



Avaya Solution & Interoperability Test Lab

Application Notes for the Grandstream Networks SIP Telephones with Avaya Communication Manager 4.0.1 and Avaya SIP Enablement Services 4.0 – Issue 0.1

Abstract

These Application Notes describe a solution comprised of Avaya Communication Manager 4.0.1, Avaya SIP Enablement Services 4.0, and Grandstream Networks SIP telephones. Grandstream GXP2020 and GXP1200 are SIP-based VoIP telephones. Grandstream GXP2020 telephone is typically used in an enterprise or small business environment and Grandstream GXP1200 telephone is used by residential or Small Office and Home Office users. During compliance testing, Grandstream telephones successfully registered with Avaya SES, placed and received calls to and from SIP and non-SIP telephones, and executed other telephony features like three-way conference, transfers, holds, etc. Information in these Application Notes has been obtained through 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 a solution comprised of Avaya Communication Manager 4.0.1, Avaya SIP Enablement Services (SES) 4.0, and Grandstream Networks SIP telephones. Grandstream GXP2020 and GXP1200 are SIP-based VoIP telephones. Grandstream GXP2020 telephone is typically used in an enterprise or small business environment and Grandstream GXP1200 telephone is used by residential or Small Office and Home Office (SoHo) users. During compliance testing, Grandstream telephones successfully registered with Avaya SES, placed and received calls to and from SIP and non-SIP telephones, and executed other telephony features like three-way conference, transfers, holds, etc. Grandstream telephones can bridge calls on a single line to establish a three-party conference. Grandstream GXP2020 supports up to six and GXP1200 is a single line telephone. Grandstream telephones support IM and Presence but no testing was done because of incompatibility with Avaya's implementation.

Figure 1 illustrates a sample configuration consisting of a pair of Avaya S8710 Media Servers, an Avaya G650 Media Gateway, an Avaya SIP Enablement Services (SES) server, and the Grandstream telephones. Avaya Communication Manager is installed on the S8710 Media Servers. The solution described herein is also extensible to other Avaya Media Servers and Media Gateways. For completeness, Avaya 4600 Series SIP IP Telephones, Avaya 4600 Series H.323 IP Telephones, and Avaya 6400 and 8400 Series Digital Telephones, are included in **Figure 1** to demonstrate calls between the SIP-based Grandstream telephones and Avaya SIP, H.323, and digital telephones. The analog PSTN telephone is also included to demonstrate calls routed by Avaya Communication Manager between the Grandstream telephones and the PSTN.

The Grandstream telephone originates a call by sending a call request (SIP INVITE message) to the Avaya SES server. The Avaya SES server routes the call over a SIP trunk to Avaya Communication Manager for origination services. If the call is destined for another local SIP telephone, then Avaya Communication Manager routes the call back over the SIP trunk to Avaya SES server for delivery to destination SIP telephone. Otherwise, Avaya Communication Manager routes the call to the PSTN, a local Avaya H.323, digital, or analog telephone, an adjunct, a vector, a hunt group, etc., depending on the destination number.

For a call arriving at Avaya Communication Manager that is destined for the Grandstream telephone, Avaya Communication Manager routes the call over the SIP trunk to the Avaya SES server for delivery to Grandstream telephone.

These application notes assume that Avaya Communication Manager and Avaya SES are already installed and basic configuration steps have been performed. Only steps relevant to this compliance test will be described in this document. For further details on configuration steps not covered in this document, consult [3] and [4].

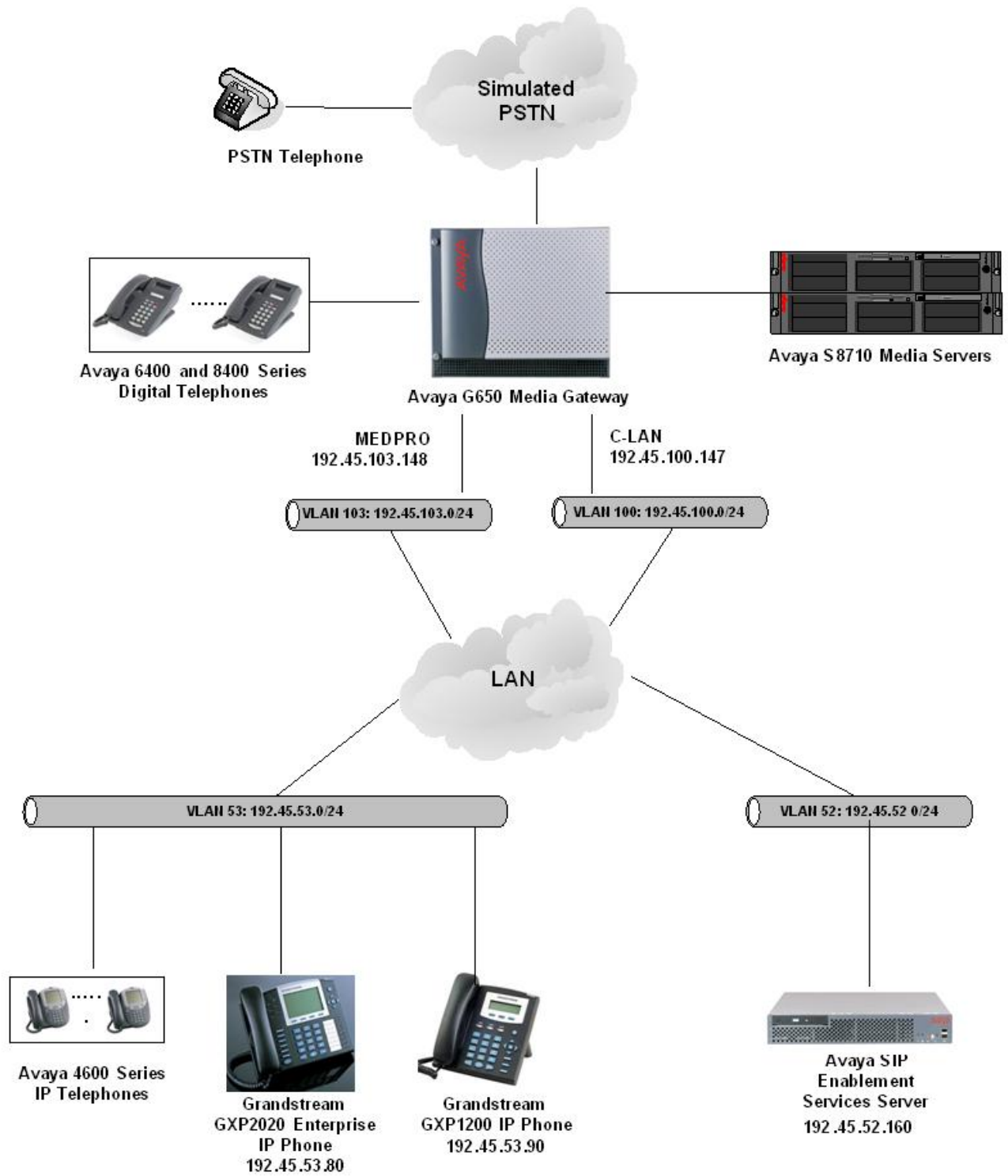


Figure 1: Sample configuration

2. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya S8710 Media Server	Avaya Communication Manager 4.0.1 (R014x.00.1.731.2)
Avaya G650 Media Gateway	-
TN2312BP IP Server Interface	HW12, FW040
TN799DP C-LAN Interface	HW01, FW024
TN2302AP IP Media Processor	HW20, FW117
Avaya SIP Enablement Services Server	SES 4.0 (SES-4.0.0.0-033.6)
Avaya 4600 Series IP Telephones	2.2.3 (4610SW SIP) 2.3 (4602SW H.323) 2.6 (4610SW H.323) 2.5 (4625SW H.323)
Avaya 6400 and 8400 Series Digital Telephones	-
Avaya Analog Telephone	-
Grandstream Networks GXP2020 Telephone	1.1.5.15
Grandstream Networks GXP1200 Telephone	1.1.5.15

3. Configure Avaya Communication Manager

This section describes a procedure for setting up a SIP trunk between Avaya Communication Manager and Avaya SES which includes steps for setting up a list of IP code set, an IP network region, a signaling group and its interface. Before a trunk can be configured, it is necessary to verify if there is enough capacity to setup an additional trunk. Also, a procedure is described here to configure SIP telephones in Avaya Communication Manager. Configuration in the following sections is only for the fields where a value needs to be entered or modified. Default values are used for all other fields. For further information related to configure Avaya Communication Manager refer to [1] and [2].

