Avaya Solution & Interoperability Test Lab

Application Notes for the G-Tek AQ10x 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 AQ10x SIP telephones. During compliance testing, G-Tek AQ10x 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.

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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 AQ10x 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 AQ10x 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].

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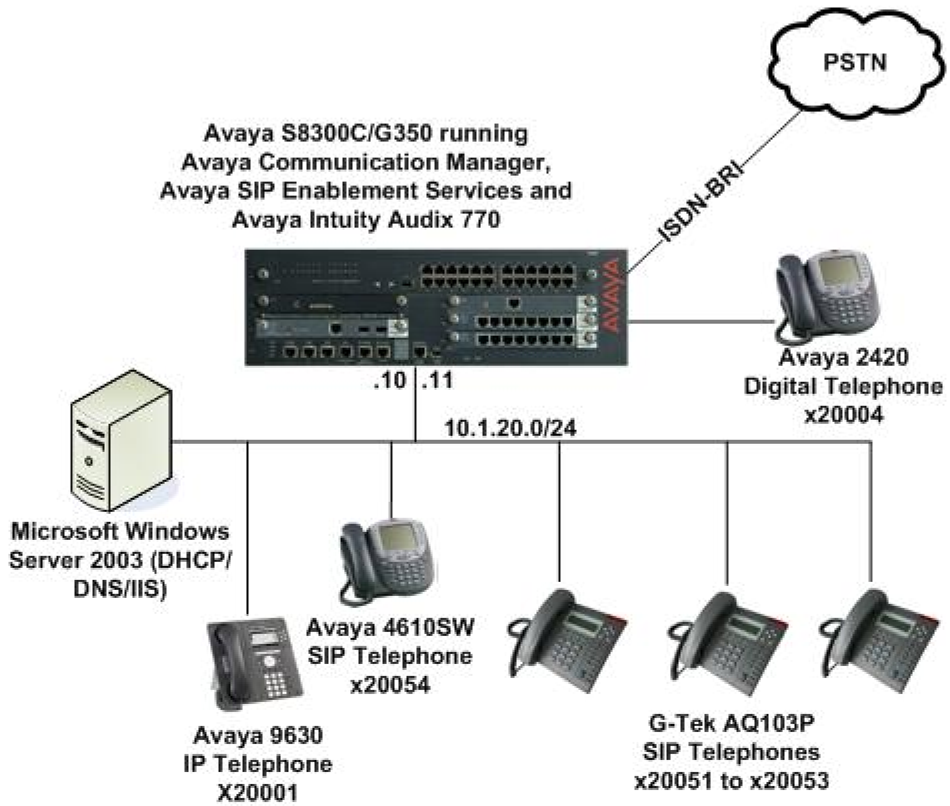

Figure 1: Sample Configuration

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 AQ103P SIP Telephones	20080903

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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.

```
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 - SCCAN: 0     0
```


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2.	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. **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.

```
display system-parameters customer-options                      Page   2 of  11
                                OPTIONAL FEATURES

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.)
```

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.	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.729** were used and **Media Encryption** was set to **none**.

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```
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.729              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: • **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.

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	`` change ip-network-region 2 Page 1 of 19 IP NETWORK REGION Region: 2 Location: Authoritative Domain: b.com Name: Local 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? n 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 ```
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 n 2 2 2 all 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 ```

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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.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td rowspan="2">1.</td><td>Enter the command add signaling-group s, where s is an available signaling group and configure the following:

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.
</td></tr>
<tr><td><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 Network Region: 2
       Far-end Domain: b.com

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

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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.

<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 configure the following:

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.

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

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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.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td rowspan="2">1.</td><td>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:

Type – Set to 6408D+.
Port – Set to X.
Name – Enter any descriptive name.
</td></tr>
<tr><td><pre>
add station 20051   Page   1 of   4
                                     STATION

Extension: 20051   Lock Messages? n   BCC: 0
     Type: 6408D+   Security Code: TN: 1
     Port: X   Coverage Path 1:   COR: 1
     Name: Alan   Coverage Path 2:   COS: 1
                                        Hunt-to Station:

STATION OPTIONS
              Loss Group: 2   Personalized Ringing Pattern: 1
             Data Module? n   Message Lamp Ext: 20051
            Speakerphone: 2-way   Mute Button Enabled? y
        Display Language: english

