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@aeicommunications.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.

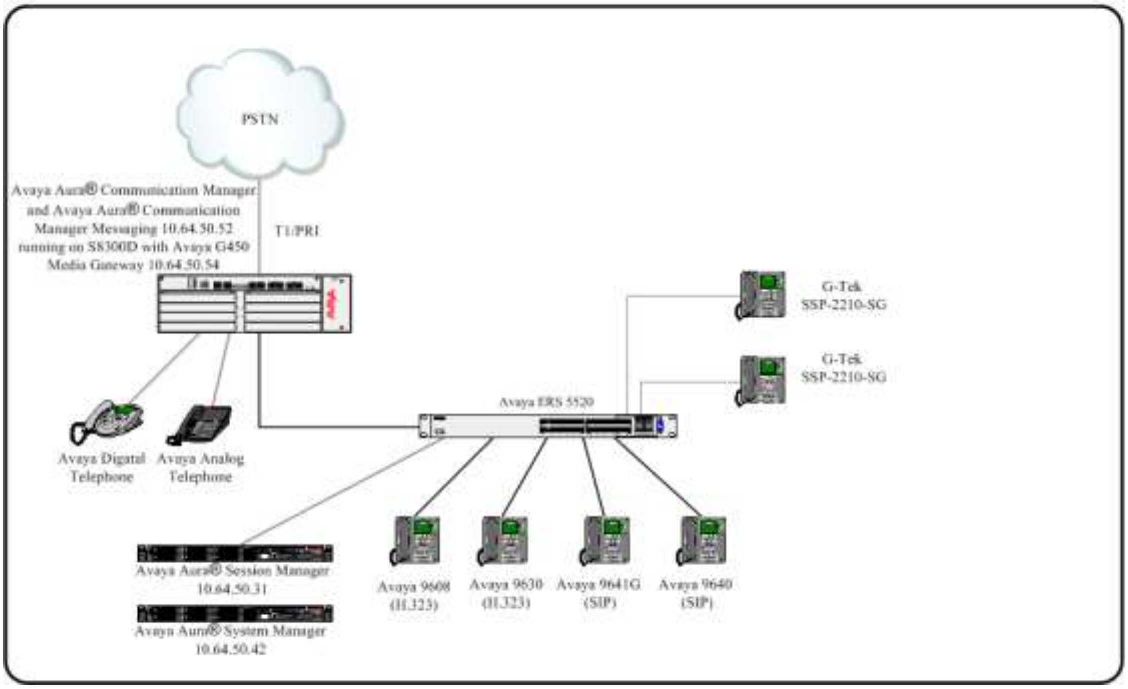


Figure 1: G-Tek/AEi SSP-2210-SG SIP Phone Reference Configuration

4. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya PBX Products	
Avaya Aura® Communication Manager	6.3 SP6
Avaya Aura® Session Manager	6.3.8
Avaya Messaging (Voice Mail) Products	
Avaya Aura® Communication Manager Messaging	6.3 SP6
Avaya Endpoints	
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
G-Tek/AEi Products	
SSP-2210-SG	190220.16.2.01D10