These steps are performed from the Avaya Communication Manager System Access Terminal (SAT) interface. Grandstream and other SIP telephones are configured as off-PBX telephones in Avaya Communication Manager. Avaya Communication Manager does not directly control an off-PBX telephone but its features and calling privileges can be applied to it by associating a local, on-PBX telephone with the off-PBX telephone. Similarly, a SIP telephone in Avaya SES is associated with an on-PBX telephone on Avaya Communication Manager. SIP Telephones register with the Avaya SES and use Avaya Communication Manager for call origination and termination services. Throughout the rest of this document, on-PBX telephones associated with SIP telephones in such a manner will be referred to as Outboard Proxy SIP (OPS) stations.

3.1. Capacity Verification

Step	Description
1.	<p>Enter the display system-parameters customer-options command. Verify that there are sufficient Maximum Off-PBX Telephones – OPS licenses. If not, contact an authorized Avaya account representative to obtain additional licenses.</p> <pre> display system-parameters customer-options Page 1 of 10 OPTIONAL FEATURES G3 Version: V13 Location: 1 RFA System ID (SID): 1 Platform: 8 RFA Module ID (MID): 1 USED Platform Maximum Ports: 44000 908 Maximum Stations: 36000 410 Maximum XMOBILE Stations: 0 0 Maximum Off-PBX Telephones - EC500: 5 0 Maximum Off-PBX Telephones - OPS: 200 50 Maximum Off-PBX Telephones - SCCAN: 0 0 </pre>
2.	<p>Proceed to Page 2 of OPTIONAL FEATURES form. Verify that the number of SIP trunks supported by the system is sufficient for the number of SIP trunks needed. If not, contact an authorized Avaya account representative to obtain additional licenses.</p> <p>Note: <i>Each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunks for the duration of the call. The license file installed on the system controls the maximum permitted.</i></p> <pre> display system-parameters customer-options Page 2 of 10 OPTIONAL FEATURES IP PORT CAPACITIES USED Maximum Administered H.323 Trunks: 200 148 Maximum Concurrently Registered IP Stations: 1000 2 Maximum Administered Remote Office Trunks: 0 0 Maximum Concurrently Registered Remote Office Stations: 0 0 Maximum Concurrently Registered IP eCons: 0 0 Max Concur Registered Unauthenticated H.323 Stations: 0 0 Maximum Video Capable H.323 Stations: 0 0 Maximum Video Capable IP Softphones: 0 0 Maximum Administered SIP Trunks: 200 153 Maximum Number of DS1 Boards with Echo Cancellation: 0 0 Maximum TN2501 VAL Boards: 1 1 Maximum G250/G350/G700 VAL Sources: 0 0 Maximum TN2602 Boards with 80 VoIP Channels: 2 0 Maximum TN2602 Boards with 320 VoIP Channels: 2 1 Maximum Number of Expanded Meet-me Conference Ports: 0 0 (NOTE: You must logoff & login to effect the permission changes.) </pre>

3.2. IP Codec Set

This section describes the steps for administering codec set in Avaya Communication Manager. This codec set is used in the IP network region for communications between Avaya Communication Manager and Avaya SES.

Step	Description
1.	<p>Enter the change ip-codec-set <c> command, where c is a number between 1 and 7, inclusive. IP codec sets are used in Section 3.3 for configuring IP network region to specify which codec sets may be used within and between network regions. For the compliance testing, G.711MU and G.729AB were used and Media Encryption was set to none as encryption is currently not supported for SIP telephony.</p>
	<pre> change ip-codec-set 2 Page 1 of 2 IP Codec Set Codec Set: 2 Audio Silence Frames Packet Codec Suppression Per Pkt Size(ms) 1: G.711MU n 2 20 2: G.729AB n 2 20 3: 4: 5: 6: 7: Media Encryption 1: none 2: 3: </pre>

3.3. IP Network Region

This section describes the steps for administering an IP network region in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES.

Step	Description
1.	<p>Enter the change ip-network-region <n> command, where n is a number between 1 and 250 inclusive and configure the following:</p> <ul style="list-style-type: none"> • Authoritative Domain – Set to the devconnect.com. This should match the SIP Domain value in Section 4, Step 2. • Intra-region IP-IP Direct Audio – Set to yes to allow direct IP-to-IP audio connectivity between endpoints registered to Avaya Communication Manager or Avaya SES in the same IP network region. • Codec Set – Set the codec set number as provisioned in Section 3.2. • Inter-region IP-IP Direct Audio – Set to yes to allow direct IP-to-IP audio connectivity between endpoints registered to Avaya Communication Manager or Avaya SES in different IP network regions.
	<pre> change ip-network-region 2 Page 1 of 19 IP NETWORK REGION Region: 2 Location: Authoritative Domain: devconnect.com Name: MEDIA PARAMETERS Intra-region IP-IP Direct Audio: yes Codec Set: 2 Inter-region IP-IP Direct Audio: yes UDP Port Min: 2048 IP Audio Hairpinning? y UDP Port Max: 65535 DIFFSERV/TOS PARAMETERS RTCP Reporting Enabled? y Call Control PHB Value: 46 RTCP MONITOR SERVER PARAMETERS Audio PHB Value: 46 Use Default Server Parameters? y Video PHB Value: 26 802.1P/Q PARAMETERS Call Control 802.1p Priority: 6 Audio 802.1p Priority: 6 Video 802.1p Priority: 5 AUDIO RESOURCE RESERVATION PARAMETERS H.323 IP ENDPOINTS RSVP Enabled? n H.323 Link Bounce Recovery? y Idle Traffic Interval (sec): 20 Keep-Alive Interval (sec): 5 Keep-Alive Count: 5 </pre>

Step	Description
2.	Proceed to Page 3 of IP network region configuration and enable inter-region connectivity between regions as per below. For this compliance testing, codec set was set to the IP codec set configured in Section 3.2.
	<pre> Page 3 of 19 Inter Network Region Connection Management src dst codec direct Total Video Dyn rgn rgn set WAN WAN-BW-limits WAN-BW-limits Intervening-regions CAC IGAR 2 1 2 y :NoLimit :NoLimit Intervening-regions CAC IGAR 2 2 2 2 3 2 4 2 5 2 6 2 7 2 8 2 9 2 10 2 11 2 12 2 13 2 14 2 15 </pre>

3.4. IP Node Names

This section describes the steps for setting IP node name for Avaya SES in Avaya Communication Manager.

Step	Description
1.	Enter the change node-names ip command and add a node name for Avaya SES along with its IP address.
	<pre> change node-names ip IP NODE NAMES Name IP Address CLAN-1A06 192.45 .100.147 MEDPRO-1A13 192.45 .103.148 SES 192.45 .52 .160 </pre> <p style="text-align: right;">Page 1 of 1</p>

3.5. SIP Signaling

This section describes the steps for administering a signaling group in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SIP Enablement Services.

Step	Description
1.	<p>Issue the command add signaling-group <s>, where s is an available signaling group and configure the following:</p> <ul style="list-style-type: none"> • Group Type – Set to sip. • Transport Method – Set to tls. • Near-end Node Name - Set to CLAN name as displayed in Section 3.4. • Far-end Node Name - Set to Avaya SES name configured in Section 3.4. • Far-end Network Region - Set to the region configured in Section 3.3. • Far-end Domain - Set to the devconnect.com. This should match the SIP Domain value in Section 4, Step 2. <pre> add signaling-group 10 Page 1 of 5 SIGNALING GROUP Group Number: 10 Group Type: sip Transport Method: tls Near-end Node Name: CLAN-1A06 Far-end Node Name: SES Near-end Listen Port: 5061 Far-end Listen Port: 5061 Far-end Network Region: 2 Far-end Domain: devconnect.com Bypass If IP Threshold Exceeded? n DTMF over IP: rtp-payload Direct IP-IP Audio Connections? y IP Audio Hairpinning? n Session Establishment Timer(min): 120 </pre>

3.6. SIP Trunking

This section describes the steps for administering a trunk group in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES.

