login Name** — Enter the extension number followed by the domain, in this case **3181@greaneyp.sil6.avaya.com**.
- **Time Zone** — Enter the appropriate time zone.



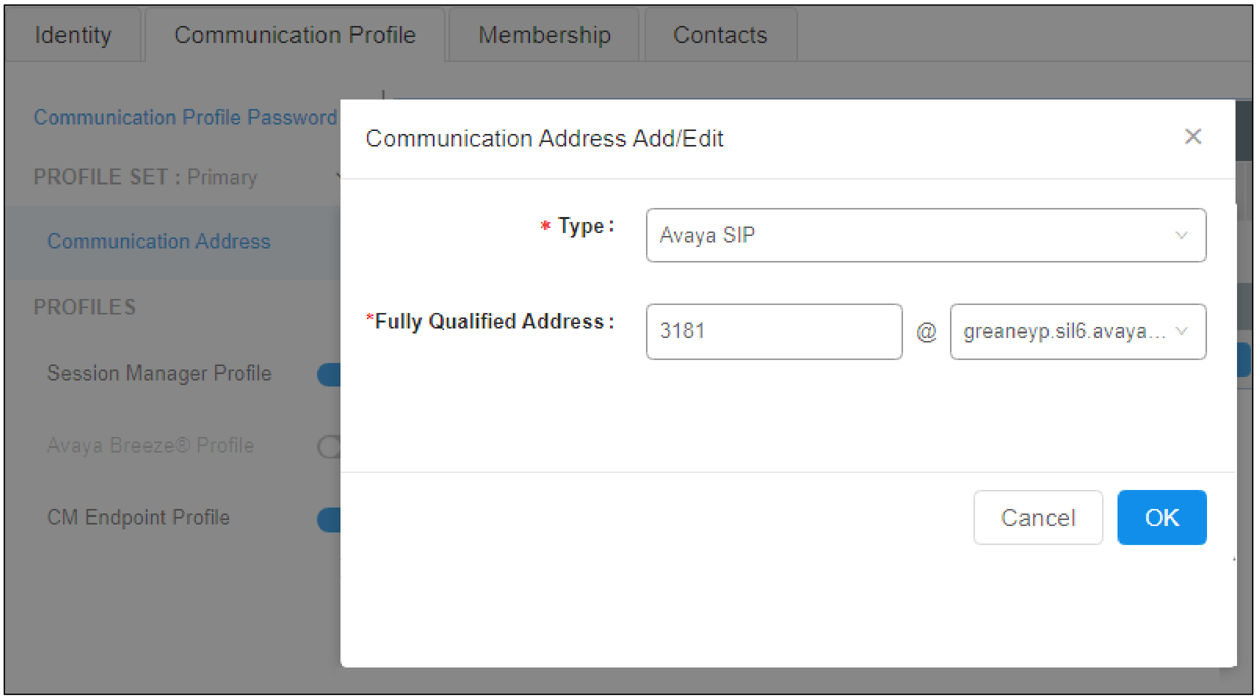
Click the **Communication Profile** tab and in the **Communication Profile Password** and **Confirm Password** fields, enter a numeric password. This will be used to register the iTurret during login and adding into Speakerbus iCMS / imanager configuration in **Section** Error! Reference source not found.. Click **OK** to continue.



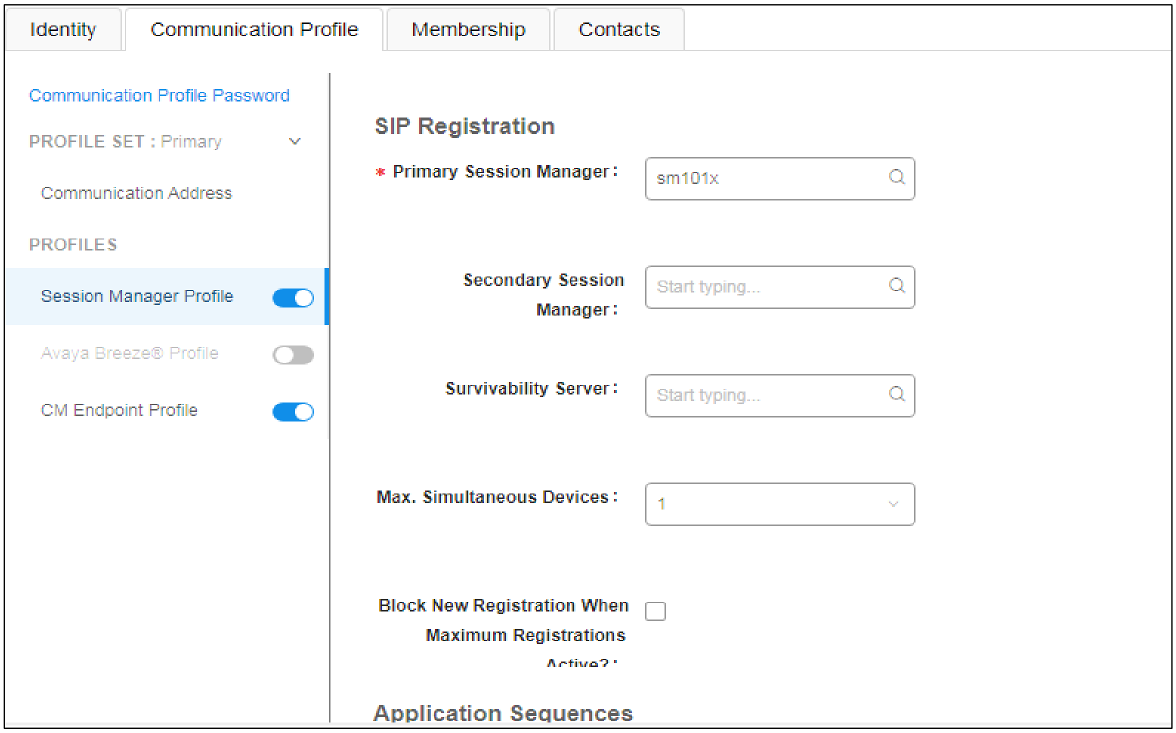
Select **Communication Address** in the left window and click **New** in the main window.

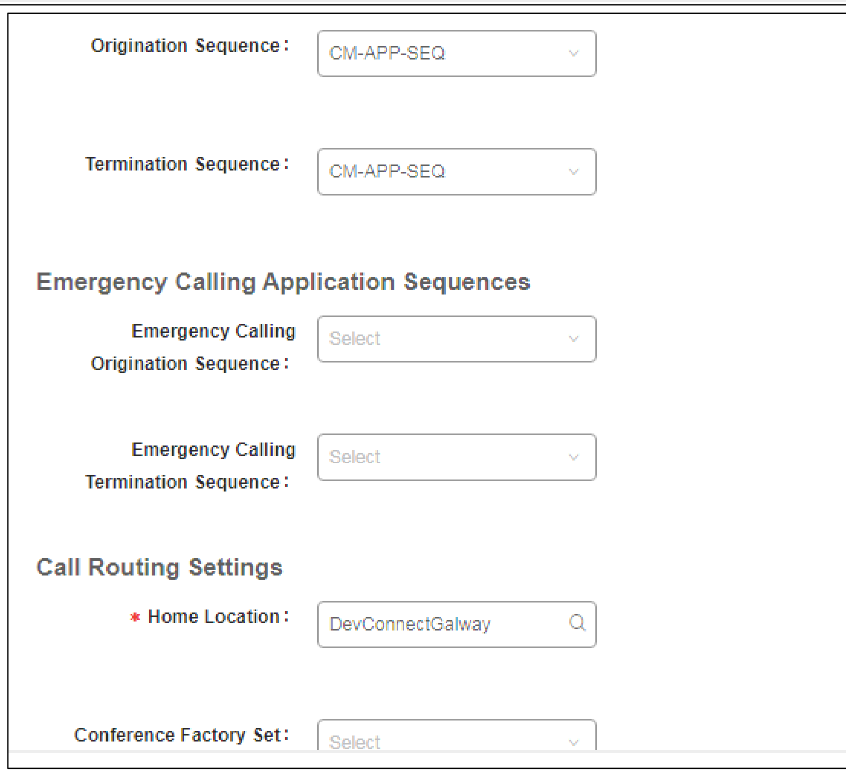


Select **Avaya SIP** from the drop-down list. In the **Fully Qualified Address** field enter the extension number as required and select the appropriate **Domain** from the drop-down list. Click **OK** when done.



Ensure **Session Manager Profile** is checked and enter the **Primary Session Manager** details, enter the **Origination Sequence** and the **Termination Sequence**. Scroll down to complete the profile. Enter the **Home Location**, this should be the location configured in **Section** Error! Reference source not found.. Click on Commit at the top of the page (not shown).

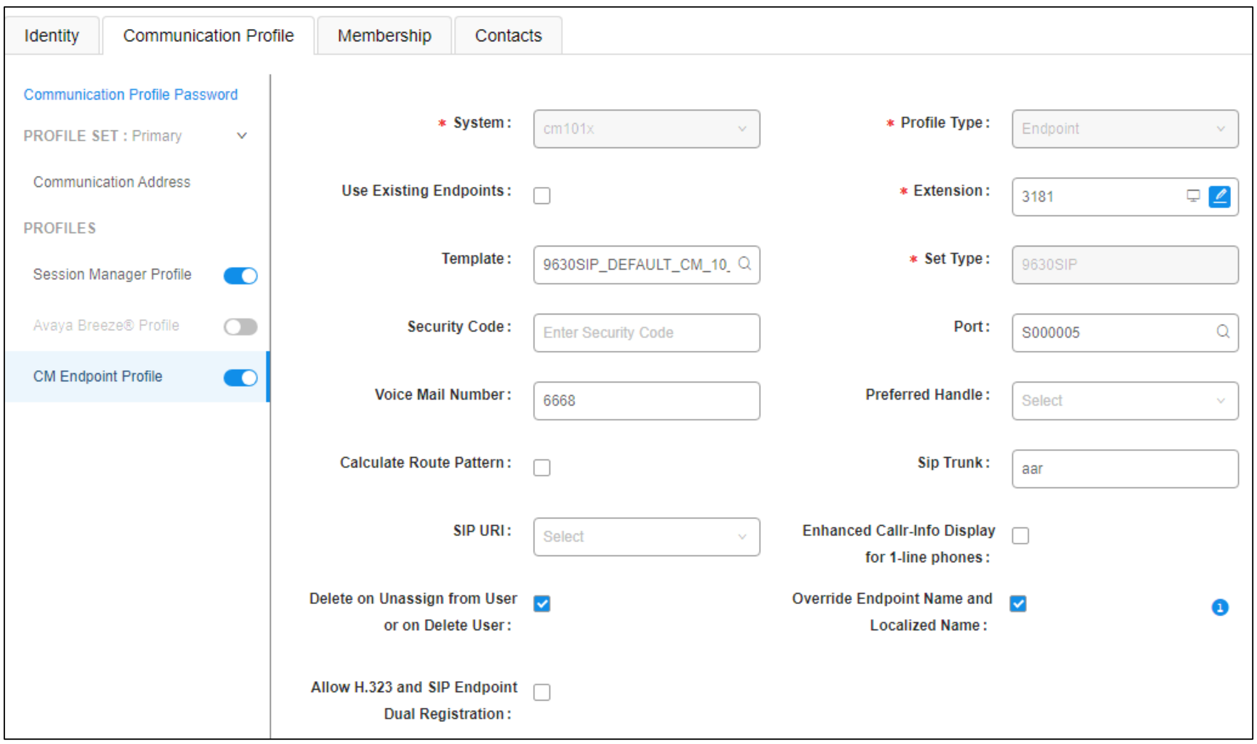




Place a tick in the **CM Endpoint Profile** bar and configure as follows:

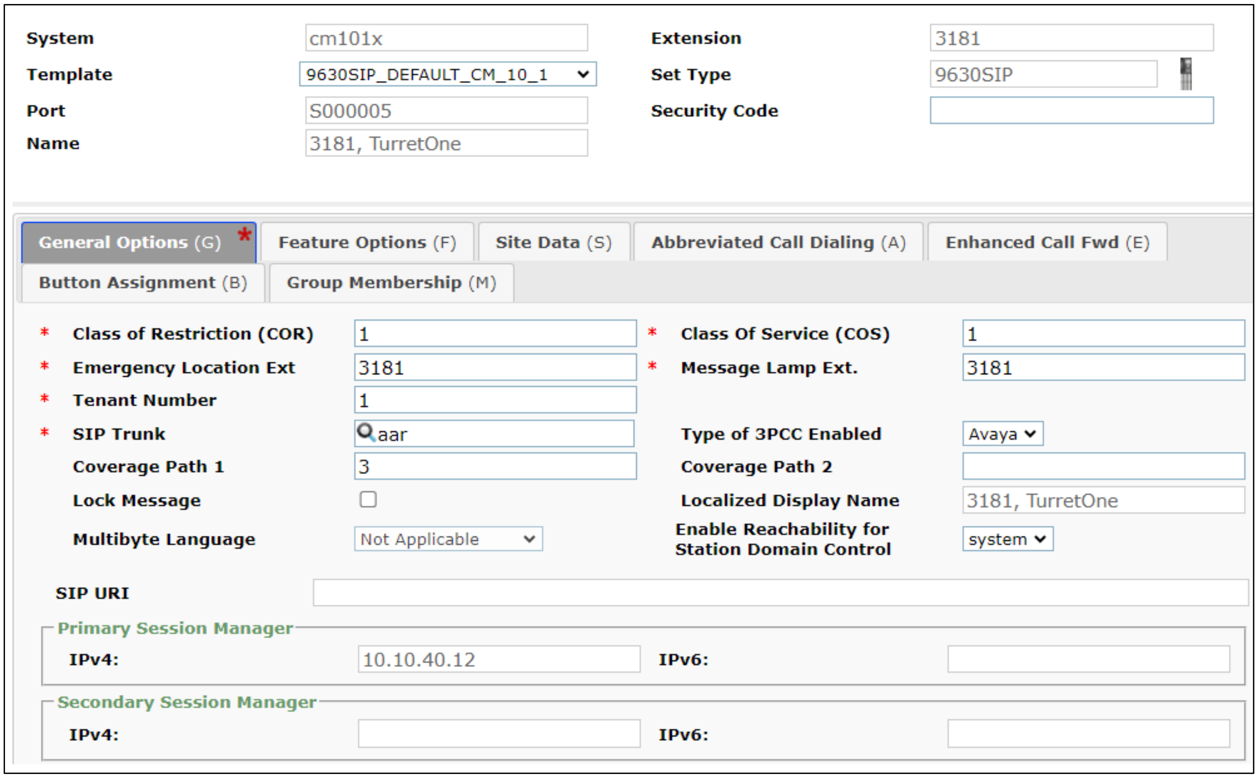
- **System** Select the relevant Communication Manager SIP Entity from the drop-down list.
- **Profile Type** Select **Endpoint** from the drop-down list.
- **Extension** Enter the required extension number, in this case **3181**.
- **Template** Select **DEFAULT_9630SIP_CM_10_1** from the drop-down list.
- **Port** Enter **IP**.
- **Sip Trunk** This was set to **aar** for compliance testing.

Click on the Endpoint Editor icon, (this is next to the **Extension** number), to open the Communication Manger configuration for this extension. This will allow the buttons to be administered as well as changes to Class of Service and Class of Restriction and other features.

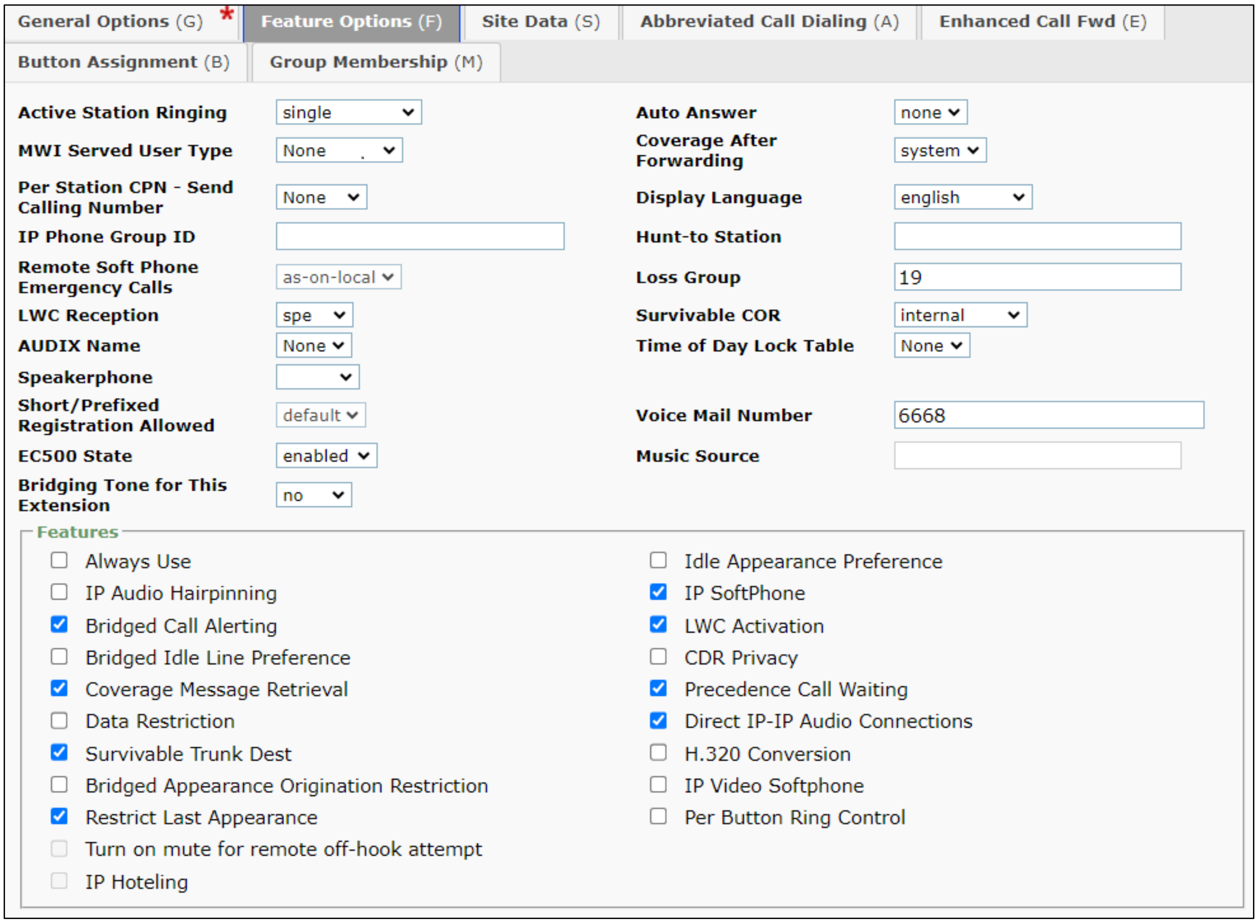


Click on the **General Options** tab and enter the following:

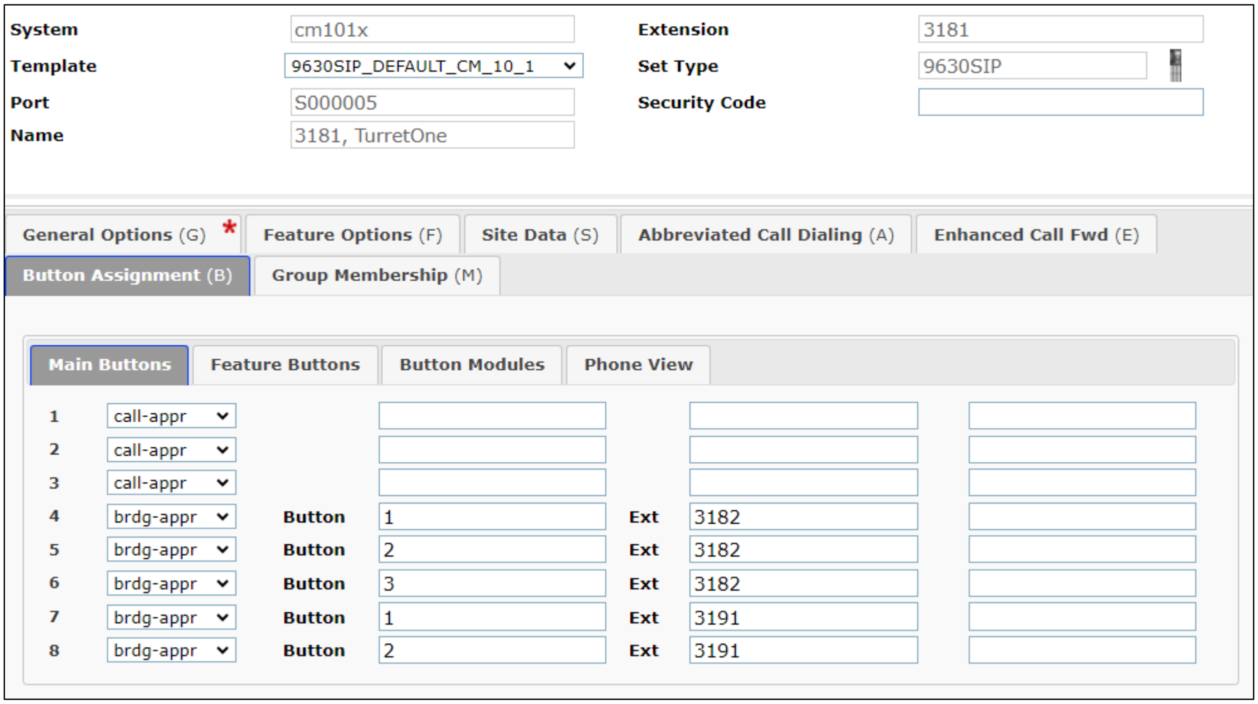
- **Class of Restriction (COR)** Enter the **COR** as configured in **Section 5.7**.
- **Emergency Location Ext** Enter **3181** (the extension for this user).
- **Tenant Number** Enter the appropriate **Tenant Number**.
- **SIP Trunk** Enter **aar**.
- **Class of Service (COS)** Enter the **COS** as configured in **Section 5.6**.
- **Message Lamp Ext.** Enter **3181** (the extension for this user).
- **Type of 3PCC Enabled** This was set to **Avaya** for compliance testing.
- **Coverage Path 1** This was set to the coverage path, as per **Section 5.8**.



