



Avaya Solution & Interoperability Test Lab

Application Notes for Grandstream GXP280, GXP1200, and GXP2020 SIP Telephones with Avaya Aura™ Communication Manager and Avaya Aura™ Session Manager – Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Aura™ Communication Manager, Avaya Aura™ Session Manager, and Grandstream GXP280/1200/2020 SIP Telephones. During compliance testing, the Grandstream SIP Telephones successfully registered with Session Manager, placed and received calls to and from SIP and non-SIP telephones, and executed other telephony features such as conference, transfer, and hold.

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 a solution comprised of Avaya Aura™ Communication Manager, Avaya Aura™ Session Manager, and Grandstream GXP280/1200/2020 SIP Telephones. Communication Manager and Session Manager have the capability to extend advanced telephony features to SIP stations. These features can be extended to non-Avaya SIP telephones such as the Grandstream SIP Telephones.

1.1. Interoperability Compliance Testing

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

1.2. Support

For technical support on Grandstream SIP Telephones, contact Grandstream's technical support at their website:

<http://www.grandstream.com/support/contactsupport.html>

The website provides contact telephone numbers, as well as access to their ticketing system and online forums.

2. Reference Configuration

Figure 1 illustrates a sample configuration consisting of Communication Manager running on an Avaya S8510 Server with a G450 Media Gateway, serving as a Feature Server for SIP endpoints. An Avaya S8720 Server with a G650 Media Gateway serves as an Access Element supporting H.323 and Digital Telephones. A Session Manager interconnects these two systems via SIP trunks and acts as a Registrar/Proxy for SIP telephones. An Avaya Aura™ System Manager is used to configure the Session Manager.

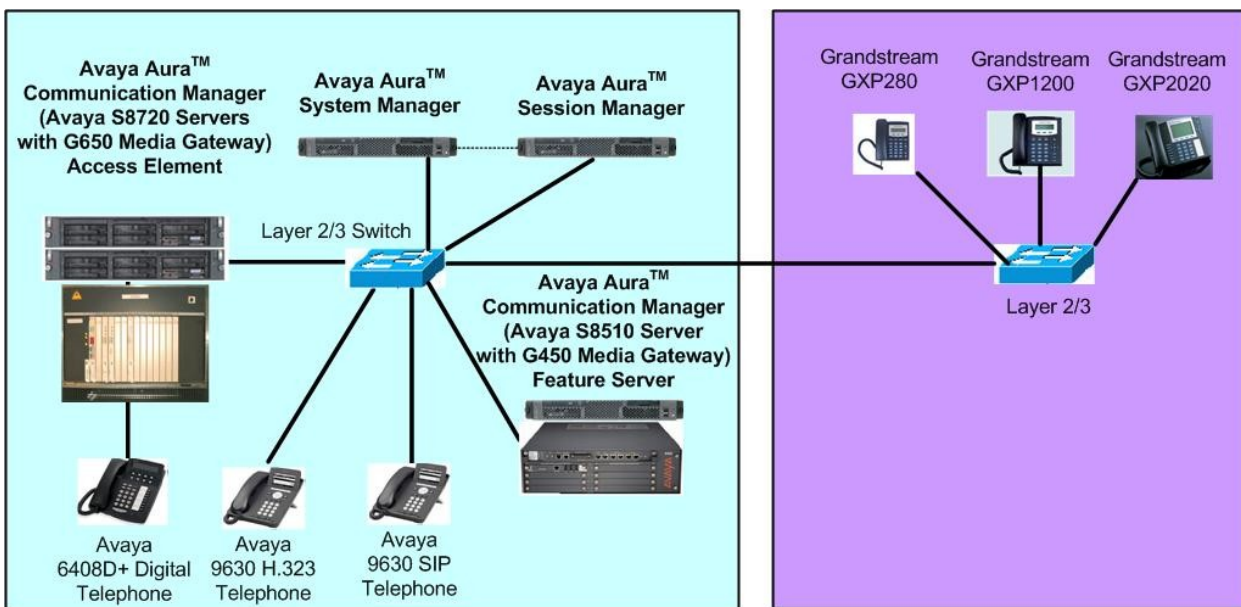


Figure 1: Sample Configuration

The Grandstream SIP Telephone originates a call by sending a call request (SIP INVITE message) to the Session Manager, which then routes the call over a SIP trunk to Communication Manager (Feature Server) for origination services. If the call is destined for another local SIP telephone, then Communication Manager routes the call back over the SIP trunk to Session Manager for delivery to the destination SIP telephone. If the call is destined for an H.323 or Digital Telephone, then Communication manager routes the call back to Session Manager for delivery to the Communication Manager (Access Element) supporting H.323 and Digital endpoints.

For a call arriving at Communication Manager (Feature Server) that is destined for the Grandstream SIP Telephones, Communication Manager routes the call over the SIP trunk to the Session Manager for delivery to the Grandstream SIP Telephone.

These application notes assume that Communication Manager and Session Manager 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 the appropriate document in the reference section at the end of this document.

