



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring G-Tek/AEi Electronics SSP-2x10-S SIP Phone with Avaya Aura® Session Manager 6.3 and Avaya Aura® Communication Manager 6.3 - Issue 1.0

Abstract

These Application Notes describe the configuration steps required for G-Tek/AEi Electronics SSP-2x10-S SIP Phone to interoperate with Avaya Aura® Session Manager 6.3 and Avaya Aura® Communication Manager 6.3.

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/AEi Electronics SSP-2x10-S SIP Phone to interoperate with an Avaya SIP infrastructure consisting of Avaya Aura® Session Manager 6.3 and Avaya Aura® Communication Manager 6.3. G-Tek/AEi SSP-2x10-S is part of a series of hospitality IP phones.

2. General Test Approach and Test Results

To verify interoperability of G-Tek/AEi SSP-2x10-S SIP Phone with Avaya Aura® Session Manager (Session manager) and Avaya Aura® Communication Manager (Communication Manager), calls were made between G-Tek/AEi telephones and Avaya telephones (SIP and H.323) using various codec settings and exercising common PBX features. The telephony features were activated and deactivated using speed-dial buttons. It also includes serviceability test where the G-Tek/AEi telephones Ethernet cables were disconnected.

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/AEi SSP-2x10-S SIP Phones with Session Manager.
- Calls between G-Tek/AEi telephones and Avaya IP telephones (SIP and H.323).
- G.711 and/or 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.
- Hospitality features like automatic wakeup activation and call, housekeeping status update and programmable buttons.
- PBX features including Multiple Call Appearances, Call Waiting, Call Hold, Transfer, Conference and Multi-Device Access
- Proper system recovery after a G-Tek/AEi telephone restart and loss of IP connection.

2.2. Test Results

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

- G.729 codec is not supported
- Blind/Attended Transfer is not supported as this is a hospitality telephone.
- Blind Conference is not supported.
- Multi-Device Access is not supported.

2.3. Support

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

- Phone: +886-2-26962665
- E-mail: support@G-Tek.com.tw

3. Reference Configuration

The diagram illustrates an enterprise site with an Avaya SIP-based network, including Session Manager, Messaging server, duplex S8800 Server running Communication Manager with a G430 Media Gateway, and Avaya SIP and H.323 IP endpoints. The enterprise site also contains three G-Tek/AEi SSP-2x10-S SIP Phones used in the compliance testing. The G-Tek/AEi phones are registered with Session Manager and are configured as endpoint users.

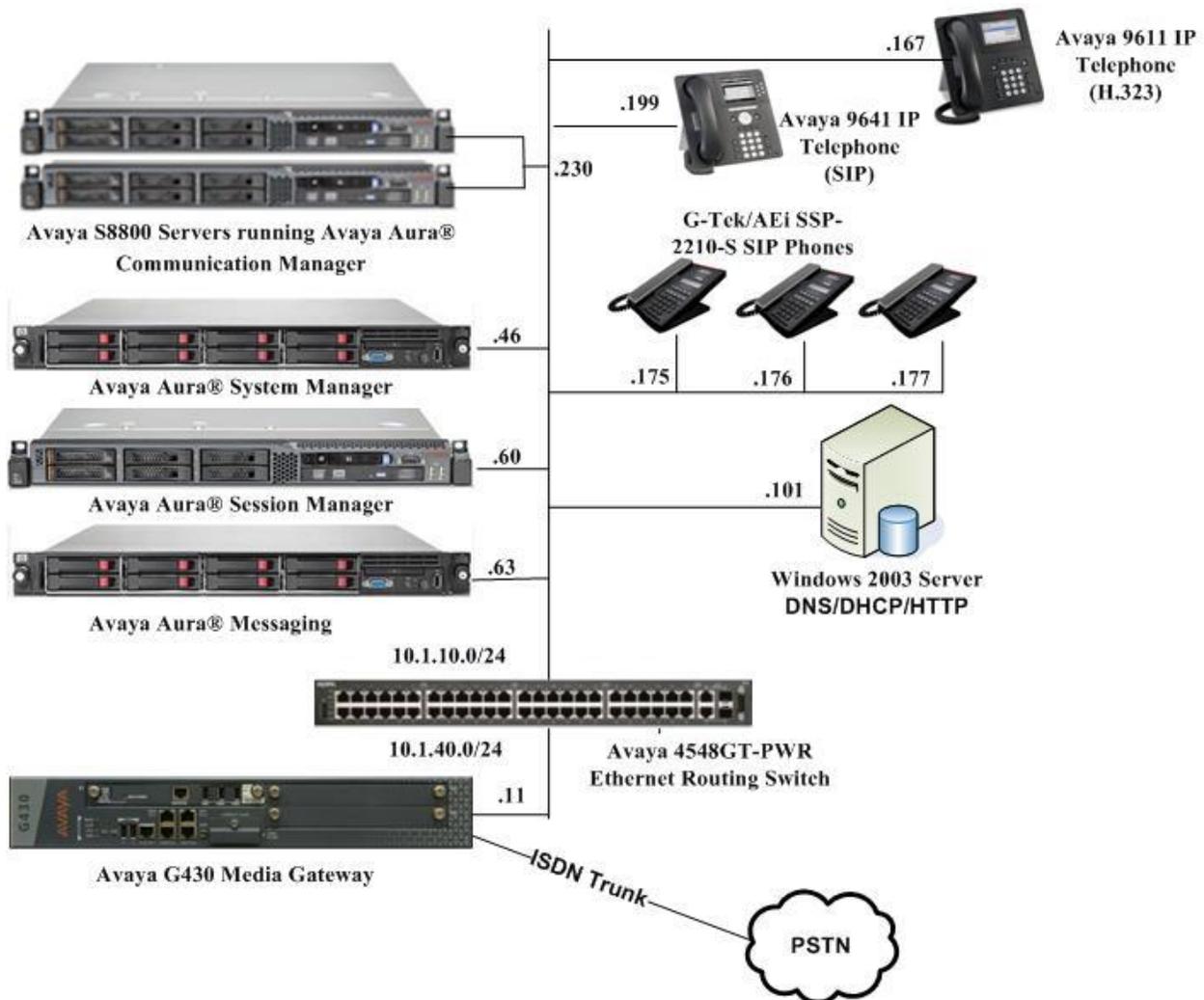


Figure 1: G-Tek/AEi SSP-2x10-S SIP Phones with Avaya SIP Solution

Table 1 lists the extensions used for this testing.