                                                  Media Complex Ext:
                                                       IP SoftPhone? n
</pre></td></tr>
<tr><td rowspan="2">2.</td><td>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.</td></tr>
<tr><td><pre>
add station 20051   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></td></tr>
</table>

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3.	Enter the **add off-pbx-telephone station-mapping** command and configure the following: • **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 Station Application Dial CC Phone Number Trunk Config Extension Prefix Selection Set **20051** **OPS** **-** **20051** **50** **1**</pre>
4.	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**.
	<pre>add off-pbx-telephone station-mapping Page 2 of 2 STATIONS WITH OFF-PBX TELEPHONE INTEGRATION Station Call Mapping Calls Bridged Location Extension Limit Mode Allowed Calls **20051** **3** **both** **all** **both**</pre>
5.	Repeat **Steps 1 - 4** as necessary to administer additional OPS stations and associations for G-Tek telephones.

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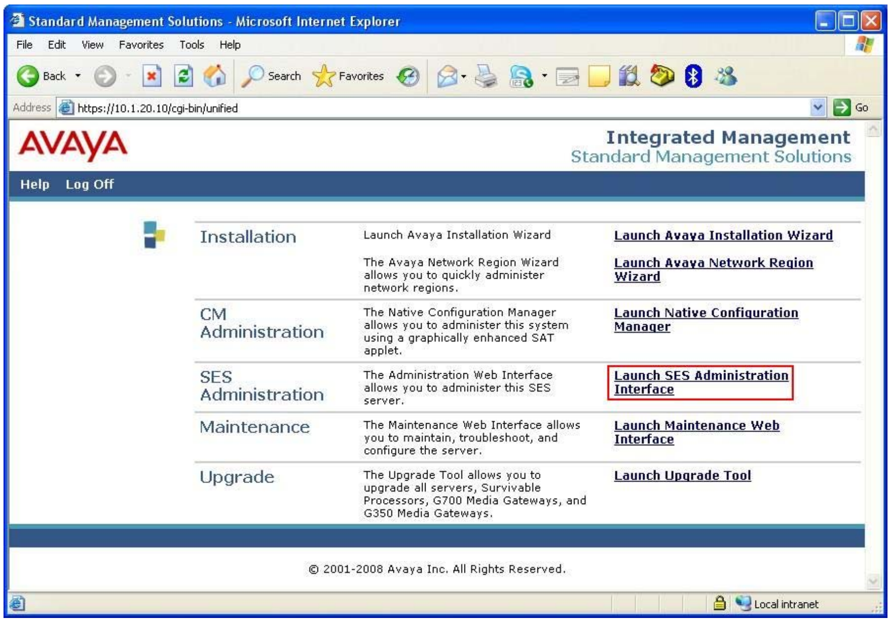
4. Configure Avaya SIP Enablement Services

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.	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.



