Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Grandstream GXV3240 and GXV3275 Multimedia IP Phones for Android™ with Avaya Aura® Session Manager and Avaya Aura® Communication Manager - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate Grandstream GXV3240 and GXV3275 Multimedia IP Phones for Android™ with Avaya Aura® Session Manager and Avaya Aura® Communication Manager.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the 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 required to integrate Grandstream GXV3240 and GXV3275 Multimedia IP Phones for Android™ with Avaya Aura® Session Manager and Avaya Aura® Communication Manager.

2. General Test Approach and Test Results

The interoperability compliance testing included both feature and serviceability testing.

The featured testing focused on verifying the ability of the Grandstream Multimedia IP Phones to register as SIP endpoints with Avaya Aura® Session Manager, establish voice and point-to-point video calls, and exercise various telephony features (e.g. hold/resume, transfer, conference, etc.).

The serviceability testing focused on verifying the ability of the Grandstream Multimedia IP Phones to handle various outages such as network disconnects and server/phone reboots.

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.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Successful registration of the Grandstream Multimedia IP Phones with Session Manager.
- Voice calls between Grandstream Multimedia IP Phones, Avaya 6200 Series analog telephone, Avaya 2400 Series digital telephone, Avaya 1600 Series IP Deskphones, Avaya 9600 Series IP Deskphones (96*x*1 models, both H.323 and SIP), Avaya E129 SIP Deskphones, Avaya E169 Media Station, Avaya one-X® Communicator (both H.323 and SIP), and Avaya Communicator (for Windows, Android, and iPad).
- Point-to-Point video calls between the Grandstream Multimedia IP Phones, Avaya one-X® Communicator (SIP and H.323), and Avaya Communicator (for Windows and iPad).
- G.711MU, G.711.A, G.729A, G.722-64k, and G.726A.32K codec support.
- Guest login/logout
- Caller ID and display updates.
- Direct IP-IP Media (i.e. media shuffling).
- Proper recognition of DTMF tones by navigating voicemail menus.
- Proper operation of voicemail with Message Waiting Indication.
- Telephony features including: Mute, Hold/Resume, Transfer, Conference, Music on Hold, Call Coverage Paths, Call Forwarding (Unconditional, Busy, and No Answer), Call Park/Answer Back, Call Pickup, and Automatic Redial.

The serviceability testing focused on verifying the ability of the Grandstream phones to handle various outages such as network disconnects and server/phone reboots.

The wireless functionality of the Grandstream phones was not tested.

2.2. Test Results

The Grandstream Multimedia IP Phones successfully passed compliance testing with the following observations:

- **G.726-32K**: G.726-32K codec negotiation failed during compliance testing; however, Grandstream has delivered a fix for this issue in their system firmware version 1.0.3.25.
- **Conference URI / display updates**: When the conference is established, the endpoints involved in the conference receive a SIP UPDATE message with a Contact header containing the "isfocus" feature tag. This feature tag indicates that the URI in the Contact header field is a conference URI. Avaya phones update their display to show the conference URI display information (e.g. Conference 2) which indicates the number of other parties in the conference. The Grandstream phones did not update their display to show the conference URI display information; however, Grandstream has delivered a fix for this issue in their system firmware version 1.0.3.25.
- **Long Hold Recall**: The Grandstream phones do not audibly or visually alert the user of a held call when the Avaya Aura® Communication Manager long hold recall timer expires.
- **481 Call Leg/Transaction Does Not Exist**: Multiple "481 Call Leg/Transaction Does Not Exist" SIP messages are generated for transfer/conference scenarios. This is essentially a race condition. For example, after the REFER for a transfer is sent, both parties send a BYE for the call leg going away. When each party receives the BYE, it responds with a 481 Call Leg/Transaction Does Not Exist (since each party has already sent its own BYE for that call leg).

2.3. Support

Grandstream can be reached using the information provided on the following web sites:

- **General Contact Information**: http://www.grandstream.com/company/contact-us
- **Support Requests:** http://esupport.grandstream.com/support/customerportal/login.php

3. Reference Configuration

Figure 1 illustrates the test configuration used to verify Grandstream Multimedia IP Phones integration with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. The configuration consists of an Avaya Aura® Communication Manager Server with an Avaya G450 Media Gateway providing connectivity to the PSTN via an ISDN-PRI trunk, Avaya Aura® System Manager, and Avaya Aura® Session Manager. Avaya Aura® Messaging was used as the voicemail system. The Grandstream endpoints registered with Session Manager as SIP Users.

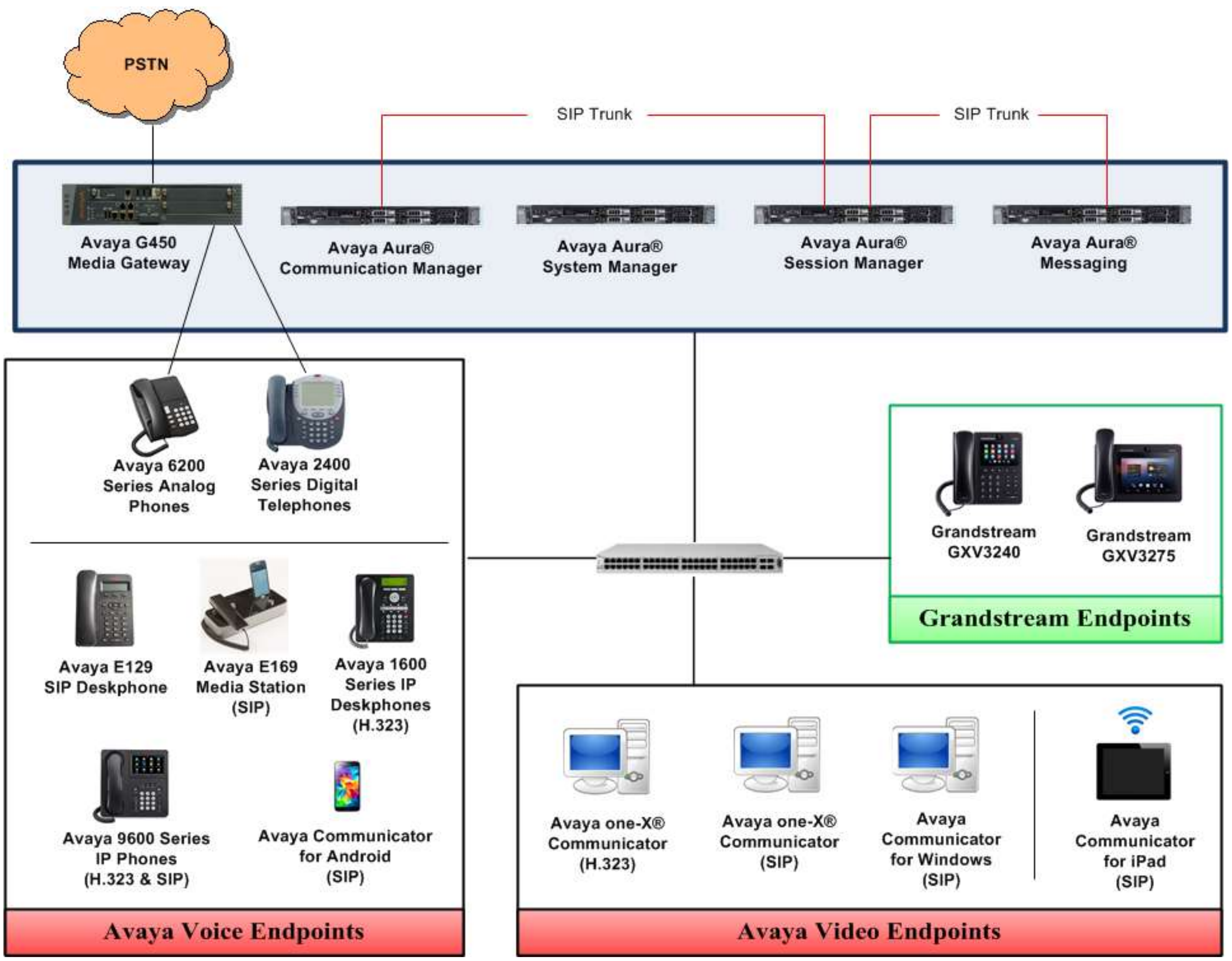


Figure 1: Grandstream Multimedia IP Phones

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager with an Avaya G450 Media Gateway	R6.3 Update 22098
Avaya Aura® System Manager	R6.3.11
Avaya Aura® Session Manager	R6.3.11
Avaya Aura® Messaging	R6.3
Avaya 6200 Series Analog Phones	-
Avaya 2400 Series Digital Telephones	R6
Avaya 1600 Series IP Deskphones	1.3.6
Avaya E129 SIP Deskphones	1.0
Avaya E169 Media Station	1.1
Avaya 9600 Series IP Deskphone (96x1 – SIP and H.323)	6.4
Avaya one-X® Communicator (SIP and H.323)	6.2.5
Avaya Communicator (for Windows, Android, and iPad)	2.1 (Windows and Android) 2.0.2 (iPad)
Grandstream GXV3240 and GXV3275 Multimedia IP Phones	Hardware Revision: V1.7A (GXV3240) V1.4B (GXV3275) System Version: 1.0.3.15 Android Version: 4.2.2