Extension	Note
10001	Avaya 9611 IP Telephone (H.323)
10049	Avaya 9640 IP Telephone (SIP)
10045 to 10047	G-Tek/AEi SSP-2x10-S SIP Phones

Table 1 – Extension Setup

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager on duplex Avaya S8800 Server	R6.3 SP3
Avaya G430 Media Gateway	34.5.1
Avaya Aura® Session Manager on Avaya S8800 Server	R6.3 SP 5
Avaya Aura® System Manager on HP DL360	R6.3 SP5 Patch 1
Avaya Aura® Messaging on HP DL360	R6.2 SP3
Avaya 9641 IP Telephone (SIP)	6.3
Avaya 9611 IP Telephone (H.323)	6.3
Avaya 4548GT-PWR Ethernet Routing Switch	V5.4.0.008
G-Tek/AEi SSP-2x10-S SIP Phones	Firmware Version: 190220.16.2.01A7

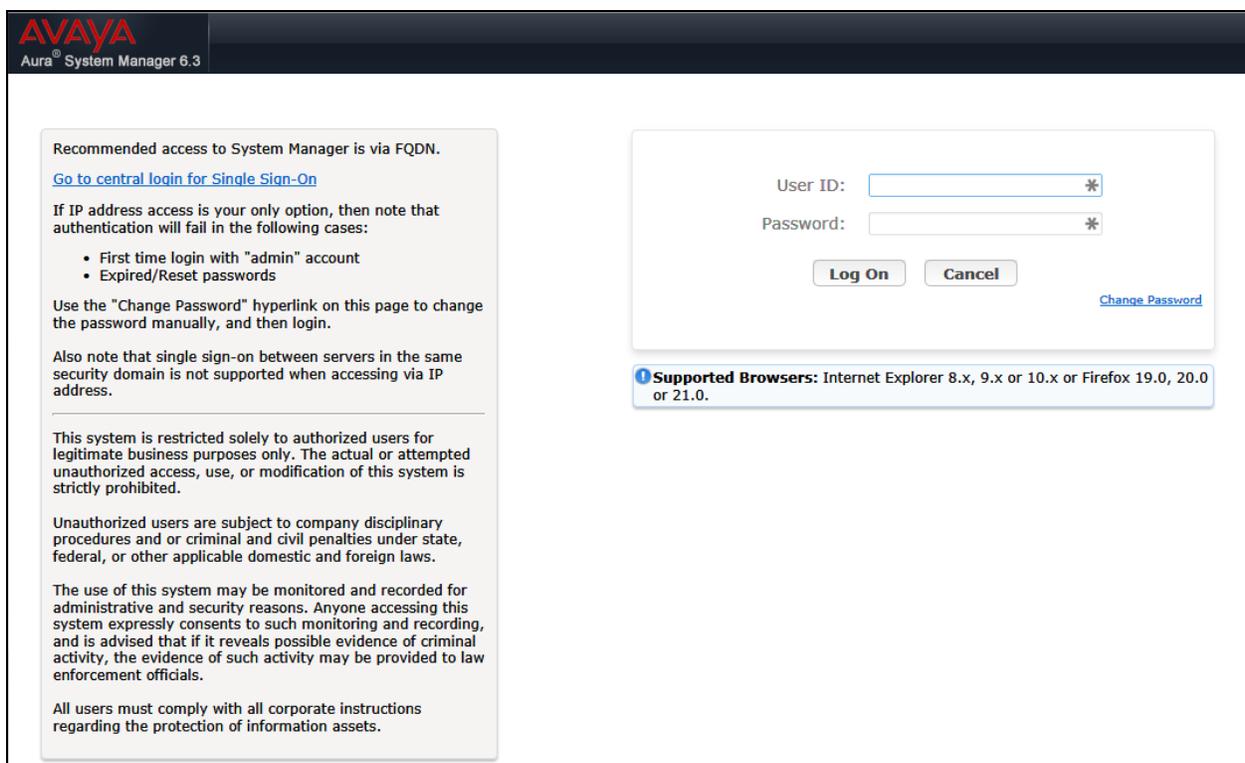
5. Configure Avaya Aura® Session Manager

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.



AVAYA
Aura® System Manager 6.3

Recommended access to System Manager is via FQDN.
[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

Use the "Change Password" hyperlink on this page to change the password manually, and then login.

Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address.

This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use, or modification of this system is strictly prohibited.

Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal, or other applicable domestic and foreign laws.

The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials.

All users must comply with all corporate instructions regarding the protection of information assets.

User ID: *

Password: *

[Change Password](#)

Supported Browsers: Internet Explorer 8.x, 9.x or 10.x or Firefox 19.0, 20.0 or 21.0.

