



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 (96x1 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.

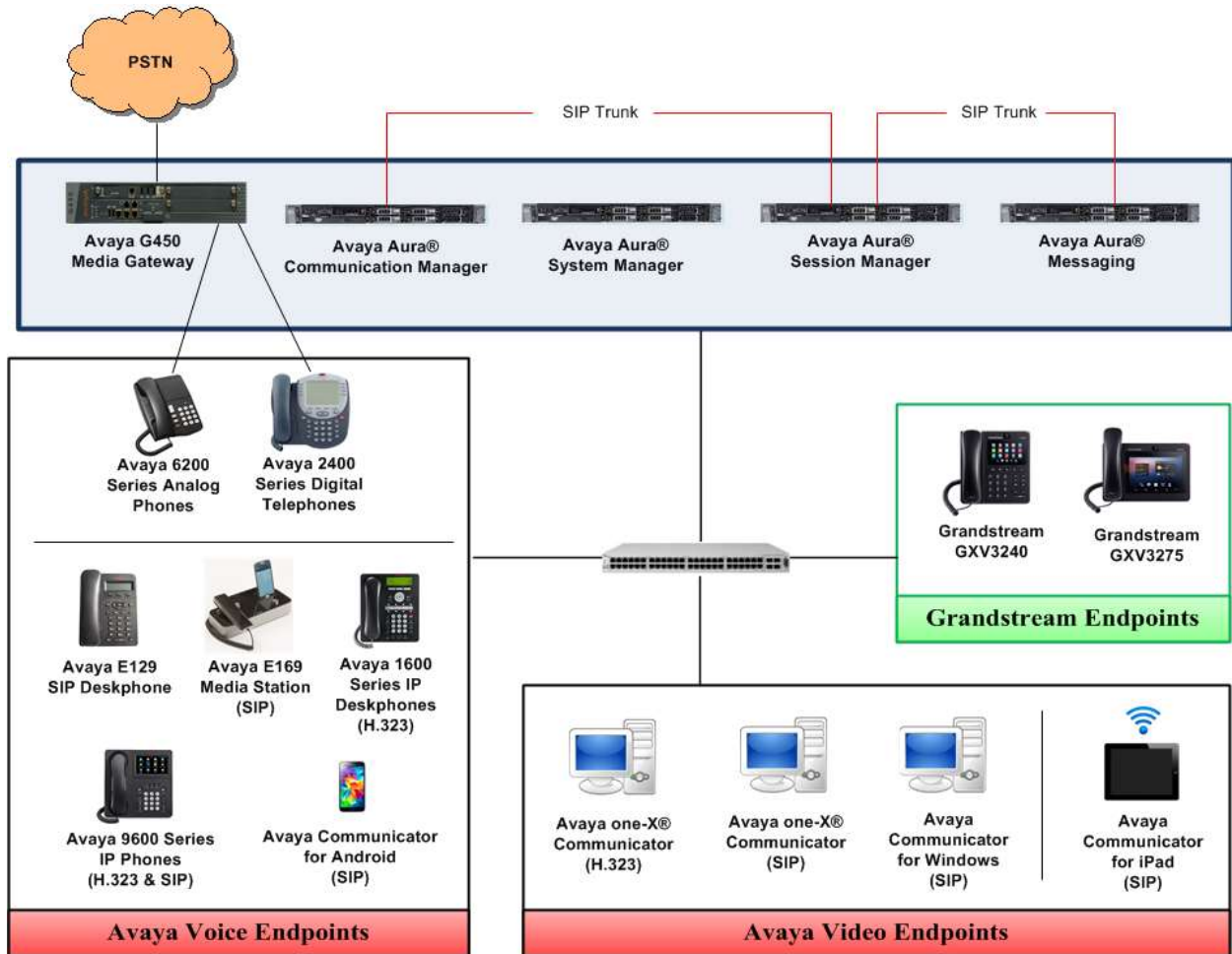


Figure 1: Grandstream Multimedia IP Phones

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

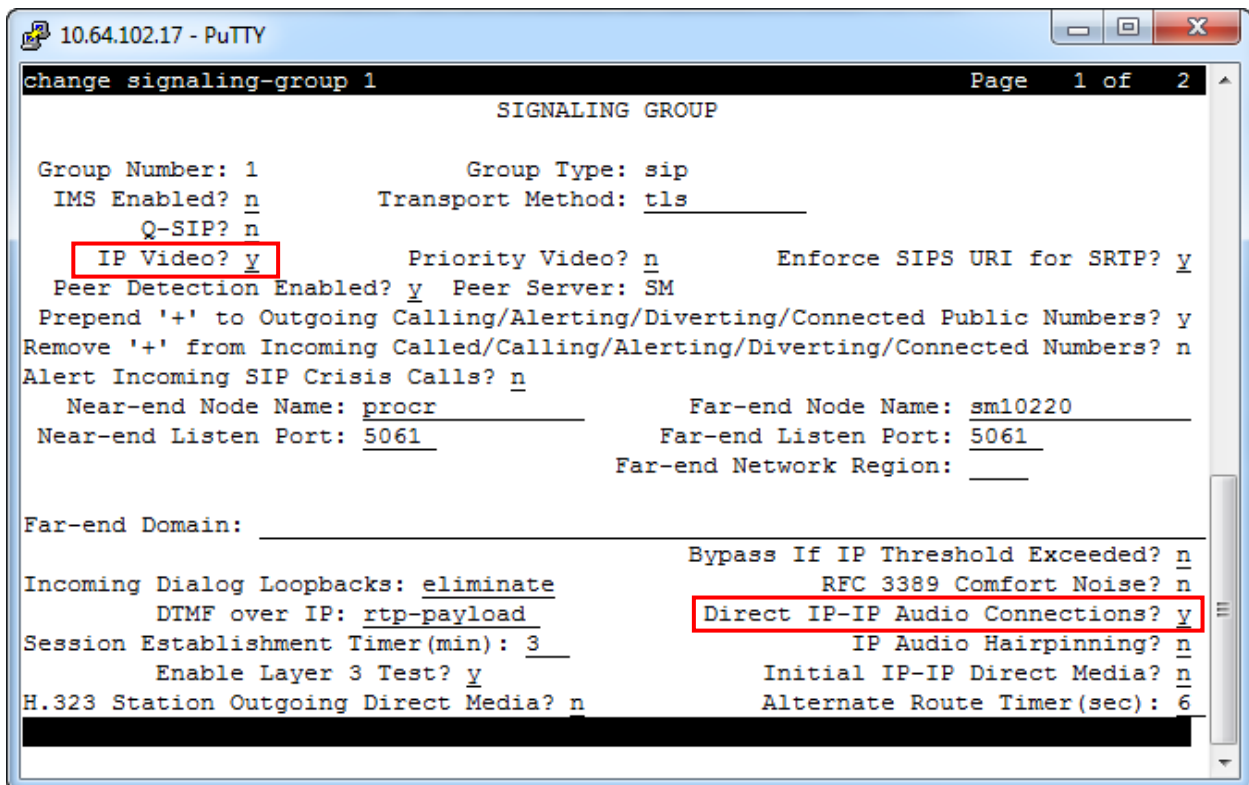