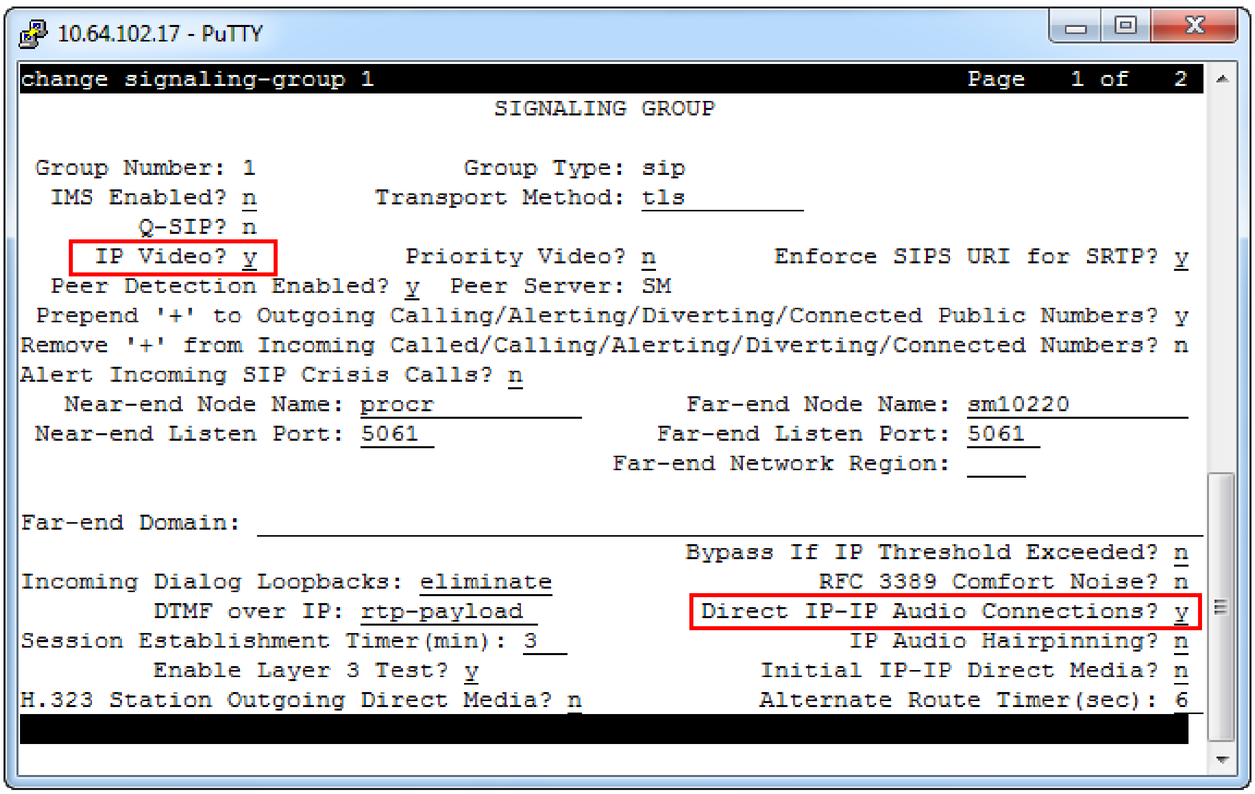
5. Configure Avaya Aura® Communication Manager

These Application Notes assume that basic Communication Manager administration has already been performed, including the configuration needed to establish a SIP trunk to Session Manager. Consult Reference **[1]** for further details if necessary.

All configuration in this section is administered using the System Access Terminal (SAT).

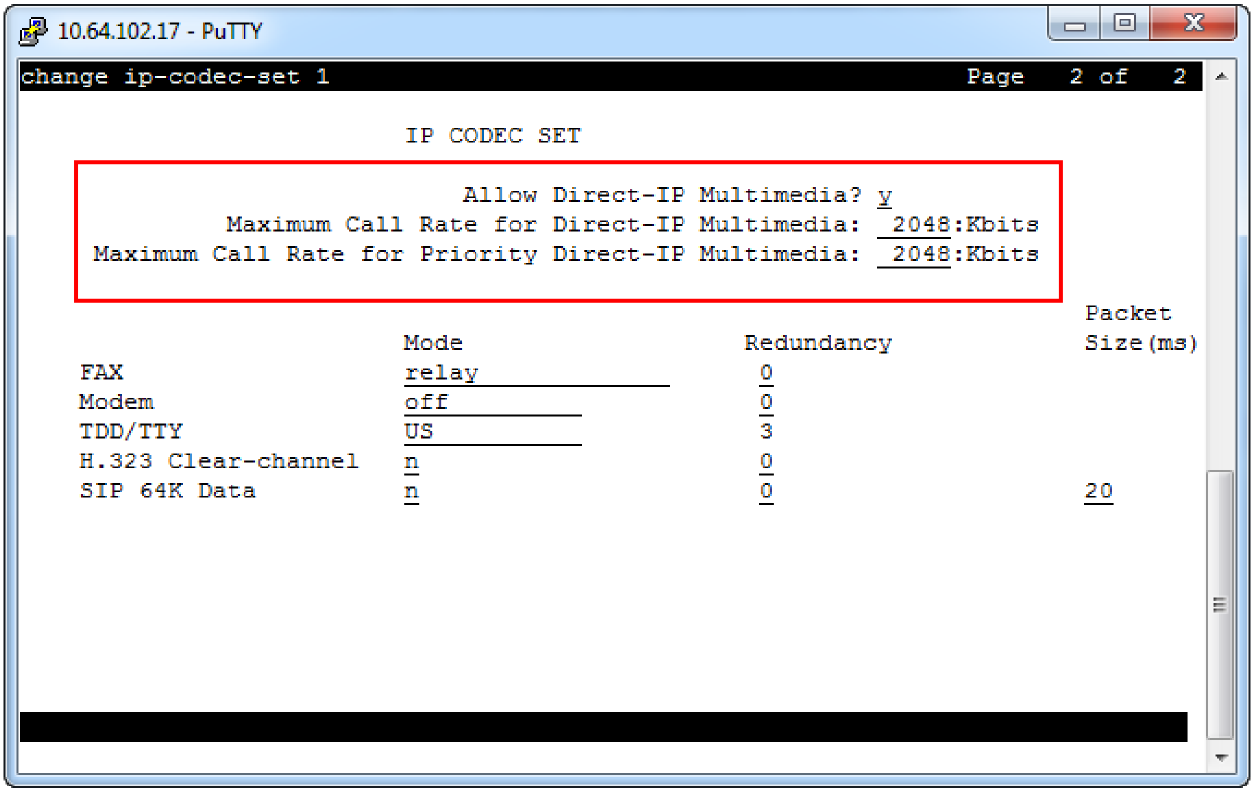
5.1. Configure Video Parameters

Enable IP Video on the signaling group for the SIP trunk between Communication Manager and Session Manager. Enter the "***change signaling-group***" command. Set **IP Video** to **y**. By default, **Direct IP-IP Audio Connections** is enabled for audio media shuffling.



