



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

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# **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.

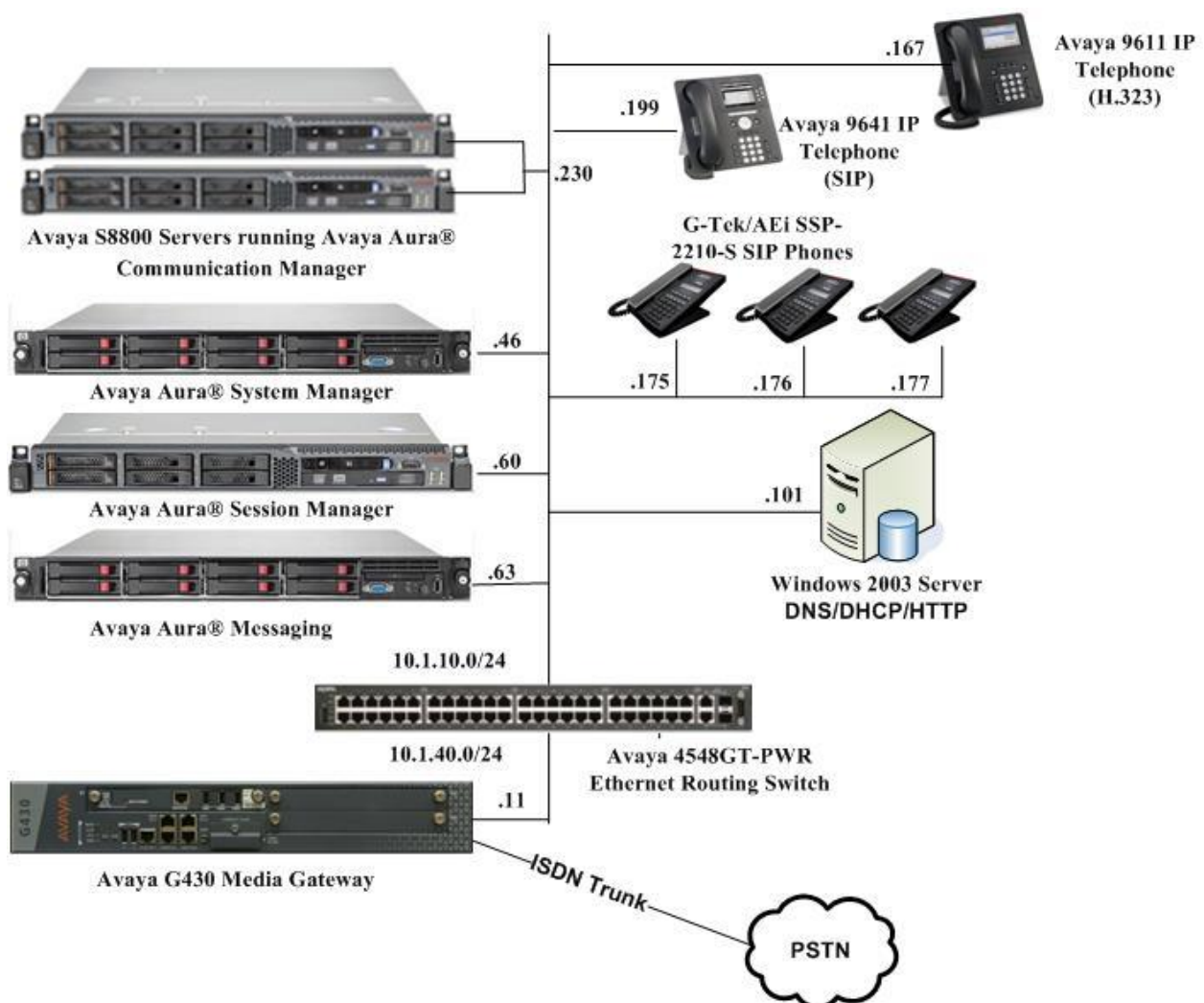
## 2.3. Support

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

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

<b>Extension</b>	<b>Note</b>
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:

<b>Equipment/Software</b>	<b>Release/Version</b>
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:

**Log On** **Cancel** [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 User Management interface. The top navigation bar includes 'Home' and 'User Management'. The left sidebar lists '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 'Users' section with a table of users. The 'New' button is highlighted with a red box.

