



Avaya Solution & Interoperability Test Lab

Application Notes for the G-Tek HL20x SIP Telephones with Avaya Communication Manager and Avaya SIP Enablement Services – Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Communication Manager, Avaya SIP Enablement Services, and G-Tek HL20x SIP telephones. During compliance testing, G-Tek HL20x SIP Telephones successfully registered with Avaya SES, placed and received calls to and from SIP and non-SIP telephones, and executed other telephony features such as three-way conference, transfers, holds, etc.

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 Communication Manager 5.1, Avaya SIP Enablement Services (SES) 5.1, and G-Tek HL20x SIP Telephones. Avaya Communication Manager and Avaya SIP Enablement Services has the capability to extend advanced telephony features to SIP stations. These features can be extended to non-Avaya SIP telephones such as the G-Tek HL20x SIP Telephones.

Figure 1 illustrates a sample configuration consisting of an Avaya Communication Manager running on an Avaya S8300C Server with the Avaya G350 Media Gateway, the Avaya SIP Enablement Services (SES) that is co-resident on the S8300C, and the G-Tek telephones. For completeness, an Avaya 4610SW SIP IP Telephone, an Avaya 9630 H.323 IP Telephone and an Avaya 2420 Digital Telephone are included in **Figure 1** to demonstrate calls between the SIP-based G-Tek telephones and Avaya SIP, H.323 and digital telephones. The MM314 LAN Media Module installed on the G350 Media Gateway provides LAN connectivity and power to the Avaya and G-Tek IP telephones through Power-over-Ethernet (PoE). Avaya IA 770 INTUITY AUDIX Messaging (IA 770) is used to support voice messaging. An audio wav file is used as the music-on hold (MOH) through the virtual Voice Announcement with LAN (VAL) feature in the G350 Media Gateway. The ISDN-BRI trunk is also included to demonstrate calls routed by Avaya Communication Manager between the G-Tek telephones and the PSTN.

The G-Tek telephone originates a call by sending a call request (SIP INVITE message) to the Avaya SES. The Avaya SES 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 for delivery 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 at Avaya Communication Manager that is destined for the G-Tek telephone, Avaya Communication Manager routes the call over the SIP trunk to the Avaya SES for delivery to G-Tek 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 [1] thru [4].

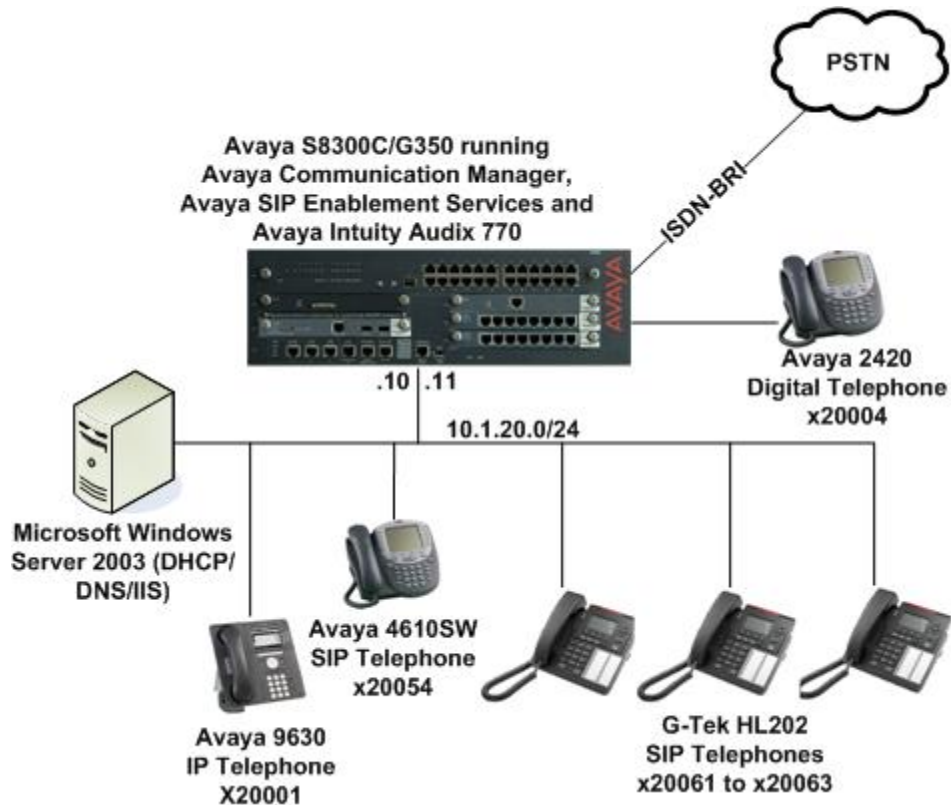


Figure 1: Sample Configuration

2. Equipment and Software Validated

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

Equipment	Software / Firmware
Avaya S8300C Server	Avaya Communication Manager, Avaya SIP Enablement Services and Avaya IA 770 5.1 (Service Pack 01.0.414.3-15962 and SES-01.0.414.3-SP1)
Avaya G350 Media Gateway - MM722AP BRI Media Module - MM712AP DCP Media Module - MM314 LAN Media Module	28.17.0 HW01, FW008 HW04, FW009 HW00 FW000
Avaya 4600 Series IP Telephones - 4610SW	2.2.2 (SIP)
Avaya 9600 Series IP Telephones - 9630	2.0 (H.323)
Avaya 2400 Series Digital Telephone	-
G-Tek HL202 SIP Telephones	20080903

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 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 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.

These steps are performed from the Avaya Communication Manager System Access Terminal (SAT) interface. G-Tek and other SIP telephones are configured as Off-PBX Stations (OPS) in Avaya Communication Manager. Avaya Communication Manager does not directly control an OPS endpoint, but its features and calling privileges can be applied to it by associating a local, on-PBX telephone with the OPS endpoint. 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.