5. Configure Avaya Aura® Communication Manager

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.

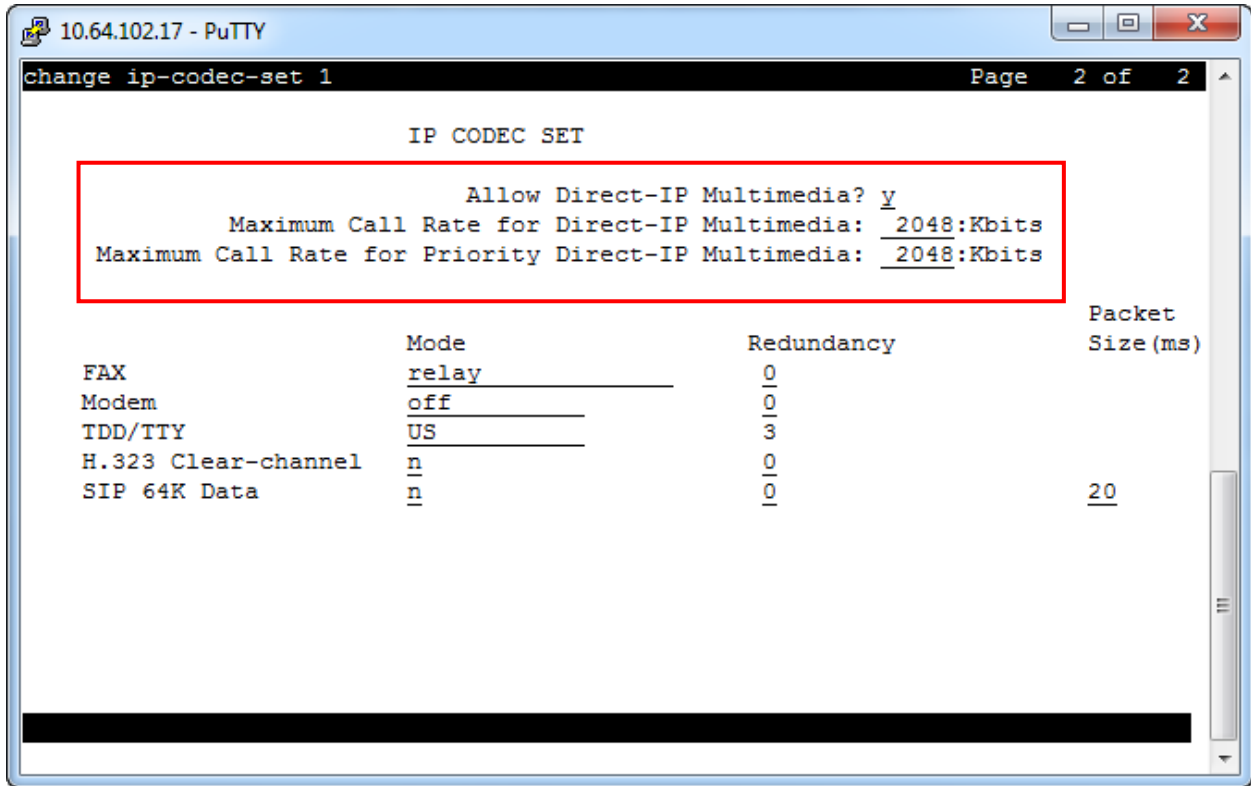


```
10.64.102.17 - PuTTY
change signaling-group 1 Page 1 of 2
SIGNALING GROUP

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:           

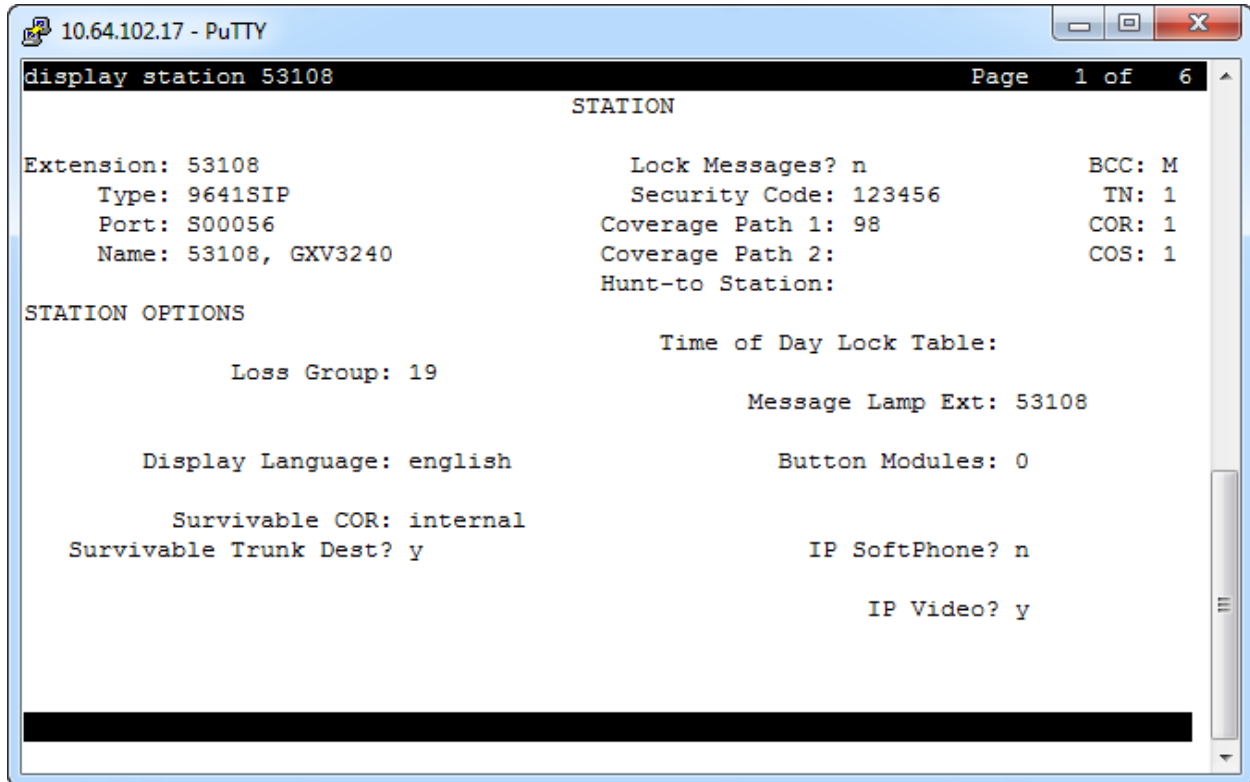
Far-end Domain:                   
Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate RFC 3389 Comfort Noise? n
DTMF over IP: rtp-payload       Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3 IP Audio Hairpinning? n
Enable Layer 3 Test? y           Initial IP-IP 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.