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 6.3 interface. The top navigation bar includes 'Home' and 'User Management'. A sidebar on the left lists 'User Management' sub-items: 'Manage Users', 'Public Contacts', 'Shared Addresses', 'System Presence ACLs', 'Communication Profile', and 'Password Policy'. The main content area is titled 'User Management' and contains a search bar and a 'Help ?' link. Below this is a 'Users' section with a toolbar containing 'View', 'Edit', 'New' (highlighted with a red box), 'Duplicate', 'Delete', and 'More Actions'. A table below the toolbar shows 9 items with columns for 'Last Name', 'First Name', 'Display Name', 'Login Name', 'SIP Handle', and 'Last Login'. The 'admin' user is listed with a last login of 'February 14, 2014 10:15:33 AM +08:00'.

	Last Name	First Name	Display Name	Login Name	SIP Handle	Last Login
<input type="checkbox"/>	ADVD	User1	ADVD User1	10070@sglab.com	+10070	
<input type="checkbox"/>	ADVD	User2	ADVD User2	10069@sglab.com	+10069	
<input type="checkbox"/>	AVAYA	SIP3	AVAYA, SIP3	10049@sglab.com	+10049	
<input type="checkbox"/>	AVAYA	SIP4	AVAYA, SIP4	10050@sglab.com	+10050	
<input type="checkbox"/>	Avaya	SIP5	AVAYA, SIP5	10051@sglab.com		
<input type="checkbox"/>	admin	admin	Default Administrator	admin		February 14, 2014 10:15:33 AM +08:00

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/AEi SSP-2x10-S 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. Enter desired **Localized Display Name** and **Endpoint Display Name**. Retain the default values in the remaining fields.

The screenshot shows the 'New User Profile' form in the Avaya Aura System Manager 6.3 interface. The form is titled 'New User Profile' and has tabs for 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Identity' tab is active. The form contains the following fields and values:

- User Provisioning Rule:** (Empty dropdown)
- Identity:**
 - Last Name:** GTek
 - Last Name (Latin Translation):** GTek
 - First Name:** AEI1
 - First Name (Latin Translation):** AEI1
 - Middle Name:** (Empty)
 - Description:** (Empty)
 - Login Name:** 10045@sglab.com
 - Authentication Type:** Basic
 - Password:** (Masked with dots)
 - Confirm Password:** (Masked with dots)
 - Localized Display Name:** GTek, AEI1
 - Endpoint Display Name:** GTek, AEI1
 - Title:** (Empty)

Buttons for 'Commit & Continue', 'Commit', and 'Cancel' are visible at the top right of the form.

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 6.3

Last Logged on at February 12, 2014 3:21 PM
Help | About | Change Password | Log off admin

Home / Users / User Management

New User Profile Commit & Continue Commit Cancel

Identity * **Communication Profile** Membership Contacts

Communication Profile

Communication Profile Password: ●●●●●●
Confirm Password: ●●●●●●

Name

Primary

Select : None

* Name: Primary

Default :

Communication Address

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 Error! Reference source not found.** Click **Add**.

Communication Address

Type	Handle	Domain
No Records found		

Type: Avaya SIP

* Fully Qualified Address: [] @ []

Scroll down and check the **Session Manager Profile** check box to expand. 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 **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter the SIP user extension from **Section Error! Reference source not found.** For **Template**, select “**DEFAULT_9630SIP_CM_6_3**”. For **Port**, select “**IP**”. Retain the default values in the remaining fields.

Session Manager Profile

SIP Registration

* Primary Session Manager: sm1

Primary	Secondary	Maximum
8	0	8

Secondary Session Manager: (None)

Survivability Server: (None)

Max. Simultaneous Devices: 1

Block New Registration When Maximum Registrations Active?

Application Sequences

Origination Sequence: cm6-duplex-app-seq

Termination Sequence: cm6-duplex-app-seq

Call Routing Settings

* Home Location: Location1

Conference Factory Set: (None)

Collaboration Environment Profile

CM Endpoint Profile

* System: CM6-duplex

* Profile Type: Endpoint

Use Existing Endpoints

* Extension: 10045 Endpoint Editor

Template: 9630_DEFAULT_CM_6_3

Set Type: 9630

Security Code:

Port: IP

Voice Mail Number:

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

Repeat **Section 5.2** to add a user for each G-Tek/AEi SSP-2x10 user. In this compliance testing, three Users with extensions “10045”, “10046” and “10047” 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/AEi SSP-2x10 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.

```
display system-parameters customer-options                               Page 1 of 11
                                OPTIONAL FEATURES

G3 Version: V16                                                         Software Package: Enterprise
Location: 2                                                             System ID (SID): 1
Platform: 28                                                            Module ID (MID): 1

                                USED
Platform Maximum Ports: 65000 392
Maximum Stations: 41000 200
Maximum XMOBILE Stations: 41000 0
Maximum Off-PBX Telephones - EC500: 41000 1
Maximum Off-PBX Telephones - OPS: 41000 35
Maximum Off-PBX Telephones - PBFMC: 41000 0
Maximum Off-PBX Telephones - PVFMC: 41000 0
Maximum Off-PBX Telephones - SCCAN: 0 0
Maximum Survivable Processors: 313 2

(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 and G.711A.

