



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring G-Tek/AEi Communications SSP-2210-SG 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 Communications SSP-2210-SG 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 Communications SSP-2210-SG SIP Phone to interoperate within an Avaya SIP infrastructure consisting of Avaya Aura® Session Manager 6.3 and Avaya Aura® Communication Manager 6.3. G-Tek/AEi SSP-2210-SG is an IP phones for the hospitality industry.

2. General Test Approach and Test Results

To verify interoperability of G-Tek/AEi SSP-2210-SG phone with Session Manager and Communication Manager, calls were made between G-Tek/AEi telephones and Avaya SIP, H.323, analog and digital telephones using various codec settings and exercising common PBX features. Various telephony features were also 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/AEi SSP-2210-SG SIP Phones with Session Manager.
- Calls between G-Tek/AEi telephones and Avaya SIP, H.323, analog, and digital telephones. Also included calls to and from PSTN.
- G.711Mu, G.711A and G729 codec support and negotiation.
- 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/Resume, Transfer, and Conference.
- 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.

2.3. Support

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

- Phone: +1-650-552-9516
- E-mail: techsupport@aeicomcommunications.com

3. Reference Configuration

The diagram illustrates an enterprise site with an Avaya SIP-based network, including Session Manager, an S8300 Board installed in an Avaya G450 Media Gateway running Communication Manager, and Avaya SIP, H.323, analog, and digital endpoints. Two G-Tek/AEi SSP-2210-SG SIP Phones, registered with Session Manager, were also used during the compliance test.

