



## **Avaya Solution & Interoperability Test Lab**

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# **Application Notes for Configuring G-Tek Electronics SSP-9x10-SM SIP Phone with Avaya Aura® Session Manager 6.1 and Avaya Aura® Communication Manager 6.0.1 - Issue 1.0**

## **Abstract**

These Application Notes describe the configuration steps required for G-Tek Electronics SSP-9x10-SM SIP Phone with DECT handset to interoperate with Avaya Aura® Session Manager 6.1 and Avaya Aura® Communication Manager 6.0.1.

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 the steps required to configure G-Tek Electronics SSP-9x10-SM SIP Phone to interoperate with an Avaya SIP infrastructure consisting of Avaya Aura® Session Manager 6.1 and Avaya Aura® Communication Manager 6.0.1. G-Tek SSP-9x10-SM is a series of business IP phones for the enterprise.

## 2. General Test Approach and Test Results

To verify interoperability of G-Tek SSP-9x10-SM SIP Phone with Session Manager and Communication Manager, calls were made between G-Tek telephones and Avaya SIP, H.323 and Digital telephones using various codec settings and exercising common PBX features. The telephony features were activated and deactivated using speed-dial buttons.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

### 2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Successful registration of G-Tek SSP-9x10-SM SIP Phones with Session Manager.
- Calls between G-Tek telephones and Avaya SIP, H.323, and digital telephones.
- G.711 and G729/B codec support and negotiation, with and without media shuffling.
- Basic features including phone display, mute/un-mute, answer, hang up, music on hold, DTMF transmission, Message Waiting Indicator (MWI) subscription and feature access code dialing.
- PBX features including Multiple Call Appearances, Hold, Transfer, and Conference.
- Proper system recovery after a G-Tek telephone restart and loss of IP connection.

### 2.2. Test Results

All test cases were executed and passed successfully. However, the following points were noted during the testing:

- When making changes in codec settings on the web interface while the phones are in talking state, call drops. There is no warning is given on web page.
- Data sheet does not indicate support of G.729 codec but the administration web interface and testing does indicate that this codec is supported.

## 2.3. Support

Technical support from G-Tek Electronics can be obtained through the following:

- Phone: +886-2-26962665 ext. 221
- E-mail: [support@G-Tek.com.tw](mailto:support@G-Tek.com.tw)

## 3. Reference Configuration

The diagram illustrates an enterprise site with an Avaya SIP-based network, including Session Manager, an S8800 Server running Communication Manager with a G450 Media Gateway, and Avaya SIP, H.323 and Digital endpoints. The enterprise site also contains two G-Tek SSP-9x10-SM SIP Phones used in the compliance testing. The G-Tek phones are registered with Session Manager and are configured as endpoint users.