3.1. 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 11 OPTIONAL FEATURES G3 Version: V15 Software Package: Standard Location: 2 RFA System ID (SID): 1 Platform: 13 RFA Module ID (MID): 1 USED Platform Maximum Ports: 900 118 Maximum Stations: 450 50 Maximum XMOBILE Stations: 0 0 Maximum Off-PBX Telephones - EC500: 450 0 Maximum Off-PBX Telephones - OPS: 450 8 Maximum Off-PBX Telephones - PBFMC: 0 0 Maximum Off-PBX Telephones - PVFMC: 0 0 Maximum Off-PBX Telephones - SCAN: 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: 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.</p>
<pre> display system-parameters customer-options OPTIONAL FEATURES Page 2 of 11 IP PORT CAPACITIES USED Maximum Administered H.323 Trunks: 200 16 Maximum Concurrently Registered IP Stations: 450 3 Maximum Administered Remote Office Trunks: 450 0 Maximum Concurrently Registered Remote Office Stations: 450 0 Maximum Concurrently Registered IP eCons: 2 0 Max Concur Registered Unauthenticated H.323 Stations: 200 0 Maximum Video Capable H.323 Stations: 200 0 Maximum Video Capable IP Softphones: 200 0 Maximum Administered SIP Trunks: 450 12 Maximum Administered Ad-hoc Video Conferencing Ports: 0 0 Maximum Number of DS1 Boards with Echo Cancellation: 0 0 Maximum TN2501 VAL Boards: 0 0 Maximum Media Gateway VAL Sources: 2 1 Maximum TN2602 Boards with 80 VoIP Channels: 0 0 Maximum TN2602 Boards with 320 VoIP Channels: 0 0 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 an IP 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 an IP network region to specify which codec sets may be used within and between network regions. For the compliance testing, G.711MU and G.729B were used and Media Encryption was set to none.</p>

```

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.729B      n            2           20
3:
4:
5:
6:
7:

Media Encryption
1: none
2:
3:

```

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.	Enter the change ip-network-region n command, where n is a number between 1 and 250 inclusive and configure the following: <ul style="list-style-type: none"> • Authoritative Domain – Set to b.com in this example. 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.

```

change ip-network-region 2                                     Page 1 of 19
                                                              IP NETWORK REGION
Region: 2
Location:                Authoritative Domain: b.com
Name: Local
MEDIA PARAMETERS
Codec Set: 2          Intra-region IP-IP Direct Audio: yes
                        Inter-region IP-IP Direct Audio: yes
UDP Port Min: 2048      IP Audio Hairpinning? n
UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS
Call Control PHB Value: 46    RTCP Reporting Enabled? y
Audio PHB Value: 46          RTCP MONITOR SERVER PARAMETERS
Video PHB Value: 26          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
H.323 IP ENDPOINTS
H.323 Link Bounce Recovery? y    RSVP Enabled? n
Idle Traffic Interval (sec): 20
Keep-Alive Interval (sec): 5
Keep-Alive Count: 5