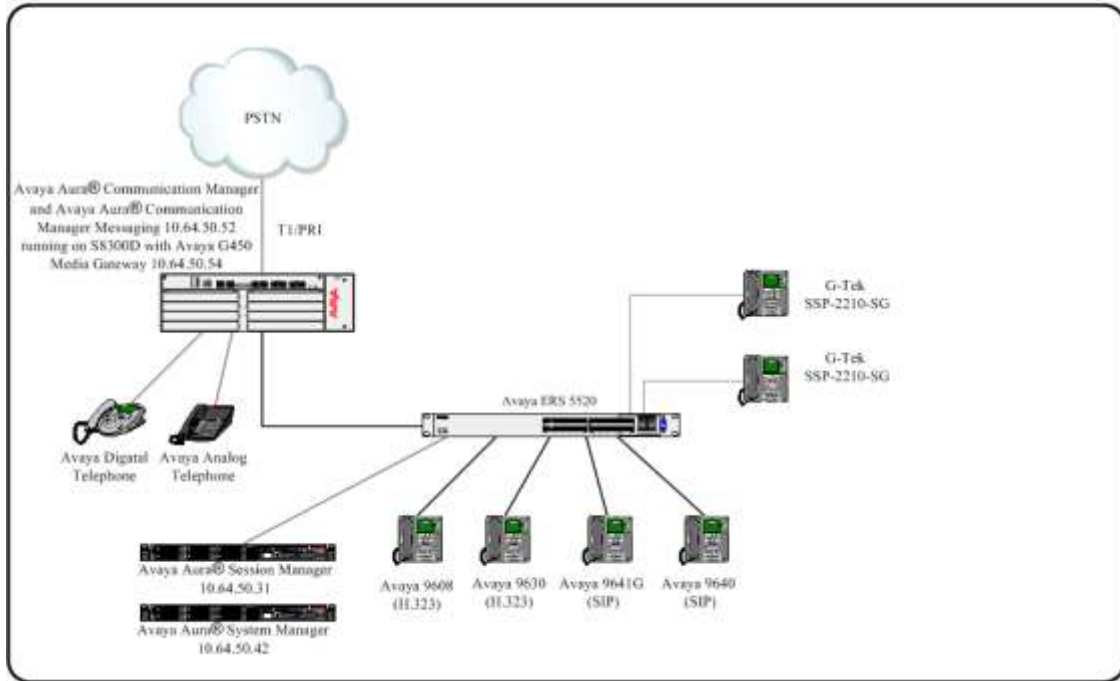


Figure 1: G-Tek/AEi SSP-2210-SG SIP Phone Reference Configuration

4. Equipment and Software Validated

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

Equipment	Software/Firmware
<i>Avaya PBX Products</i>	
Avaya Aura® Communication Manager	6.3 SP6
Avaya Aura® Session Manager	6.3.8
<i>Avaya Messaging (Voice Mail) Products</i>	
Avaya Aura® Communication Manager Messaging	6.3 SP6
<i>Avaya Endpoints</i>	
Avaya 96xx Series IP Deskphone	(H.323 3.2) (SIP 2.6)
Avaya 96x1 Series IP Deskphone	(H.323 6.4) (SIP 6.4)
Avaya Digital Telephone	R39
Avaya Analog Telephone	NA
<i>G-Tek/AEi Products</i>	
SSP-2210-SG	190220.16.2.01D10

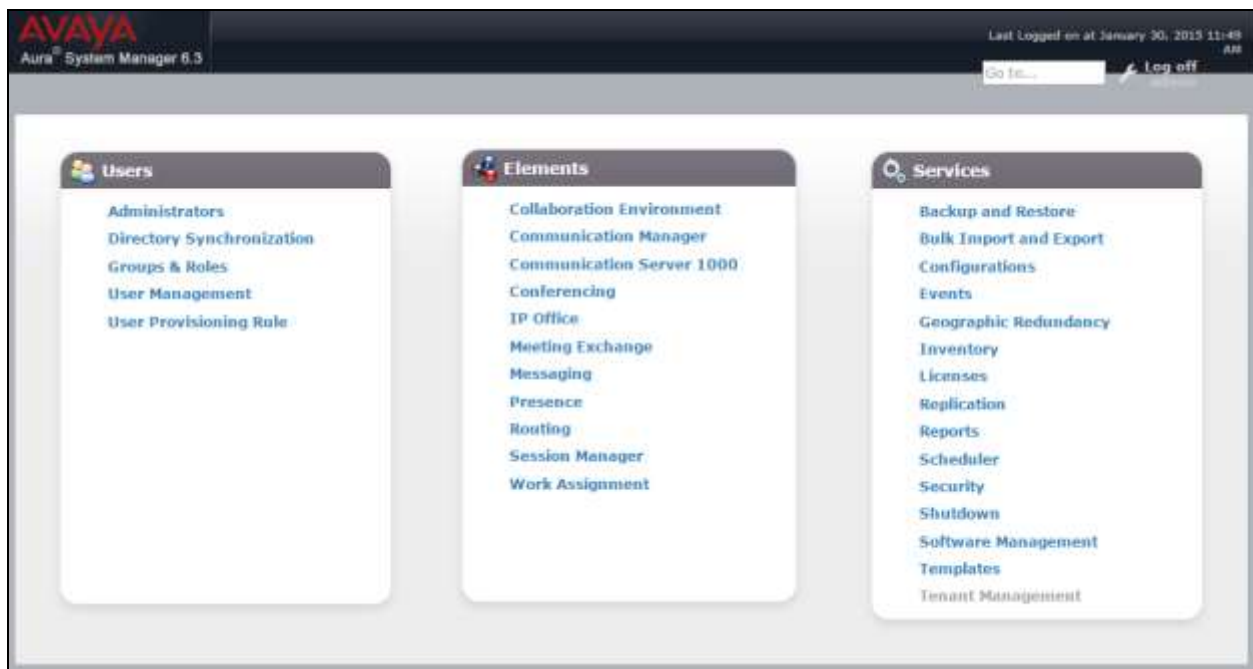
5. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring 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 and the following screen should be displayed.



5.2. Administer Users

From the main screen select **Users** → **User Management** → **Manage Users** to display the **User Management** screen below. Click **New** to add a user.

The screenshot displays the Avaya System Manager 6.3 User Management interface. The page title is "User Management" and the breadcrumb trail is "Home / Users / User Management / Manage Users". A search bar is visible at the top. The main content area shows a table of users with the following columns: Last Name, First Name, Display Name, Login Name, SIP Handle, and Last Login. The table contains 15 rows of user data, including SIP, Station, Analog, AudioCodes, and Avaya/Avtec agents and lines.

