Avaya Solution & Interoperability Test Lab

Application Notes for the Grandstream Telephones with Avaya Communication Manager 3.1.2 and Avaya SIP Enablement Services 3.1.1 – Issue 0.1

Abstract

These Application Notes describe a solution comprised of Avaya Communication Manager 3.1.2, Avaya SIP Enablement Services (SES) 3.1.1, and Grandstream Networks SIP telephones. Grandstream GXP2000 and BT200 are SIP-based VoIP telephones. Grandstream GXP2000 is typically used in an enterprise or small business environment and BT200 is used by residential or SoHo users. During compliance testing, the Grandstream Telephones successfully registered with Avaya SES, placed and received calls to and from SIP and non-SIP telephones, and established conference calls. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the Developer*Connection* Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe a solution comprised of Avaya Communication Manager 3.1.2, Avaya SIP Enablement Services (SES) 3.1.1, and Grandstream Networks SIP telephones. Grandstream GXP2000 and BT200 are SIP-based VoIP telephones. Grandstream GXP2000 is typically used in an enterprise or small business environment and BT200 is used by residential or SoHo users. Grandstream GXP2000 telephone supports up to four lines, and on each line can bridge calls to establish a 3-party conference. Grandstream BT200 supports one line.

Figure 1 illustrates a sample configuration consisting of an Avaya S8710 Media Servers, an Avaya G650 Media Gateway, an Avaya SIP Enablement Services (SES) server, and the Grandstream endpoints. 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 Telephone 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 IP telephones and the PSTN.

The Grandstream endpoint 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 endpoint, such as another Grandstream telephone or an Avaya SIP telephone, then Avaya Communication Manager routes the call back over the SIP trunk to the Avaya SES server, which in turn delivers the call to the 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 to Avaya Communication Manager that is destined for the Grandstream SIP telephone, Avaya Communication Manager routes the call over the SIP trunk to the Avaya SES server, which in turn delivers the call to the Grandstream telephone.

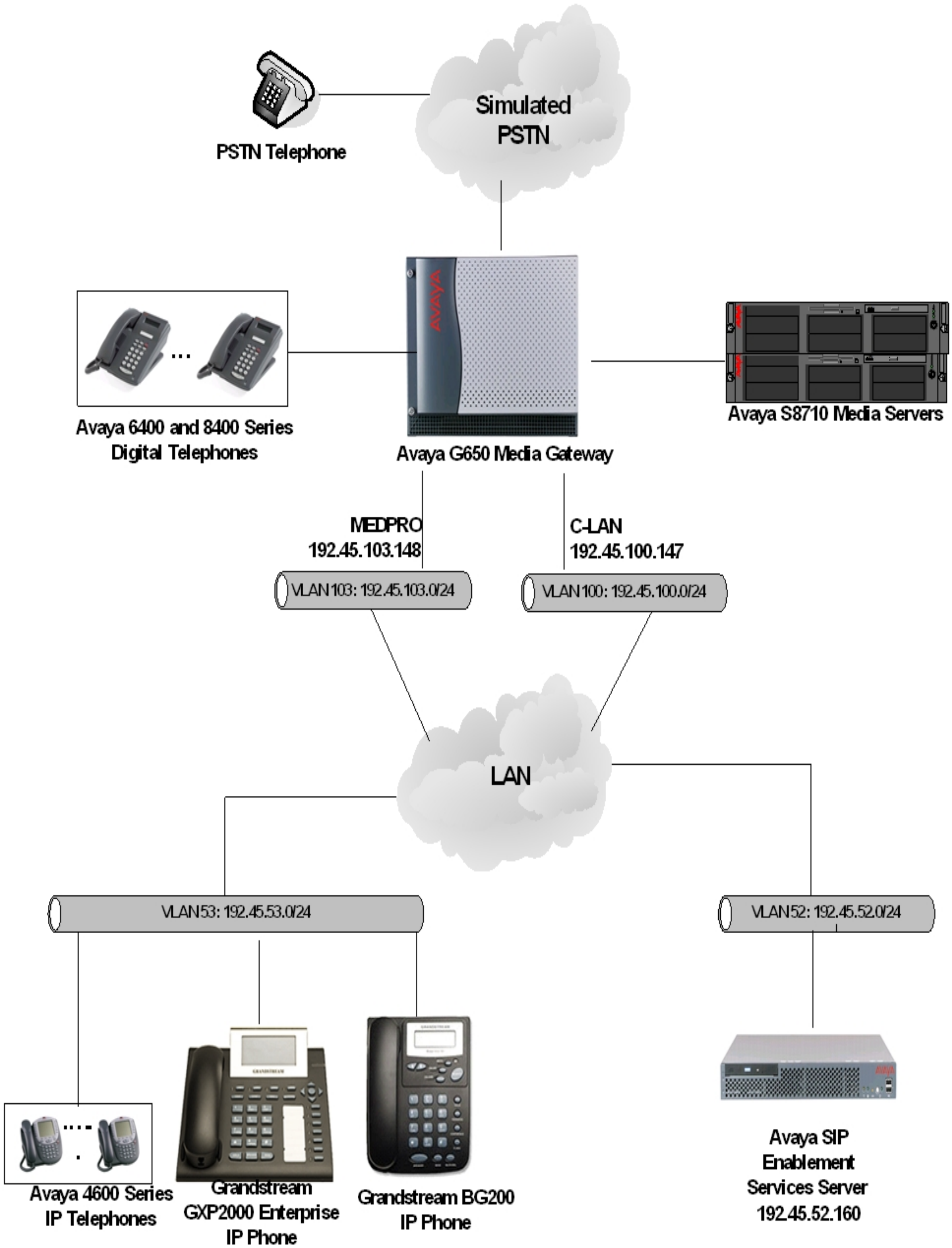


Figure 1: Sample configuration.

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 3.1.2 (R013x.01.2.632.1)
Avaya G650 Media Gateway	-
TN2312BP IP Server Interface	HW12 FW 31
TN799DP C-LAN Interface	HW01 FW 17
TN2302AP IP Media Processor	HW20 FW 112
Avaya SIP Enablement Services Server	SES 3.1.1(R03.1.1-03.1.114.0)
Avaya 4600 Series IP Telephones	2.3 (4602SW H.323) 2.5 (4625SW H.323) 2.2.3 (4610SW SIP)
Avaya 6400 and 8400 Series Digital Telephones	-
Grandstream Networks GXP2000 Telephone	1.1.2.26
Grandstream Networks BT200 Telephone	1.1.2.26
Analog Telephone	-

3. Configure Avaya Communication Manager

This section describes the steps for configuring IP codec sets and associating SIP telephone numbers with off-PBX telephone stations in Avaya Communication Manager. The steps are performed from the Avaya Communication Manager System Access Terminal (SAT) interface. IP codec sets identify the codecs that may be used in calls involving VoIP telephones. An off-PBX telephone is a phone that Avaya Communication Manager does not control, such as a cellular phone, a home telephone, or a SIP telephone. Avaya Communication Manager features and calling privileges, however, can be applied to an off-PBX telephone by associating a local, on-PBX, extension with the off-PBX telephone. This approach is taken for SIP Telephones that register with the Avaya SES server and intend to use Avaya Communication Manager for call origination and termination services. Specifically, an Administration WithOut Hardware (AWOH) on-PBX station is administered in Avaya Communication Manager and then associated with the telephone number of the SIP telephone. Similarly, on the Avaya SES server, the number of the SIP telephone is administratively associated with the extension of the on-PBX station. Throughout the rest of this document, on-PBX stations associated with SIP Telephones in such a manner will be referred to as Outboard Proxy SIP (OPS) stations.