3. Equipment and Software Validated

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

Equipment	Software / Firmware
S8720 Servers with G650 Media Gateway	Avaya Aura™ Communication Manager 5.2.1, Load R015x.02.1.016.4, Update 17774
S8510 Server with G450 Media Gateway	Avaya Aura™ Communication Manager 5.2.1, Load R015x.02.1.016.4, Update 17774
S8510 Server	Avaya Aura™ Session Manager 5.2.1 Load 5.2.1.1.521012
S8510 Server	Avaya Aura™ System Manager 5.2 Load 5.2.1.0.521001
Avaya 9630 IP Telephone (SIP)	2.5.0
Avaya 4620 IP Telephone (H.323)	2.9.1
Avaya 6408D+ Digital Telephone	-
Grandstream GXP 280	1.2.2.19
Grandstream GXP 1200	1.2.2.19*
Grandstream GXP 2020	1.2.2.19*

- * Test load 0.1.21.1 was used to verify an issue with a Long Hold Timeout and also with using TCP as the SIP transport. Very limited testing was done with TCP (just basic calls). All testing was completed using UDP.

4. Configure Communication Manager Feature Server

This section describes a procedure for setting up a SIP trunk between the Communication Manager serving as a Feature Server, and Session Manager. This includes steps for setting up a list of IP codecs, 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 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.

These steps are performed from the Communication Manager System Access Terminal (SAT) interface. Grandstream and other SIP telephones are configured as Off-PBX Stations (OPS) in Communication Manager. Communication Manager does not directly control an OPS endpoint, but its features and calling privileges can be applied to it by associating a local extension with the OPS endpoint. Similarly, a SIP telephone in Session Manager is associated with an extension on Communication Manager. SIP telephones register with Session Manager and use Communication Manager for call origination and termination services, including Feature Name Extensions (FNEs). Enter the **save translation** command after completing this section.

4.1. Capacity Verification

Step	Description
1.	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.
	<pre> display system-parameters customer-options Page 1 of 10 OPTIONAL FEATURES G3 Version: V15 Software Package: Standard Location: 1 RFA System ID (SID): 1 Platform: 12 RFA Module ID (MID): 1 USED Platform Maximum Ports: 44000 200 Maximum Stations: 450 60 Maximum XMOBILE Stations: 0 0 Maximum Off-PBX Telephones - EC500: 10 0 Maximum Off-PBX Telephones - OPS: 200 55 Maximum Off-PBX Telephones - PBFMC: 0 0 Maximum Off-PBX Telephones - PVFMC: 0 0 Maximum Off-PBX Telephones - SCCAN: 0 0 </pre>

2. Proceed to **Page 2** of **OPTIONAL FEATURES** form. Verify that the number of **Maximum Administered 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.

Note: Each SIP call between two SIP endpoints requires four SIP trunks for the duration of the call. The license file installed on the system controls the maximum permitted.

```
display system-parameters customer-options                               Page 2 of 10
                                OPTIONAL FEATURES
```

IP PORT CAPACITIES		USED
Maximum Administered H.323 Trunks:	450	100
Maximum Concurrently Registered IP Stations:	18000	1
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:	5	0
Maximum Video Capable H.323 Stations:	0	0
Maximum Video Capable IP Softphones:	5	0
<u>Maximum Administered SIP Trunks:</u>	<u>300</u>	40
Maximum Administered Ad-hoc Video Conferencing Ports:	0	0
Maximum Number of DS1 Boards with Echo Cancellation:	0	0
Maximum TN2501 VAL Boards:	10	0
Maximum Media Gateway VAL Sources:	5	0
Maximum TN2602 Boards with 80 VoIP Channels:	128	0
Maximum TN2602 Boards with 320 VoIP Channels:	128	0

4.2. IP Codec Set

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

Step	Description
1.	<p>Enter the change ip-codec-set n command, where n is a number between 1 and 7, inclusive. IP codec sets are used in Section 4.3 for configuring an IP network region to specify which codec sets may be used within and between network regions. For the compliance testing, G.722-64K, G.711MU, and G.729AB were used and Media Encryption was set to none. If only one codec should be used, then only specify the one that is to be used. Note that for G.729 interoperability between Avaya 9630 SIP Telephones, Avaya 4600 H.323 Telephones, and Grandstream phones, the G.729AB codec should be used, and the configuration file settings for the 9600 SIP Telephone should include the line: SET ENABLE_G729 "2".</p> <pre> change ip-codec-set 7 Page 1 of 2 IP Codec Set Codec Set: 7 Audio Silence Frames Packet Codec Suppression Per Pkt Size(ms) 1: G.722-64K n 2 20 2: G.711MU n 2 20 3: G.729AB n 2 20 4: 5: 6: 7: Media Encryption 1: none 2: 3: </pre>