Last Name	First Name	Display Name	Login Name	SIP Handle	Last Login
36101	SIP	36101, Stehen	36101@4427.com	36101	
36102	Station	36102, Stehen	36102@4427.com	36102	
36103	Station	36103, Stehen	36103@4427.com	36103	
36104	Station	36104, Stehen	36104@4427.com	36104	
36105	Station	36105, Stehen	36105@4427.com	36105	
36106	Station	36106, Stehen	36106@4427.com	36106	
36107	Station	36107, Stehen	36107@4427.com	36107	
36200	Analog	36200, Analog	36200@4427.com	36200	
36201	Analog	36201, Analog	36201@4427.com	36201	
AudioCodes	Analog Line 1	AudioCodes, Analog Line 1	36301@4427.com	36301	
Avaya	ara-X Agent 1	Avaya, ara-X Agent 1	40001@4427.com	+9009445001	
Avaya	ara-X Agent 2	Avaya, ara-X Agent 2	40002@4427.com	+9009445002	
61000	Avtec	Avtec Line1	61000@4427.com	61000	
61001	Avtec	Avtec Line2	61001@4427.com	61001	
61002	Avtec	Avtec Line3	61002@4427.com	61002	

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-2210-SG user extension and “z” is the domain name used for compliance testing, in this case “**d4f27.com**”. Retain the default values in the remaining fields.

The screenshot shows the Avaya Aura System Manager 6.3 interface. The main window is titled "New User Profile" and has tabs for "Identity", "Communication Profile", "Membership", and "Contacts". The "Identity" tab is selected. Below the tabs, there is a "User Provisioning Rule" dropdown menu. The "Identity" section contains the following fields:

- Last Name: SKD1203
- Last Name (Latin Translation): SKD1203
- First Name: G-Tek
- First Name (Latin Translation): G-Tek
- Middle Name: (empty)
- Description: (empty)
- Login Name: 81020@d4f27.com
- Authentication Type: Basic

Buttons for "Commit & Continue", "Commit", and "Cancel" are visible in the top right corner 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.

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**.

Scroll down to check and expand **Session Manager Profile**. For **Primary Session Manager**, **Origination Sequence**, **Termination 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 5.2.1**. For **Template**, select “**9608SIP_DEFAULT_CM_6_3**”. Retain the default values in the remaining fields.

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

Repeat **Section 5.2** to add a user for each G-Tek/AEi SSP-2210-SG.

Avaya System Manager 6.2 | Last Logged on 01 January 2015 11:00:11 | Log off

Home / User Management / Manage Users

New User Profile

Identity | **Communication Profile** | Membership | Contacts

Communication Profile

Communication Profile Password: [password field]
 Confirm Password: [password field]

Communication Address

Type	Handle	Domain
No Records Found		

Type: Anaya SIP
 Fully Qualified Address: 61020 @ .44127.com

Session Manager Profile

SIP Registration

Primary Session Manager: cm5031
 Secondary Session Manager: (None)
 Survivality Server: (None)
 Max. Simultaneous Devices: 1
 Block New Registration When Maximum Registrations Active? [checkbox]

Application Sequences

Origination Sequence: cm5052
 Termination Sequence: cm5052

Call Routing Settings

Home Location: 44127_1
 Conference Factory Set: (None)

Call History Settings

Enable Centralized Call History? [checkbox]

CM Endpoint Profile

System: cm5052
 Profile Type: Endpoint
 Use Existing Endpoints? [checkbox]
 Extension: 61020
 Template: WDSSEP_DEFAULT_CN_6_3
 Set Type: WDSSEP
 Security Code: [field]
 Port: 11
 Voice Mail Number: [field]
 Redirected Handle: (None)
 Enhanced Call-Info display for 1-line phones [checkbox]
 Delete Endpoint on Unassign of Endpoint from User or on Delete User [checkbox]
 Override Endpoint Name and Localized Name [checkbox]

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-2210-SG 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: 6400 126
Maximum Stations: 2400 60
Maximum XMOBILE Stations: 2400 0
Maximum Off-PBX Telephones - EC500: 9600 0
Maximum Off-PBX Telephones - OPS: 9600 29
Maximum Off-PBX Telephones - PBFMC: 9600 0
Maximum Off-PBX Telephones - PVFMC: 9600 0
Maximum Off-PBX Telephones - SCCAN: 0 0
Maximum Survivable Processors: 313 0

(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, and G.729.