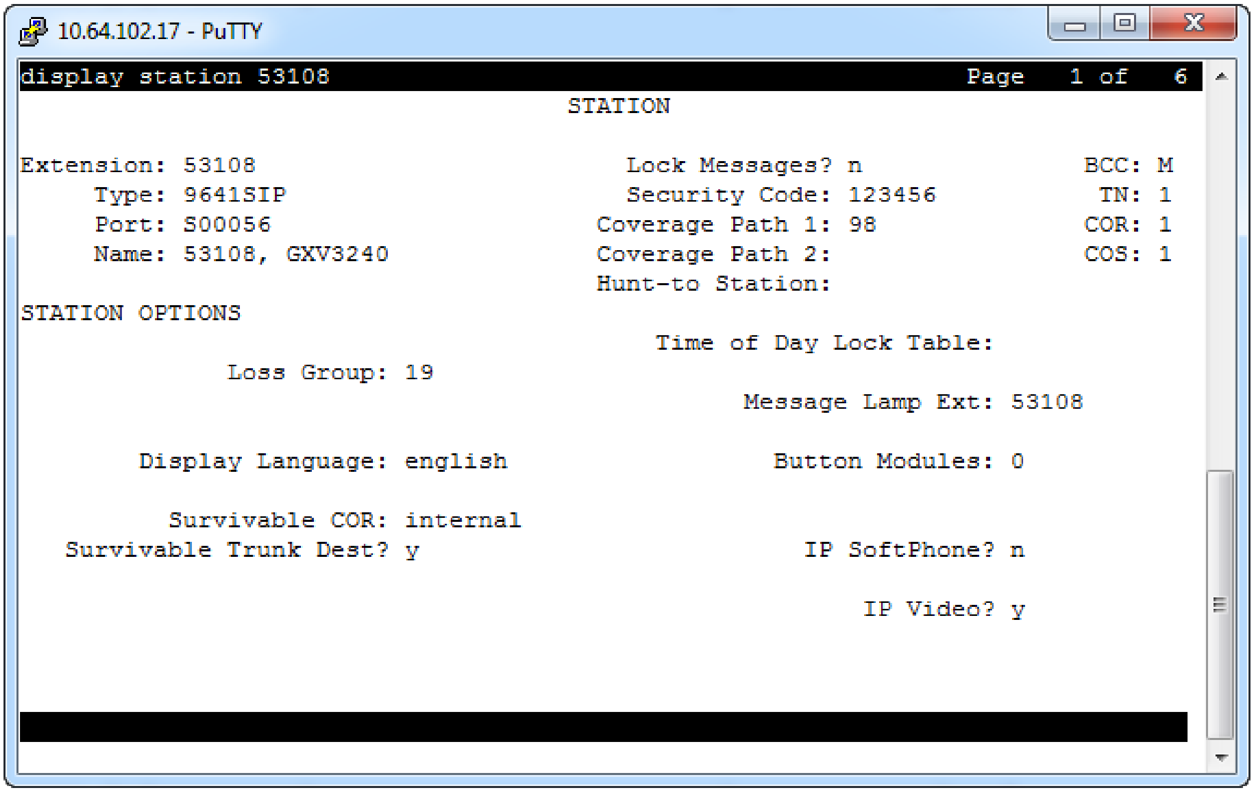
```
10.64.102.17 - PuTTY
change signaling-group 1                                        Page   1 of   2
                                SIGNALING GROUP

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? y          Priority Video? n        Enforce SIPS URI for SRTP? y
  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: sm10220
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region:


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
```

Set the multimedia parameters for the ip-codec-set used by the endpoints. Enter the "***change ip-codec-set***" command. Set **Allow Direct-IP Multimedia** to **y**. Set the two call rate parameters to desired values (**2048** was used during compliance testing).



5.2. Verify Station Configuration

It is assumed all Avaya endpoints have already been configured. The stations used for the Grandstream endpoints do not need to be configured via the SAT. Rather, stations will be automatically created in Communication Manager when the SIP Users are added in **Section 6.1**. After completing the steps in **Section 6.1** to create a SIP User, enter the "***display station***" command within the SAT to verify that station was successfully added to Communication Manager.

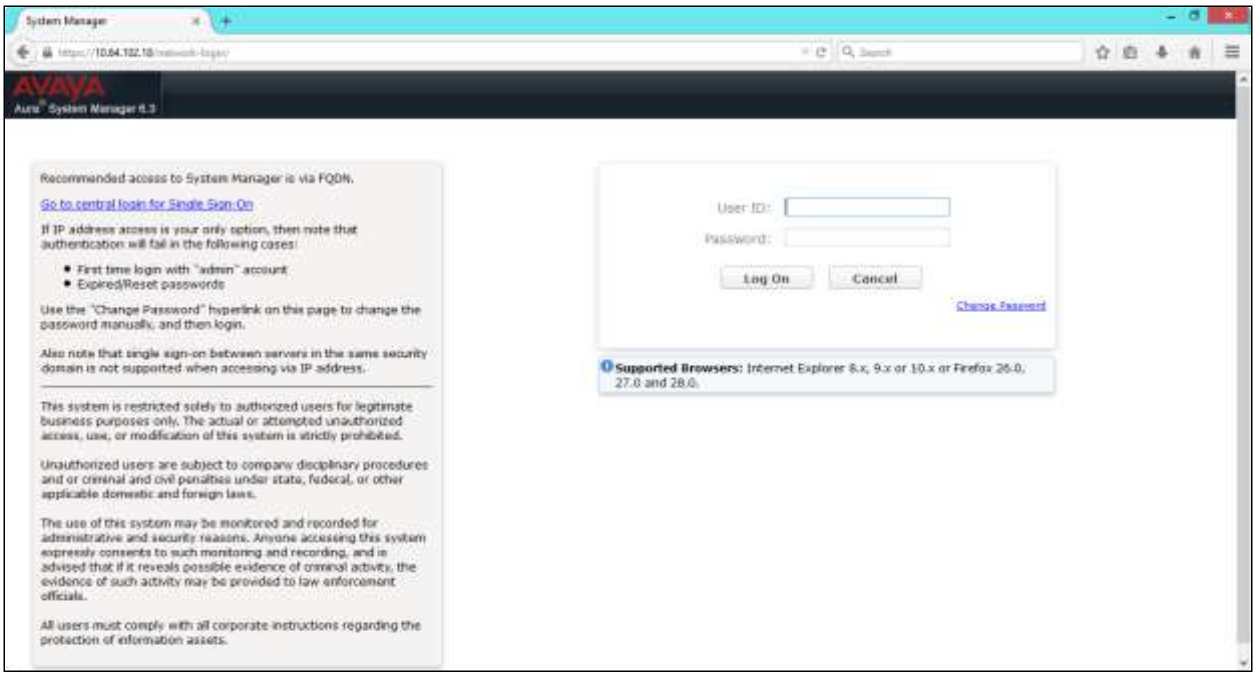


```
10.64.102.17 - PuTTY
display station 53108                                           Page   1 of   6
                                     STATION

Extension: 53108                          Lock Messages? n              BCC: M
     Type: 9641SIP                        Security Code: 123456          TN: 1
     Port: S00056                       Coverage Path 1: 98             COR: 1
     Name: 53108, GXV3240               Coverage Path 2:                COS: 1
                                        Hunt-to Station:
STATION OPTIONS
                                            Time of Day Lock Table:
              Loss Group: 19
                                                  Message Lamp Ext: 53108

        Display Language: english                   Button Modules: 0

          Survivable COR: internal
   Survivable Trunk Dest? y                           IP SoftPhone? n

                                                          IP Video? y
```

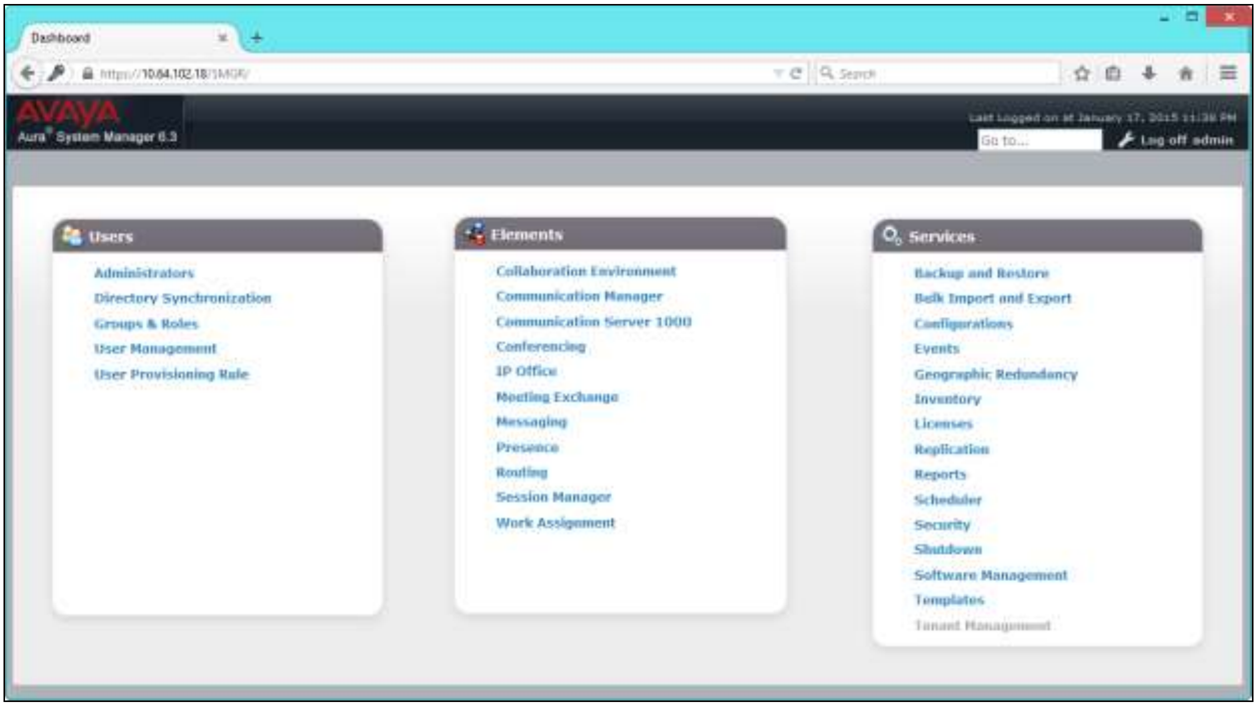
6. Configure Avaya Aura® Session Manager

Configuration of Avaya Aura® Session Manager is performed via Avaya Aura® System Manager. These Application Notes assume that basic System Manager and Session Manager administration has already been performed, including the configuration needed to establish a SIP trunk to Communication Manager. Consult Reference **[2]** for further details if necessary.