3.1. Capacity Verification

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>
Issue the command “display system-parameters customer-options”, and proceed to Page 2. 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.

Note: 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.

<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>
</td></tr>
<tr><td>2.</td><td>
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: 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>
</td></tr>
</table>

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.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Enter the change ip-codec-set <c> command, where c is a number between 1 and 7, inclusive. IP codec sets are specified in the IP Network Region forms to define which codecs may be used within and between network regions. For the compliance testing G.711MU and G.729AB were used.

Note: Media encryption for SIP calls is currently not supported in Avaya Communication Manger, Avaya SIP telephones, and Grandstream SIP telephones.
<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></td></tr>
</table>

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 SIP Enablement Services.

Step	Description
1.	Enter the **change ip-network-region <n>** command, where **n** is a number between **1** and **250**, inclusive and administer settings as per below. • **Codec Set –** Set to **Codec Set** as provisioned in **Section 3.1**. • **Authoritative Domain –** Set to the same value as **SIP Domain** on Avaya SIP Enablement Services **Section 4, step 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.

```
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
```

Step	Description
2.	Proceed to Page 3 of the **IP NETWORK REGION** form and enable inter-region connectivity between regions as per below. For purpose of these application notes, **src rgn "2"** and **dst rgn "2"** use **codec set "2"** as configured in **Section 3.1**.

```
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                                          n
 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
```

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.	Issue the command "**change node-names ip**"; and administer settings as per below. • Add a node name for Avaya SES along with the IP address. **Note:** Verify that node-names are configured for the *C-LAN* and *MEDPRO* boards.

```
change node-names ip                                          Page 1 of 1
                                   IP NODE NAMES
    Name              IP Address
CLAN-1A06           192.45 .100.147
MEDPRO-1A13         192.45 .103.148
SES                 192.45 .52 .160
```

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.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td rowspan="3">1.</td><td>Issue the command “add signaling-group <s>”, where s is an unallocated Signaling Group; and administer settings as per below.

Group Type – Set to sip.
Transport Method – Set to tls.
Far-end Listen Port – Set to 5061(default)
Near-end Node Name - Set to CLAN IP Address as displayed in Section 3.3.
Far-end Node Name - Set to IP Address of SES configured in Section 3.3.
Far-end Network Region - Set to the region configured in Section 3.2.
Far-end Domain - Set to the Authoritative Domain configured in Section 3.2, Step 1.
</td></tr>
<tr><td><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></td></tr>
<tr><td></td></tr>
</table>

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.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Issue the command “add trunk-group <t>”, where t is an unallocated Trunk Group; and administer settings as per below.

Group Type – Set to same value as Group Type configured in Section 3.4.
TAC (Trunk Access Code) – Set to any number with 1-4 digits;* and # may be used as first digit only.
Signaling Group – Set to same value as Group Number configured in Section 3.4.
Number of Members – Set to a value between 0 and 255.
Group Name - Set a trunk group name.

Note: 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.</td></tr>
<tr><td></td><td><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></td></tr>
<tr><td></td><td></td></tr>
</table>

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 SIP telephones.

Step	Description
1.	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: • **Type –** Set to **6408D+**. • **Port –** Set to **X**. • **Name –** Set station name.

```
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: GXP2000                        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
```

Step	Description
2.	Enter the **change off-pbx-telephone station-mapping <s>** command, where <**s**> is the extension of the OPS station configured in **Step 3**. On Page 1 of the **off-pbx-telephone station-mapping** form, configure the following: • **Station Extension** – Set the extension of the OPS station. • **Application** – Set to **OPS**. • **Phone Number** – Enter the number that the Grandstream IP telephone will use for registration and call termination. In the example below, the **Phone Number** is the same as the **OPS Station Extension**, but is not required to be the same. • **Trunk Selection –** Set to the **trunk** configured in **Section 3.5**. • **Configuration Set** – Set to "**1**", which during compliance testing used the default values of the off-pbx-telephone configuration-set form.

```
change off-pbx-telephone station-mapping 54007                  Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station     Application   Dial   Phone Number      Trunk       Configuration
 Extension                 Prefix                   Selection   Set
 54007         OPS             - 54007               10         1
```

Step	Description
3.	Repeat Steps **1** and **2** as necessary to administer additional OPS stations and associations for Grandstream SIP Telephones.

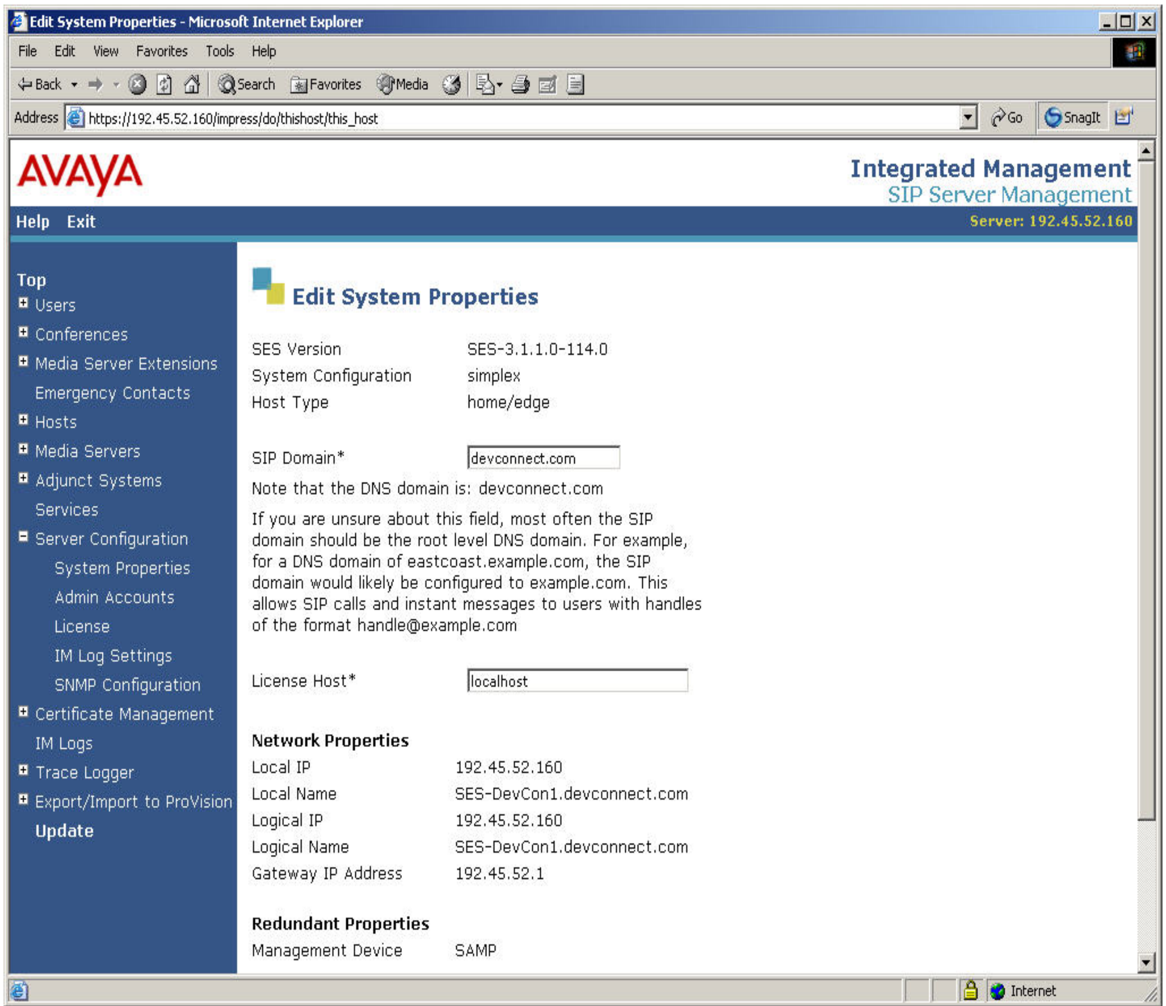
4. Configure Avaya SIP Enablement Services