Step	Description
1.	<p>Issue the command add trunk-group <t>, where t is an unallocated trunk group and configure the following:</p> <ul style="list-style-type: none"> • Group Type – Set to the Group Type field value configured in Section 3.5. • TAC (Trunk Access Code) – Set to any available trunk access code. • Signaling Group – Set to the Group Number field value configured in Section 3.5. • Number of Members – Allowed values are between 0 and 255. Set to a value large enough to accommodate the number of SIP telephone extensions being used. • Group Name – Enter any descriptive name. <p>Note: <i>Each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunks for the duration of the call. The license file installed on the system controls the maximum permitted.</i></p>
	<pre> add trunk-group 10 Page 1 of 21 TRUNK GROUP Group Number: 10 Group Type: sip CDR Reports: y Group Name: SIP-SES-DevCon1 COR: 1 TN: 1 TAC: 110 Direction: two-way Outgoing Display? n Dial Access? n Night Service: Queue Length: 0 Service Type: tie Auth Code? n Signaling Group: 10 Number of Members: 150 </pre>

3.7. SIP Stations

This section describes the steps for administering OPS stations in Avaya Communication Manager and associating the OPS station extensions with the telephone numbers of the Grandstream telephones.


Step	Description
<p>1.</p>	<p>Enter the add station <s> command, where s is an available extension in the dial plan, to administer an OPS station. On Page 1 of the station form configure the following fields:</p> <ul style="list-style-type: none"> • Type – Set to 6408D+. • Port – Set to X. • Name – Enter any descriptive name. <pre> add station 54007 Page 1 of 4 STATION Extension: 54007 Lock Messages? n BCC: 0 Type: 6408D+ Security Code: TN: 1 Port: X Coverage Path 1: COR: 1 Name: GXP2020 Coverage Path 2: COS: 1 Hunt-to Station: STATION OPTIONS Loss Group: 2 Personalized Ringing Pattern: 1 Data Module? n Message Lamp Ext: 54007 Speakerphone: 2-way Mute Button Enabled? y Display Language: english Media Complex Ext: IP SoftPhone? n </pre>
<p>2.</p>	<p>Proceed to Page 3 of the STATION form and add the required number of call-appr entries in BUTTON ASSIGNMENT field. The number of call appearances should match the Call Limit field value in Step 4.</p> <pre> add station 54007 Page 3 of 3 STATION SITE DATA Room: Headset? n Jack: Speaker? n Cable: Mounting: d Floor: Cord Length: 0 Building: Set Color: ABBREVIATED DIALING LIST1: List2: List3: BUTTON ASSIGNMENTS 1: call-appr 5: 2: call-appr 6: 3: call-appr 7: 4: 8: </pre>

Step	Description												
2.	<p>Enter the add off-pbx-telephone station-mapping command and configure the following:</p> <ul style="list-style-type: none"> • Station Extension – Set the extension of the OPS station as configured above. • Application – Set to OPS. • Phone Number – Enter the number that the Grandstream telephone will use for registration and call termination. In the example below, the Phone Number is the same as the Station Extension, but is not required to be the same. • Trunk Selection – Set to the trunk group number configured in Section 3.6. <pre>add off-pbx-telephone station-mapping Page 1 of 2 STATIONS WITH OFF-PBX TELEPHONE INTEGRATION</pre> <table border="1"> <thead> <tr> <th>Station Extension</th> <th>Application</th> <th>Dial Prefix</th> <th>Phone Number</th> <th>Trunk Selection</th> <th>Configuration Set</th> </tr> </thead> <tbody> <tr> <td>54007</td> <td>OPS</td> <td>-</td> <td>54007</td> <td>10</td> <td>1</td> </tr> </tbody> </table>	Station Extension	Application	Dial Prefix	Phone Number	Trunk Selection	Configuration Set	54007	OPS	-	54007	10	1
Station Extension	Application	Dial Prefix	Phone Number	Trunk Selection	Configuration Set								
54007	OPS	-	54007	10	1								
4.	<p>Proceed to Page 2 of station mapping form and verify that the Call Limit field value matches the number of call appearances configured in Step 2.</p> <pre>add off-pbx-telephone station-mapping 54008 Page 2 of 2 STATIONS WITH OFF-PBX TELEPHONE INTEGRATION</pre> <table border="1"> <thead> <tr> <th>Station Extension</th> <th>Call Limit</th> <th>Mapping Mode</th> <th>Calls Allowed</th> <th>Bridged Calls</th> </tr> </thead> <tbody> <tr> <td>54008</td> <td>2</td> <td>both</td> <td>all</td> <td>both</td> </tr> </tbody> </table>	Station Extension	Call Limit	Mapping Mode	Calls Allowed	Bridged Calls	54008	2	both	all	both		
Station Extension	Call Limit	Mapping Mode	Calls Allowed	Bridged Calls									
54008	2	both	all	both									
3.	<p>Repeat Steps 1 and 2 as necessary to administer additional OPS stations and associations for Grandstream telephones.</p>												

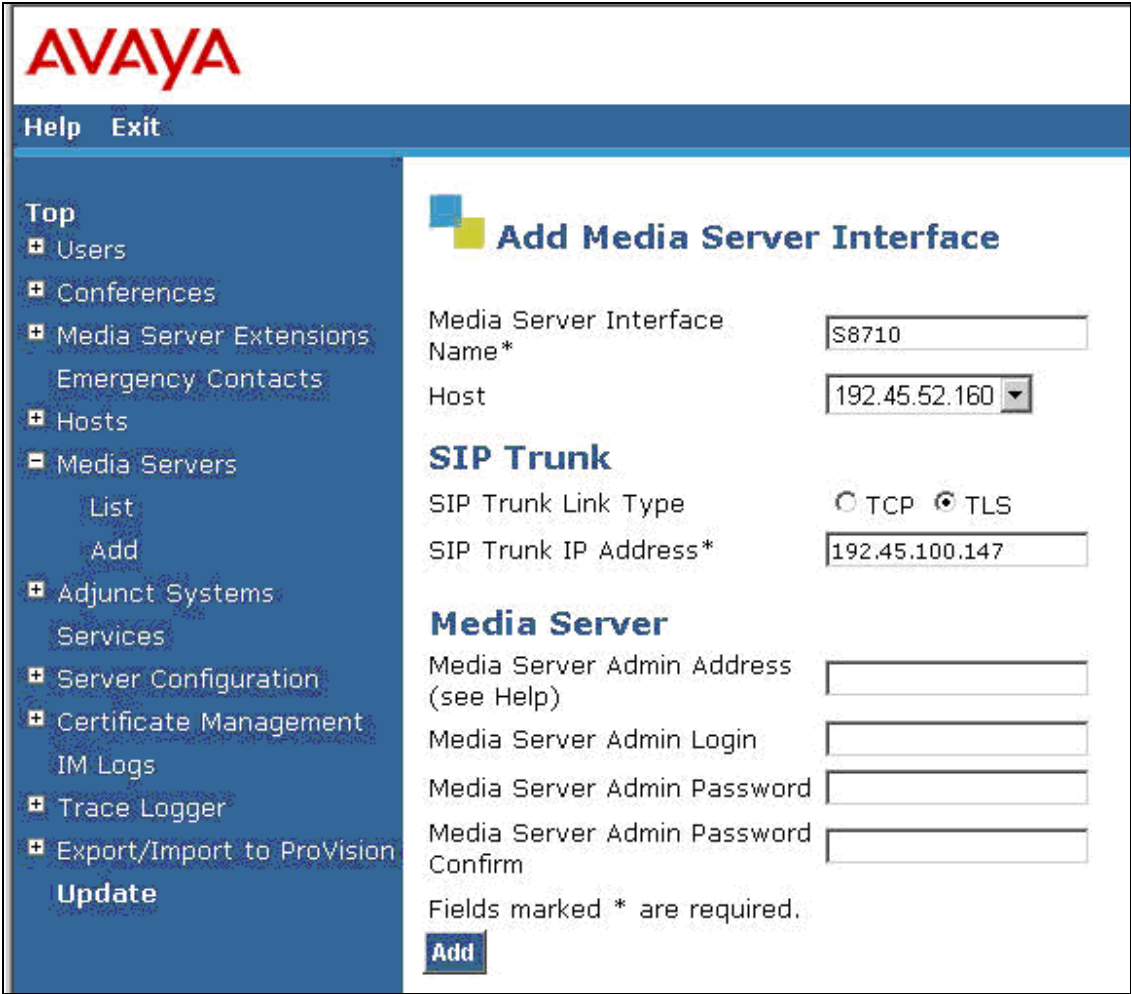
4. Configure Avaya SIP Enablement Services