Access the System Manager Administration web interface by entering **https://<*ip-address*>/SMGR** as the URL in a web browser, where <*ip-address*> is the IP address of System Manager. Log in using appropriate credentials.



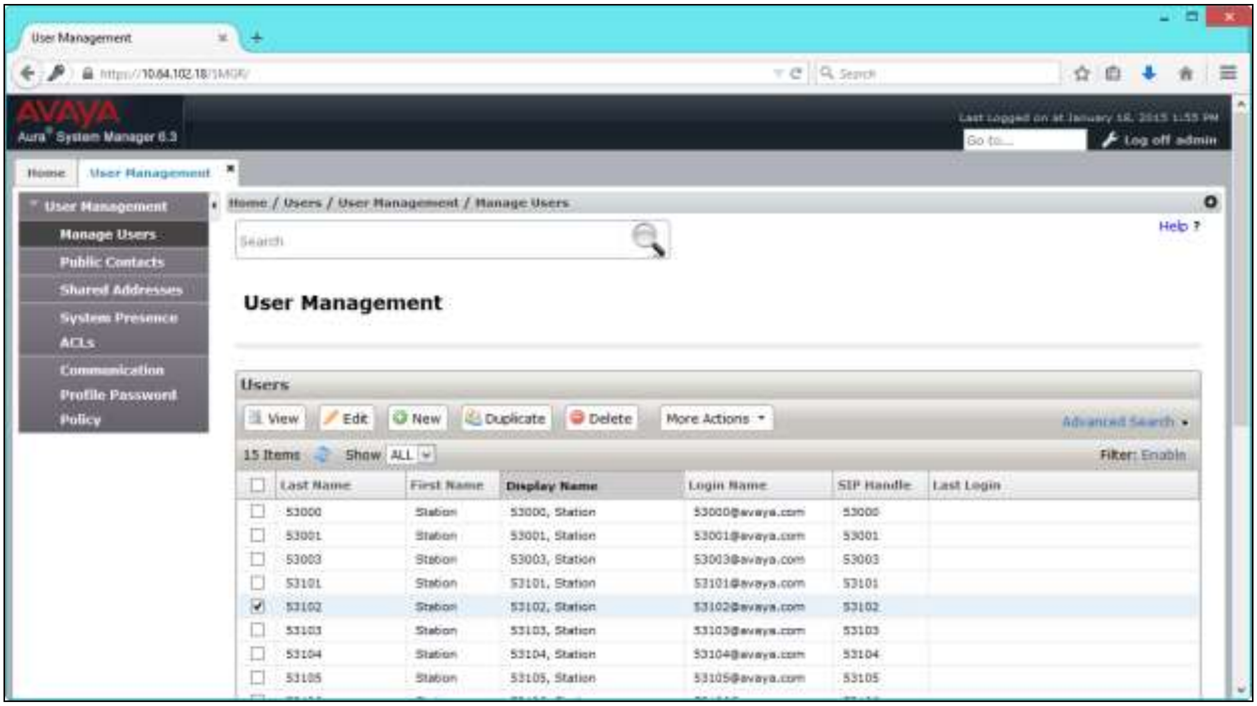
Once logged in, the following screen is displayed.



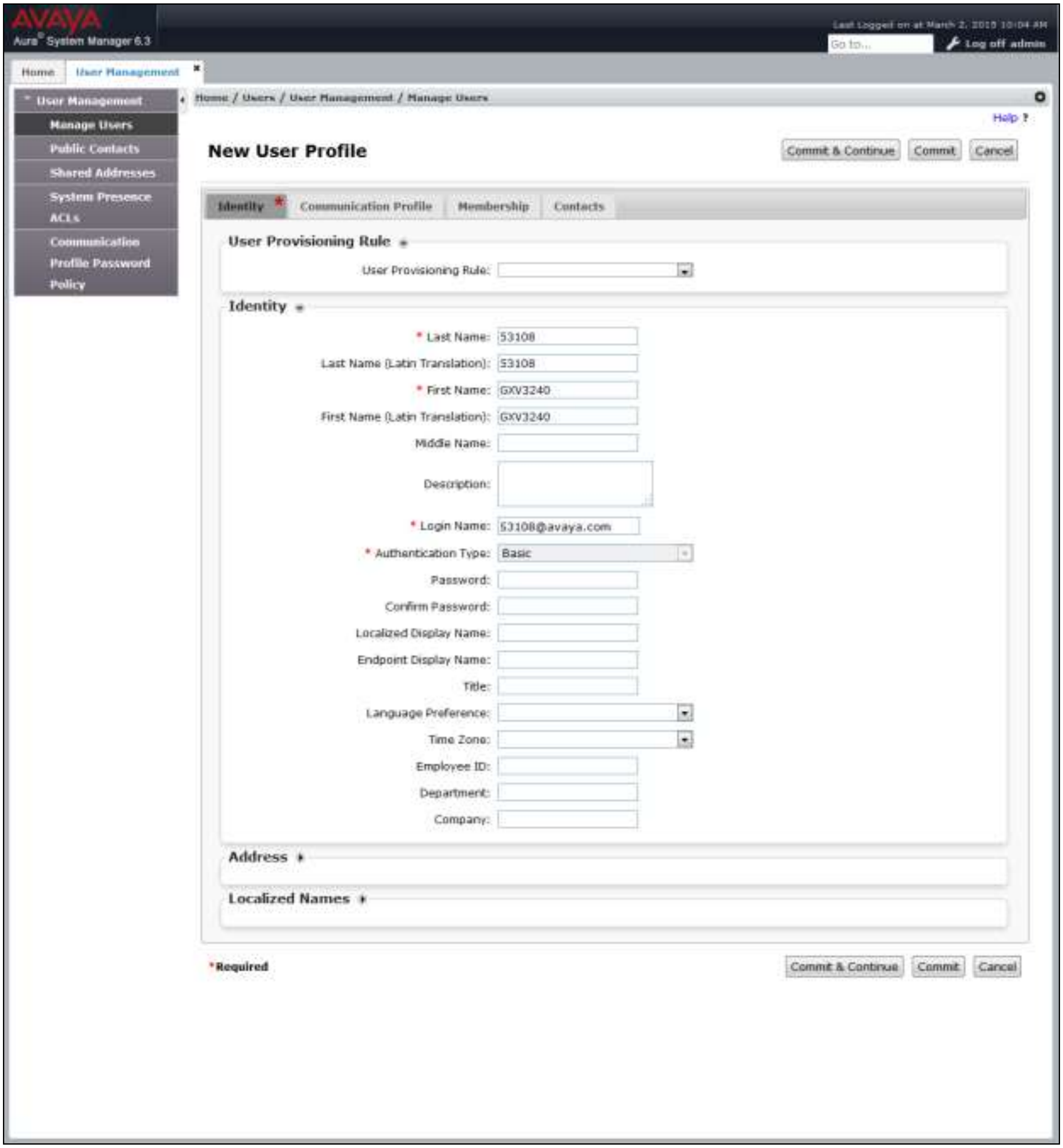
6.1. Configure Users

This document assumes all Users for Avaya SIP endpoints have already been provisioned in System Manager.