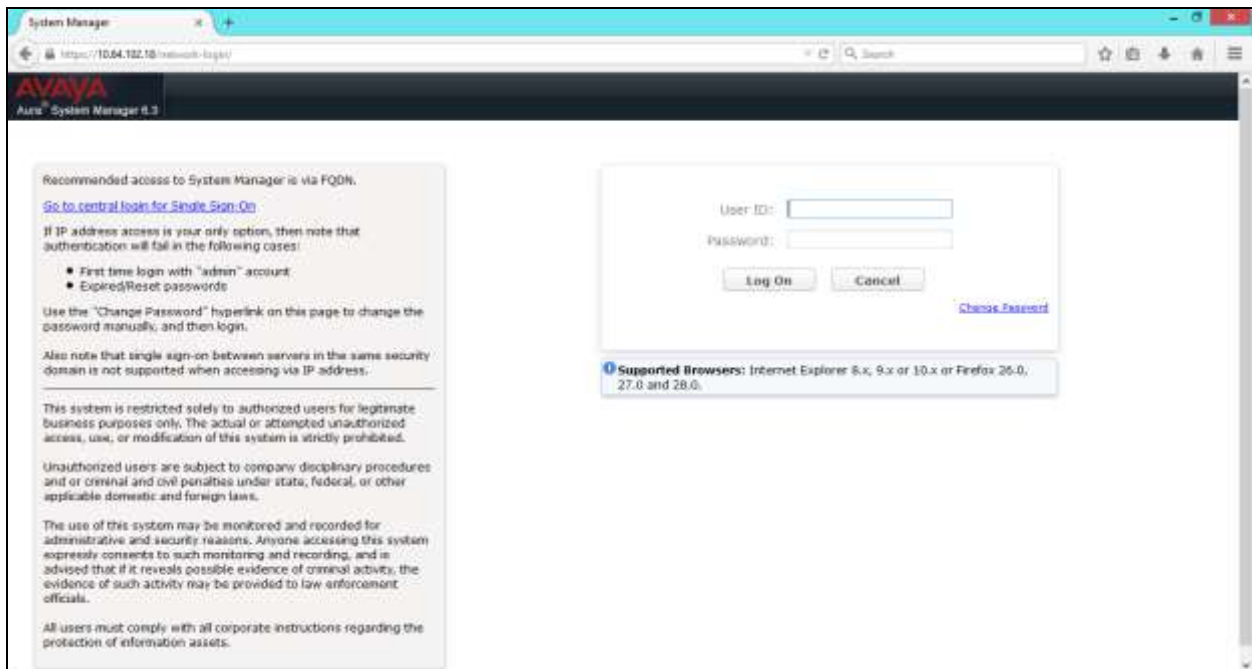


```
10.64.102.17 - PuTTY
display station 53108 Page 1 of 6
STATION
Extension: 53108           Lock Messages? n       BCC: M
  Type: 9641SIP           Security Code: 123456   TN: 1
  Port: S00056            Coverage Path 1: 98    COR: 1
  Name: 53108, GXV3240    Coverage Path 2:      COS: 1
                          Hunt-to Station:
STATION OPTIONS
      Loss Group: 19
      Display Language: english
      Survivable COR: internal
      Survivable Trunk Dest? y
      Time of Day Lock Table:
      Message Lamp Ext: 53108
      Button Modules: 0
      IP SoftPhone? n
      IP Video? y
```

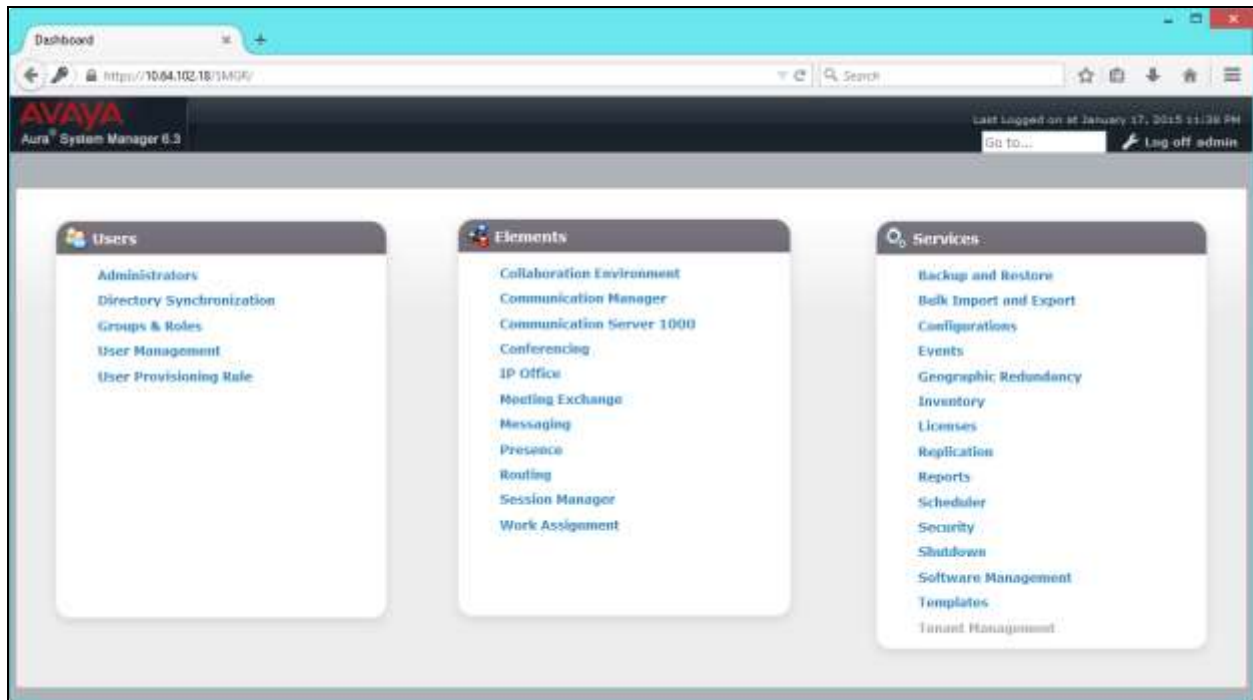

6. Configure Avaya Aura® Session Manager

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.