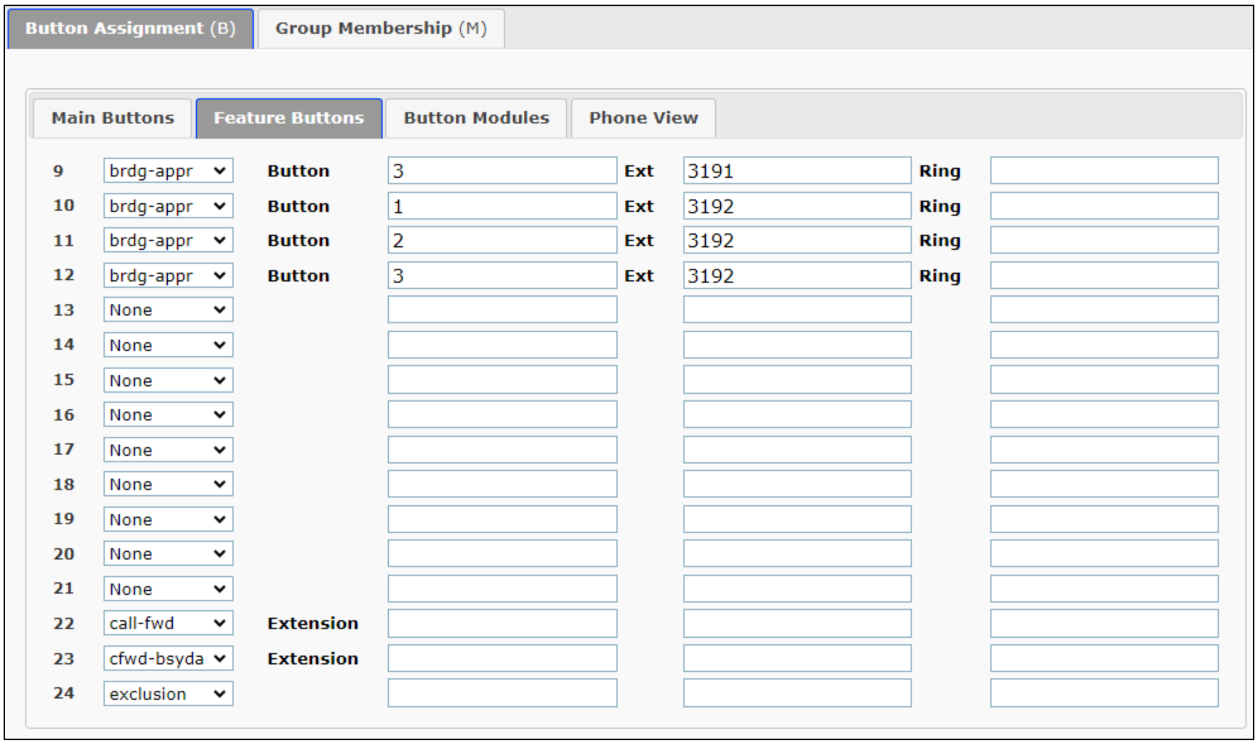
Click on the **Feature Options** tab. The screen shot below shows the Feature Options that were used during compliance testing. Ensure that **Bridged Call Alerting** is ticked as shown below, the other features are ticked as default.



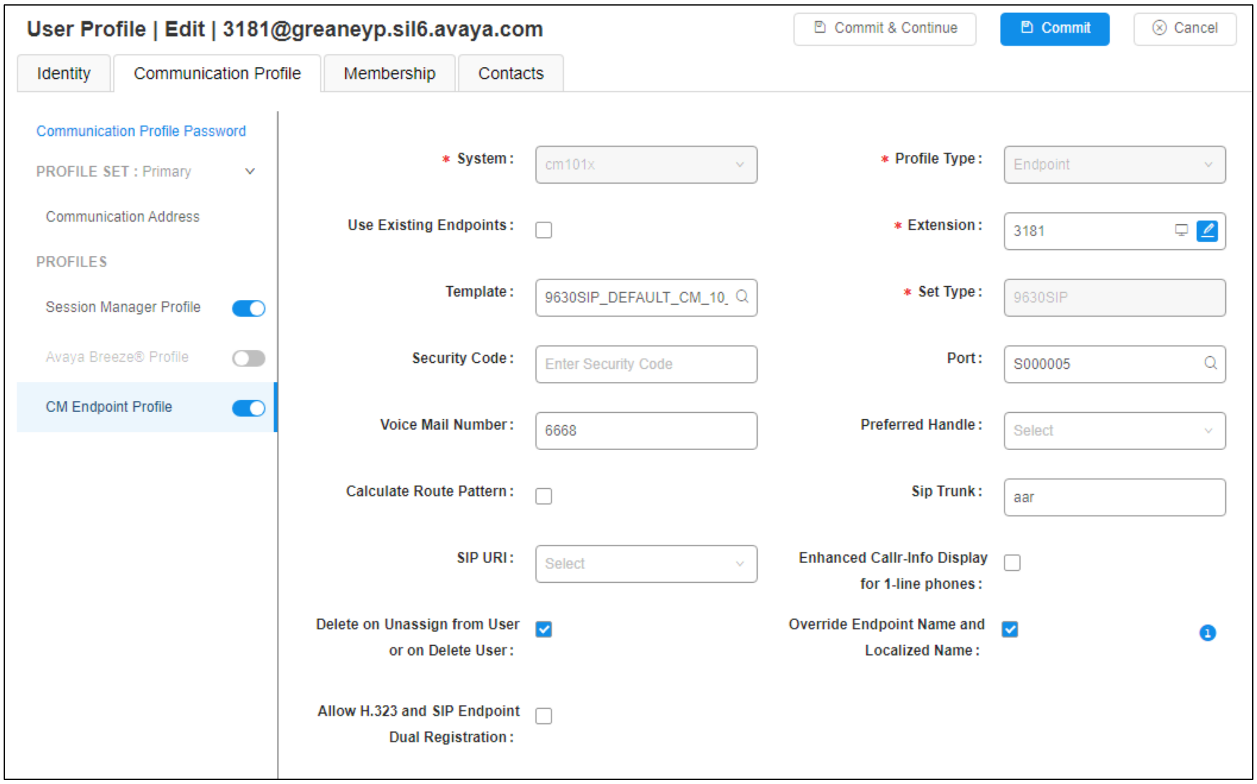
Click on the **Button Assignments tab** (**Main Buttons**) and configure Buttons **1**, **2** and **3** as **call-appr**. For compliance testing bridged appearances were configured to test 'Barge In' on buttons 4, 5 and 6. 'Privacy' buttons **7**, **8** and **9** were set to extension **3191** and **Feature Buttons 10**, **11** and **12** were set to **3192**.



Click on **Feature Buttons** and configure as per screen shot below. There were two SIP Users configured as 'Privacy Users' these were extensions **3191** and **3192**. To allow this user (3181) use Privacy, the privacy extension must be added as bridged appearances on this user's buttons as shown below. Buttons **10**, **11** and **12** were set to extension **3192**. Other features such as Call Forward and Call Forward Busy Deactivated as well as Exclusion are also added as buttons as shown. Click **Done** when all the configuration has been set correctly (not shown).

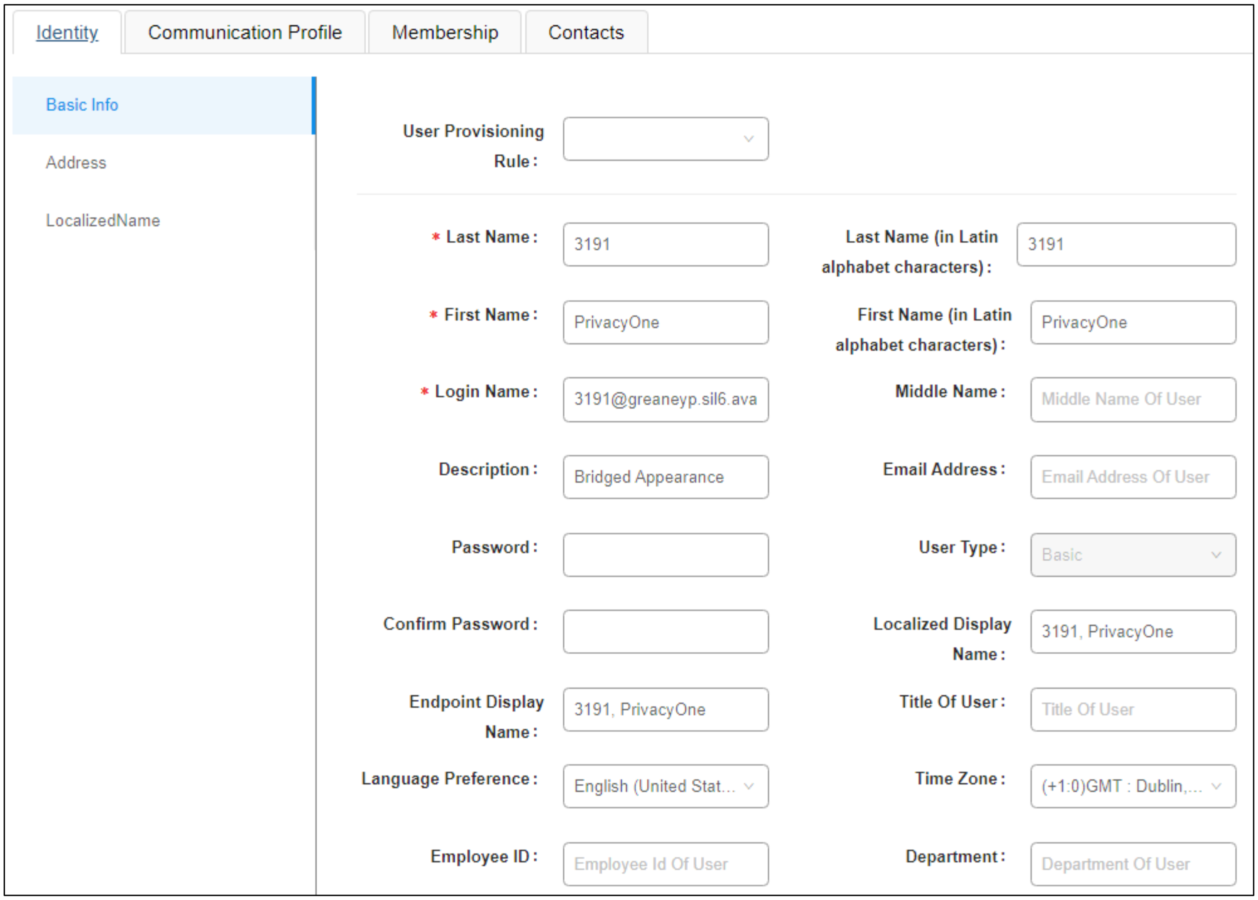


Click on **Commit** at the top of the screen to save the new user.

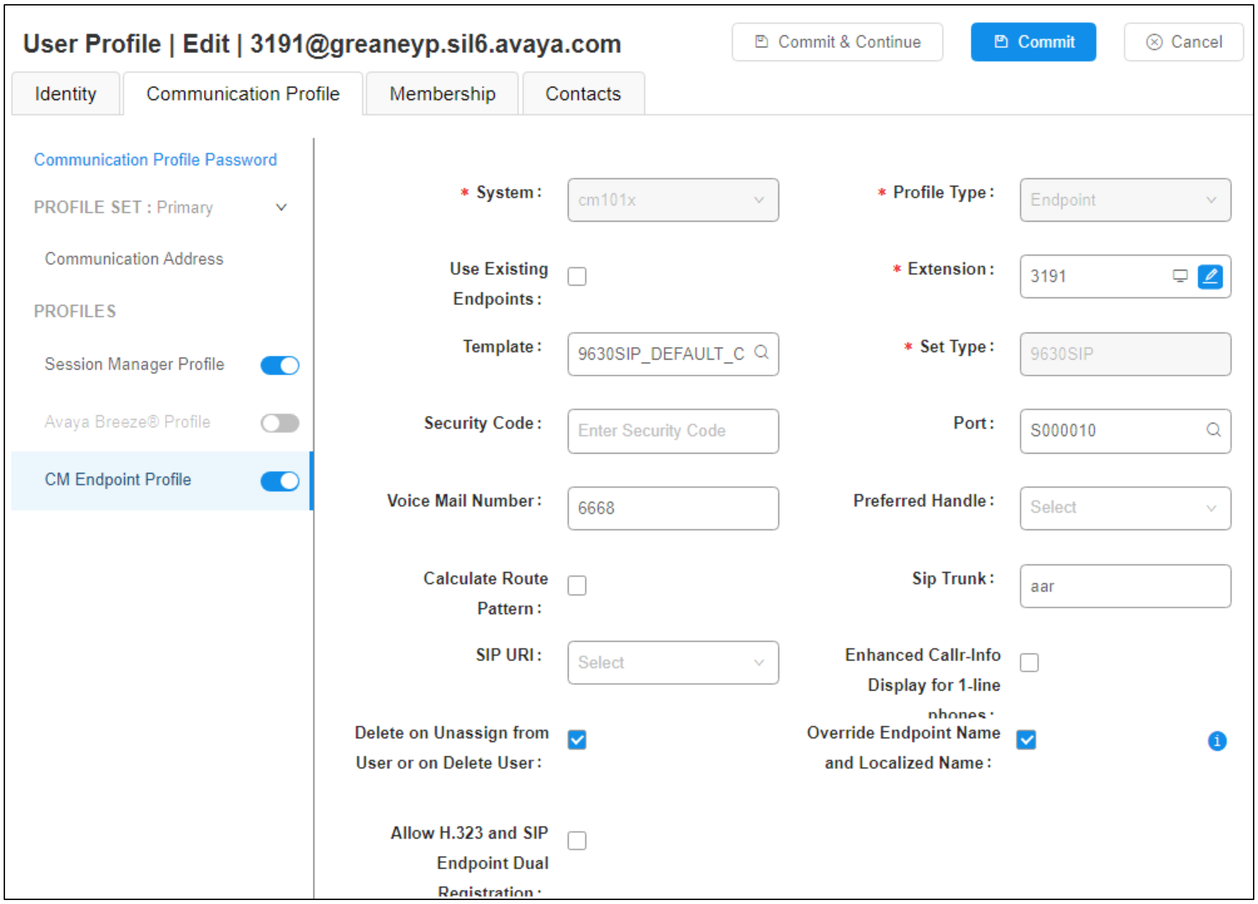


6.4. Configure Privacy Users

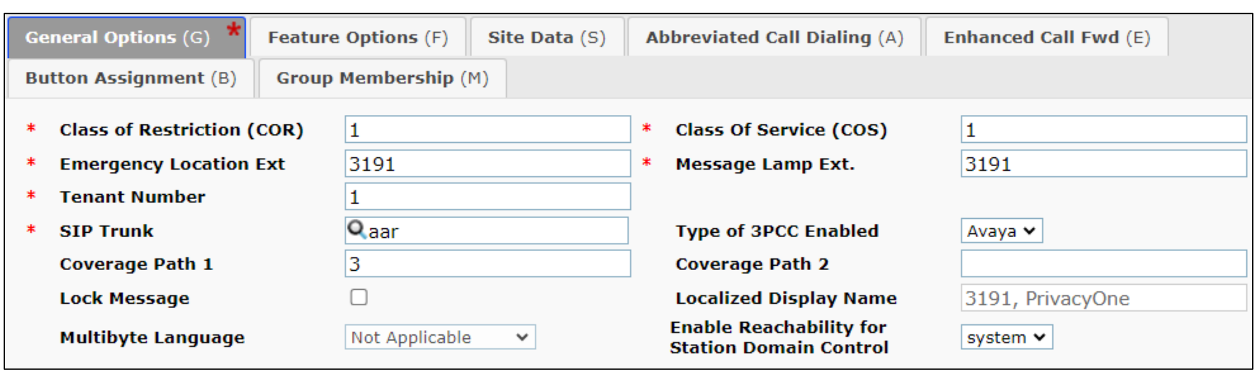
Privacy users are configured on System Manager as bridged appearances on the primary user. Add a 'Privacy User' in the same way as the primary user was configured in **Section 6.3**. Two privacy users 3191 and 3192 were created to be used by the primary user 3181. Following the same procedure as **Section 6.3**, under the **Identity** tab, enter a suitable **Name** and **Time Zone**.



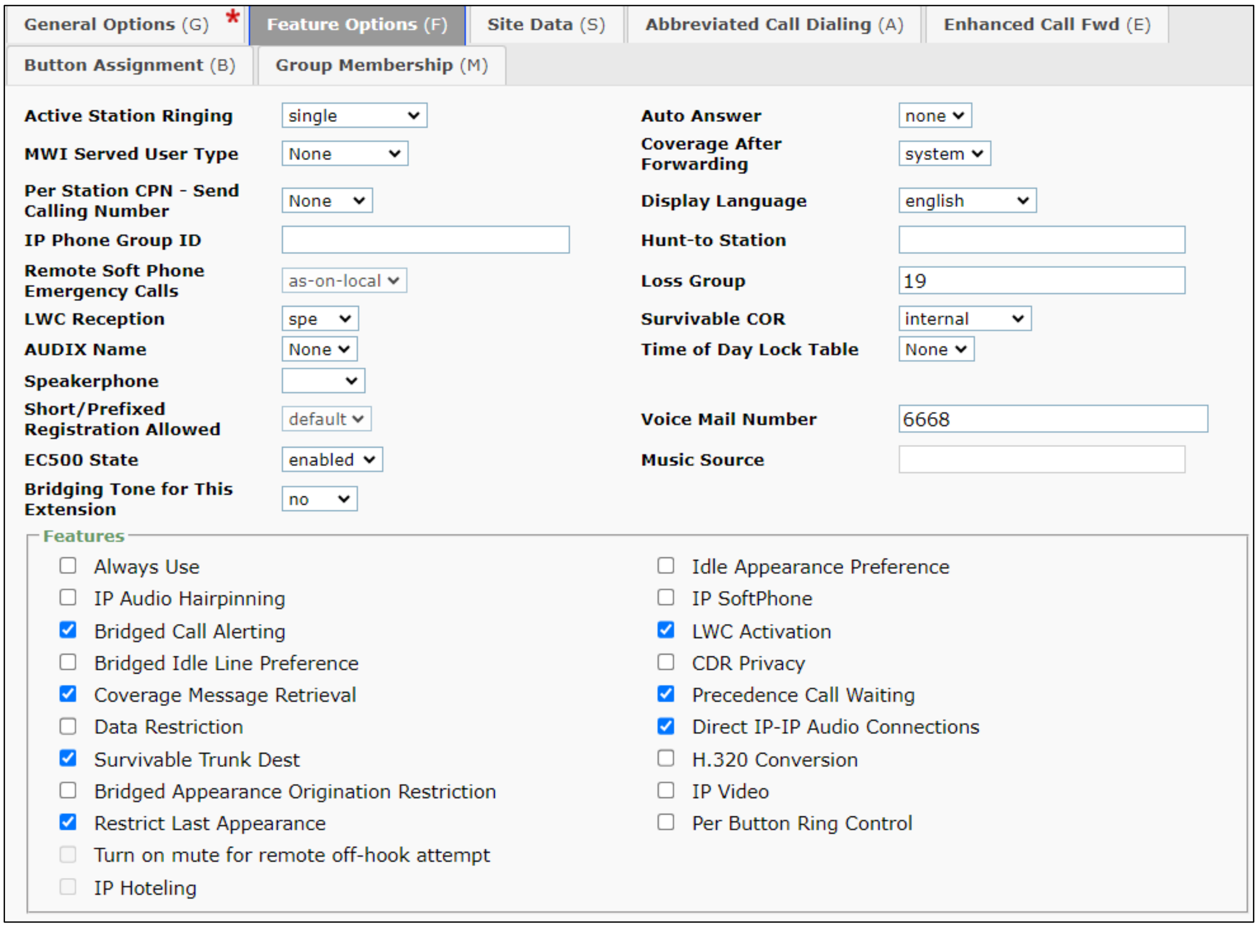
A **Communication Profile** and **Session Manager Profile** are added as per **Section 6.3**, (not shown here). Click on **CM Endpoint Profile** and enter the same **Template** information, that being **9630SIP_DEFAULT_CM_10_1**. Enter the appropriate **Extension** number (**3191**) and click on the "configure extension" icon, next to the Extension number.



