Avaya Solution & Interoperability Test Lab

Application Notes for Zenitel IP Operating Room Master with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using TCP – Issue 1.0

Abstract

These Application Notes describe the configuration steps for provisioning Zenitel IP Operating Room Master v6.1.1 to interoperate with Avaya Aura® Communication Manager R8.1 and Avaya Aura® Session Manager R8.1. Zenitel IP Operating Room Master is an IP Intercom that supports voice transmission using Session Initiation Protocol (SIP) and Transmission Control Protocol (TCP).

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 configuration steps for provisioning Zenitel IP Operating Room Master v6.1.1 to interoperate with Avaya Aura® Communication Manager R8.1 and Avaya Aura® Session Manager R8.1. Zenitel IP Operating Room Master is an IP Intercom that supports voice transmission using Session Initiation Protocol (SIP) and Transmission Control Protocol (TCP).

The IP Operating Room (OR) Master Station is an intercom station intended for use in operating theatres and clean rooms. The station front plate is totally flat and without any holes to minimize bacteria accumulation. With a large backlit display and Vingtor-Stentofon audio technology the station allows users to read caller ID, listen and talk at a distance.

During compliance testing, each IP OR Master station was set up as a SIP user on Session Manager and underwent testing of various call scenarios with other Avaya telephones.

The following models in the Zenitel IP OR Master station family were tested: IP Desk Master V2, IP Master V2, and IP Flush Master. Other models in the IP OR Master family are not covered by this compliance test.

Note: The Zenitel IP Operating Room Master phones may be referred to as 'IP OR Master station', 'IP OR Master Intercom phone', or 'IP OR Master' throughout this document, but they all refer to the same phones that were tested.

2. General Test Approach and Test Results

The general test approach was to place calls to and from the IP OR Master phones and exercise basic telephone operations. For serviceability testing, failures such as LAN cable pulls, and hardware resets were performed.

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/Smartphone 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 Smartphone interfaces, different manufacturers utilize different Smartphone/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 Zenitel IP OR Master did not include use of any specific encryption features as requested by Zenitel.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing. IP Desk Master V2, IP Master V2, and IP Flush Master models were tested. The feature testing was to verify that:

- IP OR Master successfully registers with Session Manager using the TCP protocol.
- IP OR Master successfully establishes audio calls with RTP audio to Avaya H.323, SIP and digital endpoints.
- IP OR Master successfully establishes audio calls with a simulated PSTN.
- IP OR Master successfully negotiates the appropriate audio codec.
- DTMF tones could be passed successfully to energize relay on the IP OR Master unit to allow a door to be opened from the phone sending the DTMF tone or perhaps to switch audio direction.
- IP OR Master successfully calls multiple Avaya destinations in a hunt group.
- IP OR Master successfully calls a variety of endpoints in its call list set on the IP OR Master phone.

The serviceability testing focused on verifying the ability of IP OR Master to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet cable on Session Manager.

Note: Compliance testing was carried out with the IP OR Master phones set to use TCP/RTP. Testing was also carried out with IP OR Master phones set to use TLS/SRTP and these Application Notes are labelled, *Application Notes for Zenitel IP Operating Room Master with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using TLS.*

2.2. Test Results

All test cases passed successfully with the following observation noted.

1. Call Park has a different meaning on the IP OR Master functionality than that of the Call Park feature on Communication Manager. When the Call Park function is used on IP OR Master it places multiple calls on hold. For every Direct Access Key (DAK) with Call Park configured, there can be only one active or resumed call.

2.3. Support

Technical support on Zenitel IP OR Master can be obtained through the following:

- **Phone:** +1 816 231 7200 (Americas) +47 4000 2700 (Global)
- **Email:** cs@zenitel.com
- **Web:** https://www.zenitel.com/customer-service

3. Reference Configuration

Figure 1 illustrates a test configuration that was used to compliance test the interoperability of IP OR Master with Session Manager and Communication Manager. The configuration consists of H.323 and Digital phones registering directly to Communication Manager and SIP phones registering to Session Manager using the features on Communication Manager. A SIP trunk connects Communication Manager to a simulated PSTN.

Note: The Zenitel IP OR Master phones register to Session Manager the same as the Avaya SIP phones.

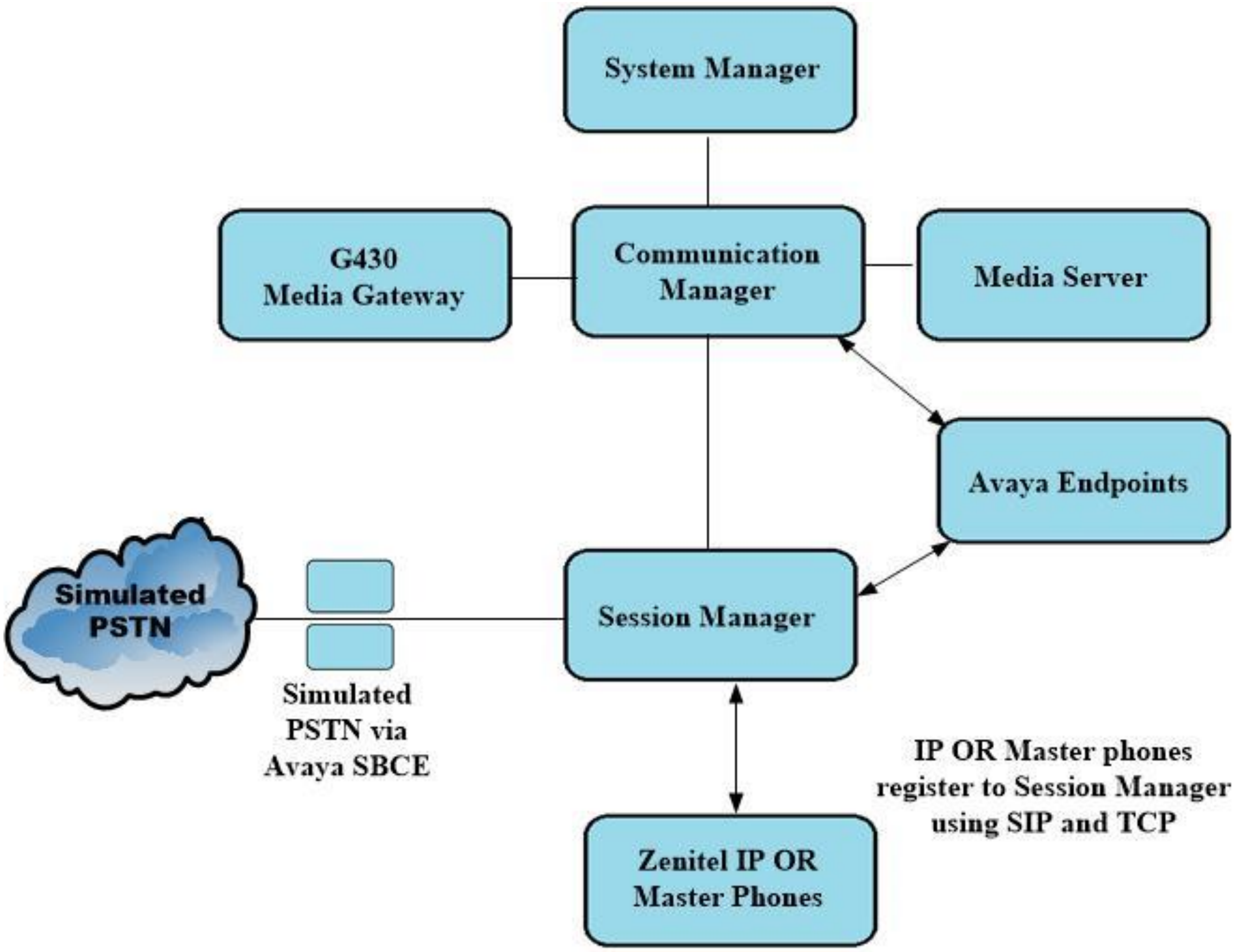


Figure 1: Configuration of Avaya Aura® Communication Manager and Avaya Aura® Session Manager with Zenitel IP Operating Room Master

4. Equipment and Software Validated

The following equipment and software were used for the compliance test.

Avaya Equipment	Software / Firmware Version
Avaya Aura® System Manager running on a virtual server	8.1.3.0 Build No. – 8.1.0.0.733078 Software Update Revision No: 8.1.3.0.1011784 Feature Pack 3
Avaya Aura® Session Manager running on a virtual server	8.1.3 Build No. – 8.1.3.0.813014
Avaya Aura® Communication Manager running on a virtual server	8.1.3 – FP3 R018x.01.0.890.0 Update ID 01.0.890.0-26568
Avaya Session Border Controller for Enterprise	8.1.1.0-26-19214
Avaya Aura® Media Server	8.0.2.138
Avaya G430 Media Gateway	41.16.0/1
Avaya J179 H.323 Deskphone	6.8304
Avaya Vantage K175 SIP Deskphone	3.0.0.1.0006
Avaya 9408 Digital Phone	2.00
Zenitel Equipment	**Software / Firmware Version**
Zenitel IP Operating Room Master - IP Desk Master - IP Master V2 - IP Flush Master	 6.1.1.0 6.1.1.0 02.11.3.0

5. Configure Avaya Aura® Communication Manager

It is assumed that a fully functioning Communication Manager is in place with the necessary licensing with SIP trunks in place to Session Manager. For further information on the configuration of Communication Manager please see **Section 10** of these Application Notes.

Note: A printout of the Signalling and Trunk groups that were used during compliance testing can be found in the **Appendix** of these Application Notes.

The following sections go through the following.

- System Parameters
- Dial Plan Analysis
- Network Region
- IP Codec

5.1. Configure System Parameters

Ensure that the SIP endpoints license is valid as shown below by using the command **display system-parameters customer-options**.

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

     G3 Version: V17                               Software Package: Enterprise
       Location: 2                                  System ID (SID): 1
       Platform: 28                                 Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 48000 168
                                      Maximum Stations: 36000 44
                              Maximum XMOBILE Stations: 36000 0
                    Maximum Off-PBX Telephones - EC500: 41000 2
                    Maximum Off-PBX Telephones -   OPS: 41000 20
                    Maximum Off-PBX Telephones - PBFMC: 41000 0
                    Maximum Off-PBX Telephones - PVFMC: 41000 0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
                         Maximum Survivable Processors: 313   1
```

5.2. Configure Dial Plan Analysis

Use the **change dialplan analysis** command to configure the dial plan using the parameters shown below. Extension numbers (**ext**) are those beginning with **1**. Feature Access Codes (**fac**) use digits **8** and **9** and use characters ***** or **#**.

```
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   ext
   3           4   udp
   6           4   ext
   8           1   fac
   9           1   fac
   *8          4   dac
   *           3   fac
   #           3   fac
```

5.3. Configure Network Region

Use **change ip-network-region x** (where x is the network region to be configured) to assign an appropriate domain name to be used by Communication Manager, in the example below **devconnect.local** is used. Note that this domain is also configured in **Section 6.1.1**.