This section describes the steps for creating SIP user accounts in Avaya SIP Enablement Services (SES) and associating the SIP users with an Avaya Communication Manager OPS station extension. The Grandstream Telephone will register with Avaya SES using the SIP user accounts.

This section assumes that the necessary Avaya SES configuration steps for establishing a SIP trunk with Avaya Communication Manager have been completed.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Open a web browser, enter 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.</td></tr>
<tr><td>2.</td><td>From the Administration Web Interface:

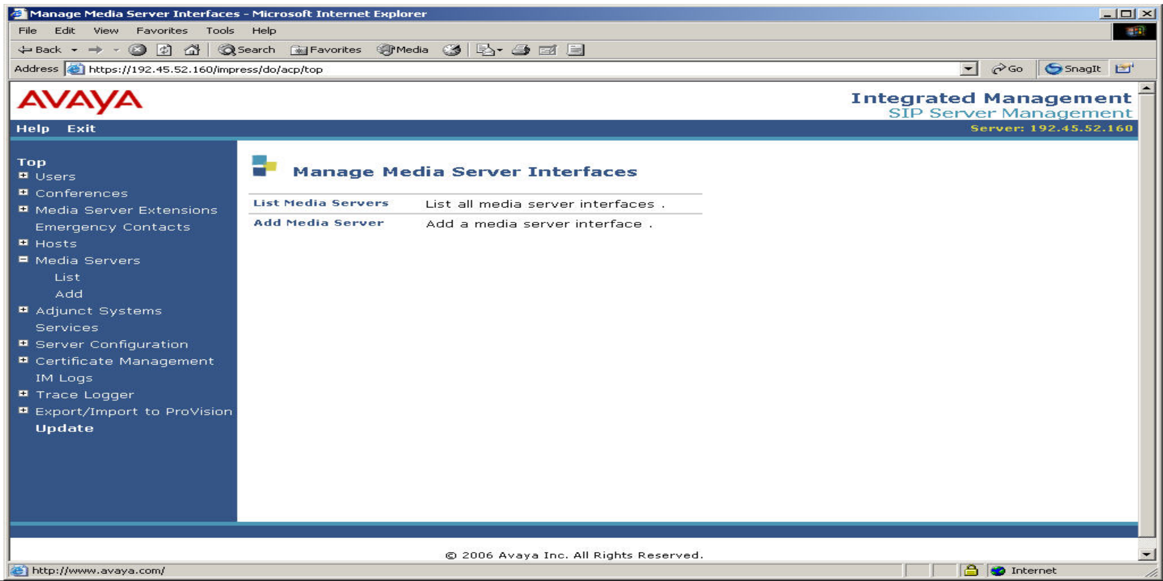
Click the + sign to expand the options under Server Configuration.
Click System Properties.
Verify the SIP Domain matches the Authoritative Domain configured for the IP NETWORK REGION on Avaya Communication Manager in Section 3.2.



</td></tr>
</table>

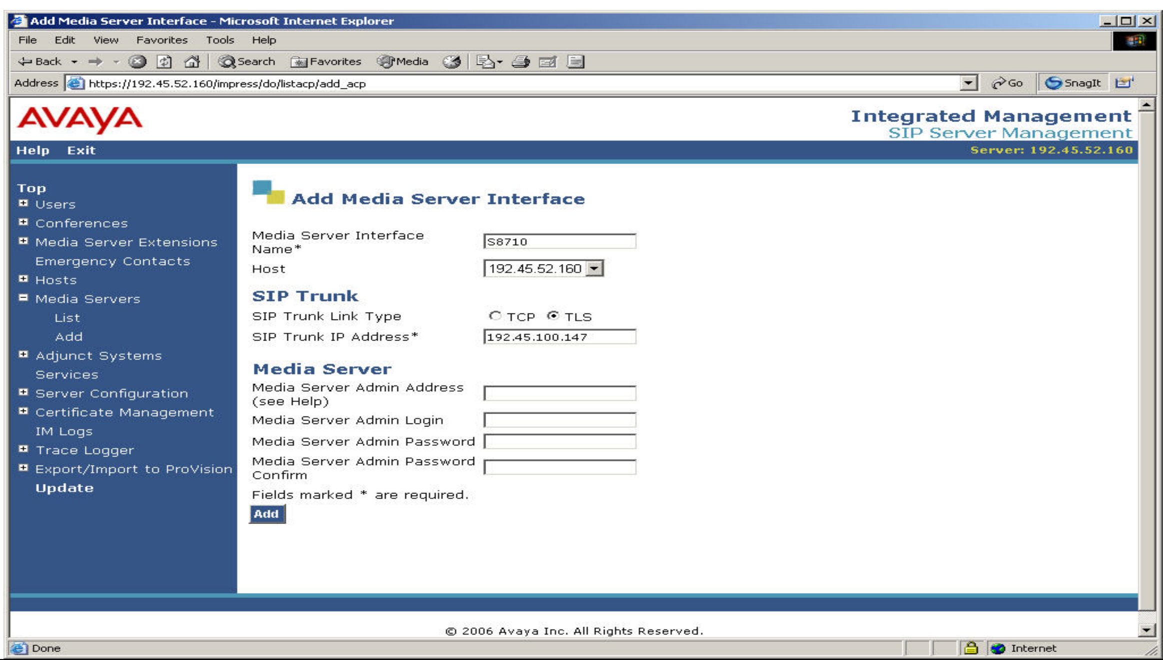
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>
To enable secure SIP trunking between Avaya SES and Avaya Communication Manager, add a Media Server corresponding to Avaya Communication Manager from the Administration Web Interface:

Click the + sign to expand the options under Media Servers.
Click Add.