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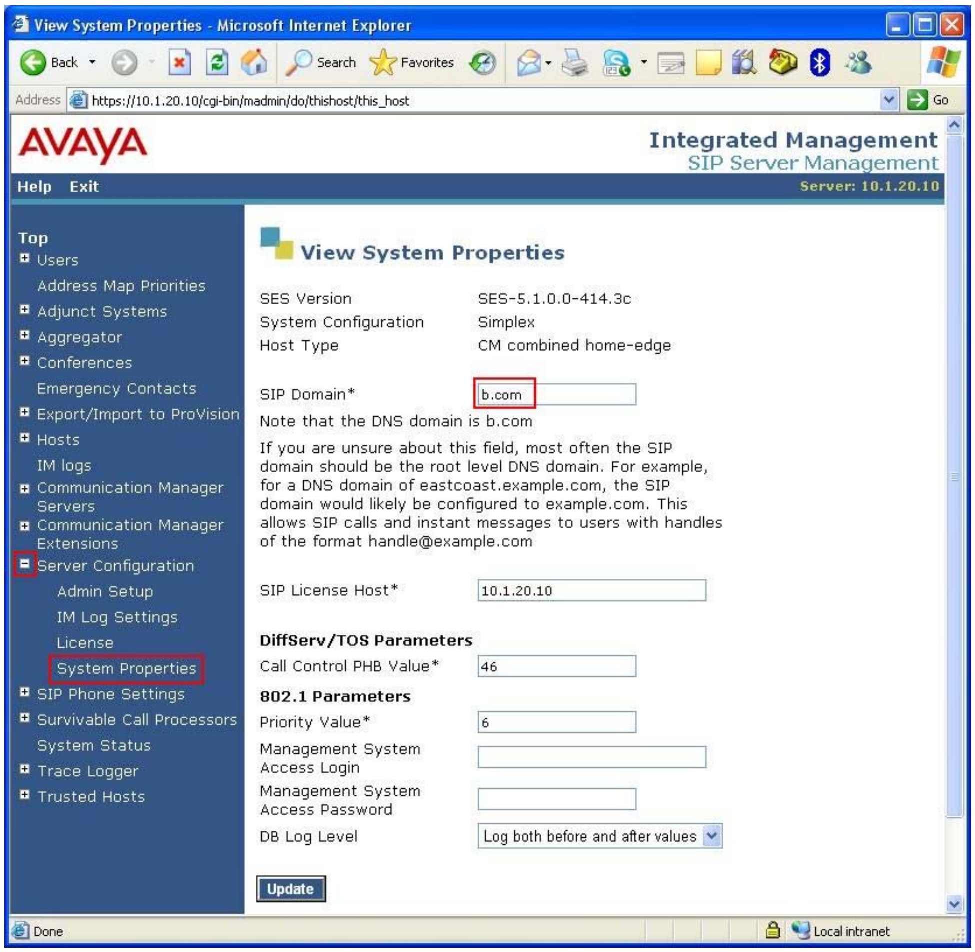
Step	Description
2.	On the **SIP Server Management** page: • Click the **+** sign to expand the options under **Server Configuration**. • Click **System Properties**. • Verify the **SIP Domain** matches the **Far-end Domain** field value configured for the signaling group on Avaya Communication Manager in **Section 3.4**. 

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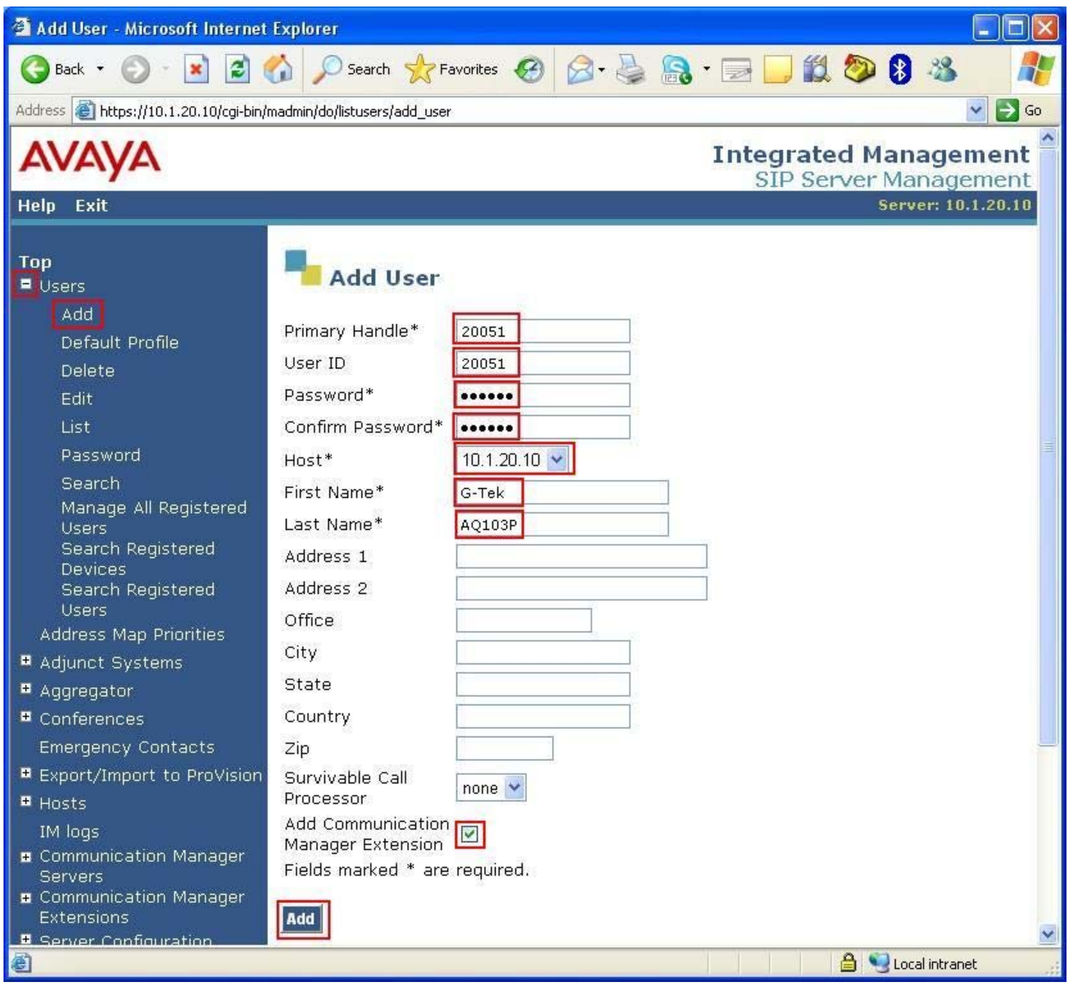
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>
In the left pane of the SIP Server Management page, expand Users and click Add. At the Add User page, configure the following:

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.

Click Add when finished and then click Continue on the next page [not shown].



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

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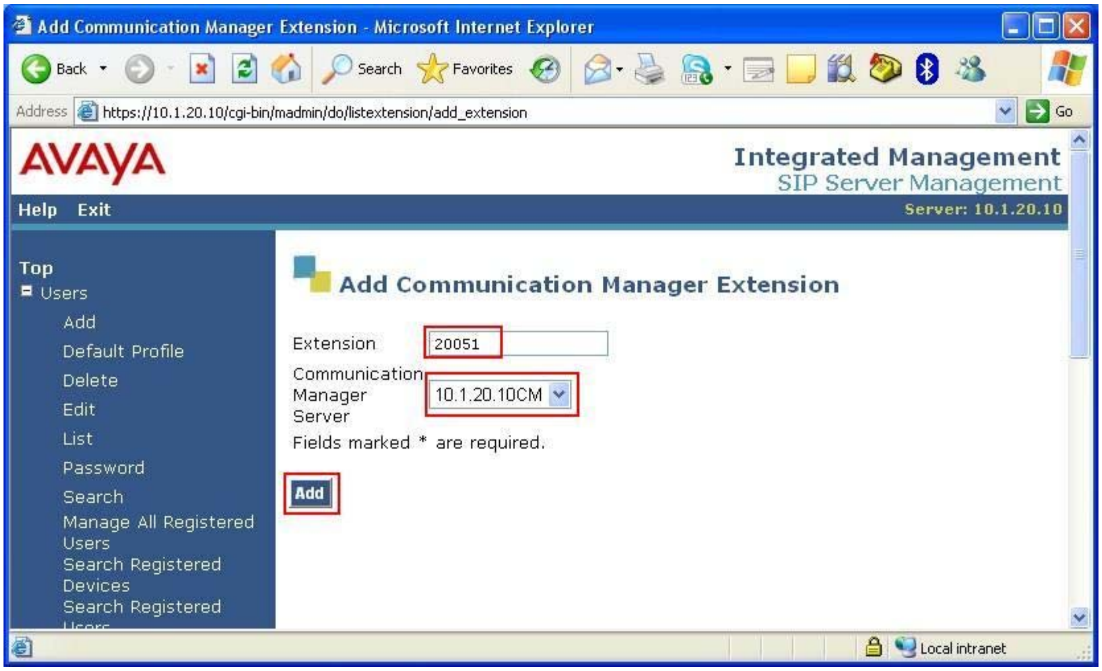
Step	Description
4.	At the **Add Communication Manager Extension** page, configure the following: • **Extension –** Set to **Phone Number** field value configured in **Section 3.6 Step 1**. • **Communication Manager Server –** Set to the co-resident Communication Manager where this OPS station is configured. • Click **Add** and then click **Continue** on the next page [not shown]. **Note:** Communication Manager Server was previously configured during the initial setup of SES. 
5.	Repeat **Steps 3 and 4** as necessary to configure additional G-Tek telephones.

5. Configure G-Tek AQ10x Telephones

This section describes the steps for configuring the G-Tek AQ10x 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.