```

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**.

```

change ip-network-region 2                                     Page 3 of 19
                                                              Inter Network Region Connection Management
src dst codec direct  WAN-BW-limits  Video  Intervening  Dyn
rgn rgn set   WAN  Units   Total Norm  Prio Shr  Regions  CAC IGAR AGL
2  1  2    y   NoLimit
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. 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>Enter 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. • Co-Resident SES – Set to y to connect to the co-resident SES. • Near-end Node Name - Set to procr. • Near-end Listen Port - Set to 6001 for co-resident Avaya SES. • Far-end Node Name - Set to procr for co-resident Avaya SES. • Far-end Network Region - Set to the region configured in Section 3.3. • Far-end Domain - Set to b.com in this example. This should match the SIP Domain value in Section 4 Step 2.
	<pre> add signaling-group 50 Page 1 of 1 SIGNALING GROUP Group Number: 50 Group Type: sip Transport Method: tls Co-Resident SES? y Near-end Node Name: procr Far-end Node Name: procr Near-end Listen Port: 6001 Far-end Listen Port: 5061 Far-end Domain: b.com Far-end Network Region: 2 Bypass If IP Threshold Exceeded? n DTMF over IP: rtp-payload Direct IP-IP Audio Connections? y IP Audio Hairpinning? n Enable Layer 3 Test? n Session Establishment Timer(min): 3 Alternate Route Timer(sec): 6 </pre>

3.5. 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
<p>1.</p>	<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 3.4. • TAC (Trunk Access Code) – Set to any available trunk access code. • Signaling Group – Set to the Group Number field value configured in Section 3.4. • 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: 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> add trunk-group 50 Page 1 of 21 TRUNK GROUP Group Number: 50 Group Type: sip CDR Reports: n Group Name: SIP Local COR: 1 TN: 1 TAC: 750 Direction: two-way Outgoing Display? n Dial Access? n Night Service: Queue Length: 0 Service Type: tie Auth Code? n Signaling Group: 50 Number of Members: 12 </pre>
<p>2.</p>	<p>Proceed to Page 4 of trunk group configuration and set Telephone Event Payload Type to 101 to match the RFC2833 Payload Type field value configured on the G-Tek telephone in Section 5 Step 4.</p>
	<pre> add trunk-group 50 Page 4 of 21 PROTOCOL VARIATIONS Mark Users as Phone? P Prepend '+' to Calling Number? n Send Transferring Party Information? n Network Call Redirection? n Telephone Event Payload Type: 101 </pre>

3.6. 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 G-Tek telephones.

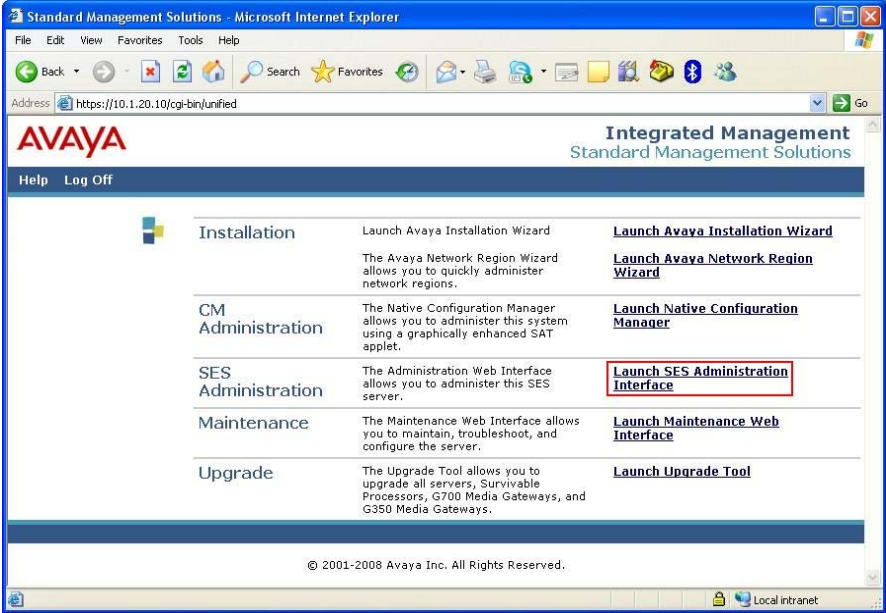
Step	Description
<p>1.</p>	<p>Enter the add station s command, where s is an available extension in the dial plan, to administer an OPS station. On Page 1 of the station form configure the following fields:</p> <ul style="list-style-type: none"> • Type – Set to 6408D+. • Port – Set to X. • Name – Enter any descriptive name. <pre> add station 20061 Page 1 of 4 STATION Extension: 20061 Lock Messages? n BCC: 0 Type: 6408D+ Security Code: TN: 1 Port: X Coverage Path 1: COR: 1 Name: Robin Coverage Path 2: COS: 1 Hunt-to Station: STATION OPTIONS Loss Group: 2 Personalized Ringing Pattern: 1 Data Module? n Message Lamp Ext: 20061 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 the BUTTON ASSIGNMENTS section. The number of call appearances should match the Call Limit field value in Step 4. Configure additional feature buttons such as no-hld-cnf (required for Conference on Answer) and auto-cback (required for Automatic Call Back) as required.</p> <pre> add station 20061 Page 3 of 4 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: no-hld-cnf 2: call-appr 6: auto-cback 3: call-appr 7: 4: 8: </pre>