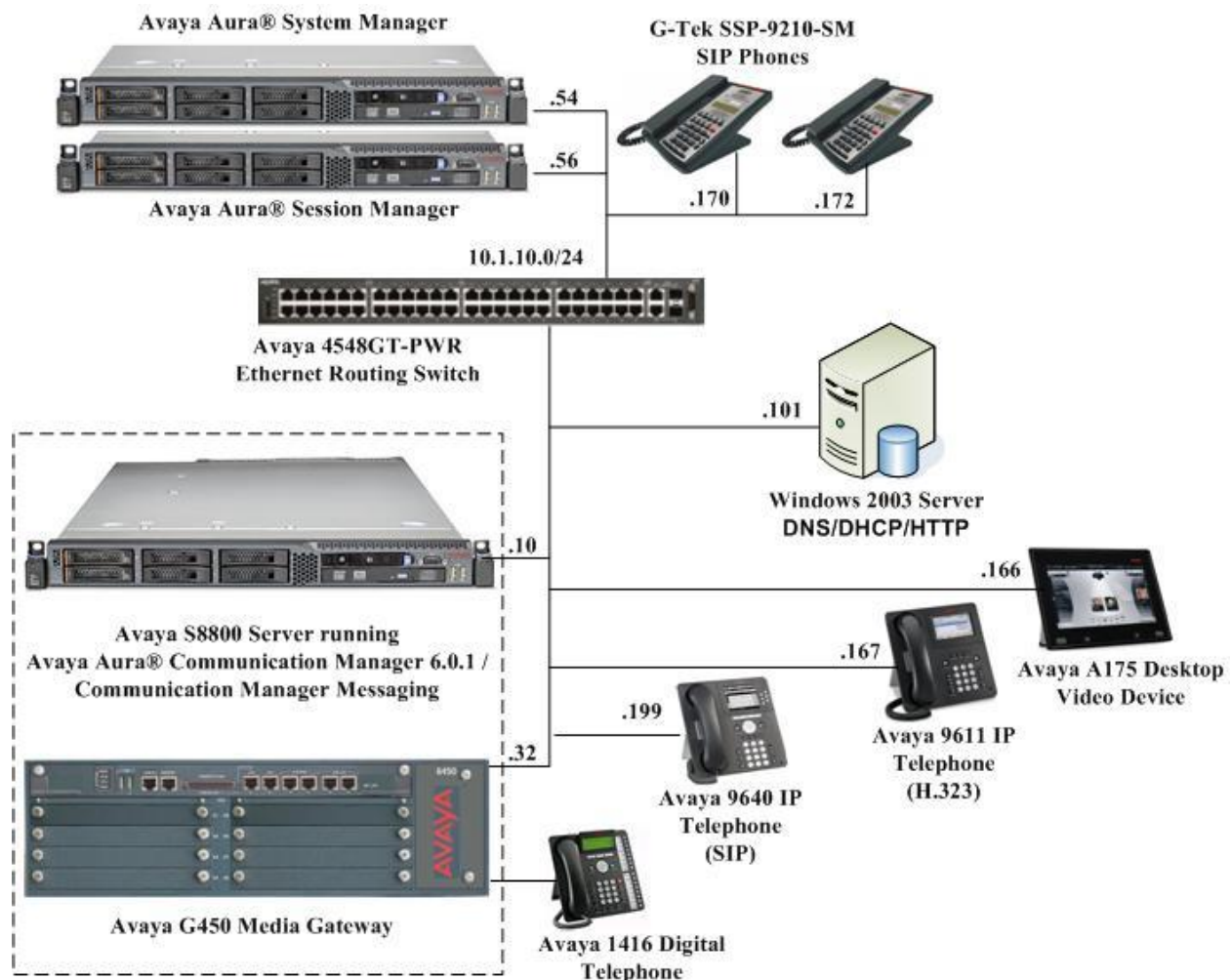


Figure 1: G-Tek SSP-9x10-SM SIP Phones with Avaya SIP Solution

**Table 1** lists the extensions used for this testing.

<b>Extension</b>	<b>Note</b>
10004	Avaya 9611 IP Telephone (H.323)
10051	Avaya 9640 IP Telephone (SIP)
481122	Avaya 1416 Digital Telephone
10062	Avaya Desktop Video Device (ADVD) (SIP)
10071 to 10072	G-Tek SSP-9x10-SM SIP Phones

**Table 1 – Extension Setup**

## **4. Equipment and Software Validated**

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

<b>Equipment/Software</b>	<b>Release/Version</b>
Avaya Aura® Communication Manager and Avaya Aura® Communication Messaging Manager on Avaya S8800 Server	R6.0.1 SP 8
Avaya G450 Media Gateway	31.20.0
Avaya Aura® Session Manager on Avaya S8800 Server	R6.1 SP 5
Avaya Aura® System Manager on Avaya S8800 Server	R6.1 SP 7
Avaya 9640IP Telephone (SIP)	2.6 SP5
Avaya 9611 IP Telephone (H.323)	6.0 SP5
Avaya 1408 Digital Telephone	-
Avaya A175 Desktop (SIP)	1.0.3
Avaya 4548GT-PWR Ethernet Routing Switch	V5.4.0.008
G-Tek SSP-9x10-SM SIP Phones	Firmware Version: 1990X.16.1.02N Codec Version: Fri Mar 26 10:47:24 2004

## 5. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Avaya Aura® Session Manager. The procedures include the following areas:

- Launch System Manager
- Administer users

### 5.1. Launch System Manager

Access the System Manager Web interface by using the URL “https://ip-address” in an Internet browser window, where “ip-address” is the IP address of the System Manager server. Log in using the appropriate credentials.