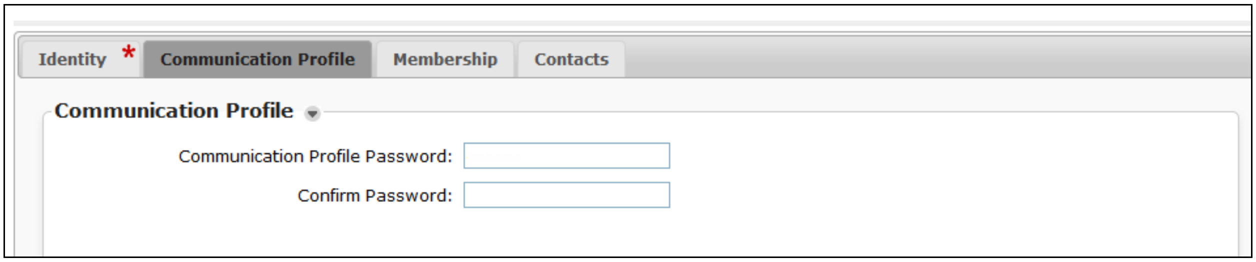
Add a User for each Grandstream phone. Navigate to **Home → Users → User Management → Manage Users**. Click the **New** button.



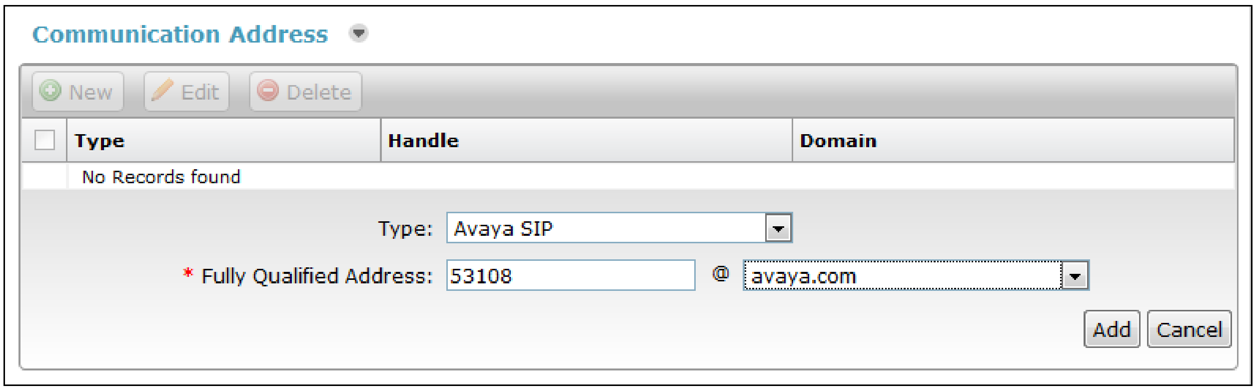
On the **Identity tab**, enter descriptive values for **Last Name** and **First Name**. Enter the **Login Name** (e.g. ***user@sipdomain***; this document assumes the SIP domain was previously configured. Consult Reference **[2]** for further details if necessary).



Click the **Communication Profile** tab. Enter the **Communication Profile Password**, and then enter it again in **Confirm Password**.



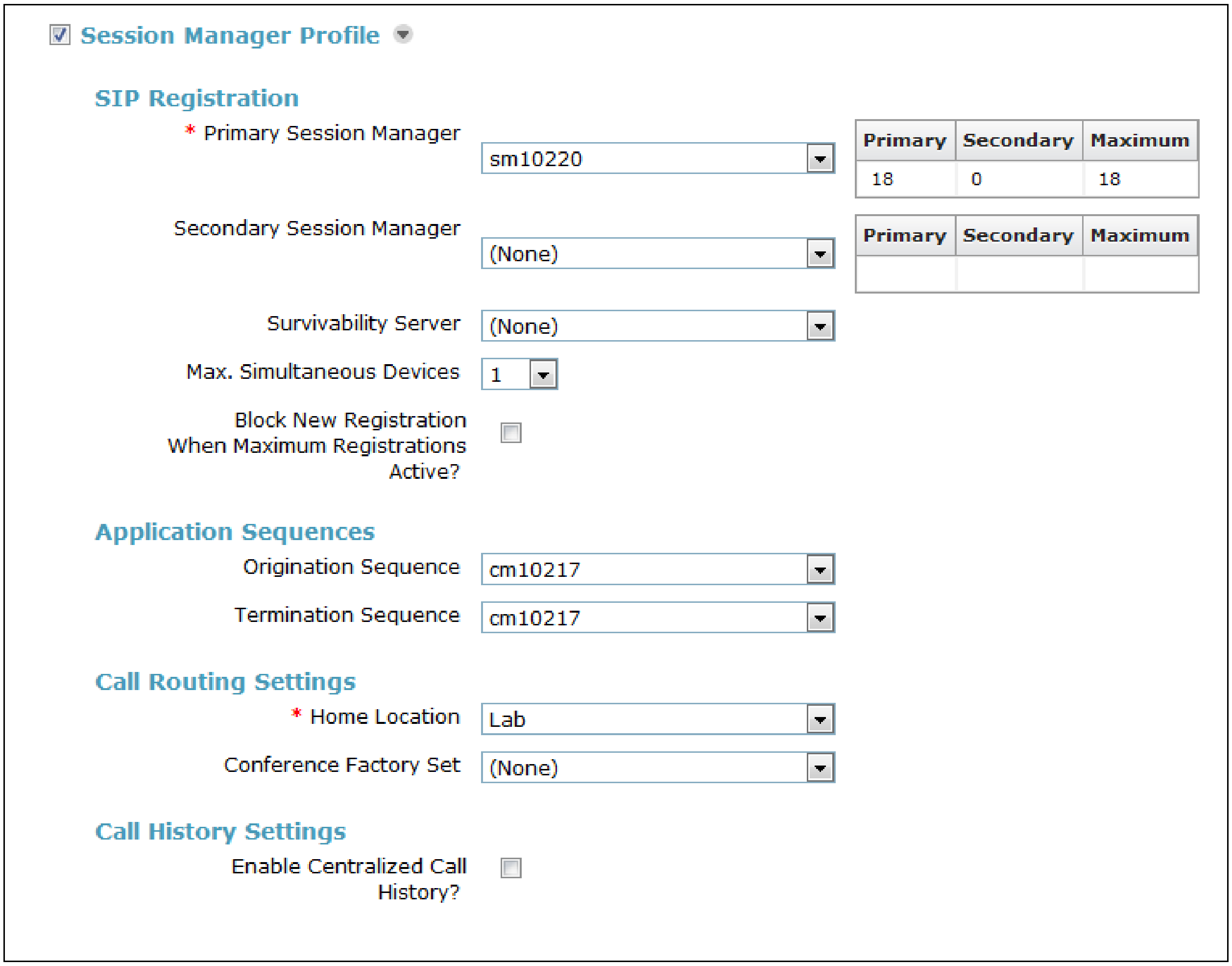
Click the **New** button under Communication Address. Select **Avaya SIP** for **Type**. Enter the **Full Qualified Address** for the user and then click the **Add** button.