```
change ip-codec-set 6                                     Page 1 of 2

                               IP Codec Set

Codec Set: 6

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

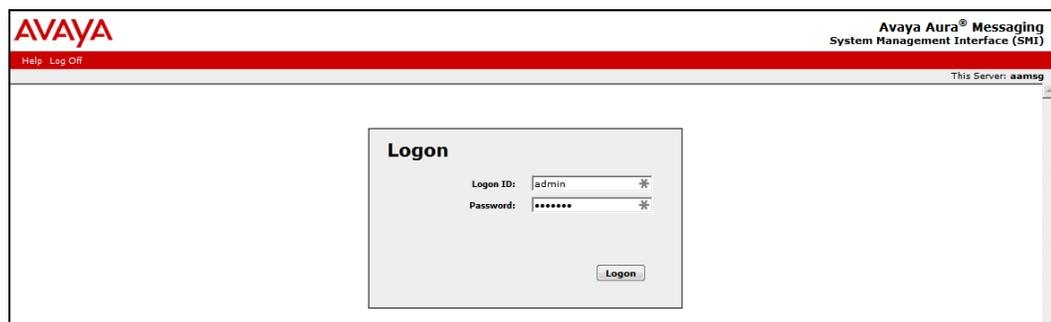
7. Configure Avaya Aura® Messaging

This section provides the procedures for configuring user mailbox in Avaya Aura® Messaging. The detail setup of the messaging server is done during the installation and shall not be detailed here. The procedures include the following areas:

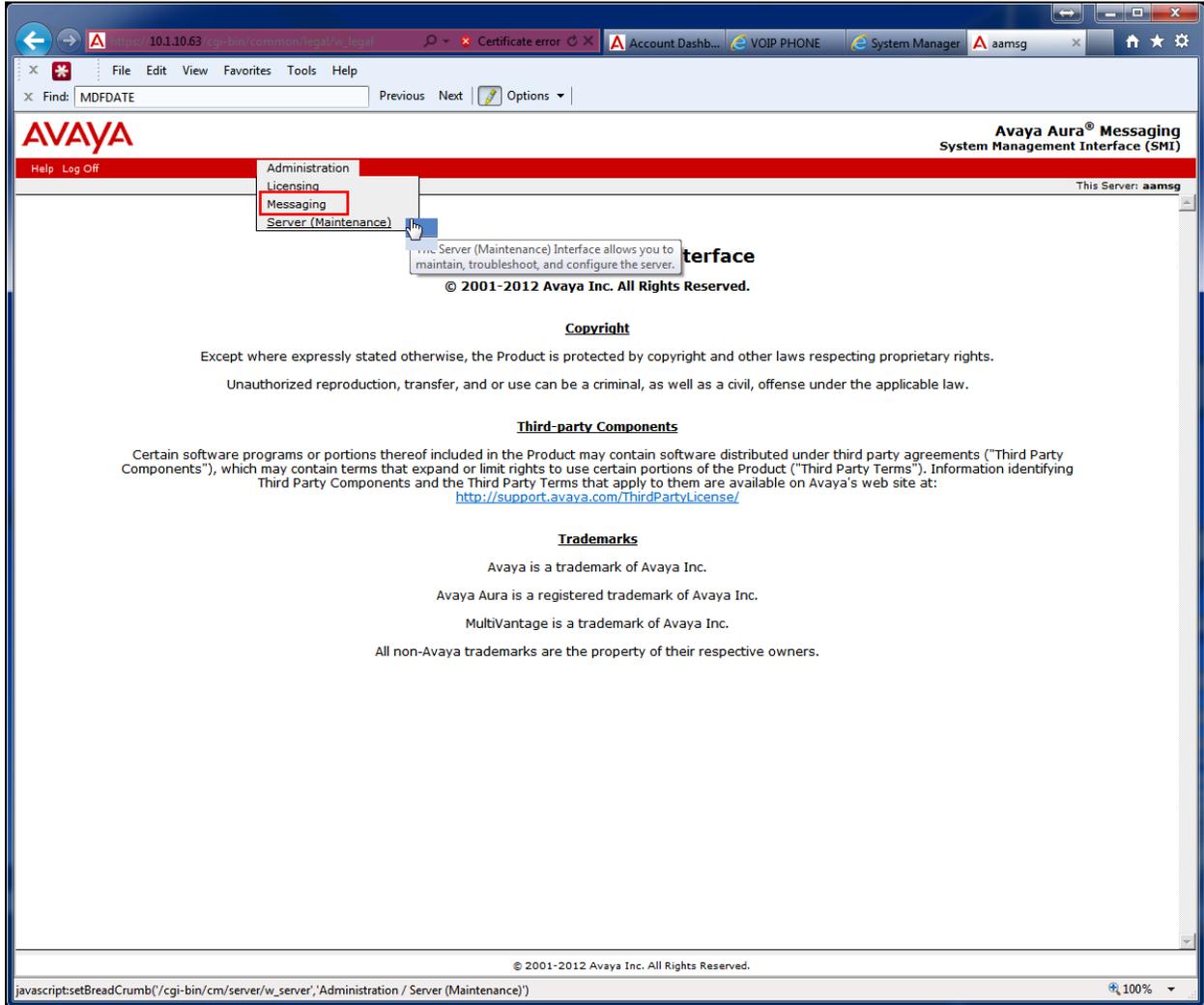
- Launch Messaging System Management Interface (SMI)
- Administer users

7.1. Launch Messaging System Management Interface (SMI)

Access the Messaging Web interface by using the URL “https://ip-address” in an Internet browser window, where “ip-address” is the IP address of the Messaging SMI. Log in using the appropriate credentials. Click **Continue** on the next page (not shown).



Select **Messaging** from the **Administration** drop down menu on top.



7.2. Administer Users

From the left panel, select **Messaging System (Storage)** → **User Management**. Click **Add** from the **Add User/Info Mailbox**.

The screenshot displays the Avaya Aura Messaging System Management Interface (SMI) for the server 'aamsg'. The interface is divided into a left navigation pane and a main content area. The navigation pane is expanded to show 'Users' under the 'Messaging System (Storage)' category. The main content area is titled 'User Management' and includes a 'Help' link. It features three sections: 'License Status' showing 'License mode: Normal'; 'Edit User/Info Mailbox' with a text input field for 'Identifier' and an 'Edit' button; and 'Add User/Info Mailbox' with two 'Add' buttons, one of which is highlighted with a red box. The top of the interface shows the Avaya logo and navigation links for 'Help' and 'Log Off'.