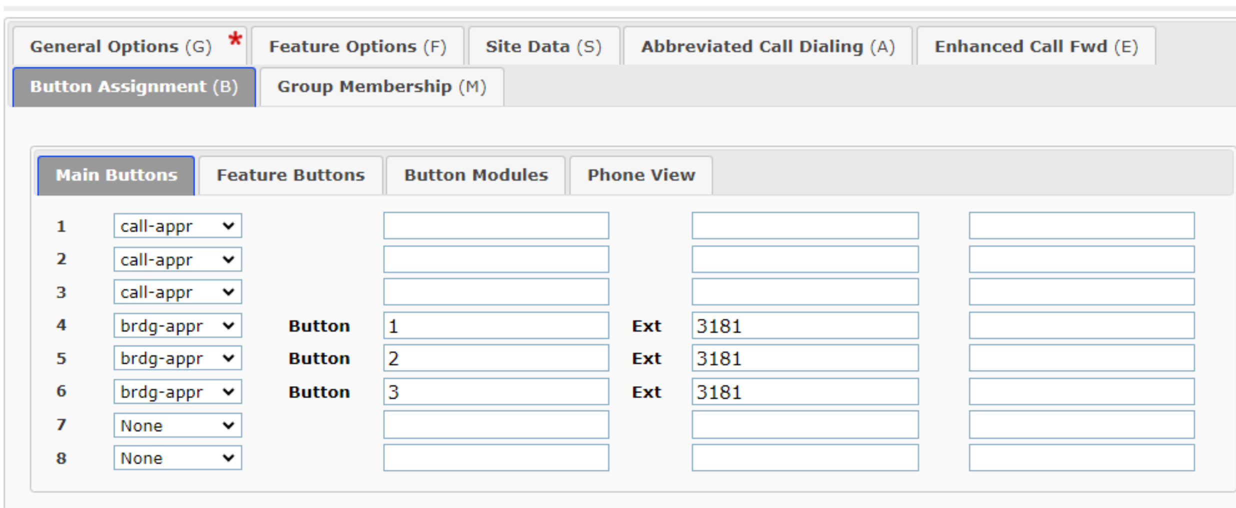
The same **COR** and **COS** that were selected for the primary user in **Section 6.3** can be used for this privacy user and again **Type of 3PCC Enabled** is set to **Avaya**.



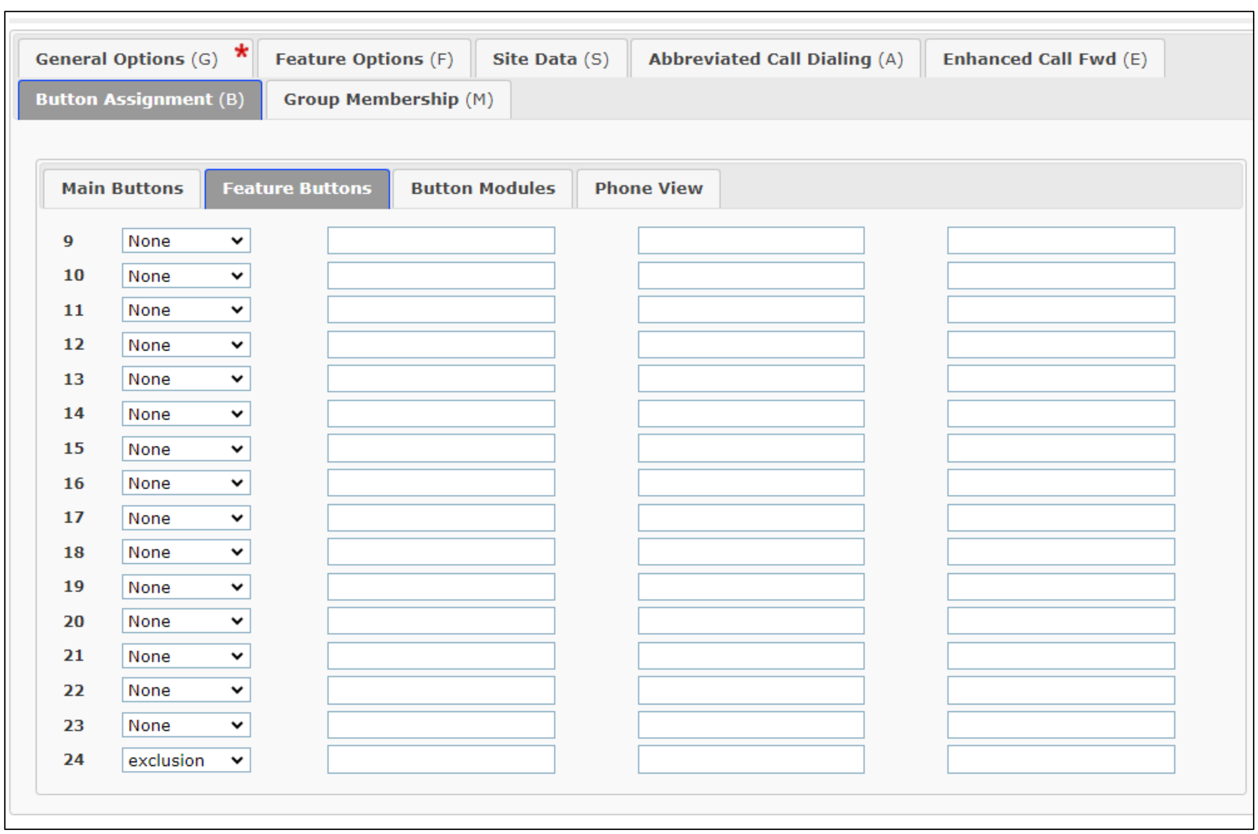
Click on the **Feature Options** tab. The screen shot below shows the Feature Options that were used during compliance testing. Ensure that **Bridged Call Alerting** is ticked as shown below, the other features are ticked as default.



Click on the **Button Assignments tab** (**Main buttons**) and configure Buttons **1**, **2** and **3** as **call-appr**. For compliance testing, buttons **4**, **5** and **6** were configured as **brdg-appr** to extension **3181** (Primary iTurret User).



Click on the **Feature Buttons** tab and ensure that Exclusion is set on one of the buttons, in this case **Button 24** was configured as **exclusion**.



7. Speakerbus iTurret Configuration

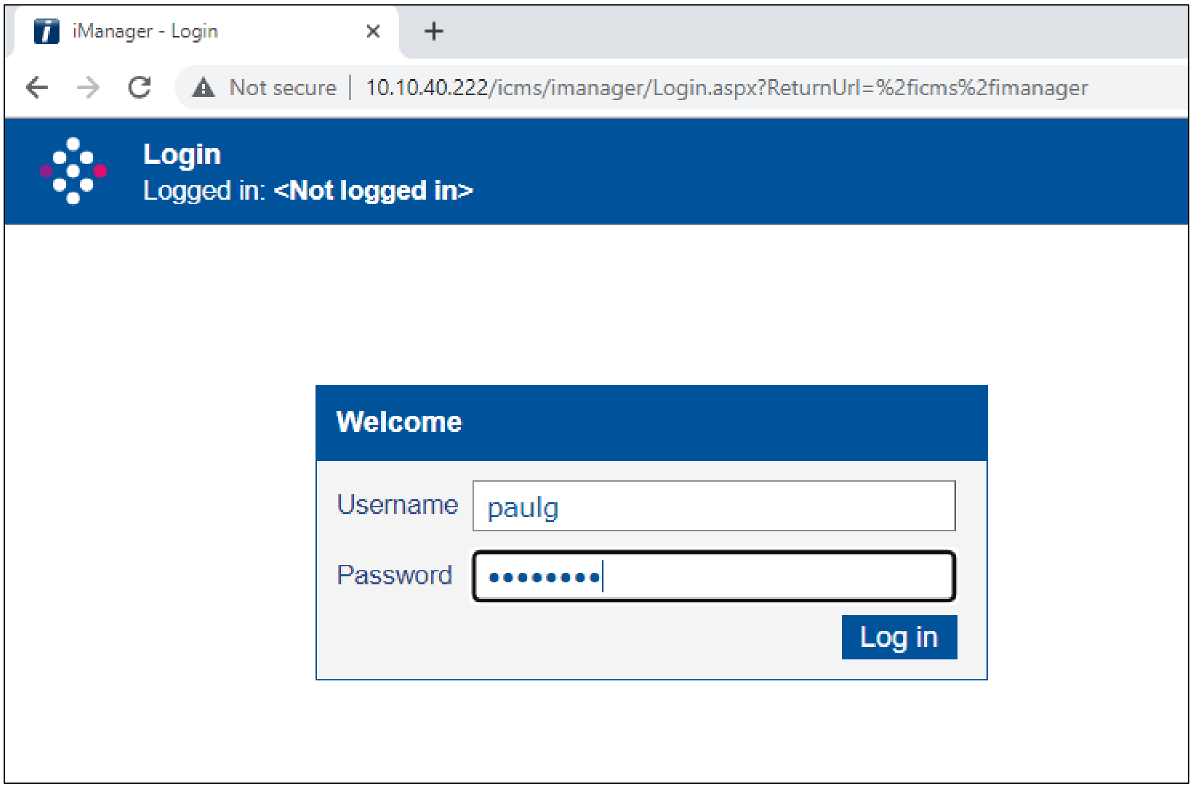
This section provides the procedure for configuring the Speakerbus iTurret via the iManager Centralised Management System (iCMS). The iCMS comprises of three components, the iManager web portal application, the iCMS communication service and the iCMS database. The iManager web portal application consists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

- Launch iManager Web Portal
- Create/Verify User Policies
- Create/Verify Device Policies
- Create Network Services
- Create Site and Call Region
- Set up device defaults
- Announce iTurrets Deskstations
- Create Users
- Create PBX (SIP Server)
- Create Dial Plan
- Create Call and Privacy Appearances
- Assign User Permissions
- Assign Ownership (of Appearances to Users)
- Assign Default Call Appearances
- Program iTurret Layout Profiles
- Synchronize Deskstations

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

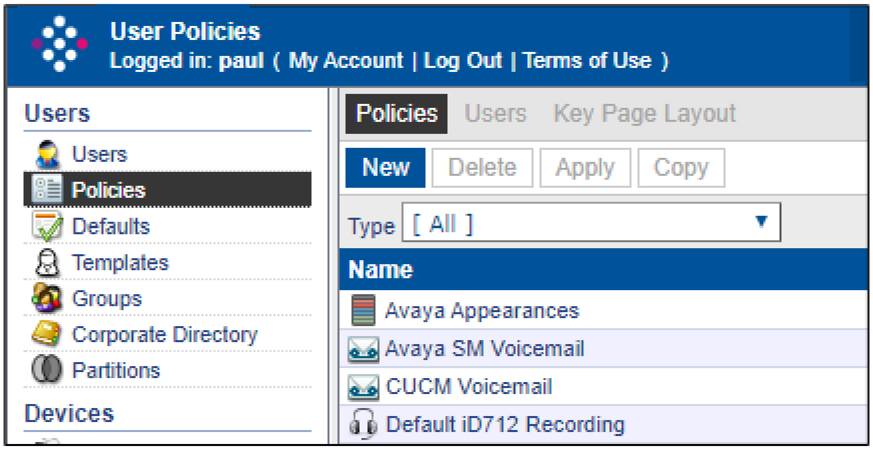
7.1. Launch iManager Web Portal

To access the iManager software interface, open a web browser and type the iManager web address, http://<ServerIP>/icms/imanager. (**Note**: If using an older version of icms / imanager, the URL is amended to http://<ServerIP>/icms/imanager). Enter the appropriate credentials and click **Log in**.

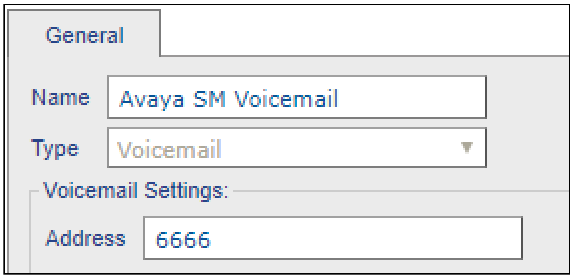


7.2. Creating/Verifying User policies

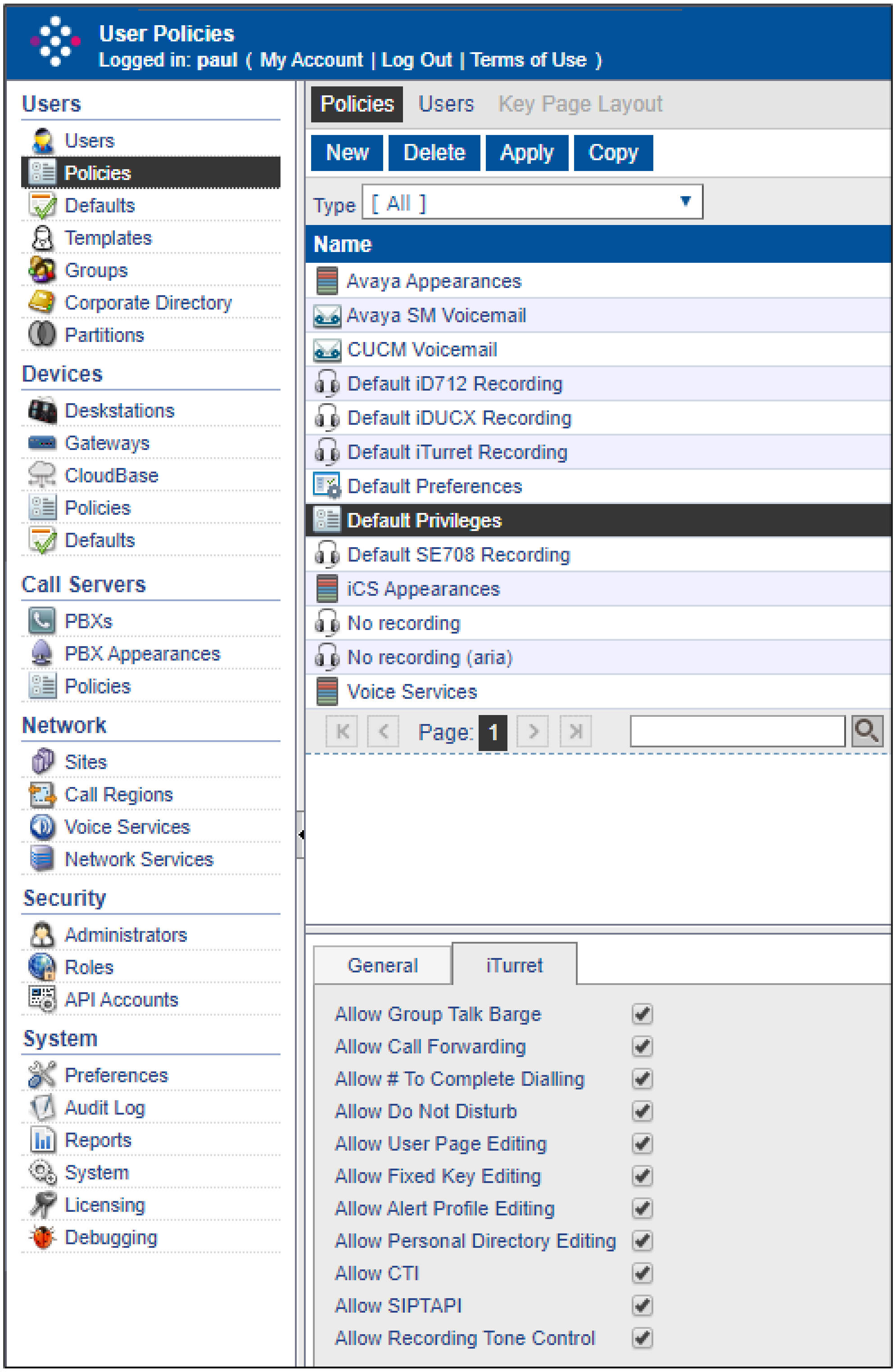
Select **Users → Policies** in the left pane and click on **New**.



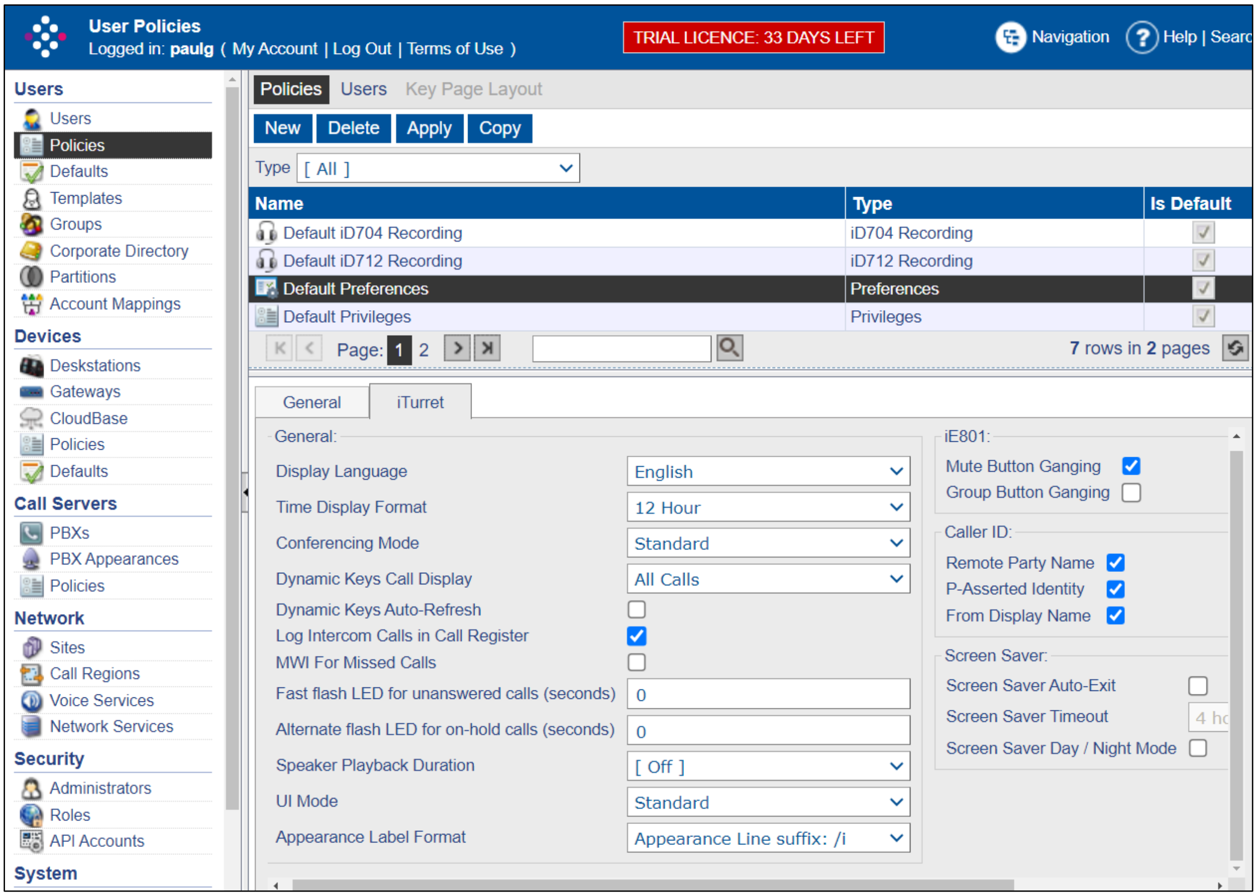
Enter an identifying **Name**, in the **Type** dropdown box select **Voicemail,** and enter a valid address for the voicemail server**,** in this case a pre-configured hunt group number for voicemail access is used. Click **OK** once completed (not shown).



Select **Users → Policies** in the left pane. Select and view the **Default Privileges** policy, (no changes to this should be required, however, it is referred to later in these Application Notes).