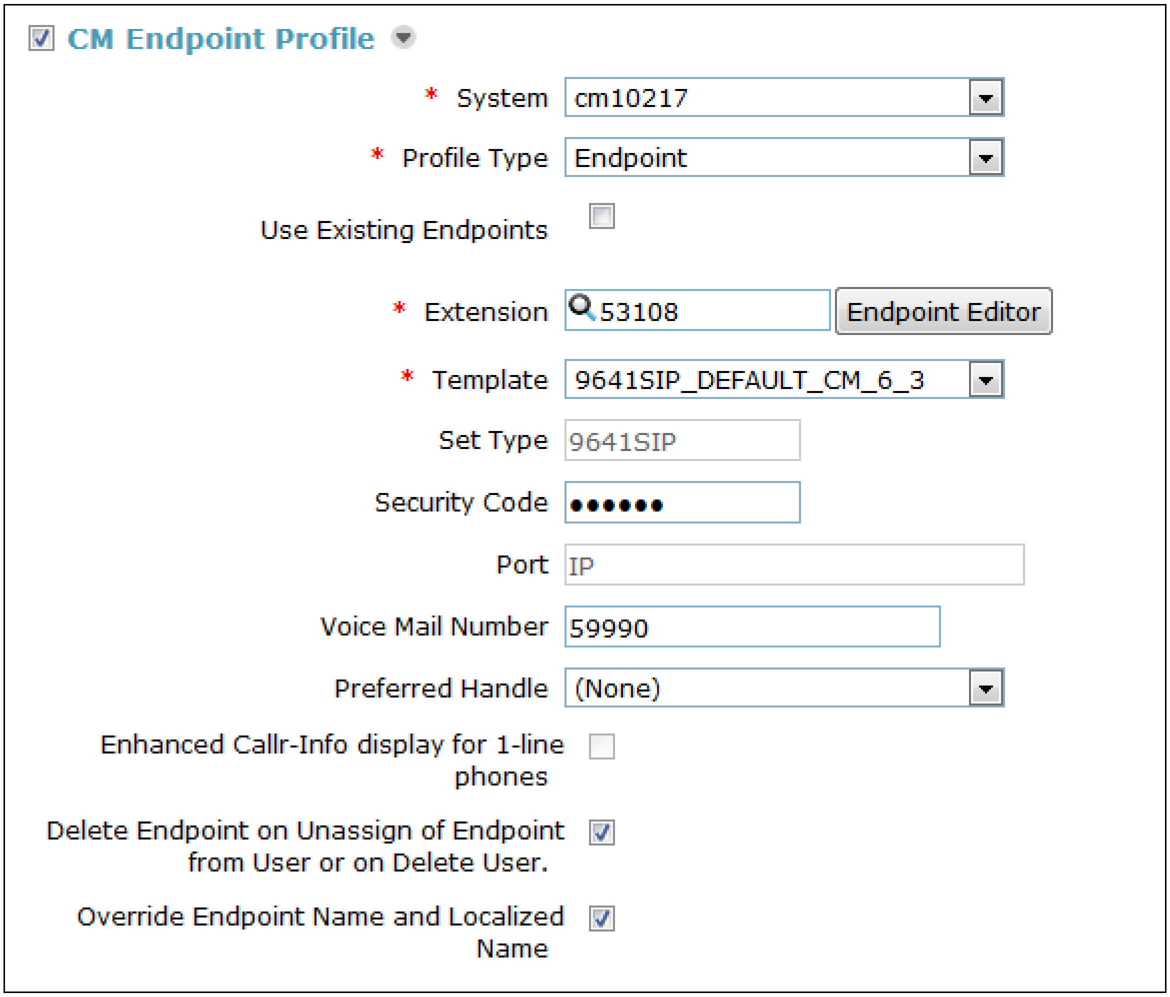
Check the box next to **Session Manager Profile**. For **Primary Session Manager**, select the Session Manager SIP Entity from the drop-down menu.

Under **Application Sequences**, select the sequence (e.g**. cm10217**) used to route calls to Communication Manager from the drop-down menu for the **Origination Sequence** and **Termination Sequence**.

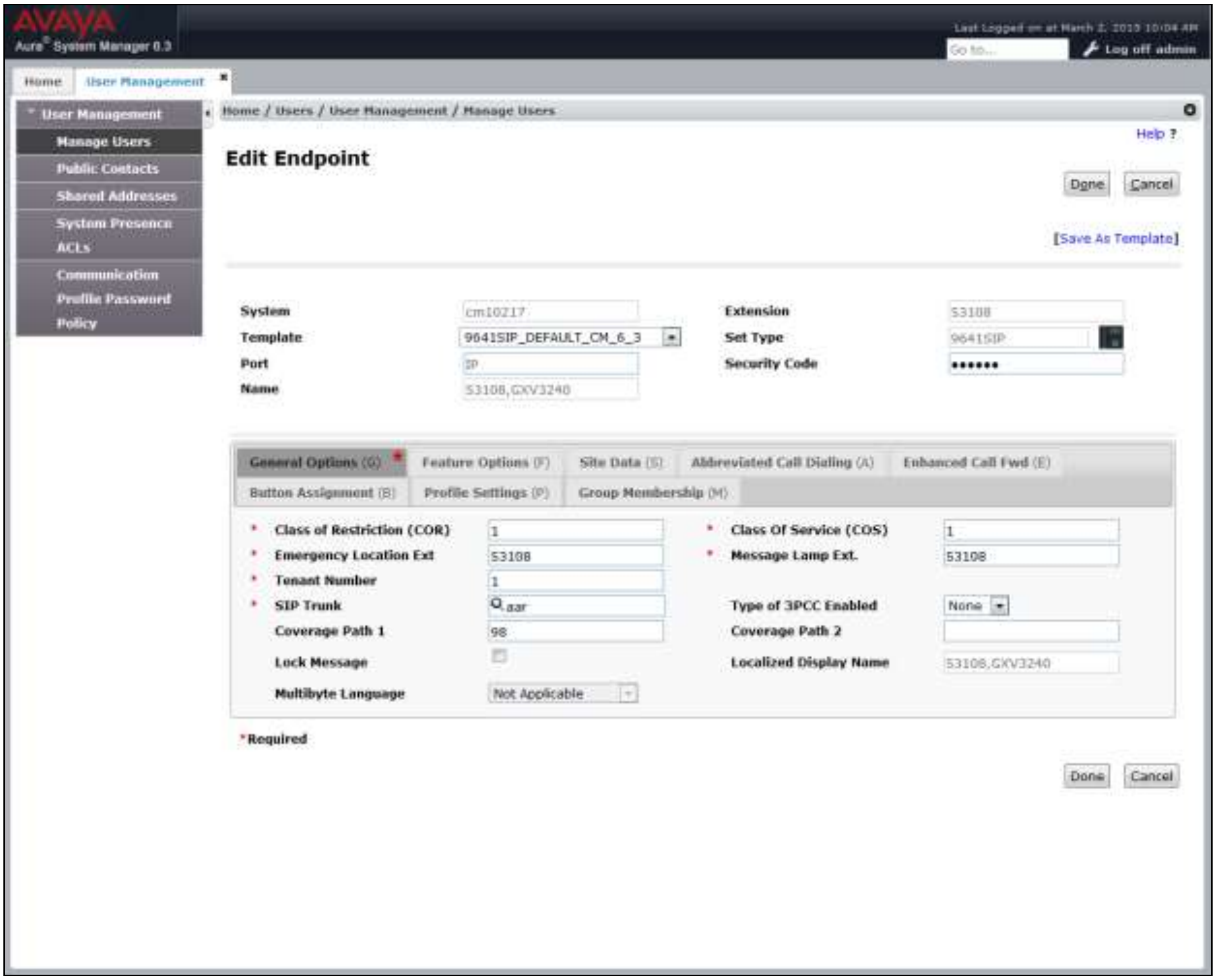
Under **Call Routing Settings**, select the **Home Location** (e.g. **Lab**) from the drop-down menu.



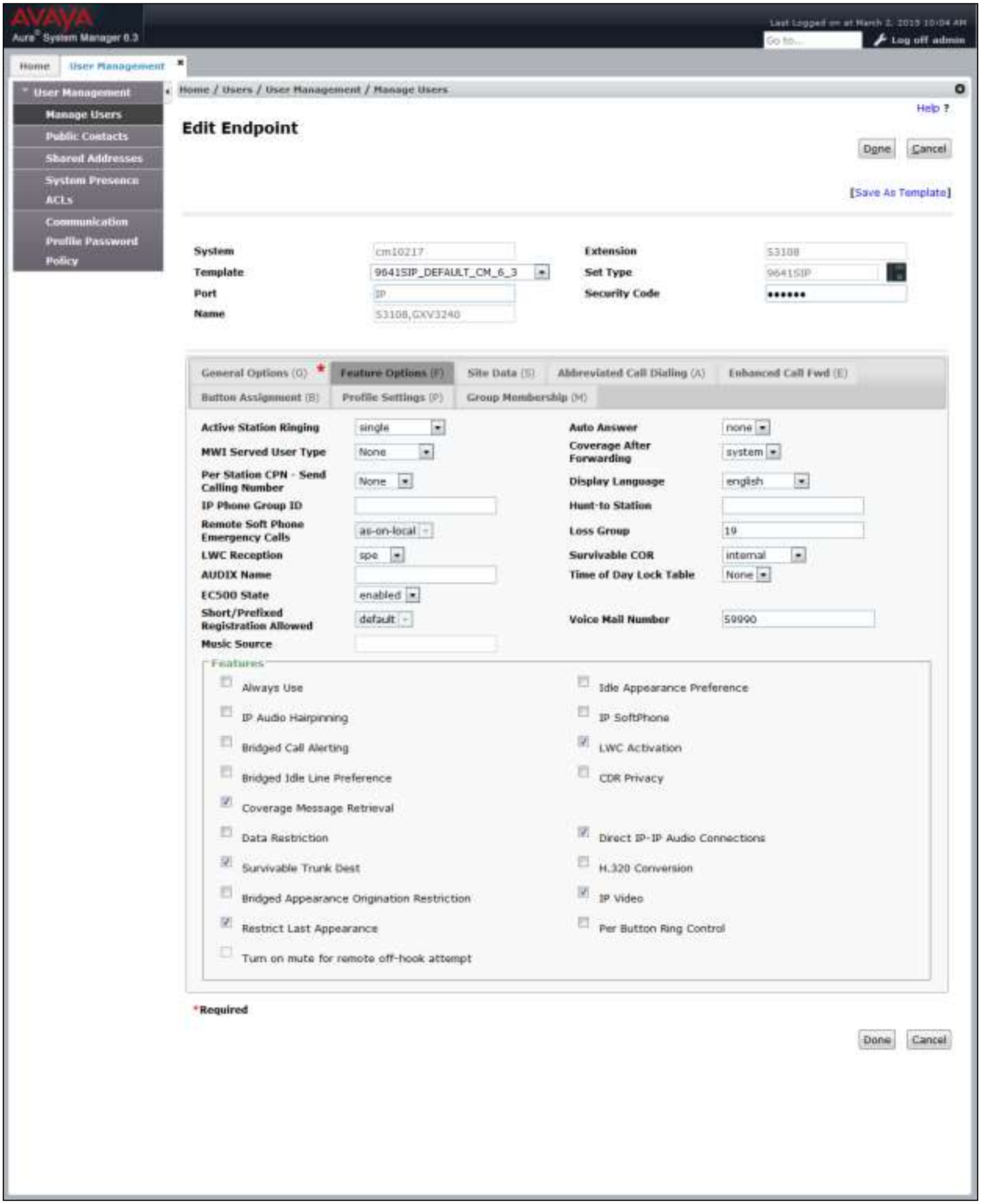
Check the box next to **CM Endpoint Profile**. Select the Communication Manager SIP Entity from the **System** drop-down menu. Select **Endpoint** for **Profile Type**. Enter the Communication Manager station **Extension** (e.g. **53108**). From the **Template** drop-down menu, select **9641SIP_DEFAULT_CM_6_3**. Enter the desired value for the station **Security Code**. For **Voice Mail Number**, enter the Avaya Aura® Messaging pilot number. Click the **Endpoint Editor** button next to **Extension** field.