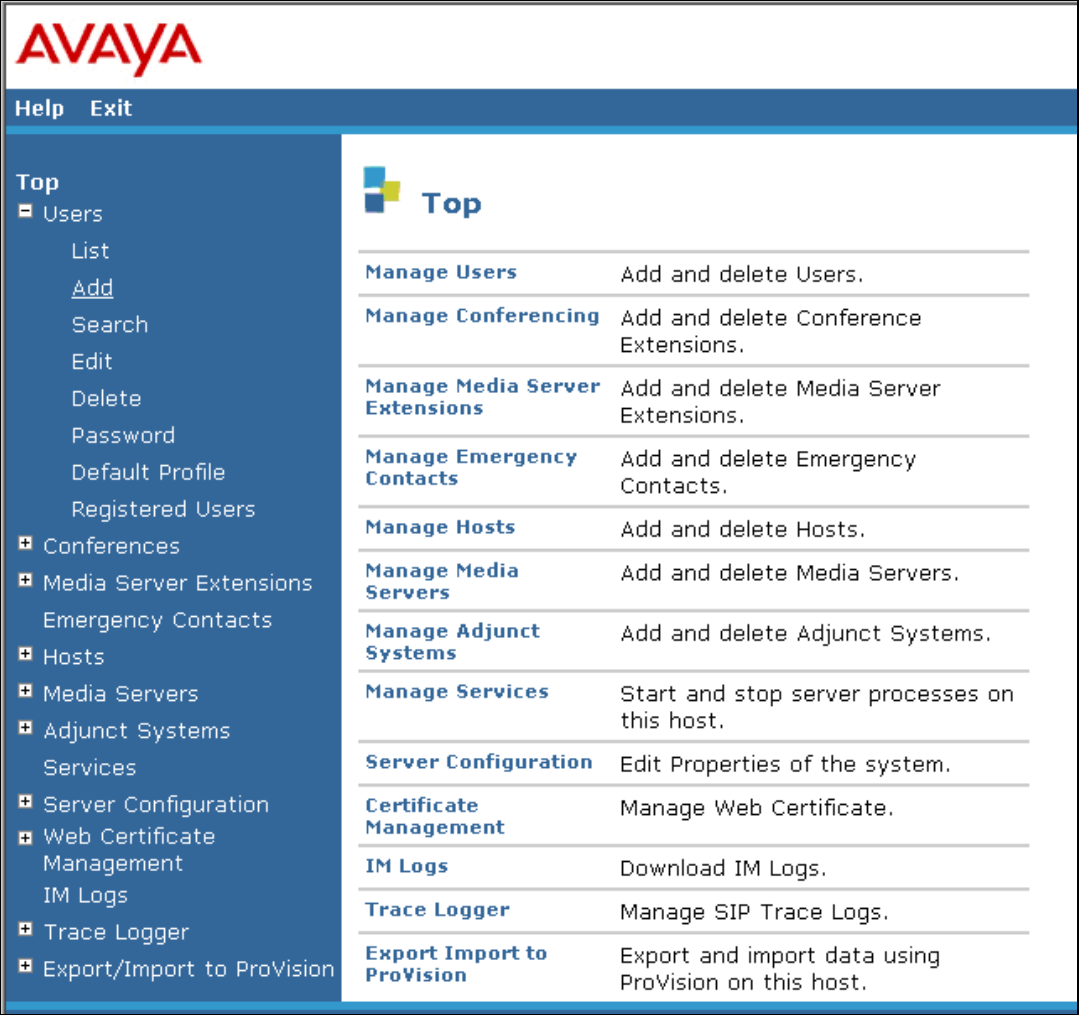
This section describes the steps for creating SIP trunk between Avaya SES and Avaya Communication Manager. Also, SIP user accounts are configured in Avaya SES and associated with an Avaya Communication Manager OPS station extension. The Grandstream telephone will register with Avaya SES using the SIP user accounts. For further information related to configure Avaya SES refer to [5] and [6].

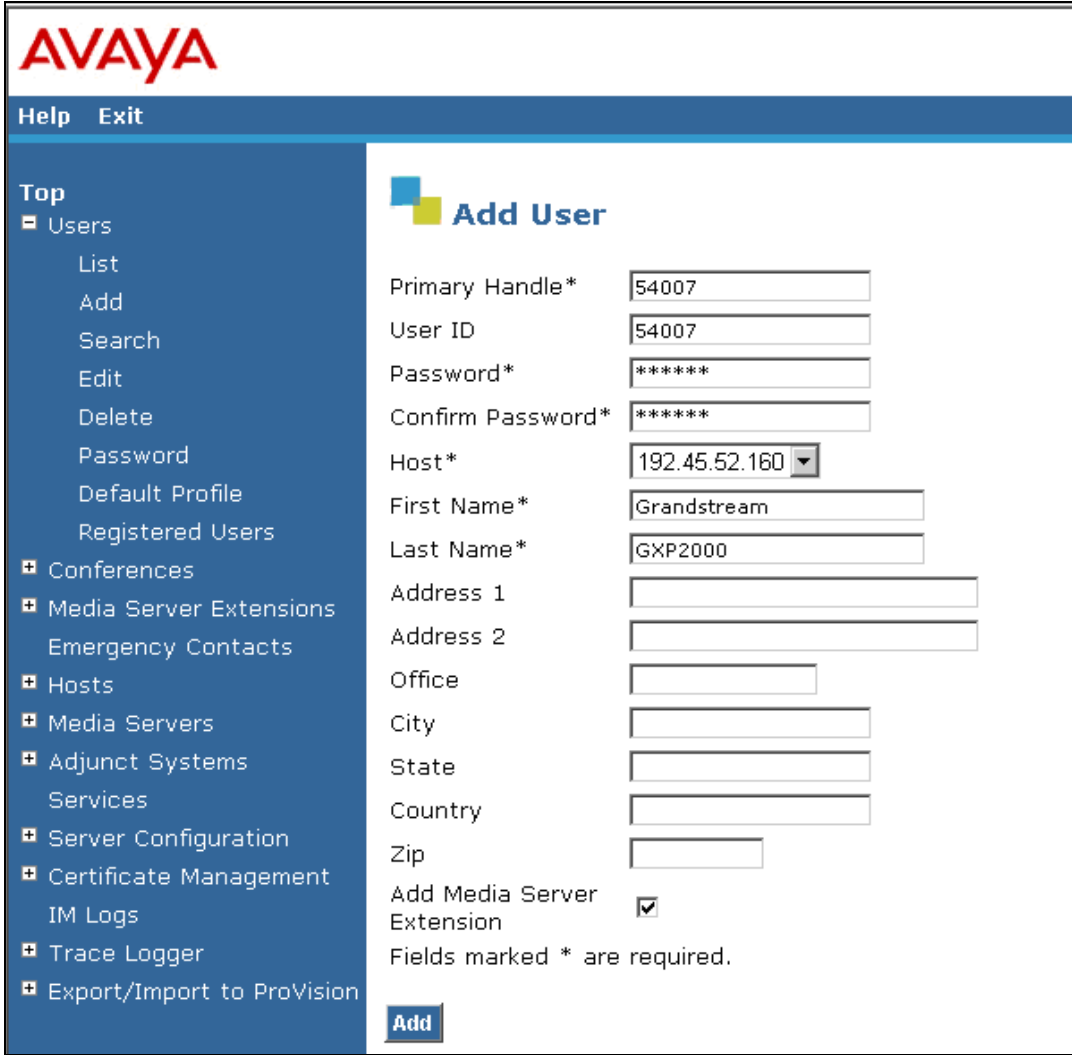
Configuration in the following steps is only for the fields where a value needs to be entered or modified. Default values are used for all other fields.