**Users**

View Edit **New** Duplicate Delete More Actions Advanced Search

9 Items Show ALL Filter: Enable

	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 displays the Avaya Aura System Manager 6.3 interface. The top header shows the Avaya logo and 'Aura System Manager 6.3'. The right header indicates the user is logged in as 'admin' on February 12, 2014, at 3:20 PM, with links for 'Help', 'About', 'Change Password', and 'Log off admin'. The left sidebar contains a 'User Management' menu with options: 'Manage Users' (selected), 'Public Contacts', 'Shared Addresses', 'System Presence ACLs', 'Communication Profile', and 'Password Policy'. The main content area is titled 'New User Profile' and includes a breadcrumb trail: 'Home / Users / User Management / Manage Users'. There are three buttons at the top right: 'Commit & Continue', 'Commit', and 'Cancel'. Below the title bar are four tabs: 'Identity' (selected), 'Communication Profile', 'Membership', and 'Contacts'. The 'Identity' tab contains a 'User Provisioning Rule' dropdown menu. Below this, the 'Identity' section includes the following fields:

- \* 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)

### 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 User Management x

User Management  
Manage Users  
Public Contacts  
Shared Addresses  
System Presence ACLs  
Communication Profile  
Password Policy

Home / Users / User Management

New User Profile

Commit & Continue 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		

\* Fully Qualified Address: [ ] @ [ ]

Add Cancel

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

New Edit Delete

Type	Handle	Domain
No Records found		

Type: Avaya SIP

\* Fully Qualified Address: [ ] @ [ ]

Add Cancel



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.

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 Codec	Silence Suppression	Frames Per Pkt	Packet 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).

**AVAYA**

Avaya Aura® Messaging  
System Management Interface (SMI)