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

                               IP CODEC SET

Codec Set: 1

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

Media Encryption
1: none
2:
3:
```

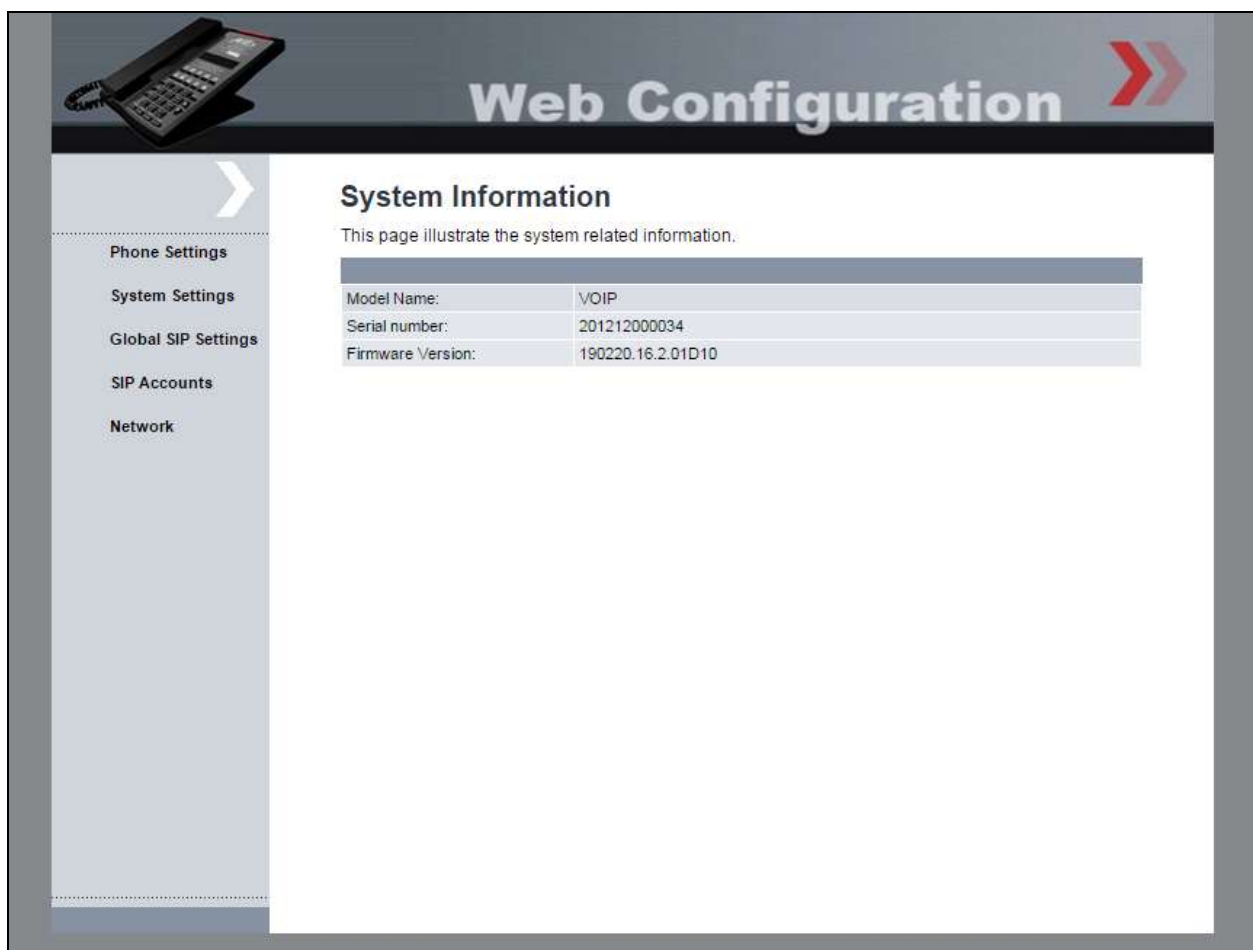
7. Configure G-Tek/AEi SSP-2210-SG SIP Phones

This section provides the procedures for configuring G-Tek/AEi SSP-2210-SG SIP Phones. The procedures include the following areas:

- Access Web Interface
- Configure SIP Account

7.1. 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.

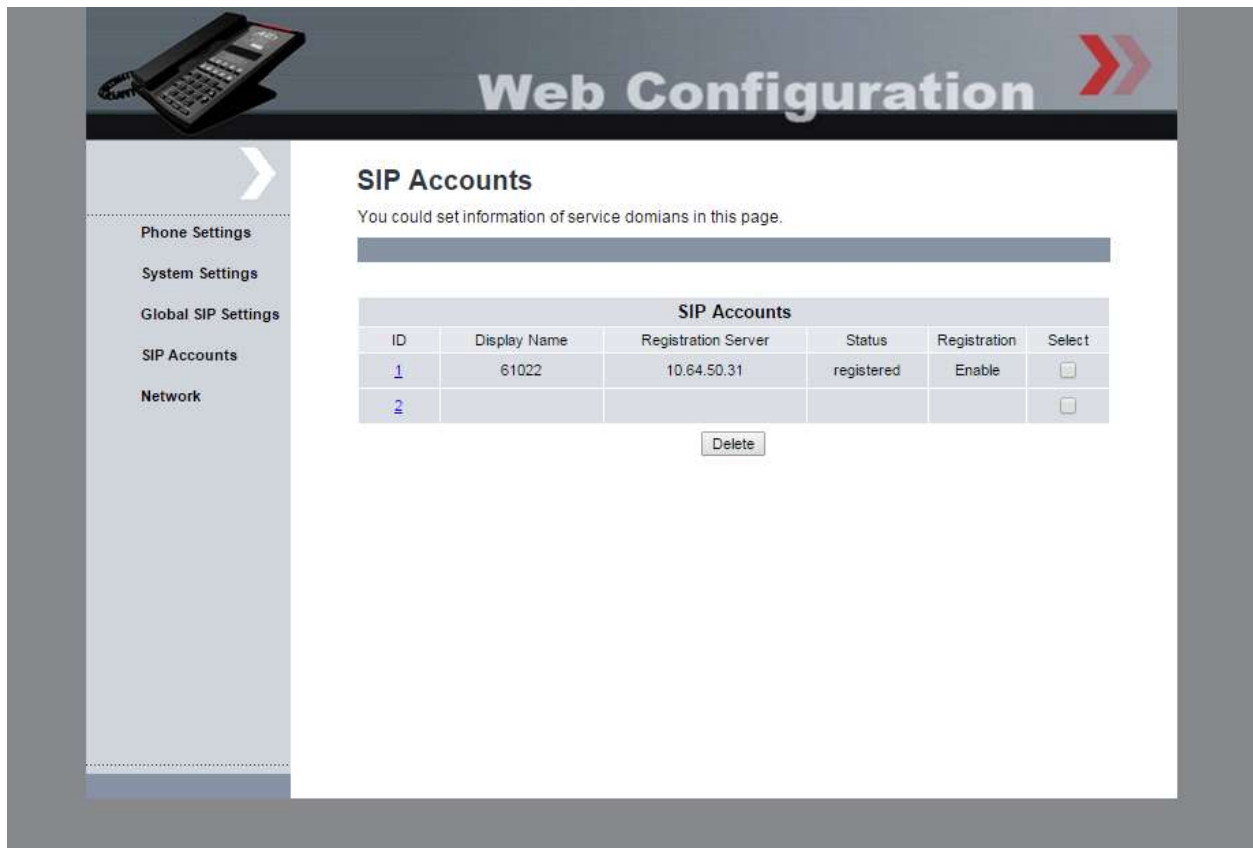


The screenshot displays the 'Web Configuration' interface for a SIP phone. At the top, there is a header with a phone icon on the left and the text 'Web Configuration' with a red double arrow icon on the right. Below the header is a navigation menu on the left with the following items: 'Phone Settings', '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:	201212000034
Firmware Version:	190220.16.2.01D10

7.2. Configure SIP Account

Select **SIP Accounts** from the left menu, and click the desired **ID**.



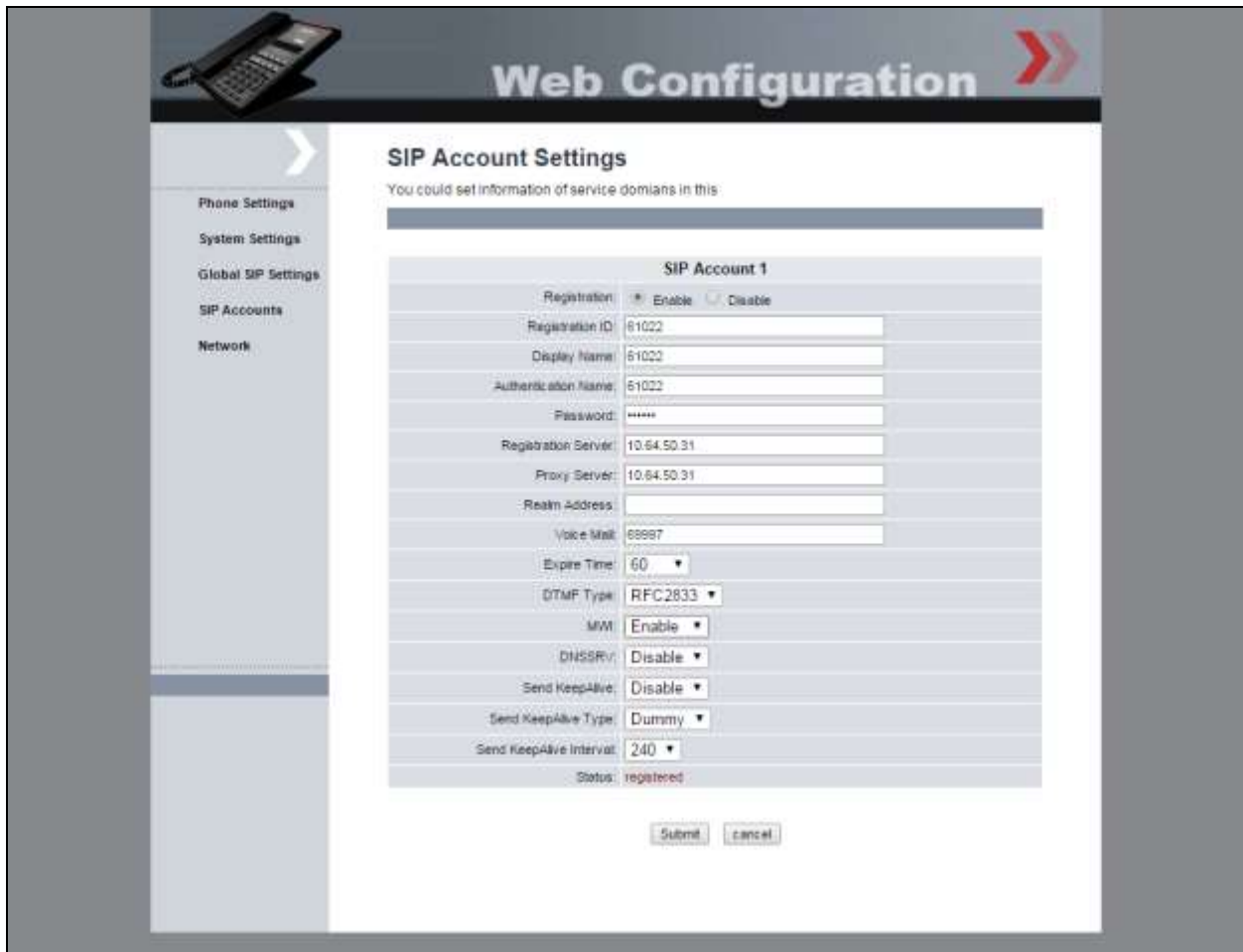