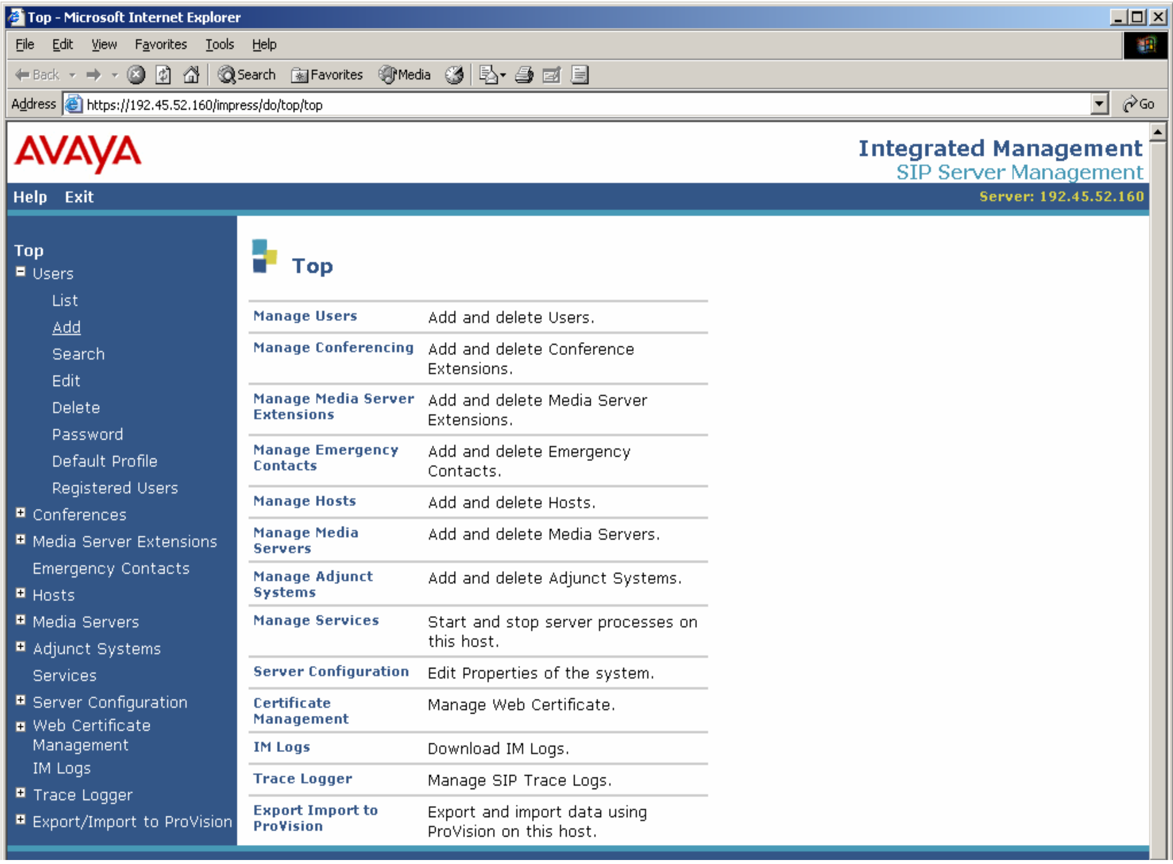
</td></tr>
<tr><td>4.</td><td>
At the Add Media Server Interface page, provision SIP Trunk parameters as follows for connectivity to Avaya Communications Manager:

SIP Trunk Link Type - Set to same value as Transport Method in Section 3.4.
SIP Trunk IP Address - Set to same value as CLAN address in Section 3.3.
Click the Add button when finished and hit the Continue button on the confirmation page [not shown].



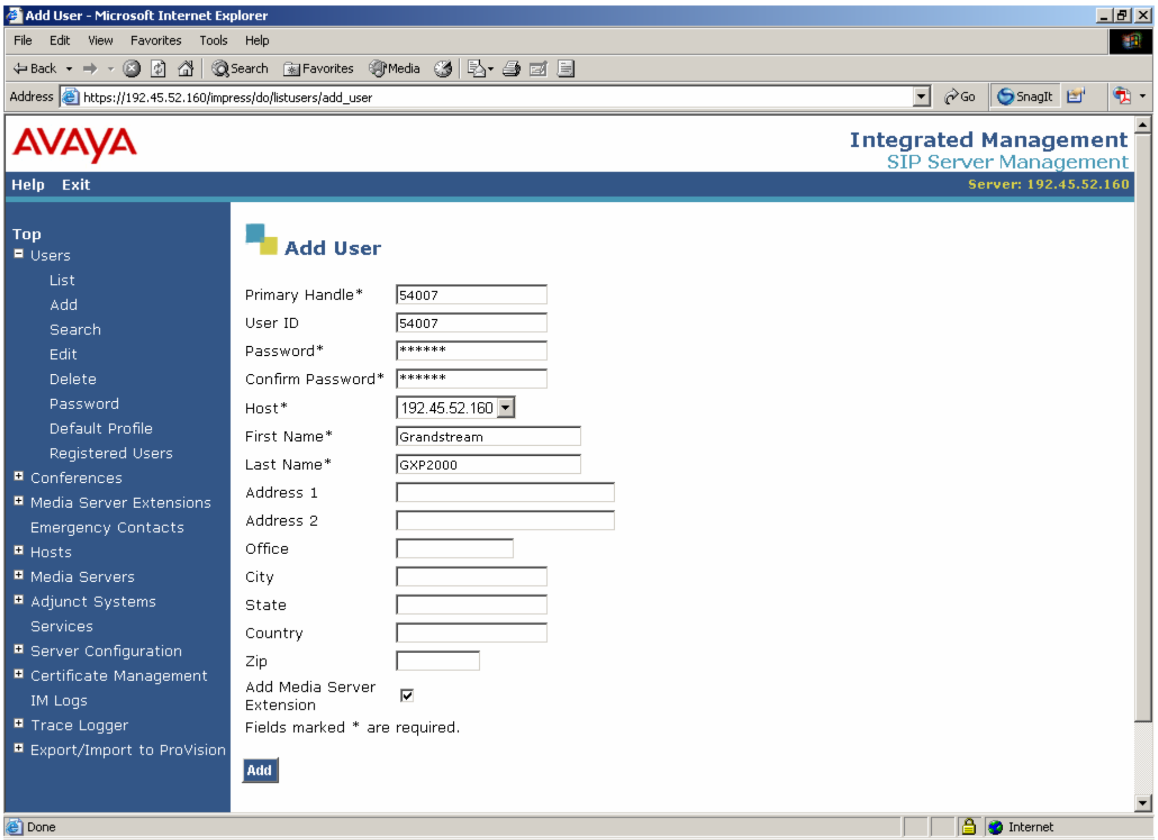
</td></tr>
</table>

Step	Description
5.	In the left pane of the SES Administration Web Interface, expand "**Users**" and click on "**Add**".