4.3. IP Network Region

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

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 as shown in the display screen below:</p> <ul style="list-style-type: none"> • Authoritative Domain – Set to avaya.com in this example. This should match the SIP Domain value configured in Session Manager. • Intra-region IP-IP Direct Audio – Set to yes to allow direct IP-to-IP audio connectivity between endpoints registered to Communication Manager or Session Manager in the same IP network region. • Inter-region IP-IP Direct Audio – Set to yes to allow direct IP-to-IP audio connectivity between endpoints registered to Communication Manager or Session Manager in different IP network regions. • Codec Set – Set the codec set number as provisioned in Section 4.2. <pre> display ip-network-region 1 Page 1 of 19 IP NETWORK REGION Region: 1 Location: 1 Name: Company X Media Parameters: Authoritative Domain: avaya.com Intra-region IP-IP Direct Audio: yes Codec Set: 7 Inter-region IP-IP Direct Audio: yes UDP Port Min: 2048 UDP Port Max: 65535 IP Audio Hairpinning? y DIFFSERV/TOS Parameters: Call Control PHB Value: 46 Audio PHB Value: 46 Video PHB Value: 26 RTCP Reporting Enabled? y RTCP Monitor Server Parameters Use Default Server Parameters? y 802.1P/Q Parameters: Call Control 802.1p Priority: 6 Audio 802.1p Priority: 6 Video 802.1p Priority: 5 AUDIO RESOURCE RESERVATION PARAMETERS RSVP Enabled? n H.323 IP Endpoints: H.323 Link Bounce Recovery? y Idle Traffic Interval (sec): 20 Keep-Alive Interval (sec): 5 Keep-Alive Count: 5 </pre>

2.	<p>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 is automatically set to the IP codec set entered in Step 1.</p>
<pre> display ip-network-region 1 Source Region: 1 Inter Network Region Connection Management I M WAN-BW-limits Video Intervening Dyn A G a dst codec direct WAN Units Total Norm Prio Shr Regions CAC R L s 1 7 2 3 4 5 6 7 8 9 10 11 12 13 14 15 Page 3 of 19 </pre>	

4.4. IP Node Names

This section describes the steps for administering a node name in Communication Manager for Session Manager to be used in the configuration of the SIP signaling group.

Step	Description
1.	<p>Use the change node-names ip command to add a new node name for Session Manager.</p> <pre> change node-names ip IP NODE NAMES Name IP Address SM1 10.1.2.170 default 0.0.0.0 procr 10.1.2.160 Page 1 of 2 </pre>

4.5. SIP Signaling

This section describes the steps for administering a signaling group in Communication Manager for communication between Communication Manager and Session Manager.

Step	Description
1.	<p>Enter the command add signaling-group n, where n is an available signaling group and configure the following as shown in the display screen below:</p> <ul style="list-style-type: none"> • Group Type – Set to sip. • Transport Method – Set to tls. • IMS Enabled – Set to y. • Near-end Node Name - Set to procr. • Near-end Listen Port - Defaults to 5061 for TLS. • Far-end Node Name - Set to the node name configured in Section 4.4. • Far-end Listen Port - Defaults to 5061 for TLS. • Far-end Network Region - Set to the Region configured in Section 4.3. • Far-end Domain - Set to avaya.com in this example. This should match the SIP Domain value configured in Session Manager. • Direct IP-IP Audio Connection – Set to y.
	<pre> display signaling-group 60 SIGNALING GROUP Group Number: 60 Group Type: sip Transport Method: tls IMS Enabled? y Near-end Node Name: procr Far-end Node Name: SM1 Near-end Listen Port: 5061 Far-end Listen Port: 5061 Far-end Network Region: 1 Far-end Domain: avaya.com Incoming Dialog Loopbacks: eliminate Bypass If IP Threshold Exceeded? n RFC 3389 Comfort Noise? n Direct IP-IP Audio Connections? y DTMF over IP: rtp-payload IP Audio Hairpinning? n Session Establishment Timer(min): 3 Enable Layer 3 Test? n Direct IP-IP Early Media? n H.323 Station Outgoing Direct Media? n Alternate Route Timer(sec): 6 </pre>

4.6. SIP Trunking

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

Step	Description
1.	<p>Issue the command add trunk-group n, where n is an unallocated trunk group and configure the following as shown in the display screen below:</p> <ul style="list-style-type: none"> • Group Type – Set to the Group Type field to sip. • Group Name – Enter any descriptive name. • TAC (Trunk Access Code) – Set to any available trunk access code. • Signaling Group – Set to the Group Number field value configured in Section 4.5. (i.e., 60) • 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. <p>Note: Each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunk members for the duration of the call. The license file installed on the system controls the maximum permitted.</p>
	<pre> display trunk-group 60 Page 1 of 21 TRUNK GROUP Group Number: 60 Group Type: sip CDR Reports: y Group Name: SMI COR: 1 TN: 1 TAC: 160 Direction: two-way Outgoing Display? n Night Service: Dial Access? n Queue Length: 0 Service Type: tie Auth Code? n Signaling Group: 60 Number of Members: 20 </pre>

4.7. SIP Stations

This section describes the steps for administering OPS stations in Communication Manager and associating the OPS station extensions with the telephone numbers of the Grandstream SIP telephones. The configuration is the same for all phones except for the number of call appearances as detailed in Step 3.