```
change ip-network-region 1                                       Page   1 of  20
                               IP NETWORK REGION
  Region: 1       NR Group: 1
Location: 1       Authoritative Domain: devconnect.local
    Name: PG Default             Stub Network Region: n
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
```

5.4. Configure IP-Codec

Use the **change ip-codec-set x** (where x is the ip-codec set used) command to designate a codec set compatible with the IP OR Master phones. During compliance testing the codecs **G.711A**, **G.729A** and **G.722** were tested.

For compliance testing the Avaya phones are set to use Media Encryption and the IP OR Master phones have no encryption set, so **none** must be present under **Media Encryption**.

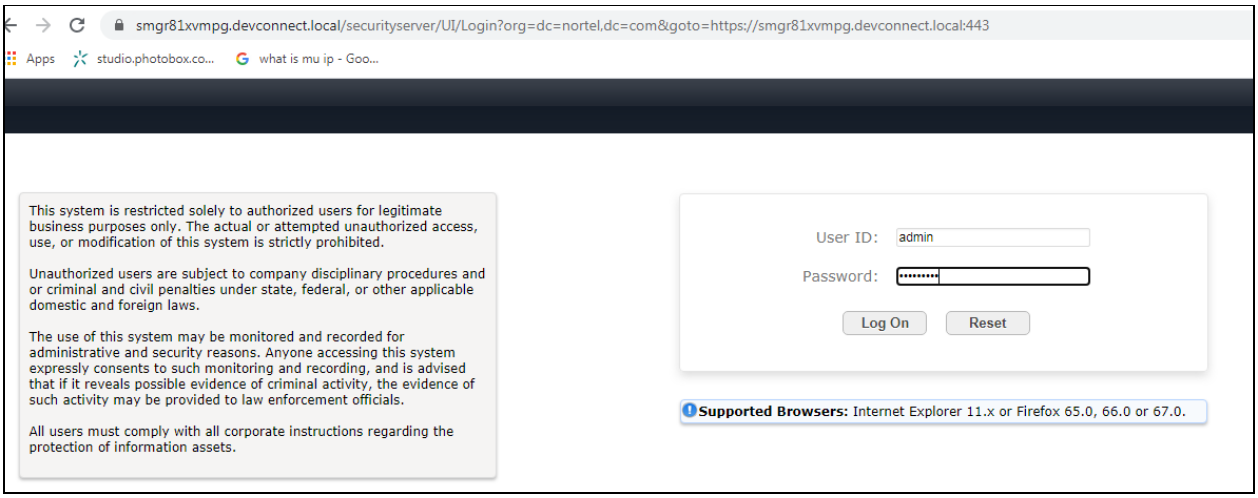
```
change ip-codec-set 1                                           Page   1 of   2
                          IP MEDIA PARAMETERS
    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711A           n          2        20
 2: G.729A           n          2        20
 3: G.722.2          n          1        20
 4: G.722-64K                   2        20
 5:
 6:
 7:
     Media Encryption                    Encrypted SRTCP: enforce-unenc-srtcp
 1: 1-srtp-aescm128-hmac80
 2: none
 3:
 4:
```

6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. Session Manager is configured via System Manager. The procedures include the following areas:

- Domains and Locations
- Adding Zenitel IP OR Master SIP Users

To make changes on Session Manager a web session is established to System Manager. Log into System Manager by opening a web browser and navigating to https://<System Manager FQDN>/SMGR. Enter the appropriate credentials for the **User ID** and **Password** and click on **Log On**.



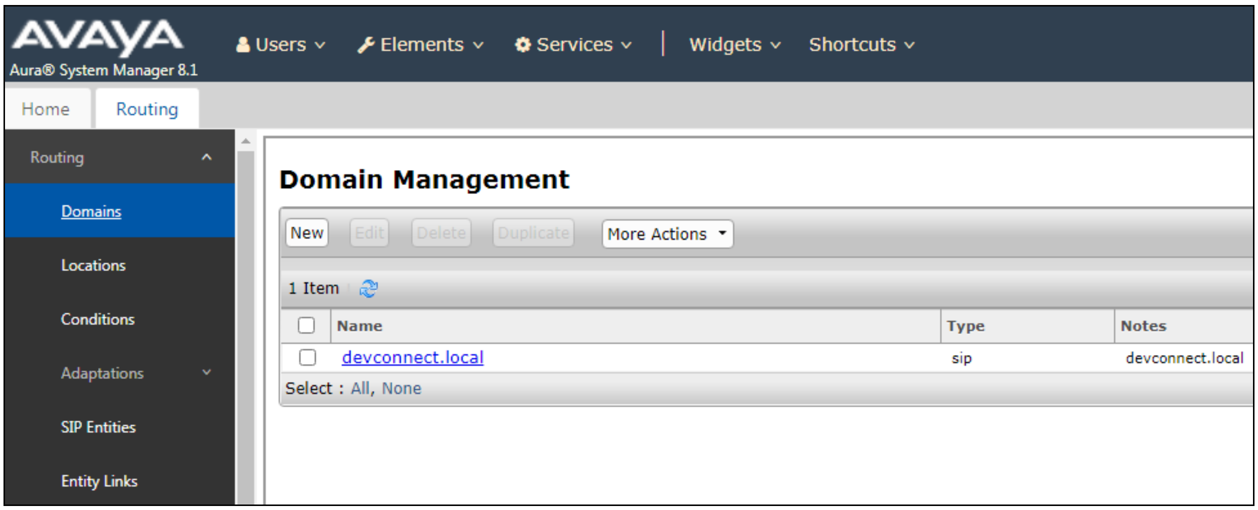
Once logged in navigate to **Elements** and click on **Routing** (not shown).