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

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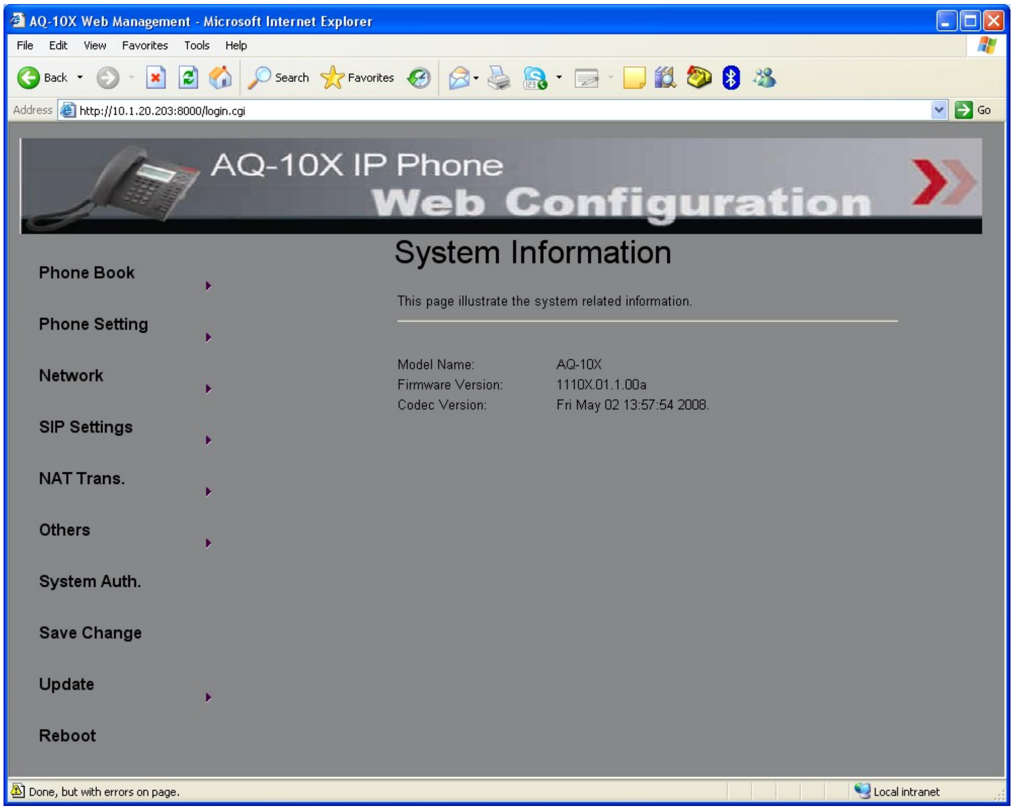
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Open a web browser and enter http://a.b.c.d:8000 for the URL, where a.b.c.d is the IP address of the G-Tek AQ10x telephone. Log in with the appropriate credentials and the System Information page will be displayed.