Step	Description
1.	<p>Enter the add station n command, where n is an available extension in the dial plan, to administer an OPS station. On Page 1 of the STATION form configure the following fields as shown in the display screen below:</p> <ul style="list-style-type: none"> • Type – Set to 9630SIP. • Port – Set to X. (Once the form is submitted, a virtual port is assigned, e.g., S00022) • Name – Enter any descriptive name.
	<pre> display station 30031 Page 1 of 6 STATION Extension: 30031 Lock Messages? n BCC: 0 Type: 9630SIP Security Code: TN: 1 Port: S00022 Coverage Path 1: 60 COR: 1 Name: Grand2020-LD Coverage Path 2: COS: 1 Hunt-to Station: STATION OPTIONS Time of Day Lock Table: Message Lamp Ext: 30031 Loss Group: 19 Button Modules: 0 Display Language: english Survivable COR: internal Survivable Trunk Dest? y IP SoftPhone? n </pre>

2. Proceed to **Page 2** of the **STATION** form and set **Direct IP-IP Connections** to **y**. Set **MWI Served User Type** to **sip-adjunct**.

```

display station 30031                               STATION                               Page 2 of 6
                                                    STATION
FEATURE OPTIONS
  LWC Reception: spe
  LWC Activation? y
  Coverage Msg Retrieval? y
  Auto Answer: non
  CDR Privacy? n
  Data Restriction? n
  Idle Appearance Preference? n
  Bridged Idle Line Preference? n
  Bridged Call Alerting? n
  Active Station Ringing: single
  H.320 Conversion? n
  Per Station CPN - Send Calling Number?
  EC500 State: enabled
  MWI Served User Type: sip-adjunct
  Coverage After Forwarding? s
  Direct IP-IP Audio Connections? y
  Emergency Location Ext: 30031
  Always Use? n IP Audio Hairpinning? n
  Precedence Call Waiting? y

```

3. Proceed to **Page 4** of the **STATION** form and add the required number of **call-appr** entries in the **BUTTON ASSIGNMENTS** section. The number of call appearances depends on the number supported by each telephone. For the GXP 2020, 3 call appearances were configured. It supports up to 6. The GXP 1200 supports 2. The GXP 280 supports one line, and a second virtual line that can be used to initiate a conference or transfer by the GXP280 user by use of the flash button. Two line appearances must be configured in Communication Manager for this to work.

```

display station 30031                               STATION                               Page 4 of 6
                                                    STATION
SITE DATA
  Room:
  Jack:
  Cable:
  Floor:
  Building:
  Headset? n
  Speaker? n
  Mounting: d
  Cord Length: 0
  Set Color:
ABBREVIATED DIALING
  List1:
  List2:
  List3:
BUTTON ASSIGNMENTS
  1: call-appr
  2: call-appr
  3: call-appr
  4:
  5:
  6:
  7:
  8:

```

4.	<p>Enter the add off-pbx-telephone station-mapping command and configure the following as shown in the change screen below:</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 SIP telephone will use for registration and call termination. In the example below, the Phone Number is the same as the Station Extension, though it is not required to be the same. • Trunk Selection – Set to aar. In this case aar is being used to route the calls.
<pre>change off-pbx-telephone station-mapping 30031 Page 1 of 3 STATIONS WITH OFF-PBX TELEPHONE INTEGRATION Station Application Dial CC Phone Number Trunk Config Dual Extension Prefix Selection Set Mode 30031 OPS - 30031 aar 1</pre>	
5.	Repeat Steps 1 - 4 as necessary to administer additional OPS stations and associations for the Grandstream SIP telephones.

4.8. Routing

AAR routing was used in this configuration. Different routing procedures can also be implemented.

Step	Description
1.	<p>Enter the change aar analysis n command, where n is the number to be routed; in this case 300 (matching any extensions starting with 300xx). On Page 1 of the form configure the following fields as shown in the screen below:</p> <ul style="list-style-type: none"> • Dialed String – Set to 300. • Total Min/Max – Set to 5 • Route Pattern - Set to the appropriate route pattern, in this case 60. • Call Type – Set to aar.
<pre>change aar analysis 3 Page 1 of 2 AAR DIGIT ANALYSIS TABLE Location: all Percent Full: 0 Dialed Total Route Call Node ANI String Min Max Pattern Type Num Reqd 300 5 5 60 aar n</pre>	

2.

Enter the **change route-pattern n** command, where n is the route-pattern to be configured, in this case **60**.

On Page 1 of the form configure the following fields as shown in the screen below:

- **Pattern name** – Set to an appropriate name.
- **Grp No** – Set to the trunk group being used, in this case **60**.
- **FRL** – Set to **0** (lowest restriction, or a higher number if appropriate).
- **No. Del Dgts** - Set to **0** (all digits are being sent).

```
change route-pattern 60                                     Page 1 of 3
                Pattern Number: 60  Pattern Name: SM FS
                SCCAN? n           Secure SIP? n
  Grp  FRL  NPA Pfx Hop Toll No.  Inserted           DCS/ IXC
  No   0    0   0  0  0  0  0  Del  Digits           QSIG
                Dgts
1: 60  0           0           Intw
2:                                     n  user
3:                                     n  user
4:                                     n  user
5:                                     n  user
6:                                     n  user

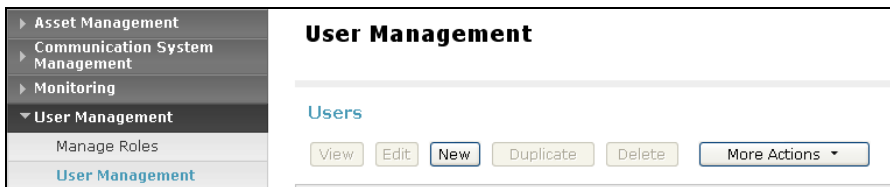
  BCC VALUE  TSC CA-TSC  ITC BCIE Service/Feature PARM No. Numbering LAR
  0 1 2 M 4 W      Request      Dgts Format
                Subaddress
1: y y y y y n n           rest           none
2: y y y y y n n           rest           none
3: y y y y y n n           rest           none
4: y y y y y n n 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

5. Configure Session Manager

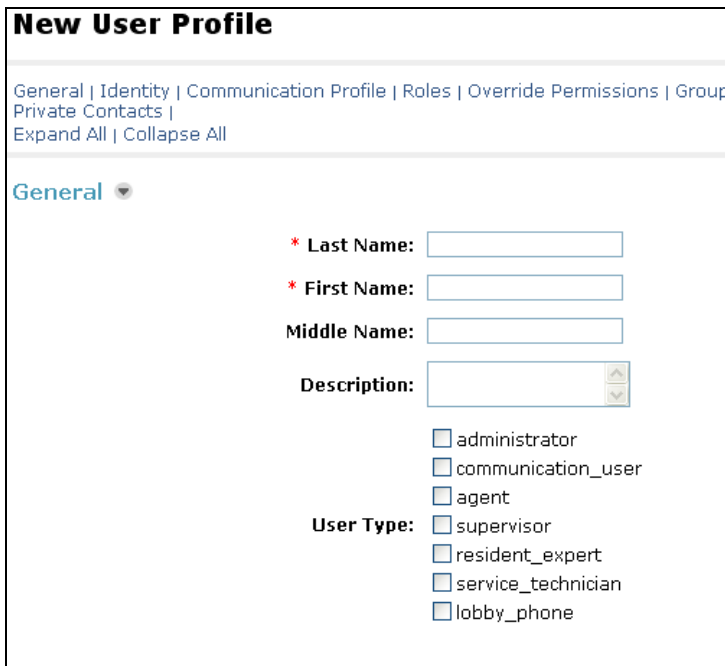
This section will describe the administration of SIP endpoints in Session Manager. It is assumed that a trunk has already been provisioned that matches the Communication Manager configuration in Sections 4.5 and 4.6. For additional references in configuring SIP trunking between Communication Manager and Session Manager see [4] and [6]. The following screens will show a sample configuration for a Grandstream endpoint, whose extension is 30031. The same procedure needs to be followed for all endpoints.

Session Manager is configured via System Manager. Use a web browser and enter “https://<ip-address>/SMGR, where <ip-address> is the IP address of System Manager

Log in using the appropriate credentials. On the main configuration page, select “User Management” under “User Management”, and click “New” to administer a new endpoint.



This will create a new User Profile. In the “General” section, enter a “Last Name” and “First Name”. Note that fields marked with * are required to be filled in.

The screenshot shows the 'New User Profile' form. At the top, there are navigation links: General | Identity | Communication Profile | Roles | Override Permissions | Group Private Contacts | Expand All | Collapse All. The 'General' section is expanded and contains the following fields:

- * Last Name: [text input]
- * First Name: [text input]
- Middle Name: [text input]
- Description: [text area]
- User Type: [checkboxes for administrator, communication_user, agent, supervisor, resident_expert, service_technician, lobby_phone]

The following screen shows what was entered for endpoint 30031.

General ▾

Last Name:

First Name:

Middle Name:

Description:

User Type:

- administrator
- communication_user
- agent
- supervisor
- resident_expert
- service_technician
- lobby_phone

In the “Identity” section, enter a “Login Name”, for example 30031@avaya.com, and the required passwords. Note that the “Shared Communication Profile” password is the one the endpoint is required to use when registering to Session Manager, therefore you must enter one. It’s also recommended to enter the display names. The “Localized Display Name” is what is displayed on a telephone when a call is made. “SMGR Login Password”, while required, was not used in this sample configuration, and can be any value.

Identity ▾

*** Login Name:**

*** Authentication Type:** ▾

SMGR Login Password:

*** Password:**

*** Confirm Password:**

Shared Communication Profile Password:

Confirm Password:

Localized Display Name:

Endpoint Display Name:

Honorific :

Language Preference: ▾

Time Zone:

The information below is what was entered for endpoint 30031. Note that the passwords are not displayed when viewing an endpoint’s configuration.

Identity ▾

Login Name: 30031@avaya.com

Authentication Type: Basic ▾

Source: local

Localized Display Name: Grand2020-LD

Endpoint Display Name: Grand2020-ED

Honorific :

Language Preference: English ▾

Time Zone:

In the “Communication Profile” section, there are three sub-sections that need to be populated: “Communication Address”, “Session Manager”, and “Station Profile”. Clicking on the arrow next to “Communication Profile” lets you see the other sections.

Communication Profile ▾

Name
Primary

Select : None

*** Name:**

Default:

Communication Address ▾

Type	SubType	Handle	Domain
sip	username	30031	avaya.com

Session Manager ▾

Session Manager Instance

Origination Application Sequence

Termination Application Sequence

Station Profile ▾

Click “New” under “Communication Address”, set “Subtype” to “username”, and fill in the extension portion of the “Fully Qualified Address”, e.g., “30031”. “@avaya.com” will be automatically filled in. Then click “Add”. This will move the entry to the table as shown in the previous screen.