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.

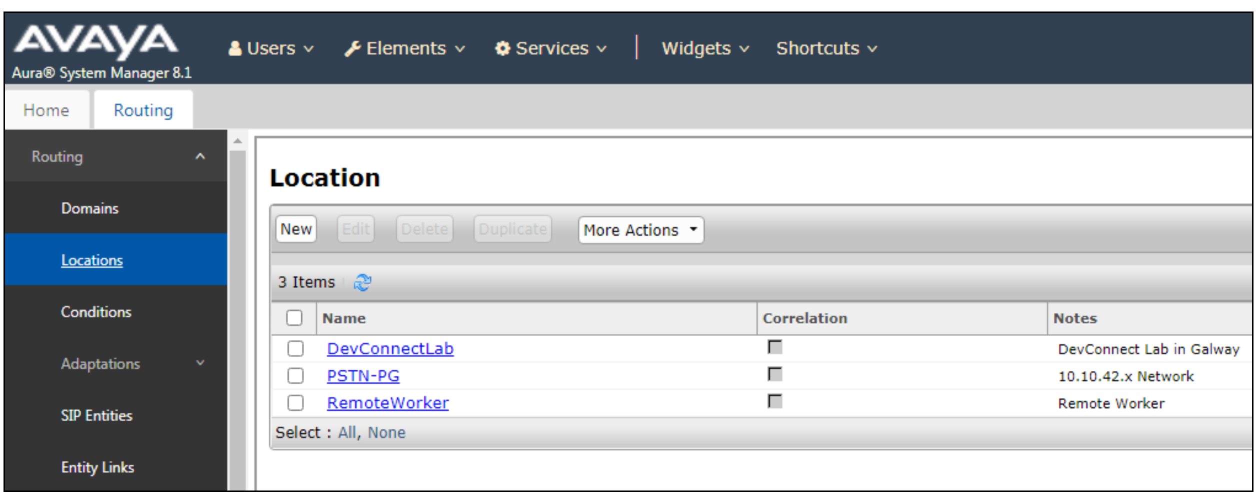
6.1.1. Display the Domain

Select **Domains** from the left window. This will display the domain configured on Session Manager. For compliance testing this domain was **devconnect.local** 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.



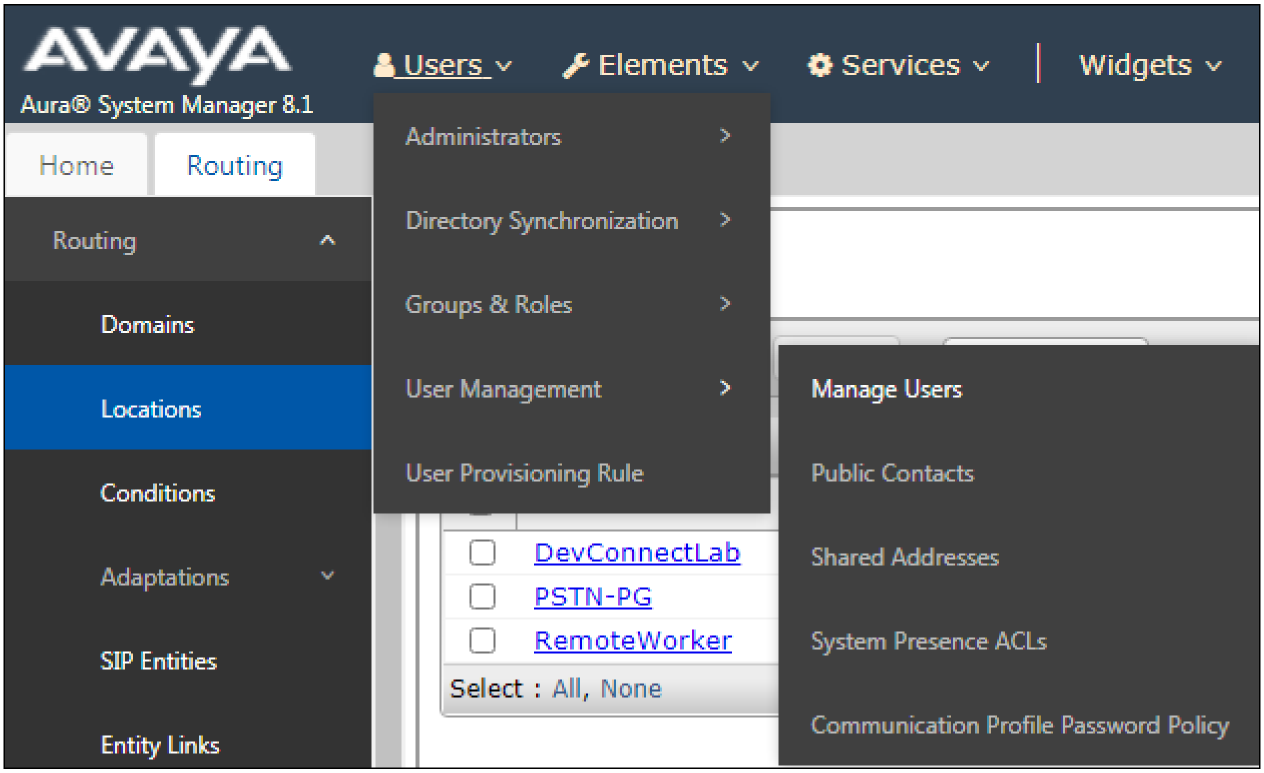
6.1.2. Display the Location

Select **Locations** from the left window and this will display the location setup. The example below shows the location **DevConnectLab_PG** 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.

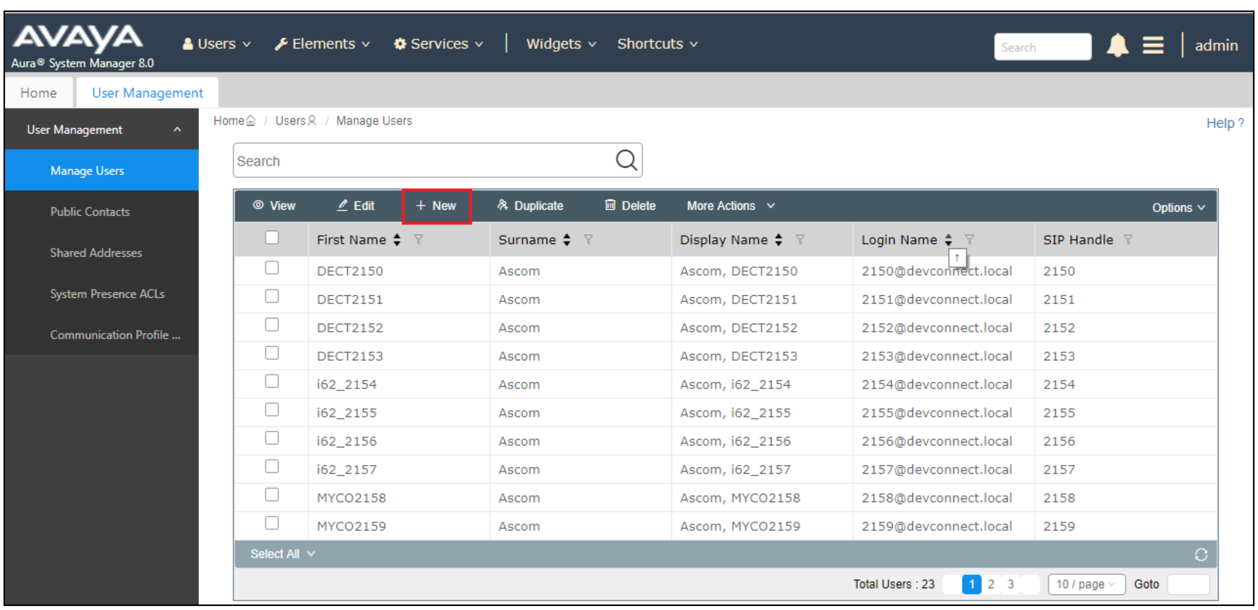


6.2. Adding Zenitel IP Operating Room Master SIP Users

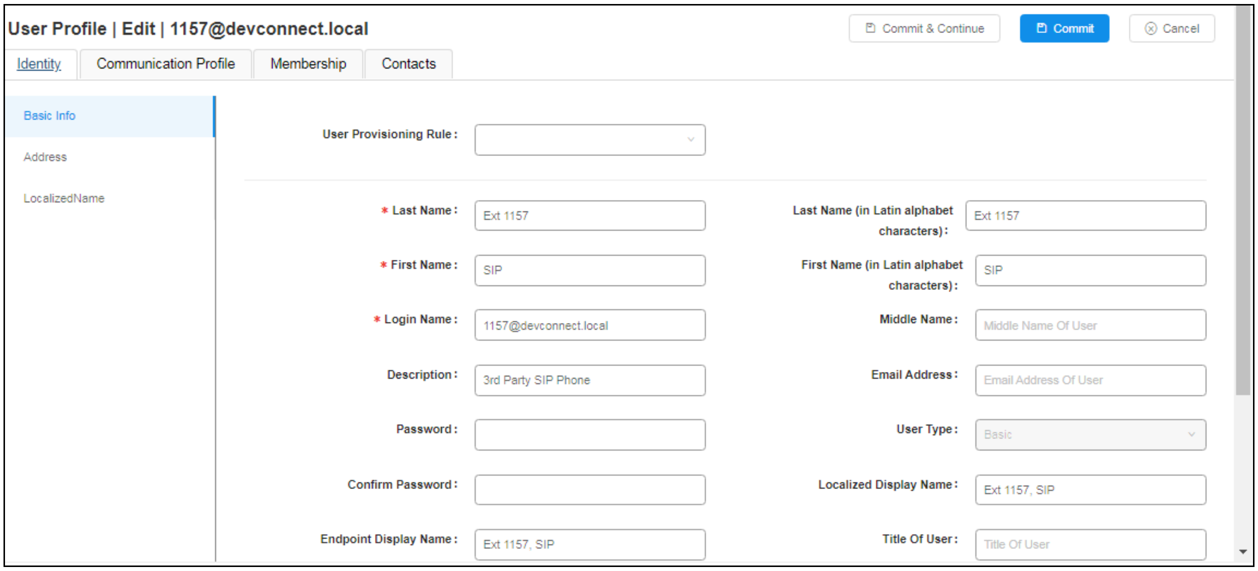
From the top of the home page click on **Users →User Management → Manager Users** as shown below.



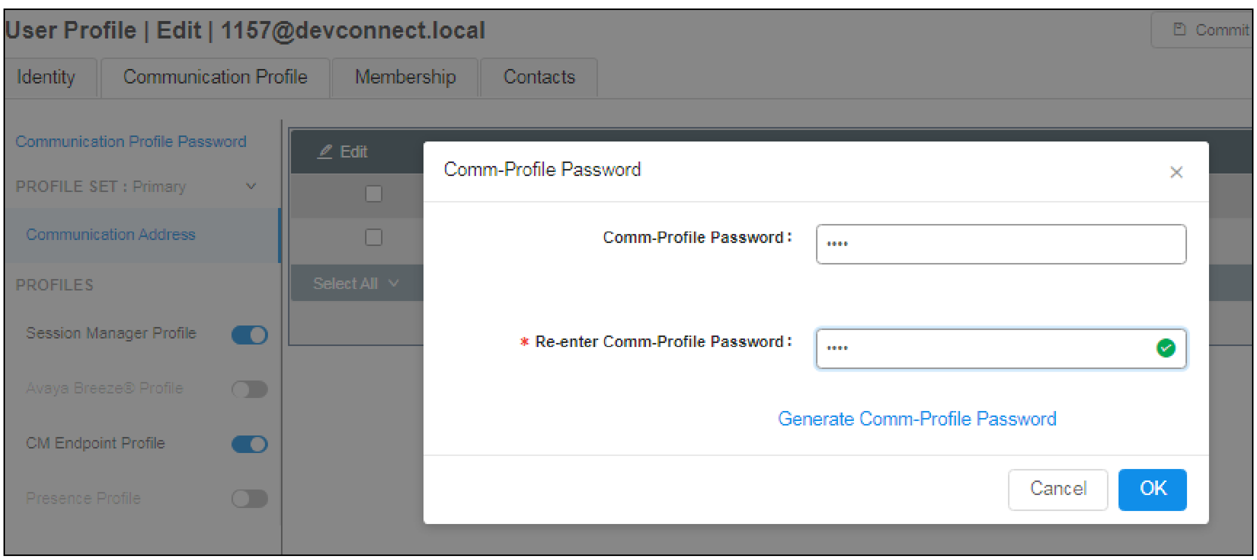
From **Manager Users** section, click on **New** to add a new SIP user.



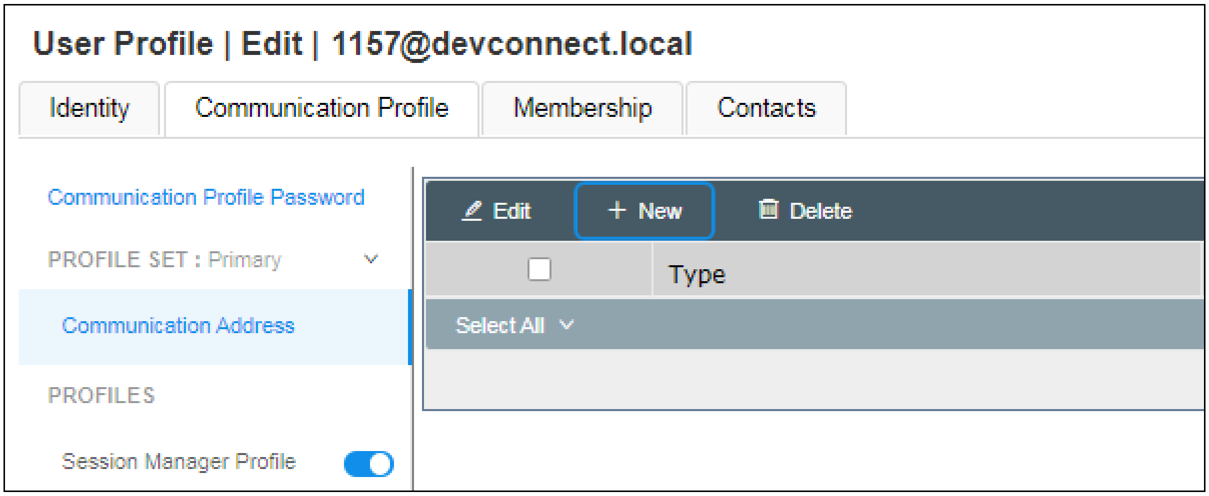
Under the **Identity** tab fill in the user's **Last Name** and **First Name** as shown below. Enter the **Login Name**, following the format of "user id@domain". The remaining fields can be left as default.



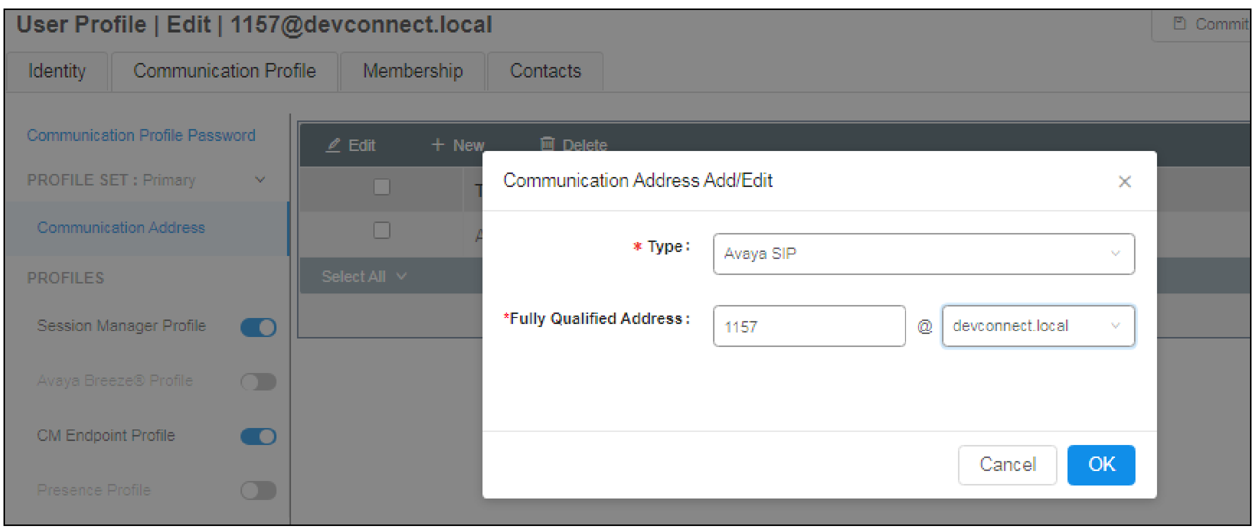
Under the **Communication Profile** tab enter **Communication Profile Password** and **Re-enter Comm-Profile Password**, note that his password is required when configuring the IP OR Master phone in **Section 7.2**.



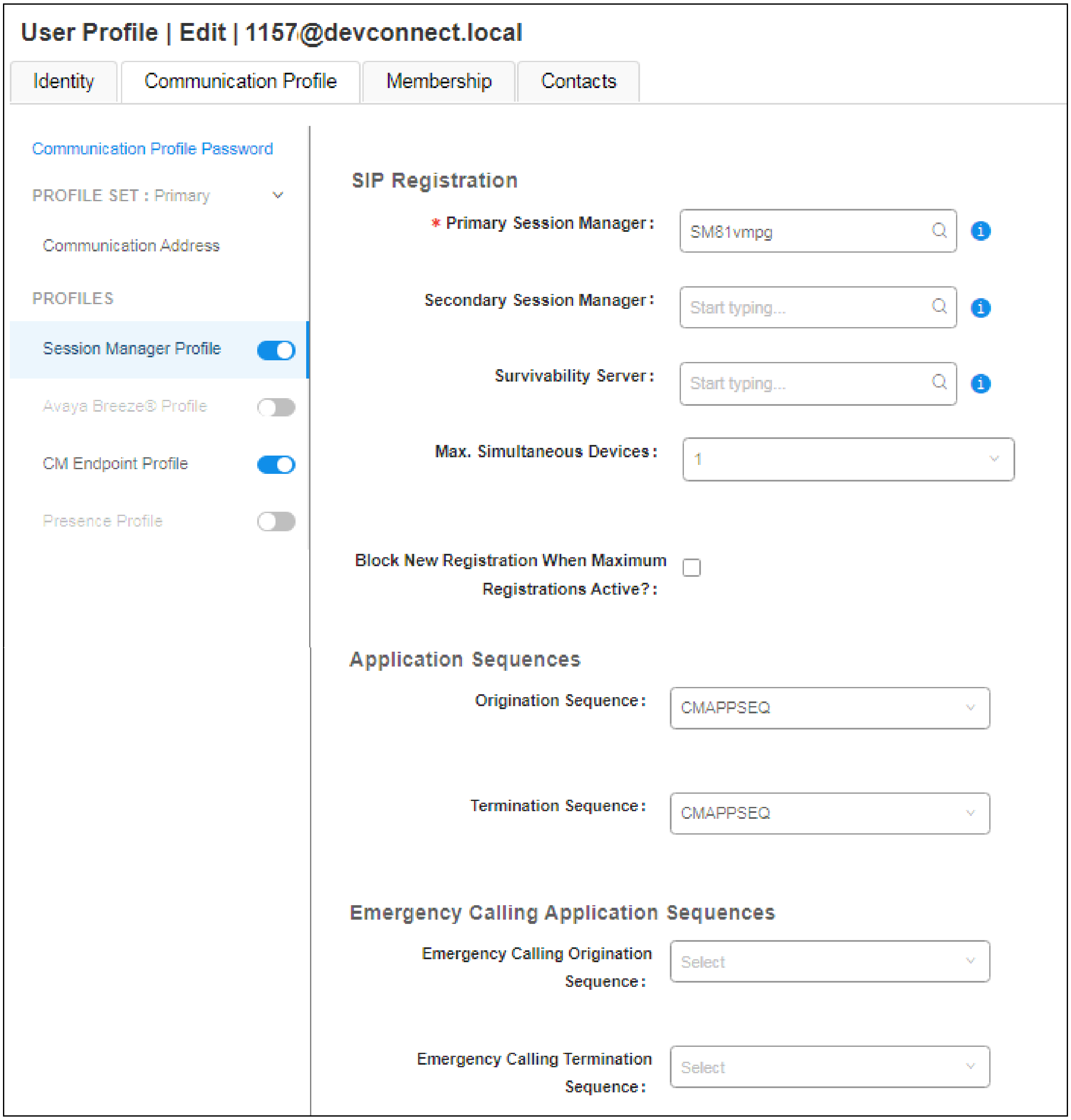
Staying on the **Communication Profile** tab, click on **New** to add a new **Communication Address**.



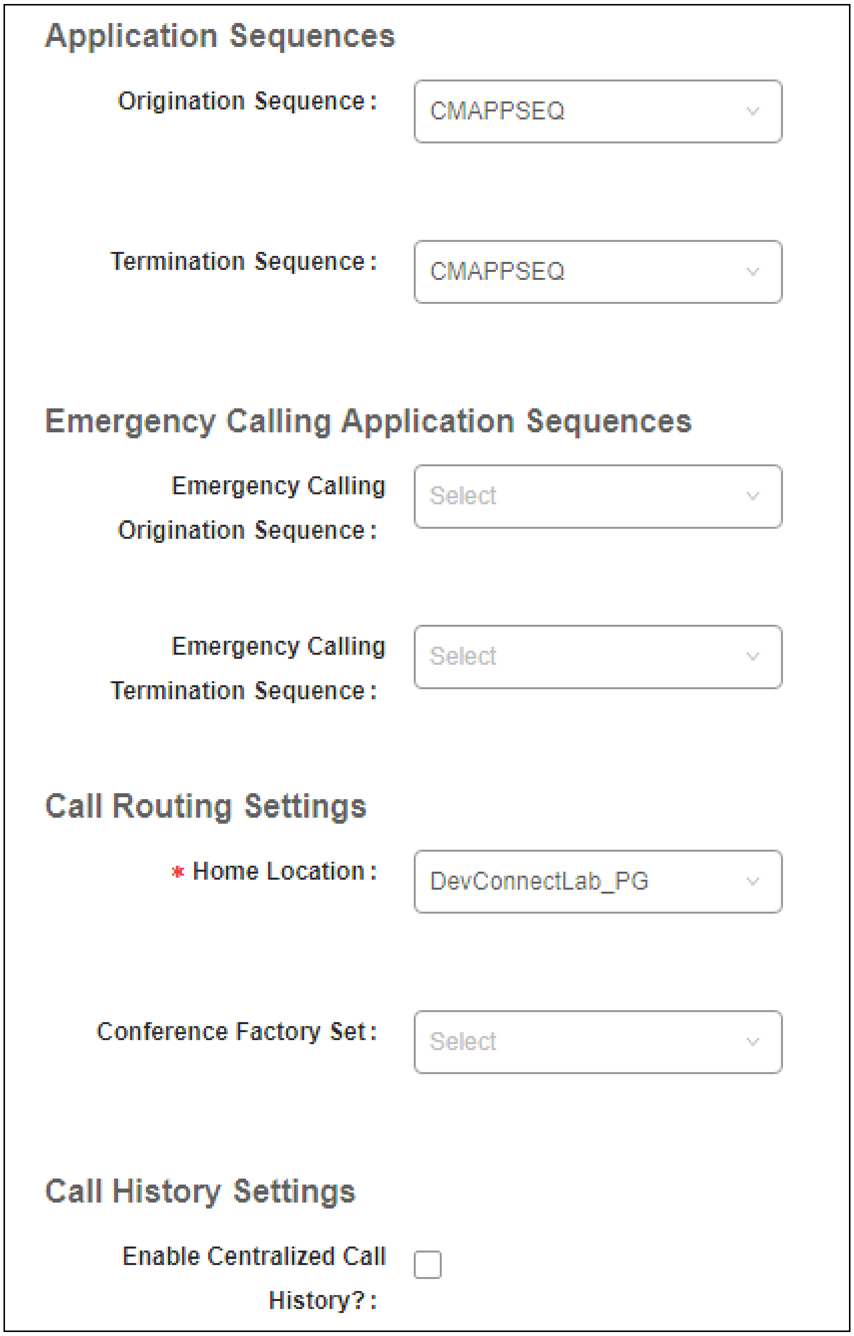
Enter the extension number and the domain for the **Fully Qualified Address** and click on **OK** once finished.



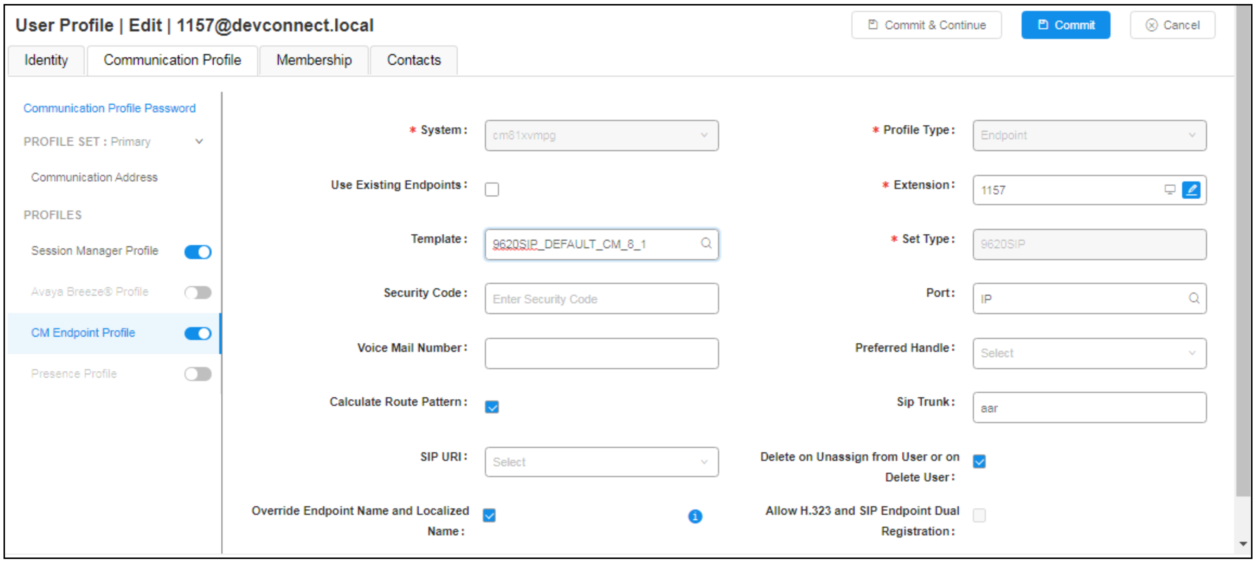