The screenshot shows the Avaya Aura® System Manager 6.1 web interface. At the top left is the Avaya logo, and to its right is the text "Avaya Aura® System Manager 6.1". Below this is a red navigation bar with the text "Home / Log On". Underneath the bar, the heading "Log On" is displayed. The main content area is divided into two columns. The left column contains a box with the following text: "Recommended access to System Manager is via FQDN." followed by a blue hyperlink "Go to central login for Single Sign-On". Below this, it states: "If IP address access is your only option, then note that authentication will fail in the following cases:" followed by a bulleted list: "• First time login with 'admin' account" and "• Expired/Reset passwords". At the bottom of this box, it says: "Use the 'Change Password' hyperlink on this page to change the password manually, and then login." The right column contains the login fields: "User ID:" followed by a text input box, and "Password:" followed by a password input box. Below these fields are two buttons: "Log On" and "Cancel". At the bottom right of the page, there is a blue hyperlink "Change Password".

## 5.2. Administer Users

In the subsequent screen (not shown), select **Users** → **User Management** → **Manage Users** to display the **User Management** screen below. Click **New** to add a user.

The screenshot shows the Avaya Aura System Manager 6.1 interface. The top navigation bar includes the Avaya logo, the title 'Avaya Aura® System Manager 6.1', and links for 'Help', 'About', 'Change Password', and 'Log off admin'. Below this is a breadcrumb trail: 'Home / Users / User Management / Manage Users - User Management'. The left sidebar contains a 'User Management' menu with options: 'Manage Users', 'Public Contacts', 'Shared Addresses', and 'System Presence ACLs'. The main content area is titled 'User Management' and features a 'Users' section with buttons for 'View', 'Edit', 'New' (circled in red), 'Duplicate', 'Delete', and 'More Actions'. Below these buttons is a table with 26 items, showing a list of users. The table has columns for 'Status', 'Name', 'Login Name', 'E164 Handle', and 'Last Login'. The 'New' button is highlighted with a red circle.

Status	Name	Login Name	E164 Handle	Last Login
1XC SIPUser1	10063@sglab.com	10063		
1XC SIPUser2	10064@sglab.com	10064		
ADVD User1	10061@sglab.com	10061		
ADVD User2	10062@sglab.com	10062		
Avaya, SIP1	10051@sglab.com	10051		

### 5.2.1. Identity

The **New User Profile** screen is displayed. Enter desired **Last Name** and **First Name**. For **Login Name**, enter “n@z”, where “n” is the first G-Tek SSP-9x10 user extension and “z” is the domain name used for compliance testing, in this case “**sglab.com**”. For **Password** and **Confirm Password**, enter the appropriate credentials for SIP user for registration. Retain the default values in the remaining fields.

The screenshot displays the Avaya Aura® System Manager 6.1 web interface. The top navigation bar includes the Avaya logo, the product name, and links for Help, About, Change Password, and Log off admin. Below this, a breadcrumb trail shows the path: Home / Users / User Management / Manage Users - New User Profile. The left sidebar contains a menu with 'User Management' expanded, showing 'Manage Users' as the active option, along with 'Public Contacts', 'Shared Addresses', and 'System Presence ACLs'. The main content area is titled 'New User Profile' and features four tabs: 'Identity' (selected), 'Communication Profile', 'Membership', and 'Contacts'. The 'Identity' tab contains the following fields: 'Last Name' (GTek), 'First Name' (SSP9210-1), 'Middle Name' (10071@sglab.com), 'Description' (empty), 'Login Name' (empty), 'Authentication Type' (Basic), 'Password' (masked with dots), and 'Confirm Password' (masked with dots). 'Commit' and 'Cancel' buttons are located at the top right of the form area.

### 5.2.2. Communication Profile

Select the **Communication Profile** tab. For **Communication Profile Password** and **Confirm Password**, enter the desired password for the SIP user to use for registration. Scroll down to the **Communication Address** sub-section, and click **New** to add a new address.

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

User Management x Home

Home / Users / User Management / Manage Users - New User Profile

Help ?

### New User Profile

Commit Cancel

Identity \* Communication Profile \* Membership Contacts

Communication Profile ▾

Communication Profile Password: ••••••

Confirm Password: ••••••

New Delete Done Cancel

Name
Primary

Select : None

\* Name: Primary

Default : ☒

Communication Address ▾

New Edit Delete

Type	Handle	Domain
No Records found		

For **Type**, retain “Avaya SIP”. For **Fully Qualified Address**, enter and select the SIP user extension and domain name from **Section 5.2.1**. Click **Add**.