<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>6.</td><td>At the Add User page, configure the following:

Primary Handle – Enter the phone number of the Grandstream telephone. The number must match the phone number entered in Section 3.6 Step 3.
Password and Confirm Password – Specify a password that the Grandstream IP telephone must use to successfully register with Avaya SES.
Host – Select the IP address or FQDN of the Avaya SES server.
First Name and Last Name – Enter descriptive names.
Check the Add Media Server Extension checkbox.
Click on “Add”.
Click “Continue” on the next page [not shown].

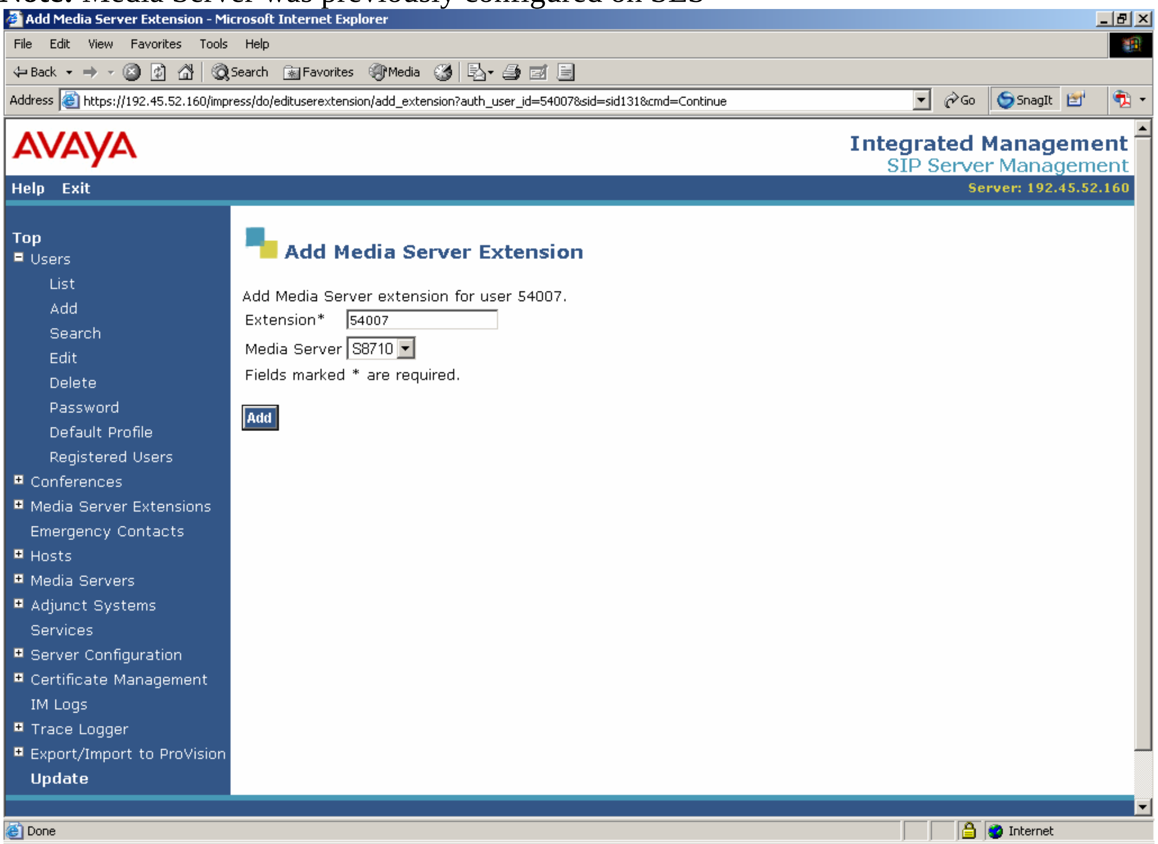


</td></tr>
</table>

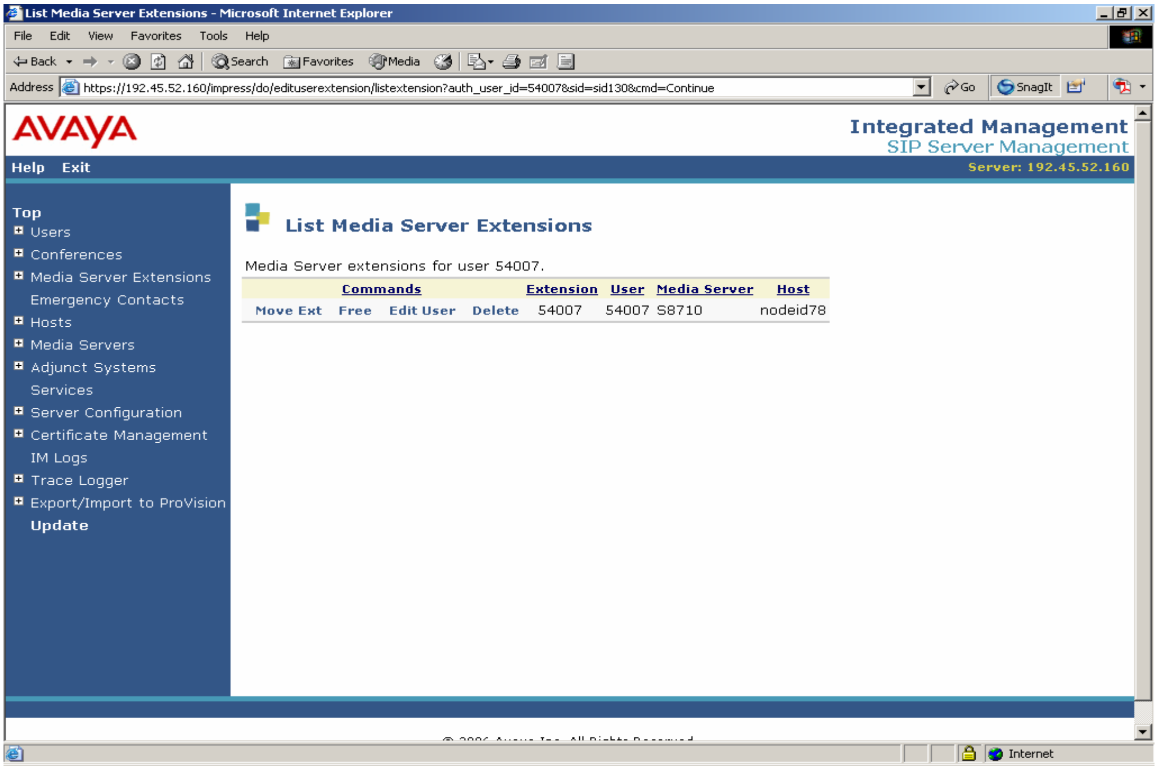
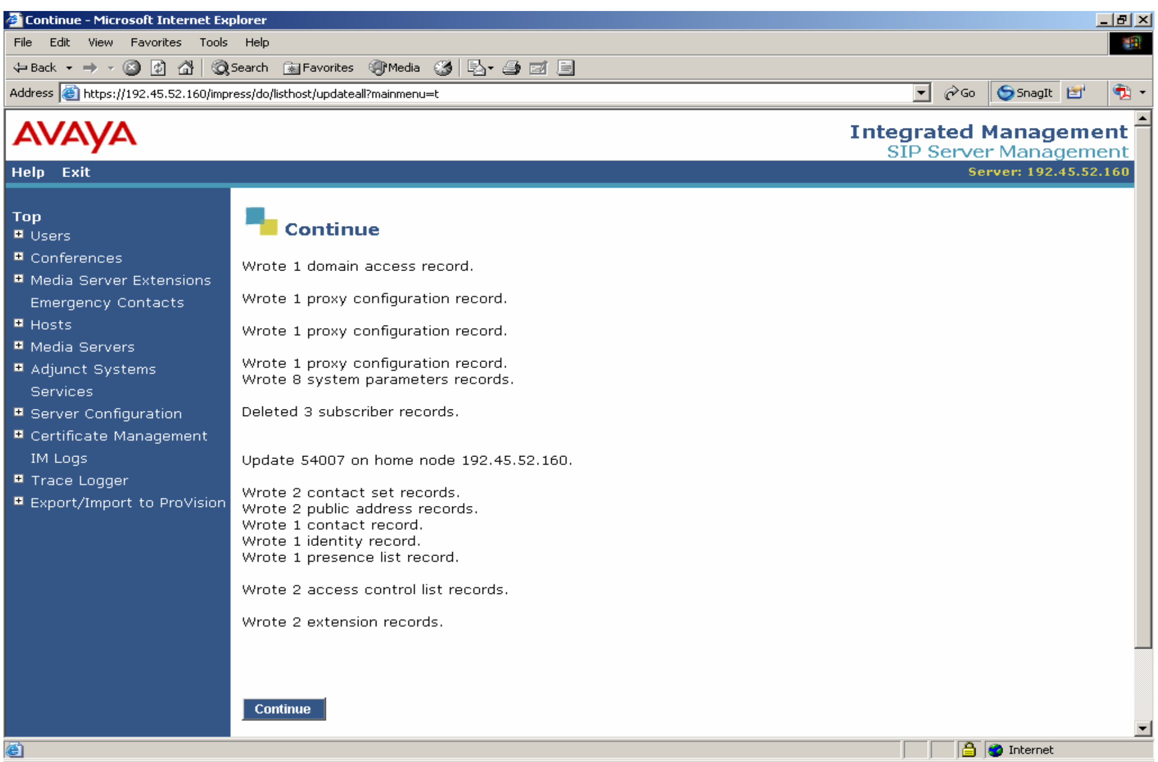