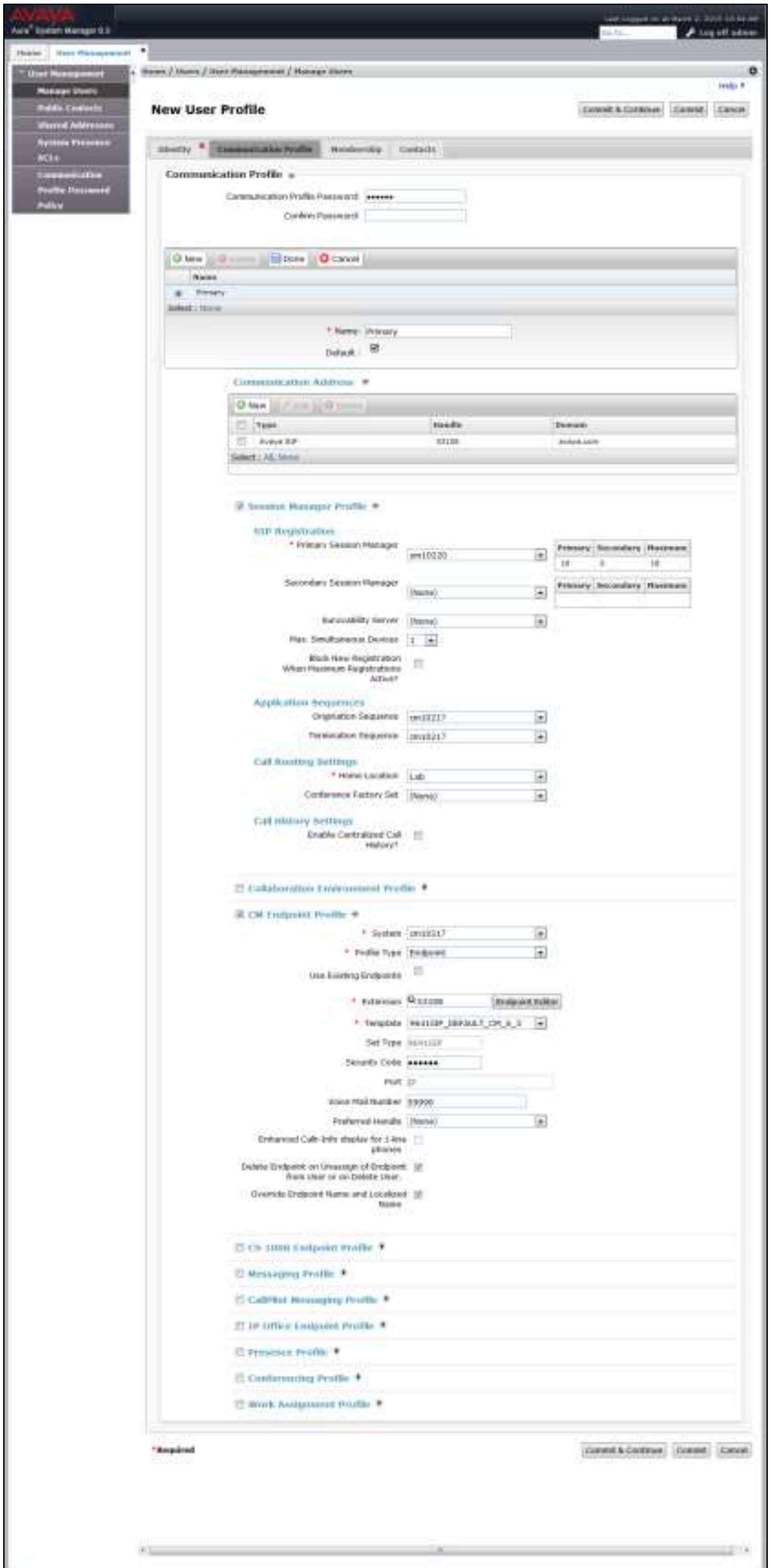


Under the **General Options** tab, set **Coverage Path 1** to the coverage path used for Avaya Aura® Messaging.



Click the **Feature Options** tab. Check the box for **IP Video** and then click the **Done** button to return to the user **Communication Profile** tab as shown on the next page. Click the **Commit** button.





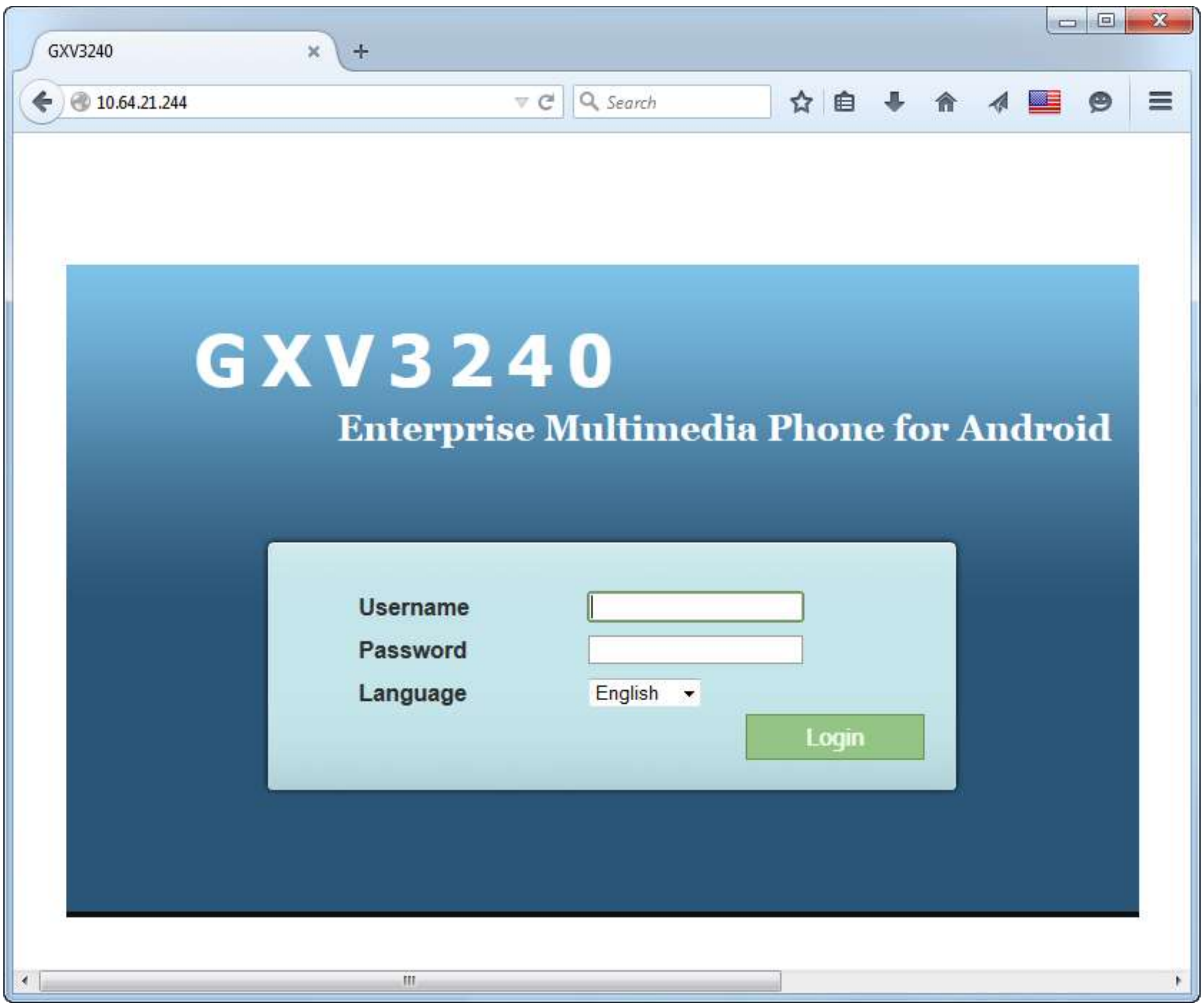
7. Configure Grandstream GXV3240 and GXV3275 Multimedia IP Phones for Android™