Ensure **Session Manager Profile** is checked and enter the **Primary Session Manager** details, enter the **Origination Sequence** and the **Termination Sequence**. Scroll down to complete the profile.



Enter the **Home Location**, this should be the location configured in **Section 6.1.2**. Click on **Commit** at the top of the page (not shown).

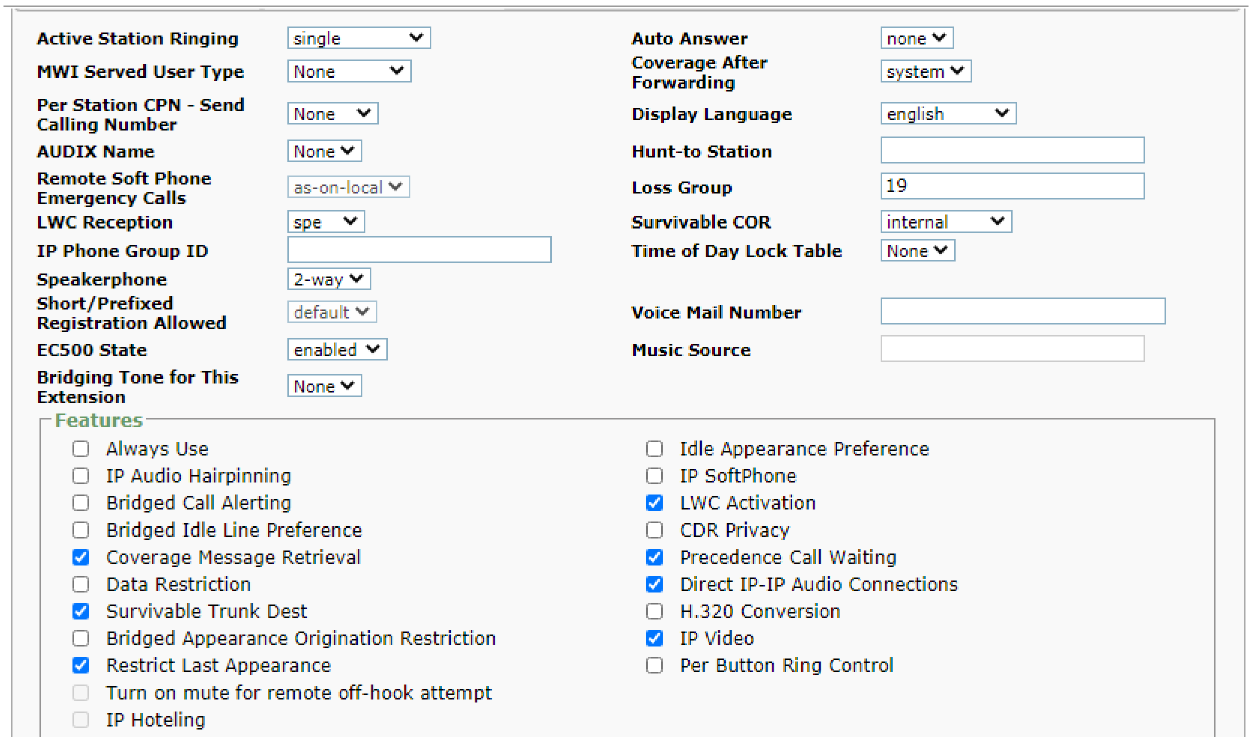


Ensure that **CM Endpoint Profile** is selected in the left window. Select the Communication Manager that is configured for the **System** and choose the **9620SIP_DEFAULT_CM_8_1** as the **Template**. **Sip Trunk** should be set to **aar**, providing that the routing is setup correctly on Communication Manager. The **Profile Type** should be set to **Endpoint** and the **Extension** is the number assigned to the IP OR Master phone. Click on **Endpoint Editor** to configure the buttons and features for that phone on Communication Manager.

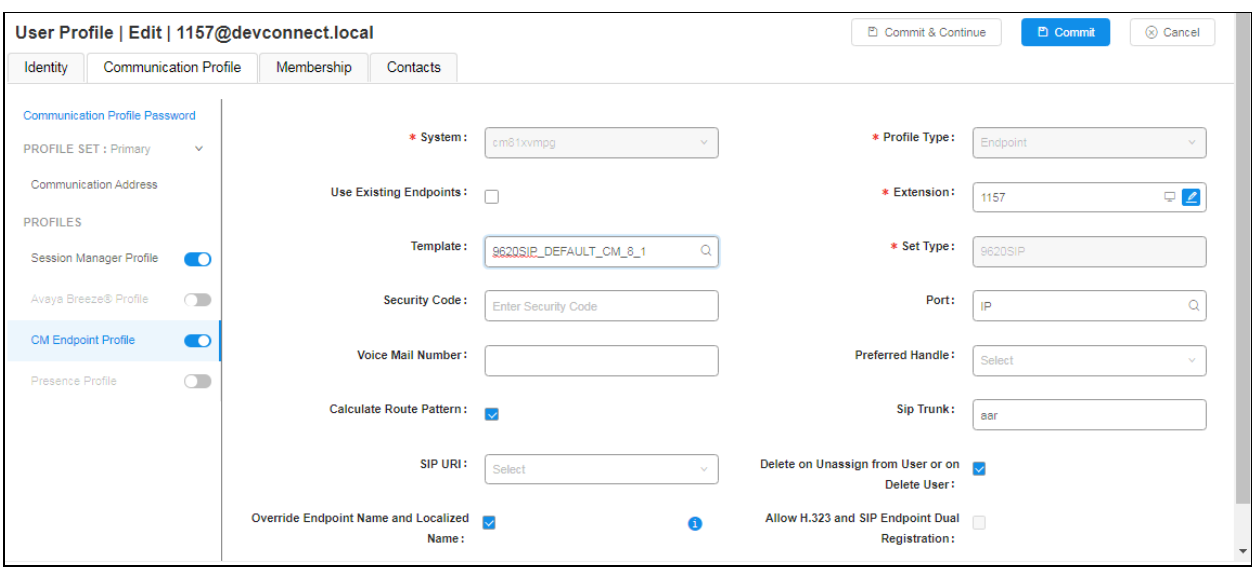


Under the **Feature Options** tab (not shown), if the IP OR Master phone is capable of video then **IP Video** can be ticked. Other tabs can be checked but for compliance testing the values were left as default. Click on **Done** (not shown) to complete.

Note: For compliance testing the default value of three call appearance buttons were used. This can be changed under the **Button Assignment** tab.

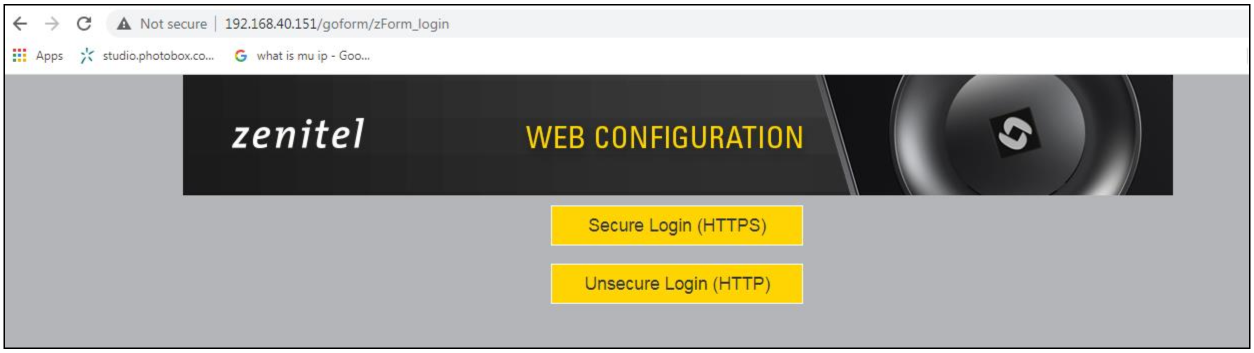


Once the **CM Endpoint Profile** is completed correctly, click on **Commit** to save the new user.

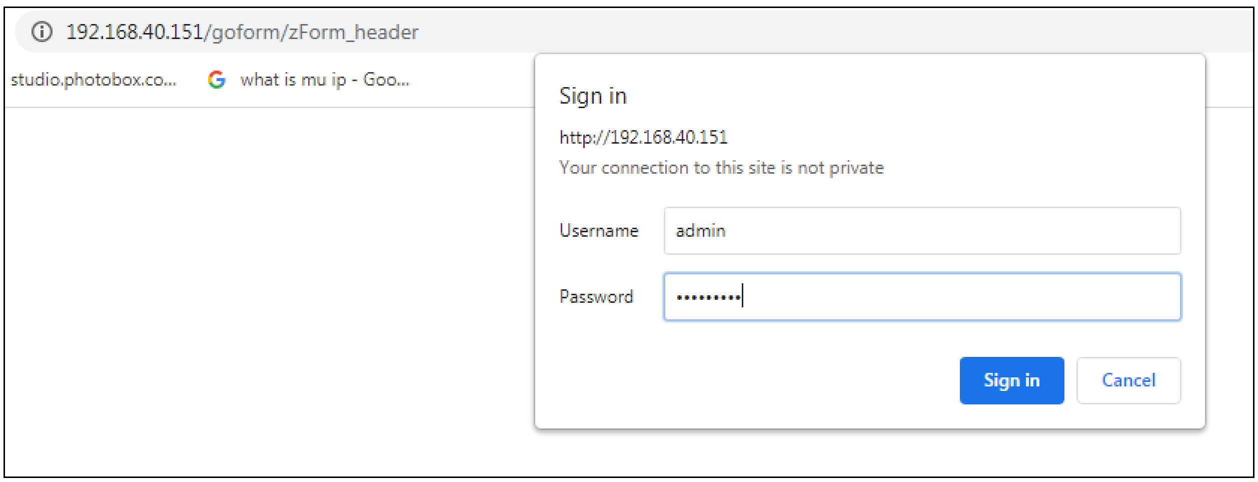


7. Configure Zenitel IP Operating Room Master Phone

The following steps detail the configuration for IP OR Master using the web interface. Access the IP OR Master web interface, enter **http://<ipaddress>** in an Internet browser window, where **<ipaddress>** is the IP address of the IP OR Master phone in question. For compliance testing **Unsecure Login (HTTP)** was chosen.

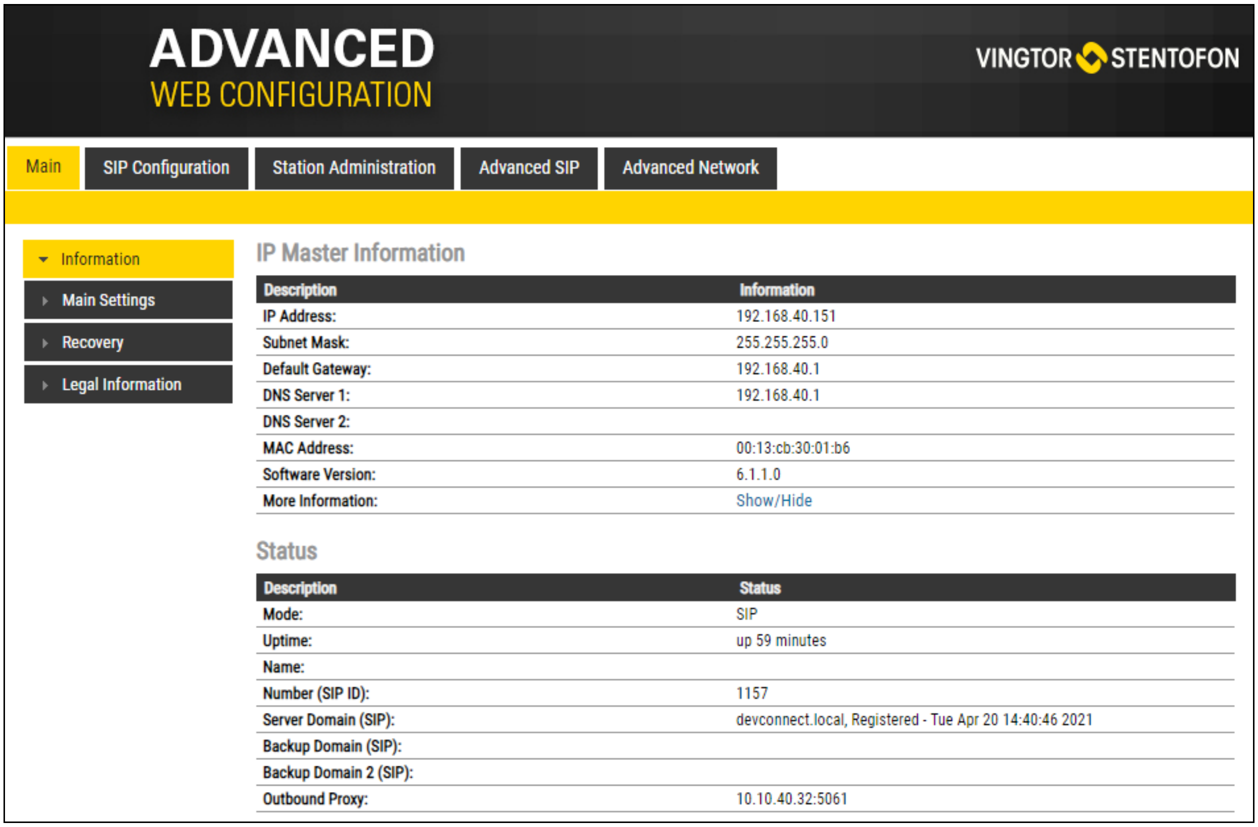


Log in with the appropriate credentials.