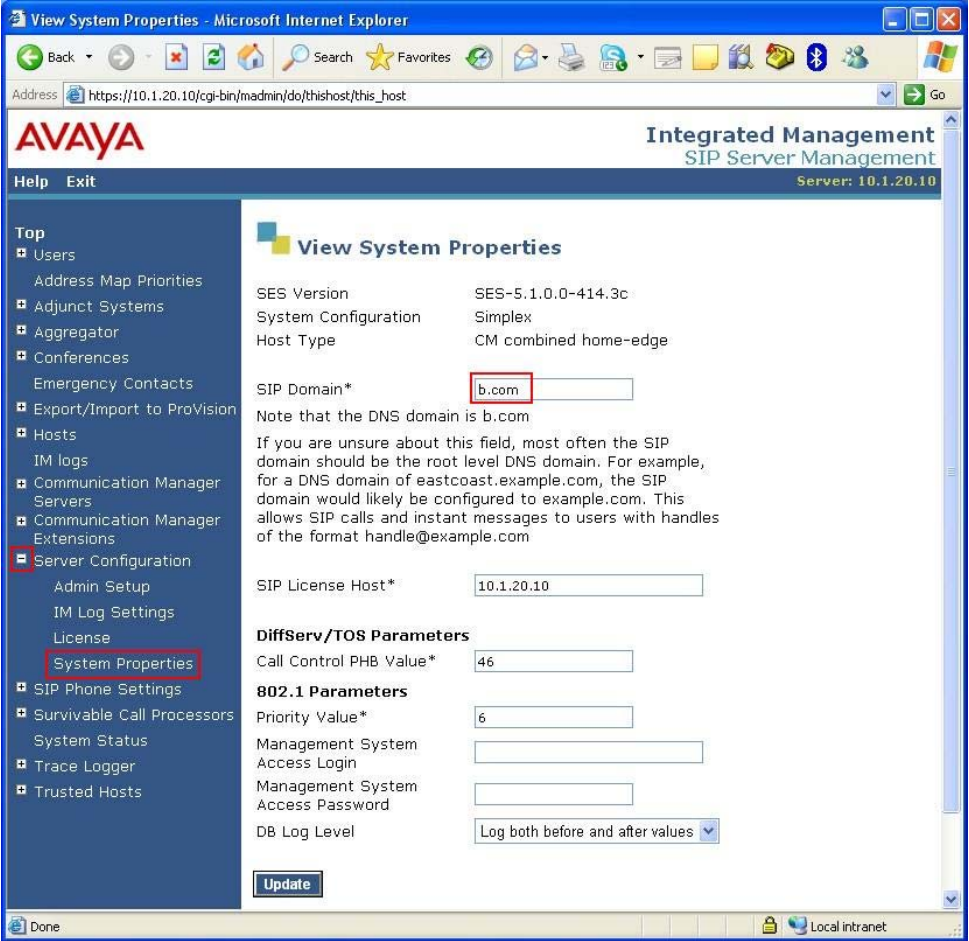
<p>3.</p>	<p>Enter the add off-pbx-telephone station-mapping command and configure the following:</p> <ul style="list-style-type: none"> • Station Extension – Set the extension of the OPS station as configured above. • Application – Set to OPS. • Phone Number – Enter the number that the G-Tek telephone 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 3.5. <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>CC</th> <th>Phone Number</th> <th>Trunk Selection</th> <th>Config Set</th> </tr> </thead> <tbody> <tr> <td>20061</td> <td>OPS</td> <td>-</td> <td></td> <td>20061</td> <td>50</td> <td>1</td> </tr> </tbody> </table>	Station Extension	Application	Dial Prefix	CC	Phone Number	Trunk Selection	Config Set	20061	OPS	-		20061	50	1
Station Extension	Application	Dial Prefix	CC	Phone Number	Trunk Selection	Config Set									
20061	OPS	-		20061	50	1									
<p>4.</p>	<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 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> <th>Location</th> </tr> </thead> <tbody> <tr> <td>20061</td> <td>3</td> <td>both</td> <td>all</td> <td>both</td> <td></td> </tr> </tbody> </table>	Station Extension	Call Limit	Mapping Mode	Calls Allowed	Bridged Calls	Location	20061	3	both	all	both			
Station Extension	Call Limit	Mapping Mode	Calls Allowed	Bridged Calls	Location										
20061	3	both	all	both											
<p>5.</p>	<p>Repeat Steps 1 - 4 as necessary to administer additional OPS stations and associations for G-Tek telephones.</p>														

4. Configure Avaya SIP Enablement Services

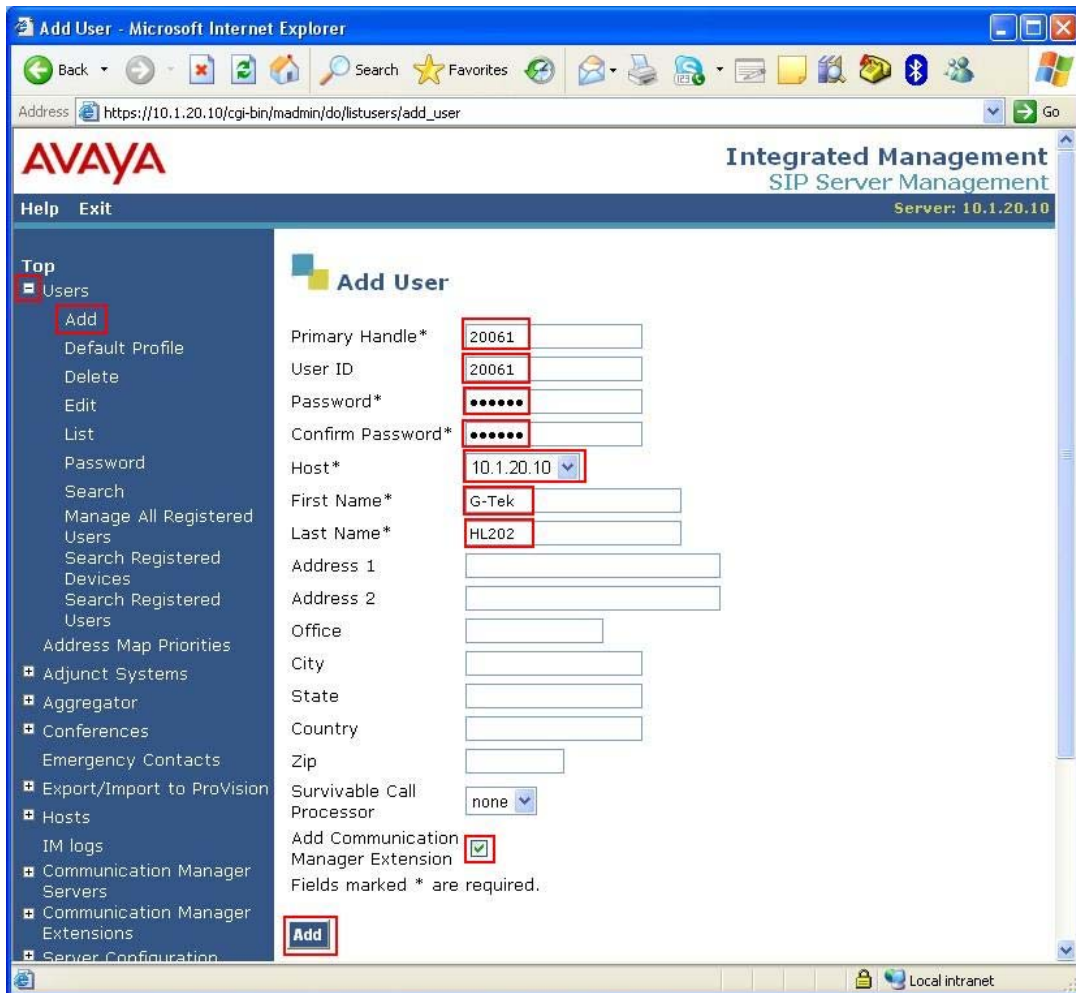
This section describes the steps for creating a SIP trunk between Avaya SES and Avaya Communication Manager. Also, SIP user accounts are configured in Avaya SES and associated with an Avaya Communication Manager OPS station extension. The G-Tek telephones will register with Avaya SES using the SIP user accounts.

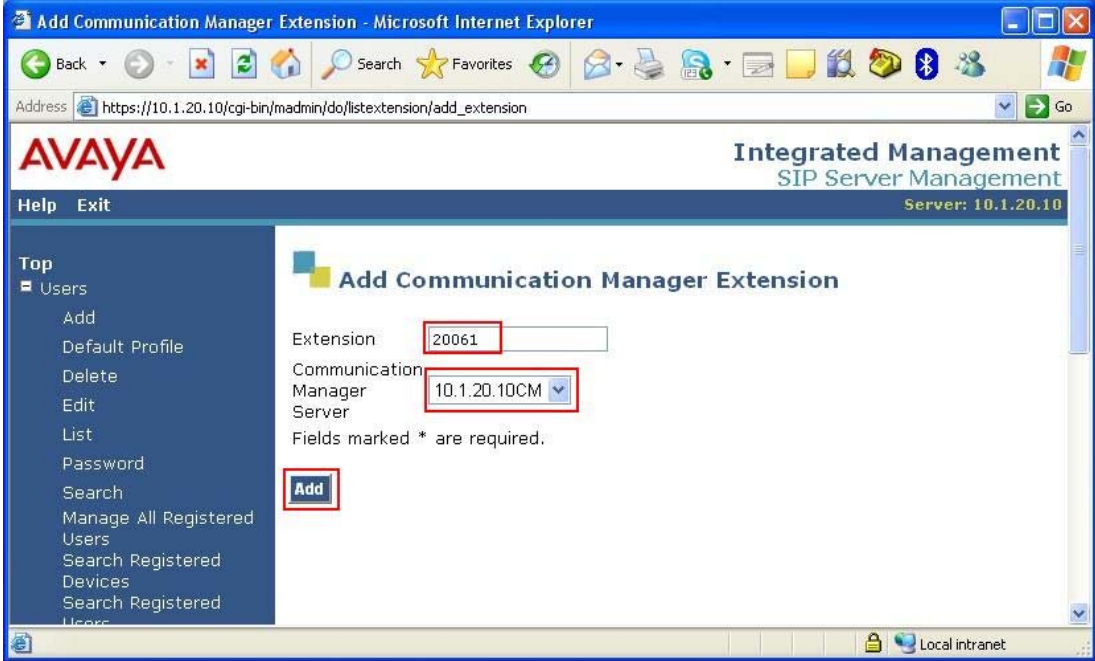
Configuration in the following steps is only for the fields where a value needs to be entered or modified. Default values are used for all other fields.

Step	Description															
1.	<p>Open a web browser, enter http://<IP address of Avaya S8300C server> for the URL, and log in with the appropriate credentials. Click on the Launch SES Administration Interface link upon successful login.</p>  <p>The screenshot shows a web browser window titled "Standard Management Solutions - Microsoft Internet Explorer". The address bar shows "https://10.1.20.10/cgi-bin/unified". The page content includes the Avaya logo and the text "Integrated Management Standard Management Solutions". Below this is a table with the following rows:</p> <table border="1"> <tr> <td>Installation</td> <td>Launch Avaya Installation Wizard The Avaya Network Region Wizard allows you to quickly administer network regions.</td> <td>Launch Avaya Installation Wizard Launch Avaya Network Region Wizard</td> </tr> <tr> <td>CM Administration</td> <td>The Native Configuration Manager allows you to administer this system using a graphically enhanced SAT applet.