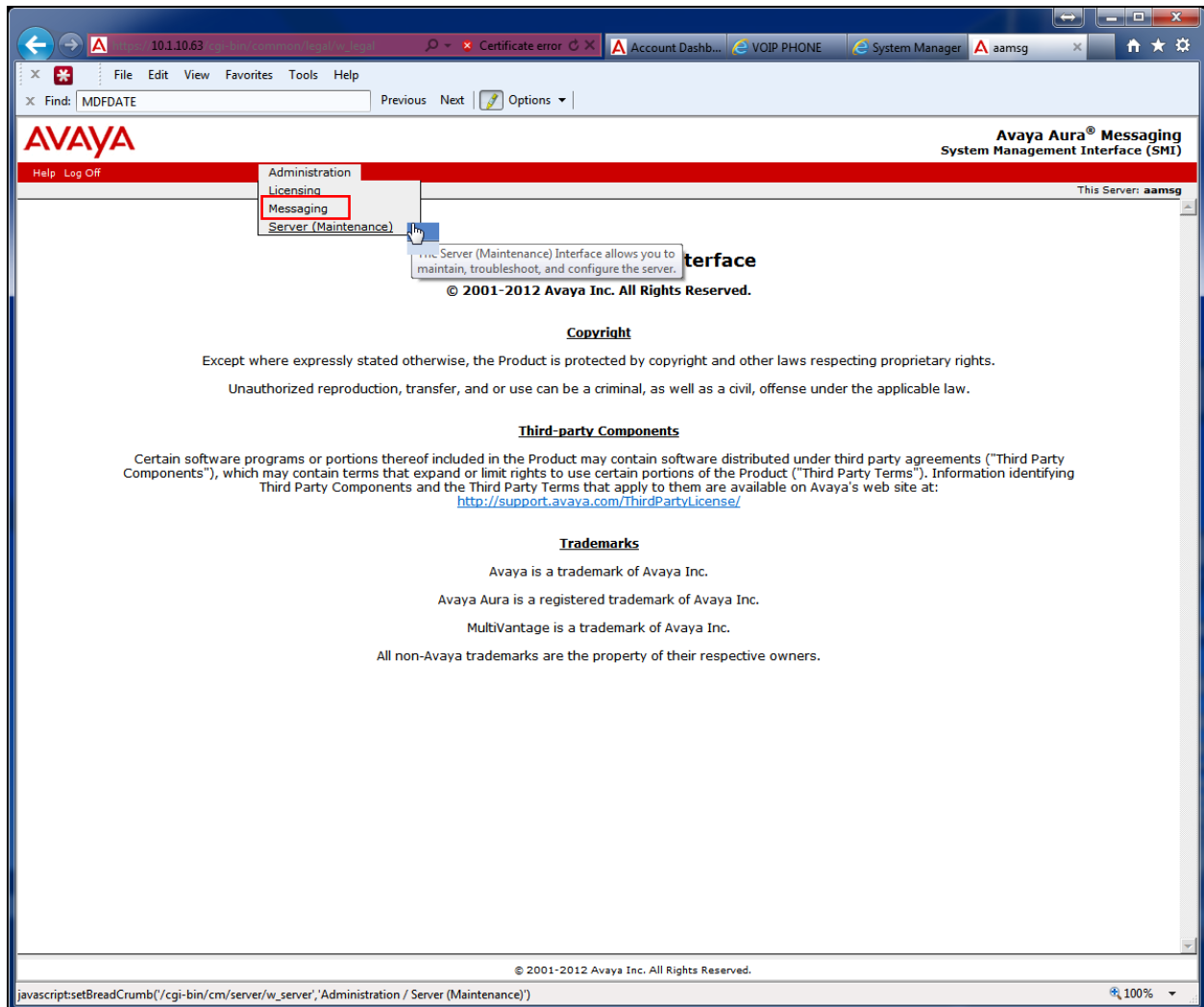
Help Log Off This Server: aams9

**Logon**

Logon ID:

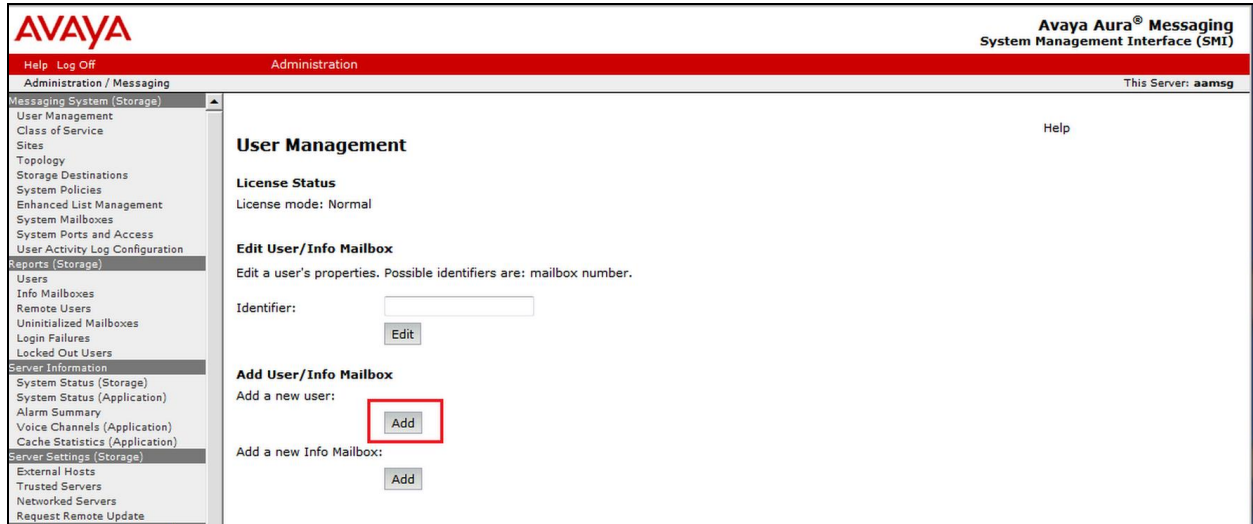
Password:

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**.