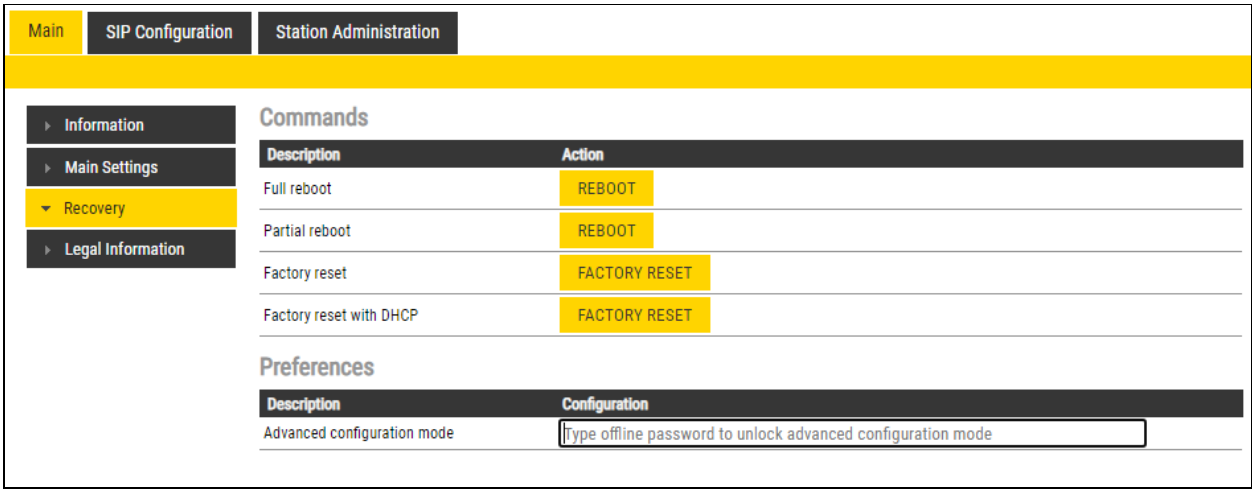
Upon logging in, information on that IP OR Master station is displayed. The following settings should be checked.

- Configure Advanced Configuration Mode
- SIP Configuration
- Direct Access Keys
- Audio

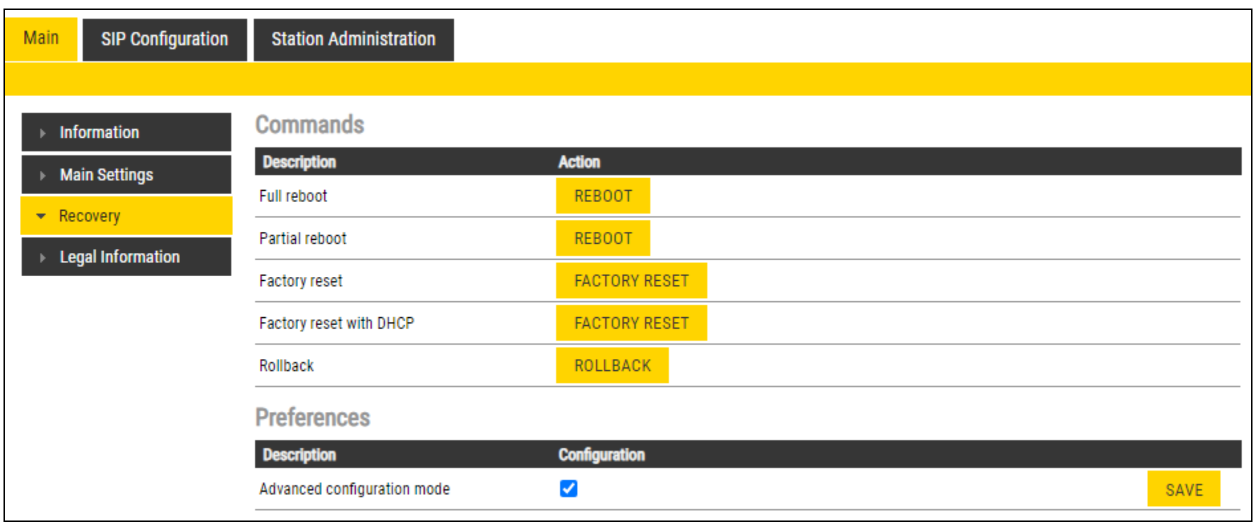


7.1. Configure Advanced Configuration Mode

Select **Recovery** from the left window and under **Preferences** enter the password for **Advanced configuration mode**, this password can be obtained from a Zenitel engineer as per **Section 2.3**.



Once the password is entered the check box will appear as shown and a tick can be placed and click on **save** to confirm.

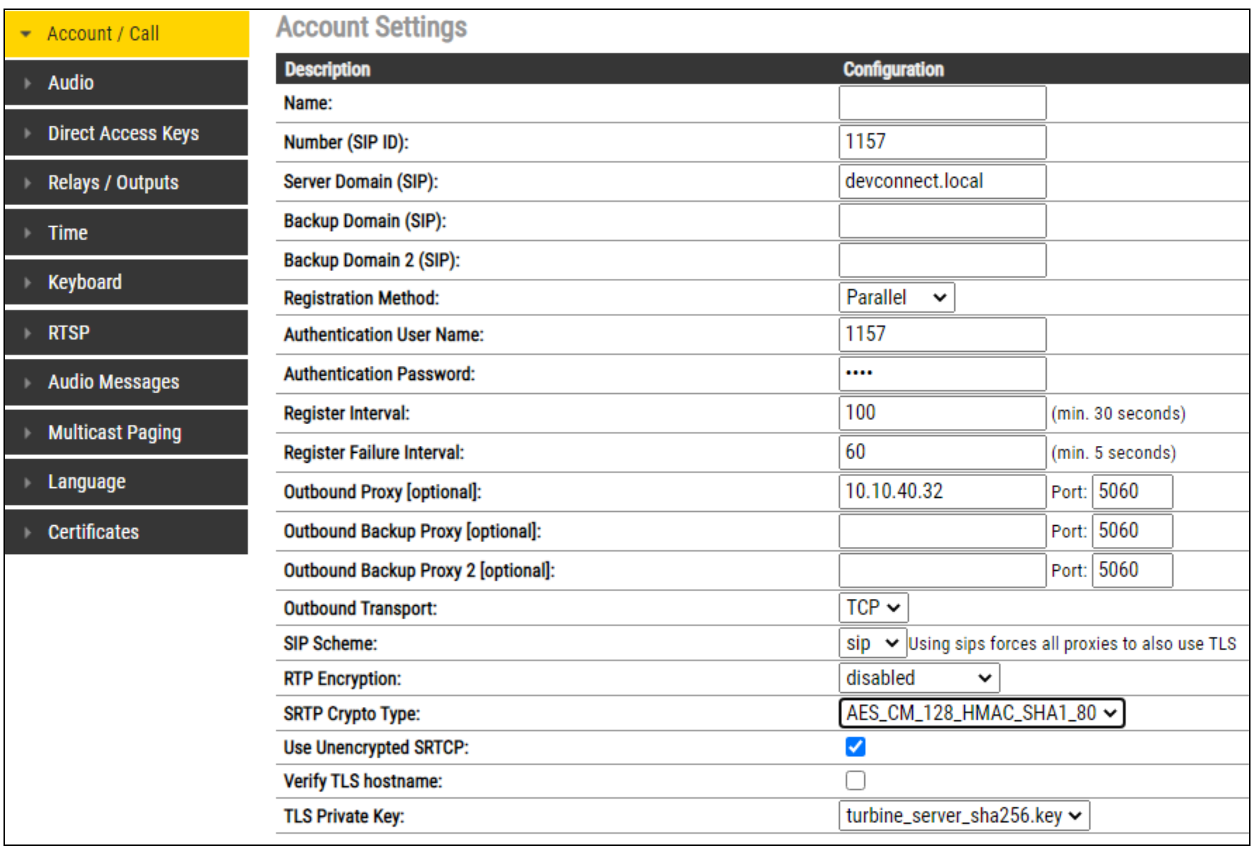


7.2. SIP Configuration

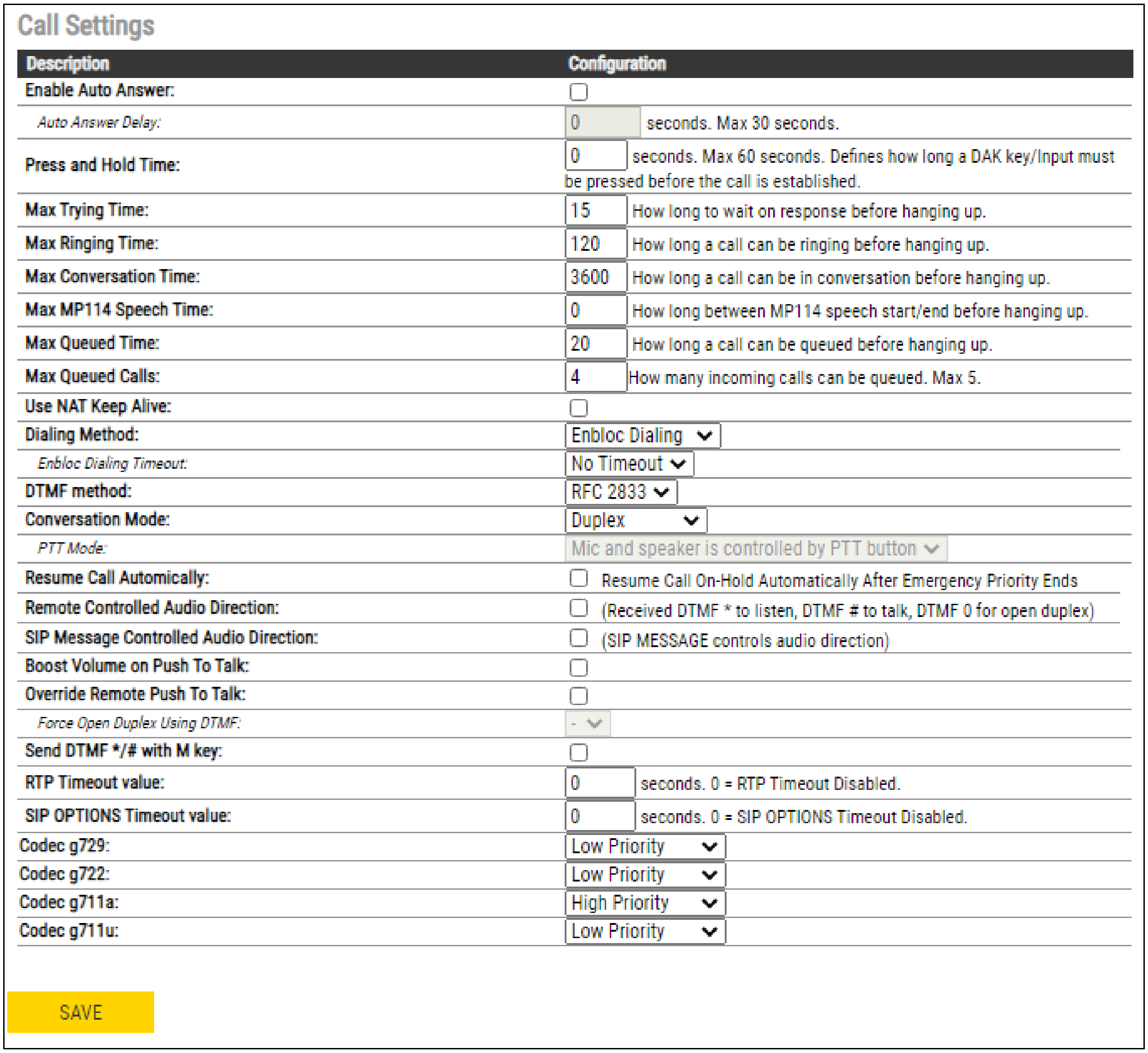
Click on **SIP Configuration → SIP** and configure the following in the **Account Settings** section:

- **Name**: Enter the desired name.
- **Number (SIP ID)**: Enter a user extension administered from **Section 6.2**.
- **Server Domain (SIP)**: Enter the Domain as per **Section 6.1.1**.
- **Authentication User Name**: Enter a user extension administered from **Section 6.2**.
- **Authentication Password**: Enter the **Login Code** from **Section 6.2**.
- **Outbound Proxy (optional)**: Enter the IP address of Session Manager and **5060** as the **Port**.
- **Outbound Transport:** Set this to **TCP**.
- **SIP Scheme:** Set this to **sip**.
- **RTP Encryption:** This is set to **disabled** for testing with RTP/TCP.
- **TLS Private Key:** This does not apply and can be left at default.

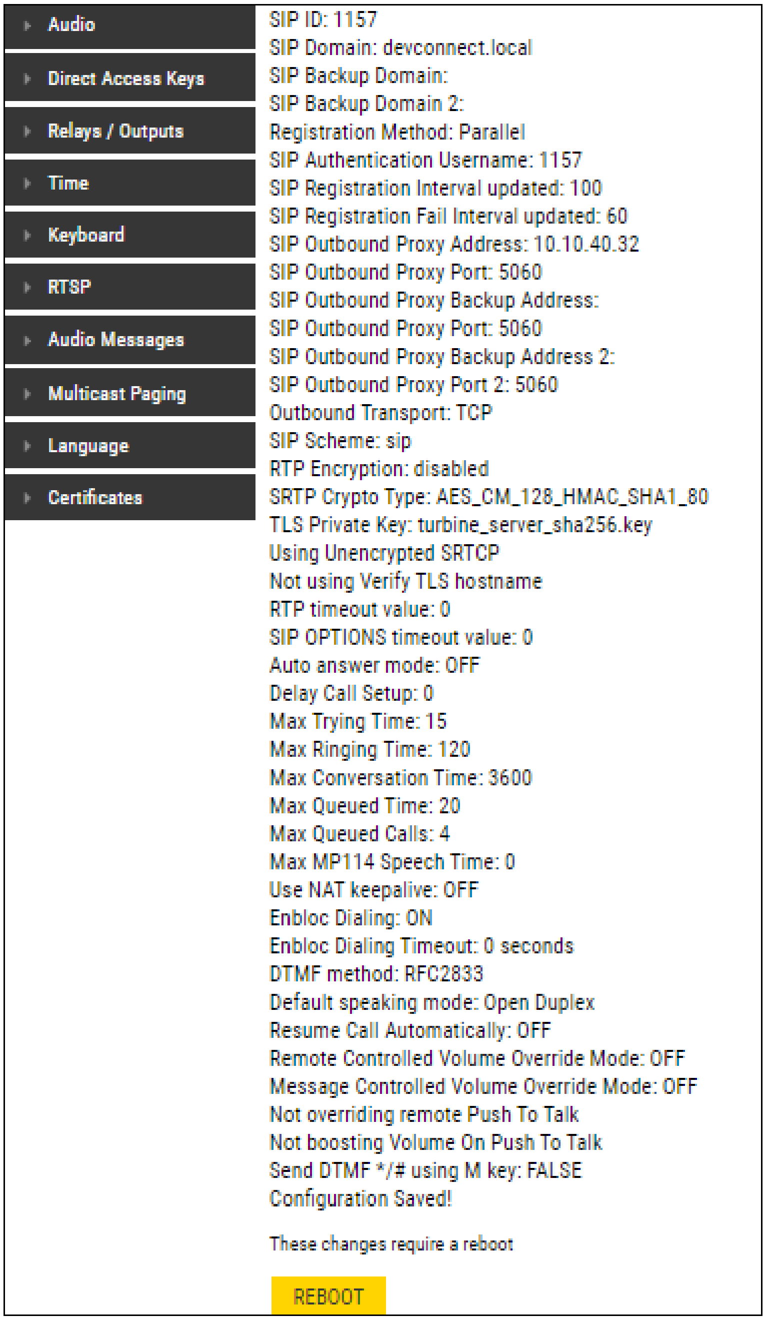
All other settings do not apply for TCP/RTP testing.



In the **Call Settings** section, configure as required the **DTMF method** as **RFC 2833** or whatever is set on Communication Manager. Configure other options as required. Click **SAVE** when done and a screen will appear (shown on the next page) to confirm the setting. The **Codec** is also set here, with **g711a** being set with the highest priority, as shown in the example below.