The screenshot displays the Avaya System Manager 6.3 User Management interface. The browser address bar shows the URL `http://10.64.102.18:11000/`. The page title is "User Management". The breadcrumb trail is "Home / Users / User Management / Manage Users". A search bar is present. The main content area shows a table of users with columns: Last Name, First Name, Display Name, Login Name, SIP Handle, and Last Login. The table contains 15 items, with the first row selected. The "New" button is visible above the table.

	Last Name	First Name	Display Name	Login Name	SIP Handle	Last Login
<input checked="" type="checkbox"/>	53000	Station	53000, Station	53000@avaya.com	53000	
<input type="checkbox"/>	53001	Station	53001, Station	53001@avaya.com	53001	
<input type="checkbox"/>	53003	Station	53003, Station	53003@avaya.com	53003	
<input type="checkbox"/>	53101	Station	53101, Station	53101@avaya.com	53101	
<input type="checkbox"/>	53102	Station	53102, Station	53102@avaya.com	53102	
<input type="checkbox"/>	53103	Station	53103, Station	53103@avaya.com	53103	
<input type="checkbox"/>	53104	Station	53104, Station	53104@avaya.com	53104	
<input type="checkbox"/>	53105	Station	53105, Station	53105@avaya.com	53105	

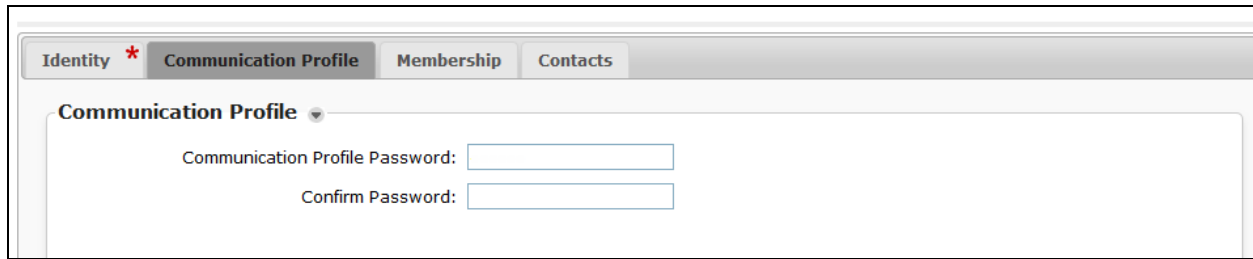
On the **Identity** tab, enter descriptive values for **Last Name** and **First Name**. Enter the **Login Name** (e.g. *user@sipdomain*; this document assumes the SIP domain was previously configured. Consult Reference [2] for further details if necessary).

The screenshot shows the 'New User Profile' form in the Avaya Aura System Manager 6.3 interface. The form is organized into several sections:

- User Provisioning Rule:** A dropdown menu.
- Identity:**
 - Last Name:** 53108 (Required)
 - Last Name (Latin Translation):** 53108
 - First Name:** GXV3240 (Required)
 - First Name (Latin Translation):** GXV3240
 - Middle Name:** (Empty)
 - Description:** (Text area)
 - Login Name:** 53108@avaya.com (Required)
 - Authentication Type:** Basic (Required)
 - Password:** (Text field)
 - Confirm Password:** (Text field)
 - Localized Display Name:** (Text field)
 - Endpoint Display Name:** (Text field)
 - Title:** (Text field)
 - Language Preference:** (Dropdown menu)
 - Time Zone:** (Dropdown menu)
 - Employee ID:** (Text field)
 - Department:** (Text field)
 - Company:** (Text field)
- Address:** (Text area)
- Localized Names:** (Text area)

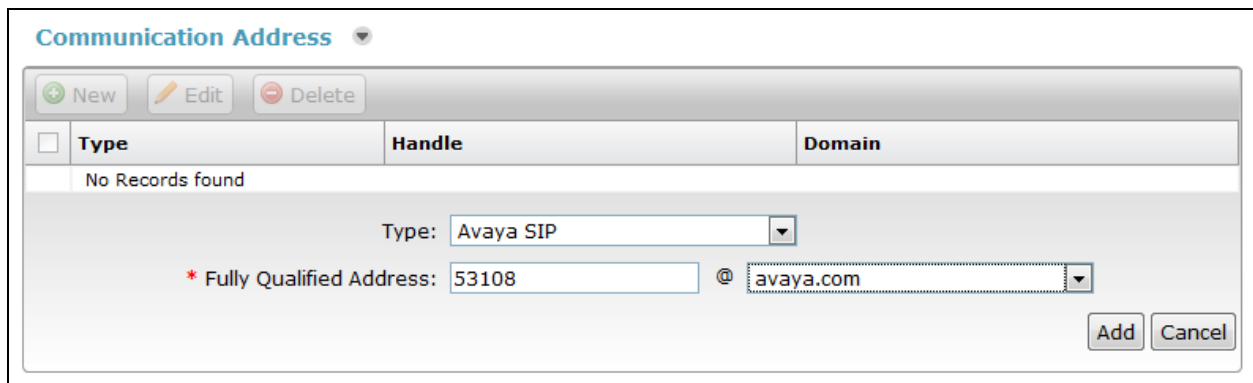
Buttons at the bottom right include 'Commit & Continue', 'Commit', and 'Cancel'. A legend indicates that fields with an asterisk are required.