The screenshot displays the 'Web Configuration' interface for SIP Accounts. The left sidebar contains the following menu items: Phone Settings, System Settings, Global SIP Settings, SIP Accounts (highlighted), and Network. The main content area is titled 'SIP Accounts' and includes the text: 'You could set information of service domians in this page.' Below this text is a table with the following data:

SIP Accounts					
ID	Display Name	Registration Server	Status	Registration	Select
1	61022	10.64.50.31	registered	Enable	<input type="checkbox"/>
2					<input type="checkbox"/>

Below the table is a 'Delete' button.

Set **Registration:** to **Enable**. Enter username configured on Session Manager for **Registration ID:**, **Display Name:**, and **Authentication Name:**. Provide user password for **Password:**. In the **Registration Server** and **Proxy Server** fields enter the IP Address of Session Manager's signaling interface. Enter **Voice Mail:** Number, select **RFC2833** for **DTMF Type:** and select **Enable** for **MWI:**

Click **Submit**, and **Status:** should display **Registered**.



The screenshot shows a web configuration page titled "Web Configuration" with a "SIP Account Settings" section. The page includes a navigation menu on the left with options like "Phone Settings", "System Settings", "Global SIP Settings", "SIP Accounts", and "Network". The main content area displays the configuration for "SIP Account 1".