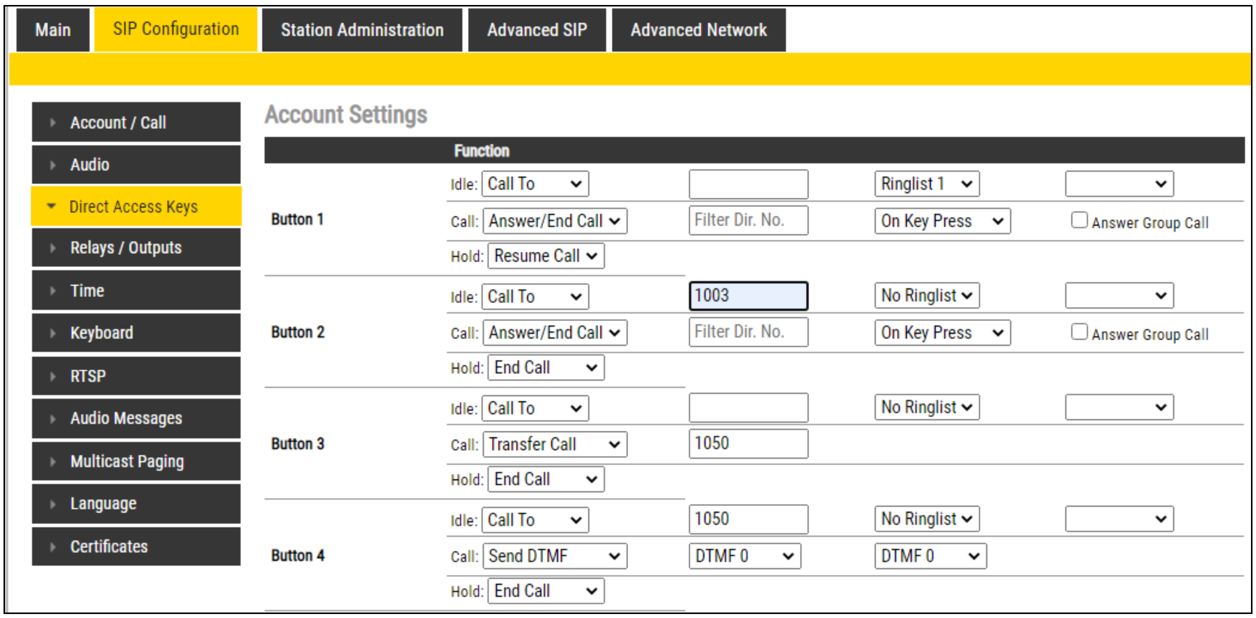


At this point the phone needs to be rebooted in order to save the SIP configuration, however this can be rebooted at a later stage should one wish to proceed with the configuration.



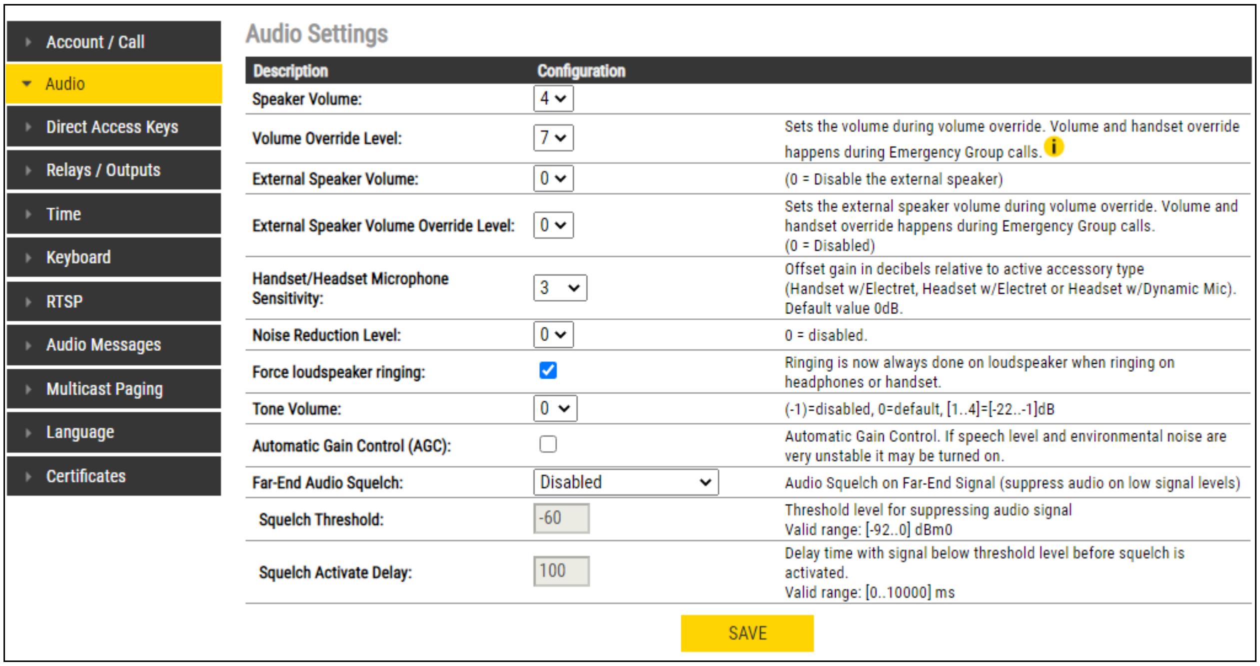
7.3. Configure Direct Access Keys

Click on the **Direct Access Keys** in the left window, this will bring up the functions as shown below where an extension to call can be assigned to the call button of the IP OR Master Intercom. This extension was an Avaya telephone, so when the button is pressed this telephone is called. **Button 1** is set to call **Ringlist 1** which is a list of Avaya phones to be called in sequence. **Button 2** is set to call a specific Avaya extension **1003**. In the **Idle** field, select **Call To** from the drop down and enter the extension to be called when the button key is pushed. In the **Call** field, select **Answer/End Call** and **On Key Press**. The buttons can be changed to use Hold or Transfer and other call features should they be required, Buttons 3 and 4 are examples of such.

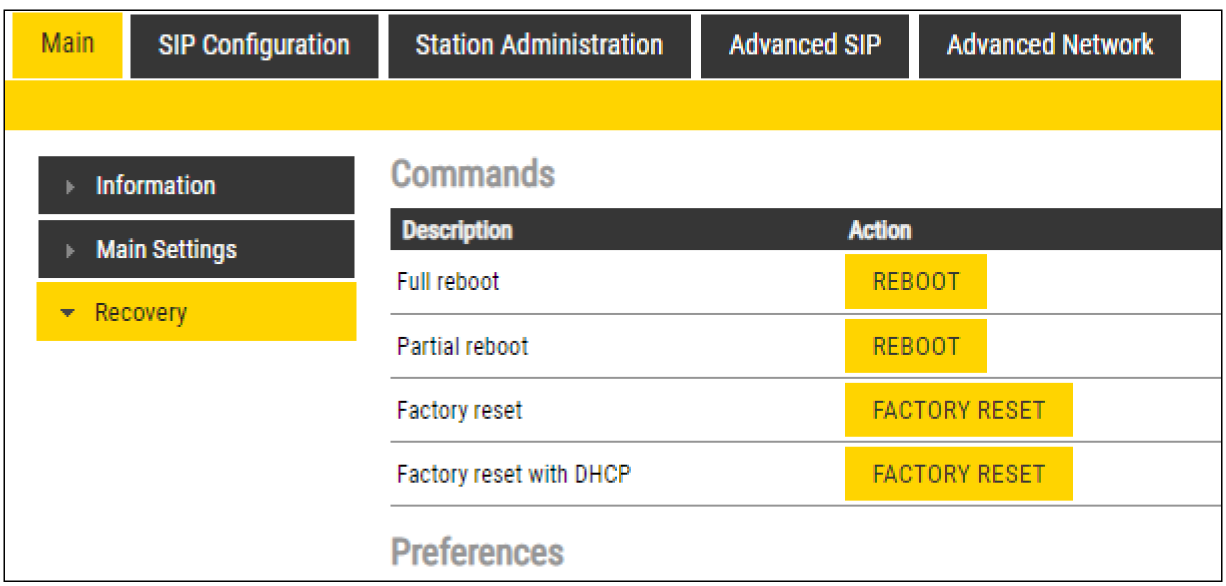


7.4. Configure Audio

Click on **Audio** in the left window, the volume of the speaker can be changed here.



If the phone was not rebooted earlier during the SIP configuration then click the **Main** tab and then click on **Recovery** as shown below. The telephone can be rebooted from this page.

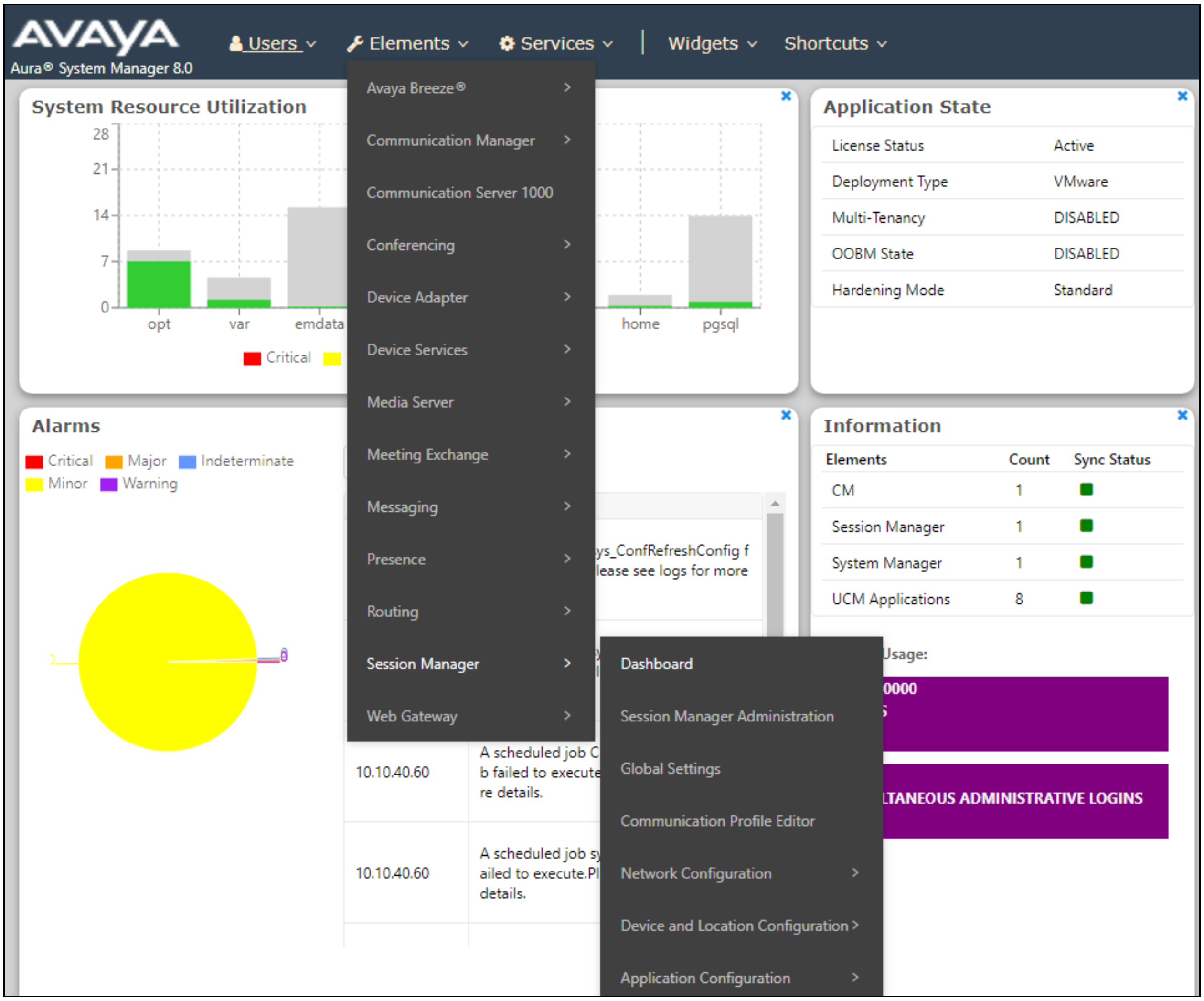


8. Verification Steps

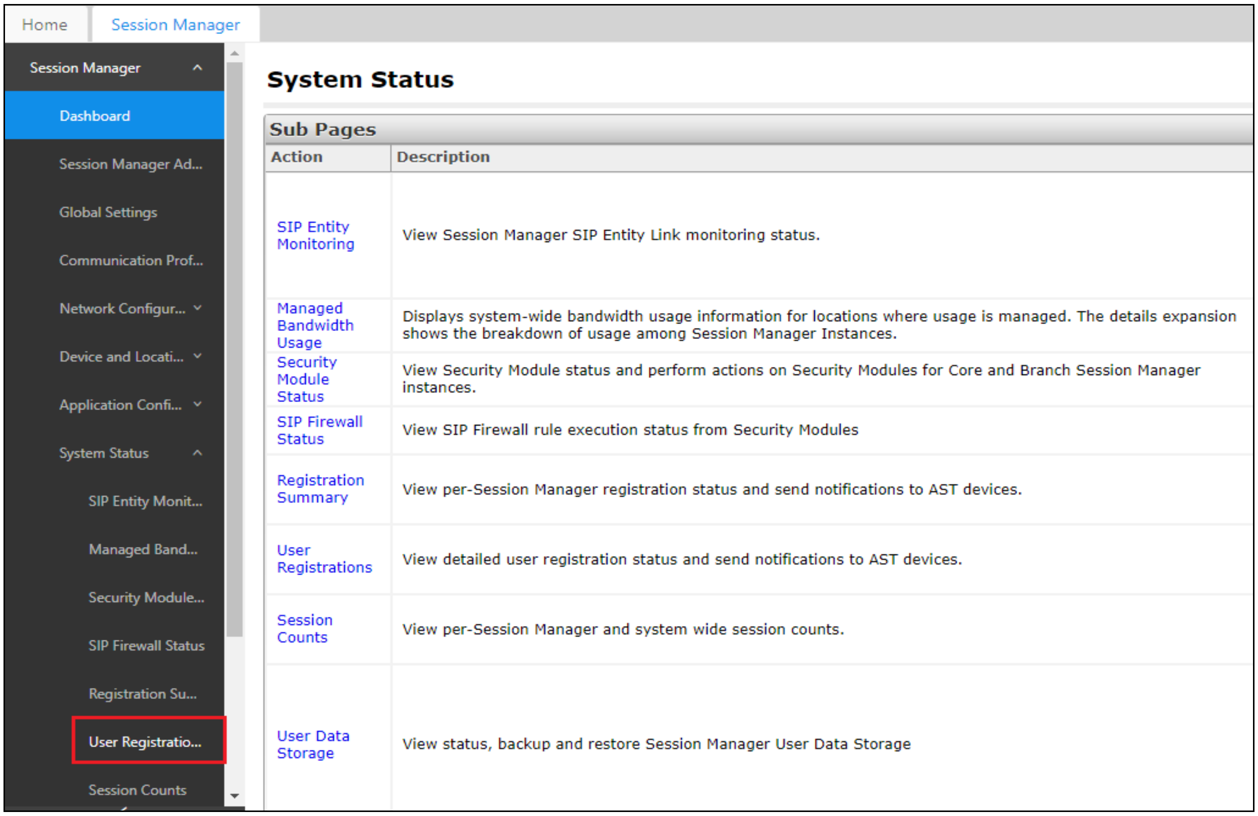
The following steps can be taken to ensure that connections between IP OR Master phones and Session Manager are up.

8.1. Session Manager Registration

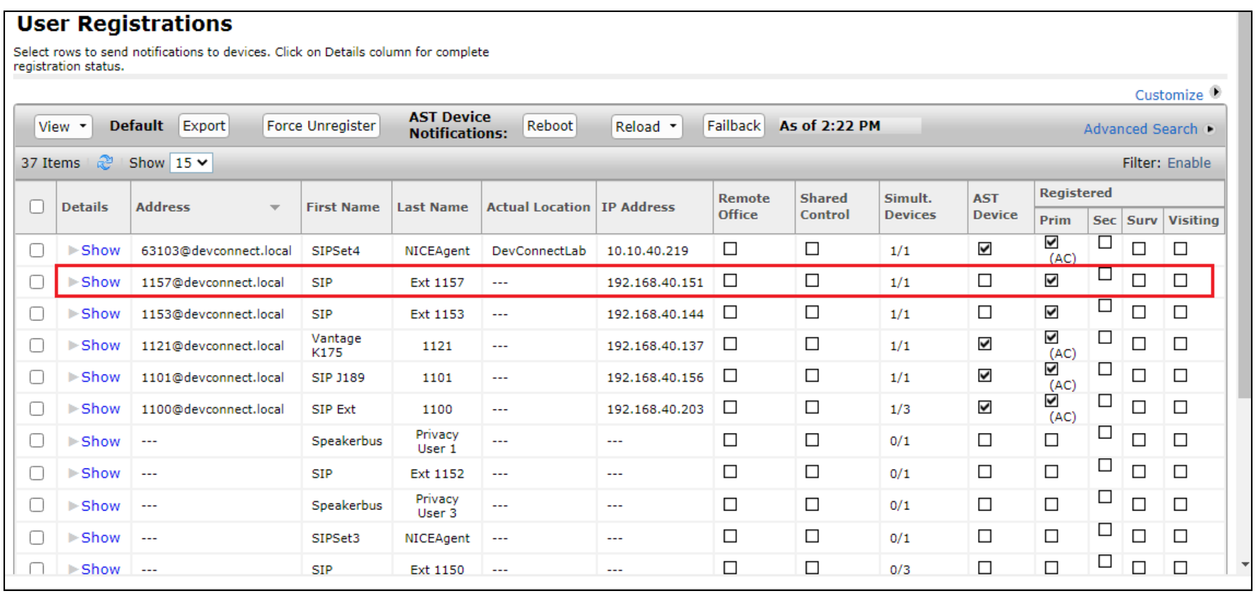