Enter the appropriate names for **First name, Last name, Display name and ASCII name**. Select the appropriate **Site** of the messaging server and in this case Avaya Aura Messaging is selected. Enter 10045 for the **Mailbox number and Extension number**. Ensure the **MWI** is enabled. For **New Password and Confirm Password**, enter the appropriate credentials for user mailbox. Repeat this to creating another 2 mailboxes for User 10046 and 10047.

AVAYA Avaya Aura® Messaging System Management Interface (SMI)

Help Log Off Administration Administration / Messaging This Server: aamsg

User Management > Properties for New User

Help

User Properties

First name: AEI One
Last name: GTEK
Display name: GTEK, AEI1
ASCII name: GTEK, AEI1

Site: Avaya Aura Messaging

Mailbox number: 10045
Extension: 10045

Include in Auto Attendant directory

Additional extensions:

Class of Service: Standard

Pronounceable name: AEI One

MWI enabled: Yes

Miscellaneous 1:
Miscellaneous 2:

New password:
Confirm password:

8. Configure G-Tek/AEi SSP-2x10 SIP Phones

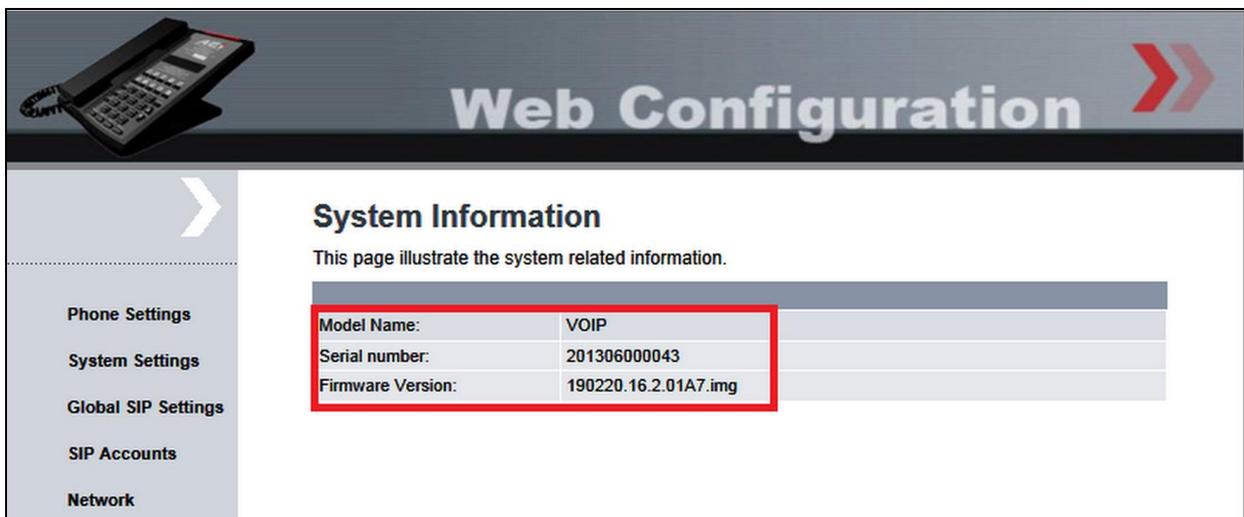
This section provides the procedures for configuring G-Tek/AEi SSP-2x10 SIP Phones on User 10045. The procedures include the following areas:

- Access Web Interface
- Configure LAN Port Settings
- Configure SIP Account Settings
- Configure Codec Settings
- Configure SIP Settings

This is repeated for User 10046 and 10047.