Step	Description
1.	Open a web browser, enter <a href="http://<IP address of Avaya SES server>/admin">http://<IP address of Avaya SES server>/admin for the URL, and log in with the appropriate credentials. Click on the Launch Administration Web Interface link upon successful login.
2.	<p>On the SIP Server Management page:</p> <ul style="list-style-type: none"> Click the + sign to expand the options under Server Configuration. Click System Properties. Verify the SIP Domain matches the Far-end Domain field value configured for the signaling group on Avaya Communication Manager in Section 3.5. 

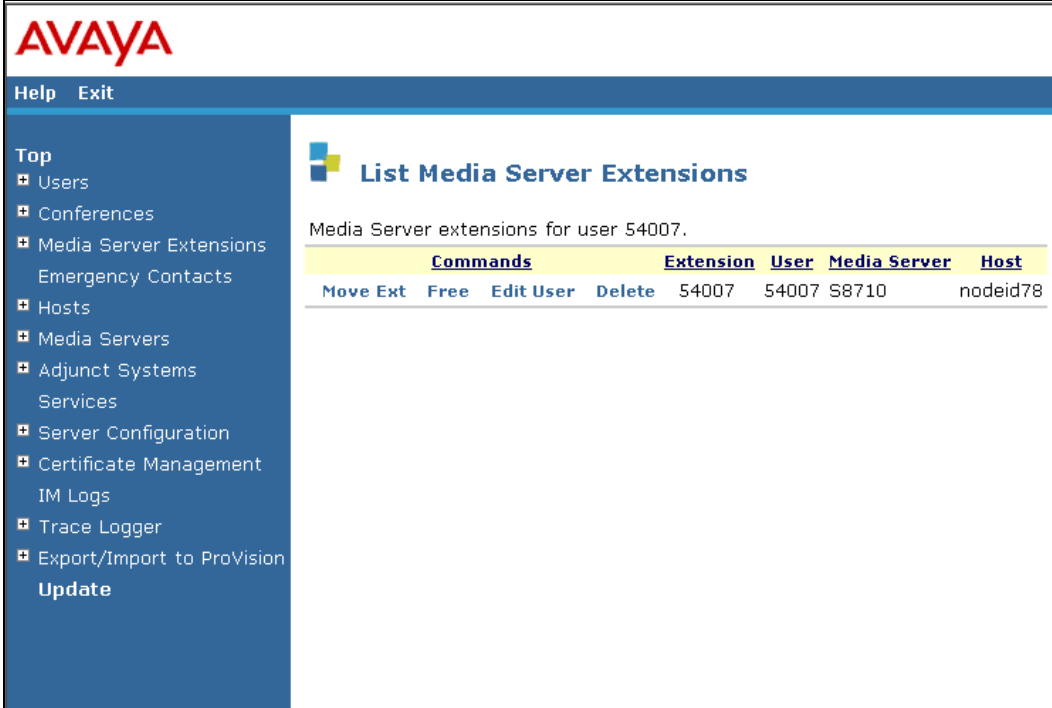
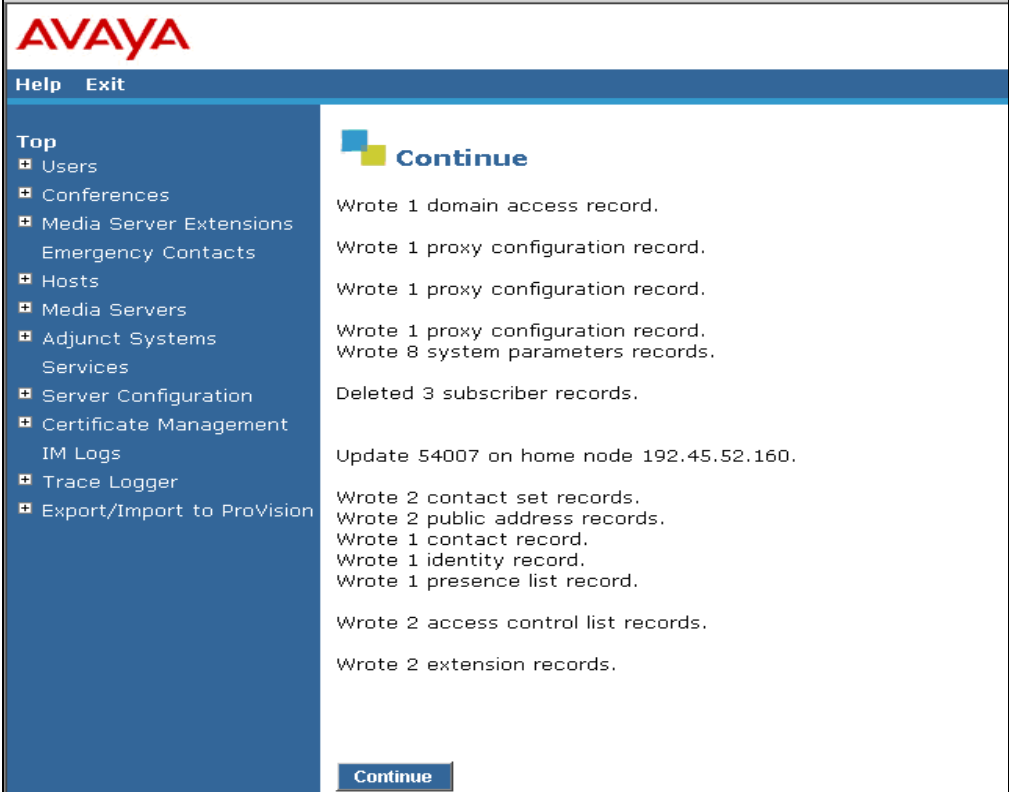
Step	Description						
3.	<p>To enable secure SIP trunking between Avaya SES and Avaya Communication Manager, add a media server corresponding to Avaya Communication Manager from the SIP Server Management page:</p> <ul style="list-style-type: none"> • Click the + sign to expand the options under Media Servers. • Click Add. <div data-bbox="358 380 1352 1136" style="border: 1px solid black; padding: 10px;"> <p>The screenshot shows the Avaya web interface. At the top left is the Avaya logo. Below it is a navigation menu with 'Help' and 'Exit' links. The menu is expanded to show 'Media Servers' with sub-options 'List' and 'Add'. The main content area is titled 'Manage Media Server Interfaces' and contains a table with the following data:</p> <table border="1"> <thead> <tr> <th>Function</th> <th>Description</th> </tr> </thead> <tbody> <tr> <td>List Media Servers</td> <td>List all media server interfaces .</td> </tr> <tr> <td>Add Media Server</td> <td>Add a media server interface .</td> </tr> </tbody> </table> </div>	Function	Description	List Media Servers	List all media server interfaces .	Add Media Server	Add a media server interface .
Function	Description						
List Media Servers	List all media server interfaces .						
Add Media Server	Add a media server interface .						