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



The screenshot shows a web interface with four tabs: Identity, Communication Profile, Membership, and Contacts. The 'Communication Profile' tab is selected. Below the tabs, there is a section titled 'Communication Profile' with a dropdown arrow. Underneath, there are two text input fields: 'Communication Profile Password:' and '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.



The screenshot shows the 'Communication Address' management interface. At the top, there are three buttons: 'New', 'Edit', and 'Delete'. Below these is a table with columns for 'Type', 'Handle', and 'Domain'. The table is currently empty, showing 'No Records found'. Below the table, there are two dropdown menus: 'Type' (set to 'Avaya SIP') and 'Fully Qualified Address' (set to '53108'). To the right of the address field is an '@' symbol and a dropdown menu for the domain (set to 'avaya.com'). At the bottom right, there are 'Add' and 'Cancel' buttons.

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.

Session Manager Profile ▼

SIP Registration

* Primary Session Manager ▼

Primary	Secondary	Maximum
18	0	18

Secondary Session Manager ▼

Primary	Secondary	Maximum

Survivability Server ▼

Max. Simultaneous Devices ▼

Block New Registration When Maximum Registrations Active?

Application Sequences

Origination Sequence ▼

Termination Sequence ▼

Call Routing Settings

* Home Location ▼

Conference Factory Set ▼

Call History Settings

Enable Centralized Call History?

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.

CM Endpoint Profile ▼

* System ▼

* Profile Type ▼

Use Existing Endpoints

* Extension

* Template ▼

Set Type

Security Code

Port

Voice Mail Number

Preferred Handle ▼

Enhanced Callr-Info display for 1-line phones

Delete Endpoint on Unassign of Endpoint from User or on Delete User.

Override Endpoint Name and Localized Name

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

The screenshot shows the 'Edit Endpoint' configuration page in the Avaya Aura System Manager. The page is titled 'Edit Endpoint' and has a breadcrumb trail: 'Home / Users / User Management / Manage Users'. The left sidebar contains a navigation menu with options: 'User Management', 'Manage Users', 'Public Contacts', 'Shared Addresses', 'System Presence', 'ACLs', 'Communication', 'Profile Password', and 'Policy'. The main content area is divided into several sections:

- Basic Information:** Fields for System (cm10217), Extension (53108), Template (9641SIP_DEFAULT_CM_6_3), Set Type (9641SIP), Port (SP), Security Code (*****), and Name (53108,GKV3240).
- General Options (G):** This tab is active and contains the following fields:
 - Class of Restriction (COR): 1
 - Emergency Location Ext: 53108
 - Tenant Number: 1
 - SIP Trunk: gar
 - Coverage Path 1: 98
 - Lock Message:
 - Multibyte Language: Not Applicable
 - Class of Service (COS): 1
 - Message Lamp Ext.: 53108
 - Type of 3PCC Enabled: None
 - Coverage Path 2: (empty)
 - Localized Display Name: 53108,GKV3240

Buttons for 'Done', 'Cancel', and '[Save As Template]' are visible. A 'Help ?' link is also present in the top right corner.

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.

The screenshot displays the 'Edit Endpoint' configuration page in Avaya System Manager. The page is divided into several sections:

- Header:** AVAYA logo, 'Aure System Manager 6.3', and user information (Last Logged on at March 2, 2015 10:04 AM, Log off admin).
- Navigation:** Home, User Management, and a sidebar menu with options like Manage Users, Public Contacts, Shared Addresses, System Presence, ACLs, Communication Profile Password, and Policy.
- Breadcrumbs:** Home / Users / User Management / Manage Users.