Communication Address ▾

New Edit Delete

Type	Handle	Domain
No Records found		

Type: Avaya SIP ▾

\* Fully Qualified Address: 10071 @ sglab.com ▾

Add Cancel



Scroll down to check and expand **Session Manager Profile**. For **Primary Session Manager**, **Origination Application Sequence**, **Termination Application Sequence**, and **Home Location**, select the values corresponding to the applicable Session Manager. Retain the default values in the remaining fields. These settings are configured during the initial setup of Session Manager.

Scroll down to check and expand **Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter the SIP user extension from **Section 5.2.1**. For **Template**, select “**DEFAULT\_9630SIP\_CM\_6\_0**”. For **Port**, select “**IP**”. Retain the default values in the remaining fields.

☒ **Session Manager Profile**

\* Primary Session Manager

me1-sm

Secondary Session Manager

(None)

Primary	Secondary	Maximum
26	0	26

Origination Application Sequence

cm6-site1-app-seq

Termination Application Sequence

cm6-site1-app-seq

Survivability Server

(None)

\* Home Location

Location1

☒ **Endpoint Profile**

\* System

cm6-site1

\* Profile Type

Endpoint

Use Existing Endpoints

☐

\* Extension

10071

Endpoint Editor

\* Template

DEFAULT\_9630SIP\_CM\_6\_0

Set Type

9630SIP

Security Code

\* Port

IP

Voice Mail Number

Delete Endpoint on Unassign of Endpoint from User or on Delete User.

☐

Scroll down to check and expand on **Messaging Profile**. For **System**, select the value corresponding to the applicable Communication Messaging Manager. For **Mailbox Number**, select the same user extension number. For Template, select “**DEFAULT\_CMM\_6\_0**”. For **Password**, enter the appropriate mailbox password. Click “**Delete Subscriber on Unassign of Subscriber from User or on Delete User**”.

✓ **Messaging Profile**

**System** cm6-site1-cmm

**Use Existing Subscriber on System** ☐

\* **Mailbox Number** 10071 Messaging Editor

**Template** DEFAULT\_CMM\_6\_0

\* **Password** •••••

**Delete Subscriber on Unassign of Subscriber from User or on Delete User** ☒

Click **Commit** to complete the creation of the new user (not shown).

Repeat **Section 5.2** to add a user for each G-Tek SSP-9X10 user. In the compliance testing, two users with extensions “10071” and “10072” were added.

## 6. Configure Avaya Aura® Communication Manager

This section provides the procedures for configuring Communication Manager. The procedures include the following areas:

- Verify Communication Manager license
- Administer IP codec set

### 6.1. Verify Communication Manager License

Log in to the System Access Terminal (SAT) to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the “**display system-parameters customer-options**” command to verify that there is sufficient capacity for SIP stations by comparing the **Maximum Off-PBX Telephones - OPS** field value with the corresponding value in the **USED** column. The difference between the two values needs to be greater than or equal to the number of G-Tek SSP-9x10 SIP Phone extensions.

The license file installed on the system controls the maximum permitted. If there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.

<b>display system-parameters customer-options</b>	Page 1 of 11
OPTIONAL FEATURES	
G3 Version: V16	Software Package: Enterprise

```
System ID (SID): 1
Module ID (MID): 1
```

(NOTE: You must logoff & login to effect the permission changes.)

## 6.2. Administer IP Codec Set

Use the **change ip-codec-set n** command, where **n** is the existing codec set number associated with the SIP trunk group to Session Manager. Update the audio codec types in the **Audio Codec** fields as necessary to include G.711MU, G.711A, G.729 and G.729B.

change ip-codec-set 6

Page 1 of 2

IP Codec Set

Codec Set: 6

	Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size(ms)
1:	G.711A	n	2	20
2:	G.711MU	n	2	20
3:	G.729	n	2	20
4:	G.729B	n	2	20
5:				
6:				
7:				