Step	Description
4.	<p>At the Add Media Server Interface page, provision SIP Trunk parameters as follows for connectivity to Avaya Communications Manager:</p> <ul style="list-style-type: none"> • SIP Trunk Link Type - Set to the Transport Method field value in Section 3.5. • SIP Trunk IP Address - Set to the CLAN IP address as displayed in Section 3.4. • Click Add when finished and then click Continue on the confirmation page [not shown]. <div data-bbox="293 415 1417 1402" style="border: 1px solid black; padding: 10px;">  </div>

Step	Description																										
5.	<p>In the left pane of the SIP Server Management page, expand Users and click Add.</p>  <p>The screenshot displays the Avaya SIP Server Management web interface. At the top left is the Avaya logo. Below it is a navigation bar with 'Help' and 'Exit'. The main interface is split into two panes. The left pane is a dark blue sidebar with a tree view. The 'Users' folder is expanded, showing sub-items: List, Add, Search, Edit, Delete, Password, Default Profile, and Registered Users. Other folders include Conferences, Media Server Extensions, Hosts, Media Servers, Adjunct Systems, Services, Server Configuration, Web Certificate Management, IM Logs, Trace Logger, and Export/Import to ProVision. The right pane is white and titled 'Top'. It contains a list of management functions, each with a description: <table border="1" data-bbox="678 464 1344 1203"> <tr> <td>Manage Users</td> <td>Add and delete Users.</td> </tr> <tr> <td>Manage Conferencing</td> <td>Add and delete Conference Extensions.</td> </tr> <tr> <td>Manage Media Server Extensions</td> <td>Add and delete Media Server Extensions.</td> </tr> <tr> <td>Manage Emergency Contacts</td> <td>Add and delete Emergency Contacts.</td> </tr> <tr> <td>Manage Hosts</td> <td>Add and delete Hosts.</td> </tr> <tr> <td>Manage Media Servers</td> <td>Add and delete Media Servers.</td> </tr> <tr> <td>Manage Adjunct Systems</td> <td>Add and delete Adjunct Systems.</td> </tr> <tr> <td>Manage Services</td> <td>Start and stop server processes on this host.</td> </tr> <tr> <td>Server Configuration</td> <td>Edit Properties of the system.</td> </tr> <tr> <td>Certificate Management</td> <td>Manage Web Certificate.</td> </tr> <tr> <td>IM Logs</td> <td>Download IM Logs.</td> </tr> <tr> <td>Trace Logger</td> <td>Manage SIP Trace Logs.</td> </tr> <tr> <td>Export Import to ProVision</td> <td>Export and import data using ProVision on this host.</td> </tr> </table> </p>	Manage Users	Add and delete Users.	Manage Conferencing	Add and delete Conference Extensions.	Manage Media Server Extensions	Add and delete Media Server Extensions.	Manage Emergency Contacts	Add and delete Emergency Contacts.	Manage Hosts	Add and delete Hosts.	Manage Media Servers	Add and delete Media Servers.	Manage Adjunct Systems	Add and delete Adjunct Systems.	Manage Services	Start and stop server processes on this host.	Server Configuration	Edit Properties of the system.	Certificate Management	Manage Web Certificate.	IM Logs	Download IM Logs.	Trace Logger	Manage SIP Trace Logs.	Export Import to ProVision	Export and import data using ProVision on this host.
Manage Users	Add and delete Users.																										
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Step	Description
6.	<p>At the Add User page, configure the following:</p> <ul style="list-style-type: none"> • Primary Handle – Enter the phone number of the Grandstream telephone. This number was configured in Section 3.7, Step 3. • User ID – Set to any descriptive name. • Password and Confirm Password – Specify a password that the Grandstream telephone will use to register with Avaya SES. • Host – Select the IP address or Fully Qualified Domain Name (FQDN) of the Avaya SES server. • First Name and Last Name – Enter descriptive names. • Check the Add Media Server Extension checkbox. • Click Add when finished and then click Continue on the next page [not shown]. <div data-bbox="321 596 1386 1644" style="border: 1px solid black; padding: 10px;">  </div>


Step	Description
<p>7.</p>	<p>At the Add Media Server Extension page, configure the following:</p> <ul style="list-style-type: none"> • Extension – Set to Phone Number field value configured in Section 3.7, Step 3. • Media Server – Set to the media server where this OPS station is configured. • Click “Add” and then click Continue on the next page [not shown]. <p>Note: Media Server was previously configured on SES</p> 
<p>8.</p>	<p>Repeat Steps 5 – 7 as necessary to configure additional Grandstream telephones.</p>

Step	Description																
<p>9.</p>	<p>Click Update at the bottom of the left panel to save the configuration completed in the above steps.</p>  <p>The screenshot shows the Avaya web interface. The left sidebar is expanded to 'Media Server Extensions' and the 'Update' button is highlighted. The main content area shows a table with one row of data for extension 54007.</p> <table border="1" data-bbox="633 499 1372 556"> <thead> <tr> <th colspan="4">Commands</th> <th>Extension</th> <th>User</th> <th>Media Server</th> <th>Host</th> </tr> </thead> <tbody> <tr> <td>Move Ext</td> <td>Free</td> <td>Edit User</td> <td>Delete</td> <td>54007</td> <td>54007</td> <td>S8710</td> <td>nodeid78</td> </tr> </tbody> </table>	Commands				Extension	User	Media Server	Host	Move Ext	Free	Edit User	Delete	54007	54007	S8710	nodeid78
Commands				Extension	User	Media Server	Host										
Move Ext	Free	Edit User	Delete	54007	54007	S8710	nodeid78										
<p>10.</p>	<p>Click Continue at the bottom of the right panel.</p>  <p>The screenshot shows the Avaya web interface. The left sidebar is expanded to 'Media Server Extensions' and the 'Continue' button is highlighted. The main content area shows a list of actions performed, such as writing domain access records, proxy configuration records, and deleting subscriber records.</p>																

5. Configure Grandstream Telephones

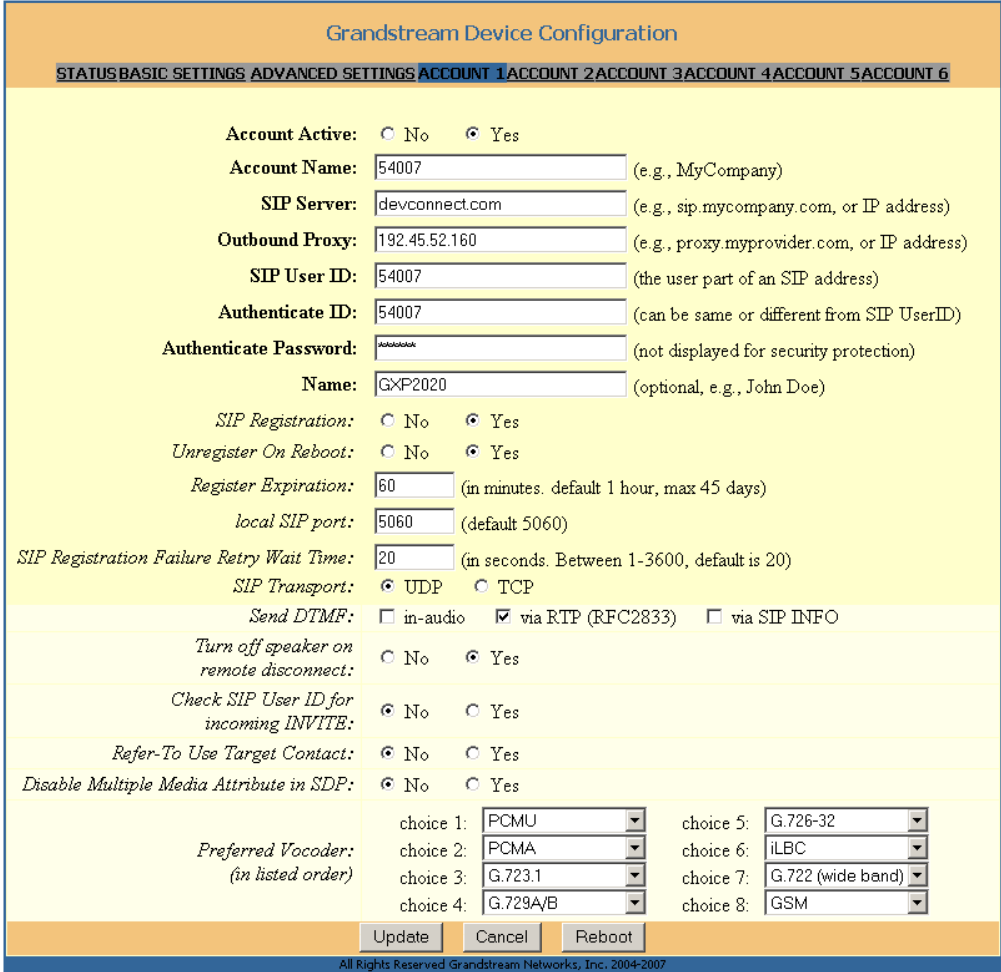
This section describes the steps for configuring the Grandstream telephones. Grandstream GXP2020 and GXP1200 have similar configuration steps except Grandstream GXP2020 supports up to six separate SIP accounts whereas GXP1200 supports up to two separate SIP Accounts. This section assumes that the Grandstream telephone's IP address is already configured. Configuration steps described in this section apply only to the fields where a value needs to be modified or entered. Default values are used for all other fields. Screenshots shown here are for GXP2020 but GXP1200 has similar screens. For further information on Grandstream telephones refer to [5] and [6].