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>7.</td><td>
At the **Add Media Server Extension** screen, configure the following:

- **Extension –** Set it to the corresponding Avaya Communication Manager OPS station configured in **Section 3.6 Step 3**.
- **Media Server** – Set to the Media Server where this OPS station is configured.
- Click on "**Add**".
- Click **"Continue"** on the next page [not shown].

Note: Media Server was previously configured on SES



</td></tr>
<tr><td>8.</td><td>Repeat Steps 2 – 7 as necessary to configure SIP users for additional Grandstream SIP Telephones.</td></tr>
</table>

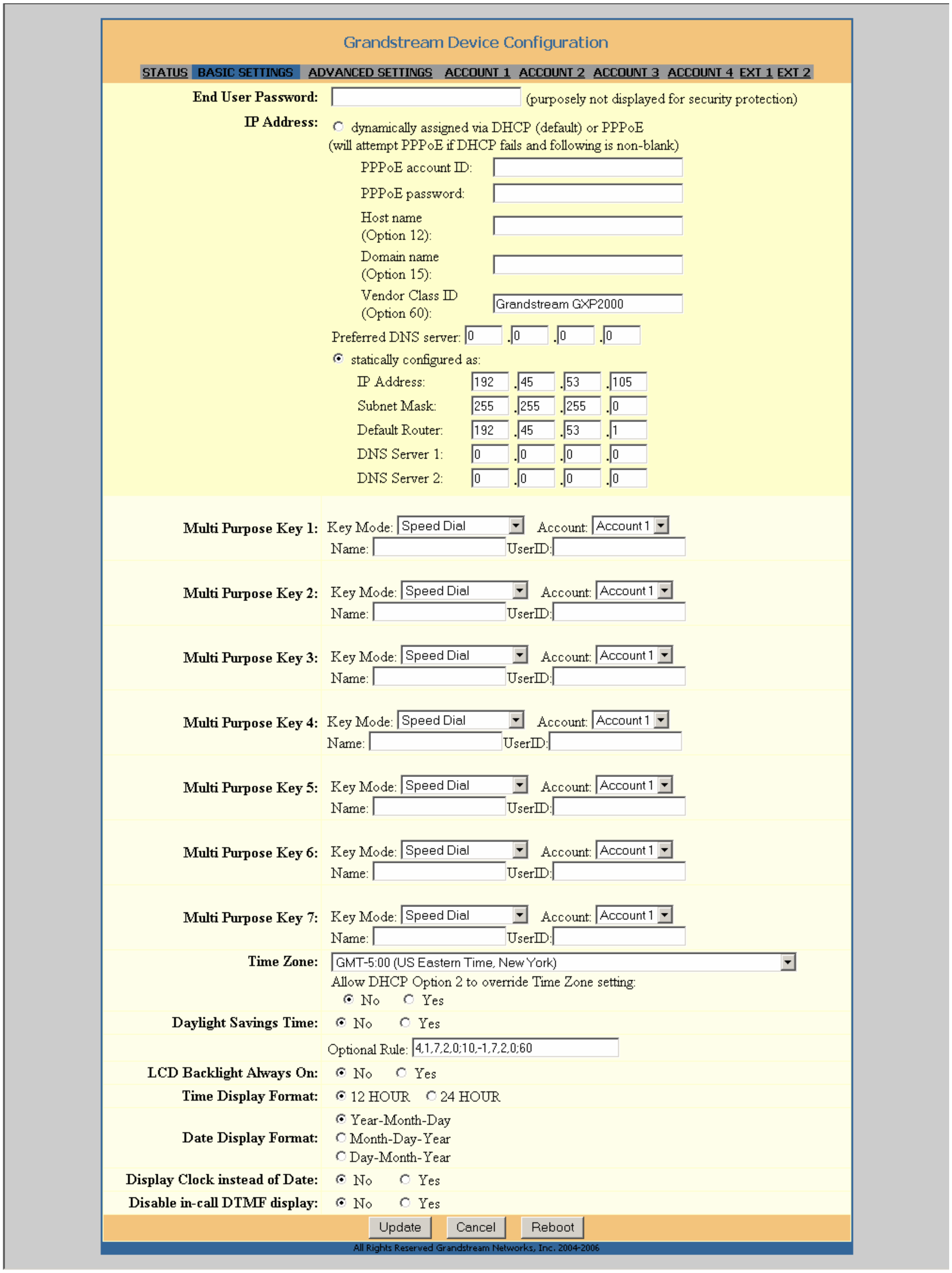
Step	Description
9.	Click on “**Update**” at the bottom of the left pane. 
10.	Click on “**Continue**” at the bottom of the right pane. 