Communication Address ▼

New Edit Delete

	Type	SubType	Handle	Domain
No Records found				

Type: sip ▼

SubType: ▼

* Fully Qualified Address: @

Add Cancel

Click on the box next to “Session Manager”, and select the appropriate “Session Manager” from the list. Select the appropriate “Origination and Termination Application Sequence”.

Session Manager ▼

* Session Manager Instance Select ▼

Origination Application Sequence (None) ▼

Termination Application Sequence (None) ▼

The screen below shows what was used for endpoint 30031.

Session Manager ▼

Session Manager Instance SM1

Origination Application Sequence CM FS App Sequence

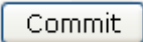
Termination Application Sequence CM FS App Sequence

Click on the box next to “Station Profile”, and enter the appropriate “System”, this is the Communication Manager Feature Server supporting the endpoint. Check mark “Use Existing Stations”, this will use the station previously entered in Communication Manager. Note that leaving this field un-checked will force System Manager to attempt to create the station in Communication Manager. Enter an “Extension”, and select the appropriate “Template”. Leave the “Security Code” blank. Select “IP” for the “Port” field.

The screenshot shows a configuration form for a station profile. At the top left, there is a checked checkbox labeled "Station Profile" with a dropdown arrow. Below this, the "System" field is a dropdown menu showing "S8510-FS". The "Use Existing Stations" checkbox is checked. The "Extension" field is a text input containing "30031". The "Template" field is a dropdown menu showing "DEFAULT_9630SIP". The "Set Type" field is a text input containing "9630SIP". The "Security Code" field is an empty text input. The "Port" field is a dropdown menu showing "IP". At the bottom, there is a checkbox labeled "Delete Station on Unassign of Station from User" which is unchecked.

The screen below shows what was used for endpoint 30031.

The screenshot shows the same configuration form as above, but with some changes. The "System" field is still "S8510-FS". The "Extension" field is "30031". The "Set Type" field is "9630SIP". The "Security Code" field is empty. The "Port" field is now a text input containing "S00020". The "Delete Station on Unassign of Station from User" checkbox remains unchecked.

When done click  at the bottom of the web page. Repeat the above steps for each endpoint to be configured.

6. Configure Grandstream SIP Telephones

This section describes the basic configuration of the Grandstream SIP Telephones. For additional details, see [8] available at <http://www.grandstream.com/>.

Three SIP telephones were tested: Grandstream GXP 280, Grandstream GXP 1200, and Grandstream GXP 2020. The configuration was done using the web interface. Some configuration can also be done on the telephone itself.

The configuration steps are similar for all three telephones, the main difference being the number of accounts or line appearances that each telephone supports. The GXP 2020 supports up to 6 lines, only three were used in the testing. The GXP 1200 supports 2 lines. The GXP 280 supports one line, and a second virtual line that can be used to initiate a conference or transfer by the GXP280 user, by use of the flash button. Two line appearances must be configured in Communication Manager for this to work. The user can be active on a call, hit the “Flash” button, dial a second number, then hit the “Conference” or “Transfer” buttons as appropriate. Make sure the number of lines used matches what is configured in Communication Manager.

The steps below show the configuration screens for the GXP 2020.

Access the GXP 2020 configuration screen by entering `http://<ip-address>` in a browser, where ip-address is the IP address of the telephone. This can be statically configured on the phone, or via DHCP. A static IP address was used in this testing. Enter the appropriate credentials to log in to the phone. The initial IP Address can be configured on the phone by navigating through the Menu keys (Menu → Config → Network).

The “Basic Settings” tab allows selection of DHCP or Static IP information. It also provides fields for configuring “Multi purpose Keys”. The screen below shows a configuration for a “Call Pickup FNE” button.

Grandstream Device Configuration

STATUS	BASIC SETTINGS	ADVANCED SETTINGS	EXT 1	EXT 2
ACCOUNT 1	ACCOUNT 2	ACCOUNT 3	ACCOUNT 4	ACCOUNT 5

End User Password: (purposely not displayed for security protection)

IP Address: dynamically assigned via DHCP (default) or PPPoE
(will attempt PPPoE if DHCP fails and following is non-blank)

PPPoE account ID:

PPPoE password:

Host name
(Option 12):

Domain name
(Option 15):

Vendor Class ID
(Option 60):

Preferred DNS server:

statically configured as:

IP Address:

Subnet Mask:

Gateway:

DNS Server 1:

DNS Server 2:

Multi Purpose Key 1: Key Mode: Account:

Name: UserID:

The “Advanced Settings” tab was not modified during the test, other than to configure a server to update the phone’s firmware, and for logging/troubleshooting.

The “Account 1” setting has several fields that were used. The following ones are worth noting:

SIP Server: IP address of Session Manager

Outbound Proxy: IP address of Session Manager

SIP User ID: Endpoint extension

Authenticate ID: Endpoint extension

Authenticate Password: Security password assigned in Session Manager (Shared Communication Profile password)

SIP Registration: Yes

Local SIP Port: 5060

SIP Transport: UDP (All of the testing was completed with UDP. Basic testing was done with TCP, and requires a fix from Grandstream. See Section 7 for details).

STATUS	BASIC SETTINGS	ADVANCED SETTINGS	EXT 1	EXT 2
ACCOUNT 1	ACCOUNT 2	ACCOUNT 3	ACCOUNT 5	ACCOUNT 6
<p>Account Active: <input type="radio"/> No <input checked="" type="radio"/> Yes</p> <p>Account Name: <input type="text"/> (e.g., MyCompany)</p> <p>SIP Server: <input type="text" value="10.1.2.170"/> (e.g., sip.mycompany.com, or IP address)</p> <p>Outbound Proxy: <input type="text" value="10.1.2.170"/> (e.g., proxy.myprovider.com, or IP address)</p> <p>SIP User ID: <input type="text" value="30031"/> (the user part of an SIP address)</p> <p>Authenticate ID: <input type="text" value="30031"/> (can be same or different from SIP UserID)</p> <p>Authenticate Password: <input type="password"/> (not displayed for security protection)</p> <p>Name: <input type="text"/> (optional, e.g., John Doe)</p> <p>Use DNS SRV: <input checked="" type="radio"/> No <input type="radio"/> Yes</p> <p>User ID is phone number: <input checked="" type="radio"/> No <input type="radio"/> Yes</p> <p>SIP Registration: <input type="radio"/> No <input checked="" type="radio"/> Yes</p> <p>Unregister On Reboot: <input checked="" type="radio"/> No <input type="radio"/> Yes</p> <p>Register Expiration: <input type="text" value="60"/> (in minutes. default 1 hour, max 45 days)</p> <p>local SIP port: <input type="text" value="5060"/> (default 5060)</p> <p>SIP Registration Failure Retry Wait Time: <input type="text" value="20"/> (in seconds. Between 1-3600, default is 20)</p> <p>SIP T1 Timeout: <input type="text" value="1 sec"/></p> <p>SIP T2 Interval: <input type="text" value="4 sec"/></p> <p>SIP Transport: <input checked="" type="radio"/> UDP <input type="radio"/> TCP</p>				

SUBSCRIBE for MWI: Yes

Send DTMF: via RTP (RFC2833) must be checked

SUBSCRIBE for MWI: <input type="radio"/> No <input checked="" type="radio"/> Yes
SUBSCRIBE for Registration Event: <input checked="" type="radio"/> No <input type="radio"/> Yes
PUBLISH for Presence: <input checked="" type="radio"/> No <input type="radio"/> Yes
Proxy-Require: <input type="text"/>
Voice Mail UserID: <input type="text"/> (UserID for voice mail system)
Send DTMF: <input checked="" type="checkbox"/> in-audio <input checked="" type="checkbox"/> via RTP (RFC2833) <input type="checkbox"/> via SIP INFO

Communication Manager will select which codec should be used. That codec must be in the list.

Preferred Vocoder: (in listed order)	choice 1: <input type="text" value="PCMU"/>	choice 5: <input type="text" value="G.726-32"/>
	choice 2: <input type="text" value="PCMA"/>	choice 6: <input type="text" value="iLBC"/>
	choice 3: <input type="text" value="G.723.1"/>	choice 7: <input type="text" value="G.722 (wide band)"/>
	choice 4: <input type="text" value="G.729A/B"/>	choice 8: <input type="text" value="GSM"/>

7. General Test Approach and Test Results

The general test approach was to place calls to and from the Grandstream SIP telephones and exercise basic telephone features. The main objectives were to verify:

- Registration with Session Manager.
- Ability to place basic calls.
- Successful negotiation of codecs.
- Proper codec operation for G.711MU, G.722, and G.729.
- Successful shuffling for VoIP calls.
- Successful transmission of DTMF during a call.
- Successful MWI operation.
- Ability to exercise certain features: Hold, Music on Hold, Mute, Redial, Transfer, Conference, Call-Forwarding, Calling Number Block, and select FNEs.
- Proper displays.
- Failure and Recovery.
- Long duration calls (24 hours).

All test cases were successfully completed with the exception of the following observations:

- 1) There was an issue with using TCP as the SIP transport. Grandstream provided Test Load 0.1.21.1 and it corrected this problem. Very limited testing was done with TCP (just basic calls). All testing was completed using UDP as the transport.

2) There was an issue with a Long Hold Timeout, a Communication Manager feature that alerts a phone when a call has been on hold past the specified time. Grandstream provided Test Load 0.1.21.1 and it corrected this problem.