</td> <td>Launch Native Configuration Manager</td> </tr> <tr> <td>SES Administration</td> <td>The Administration Web Interface allows you to administer this SES server.</td> <td>Launch SES Administration Interface</td> </tr> <tr> <td>Maintenance</td> <td>The Maintenance Web Interface allows you to maintain, troubleshoot, and configure the server.</td> <td>Launch Maintenance Web Interface</td> </tr> <tr> <td>Upgrade</td> <td>The Upgrade Tool allows you to upgrade all servers, Survivable Processors, G700 Media Gateways, and G350 Media Gateways.</td> <td>Launch Upgrade Tool</td> </tr> </table> <p>At the bottom of the page, it says "© 2001-2008 Avaya Inc. All Rights Reserved." and "Local intranet" is visible in the status bar.</p>	Installation	Launch Avaya Installation Wizard The Avaya Network Region Wizard allows you to quickly administer network regions.	Launch Avaya Installation Wizard Launch Avaya Network Region Wizard	CM Administration	The Native Configuration Manager allows you to administer this system using a graphically enhanced SAT applet.	Launch Native Configuration Manager	SES Administration	The Administration Web Interface allows you to administer this SES server.	Launch SES Administration Interface	Maintenance	The Maintenance Web Interface allows you to maintain, troubleshoot, and configure the server.	Launch Maintenance Web Interface	Upgrade	The Upgrade Tool allows you to upgrade all servers, Survivable Processors, G700 Media Gateways, and G350 Media Gateways.	Launch Upgrade Tool
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Upgrade	The Upgrade Tool allows you to upgrade all servers, Survivable Processors, G700 Media Gateways, and G350 Media Gateways.	Launch Upgrade Tool														

Step	Description
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 3.4.  <p>The screenshot shows a web browser window titled "View System Properties - Microsoft Internet Explorer". The address bar shows "https://10.1.20.10/cgi-bin/admin/do/thishost/this_host". The page header includes the Avaya logo and "Integrated Management SIP Server Management" with "Server: 10.1.20.10". The main content area is titled "View System Properties" and displays various system configuration parameters. The "SIP Domain*" field is highlighted with a red box and contains the value "b.com". Below it, a note states "Note that the DNS domain is b.com" and provides instructions on how to determine the SIP domain. Other fields include "SIP License Host*" (10.1.20.10), "Call Control PHB Value*" (46), "Priority Value*" (6), and "DB Log Level" (Log both before and after values). An "Update" button is located at the bottom of the form.</p>

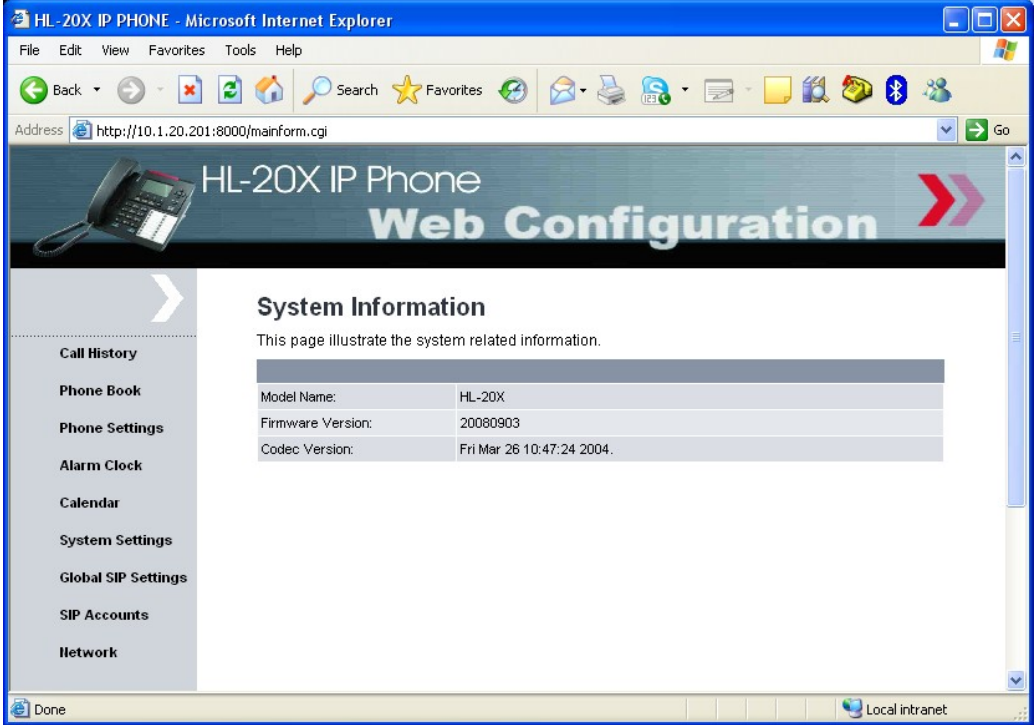
Step	Description
3.	<p>In the left pane of the SIP Server Management page, expand Users and click Add. At the Add User page, configure the following:</p> <ul style="list-style-type: none"> • Primary Handle – Enter the phone number of the G-Tek telephone. This number was configured in Section 3.6 Step 1. • User ID – Set to any descriptive name. • Password and Confirm Password – Specify a password that the G-Tek telephone will use to register with Avaya SES. • Host – Select the IP address of the co-resident Avaya SES server. • First Name and Last Name – Enter descriptive names. • Check the Add Media Server Extension checkbox. <p>Click Add when finished and then click Continue on the next page [not shown].</p>

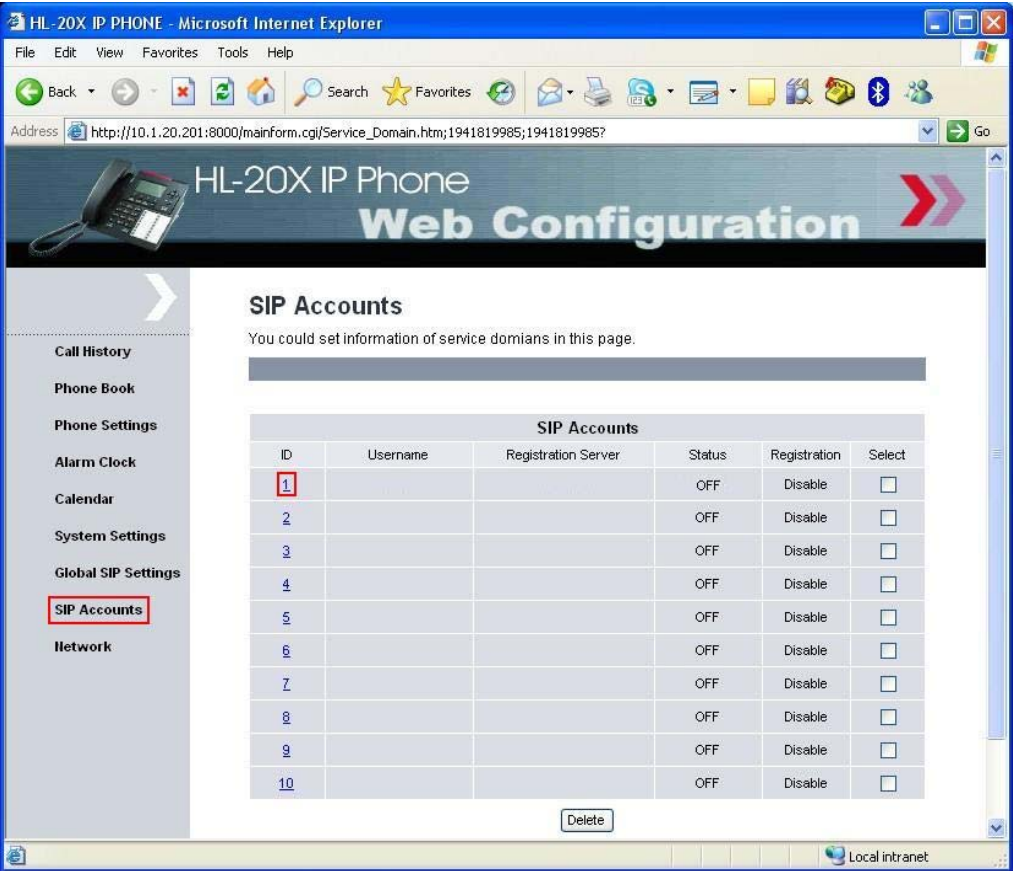


Step	Description