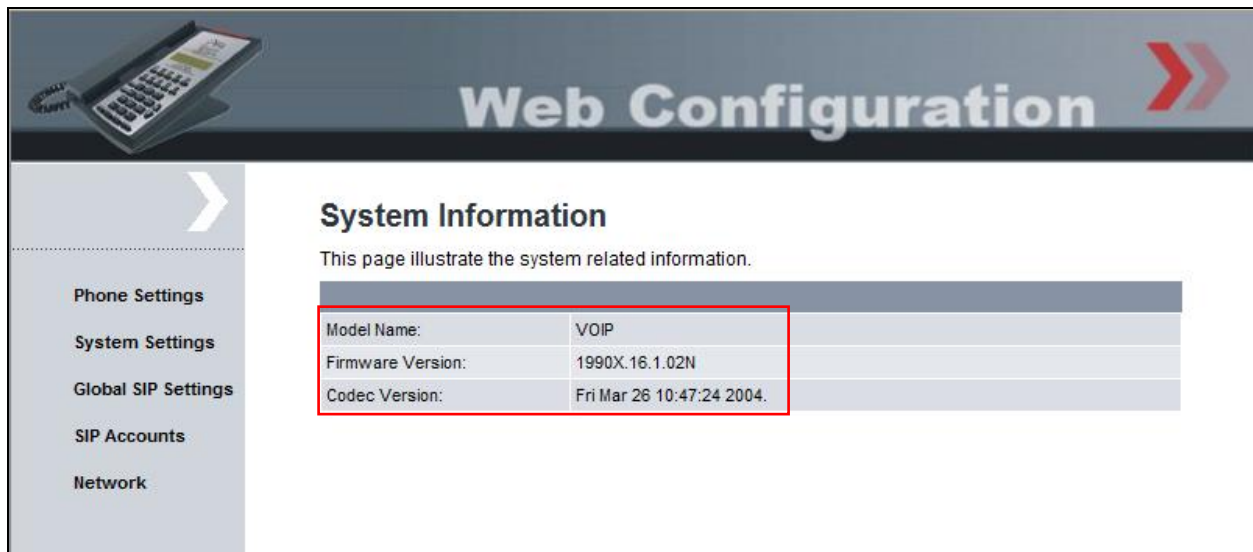
## 7. Configure G-Tek SSP-9x10 SIP Phones

This section provides the procedures for configuring G-Tek SSP-9x10 SIP Phones. The procedures include the following areas:

- Access Web Interface
- Configure SIP Account and DTMF Settings
- Configure Audio Codecs
- Configure Tone Settings
- Reboot after configuration

### 7.1. Access Web Interface

Enter <http://<ip-addr>:8000/>, where <ip-addr> is the IP address of the G-Tek phone, into the address bar of web browser and log in using a valid account. The **System Information** screen is displayed.



The screenshot displays the 'Web Configuration' interface for a G-Tek SIP phone. At the top, there is a header with a phone icon and the text 'Web Configuration' followed by a red double arrow. Below the header, a left sidebar contains a list of menu items: 'Phone Settings', 'System Settings', 'Global SIP Settings', 'SIP Accounts', and 'Network'. The main content area is titled 'System Information' and includes the text 'This page illustrate the system related information.' Below this text is a table with system information. The table has two columns: 'Model Name:', 'Firmware Version:', and 'Codec Version:' in the first column, and their corresponding values in the second column. The values are 'VOIP', '1990X.16.1.02N', and 'Fri Mar 26 10:47:24 2004.' respectively. The table is highlighted with a red border.

Model Name:	VOIP
Firmware Version:	1990X.16.1.02N
Codec Version:	Fri Mar 26 10:47:24 2004.

## 7.2. Configure SIP Account and DTMF Settings

Select **SIP Accounts** from the left menu. Set the **Active** field to **Enable**. Enter the IP address of Session Manager signaling interface as shown in **Figure 1** for the **Registration Server** and **Proxy Server** fields. For the fields **Registration ID**, **Display Name**, **Authorization Name** and **Password**, enter the account details as shown below to match the user settings in Session Manager added in **Section 5.2**. Set **MWI** to **Enable**. Set the **Voice Mail** number to dial as **10000**. This number is map to the message speed dial key on the base. Check the default **DTMF Type** settings is **RFC2833**. Click **Submit** to continue.