Note: Due to the page size, only the most relevant fields have been included in the screen shots.

Step	Description
1.	<p>Open a web browser and enter http://a.b.c.d for the URL, where a.b.c.d is the IP address of the Grandstream telephone. Enter the password and click Login to proceed to the next screen.</p> 

Step	Description
2.	<p>Select the BASIC SETTINGS tab and check the statically configured as option to configure as follows:</p> <ul style="list-style-type: none"> • IP Address – Set the IP address. • Subnet Mask – Set the subnet mask. • Default Router – Set the default router. • Click Update to modify the values. <div data-bbox="285 422 1425 1205" style="border: 1px solid black; padding: 10px; background-color: #ffffcc;"> <p style="text-align: center; color: #0056b3;">Grandstream Device Configuration</p> <p style="text-align: center; font-weight: bold; color: #0056b3;"> STATUS BASIC SETTINGS ADVANCED SETTINGS ACCOUNT 1 ACCOUNT 2 ACCOUNT 3 ACCOUNT 4 ACCOUNT 5 ACCOUNT 6 </p> <p>End User Password: <input type="text"/> (purposely not displayed for security protection)</p> <p>IP Address: <input type="radio"/> dynamically assigned via DHCP (default) or PPPoE (will attempt PPPoE if DHCP fails and following is non-blank)</p> <p> PPPoE account ID: <input type="text"/></p> <p> PPPoE password: <input type="text"/></p> <p> Host name (Option 12): <input type="text"/></p> <p> Domain name (Option 15): <input type="text"/></p> <p> Vendor Class ID (Option 60): <input type="text" value="Grandstream GXP2020"/></p> <p>Preferred DNS server: <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/></p> <p><input checked="" type="radio"/> statically configured as:</p> <p> IP Address: <input type="text" value="192"/> <input type="text" value="45"/> <input type="text" value="53"/> <input type="text" value="80"/></p> <p> Subnet Mask: <input type="text" value="255"/> <input type="text" value="255"/> <input type="text" value="255"/> <input type="text" value="0"/></p> <p> Default Router: <input type="text" value="192"/> <input type="text" value="45"/> <input type="text" value="53"/> <input type="text" value="1"/></p> <p> DNS Server 1: <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/></p> <p> DNS Server 2: <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/></p> <p style="text-align: center;"> <input type="button" value="Update"/> <input type="button" value="Cancel"/> <input type="button" value="Reboot"/> </p> <p style="text-align: center; font-size: small; color: #0056b3;">All Rights Reserved Grandstream Networks, Inc. 2004-2007</p> </div>

Step	Description
3.	<p>Select the ADVANCED SETTINGS tab and configure as follows:</p> <ul style="list-style-type: none"> • Layer 3 QoS – Set to the desired value between 0 and 63. For compliance testing, a value of 48 was used. • 802.1p priority value – Set to the desired value between 0 and 7. For compliance testing, a value of 6 was used. • Click Update to modify the values. <div data-bbox="277 407 1422 989" style="border: 1px solid black; padding: 10px; margin: 10px 0;"> <p style="text-align: center; color: #4F81BD;">Grandstream Device Configuration</p> <p style="text-align: center; background-color: #F0F0F0; margin: 0;"> STATUS ADVANCED SETTINGS ACCOUNT 1 ACCOUNT 2 ACCOUNT 3 ACCOUNT 4 ACCOUNT 5 ACCOUNT 6 </p> <p>Admin Password: <input type="text"/> (purposely not displayed for security protection)</p> <p><i>Silence Suppression:</i> <input checked="" type="radio"/> No <input type="radio"/> Yes</p> <p><i>Voice Frames per TX:</i> <input type="text" value="2"/> (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)</p> <p><i>Layer 3 QoS:</i> <input type="text" value="48"/> (Diff-Serv or Precedence value)</p> <p><i>Layer 2 QoS:</i> 802.1Q/VLAN Tag <input type="text" value="53"/> 802.1p priority value <input type="text" value="6"/> (0-7)</p> <p><i>No Key Entry Timeout:</i> <input type="text" value="4"/> (in seconds, default is 4 seconds)</p> <p><i>Use # as Dial Key:</i> <input type="radio"/> No <input checked="" type="radio"/> Yes</p> <p><i>local RTP port:</i> <input type="text" value="5004"/> (1024-65535, default 5004)</p> <p><i>Use random port:</i> <input checked="" type="radio"/> No <input type="radio"/> Yes</p> <p><i>keep-alive interval:</i> <input type="text" value="20"/> (in seconds, default 20 seconds)</p> <p><i>DTMF Payload Type:</i> <input type="text" value="101"/></p> <p style="text-align: right;"> <input type="button" value="Update"/> <input type="button" value="Cancel"/> <input type="button" value="Reboot"/> </p> <p style="font-size: small; text-align: center;">All Rights Reserved Grandstream Networks, Inc. 2004-2007</p> </div>

Step	Description
4.	<p>Select the ACCOUNT1 tab and configure as follows:</p> <ul style="list-style-type: none"> • Account Name – Set to the Primary Handle field value configured in Section 4, Step 6. • SIP Server – Set to the SIP Domain field value configured in Section 4, Step 2. • Outbound Proxy – Set to the Avaya SES server IP address. • SIP User ID – Set to the User Id field value configured in Section 4, Step 6. • Authenticate ID – Set to the User Id field value configured in Section 4, Step 6. • Authenticate Password – Set to the Password field value configured in Section 4, Step 6. • Name – Enter any descriptive name. • SIP Transport – Set to UDP. • Send DTMF – set to via RTP. • Turn off speaker on remote disconnect – Set the value to Yes. • Special Vocoder – This should have at least one of the codecs configured in Section 3.2. • Click Update. • Repeat this step to configure additional accounts. For GXP2020, up to six accounts can be configured and for GXP1200, up to two accounts can be configured.
	 <p>The screenshot displays the 'Grandstream Device Configuration' web interface. At the top, there are tabs for 'STATUS', 'BASIC SETTINGS', 'ADVANCED SETTINGS', and 'ACCOUNT 1' through 'ACCOUNT 6'. The 'ACCOUNT 1' tab is selected. The configuration form includes the following fields and options:</p> <ul style="list-style-type: none"> Account Active: Radio buttons for 'No' and 'Yes' (selected). Account Name: Text input field containing '54007'. SIP Server: Text input field containing 'devconnect.com'. Outbound Proxy: Text input field containing '192.45.52.160'. SIP User ID: Text input field containing '54007'. Authenticate ID: Text input field containing '54007'. Authenticate Password: Password input field (masked). Name: Text input field containing 'GXP2020'. SIP Registration: Radio buttons for 'No' and 'Yes' (selected). Unregister On Reboot: Radio buttons for 'No' and 'Yes' (selected). Register Expiration: Text input field containing '60'. local SIP port: Text input field containing '5060'. SIP Registration Failure Retry Wait Time: Text input field containing '20'. SIP Transport: Radio buttons for 'UDP' (selected) and 'TCP'. Send DTMF: Checkboxes for 'in-audio', 'via RTP (RFC2833)' (checked), and 'via SIP INFO'. Turn off speaker on remote disconnect: Radio buttons for 'No' and 'Yes' (selected). Check SIP User ID for incoming INVITE: Radio buttons for 'No' (selected) and 'Yes'. Refer-To Use Target Contact: Radio buttons for 'No' (selected) and 'Yes'. Disable Multiple Media Attribute in SDP: Radio buttons for 'No' (selected) and 'Yes'. Preferred Vocoder: Eight dropdown menus for codecs: choice 1: PCMU, choice 2: PCMA, choice 3: G.723.1, choice 4: G.729A/B, choice 5: G.726-32, choice 6: iLBC, choice 7: G.722 (wide band), choice 8: GSM. <p>At the bottom of the form are three buttons: 'Update', 'Cancel', and 'Reboot'. A footer note reads 'All Rights Reserved Grandstream Networks, Inc. 2004-2007'.</p>

6. Interoperability Compliance Testing

The focus of the interoperability compliance testing was primarily on verifying call establishment on the Grandstream telephones and operations such as dialing methods (manual, re-dial, and phone book), hold, mute, transfer and conference. Grandstream telephones interactions with SES, Avaya Communication Manager, and Avaya SIP, H.323, and digital telephones were also verified.