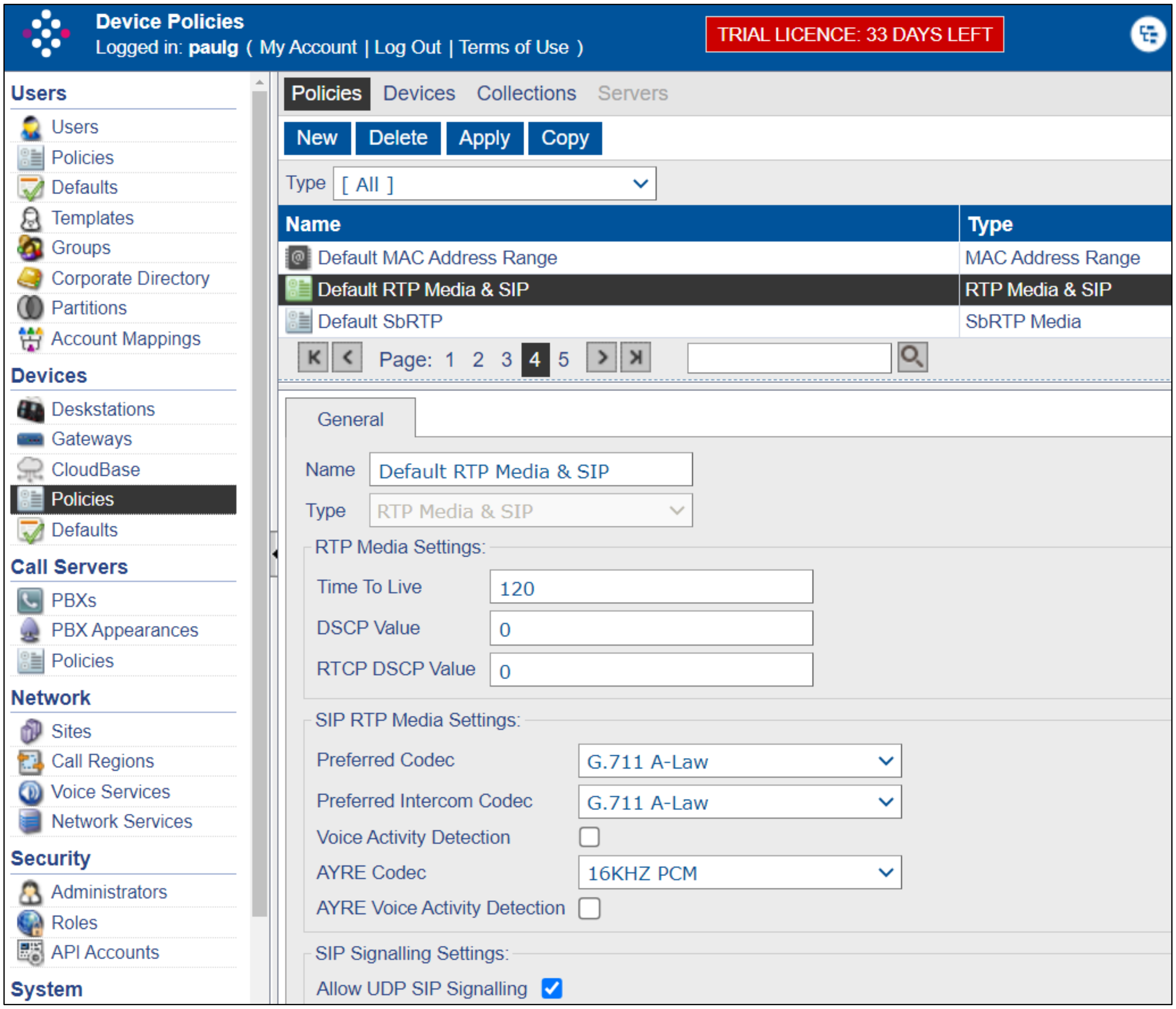


Select **Users → Policies** in the left pane. Select the **Default Preferences** policy, click the **iTurret** tab and review the default settings (no changes should be needed to these; however, they are referred to later in these Application Notes).

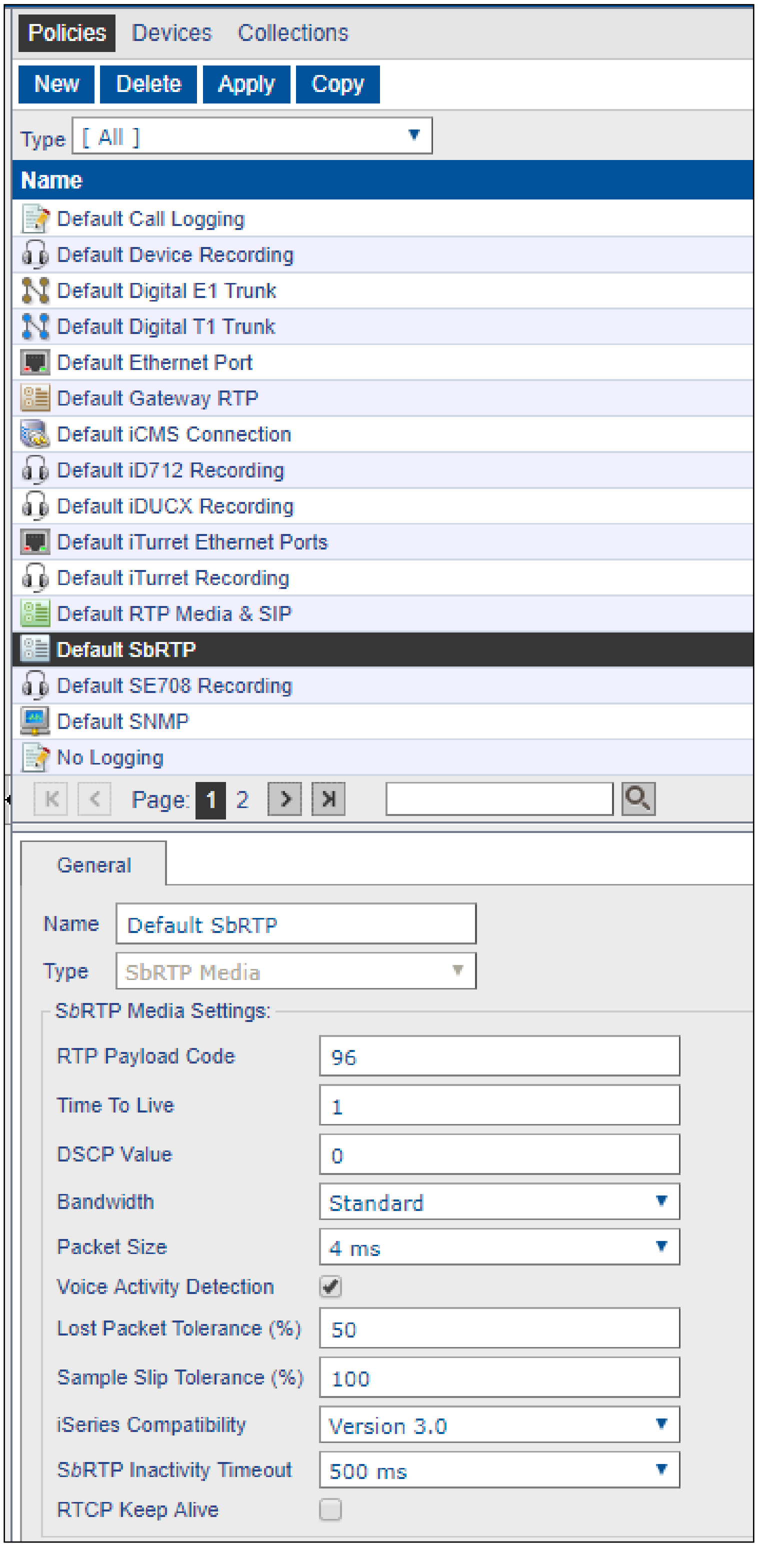


7.3. Creating/Verifying Device Policies

Select **Devices → Policies** in the left pane. Select the **Default RTP Media & SIP** policy, if leaving the SIP signaling protocol setting at default UDP, then no changes should be needed to these; however, they are referred to later in these Application Notes. If using TCP, then untick the "Allow UDP SIP Signaling" flag and press OK (not shown).



Staying on Polices, select and view the **Default SbRTP** policy (no changes should be needed to these; however, they are referred to later in these Application Notes).



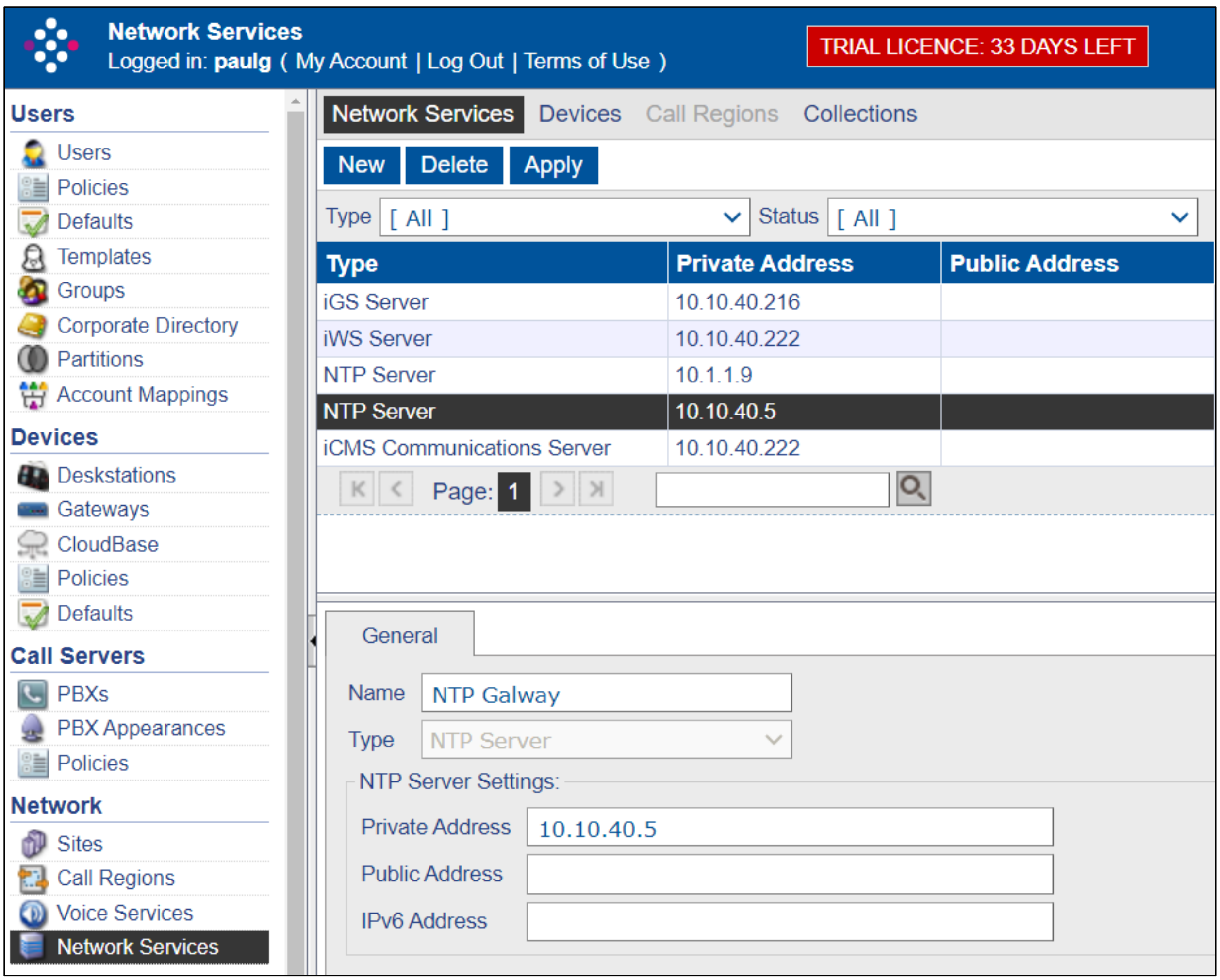
7.4. Creating Network Services

A network service is an addressable entity that a device uses to contact the relevant service when and where required. Defining network services here merely defines the network service configuration, it does not cause it to be used by any devices. Network services can be assigned to devices via the device configuration or via a policy, depending on the network service type. Confirm that CMS comms and seen in the list view with the correct details.

Note: Refer to the *Speakerbus iManager Administrator`s Guide*.

To create an NTP Server, select **Network → Network Services** in the left pane, click **New** and select NTP Server from the dropdown menu (not shown).
Complete the following fields.

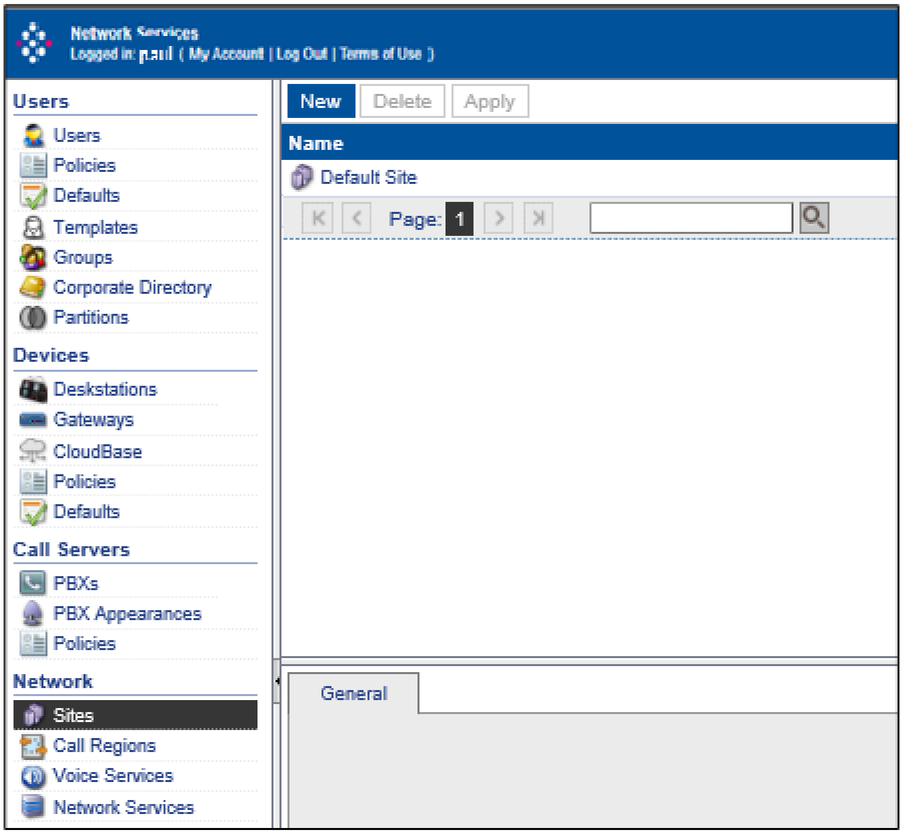
- **Name** Enter a descriptive name for the site.
- **Private Address** Enter the IP address of the NTP server.



7.5. Creating Site and Call Region

A site represents the location where the Speakerbus iSeries equipment is installed. To create a Site, select **Network** → **Sites** in the left pane, click **New**.

Note 1: A Default Site is available and can be used if required.



Complete the following fields:

- **Name** Enter a descriptive name for the site.
- **Remote Site** Leave unticked for most cases.

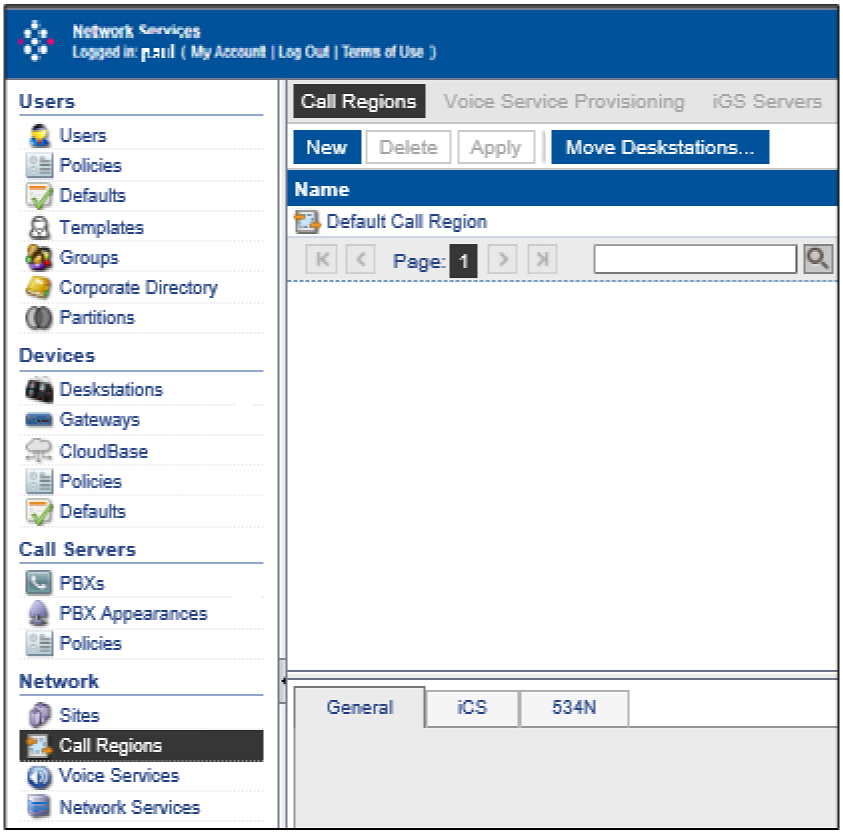
Click **OK** once completed.

Note 2: Only tick remote site when using an iTurret device at home connecting to a corporate network via a VPN link.

A call region represents part of an organisation's network over which all devices associated with the call region can communicate call audio and call signalling.

To create a Call Region, select **Network → Call Region** in the left pane, click **New**.

Note 3: A Default Call Region is available and can be used if required.



Complete the following fields:

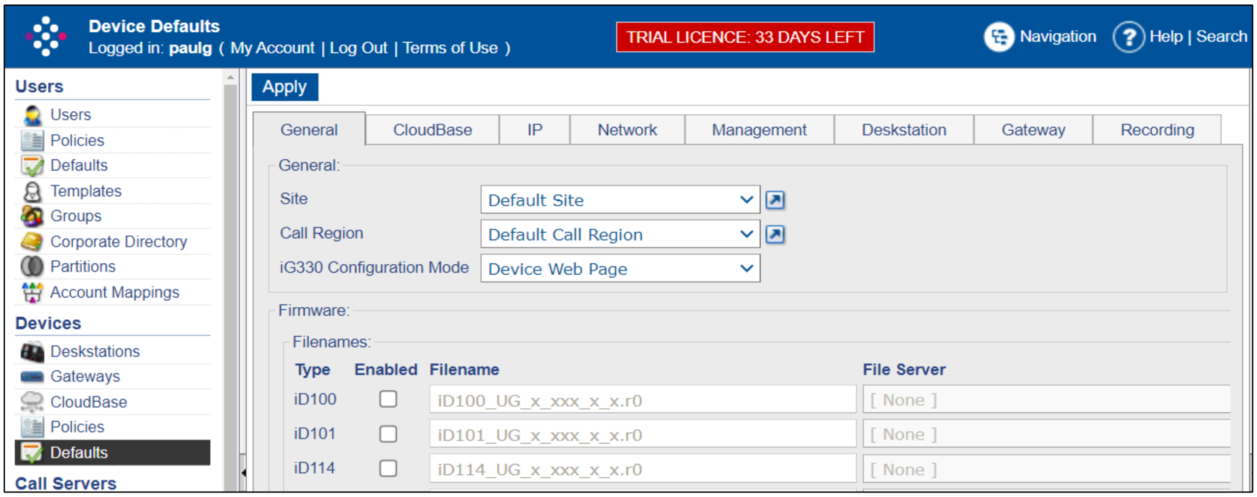
- **Name** Enter a descriptive name for the call region.
- **Partition Checking** Leave unticked for most cases.
- **Priority for P2P** Leave unticked for most cases.
- **IGMP Auto-leave** Leave unticked for most cases.
- **DMVS Intercom Calls** Leave unticked for most cases.

Note: Refer to the *Speakerbus iManager Administrator`s Guide.*

Click **OK** once completed.

7.6. Check Device Defaults

The default configuration is used when a new device is created either from an auto-announce or from iManager. Select **Device → Defaults** in the left pane.



Confirm the following fields are set.

General Tab

- **Site** Set with what created in **Section 7.5**.
- **Call Region** Set with what created in **Section 7.5**.

IP Tab

- **NTP Server** Set with what created in **Section 7.4**.

Network Tab

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

Management Tab

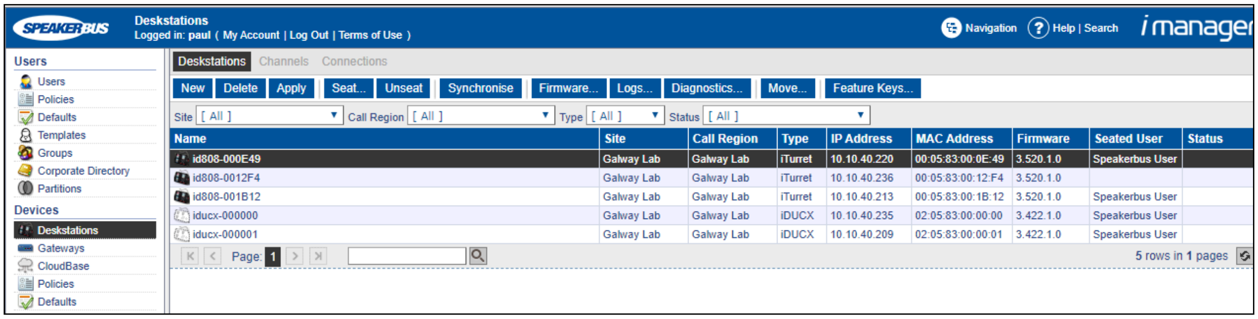
- **iCMS Communication Policy** is set to the default.
- **iCMS Communication Server** is set to Auto-Locate iCMS if using DHCP / DNS.
- **Enable Live Updates** Ticked.

Note: Refer to the *Speakerbus iManager Administrator`s Guide*.

Click **APPLY** once completed.