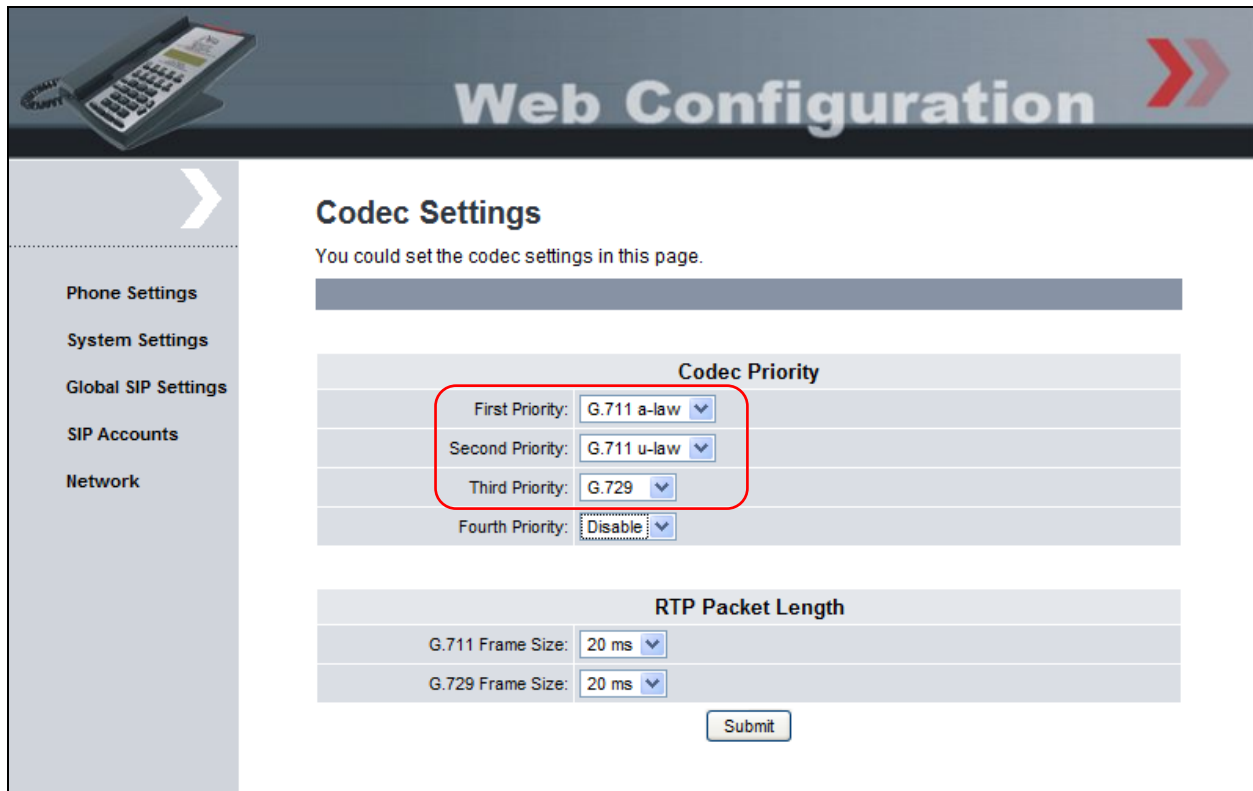
The image shows a web configuration interface for SIP Account Settings. The left sidebar contains a menu with the following items: Phone Settings, System Settings, Global SIP Settings, SIP Accounts (highlighted), and Network. The main content area is titled "SIP Account Settings" and includes a sub-header "You could set information of service domains in this". Below this is a form for "SIP Account 1". The form fields are as follows:

SIP Account 1	
Active:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Registration ID:	10071
Display Name:	10071
Authentication Name:	10071
Password:	*****
Registration Server:	10.1.10.56
Proxy Server:	10.1.10.56
Proxy Address:	
Voice Mail:	10000
Expire Time:	60
DTMF Type:	RFC2833
Send KeepAlive:	Disable
MWI:	Enable
Mode:	Single
Status:	registered

At the bottom of the form are two buttons: "Submit" and "cancel".

### 7.3. Configure Audio Codecs

Select **Global SIP Settings** → **Codec Settings** from the left menu. In the **Codec Priority** section, prioritize the audio codecs accordingly. Click **Submit** to continue.