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

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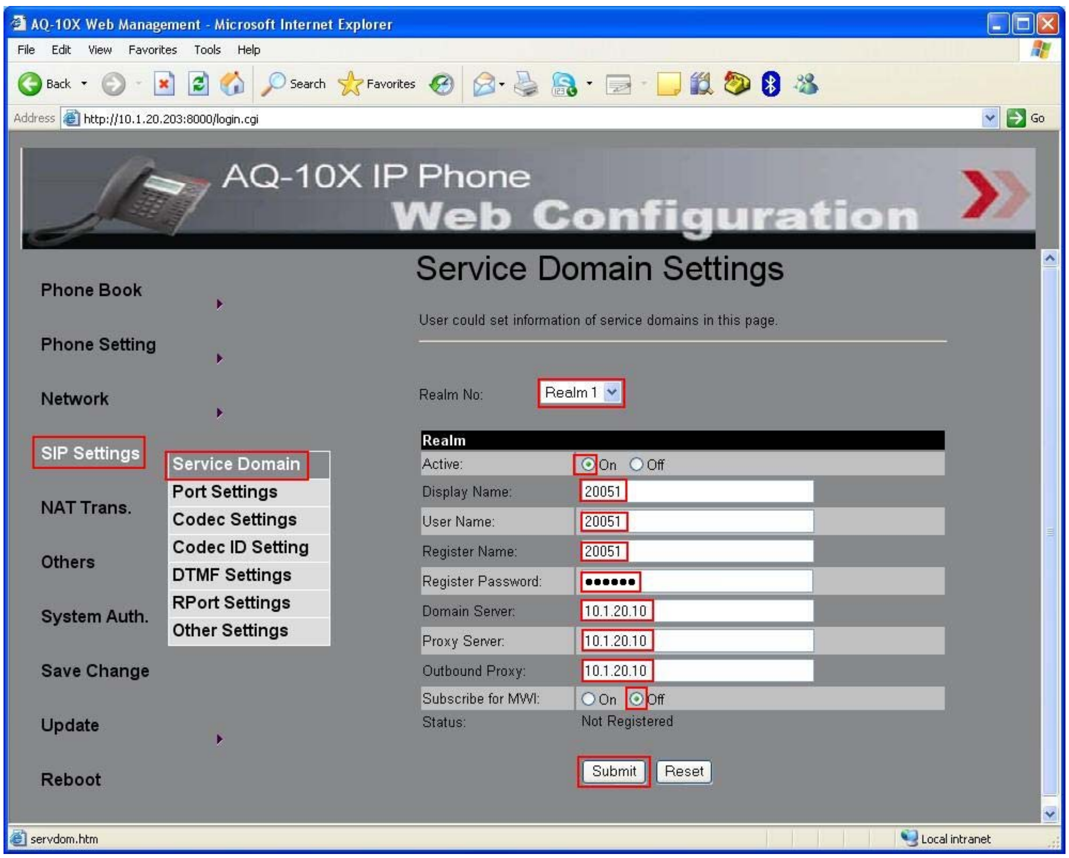
Step	Description
2.	Click **SIP Settings > Service Domain** on the left navigation menu. At the Service Domain Settings page, configure the following: • **Realm No –** Select **Realm 1**. • **Active –** Set to **On**. • **Display Name, User Name** and **Register Name –** Set to the **Primary Handle** field value configured in **Section 4 Step 3**. • **Register Password –** Set to the **Password** field value configured in **Section 4 Step 3**. • **Domain Server, Proxy Server** and **Outbound Proxy –** Set to the Avaya SES server IP address. • **Subscribe for MWI –** Select **Off**. Click **Submit** to continue. 

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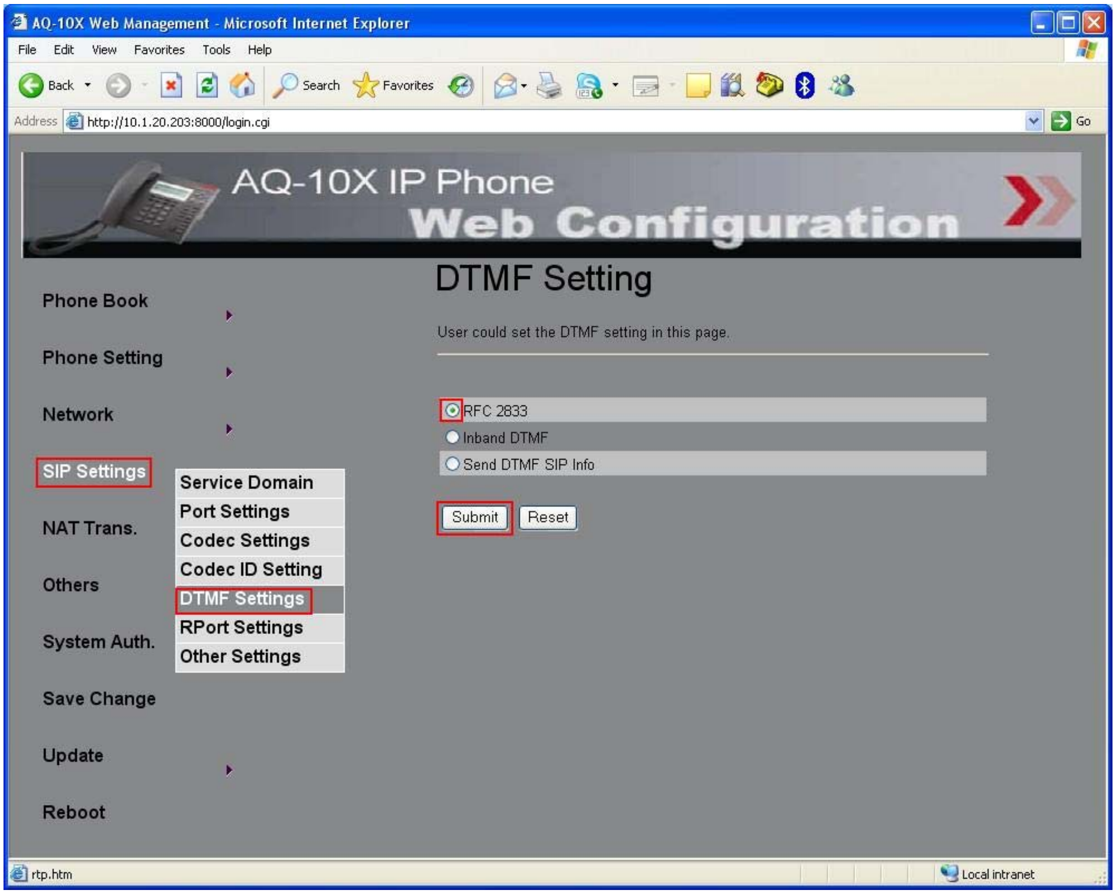
Step	Description
3.	Click **SIP Settings** > **DTMF Settings** on the left navigation menu. At the DTMF Settings page, select **RFC 2833** and click **Submit** to continue. 