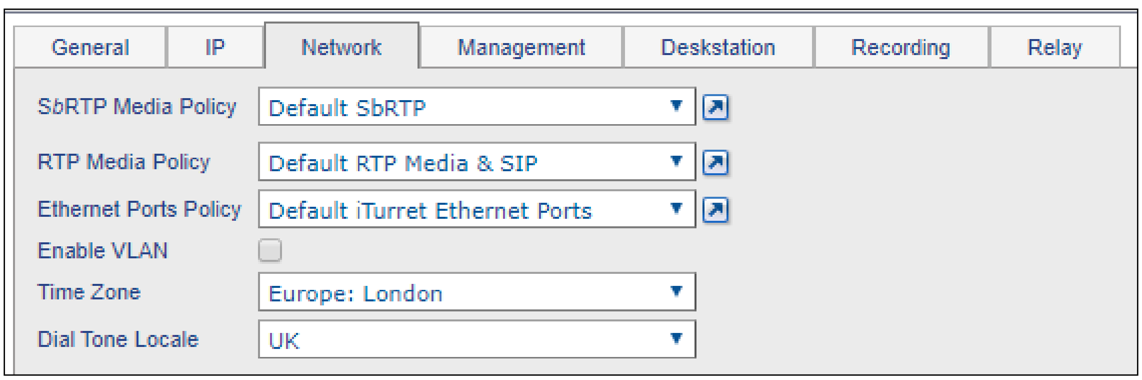
7.7. Announce iTurret Deskstation

The iTurret deskstations will automatically announce to the iCMS server if appropriate **DHCP** and **DNS** records were created prior to the iTurret deskstations being connected to the IP network and powered up. To view the newly registered deskstations, select **Devices →Deskstations** in the left pane, confirm they are seen as below.



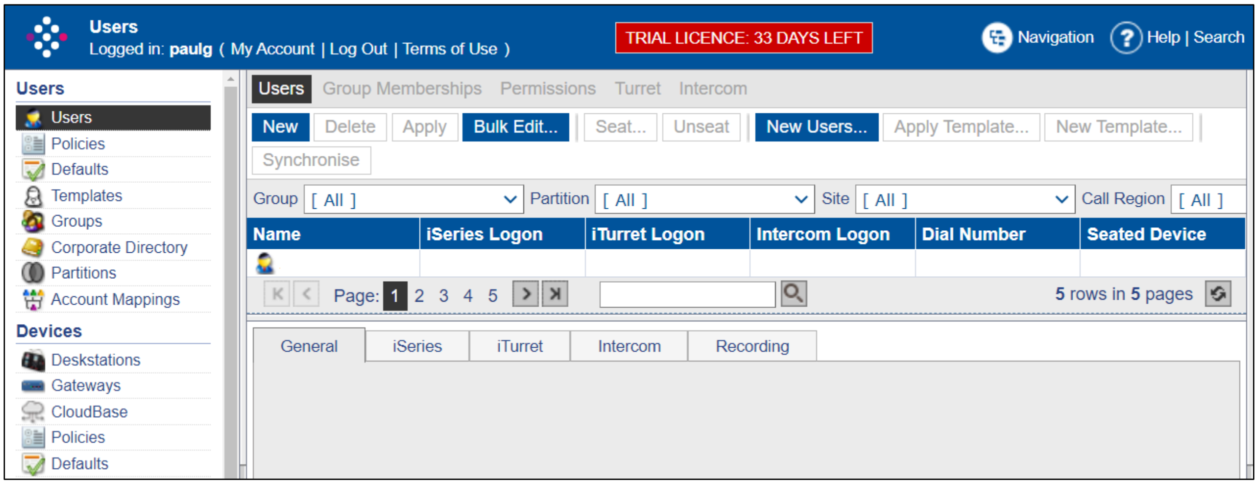
In the **Network** tab, verify the following are configured as mentioned above:

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



7.8. Create Users

To create a User, select **Users** →**Users**, click **New**.



Confirm the following fields are set:

General Tab

- **Name** Enter a descriptive name for the call region.
- **Privileges Policy** This should be set to the default in **Section 7.2**.
- **Preferences Policy** This should be set to the default in **Section 7.2**.

iTurret Tab

- **Logon Name** Enter a relevant logon name (8 – 16 characters in length).
- **Logon Password** Enter a relevant logon password.
- **Verify Password** Enter a relevant logon password (should match above).
- **Voicemail Policy** This should be set to the policy in **Section 7.2**.

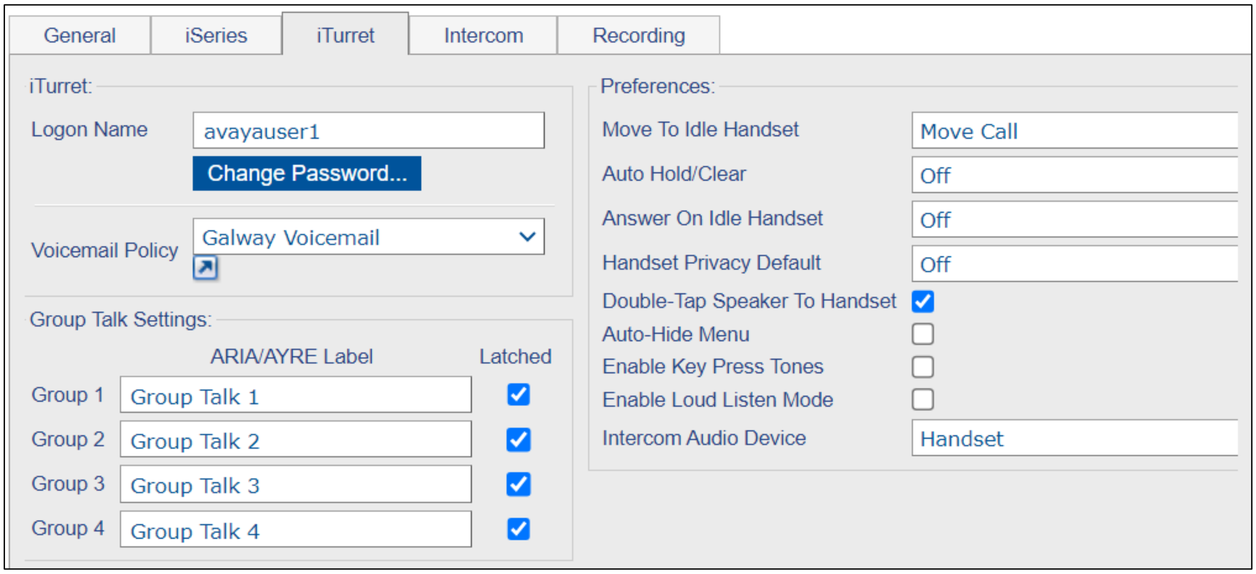
All other areas can be left at defaults (refer to the *Speakerbus iManager Administrator`s Guide*).

Click **OK** once completed.

Within the **iTurret** tab, provide the **logon** credentials by clicking on the **Change Password** button and enter a **Login Name** and **Password** (not shown) and enter the following:

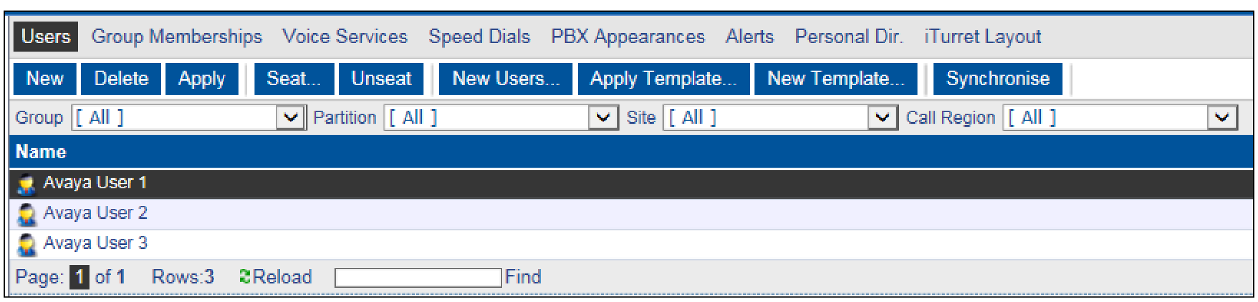
- **Voicemail Policy** Select the voicemail policy as configured in **Section 7.2**.
- **Move to Idle Handset Mode** Select **Move Call** from the drop-down list.
- **Enable Latching** Tick **Group Button 1**, **2**, **3** and **4**.

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



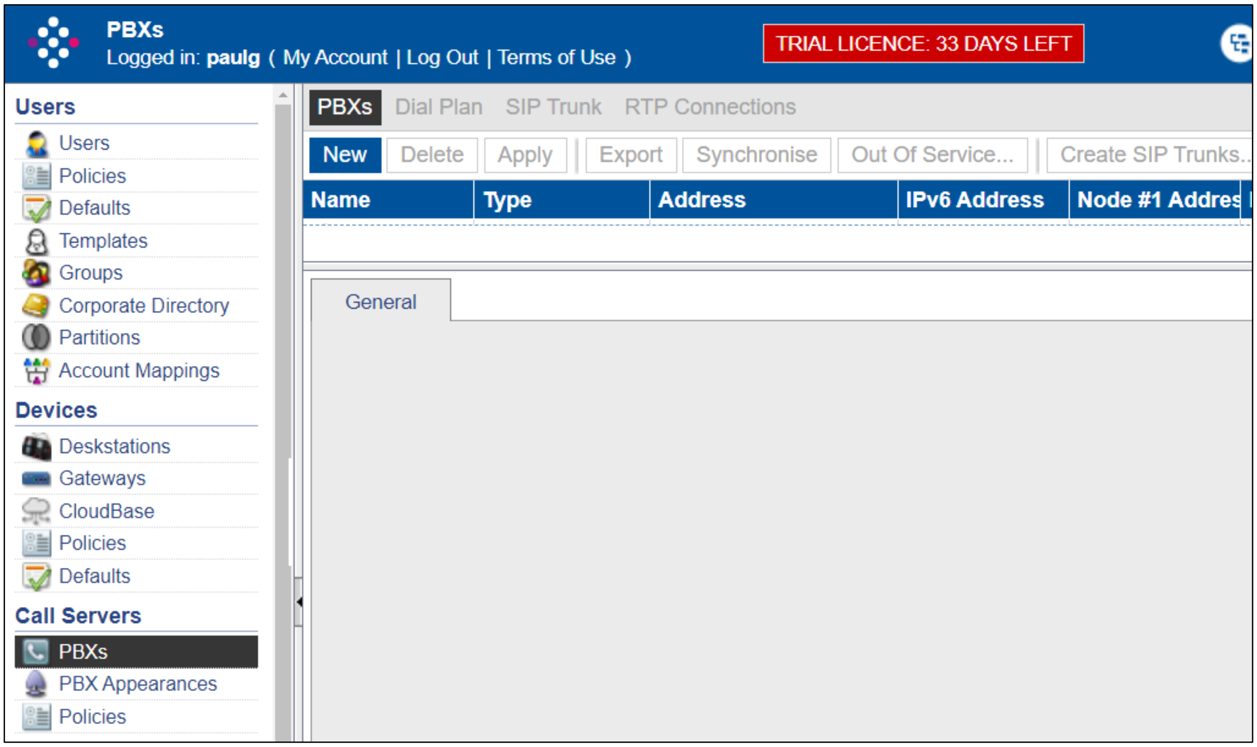
Repeat the previous steps to add more users.

Once the users are added, set up the PBX appearances for these users and then add them as Default PBX Appearances, see subsequent sections for further details.



7.9. Create PBX (SIP Server)

To create a PBX, select **Call Servers →PBXs,** click **New**.

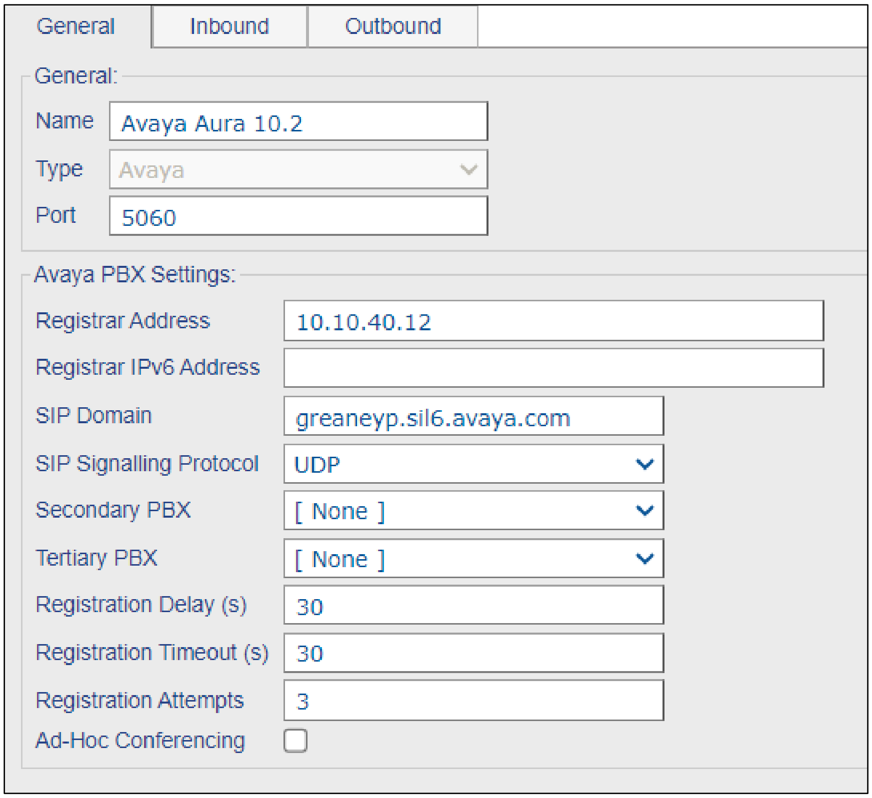


Complete the following fields (shown on next page):

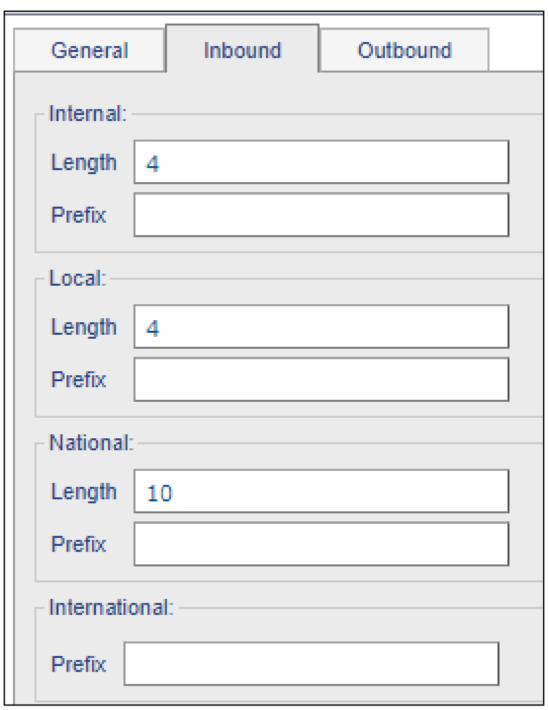
- **Name** Enter a descriptive name for the SIP/PBX server.
- **Type** Select **Avaya** from the dropdown list.
- **Port** Enter **5060**.
- **Registrar Address** Enter the IP address of the Primary Session Manager.
- **SIP Domain** Enter the appropriate SIP Domain.
- **SIP Signaling Protocol** This can be set to **UDP** or **TCP**.

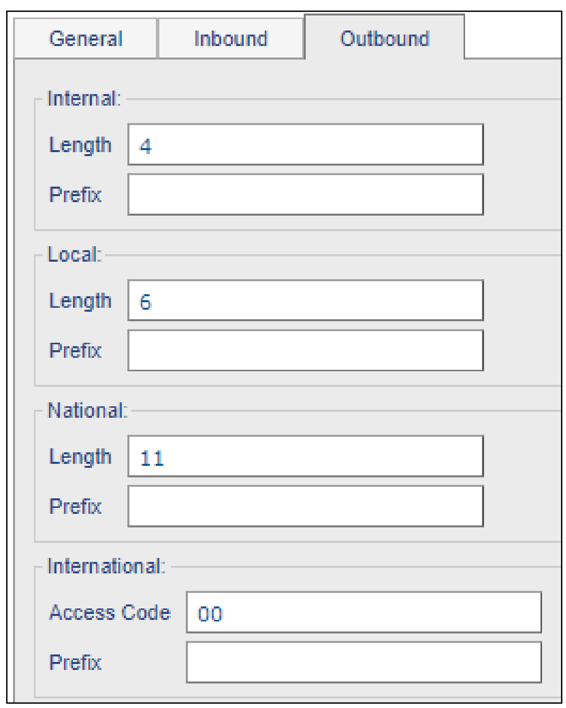
Note 1: A server locater record (SRV) for the registrar address and SIP domain may be created on DNS if the registrar address is set to **greaneyp.sil6.avaya.com**, in the example below it will not be required. Refer to the *Speakerbus iManager Administrator`s Guide* for the correct configuration of DNS.

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



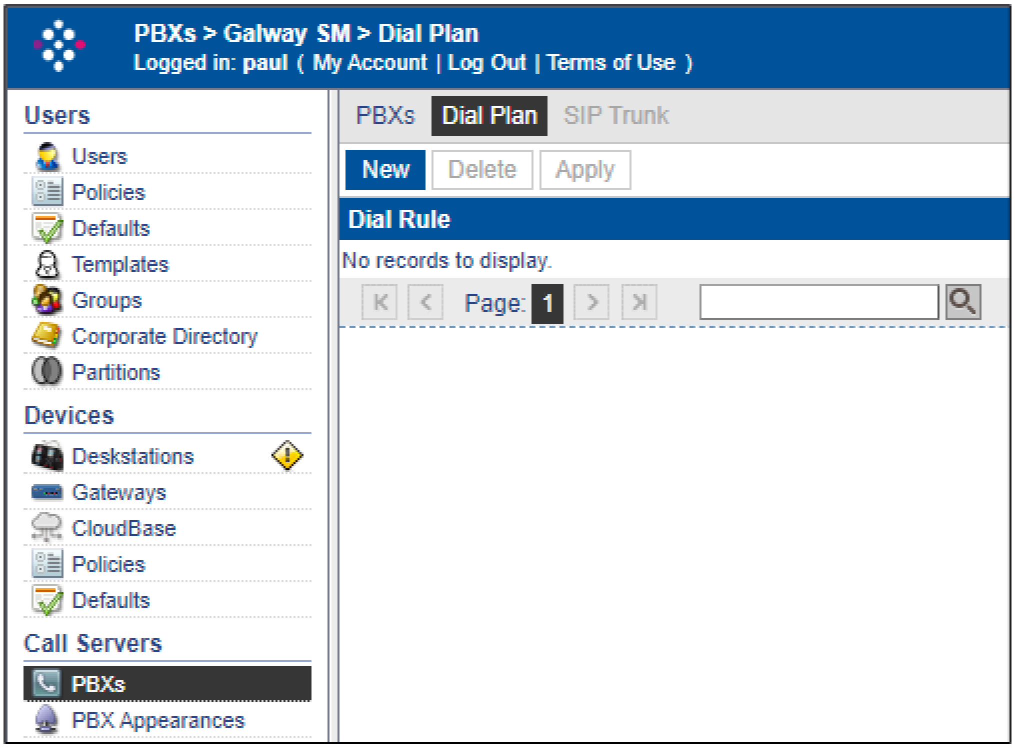
The **Outbound** and **Inbound** tabs are left with their default values, Click **OK** (not shown).



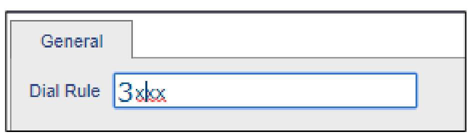


7.10. Create Dial Plan

To create a PBX specific dial plan, select **Call Servers →PBXs**, select the **Dial Plan** tab**,** click **New**.



Under the **General** tab fill in the **Dial Rule**. Press **OK** when completed.