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:

Navigation Menu:

- Messaging System (Storage)
  - User Management
  - Class of Service
  - Sites
  - Topology
  - Storage Destinations
  - System Policies
  - Enhanced List Management
  - System Mailboxes
  - System Ports and Access
  - User Activity Log Configuration
- Reports (Storage)
- Users
  - Info Mailboxes
  - Remote Users
  - Uninitialized Mailboxes
  - Login Failures
  - Locked Out Users
- Server Information
  - System Status (Storage)
  - System Status (Application)
  - Alarm Summary
  - Voice Channels (Application)
  - Cache Statistics (Application)
- Server Settings (Storage)
  - External Hosts
  - Trusted Servers
  - Networked Servers
  - Request Remote Update
- IMAP/SMTP Settings (Storage)
  - General Options
  - Mail Options
  - IMAP/SMTP Status
- Telephony Settings (Application)
  - Telephony Integration
- Server Settings (Application)
  - Dial Rules
  - Cluster
  - System Parameters
  - Languages
  - Log Configuration
- Advanced (Application)
  - System Operations
  - Timeouts
  - AxC Address
  - Miscellaneous
  - Core Files
- Utilities
- Messaging DB Audits (Storage)

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

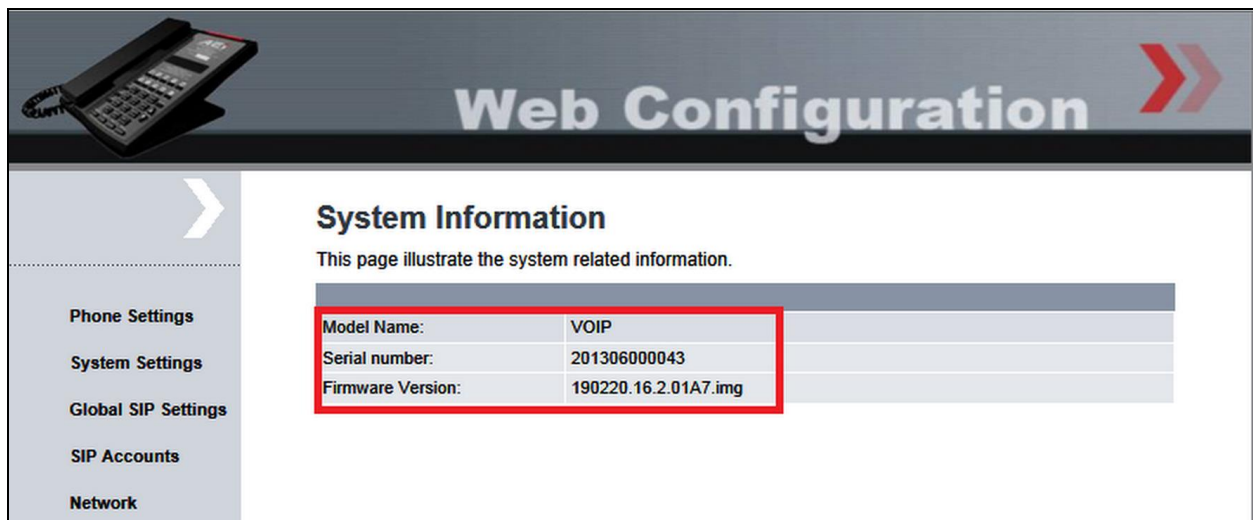
This section provides the procedures for configuring G-Tek/AEi SSP-2x10 SIP Phones on User 10045. The procedures include the following areas:

- 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.