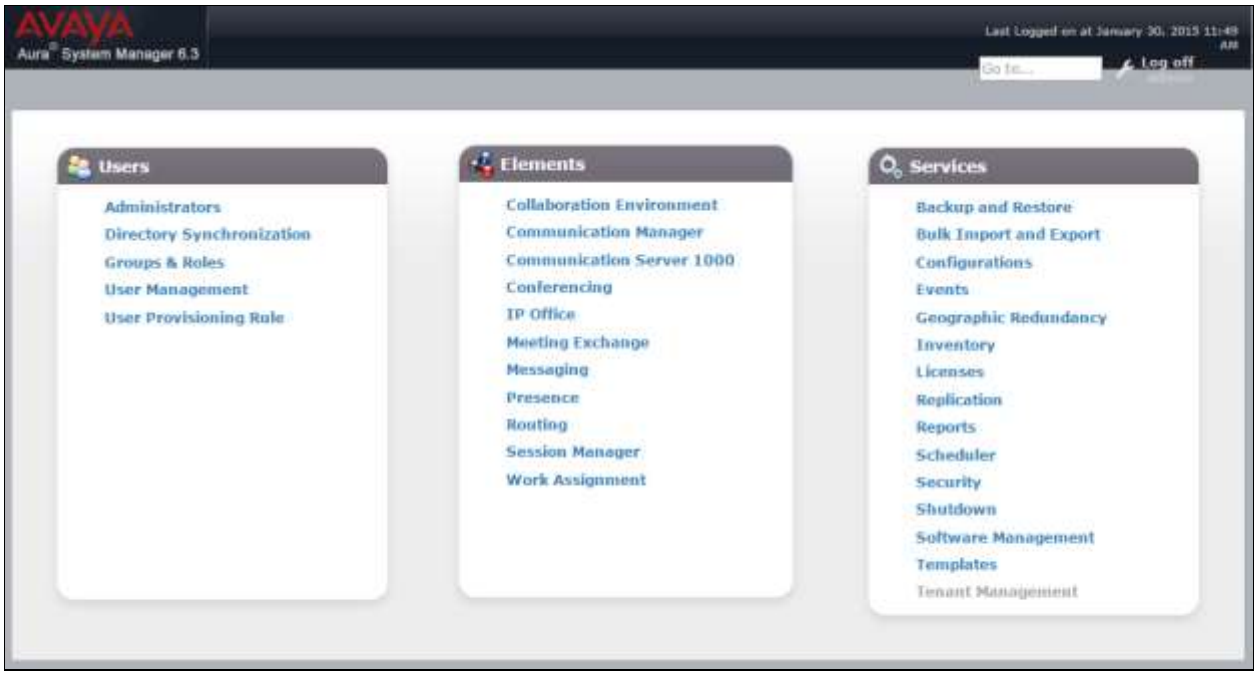
5. Configure Avaya Aura® Session Manager