6.1. General Test Approach

The general test approach was to place calls to and from the Grandstream GXP2020 and GXP1200 telephones and exercise basic telephone operations. The main objectives were to verify that:

- Grandstream telephones successfully register with Avaya SES.
- Grandstream telephones successfully establish calls with Avaya SIP, H.323, and digital telephones attached to Avaya SES or Avaya Communication Manager.
- Grandstream telephones successfully establish calls with PSTN telephones through Avaya Communication Manager.
- Grandstream telephones successfully handle concurrent calls.
- Grandstream telephones successfully negotiate the right codec.
- Grandstream telephones successfully shuffle for VoIP calls.
- Grandstream telephones successfully transmit DTMF during a call.
- Grandstream telephones successfully hold and transfer a call.
- Grandstream telephones establish a three party conference call, and display calling party number.
- Grandstream telephones successfully tags layer-2 (802.1p) and layer-3 (DiffServ) QoS packets.

For serviceability testing, failures such as cable pulls and hardware resets were applied. For performance testing, a conference call involving two Grandstream telephones and two Avaya telephones was formed as follows:

A call was established between an Avaya telephone and a Grandstream telephone. The Grandstream telephone then used its second call appearance to establish a call with another Grandstream telephone, and bridged the two lines together, forming a 3-party conference. The second Grandstream telephone then used its second call appearance to establish a call with another Avaya telephone, and bridged its two lines together, effectively forming a 4-party conference.

6.2. Test Results

The test objectives of **Section 6.1** were verified. For serviceability testing, the Grandstream telephones operated properly after recovering from failures such as cable disconnects, and resets of the Grandstream telephones, the Avaya SES server, and Avaya Communication Manager. For performance testing, the conference call was successfully maintained for approximately two hours. Grandstream telephones successfully shuffled to communicate directly with the other telephones. Grandstream telephones successfully negotiated the codec to be used and properly tagged layer-2 and layer-3 QoS packets.

The following observations were made during testing:

- Grandstream telephones cannot mute all parties if it initiates the conference. Only the last party added is muted.
- Grandstream telephones only support UDP as SIP transport.

Grandstream Networks will address and attempt to resolve the above observations in future firmware releases. Contact Grandstream Networks (www.grandstream.com) for further updates.

7. Verification Steps

The following steps may be used to verify the configuration:

- Verify that the Grandstream telephones successfully register with the Avaya SES server by following the **Users -> Registered Users** links on the SES Administration Web Interface.
- Place calls to and from the Grandstream telephone and verify that the calls are successfully established with two-way talk path.
- From the Avaya Communication Manager System Access Terminal (SAT) interface, perform the following steps to verify:
 - Audio codec used between two telephones
 - Shuffling between two telephones

Step	Description
1.	<p>Enter status trunk <t> command, where t is the SIP trunk configured in Section 3.6. Note down the Member with Service State set to in-service/active. In this example, 0010/002 and 0010/006 are active and either member can be used to verify whether calls shuffled and which codec was used.</p>
	<pre>Status trunk 10 TRUNK GROUP STATUS Member Port Service State Mtce Connected Ports Busy 0010/001 T00046 in-service/idle no 0010/002 T00047 in-service/active no T0051 0010/003 T00048 in-service/idle no 0010/004 T00049 in-service/idle no 0010/005 T00050 in-service/idle no 0010/006 T00051 in-service/active no T0047 0010/007 T00052 in-service/idle no 0010/008 T00053 in-service/idle no 0010/009 T00054 in-service/idle no 0010/010 T00055 in-service/idle no</pre>

Step	Description
2.	<p>Enter status trunk <m>, where m is the member in active state as noted in the previous step for verification of codec used and shuffling status:</p> <ul style="list-style-type: none"> • Codec – The codec used for Audio is G.711MU in this example. • Shuffling - If the Near-end IP Addr and Far-end IP Addr for Audio are using the same port and the Audio Connection Type is ip-direct, it signifies that shuffling was successful. In this example, shuffling was successful.
	<pre> status trunk 10/2 Page 1 of 2 TRUNK STATUS Trunk Group/Member: 0010/002 Service State: in-service/active Port: T00047 Maintenance Busy? No Signalling Group ID: Connected Ports: T0051 Port Near-end IP Addr : Port Far-end IP Addr : Port Signaling: 01A0617 192. 45.100.147 : 5061 192. 45. 52.160 : 5061 G.711MU Audio: 192. 45. 53.101 : 34008 192. 45. 53.102 : 34008 Video: Video Codec: Authentication Type: None Audio Connection Type: ip-direct </pre>

Step	Description
3.	<p>Select the STATUS tab at the Grandstream Device Configuration screen and verify the following:</p> <ul style="list-style-type: none"> • Verify the IP Address is correct. • Verify the Software Version is correct. • Verify the Accounts configured in Section 5, Step 3 are registered with Avaya SES. <div data-bbox="280 417 1422 982" style="border: 1px solid black; padding: 10px; margin: 10px 0;"> <p style="text-align: center; color: #0056b3;">Grandstream Device Configuration</p> <p style="text-align: center; background-color: #e0e0e0; margin: 0;">STATUS BASIC SETTINGS ADVANCED SETTINGS ACCOUNT 1 ACCOUNT 2 ACCOUNT 3 ACCOUNT 4 ACCOUNT 5 ACCOUNT 6</p> <p>MAC Address: 00:0B:82:12:12:42</p> <p>IP Address: 192.45.53.80</p> <p>Product Model: GXP2020</p> <p>Software Version: Program-- 1.1.5.15 Bootloader-- 1.1.5.6</p> <p>System Up Time: 0 day(s) 21 hour(s) 51 minute(s)</p> <p>Registered: Account 1 Yes Account 2 Yes Account 3 No Account 4 No Account 5 No Account 6 No</p> <p>PPPoE Link Up: disabled</p> <p style="font-size: small; text-align: center;">All Rights Reserved Grandstream Networks, Inc. 2004-2007</p> </div>

8. Support

For technical support on Grandstream Networks telephones, consult the support pages at http://www.grandstream.com/contact_us.html or contact Grandstream Networks technical support at:

- Telephone: 1- (617) 566 9300
- E-mail: support@grandstream.com

9. Conclusion

These Application Notes describe a solution comprised of Avaya Communication Manager 4.0.1, Avaya SES 4.0, and Grandstream Networks SIP telephones. Grandstream GXP2020 and GXP1200 are SIP-based VoIP telephones. Grandstream GXP2020 telephone is typically used in an enterprise or small business environment and Grandstream GXP1200 telephone is used by residential or SoHo users. During compliance testing, Grandstream telephones successfully registered with Avaya SES, placed and received calls to and from SIP and non-SIP telephones, and executed other telephony features like three-way conference, transfers, hold, etc. The objective of **Section 6.1** were met with some exceptions noted in **Section 6.2**.

10. Additional References

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

[1] *Administrator Guide for Avaya Communication Manager*, Issue 2.1, May 2006, Document Number 03-300509

[2] *Administration for Network Connectivity for Avaya Communication Manager*, Issue 11, February 2006, Document Number 555-233-504

[3] *SIP Support in Release 3.1 of Avaya Communication Manager*, Issue 6, February 2006, Document Number 555-245-206

[4] *Installing and Administering SIP Enablement Services R4.0*, Issue 2.0, August 2006, Document Number 03-600768

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

[5] Grandstream GXP2020 user manual GXP2020UsersManual.pdf

[6] Grandstream GXP1200 user manual GXP1200UserManual.pdf

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