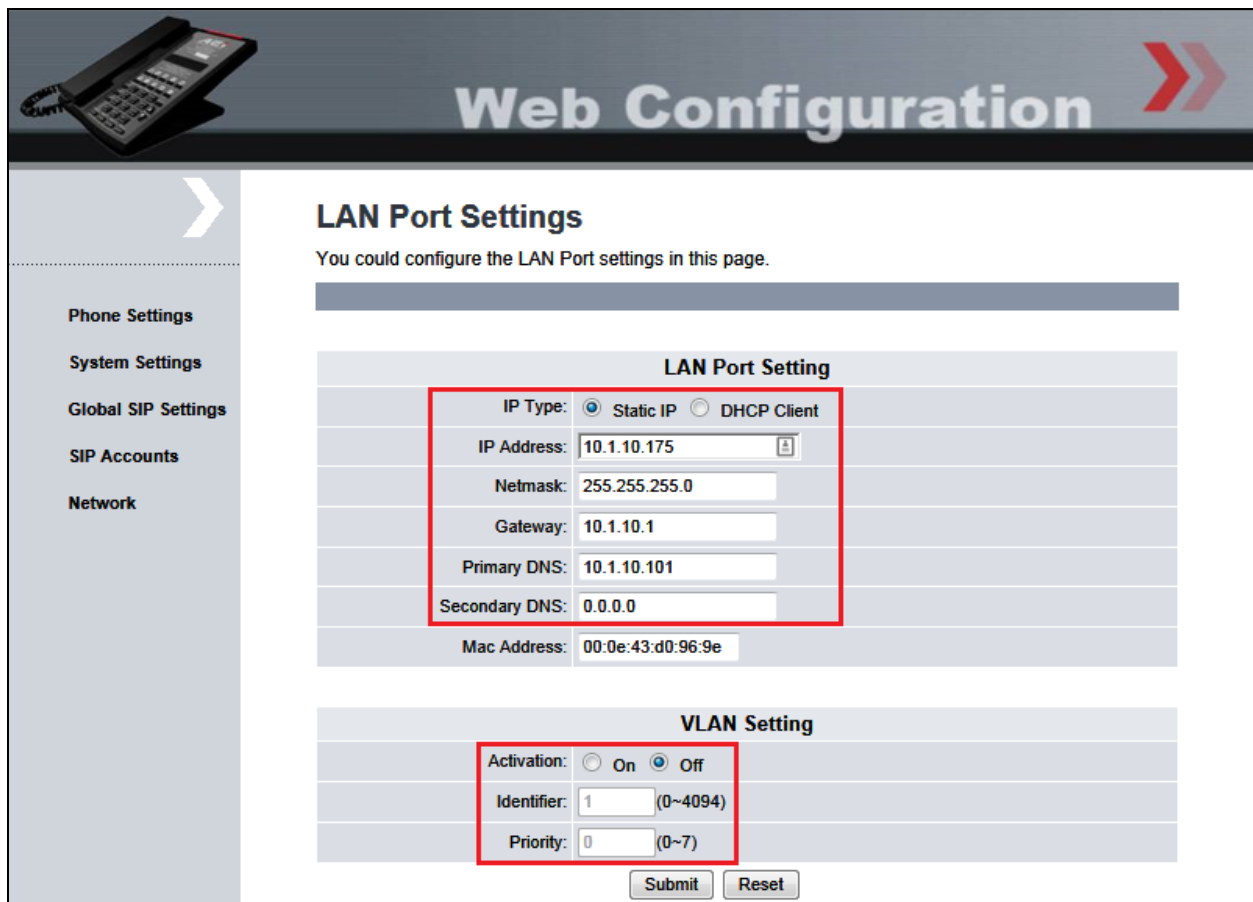


The screenshot displays the 'Web Configuration' interface for a G-Tek/AEi SIP phone. The header features a phone icon and the title 'Web Configuration' with a red double-arrow icon. A left sidebar contains navigation links: 'Phone Settings', 'System Settings', 'Global SIP Settings', 'SIP Accounts', and 'Network'. The main content area is titled 'System Information' and includes the text 'This page illustrate the system related information.' Below this is a table with system details. The table has three rows: 'Model Name' with value 'VOIP', 'Serial number' with value '201306000043', and 'Firmware Version' with value '190220.16.2.01A7.img'. The first two rows of the table are highlighted with a red border.

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 image shows a web configuration interface for a device. The top header features a telephone icon and the text "Web Configuration" with a red double arrow icon. A left sidebar contains a menu with "Phone Settings", "System Settings", "Global SIP Settings", "SIP Accounts", and "Network" (which is highlighted with a white arrow). The main content area is titled "LAN Port Settings" and includes the instruction "You could configure the LAN Port settings in this page." Below this, there are two sections: "LAN Port Setting" and "VLAN Setting".

**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)

At the bottom of the form are "Submit" and "Reset" buttons.