8.1. Access Web Interface

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

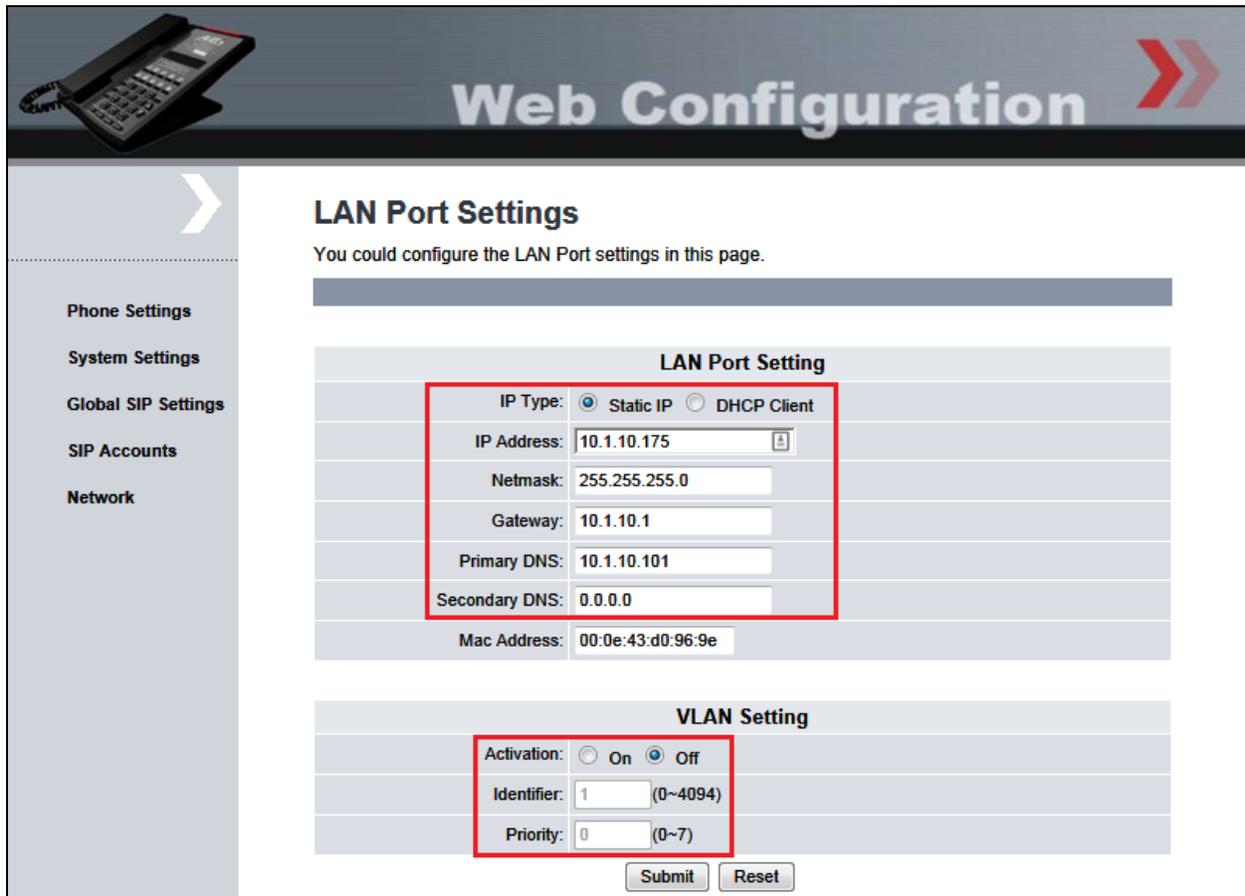


The screenshot displays the 'Web Configuration' interface for a G-Tek/AEi SIP phone. The main heading is 'Web Configuration' with a red double arrow icon. Below this is a navigation menu on the left with options: Phone Settings, System Settings, Global SIP Settings, SIP Accounts, and Network. The main content area is titled 'System Information' and contains the text 'This page illustrate the system related information.' Below this text is a table with the following data:

Model Name:	VOIP
Serial number:	201306000043
Firmware Version:	190220.16.2.01A7.img

8.2. Configure LAN Port Settings

On the left panel, select **Network** → **LAN Port Settings** and configure either as **DHCP Client** (default) or **Static IP** for the LAN connection. As DHCP Client, the LAN Port setting will be automatically populated. In our testing, Static IP is used and the appropriate **IP Address**, **Netmask**, **Gateway**, **Primary/Secondary DNS** and **VLAN Setting** (if any) are manually configured. Click **Submit** to effect the changes.



The screenshot displays the 'Web Configuration' interface. On the left is a navigation menu with 'Network' selected. The main area is titled 'LAN Port Settings' and contains two sections: 'LAN Port Setting' and 'VLAN Setting'. The 'LAN Port Setting' section has a red box around the 'IP Type' (Static IP), 'IP Address' (10.1.10.175), 'Netmask' (255.255.255.0), 'Gateway' (10.1.10.1), 'Primary DNS' (10.1.10.101), and 'Secondary DNS' (0.0.0.0) fields. The 'VLAN Setting' section has a red box around the 'Activation' (Off), 'Identifier' (1), and 'Priority' (0) fields. 'Submit' and 'Reset' buttons are at the bottom.

LAN Port Setting	
IP Type:	<input checked="" type="radio"/> Static IP <input type="radio"/> DHCP Client
IP Address:	10.1.10.175
Netmask:	255.255.255.0
Gateway:	10.1.10.1
Primary DNS:	10.1.10.101
Secondary DNS:	0.0.0.0
Mac Address:	00:0e:43:d0:96:9e

VLAN Setting	
Activation:	<input type="radio"/> On <input checked="" type="radio"/> Off
Identifier:	1 (0~4094)
Priority:	0 (0~7)

Submit Reset

8.3. Configure SIP Account Settings

Select **SIP Accounts** from the left panel. Select the **Registration** radio button 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 mapped to the message speed dial key on the base. Check the default **DTMF Type** settings is **RFC2833**. Click **Submit** to continue.