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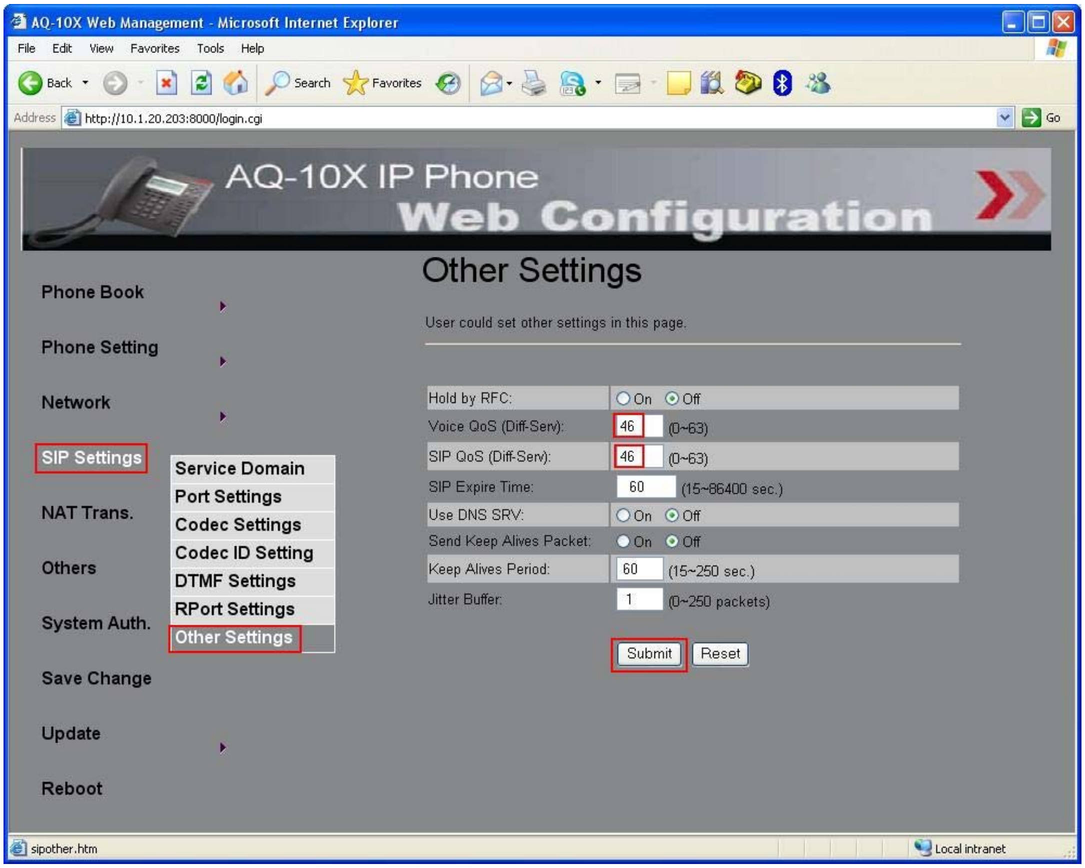
Step	Description
4.	Click **SIP Settings > Other Settings** on the left navigation menu. At the Other Settings page, set **Voice QoS (Diff-Serv)** and **SIP QoS (Doff-Serv)** to the desired value between 0 and 63. For compliance testing, a value of **46** was used. Click **Submit** to continue. 