### 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.



**Web Configuration**

**SIP Account Settings**  
You could set information of service domains in this

**SIP Account 1**

Registration: ☒ Enable ☐ 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

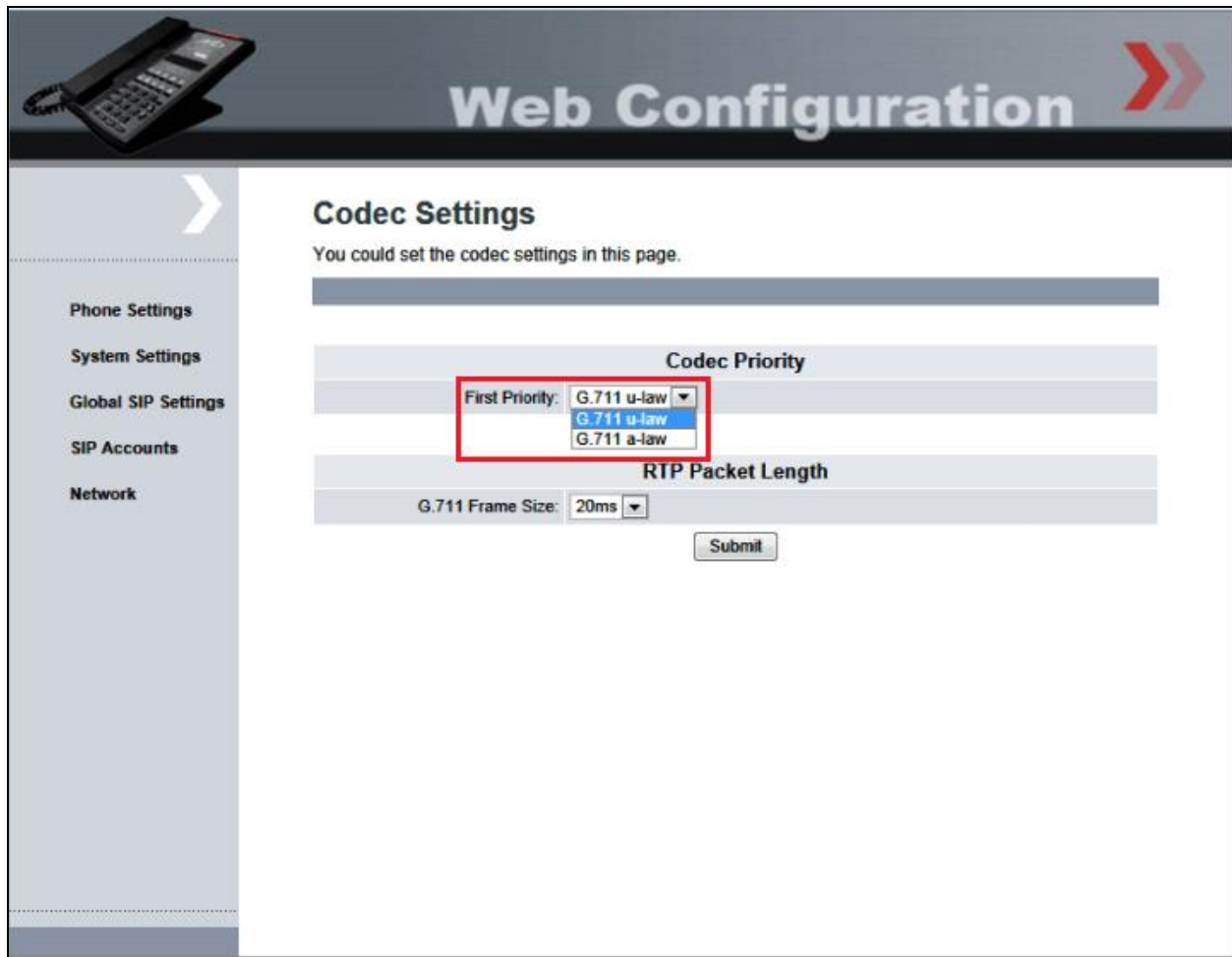
MWI: Enable

Status: registered

Submit 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 image shows a web configuration interface for a phone system. At the top, there is a header with a phone icon and the text "Web Configuration". Below the header, there is a left sidebar with a list of settings: "Phone Settings", "System Settings", "Global SIP Settings", "SIP Accounts", and "Network". The "Global SIP Settings" option is selected, and a right arrow indicates the next step is "Codec Settings". The main content area is titled "Codec Settings" and contains the text "You could set the codec settings in this page." Below this, there is a section for "Codec Priority" with a dropdown menu for "First Priority". The dropdown menu is open, showing three options: "G.711 u-law", "G.711 u-law", and "G.711 a-law". The "G.711 u-law" option is highlighted. Below the "Codec Priority" section, there is a section for "RTP Packet Length" with 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**