- Form Fields:**
 - System:** cm10217
 - Extension:** 53108
 - Template:** 9641SIP_DEFAULT_CM_6_3
 - Set Type:** 9641SIP
 - Port:** IP
 - Security Code:** *****
 - Name:** 53108,GXV3240
- Feature Options Tab:**
 - Active Station Ringing:** single
 - MWI Served User Type:** None
 - Per Station CPN - Send Calling Number:** None
 - IP Phone Group ID:** [Empty]
 - Remote Soft Phone Emergency Calls:** on-local
 - LWC Reception:** spe
 - AUDIX Name:** [Empty]
 - EC500 State:** enabled
 - Short/Prefixed Registration Allowed:** default
 - Music Source:** [Empty]
 - Auto Answer:** none
 - Coverage After Forwarding:** system
 - Display Language:** english
 - Hunt-to Station:** [Empty]
 - Loss Group:** 19
 - Survivable COR:** intamal
 - Time of Day Lock Table:** None
 - Voice Mail Number:** 50000
- Features List:**
 - Always Use
 - IP Audio Hairpinning
 - Bridged Call Alerting
 - Bridged Idle Line Preference
 - Coverage Message Retrieval
 - Data Restriction
 - Survivable Trunk Dest
 - Bridged Appearance Origination Restriction
 - Restrict Last Appearance
 - Turn on mute for remote off-hook attempt
 - Idle Appearance Preference
 - IP SoftPhone
 - LWC Activation
 - CDR Privacy
 - Direct IP-IP Audio Connections
 - H.320 Conversion
 - IP Video
 - Per Button Ring Control
- Buttons:** Done, Cancel, [Save As Template]
- Legend:** *Required



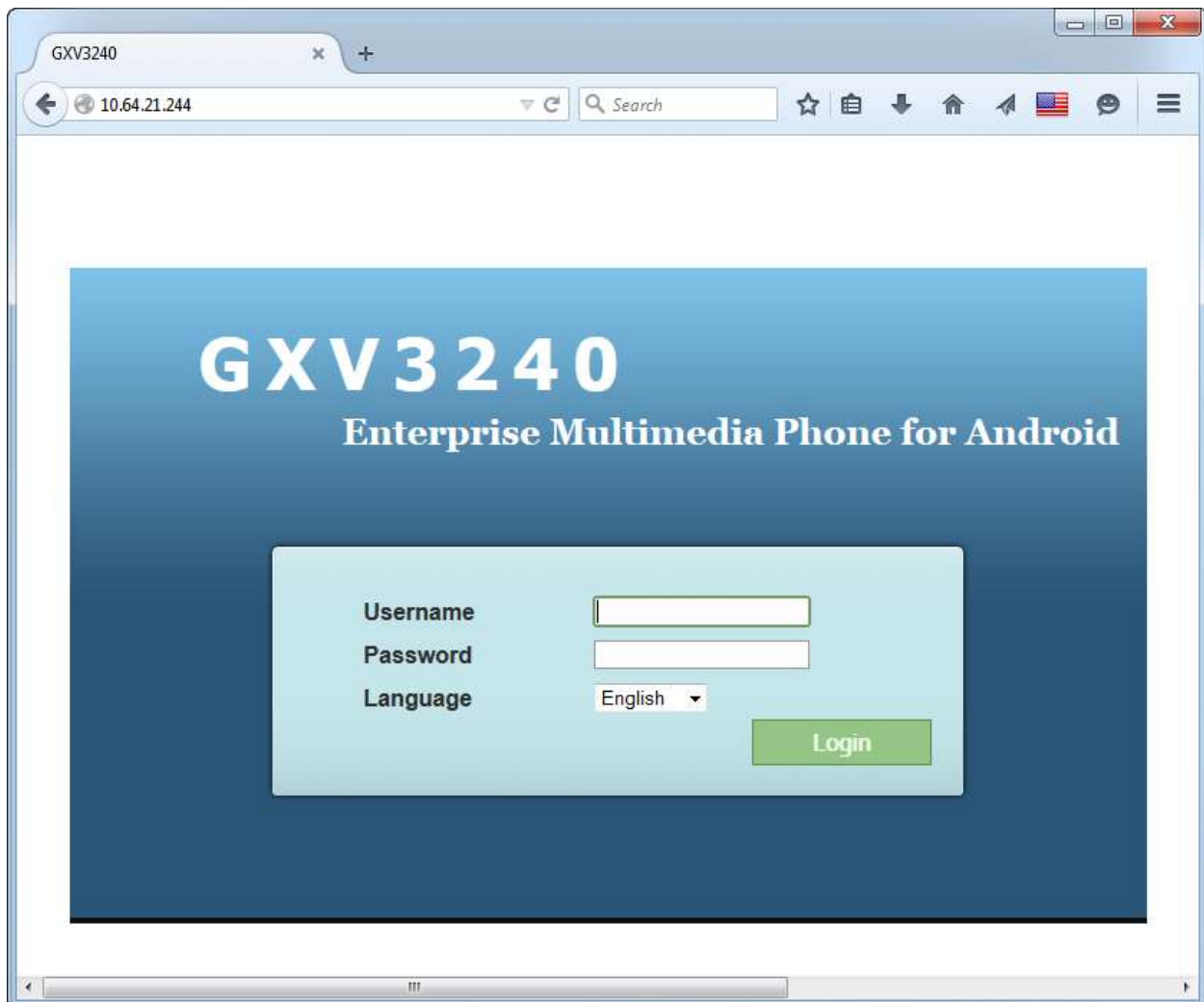
7. Configure Grandstream GXV3240 and GXV3275 Multimedia IP Phones for Android™

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.