It is assumed that the basic configuration for the Grandstream phones has already been implemented and that the phones are ready for the integration with Avaya Aura® Session Manager. The sub-sections below provide only the steps required to configure the Grandstream to interoperate with Avaya.

Note: Only the configuration steps required to integrate the Grandstream GXV3240 phone model with Avaya are shown in the sub-sections below; however the same steps are required and applicable to the Grandstream GXV3275 phone model.

7.1. Web Interface

Access the phone web interface by opening a web browser and entering the following URL: **http://*ip-address***, where *ip-address* is the IP address of the phone. Log in using appropriate credentials.



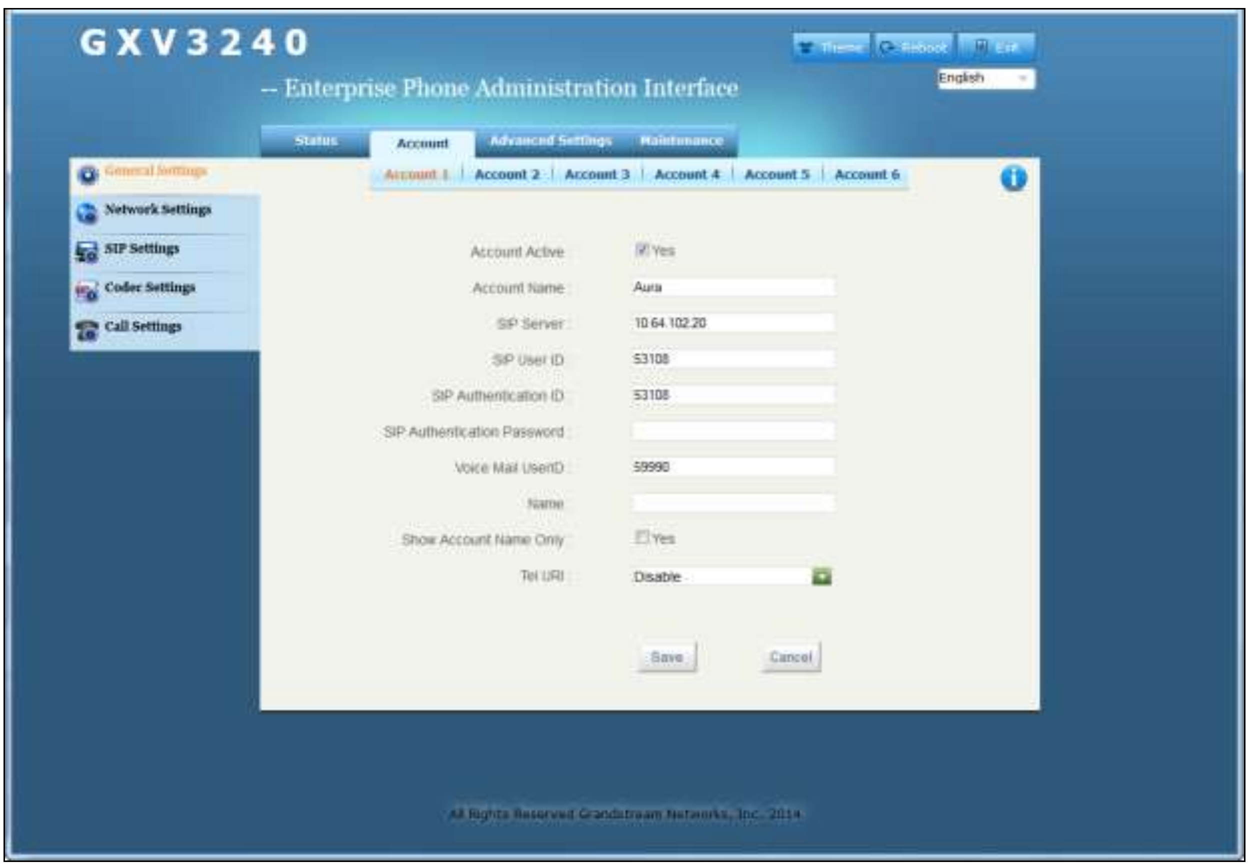
After making and saving any configuration changes throughout the remainder of **Section 7**, an **Apply** button will appear at the top of the web interface. Click the **Apply** button (not shown) to immediately apply the changes.

7.2. Configure General Settings

Click the **Account** tab and select an Account sub-tab to configure (e.g. **Account 1**). Click **General Settings** on the left. Configure the following parameters:

- **Account Active**: check the **Yes** box.
- **Account Name**: Enter descriptive text.
- **SIP Server**: Enter the Session Manager signaling IP address.
- **SIP User ID**: Enter the SIP User created in **Section 6.1**.
- **SIP Authentication ID**: Enter the SIP User created in **Section 6.1**.
- **SIP Authentication Password**: Enter the password for the User created in **Section 6.1**.
- **Voice Mail UserID**: Enter the Avaya Aura® Messaging pilot number. Configuration of Avaya Aura® Messaging is outside the scope of this document. The mailboxes for the Grandstream phones are configured the same and Avaya phones).
- **Name**: Enter descriptive text (optional)

Use the default values for the remaining fields. Click the **Save** button at the bottom of the screen.



7.3. Configure SIP Settings

Click the **SIP Settings** tab on the left. Check the **Yes** box for **SUBSCRIBE for MWI**. Use the default values for the remaining fields. Click the **Save** button at the bottom of the screen.

7.4. Configure Code Settings

Click the **Codec Settings** tab on the left. The settings on the screen below show that values that were used during compliance testing; however, **Preferred Vocoder** settings were modified occasionally during the testing to test various codecs. After making any changes, click the **Save** button at the bottom of the screen.

7.5. Configure Packetization Mode

By default, the Grandstream phones support packetization mode set to 1. At the time of testing, the Avaya video endpoints did not support packetization mode set to 1. Therefore, the packetization mode value on the Grandstream phones was changed to 0.

To change the packetization mode value on the Grandstream phone, open a secure shell session to the phone's IP address. Log in using appropriate credentials. The commands required to change the value (and output) are shown below.

```
GXV3240 > config
CONFIG > get 957
957=1
CONFIG > set 957 0
Set 957=0
CONFIG> commit
nvram commited
CONFIG > exit
GXV3240 > reboot
You have chosen to reboot.
Do you want to continue (y/N)? y
```

8. Verification Steps

This section includes steps that can be followed to verify the configuration.

8.1. Verify Point to Point Audio and Video Calls

Place point to point audio and video calls between the Grandstream phones and Avaya endpoints. Verify 2-way audio as well as 2-way video for the video calls.

9. Conclusion

These Application Notes describe the procedures required for the Grandstream Multimedia IP Phones to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager to support the reference configuration shown in **Figure 1**. Refer to **Section 2.2** for testing results and any observations noted during testing.

10. Additional References

Product documentation for Avaya products may be found at: http://support.avaya.com.

[1] Administering Avaya Aura® Communication Manager, Release 6.3, Issue 10, June 2014.
[2] Administering Avaya Aura® Session Manager, Release 6.3, Issue 7, September 2014.
[3] Administering Avaya Aura® Messaging, Release 6.3.2, Issue 1, December 2014.

Product documentation for Grandstream products may be found at: http://www.grandstream.com.

[4] GXV3240 IP Multimedia Phone for Android™ Administration Guide
[5] GXV3275 IP Multimedia Phone for Android™ Administration Guide

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.