<p>4.</p>	<p>At the Add Communication Manager Extension page, configure the following:</p> <ul style="list-style-type: none"> • Extension – Set to Phone Number field value configured in Section 3.6 Step 1. • Communication Manager Server – Set to the co-resident Communication Manager where this OPS station is configured. • Click Add and then click Continue on the next page [not shown]. <p>Note: Communication Manager Server was previously configured during the initial setup of SES.</p> 
<p>5.</p>	<p>Repeat Steps 3 and 4 as necessary to configure additional G-Tek telephones.</p>

5. Configure G-Tek HL20x Telephones

This section describes the steps for configuring the G-Tek HL20x telephones. This section assumes that the G-Tek 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.

Step	Description						
1.	<p>Open a web browser and enter http://a.b.c.d:8000 for the URL, where a.b.c.d is the IP address of the G-Tek HL20x telephone. Log in with the appropriate credentials and the System Information page will be displayed.</p>  <table border="1" data-bbox="597 1010 1279 1108"><tr><td>Model Name:</td><td>HL-20X</td></tr><tr><td>Firmware Version:</td><td>20080903</td></tr><tr><td>Codec Version:</td><td>Fri Mar 26 10:47:24 2004.</td></tr></table>	Model Name:	HL-20X	Firmware Version:	20080903	Codec Version:	Fri Mar 26 10:47:24 2004.
Model Name:	HL-20X						
Firmware Version:	20080903						
Codec Version:	Fri Mar 26 10:47:24 2004.						

Step	Description																																																																								
2.	<p>Click SIP Accounts on the left navigation menu and click ID 1 to configure the SIP account 1.</p>  <p>The screenshot shows a web browser window titled "HL-20X IP PHONE - Microsoft Internet Explorer". The address bar shows the URL: http://10.1.20.201:8000/mainform.cgi/Service_Domain.htm;1941819985;1941819985?. The page header includes "HL-20X IP Phone" and "Web Configuration". On the left is a navigation menu with items: Call History, Phone Book, Phone Settings, Alarm Clock, Calendar, System Settings, Global SIP Settings, SIP Accounts (highlighted with a red box), and Network. The main content area is titled "SIP Accounts" and contains a table with the following data:</p> <table border="1"> <thead> <tr> <th colspan="6">SIP Accounts</th> </tr> <tr> <th>ID</th> <th>Username</th> <th>Registration Server</th> <th>Status</th> <th>Registration</th> <th>Select</th> </tr> </thead> <tbody> <tr> <td>1</td> <td></td> <td></td> <td>OFF</td> <td>Disable</td> <td><input type="checkbox"/></td> </tr> <tr> <td>2</td> <td></td> <td></td> <td>OFF</td> <td>Disable</td> <td><input type="checkbox"/></td> </tr> <tr> <td>3</td> <td></td> <td></td> <td>OFF</td> <td>Disable</td> <td><input type="checkbox"/></td> </tr> <tr> <td>4</td> <td></td> <td></td> <td>OFF</td> <td>Disable</td> <td><input type="checkbox"/></td> </tr> <tr> <td>5</td> <td></td> <td></td> <td>OFF</td> <td>Disable</td> <td><input type="checkbox"/></td> </tr> <tr> <td>6</td> <td></td> <td></td> <td>OFF</td> <td>Disable</td> <td><input type="checkbox"/></td> </tr> <tr> <td>7</td> <td></td> <td></td> <td>OFF</td> <td>Disable</td> <td><input type="checkbox"/></td> </tr> <tr> <td>8</td> <td></td> <td></td> <td>OFF</td> <td>Disable</td> <td><input type="checkbox"/></td> </tr> <tr> <td>9</td> <td></td> <td></td> <td>OFF</td> <td>Disable</td> <td><input type="checkbox"/></td> </tr> <tr> <td>10</td> <td></td> <td></td> <td>OFF</td> <td>Disable</td> <td><input type="checkbox"/></td> </tr> </tbody> </table> <p>Below the table is a "Delete" button. The browser's status bar at the bottom shows "Local intranet".