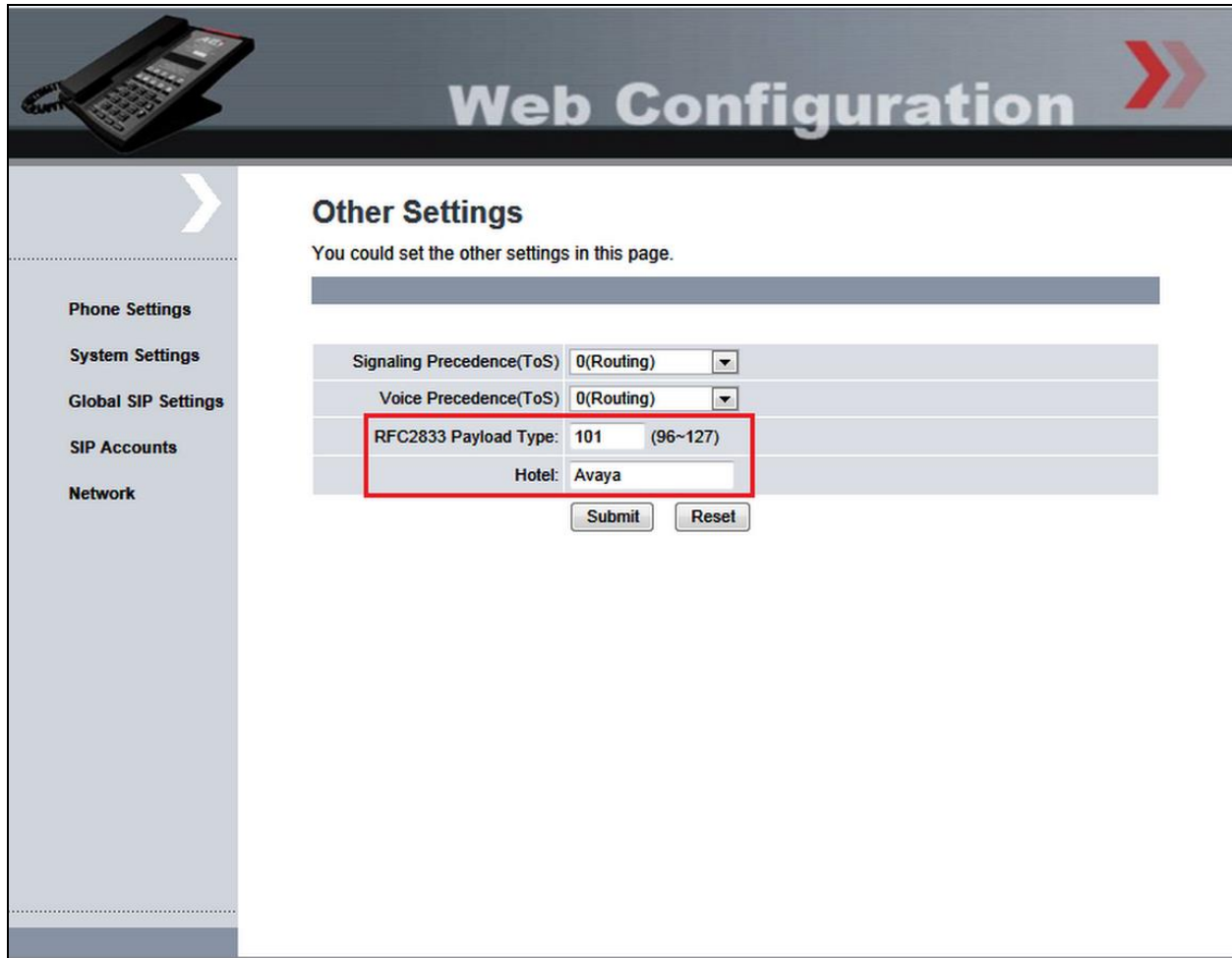
First Priority:

**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 image shows a web configuration interface for a phone system. At the top, there is a header with a phone icon on the left and the text "Web Configuration" in the center, followed by a red double arrow icon on the right. Below the header is a left sidebar with a list of settings categories: "Phone Settings", "System Settings", "Global SIP Settings", "SIP Accounts", and "Network". The "Global SIP Settings" category is selected, and a right arrow indicates the current page. The main content area is titled "Other Settings" and contains the text "You could set the other settings in this page." Below this text is a table of settings. The table has two columns: the setting name and the value. The settings are: "Signaling Precedence(ToS)" with a value of "0(Routing)", "Voice Precedence(ToS)" with a value of "0(Routing)", "RFC2833 Payload Type:" with a value of "101" and a range "(96~127)", and "Hotel:" with a value of "Avaya". The "RFC2833 Payload Type:" and "Hotel:" rows are highlighted with a red border. At the bottom of the table are two buttons: "Submit" and "Reset".