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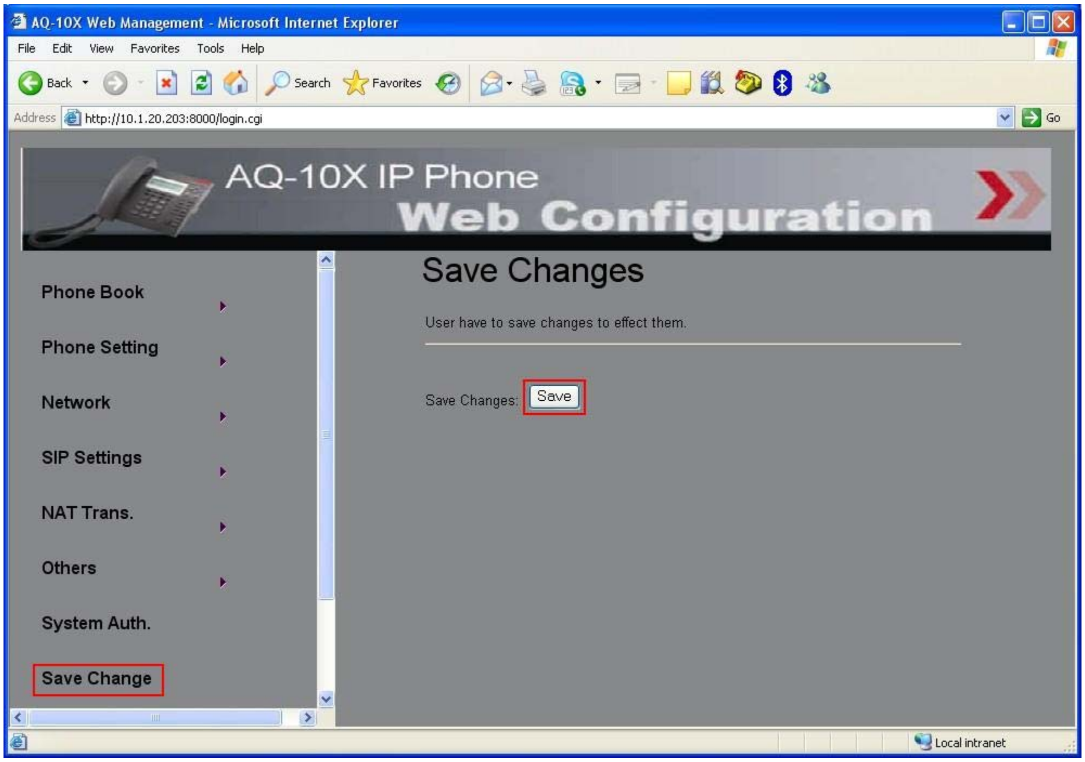
Step	Description
5.	Click **Save Change** on the left navigation menu. At the Save Changes page, click **Save** to save the changes made and the telephone will reboot. 

6. Interoperability Compliance Testing

The focus of the interoperability compliance testing was primarily on verifying call establishment on the G-Tek AQ10x SIP telephones and operations such as dialing methods (manual, re-dial, and phone book), hold, mute, transfer and conference. G-Tek AQ10x 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 AQ10x 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.

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- 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:

- The G-Tek telephone could not setup a 3-party conference successfully.
- OPS features such as Call Pickup, Directed Call Pickup, Extended Group Call Pickup, 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

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<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>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.
<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></td></tr>
<tr><td>2.</td><td>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:

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. 20.206 : 20000   10. 1. 20.203 : 20000
            Video:
      Video Codec:
                                          Authentication Type: None
    Audio Connection Type: ip-direct
</pre></td></tr>
</table>

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8. Support

For technical support on G-Tek AQ10x 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 AQ10x 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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