The image shows a web configuration interface for a device. At the top, there is a header with a phone icon and the text "Web Configuration". Below this is a left sidebar with a menu containing "Phone Settings", "System Settings", "Global SIP Settings", "SIP Accounts", and "Network". The "Global SIP Settings" option is selected, and the "Codec Settings" page is displayed. The page title is "Codec Settings" with a subtitle "You could set the codec settings in this page." The main content area has two sections: "Codec Priority" and "RTP Packet Length". The "Codec Priority" section has four rows, each with a label and a dropdown menu. The first three rows are highlighted with a red box. The "RTP Packet Length" section has two rows, each with a label and a dropdown menu. A "Submit" button is located at the bottom right of the form.

Codec Priority	
First Priority:	G.711 a-law
Second Priority:	G.711 u-law
Third Priority:	G.729
Fourth Priority:	Disable


  

RTP Packet Length	
G.711 Frame Size:	20 ms
G.729 Frame Size:	20 ms

Submit

## 7.4. Configure Tone Settings

Select **Phone Settings** → **Tone Settings** from the left menu. Select “**musiconhold.snd**” for **Music On Hold** in order to provide music while the call is on hold. Set **Call Waiting Tone** to **Enable**. Click **Submit** to continue.



The screenshot shows the 'Web Configuration' interface. On the left is a sidebar menu with 'Phone Settings' selected. The main area is titled 'Tone Setting' and contains a form with the following fields: 'Ringtone Name' (set to 'ringtone.snd'), 'Music On Hold' (set to 'musiconhold.snd'), 'Reorder Tone Play Time' (set to '30'), 'Call Waiting Tone' (set to 'Enable'), and 'Call Waiting Tone Repeat' (set to 'Disable'). The 'Music On Hold' and 'Call Waiting Tone' fields are highlighted with red boxes. At the bottom of the form are 'Submit' and 'Reset' buttons.

## 7.5. Reboot after configuration

Select **System Settings** → **Reboot** from the left menu to reboot the phone after setting the various parameters for the phones.



This screenshot shows the 'Web Configuration' interface with the 'System Settings' menu open. The 'Reboot' option is highlighted with a red box. The main area still shows the 'Tone Setting' form, which is identical to the one in the previous screenshot, with 'Music On Hold' and 'Call Waiting Tone' highlighted by red boxes. The 'Submit' and 'Reset' buttons are visible at the bottom.



## 8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Session Manager and G-Tek SSP-9x10 SIP Phones.

From the System Manager Web interface, select **Elements → Session Manager → System Status → User Registrations** to display the **User Registrations** screen. Verify that the users from **Section 5.2** are registered, as shown below with a check in the **Registered Prim** column.

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Session Manager \* Home

Home / Elements / Session Manager / System Status / User Registrations - User Registrations

Help ?

### User Registrations

Select rows to send notifications to AST devices. Click on Details column for complete registration status.

Customize ▶

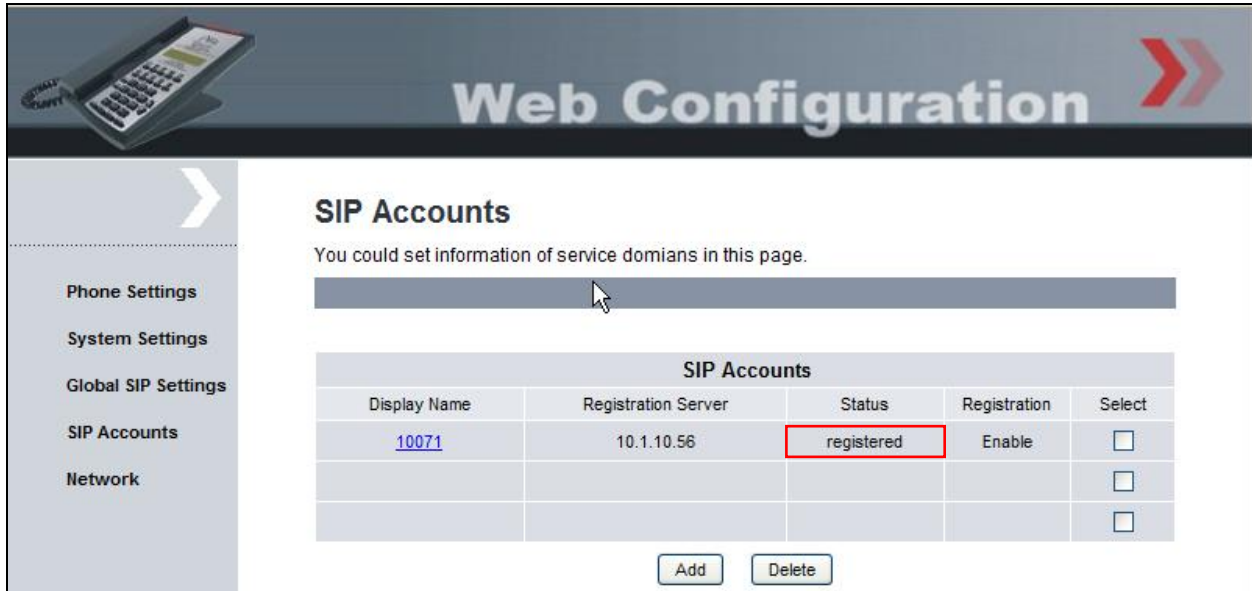
AST Device Notifications: Reboot Reload ▼ Failback As of 1:58 PM