</p>	SIP Accounts						ID	Username	Registration Server	Status	Registration	Select	1			OFF	Disable	<input type="checkbox"/>	2			OFF	Disable	<input type="checkbox"/>	3			OFF	Disable	<input type="checkbox"/>	4			OFF	Disable	<input type="checkbox"/>	5			OFF	Disable	<input type="checkbox"/>	6			OFF	Disable	<input type="checkbox"/>	7			OFF	Disable	<input type="checkbox"/>	8			OFF	Disable	<input type="checkbox"/>	9			OFF	Disable	<input type="checkbox"/>	10			OFF	Disable	<input type="checkbox"/>
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Step	Description
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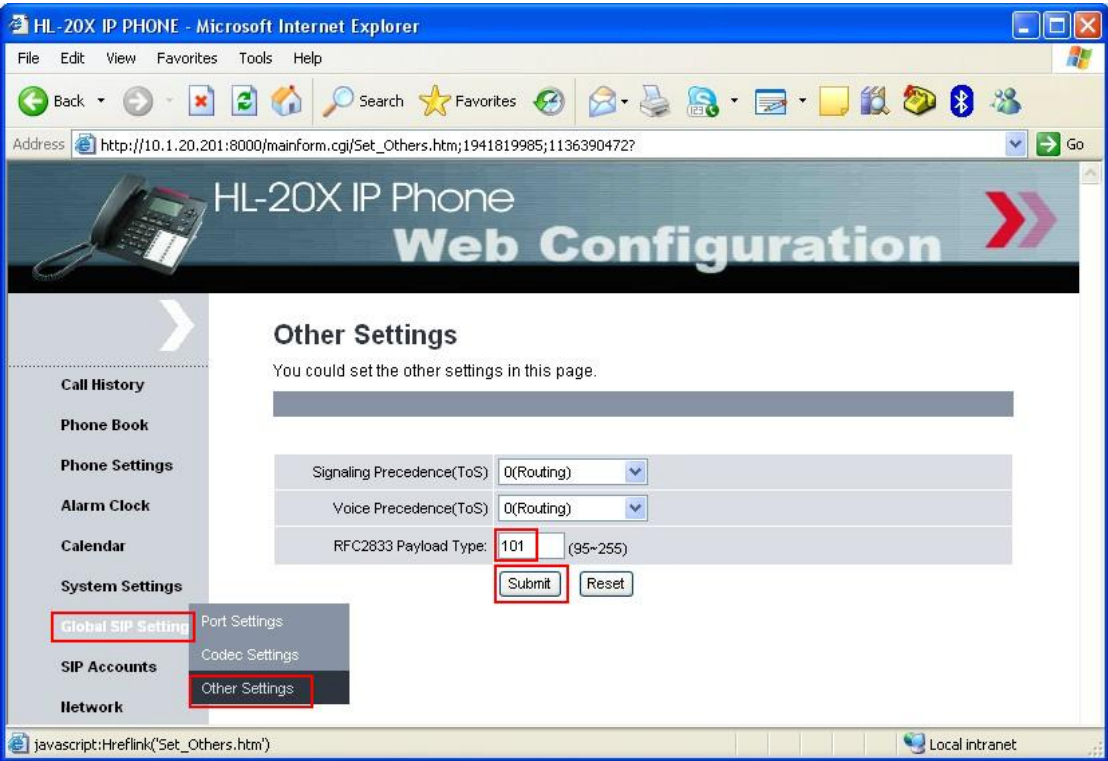
- | | |
|----|---|
| 3. | <p>At the SIP Account Settings page, configure the following:</p> <ul style="list-style-type: none">• Registration – Set to Enable.• Registration ID – Set to the Primary Handle field value configured in Section 4 Step 3.• Username – Set to the Primary Handle field value configured in Section 4 Step 3.• Password – Set to the Password field value configured in Section 4 Step 3.• Registration Server – Set to the Avaya SES server IP address.• Expire Time – Select 3600.• DTMF Type – Select RFC2833.• Authentication Name – Set to the Primary Handle field value configured in Section 4 Step 3.• Proxy Server – Set to the Avaya SES server IP address. |
|----|---|

Click **Submit** to continue.

The screenshot shows the 'SIP Account Settings' page for 'SIP Account 1'. The configuration fields are as follows:

Field	Value
Registration:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Registration ID:	20061
Username:	20061
Password:	*****
Registration Server:	10.1.20.10
Expire Time:	3600
DTMF Type:	RFC2833
Authentication Name:	20061
Proxy Server:	10.1.20.10
Voice Mail:	
Ping before register:	Disable
Send KeepAlive:	<input type="radio"/> On <input checked="" type="radio"/> Off
Status:	OFF

Buttons:

Step	Description
4.	<p>Click Global SIP Settings > Other Settings on the left navigation menu. At the Other Settings page, verify that the value for RFC2833 Payload Type matches the Telephone Event Payload Type field value in Section 3.5 Step 2.</p> 

6. Interoperability Compliance Testing

The focus of the interoperability compliance testing was primarily on verifying call establishment on the G-Tek HL20x SIP telephones and operations such as dialing methods (manual, re-dial, and phone book), hold, mute, transfer and conference. G-Tek HL20x SIP 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 G-Tek HL20x SIP telephones and exercise basic telephone operations. The main objectives were to verify that:

- G-Tek telephones successfully register with Avaya SES.
- G-Tek telephones successfully establish calls with Avaya SIP, H.323, and Digital telephones attached to Avaya SES or Avaya Communication Manager.
- G-Tek telephones successfully establish calls with PSTN telephones through Avaya Communication Manager.
- G-Tek telephones successfully handle concurrent calls.

- G-Tek telephones successfully negotiate the right codec.
- G-Tek telephones successfully shuffle for VoIP calls.
- G-Tek telephones successfully transmit DTMF during a call.
- G-Tek telephones successfully hold and transfer a call.
- G-Tek telephones establish a three party conference call, and display calling party number.

6.2. Test Results