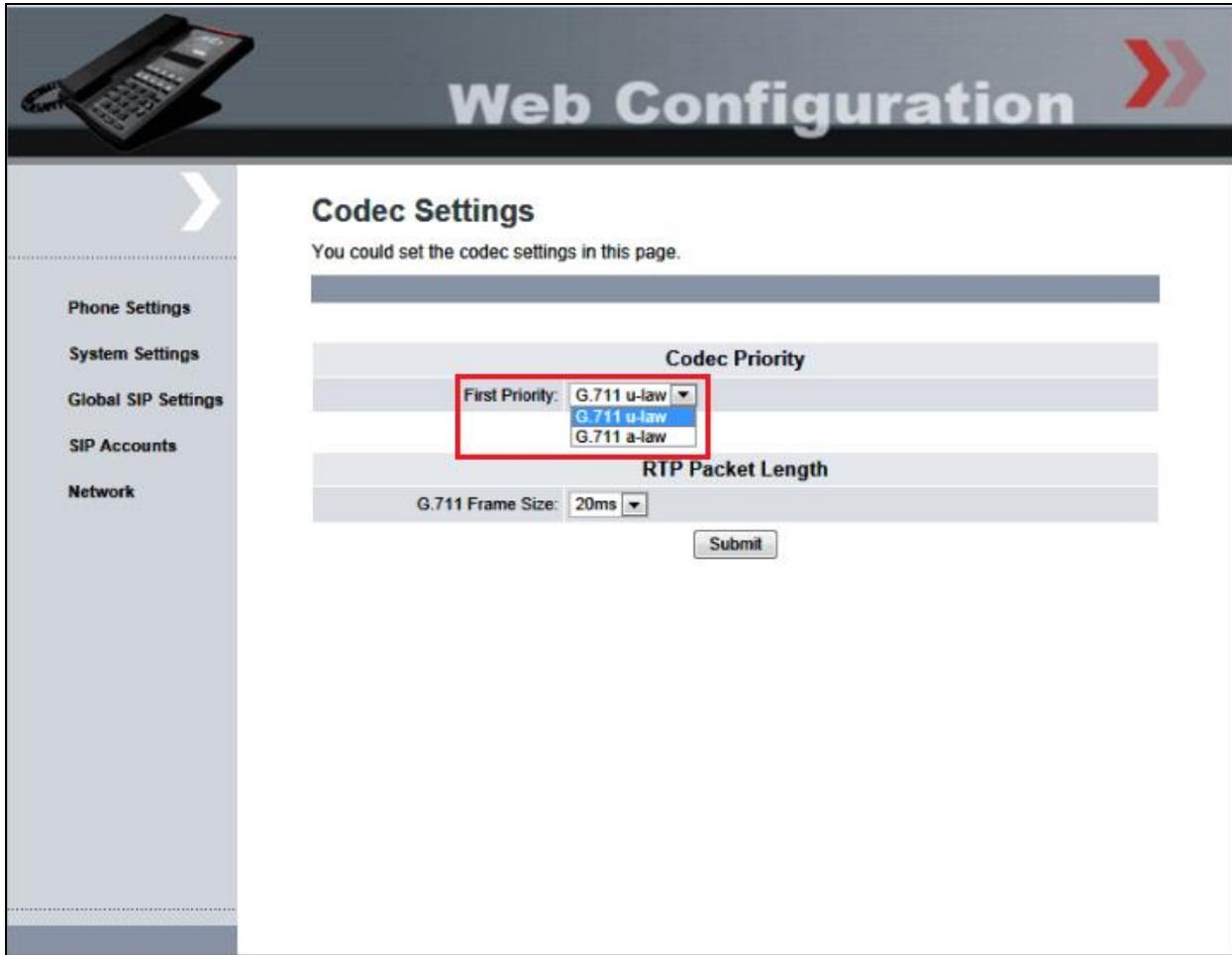
The screenshot shows the 'Web Configuration' interface for 'SIP Account Settings'. The left sidebar contains a navigation menu with the following items: Phone Settings, System Settings, Global SIP Settings, SIP Accounts, and Network. The main content area is titled 'SIP Account Settings' and includes a sub-header 'You could set information of service domians in this'. Below this is a form for 'SIP Account 1' with the following fields and values:

SIP Account 1	
Registration:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Registration ID:	10045
Display Name:	10045
Authentication Name:	10045
Password:	••••••
Registration Server:	10.1.10.60
Proxy Server:	10.1.10.60
Realm Address:	
Voice Mail:	10000
Expire Time:	60
DTMF Type:	RFC2833
MWI:	Enable
Status:	registered

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

8.4. Configure Codec Settings

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



The screenshot shows a web configuration page titled "Web Configuration" with a telephone icon on the left. The main content area is titled "Codec Settings" and includes the instruction "You could set the codec settings in this page." Below this, there are three sections: "Codec Priority", "RTP Packet Length", and "G.711 Frame Size". The "Codec Priority" section has a dropdown menu for "First Priority" with a red box around it, showing a list of options: "G.711 u-law", "G.711 u-law", and "G.711 a-law". The "RTP Packet Length" section has a dropdown menu for "G.711 Frame Size" set to "20ms". A "Submit" button is located at the bottom right of the form.

Web Configuration

Codec Settings

You could set the codec settings in this page.

Codec Priority

First Priority:

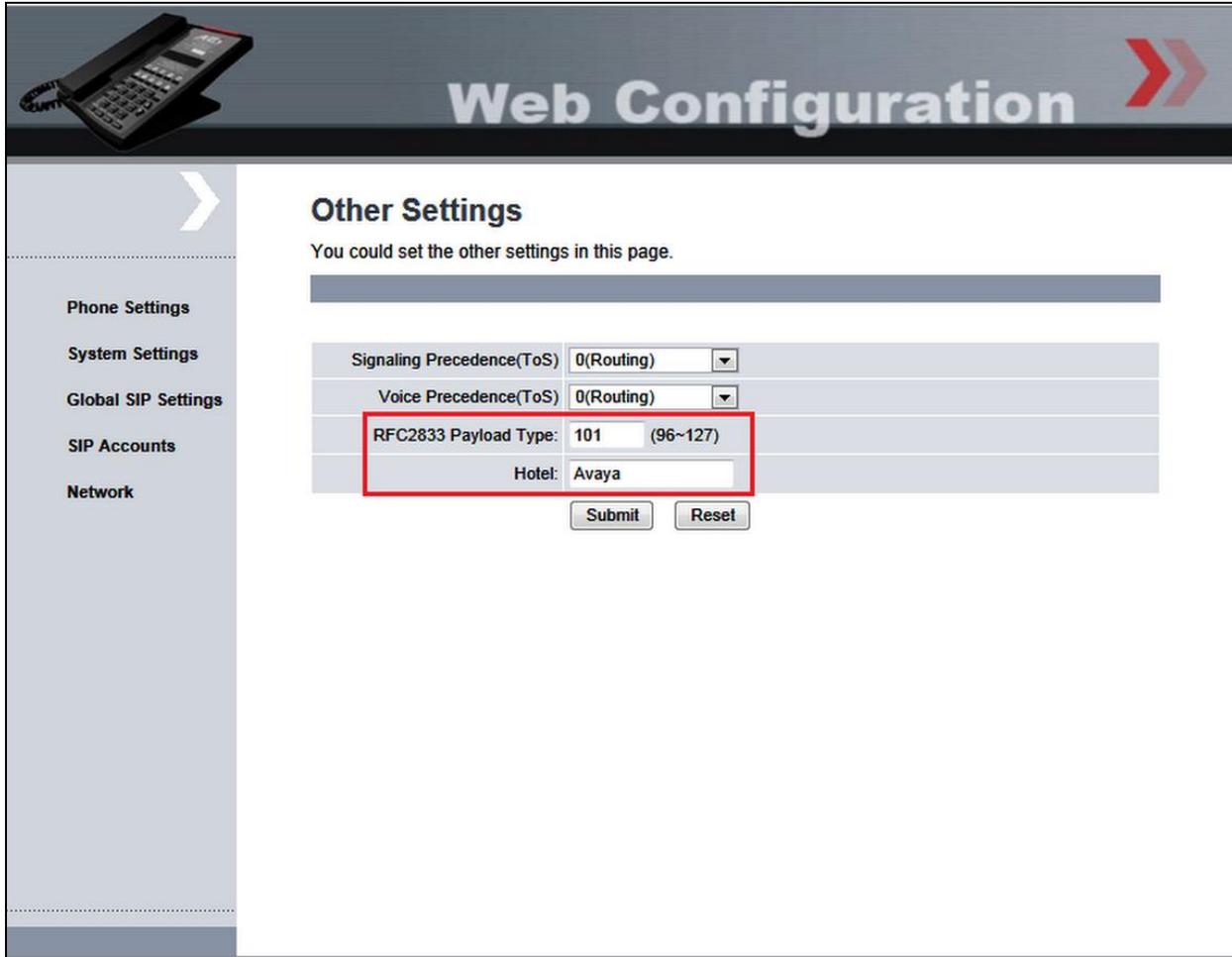
- G.711 u-law
- G.711 u-law
- G.711 a-law