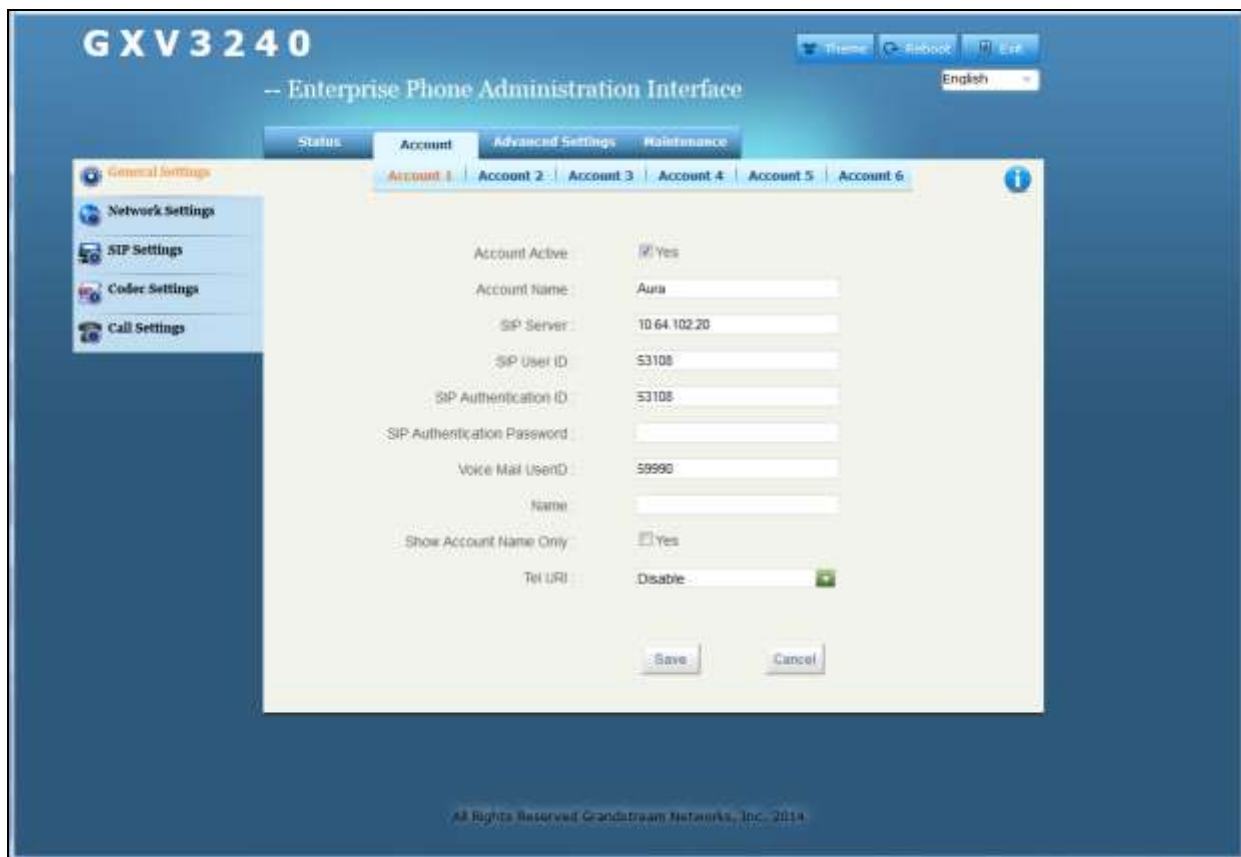
After making and saving any configuration changes throughout the remainder of **Section 7**, an **Apply** button will appear at the top of the web interface. Click the **Apply** button (not shown) to immediately apply the changes.

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.

The screenshot displays the GXV3240 Enterprise Phone Administration Interface. The interface is titled "GXV3240 -- Enterprise Phone Administration Interface" and includes a language dropdown set to "English". The main navigation menu on the left includes: General Settings, Network Settings, SIP Settings, **Codec Settings** (highlighted), and Call Settings. The top navigation bar shows tabs for Status, Account, Advanced Settings, and Maintenance. Under the Account tab, there are sub-tabs for Account 1 through Account 6. The "Codec Settings" page for Account 1 is shown, featuring the following configuration options:

- DTMF: In audio RFC2833 SIP INFO
- DTMF Payload Type: 101
- Preferred Vocoder: Available (G722, G726-32, ILBC, Opus) and Selected (PCMU, PCMA, G729A/B)
- ILBC Frame Size: 30 ms
- Enable RFC3168 Support: Yes
- H.264 Image Size: CIF
- H.264 Profile Type: Baseline Profile
- Video Bit Rate: 384 kbps
- SOP Bandwidth Attribute: Media Level
- H.264 Payload Type: 99
- SRTP Mode: Disable
- SRTP Key Length: AES 128&256 bit
- Silence Suppression: Yes
- Voice Frames Per TX: 2
- Jitter Buffer Maximum (ms): 50
- RTCP Destination: [Empty field]

At the bottom of the settings area are "Save" and "Cancel" buttons. The footer of the interface reads "All Rights Reserved Grandstream Networks, Inc. 2014".

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 committed
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

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