Log into System Manager as done previously in **Section 6**. Navigate to **Session Manager → Dashboard**.



Under **System Status** in the left window, select **User Registrations** to display all the SIP users that are currently registered with Session Manager.



The user(s) should show as being registered as seen below.



8.2. Zenitel IP OR Master Registration

To verify that the IP OR Master phone is registered correctly, log into the phone as per **Section 7**, the home page will show the following information where it can be seen if the phone is **Registered**.

IP Master Information

Description	Information
IP Address:	192.168.40.151
Subnet Mask:	255.255.255.0
Default Gateway:	192.168.40.1
DNS Server 1:	192.168.40.1
DNS Server 2:	
MAC Address:	00:13:cb:30:01:b6
Software Version:	6.1.1.0
More Information:	Show/Hide

Status

Description	Status
Mode:	SIP
Uptime:	up 59 minutes
Name:	
Number (SIP ID):	1157
Server Domain (SIP):	devconnect.local, Registered - Tue Apr 20 14:40:46 2021
Backup Domain (SIP):	
Backup Domain 2 (SIP):	
Outbound Proxy:	10.10.40.32:5061

9. Conclusion

These Application Notes describe the configuration steps required for Zenitel IP Operating Room Master to successfully interoperate with Avaya Aura® Communication Manager R8.1 and Avaya Aura® Session Manager R8.1 over TCP by registering the IP Operating Room Master phones with Session Manager as third-party SIP phones. Please refer to **Section 2.2** for test results and observations.

10. Additional References

This section references the product documentation relevant to these Application Notes. Product documentation for Avaya products may be found at http://support.avaya.com.

1. *Avaya Aura® Communication Manager Feature Description and Implementation*, Release 8.1.x
2. *Administering Avaya Aura® Session Manager*, Release 8.1.x

The Zenitel IP Operating Room Master documentation can be found by contacting Zenitel at http://www.zenitel.com.

Appendix

Signaling Group