5. Configure the Grandstream Telephone

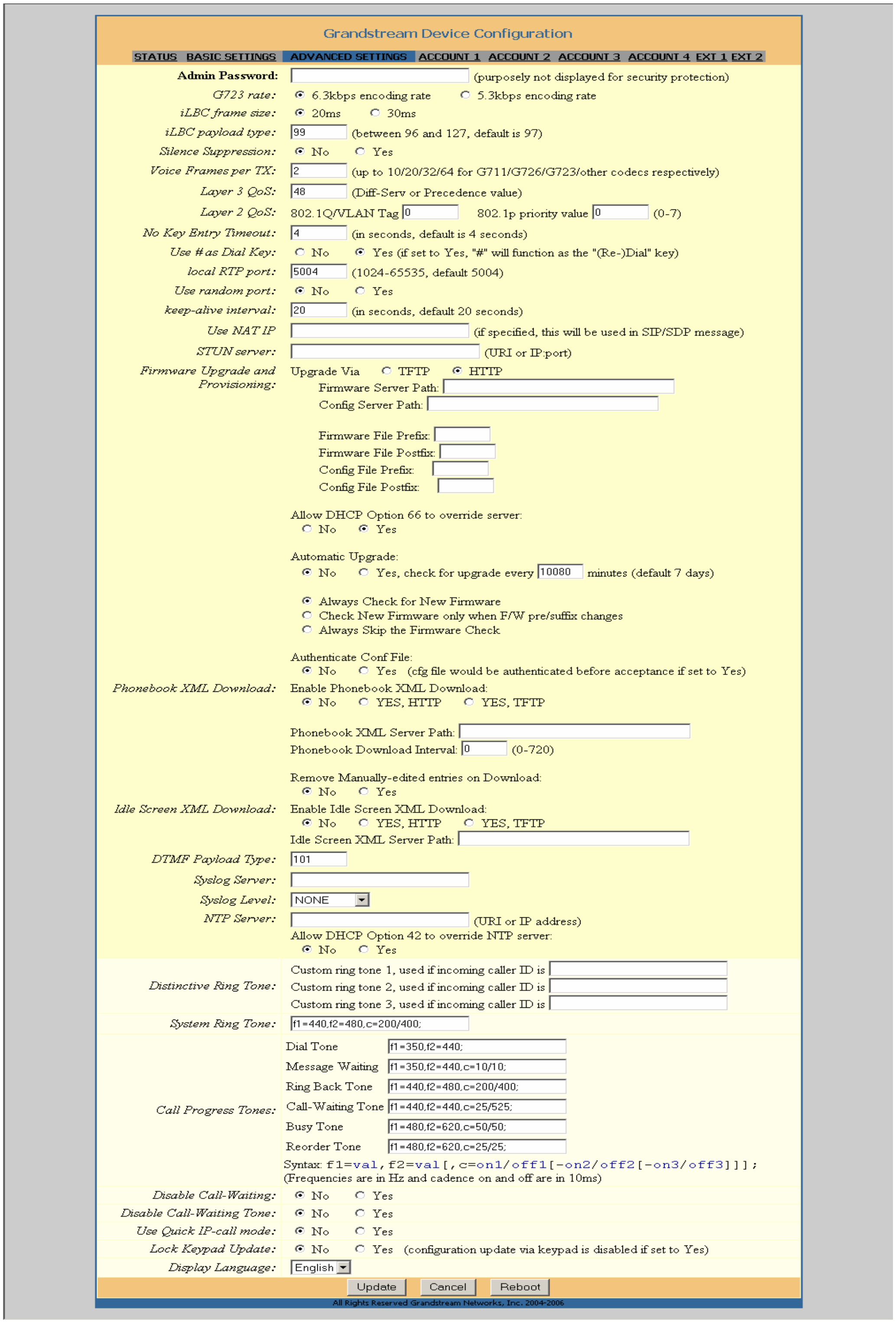
This section describes the steps for configuring the Grandstream Telephone. This section assumes that the Grandstream Telephone's IP address is already configured.

Step	Description
1.	Open a web browser, 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 on **"Login"** button to proceed to the next screen. **Note:** Following steps are performed to configure GXP2000 but the configuration for BT200 endpoint is exactly same except BT200 has only one **ACCOUNT** to be configured in its pull-down menu. 

Step	Description
2.	At **BASIC SETTINGS** screen, configure the following: • **IP Address** – Set the IP address if required. • **Subnet Mask –** Set the subnet mask. • **Default Router –** Set the default router. • Click **“Update”** to modify the values.