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

Submit Reset

## 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

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.

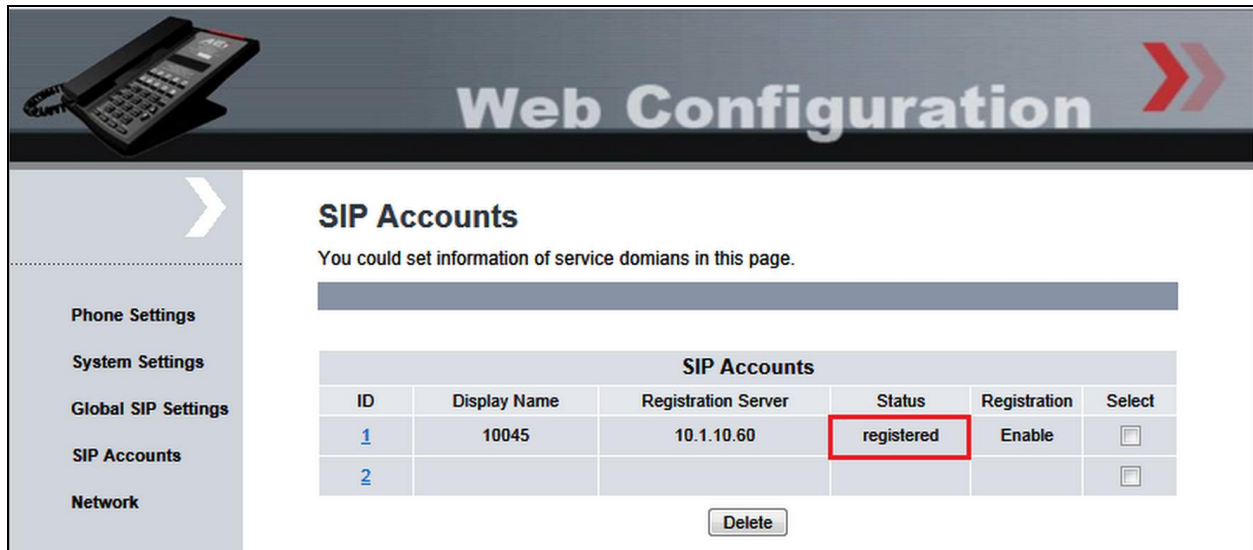
View Default Force Unregister AST Device Notifications: Reboot Reload Fallback As of 11:32 AM Customize Advanced Search

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
<input type="checkbox"/>	Show	---	SIP5	Avaya	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	10045@sglab.com	AEI1	GTEk	Location1	10.1.10.175:5060	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	(AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	10046@sglab.com	AEI2	GTEk	Location1	10.1.10.176:5060	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	(AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	10047@sglab.com	AEI3	GTEk	Location1	10.1.10.177:5060	<input type="checkbox"/>	<input type="checkbox"/>	1/3	<input type="checkbox"/>	<input checked="" type="checkbox"/>	(AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	10049@sglab.com	SIP3	AVAYA	Location1	10.1.10.168:5060	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	(AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	10050@sglab.com	SIP4	AVAYA	Location1	10.1.10.156:5060	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	(AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	10069@sglab.com	User2	ADVD	Location1	10.1.10.166:5061	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	(AC)	<input checked="" type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	10070@sglab.com	User1	ADVD	Location1	10.1.10.164:5060	<input type="checkbox"/>	<input type="checkbox"/>	2/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	(AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	10070@sglab.com	User1	ADVD	Location1	10.1.10.164:5061	<input type="checkbox"/>	<input type="checkbox"/>	2/1	<input type="checkbox"/>	<input type="checkbox"/>	(AC)	<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**.



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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