RTP Packet Length

G.711 Frame Size:

8.5. Configure SIP Settings

Select **Global SIP Settings** → **Other Settings** from the left menu. Check that **RFC2833 Payload Type** is set to 101. Enter the appropriate name for the **Hotel**. Click **Submit** to continue.



The screenshot shows the 'Web Configuration' interface. The main heading is 'Web Configuration' with a red double arrow icon. The left sidebar contains a navigation menu with the following items: Phone Settings, System Settings, Global SIP Settings, SIP Accounts, and Network. The 'Global SIP Settings' item is highlighted with a white arrow. The main content area is titled 'Other Settings' and includes the text 'You could set the other settings in this page.' Below this is a table of settings:

Signaling Precedence(ToS)	0(Routing)
Voice Precedence(ToS)	0(Routing)
RFC2833 Payload Type:	101 (96~127)
Hotel:	Avaya

At the bottom of the settings table are two buttons: 'Submit' and 'Reset'. The 'RFC2833 Payload Type' and 'Hotel' rows are highlighted with a red border.

9. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Session Manager and G-Tek/AEi SSP-2x10 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 6.3

Last Logged on at February 14, 2014 10:57 AM
Help | About | Change Password | Log off admin

Home Session Manager x

Session Manager
Dashboard
Session Manager
Administration
Communication Profile Editor
Network Configuration
Device and Location Configuration
Application Configuration
System Status
SIP Entity Monitoring
Managed Bandwidth Usage
Security Module Status
SIP Firewall Status
Registration
Summary
User Registrations

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

User Registrations

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

View: Default Force Unregister AST Device Notifications: Reboot Reload Fallback As of 11:32 AM

9 Items Show ALL Filter: Enable

Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered		
										Prim	Sec	Surv
Show	---	SIP5	Avaya	---	---			0/1				
Show	10045@sglab.com	AEI1	GTek	Location1	10.1.10.175:5060			1/1		<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Show	10046@sglab.com	AEI2	GTek	Location1	10.1.10.176:5060			1/1		<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Show	10047@sglab.com	AEI3	GTek	Location1	10.1.10.177:5060			1/3		<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Show	10049@sglab.com	SIP3	AVAYA	Location1	10.1.10.168:5060			1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Show	10050@sglab.com	SIP4	AVAYA	Location1	10.1.10.156:5060			1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Show	10069@sglab.com	User2	ADVD	Location1	10.1.10.166:5061			1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
Show	10070@sglab.com	User1	ADVD	Location1	10.1.10.164:5060			2/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Show	10070@sglab.com	User1	ADVD	Location1	10.1.10.164:5061			2/1		<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>

Select : All, None

From the web interface of the G-Tek/AEi SSP-2x10 phone, click **SIP Accounts** from the left menu. Verify that the **Status** field shows as **registered**.

SIP Accounts					
ID	Display Name	Registration Server	Status	Registration	Select
1	10045	10.1.10.60	registered	Enable	<input type="checkbox"/>
2					<input type="checkbox"/>

10. Conclusion

These Application Notes describe the configuration steps required for G-Tek/AEi Electronics SSP-2x10-S SIP Phones to successfully interoperate with Avaya Aura® Session Manager 6.3 SP5 and Avaya Aura® Communication Manager 6.3 SP3. All feature and serviceability test cases were completed successfully with observations noted in **Section 2.2**.

11. 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 6.3, Doc ID 03-300509, October 2013.

[2] *Administering Avaya Aura® Session Manager*, Release 6.3, Issue 3, October 2013.

[3] *Administering Avaya Aura® Messaging*, Release 6.2, Issue 2.2, December 2013.

The following documents can be obtained from GTek/AEi.

[4] *SSP-2210-S SSP-2110-S datasheet*

[5] *SSP-2210-S SSP-2110-S QIG Version 1.00*

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