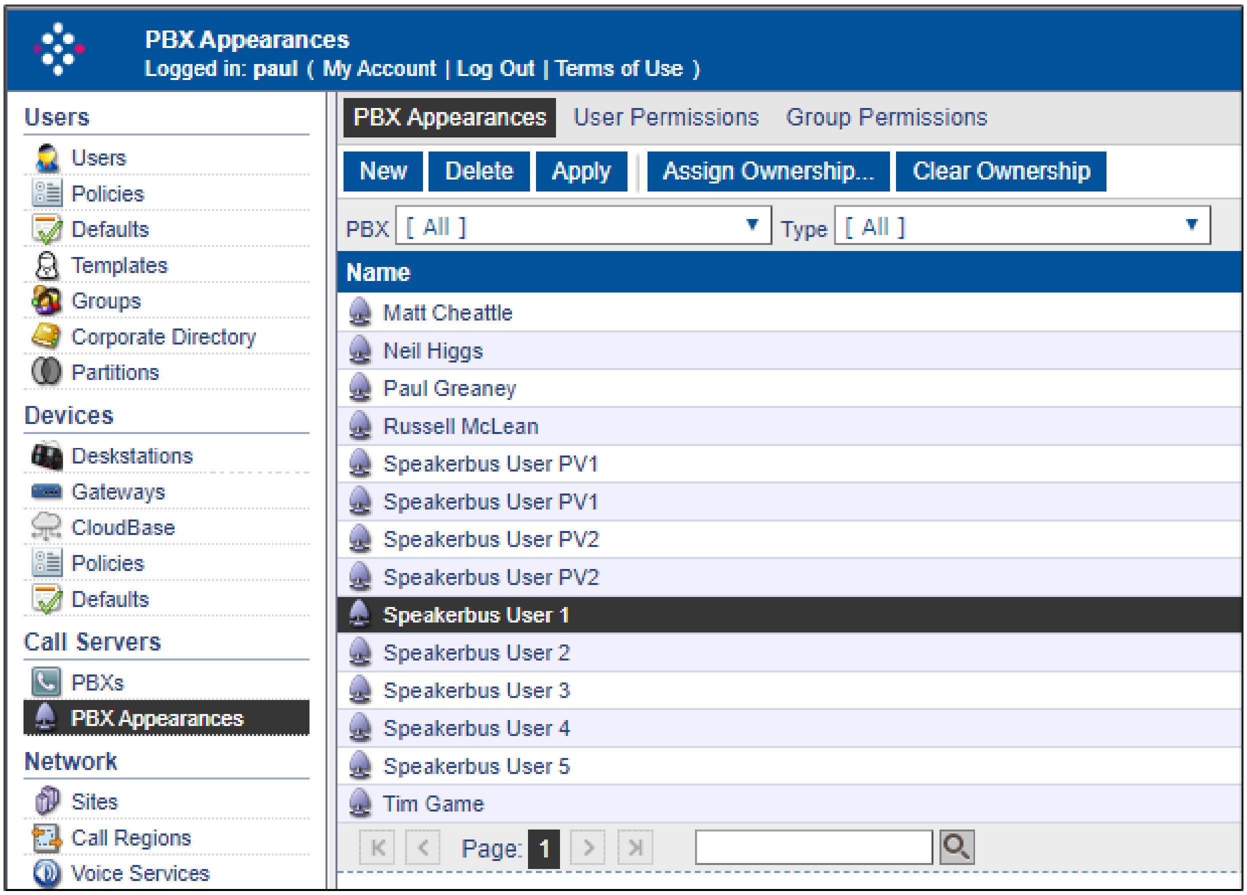
Repeat this for all valid extension formats.

7.11. Create Call and Privacy Appearances

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

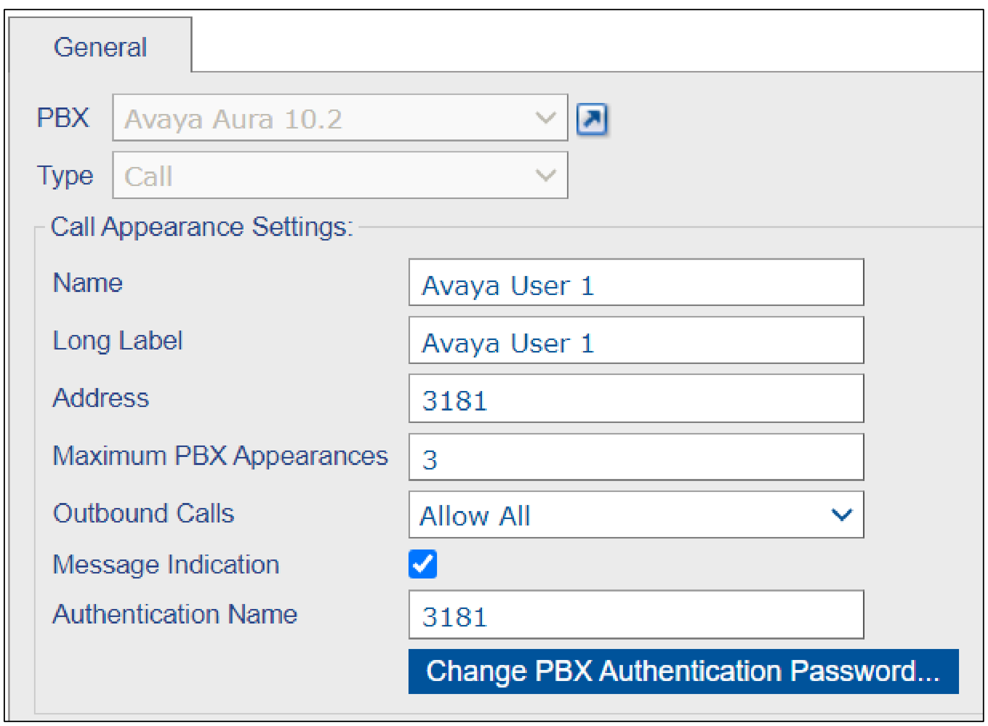
7.11.1. Create Call Appearances

To create the main appearance, click **Call Servers →PBX Appearances** in the left pane, click on **New**.



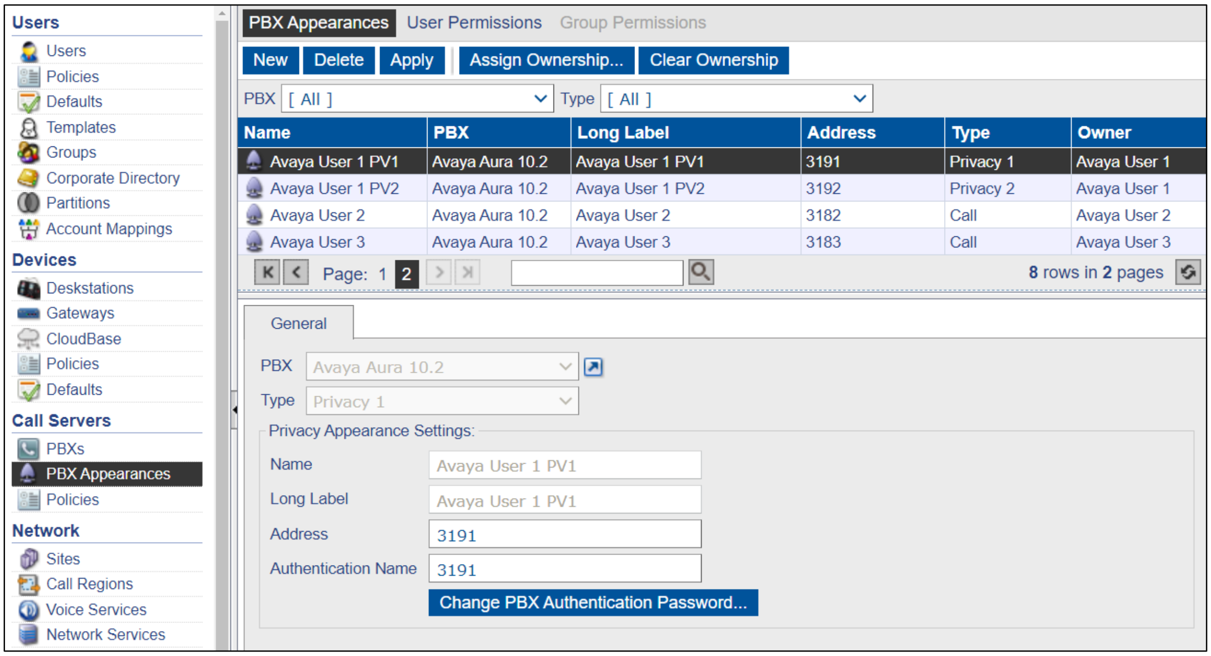
Select the PBX created in **Section 7.5** (in this case **Avaya Aura 10.2**), then select the **Type** of appearance to be created (which is **Call** in this case) and configure the following under the **General** tab:

- Provide a descriptive name for the appearance in the **Name** field, such as the extension or user's name.
- Set the **Long Label** field to the label that will be displayed for the call appearance button on the iTurret deskstation. The **Address** field should also be set to the appearance extension.
- Set the **Maximum Appearance** field to the number of call appearances configured on the station in System Manager (the number of call appearance buttons dictates the number of calls on the system the user can have directed to them). When all of the call appearances are not idle the user is considered busy and no further calls can be routed to them. Up to a maximum of 10 call appearances may be configured on Communication Manager for each iTurret deskstation.
- Check the **Message Indication** checkbox for voice mail purposes and the **Allow Outbound Calls**.
- The **Authentication Name** and **Authentication Password** fields should be set to the extension and password configured on System Manager in **Section 6.3**. These are the credentials that the iTurret deskstation will use to authenticate and register with Session Manager. Use the default values for the other fields. Click **OK** (not shown).

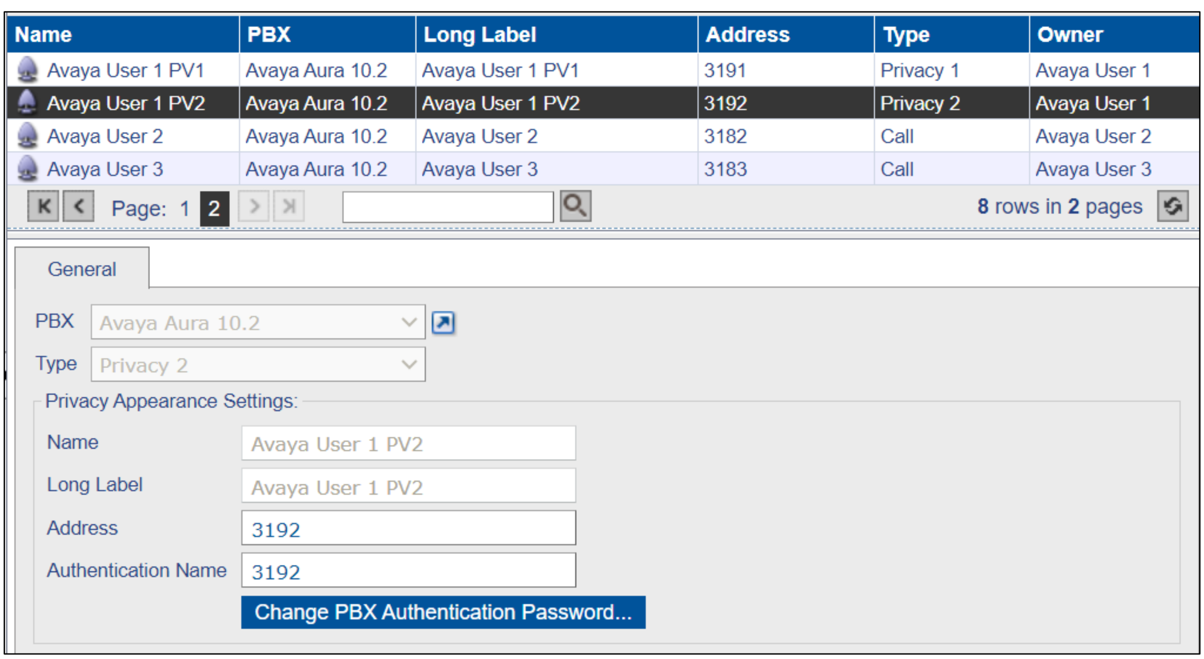


7.11.2. Create Privacy Appearances

Repeat the procedure in **Section 7.11.1** for the two corresponding privacy appearances. Click the **New** button to add another appearance. In the **General** tab select the **PBX** created in **Section 7.5**, set the **Type** field to **Privacy 1** and complete the **Address, Authentication Name** and **Authentication Password** fields. The last two fields should be identical to the setup in System Manager for registration to occur. Press **OK** (not shown) to commit the created appearance.

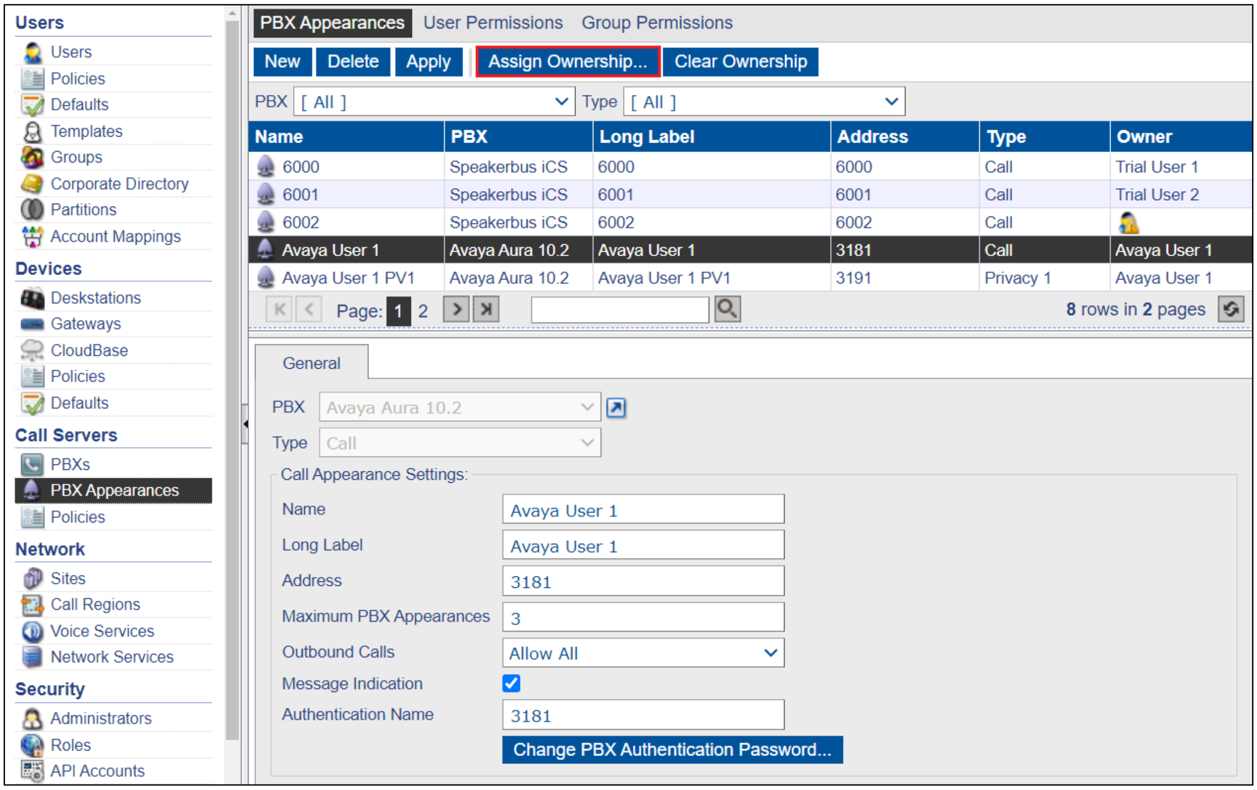


Similar details for the second privacy user.

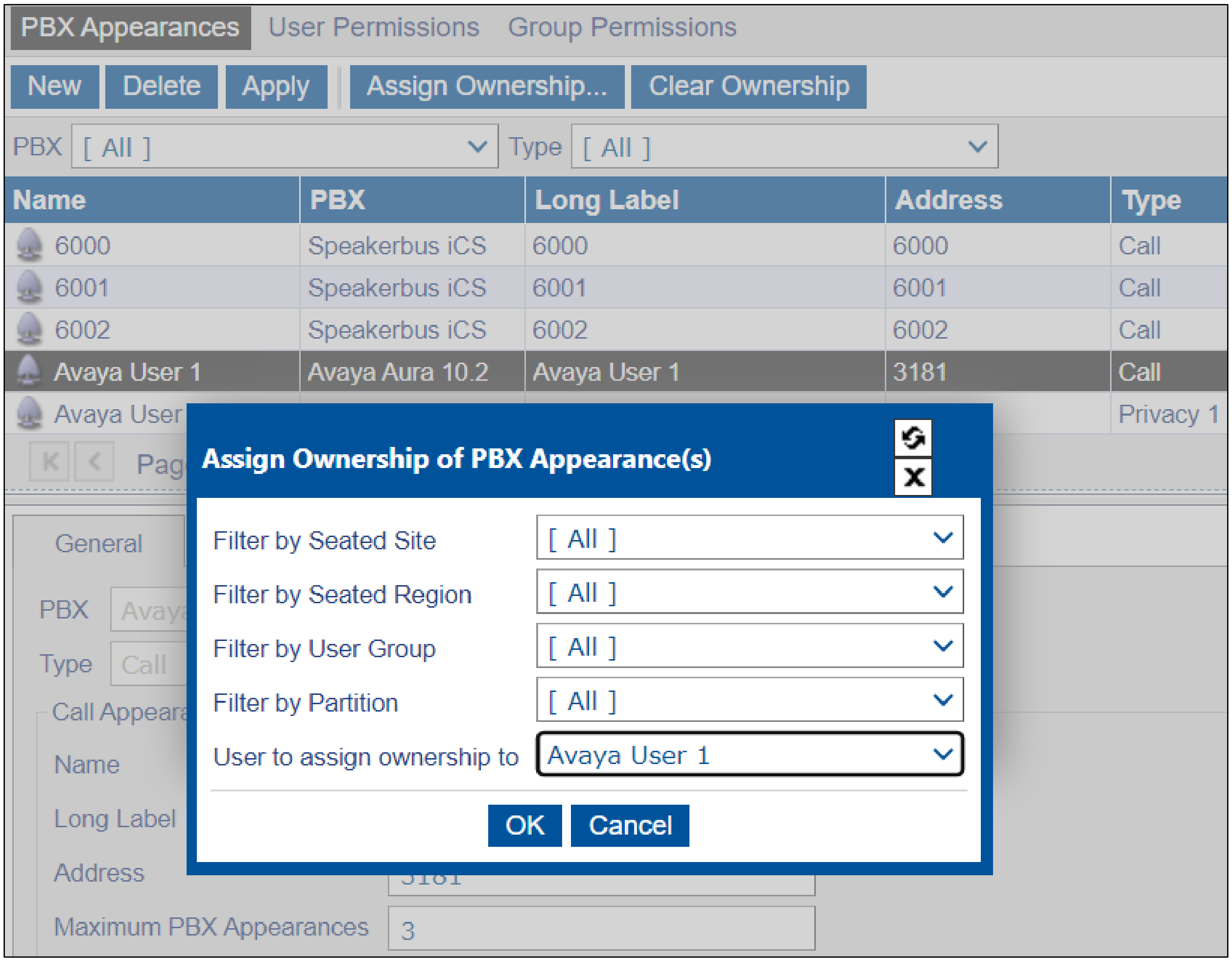


7.12. Assign Ownership

Appearance ownership must be assigned to a user as it enables the iTurret to distinguish between the owner of the call appearance as opposed to someone who is bridged on to that appearance. Select **Call Servers →PBX Appearances** in the left pane and click on the **Assign Ownership** button.

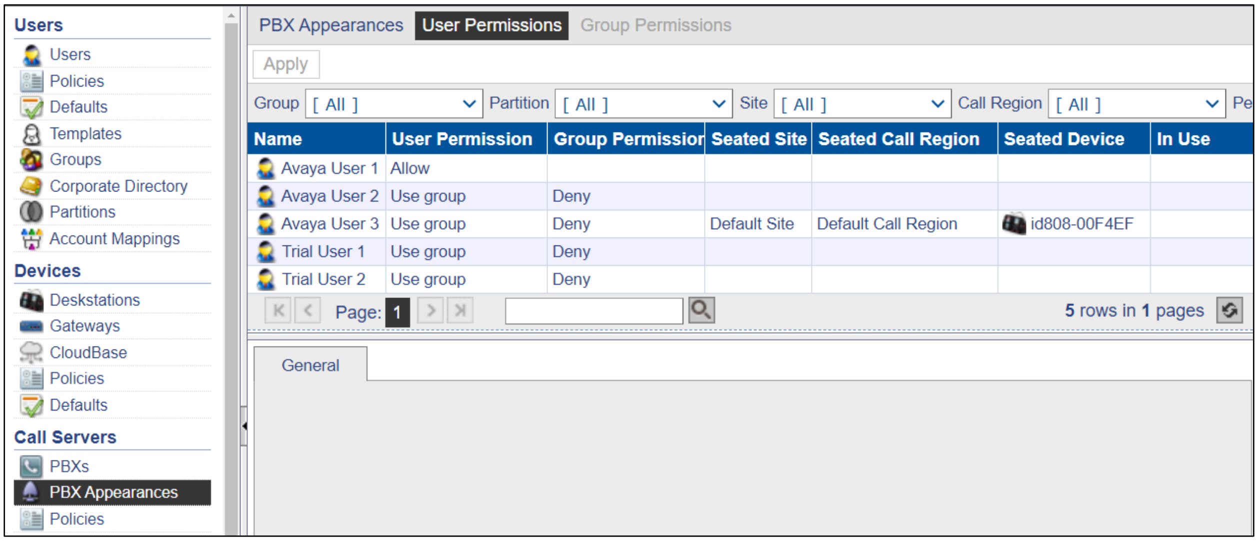


Filter accordingly and select the user from the **User to assign ownership to** drop down list. Click **OK**.