3) For shuffling to work with G.729 between the Grandstream endpoints, Avaya 9600 SIP Endpoints, and Avaya 4600 H.323 Endpoints, the codec type G.729AB had to be used, and the configuration file settings for the 9600 SIP phone had to include the line: SET ENABLE_G729 "2". This enables AnnexB support on the 9600 SIP phones. If this is not done, the call initially shuffles, but when the refresh interval expires (~90 seconds) the call is dropped by Communication Manager. Session Manager sends AnnexB=no, and the Grandstream phone responds without AnnexB, which by spec implies yes. Grandstream plans on fixing this in a future release.

4) MTU size should be kept at 1500 (the default). Changing it to a different size can cause problems with the Grandstream endpoints.

8. Verification Steps

The following steps may be used to verify the configuration:

- The web interface can be used to verify the endpoints are registered. The screen below shows three “Accounts” were registered. The endpoint’s IP address and Software Version are also shown. This information is also available directly on the Telephone.

The screenshot displays the 'Grandstream Device Configuration' web interface. At the top, there is a navigation menu with tabs for 'STATUS', 'BASIC SETTINGS', 'ADVANCED SETTINGS', 'EXT 1', and 'EXT 2'. Below the menu, there are sub-tabs for 'ACCOUNT 1', 'ACCOUNT 2', 'ACCOUNT 3', 'ACCOUNT 4', 'ACCOUNT 5', and 'ACCOUNT 6'. The main content area shows the following configuration details:

- MAC Address: 00:0B:82:1F:90:65
- IP Address: 10.3.3.213
- Product Model: GXP2020 (HW0.3B)
- Part Number: 9620000803B
- Software Version: Program-- 1.2.2.19 Bootloader-- 1.1.6.6
- System Up Time: 0 day(s) 4 hour(s) 31 minute(s)
- System Time: 4:31am
Tuesday, January 2, 1900
- Registered: Account 1 Registered
Account 2 Registered
Account 3 Registered
Account 4 Not Registered
Account 5 Not Registered
Account 6 Not Registered
- PPPoE Link Up: disabled

- Place calls to and from the SIP telephones and verify that the calls are successfully established with two-way talk path.
- From the Communication Manager (Feature Server) 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 n command, where n is the SIP trunk configured in Section 4.6. Note down the Member with Service State set to in-service/active. In this example, 0060/006 and 0060/007 are active and either member can be used to verify whether calls shuffled and which codec was used.</p> <pre> status trunk 60 Page 1 TRUNK GROUP STATUS Member Port Service State Mtce Connected Ports Busy 0060/001 T00199 in-service/idle no 0060/002 T00200 in-service/idle no 0060/003 T00201 in-service/idle no 0060/004 T00202 in-service/idle no 0060/005 T00203 in-service/idle no 0060/006 T00204 in-service/active no T00094 0060/007 T00205 in-service/active no T00063 </pre>
2.	<p>Enter status trunk n, where n 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 Type – The codec used for Audio is G.711MU in this example. • Shuffling - If the Near-end and Far-end IP addresses for Audio belong to the Grandstream SIP telephones and the Audio Connection Type is ip-direct, it signifies that shuffling was successful. In this example, shuffling was successful. <pre> status trunk 60/6 Page 2 of 3 CALL CONTROL SIGNALING Near-end Signaling Loc: 01A0017 Signaling IP Address Port Near-end: 10.1.2.160 : 5060 Far-end: 10.1.2.170 : 5060 H.245 Near: H.245 Far: H.245 Signaling Loc: H.245 Tunneled in Q.931? no Audio Connection Type: ip-direct Authentication Type: None Near-end Audio Loc: Codec Type: G.711MU Audio IP Address Port Near-end: 10.3.3.213 : 5058 Far-end: 10.3.3.215 : 5032 Video Near: Video Far: Video Port: Video Near-end Codec: Video Far-end Codec: </pre>

9. Conclusion

These Application Notes describe a solution comprised of Communication Manager, Session Manager and Grandstream SIP Telephones. During compliance testing, Grandstream SIP telephones successfully registered with Session Manager, placed and received calls to and from SIP and non-SIP telephones, and executed other telephony features like three-party conference,

transfers, hold, etc. All test cases were successfully completed, with the exception of the observations noted in section 7.

10. Additional References

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

[1] *Administering Avaya Aura™ Communication Manager*, Release 5.2, Issue 5.0, May 2009, Document Number 03-300509.

[2] *Administering Network Connectivity on Avaya Aura™ Communication Manager*, Issue 14, May 2009, Document Number 555-233-504.

[3] *SIP Support in Avaya Aura™ Communication Manager Running on Avaya S8xxx Servers*, Issue 9, May 2009, Document Number 555-245-206.

[4] *Administering Avaya Aura™ Session Manager*, Release 5.2, Issue 2.0, November 2009, Document Number 03-603324.

[5] *Avaya Aura™ Communication Manager Screen Reference*, Issue 1.0, May 2009, Document Number 03-602878.

[6] *Administering Avaya Aura™ Communication Manager as a Feature Server*, Release 5.2, Issue 1.2, January 2010, Document Number 03-603479.

[7] *Configuring 9600-Series SIP Phones with Avaya Aura™ Session Manager Release 5.2* — Issue 1.0, February 2010, Avaya Solution Interoperability Lab Application Notes.

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

[8] *GXP User Manual*, November 2009.

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