```
display signaling-group 1                                       Page   1 of   3
                                SIGNALING GROUP

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? y                                   Enforce SIPS URI for SRTP? n
  Peer Detection Enabled? y  Peer Server: SM                         Clustered? n
 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: SM81vmpg
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                       Far-end Network Region: 1

Far-end Domain:
                                             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
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 6
```

Trunk Group Page 1

```
display trunk-group 1                                           Page   1 of   5
                                TRUNK GROUP

Group Number: 1                   Group Type: sip           CDR Reports: y
  Group Name: SIP PHONES                 COR: 1        TN: 1        TAC: *801
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n
                                             Member Assignment Method: auto
                                                      Signaling Group: 1
                                                    Number of Members: 10
```

Page 2

```
display trunk-group 1                                              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): 600

 Disconnect Supervision - In? y  Out? y


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





 Caller ID for Service Link Call to H.323 1xC: station-extension
```

Page 3

```
display trunk-group 1                                              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

                                                  Hold/Unhold Notifications? y
                                Modify Tandem Calling Number: no
               Send UCID? y



 Show ANSWERED BY on Display? y

 DSN Term? n
```

Page 4

```
display trunk-group 1                                               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
```

Page 5

```
trunk-group 1                                                       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: P-Asserted-Identity
            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
```

`

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.