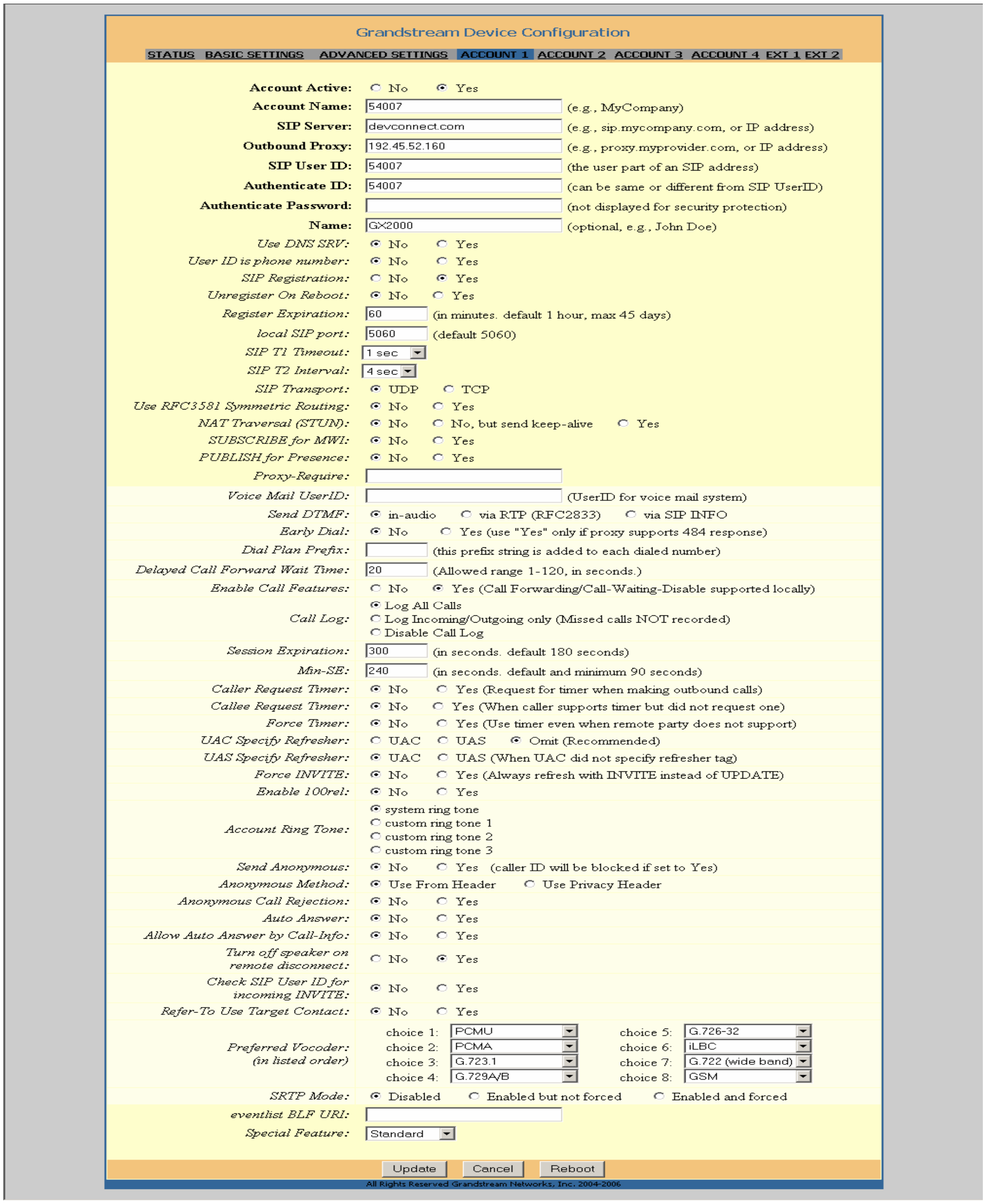


Step	Description
3.	At **ADVANCED SETTINGS** screen, configure the following: • **Layer 3 QoS –** Set to the desired value. For compliance testing, we set it to **34** and **48**. • **802.1p priority value –** Set to the desired value between 0 and 7. For compliance testing, we set it **0** and **6**. • Click **“Update”** to modify the values.



<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.</td><td>At ACCOUNT1 screen, configure the following:

Account Name – Set to the Primary Handle configured in Section 4, Step 6.
SIP Server – Set to the SIP Domain configured in Section 4, Step 2.
Outbound Proxy – Set to the Avaya SES server IP address.
SIP User ID – Set to the Primary Handle configured in Section 4, Step 6.
Authenticate ID – Set to the User Id configured in Section 4, Step 6.
Authenticate Password – Set to the Password configured in Section 4, Step 6.
Name – Any String for identification purposes.
Turn off speaker on remote disconnect – Set the value to Yes.
Click “Update” to modify the values.



</td></tr>
</table>

6. Interoperability Compliance Testing

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

6.1. General Test Approach

The general test approach was to place calls to and from the Grandstream SIP telephone and exercise basic telephone operations on the Grandstream Telephone. The main objectives were to verify that:

- The Grandstream Telephone successfully registers with Avaya SES.
- The Grandstream Telephone successfully establishes calls with Avaya SIP, H.323, and digital telephones attached to Avaya SES or Avaya Communication Manager.
- The Grandstream Telephone successfully establishes calls with PSTN telephones through Avaya Communication Manager.
- The Grandstream Telephone successfully handles concurrent calls on its two lines.
- The Grandstream Telephone successfully negotiates the right codec.
- The Grandstream Telephone successfully shuffles for VOIP calls.
- The Grandstream Telephone successfully transmits DTMF during a call.
- The Grandstream Telephone successfully handles layer-3 (DiffServ) QoS for Audio.
- The Grandstream Telephone successfully handles layer-2 (802.1p) QoS for Audio.

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 line 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 line 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 shuffles to communicate directly with the other endpoint. Grandstream Telephones successfully negotiated the codec to be used.

The following observations were made during testing:

- Grandstream Telephone does not support de-registration but when the telephone is rebooted, it automatically re-registers with Avaya SES.
- Grandstream Telephone does not support VLAN tagging.
- Grandstream Telephone cannot mute all parties if it initiates the conference. Only the first called party is muted.
- Grandstream Telephone fail to shuffle if both the endpoints are Grandstream telephones. A workaround is to configure both telephones to support the same set of codecs and these codecs should be unique.
- Grandstream Telephone terminates the call after a certain time when the call is muted or put on hold. Grandstream supports a configurable session timer which is incompatible with Avaya SIP implementation. A workaround is to make the session timer large enough for the SIP trunk configured in Avaya Communication Manager and Grandstream Telephone.
- Grandstream Telephone has a delay of about 5 seconds when the audio is muted/unmuted.
- Grandstream Telephone BT200 re-registers with Avaya SES under some extreme circumstances.
- Grandstream Telephone is not compatible with Avaya SES for Presence and IM implementation.

Grandstream Networks expects to resolve the above observations in future 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, use following commands to verify that the calls successfully shuffled between two SIP telephones:

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td rowspan="3">1.</td><td>Check the ports which are active for the SIP trunk being used by using the following command:
"status trunk 10"
Note down the members in active state. In our example, 10/2 and 10/6 are active.</td></tr>
<tr><td><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/idle   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></td></tr>
<tr><td></td></tr>
</table>

Step	Description
2.	Issue the following command for the ports in active state: • **“status trunk 10/2”** • Note that the **Near-end IP Addr** and **Far-end IP Addr** for **Audio** are talking on the same port and **Audio Connection Type** is **ip-direct**. This signifies that the endpoints have shuffled and talking to each other directly.

```
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
```

Step	Description
3.	Note on the second page of the status screen, it verifies that both endpoints are using the same codec **g711u**.

```
status trunk 10/2                                              Page 2 of 2
                     SRC PORT TO DEST PORT TALKPATH
src port: T00047
T00047:TX:192.45.53.101:34008/g711u/20ms
T00051:TX:192.45.53.102:34008/g711u/20ms













Dst port: T00051
```

8. Support

For technical support on Grandstream Networks telephones, consult the support pages at http://www.grandstream.com/y-services.htm or contact Grandstream Networks technical support at:

- Telephone: 1- (617) 566 9300
- E-mail: Provide email address if available

9. Conclusion

These Application Notes describe a solution comprised of Avaya Communication Manager 3.1.2, Avaya SIP Enablement Services (SES) 3.1.1, and Grandstream Networks SIP telephones. Grandstream GXP2000 and BT200 are SIP-based VoIP telephones. Grandstream GXP2000 is typically used in an enterprise or small business environment and BT200 is used by residential or SoHo users. During compliance testing, the Grandstream Telephones successfully registered with Avaya SES, placed and received calls to and from SIP and non-SIP telephones, and established conference calls.

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 R3.1.1*, Issue 2.0, August 2006, Document Number 03-600768

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

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya Developer*Connection* Program at devconnect@avaya.com.