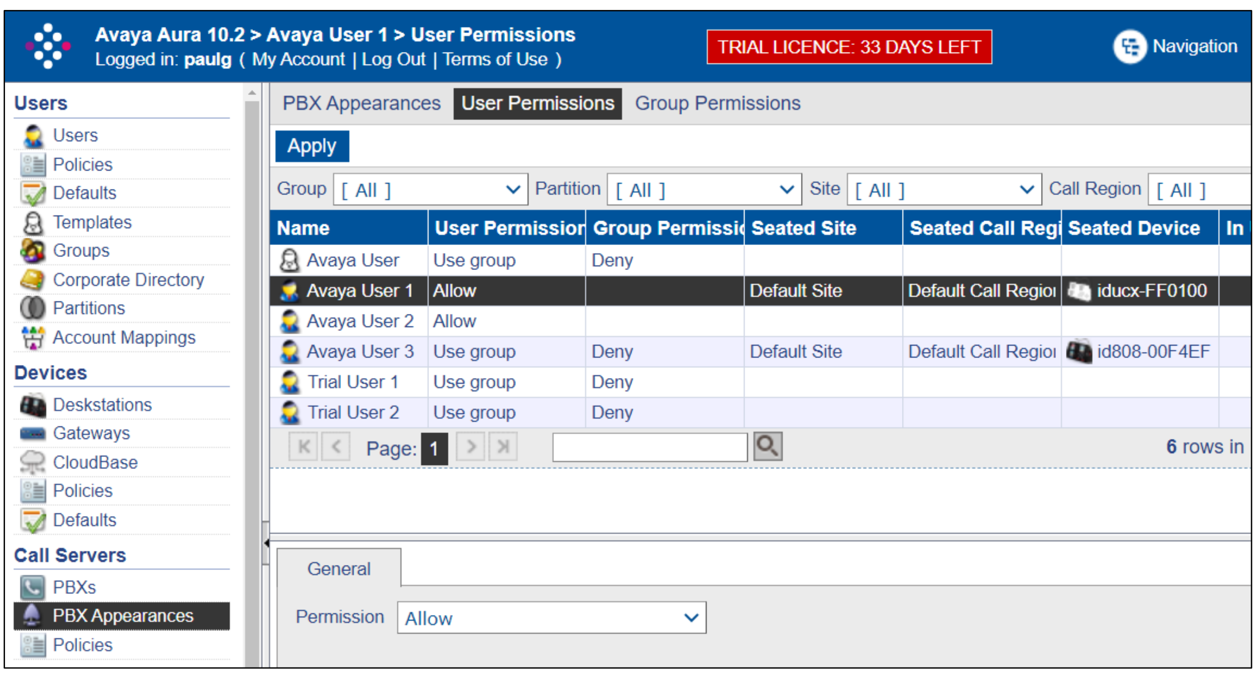


7.13. Assign User Permissions

Appearance permissions must be assigned to the created users. Select **Call Servers →PBX Appearances** in the left pane, select the **Call Appearance** from the list, and select the **User Permissions** tab at the top of the page.

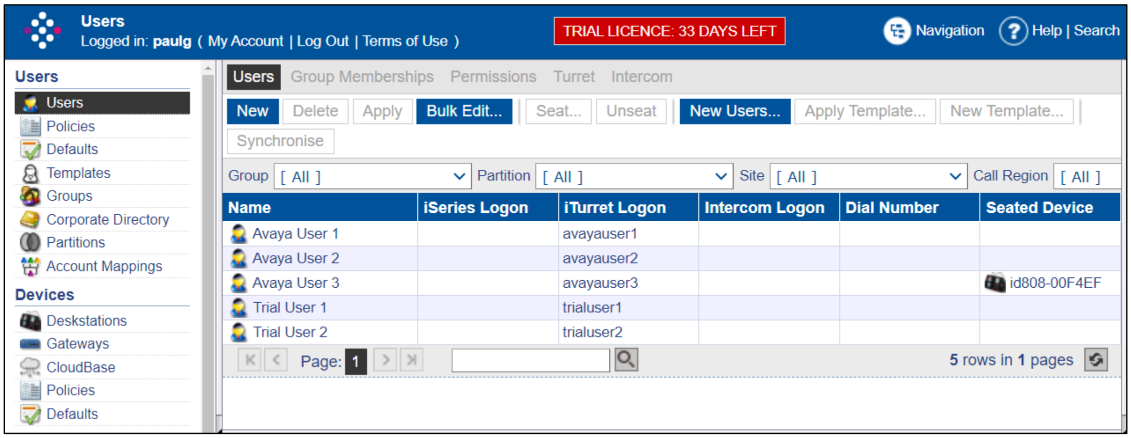


Select the user to give permissions to and select **Allow** from the **Permissions** drop down list and click **Apply**.



7.14. Set Default Appearance

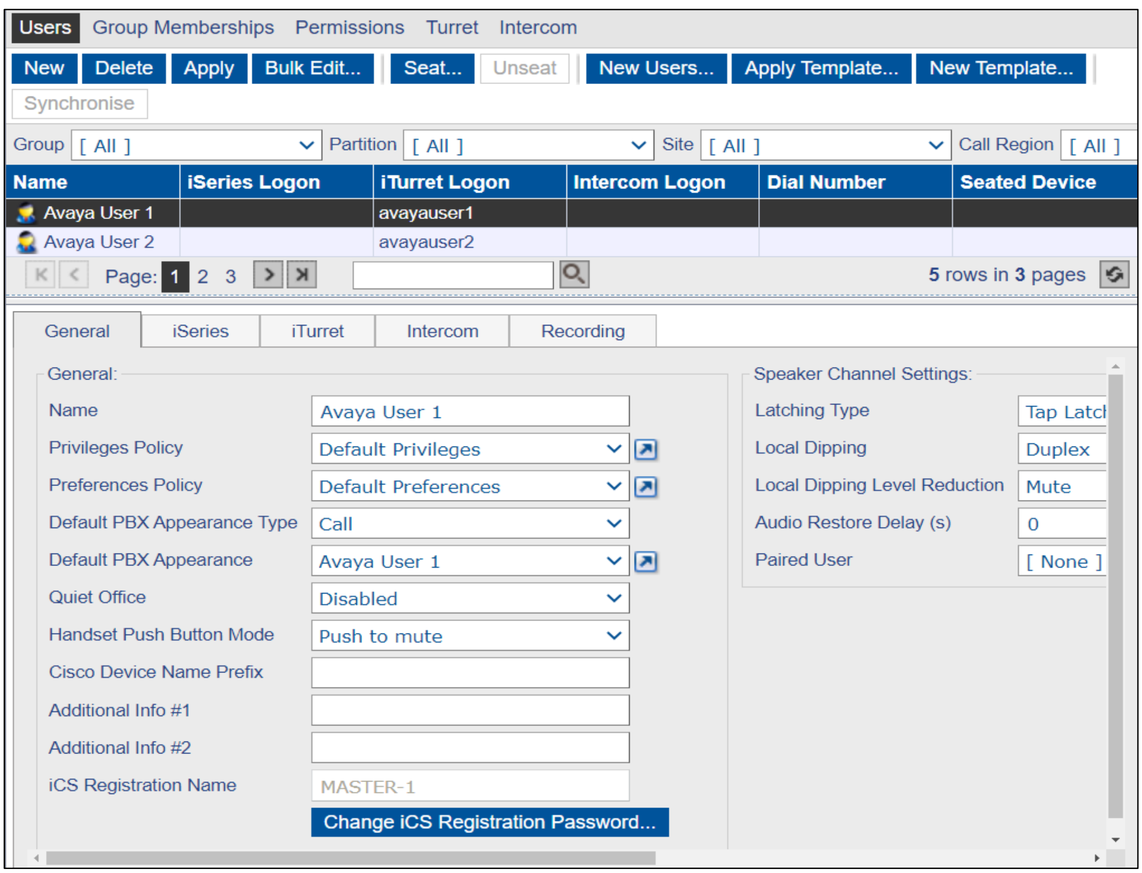
Select **Users** →**Users** in the left pane.



Within the **General** tab fill in the following:

- **Default PBX Appearance Type** Select Call from the drop-down list.
- **Default PBX Appearance** Select the appropriate user from the drop-down list.

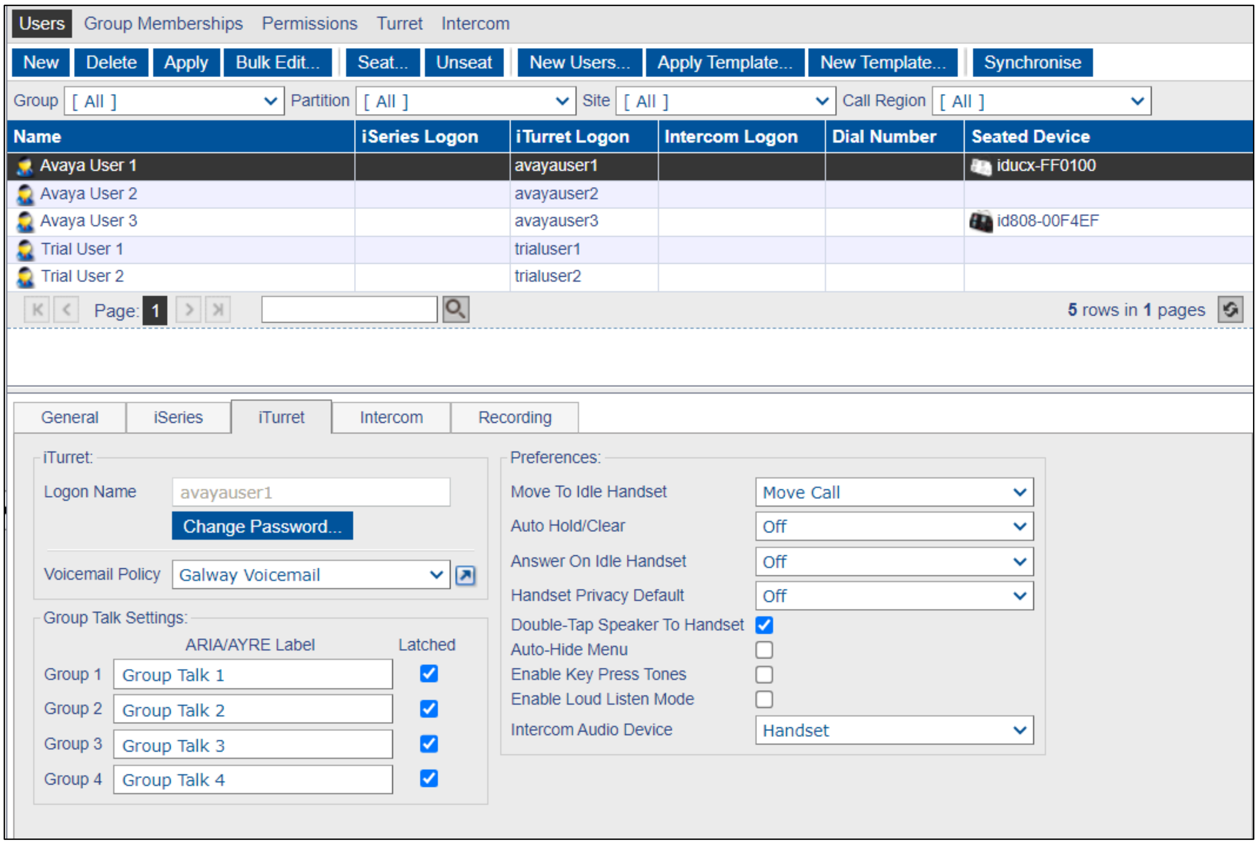
Click **Apply** once completed.



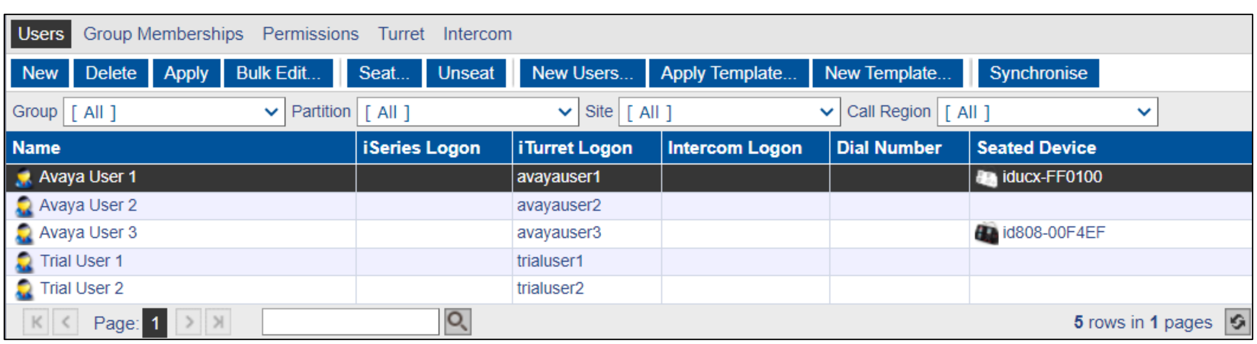
Within the **iTurret** tab, provide the **logon** credentials by clicking on the **Change Password** button and enter a **Login Name** and **Password** (not shown) and enter the following:

- **Voicemail Policy** Select the voicemail policy as configured in **Section 7.2**.
- **Move to Idle Handset Mode** Select **Move Call** from the drop-down list.
- **Enable Latching** Tick **Group Button 1**, **2**, **3** and **4**.

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



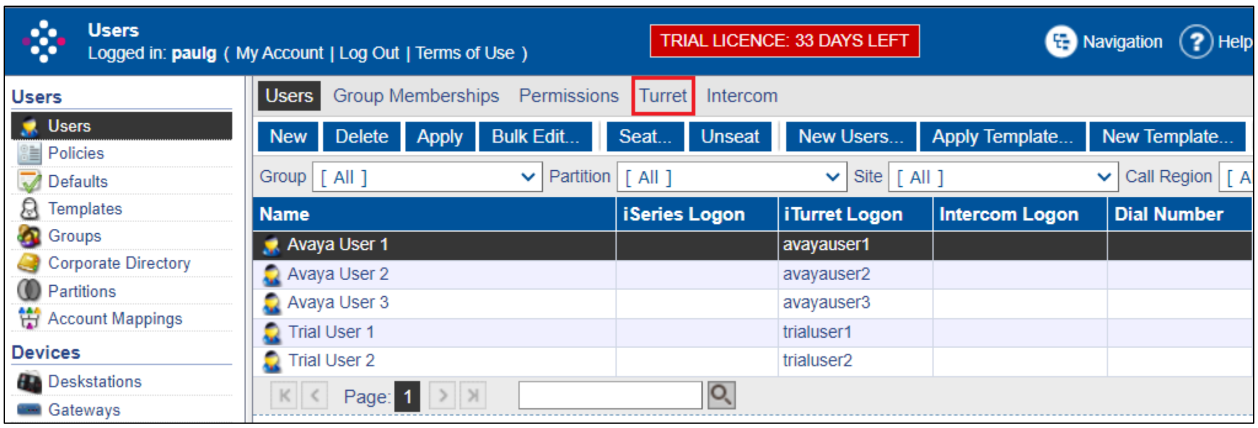
Repeat the previous steps to add more users. Once you have added the users, you can set up the PBX appearances for these users and then add them as Defaults PBX Appearance, see subsequent sections for further details.



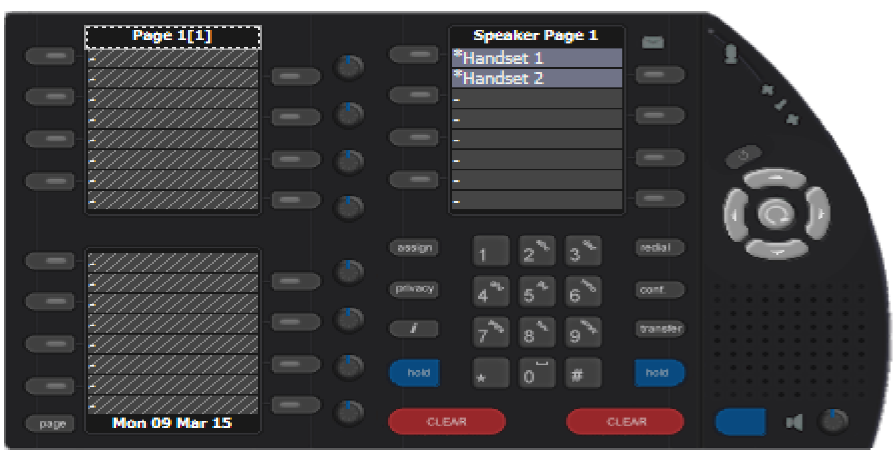
7.15. Program iTurret Layout Profiles

The programming of the iTurret Deskstations can be carried out by Speakerbus or Avaya engineer. For information on the types of keys available and administration of the iTurret layout, refer to the *Speakerbus iManager Administrator`s Guide.*

To add the above appearances to the iTurret layout, go to the user and select the **Turret**, as per the screenshot below.



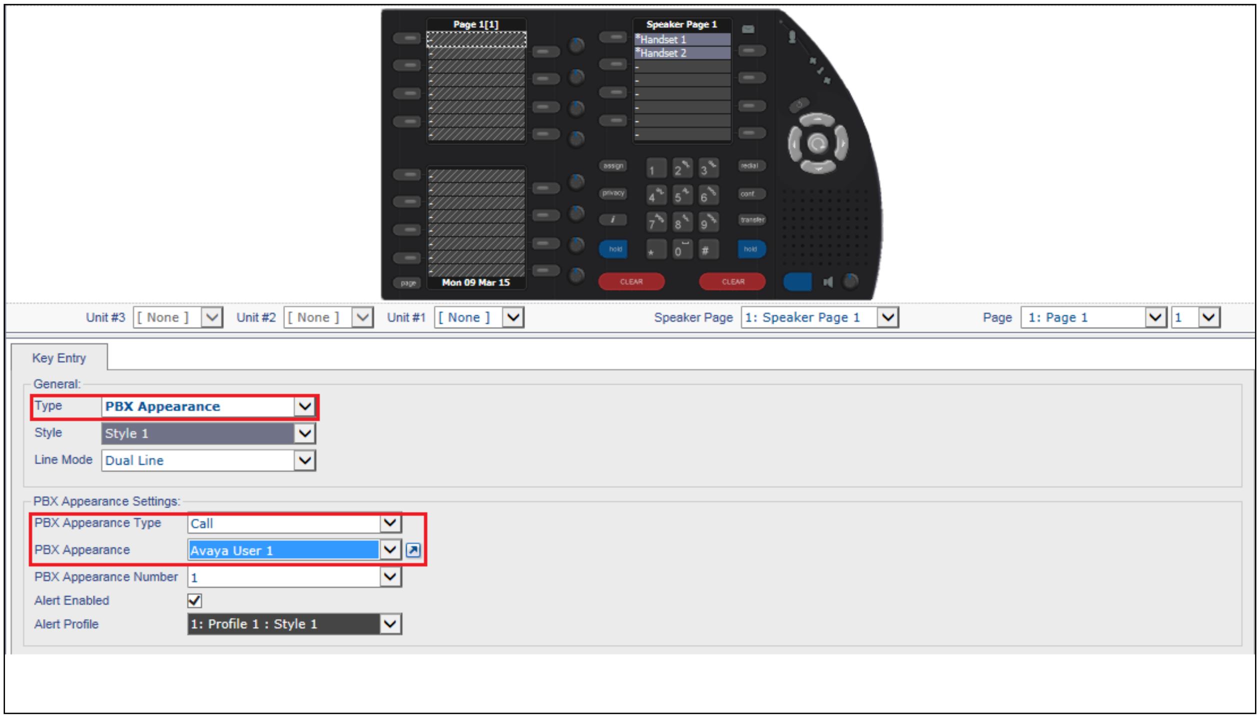
When selected the following layout is observed for a blank iTurret profile with ***Handset 1** and ***Handset 2** configured.



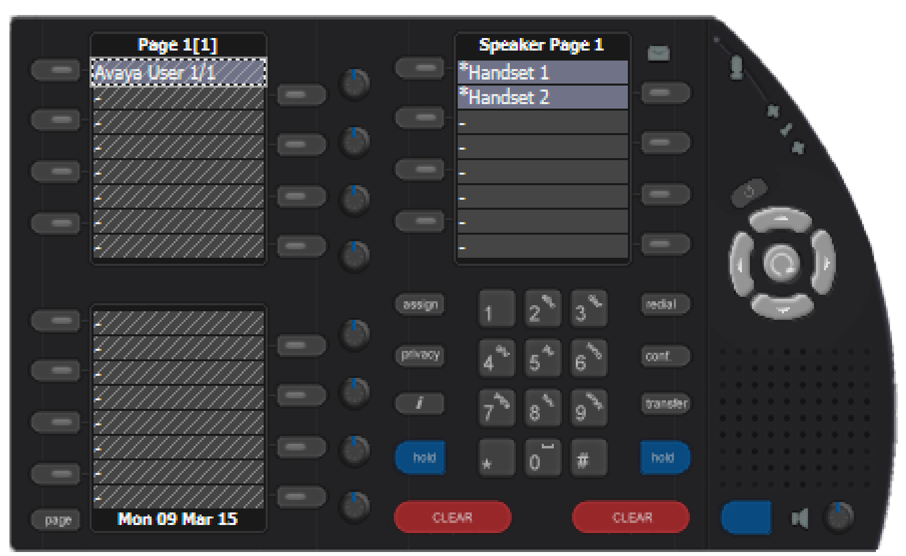
To add the keys for the call appearances, select a key (with hatching) and enter the following:

- **Type** Select **PBX Appearance** from the drop-down box.
- **PBX Appearance Type** Select **Call**, from the drop-down box.
- **PBX Appearance** Select the appearance given to this user (i.e., **Avaya User 1**).