The test objectives of **Section 6.1** were verified. G-Tek telephones successfully shuffled to communicate directly with the other telephones and negotiated the codec.

The following observations were made during testing:

- G-Tek telephones do not return to on-hook when an incomplete or non-existent is dialed.
- A call that was parked on a G-Tek telephone cannot be retrieved on the same telephone.
- OPS features such as Conference on Answer, Drop Last Added Party and Priority Call do not work.

G-Tek will address and attempt to resolve the above observations in future firmware releases. Contact G-Tek for further updates.

7. Verification Steps

The following steps may be used to verify the configuration:

- Verify that the G-Tek telephones successfully register with the Avaya SES server by using the **Users -> Search Registered Users** link on the SIP Server Management Web Interface.
- Place calls to and from the G-Tek telephones 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 3.5. Note down the Member with Service State set to in-service/active. In this example, 0050/001 and 0050/005 are active and either member can be used to verify whether calls shuffled and which codec was used.</p> <pre> status trunk 50 TRUNK GROUP STATUS Member Port Service State Mtce Connected Ports Busy 0050/001 T00011 in-service/active no T00015 0050/002 T00012 in-service/idle no 0050/003 T00013 in-service/idle no 0050/004 T00014 in-service/idle no 0050/005 T00015 in-service/active no T00011 0050/006 T00016 in-service/idle no 0050/007 T00017 in-service/idle no 0050/008 T00018 in-service/idle no 0050/009 T00019 in-service/idle no 0050/010 T00020 in-service/idle no 0050/011 T00021 in-service/idle no 0050/012 T00022 in-service/idle no </pre>
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 belongs to the G-Tek telephones and the Audio Connection Type is ip-direct, it signifies that shuffling was successful. In this example, shuffling was successful. <pre> status trunk 50/1 Page 1 of 2 TRUNK STATUS Trunk Group/Member: 0050/001 Service State: in-service/active Port: T00011 Maintenance Busy? no Signaling Group ID: IGAR Connection? no Connected Ports: T00015 Port Near-end IP Addr : Port Far-end IP Addr : Port Signaling: 01A0017 10. 1. 20. 10 : 6001 10. 1. 20. 10 : 5061 G.711MU Audio: 10. 1. 20. 202 : 10010 10. 1. 20. 201 : 10012 Video: Video Codec: Audio Connection Type: ip-direct Authentication Type: None </pre>

8. Support

For technical support on G-Tek HL20x SIP telephones, contact G-Tek technical support at:

- Telephone: +886-2-26962665 ext 221
- E-mail: paul@gtek.com.tw

9. Conclusion

These Application Notes describe a solution comprised of Avaya Communication Manager 5.1, Avaya SIP Enablement Services 5.1 and G-Tek HL20x SIP telephones. During compliance testing, G-Tek 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 objectives of **Section 6.1** were met with some exceptions as 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*, Release 5.0, Issue 4.0, January 2008, Document Number 03-300509.

[2] *Administration for Network Connectivity for Avaya Communication Manager*, Issue 13, January 2008, Document Number 555-233-504.

[3] *SIP Support in Avaya Communication Manager Running on Avaya S8xxx Servers*, Issue 8, January 2008, Document Number 555-245-206.

[4] *Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services*, Issue 6.0, June 2008, Document Number 03-600768.

Product information for G-Tek products may be found at support@gtek.com.tw

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