Advanced Search ▶

25 Items Refresh Show 15 ▼ Filter: Enable

	Details	Address	Login Name ▲	First Name	Last Name	Location	IP Address	AST Device	Registered		
									Prim	Sec	Surv
<input type="checkbox"/>	▶ Show	---	10065@sglab.com	One	G502N	Location1	---	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	▶ Show	---	10066@sglab.com	Two	G502N	Location1	---	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	▶ Show	---	10067@sglab.com	One	IP652	Location1	---	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	▶ Show	---	10068@sglab.com	Two	IP652	Location1	---	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	▶ Show	10071@sglab.com	10071@sglab.com	SSP9210-1	GTek	Location1	10.1.10.170:5060	<input type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	▶ Show	10072@sglab.com	10072@sglab.com	SSP9210-2	GTek	Location1	10.1.10.172:5060	<input type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	▶ Show	10073@sglab.com	10073@sglab.com	SGR8206-1	GTek	Location1	10.1.10.177:5060	<input type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	▶ Show	10074@sglab.com	10074@sglab.com	SGR8206-2	GTek	Location1	10.1.10.173:5060	<input type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>

From the web interface of the G-Tek SSP-9x10 phone, click **SIP Accounts** from the left menu. Verify that the **Status** field shows as **registered**.



The screenshot shows the 'Web Configuration' interface of a G-Tek SSP-9x10 phone. The left sidebar contains a menu with the following items: Phone Settings, System Settings, Global SIP Settings, SIP Accounts (highlighted), and Network. The main content area is titled 'SIP Accounts' and includes a sub-header 'You could set information of service domains in this page.' Below this is a table with the following columns: Display Name, Registration Server, Status, Registration, and Select. The first row of the table shows a Display Name of '10071', a Registration Server of '10.1.10.56', and a Status of 'registered' (highlighted with a red box). The Registration column shows 'Enable' and the Select column shows a checkbox. Below the table are 'Add' and 'Delete' buttons.

Display Name	Registration Server	Status	Registration	Select
<a href="#">10071</a>	10.1.10.56	registered	Enable	<input type="checkbox"/>
				<input type="checkbox"/>
				<input type="checkbox"/>

## 9. Conclusion

These Application Notes describe the configuration steps required for G-Tek Electronics SSP-9x10-SMK SIP Phones to successfully interoperate with Avaya Aura® Session Manager 6.1 and Avaya Aura® Communication Manager 6.0.1. All feature and serviceability test cases were completed successfully.

## 10. Additional References

This section references documentation relevant to these Application Notes. Avaya product documentation is available at <http://support.avaya.com>.

[1] *Administering Avaya Aura® Communication Manager*, Release 6.0, Doc ID 03-300509, June 2010.

[2] *Administering Avaya Aura® Session Manager*, Release 6.1, Doc ID 03-603324, Issue 1, November 2010.

[3] *Implementing Avaya Aura® Communication Manager Messaging*, Release 6.0, Doc ID 18-603644, June 2010

[4] *Administrator Guide for SSP9210-SM/SAX-8210P/SGR8210-SMK*

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