SIP Account 1	
Registration:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Registration ID:	61022
Display Name:	61022
Authentication Name:	61022
Password:	*****
Registration Server:	10.64.50.31
Proxy Server:	10.64.50.31
Realm Address:	
Voice Mail:	68887
Expire Time:	60
DTMF Type:	RFC2833
MWI:	Enable
DNSSRV:	Disable
Send KeepAlive:	Disable
Send KeepAlive Type:	Dummy
Send KeepAlive Interval:	240
Status:	registered

Buttons: Submit, Cancel

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Session Manager and G-Tek/AEi SSP-2210-SG 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.

The screenshot displays the Avaya System Manager 6.3 interface. The left sidebar contains a navigation menu with categories like Session Manager, Device and Location Configuration, Application Configuration, System Status, and System Tools. The main content area is titled "User Registrations" and includes a table of registered users. The table has the following columns: Details, Address, First Name, Last Name, Actual Location, IP Address, Remote Office, Shared Control, Simult. Devices, AST Device, and Registered (Prim, Sec, Surv). A single user is listed: Chuck Cheese, IP Address 10.64.33.212, with a checked box in the Registered Prim column. The interface also shows various controls like "View", "AST Device Notifications", and "Filter: Disable, Apply, Clear".

Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered		
										Prim	Sec	Surv
Show	61620@4427.com	Chuck	Cheese	---	10.64.33.212	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

From the web interface of the G-Tek/AEi SSP-2210-SG phone, select **SIP Accounts** from the left menu. Verify that the **Status** field shows **registered**.

Web Configuration

SIP Accounts

You could set information of service domians in this page.

SIP Accounts					
ID	Display Name	Registration Server	Status	Registration	Select
1	61022	10.64.50.31	registered	Enable	<input type="checkbox"/>
2					<input type="checkbox"/>

Delete

9. Conclusion

These Application Notes describe the configuration steps required for G-Tek/AEi Communications SSP-2210-SG SIP Phones to successfully interoperate with Avaya Aura® Session Manager 6.3 and Avaya Aura® Communication Manager 6.3. 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, Document 03-300509
- [2] Administering Avaya Aura® Session Manager, Document 03-603324

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