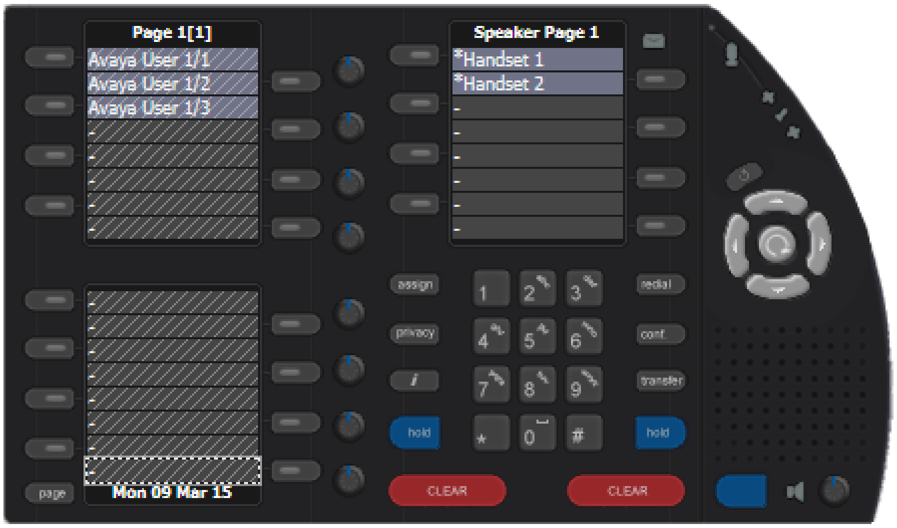
Click the **OK** button (not shown).



Once done the layout will look as follows.



Add two further instances of this appearance to the next two keys in the same way as above. The new iTurret layout will look as follows.

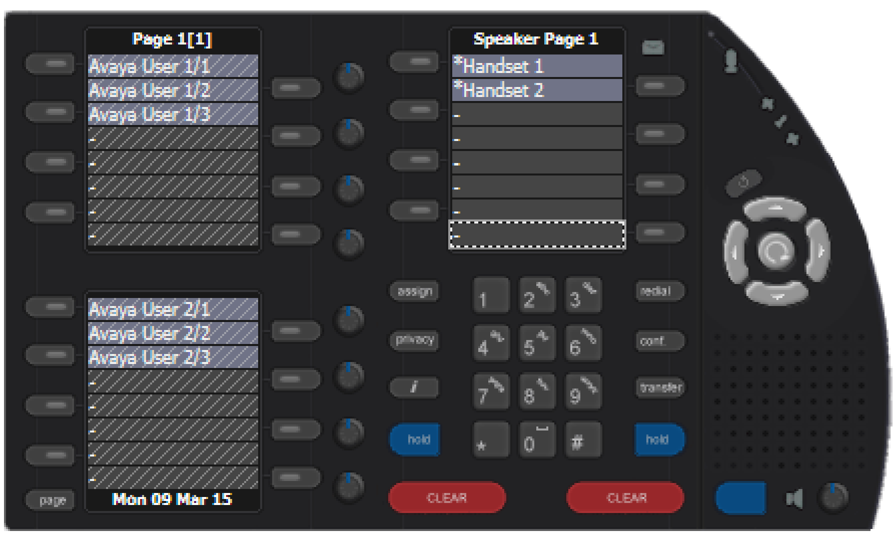


7.15.1. Add bridged appearances

To add bridged appearances, repeat **Section 7.11** and enter the following:

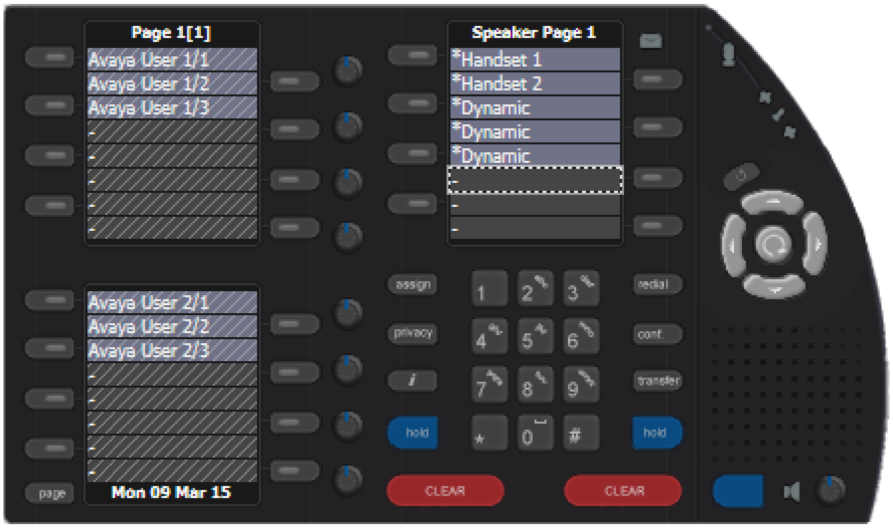
- **Type** Select **PBX Appearance** from the drop-down box
- **PBX Appearance Type** Select **Call**, from the drop-down box
- **PBX Appearance** Select the call appearance you have permissions to, but isn`t owned by this user (thus, it`s a bridged appearance)

Click the **OK** button (not shown). Repeat this step three times. The example below shows Avaya User 2 three times.



7.15.2. Add dynamic keys

Add three dynamic keys under the **handset 2 key** in the iTurret Layout using the procedure in **Section 7.11**, select the next available key under ***Handset 2** key and select **Dynamic** from the **Type** drop down box. The remaining fields are left at default. Click the **OK** button. Repeat this step three times. The example below shows the three dynamic keys added.

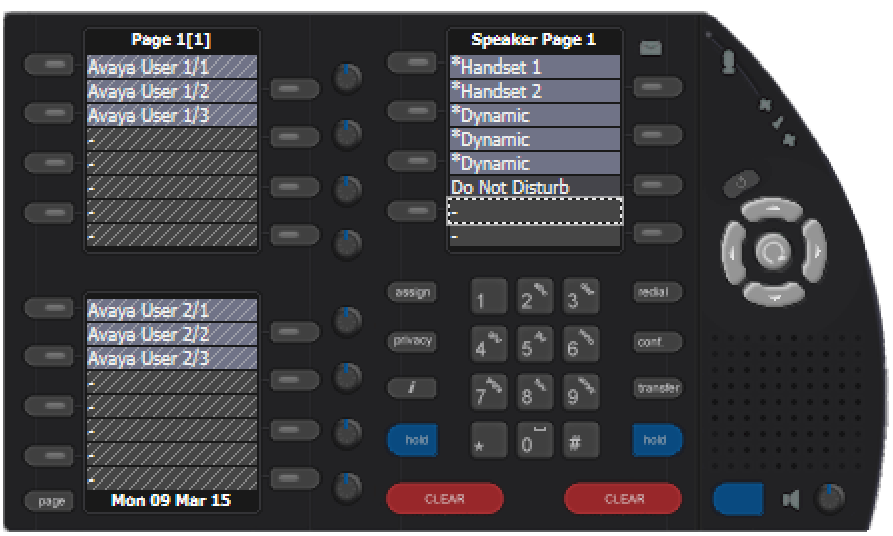


7.15.3. Add Do Not Disturb key

To add a single function key for **Do Not Disturb**, in the iTurret Layout, using the procedure in **Section 7.11**, select the next available key under the last **Dynamic** key and enter the following:

- **Type** Select **Function** from the drop-down box.
- **Function Type** Select **Do Not Disturb** from the drop-down box.

Click the **OK** button. Once done the layout will look as below.

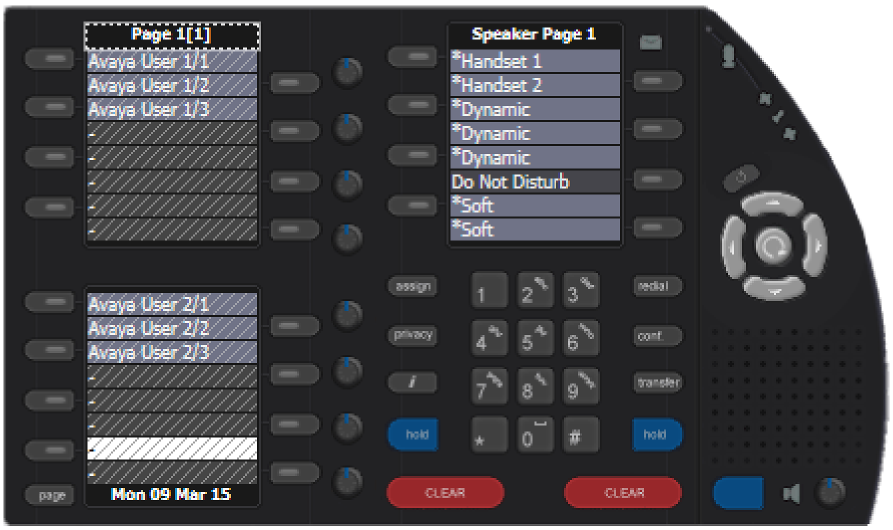


7.15.4. Add soft function keys

To add two soft function keys, in the iTurret Layout, using the procedure in **Section 7.11**, select the next available key under the Do Not Disturb key and enter the following:

- **Type** Select **Soft Function** from the drop-down box.
- **Function Type** Select **General** from the drop-down box.

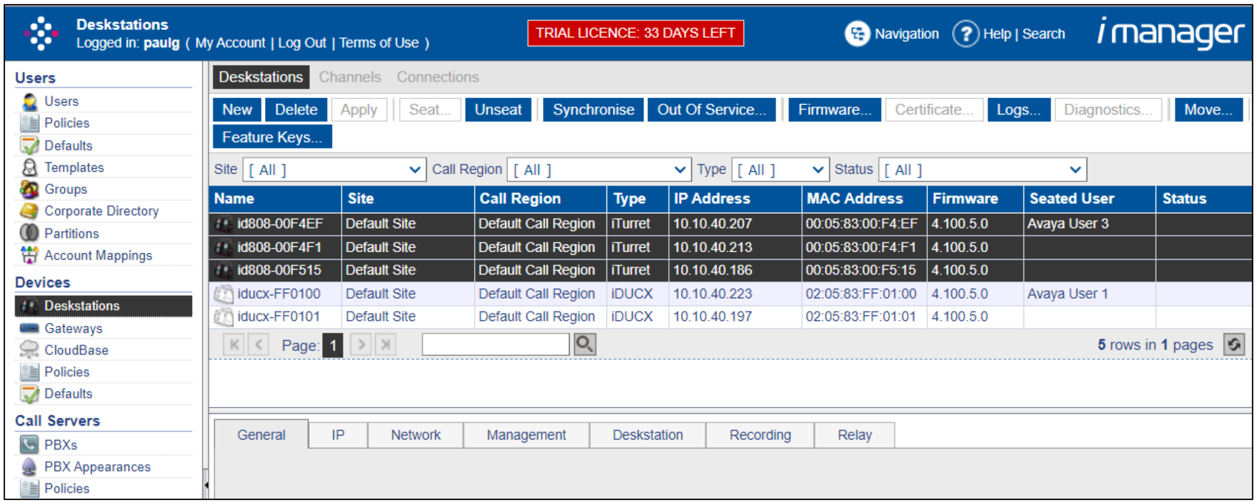
Click the **OK** button. Repeat this step two times. Once done the layout will look as below.



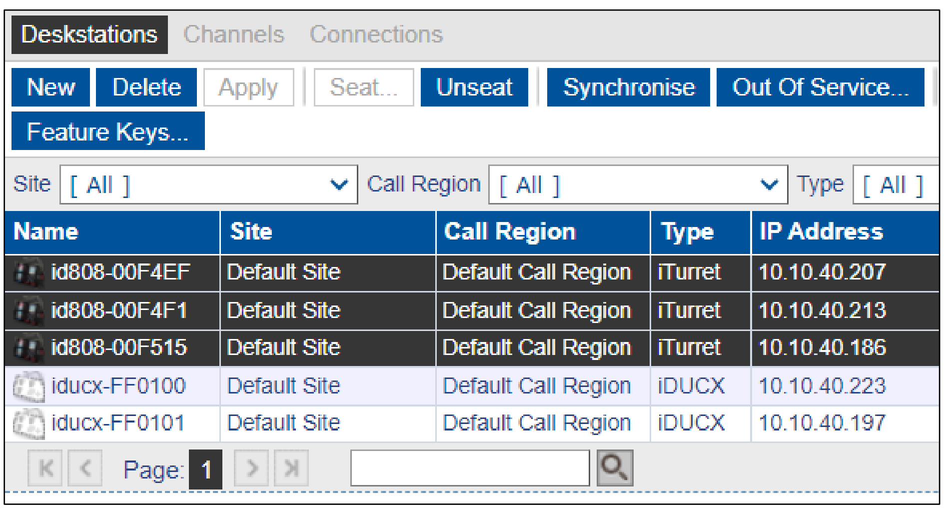
For more information on the types of keys available and adding, editing or removing, refer to the *Speakerbus iManager Administrator`s Guide.*

7.16. Synchronise Deskstations

Any changes made to the profile within iManager will be updated on the iTurret device after **OK** or **Apply** is pressed. However, some changes will require a synchronization to push the new configuration to the iTurret without disruption to the user. Select **Devices →Deskstations** and select the desired deskstations.



Click the **Synchronise** button.

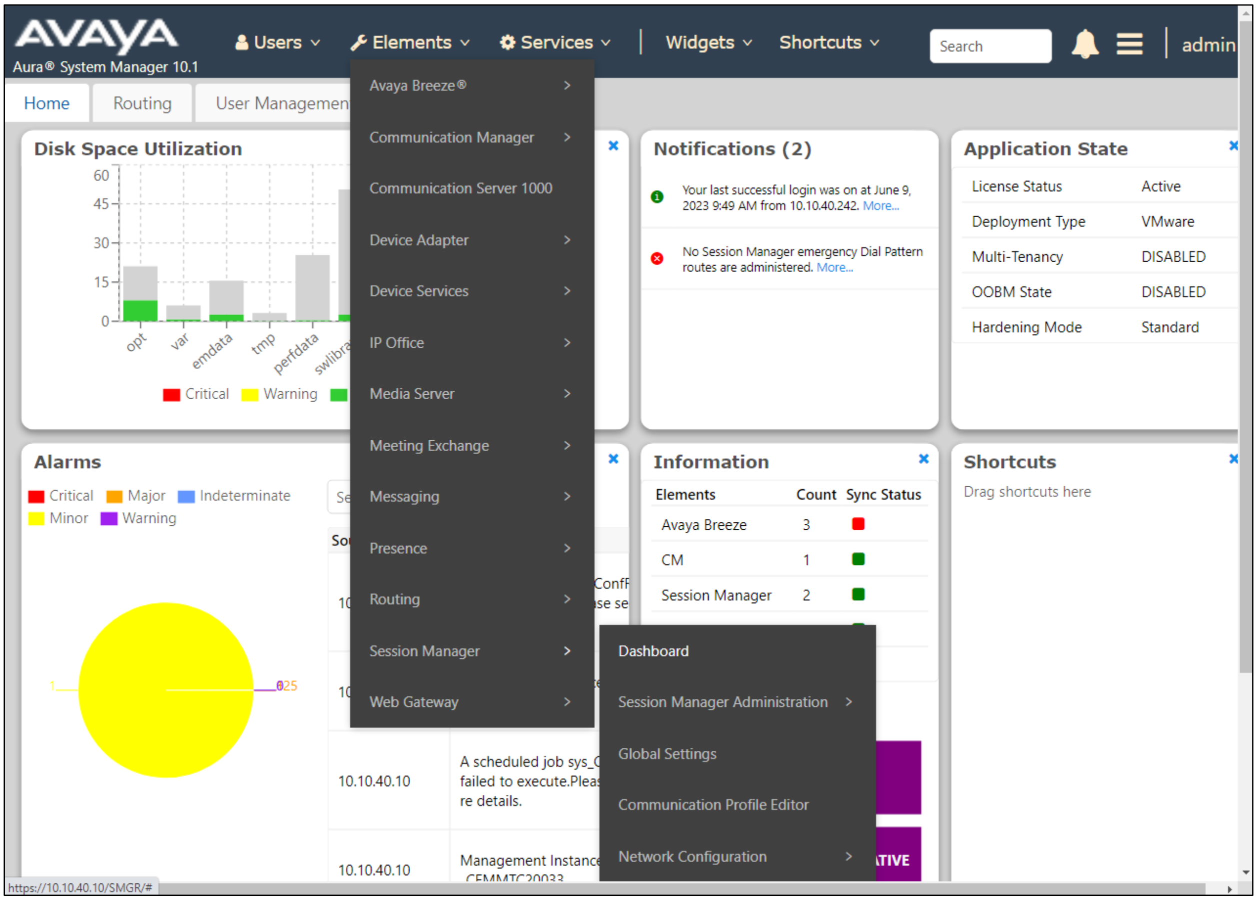


8. Verification Steps

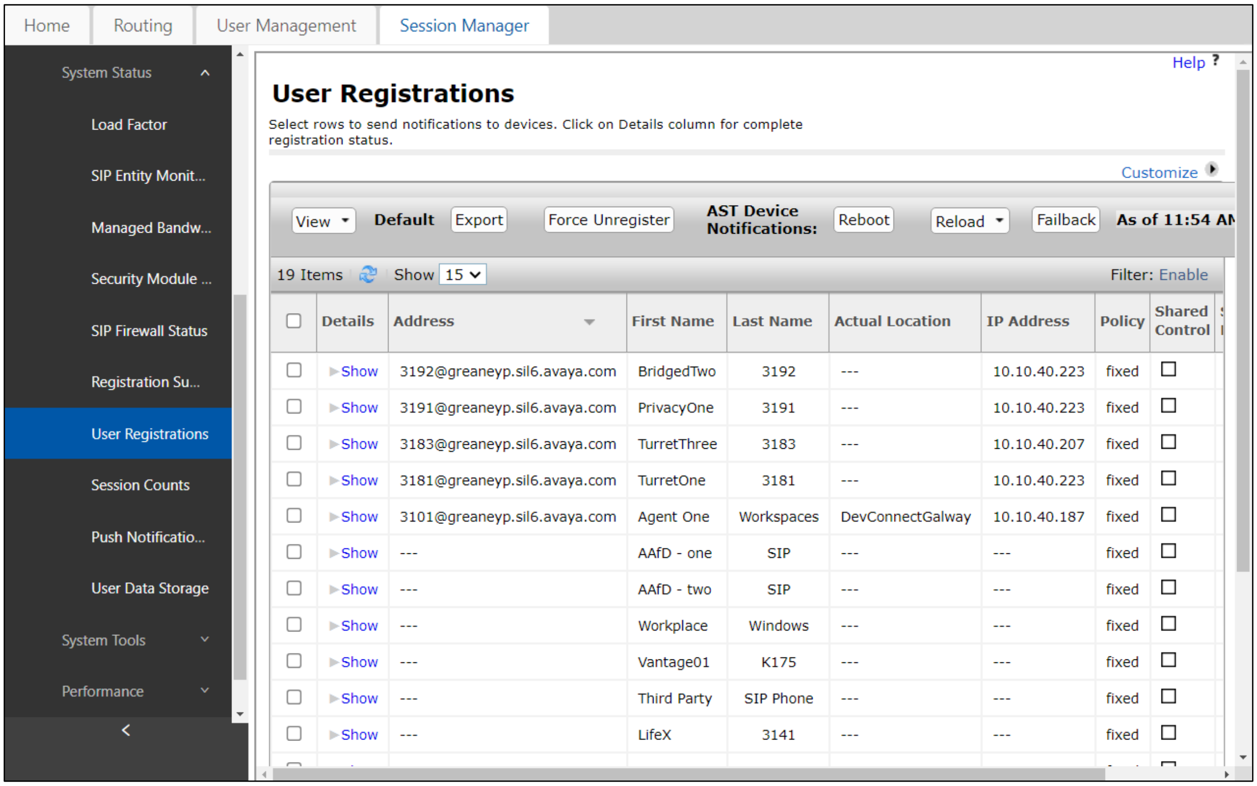
This section provides the tests that can be performed to verify correct configuration of the Avaya and Speakerbus solution.

8.1. Verify iTurret registration with Avaya Aura® Session Manager

To verify that the iTurret have successfully registered with Session Manager, from the System Manager Web interface click on **Elements → Session Manager**.



From the left window, click on **System Status → User Registrations**. This will display a summary of registered stations on each Session Manager as shown below. Note that **3181**, **3191** and **3192** are all registered which is a good indication that this iTurret is registered correctly with Session Manager.



8.2. Verify iTurret status

On the iTurret, verify that the status icons are green . These status icons indicate whether iTurret is connected to the network, iCMS server, and SIP registrar (i.e., Session Manager). Refer to **Section 10** for more details.

9. Conclusion

These Application Notes describe the compliance tested configuration of the Speakerbus iTurret v4.1 with Avaya Aura® Communication Manager 10.1 and Avaya Aura® Session Manager10.1. All tests passed with any observations noted in **Section 2.2**.

10. Additional References

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

[1] *Administering Avaya Aura® Communication Manager*, Release 10.1.
[2] *Avaya Aura® Communication Manager Feature Description and Implementation,* Release 10.1.
[3] *Administering Avaya Aura® Session Manager*, Release 10.1.
[4] *Administering Avaya Aura® System Manager*, Release 10.1.
[5] *Speakerbus iCMS Administrators Guide v4.0 R46*
[6] *Speakerbus Aria Touch User Guide R5*

Product Documentation for Speakerbus can be requested from info@speakerbus.com

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.