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

- Launch System Manager
- Administer users

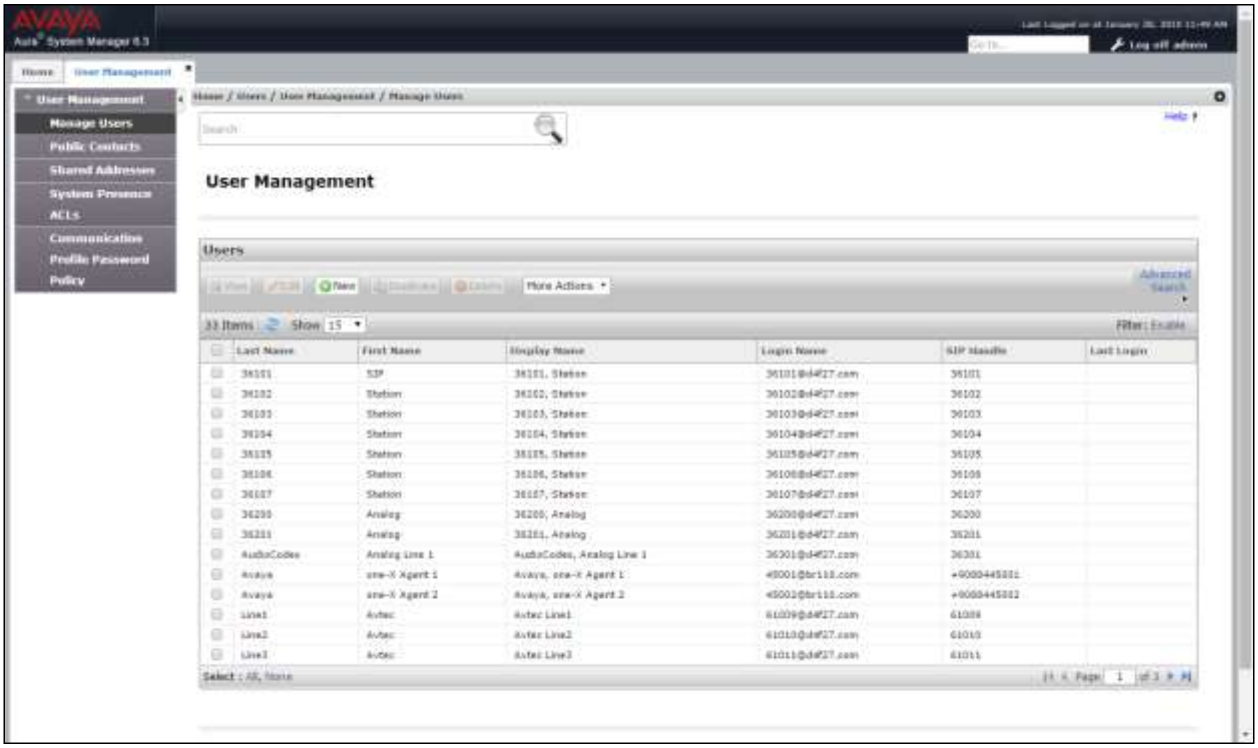
5.1. Launch System Manager

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.



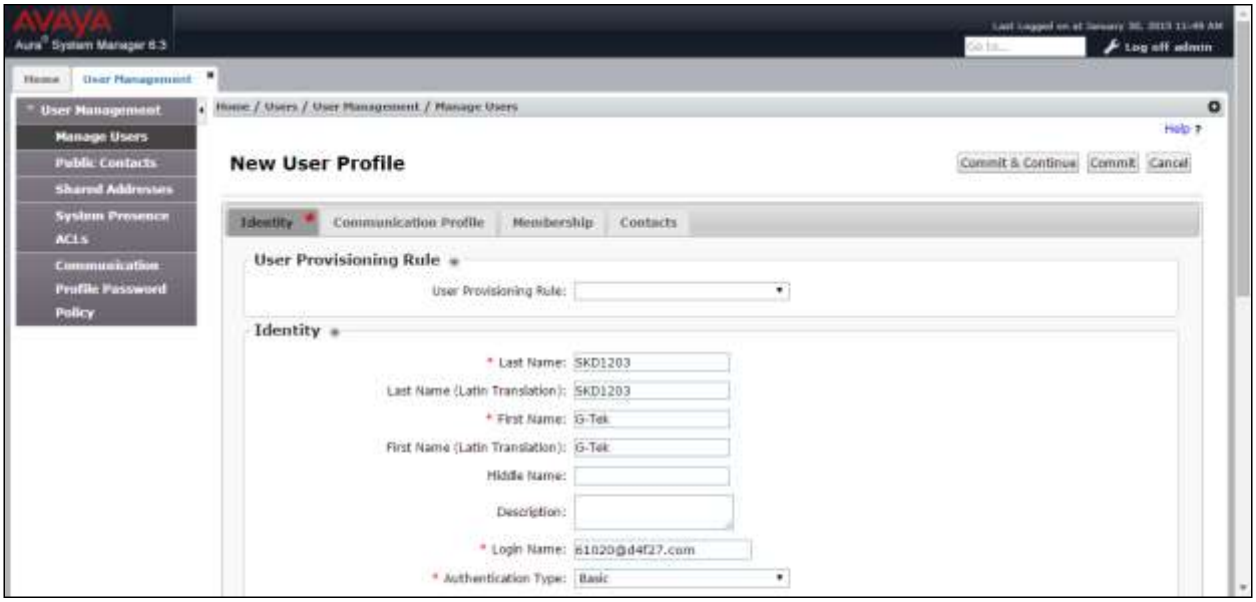
5.2. Administer Users

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



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.



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.

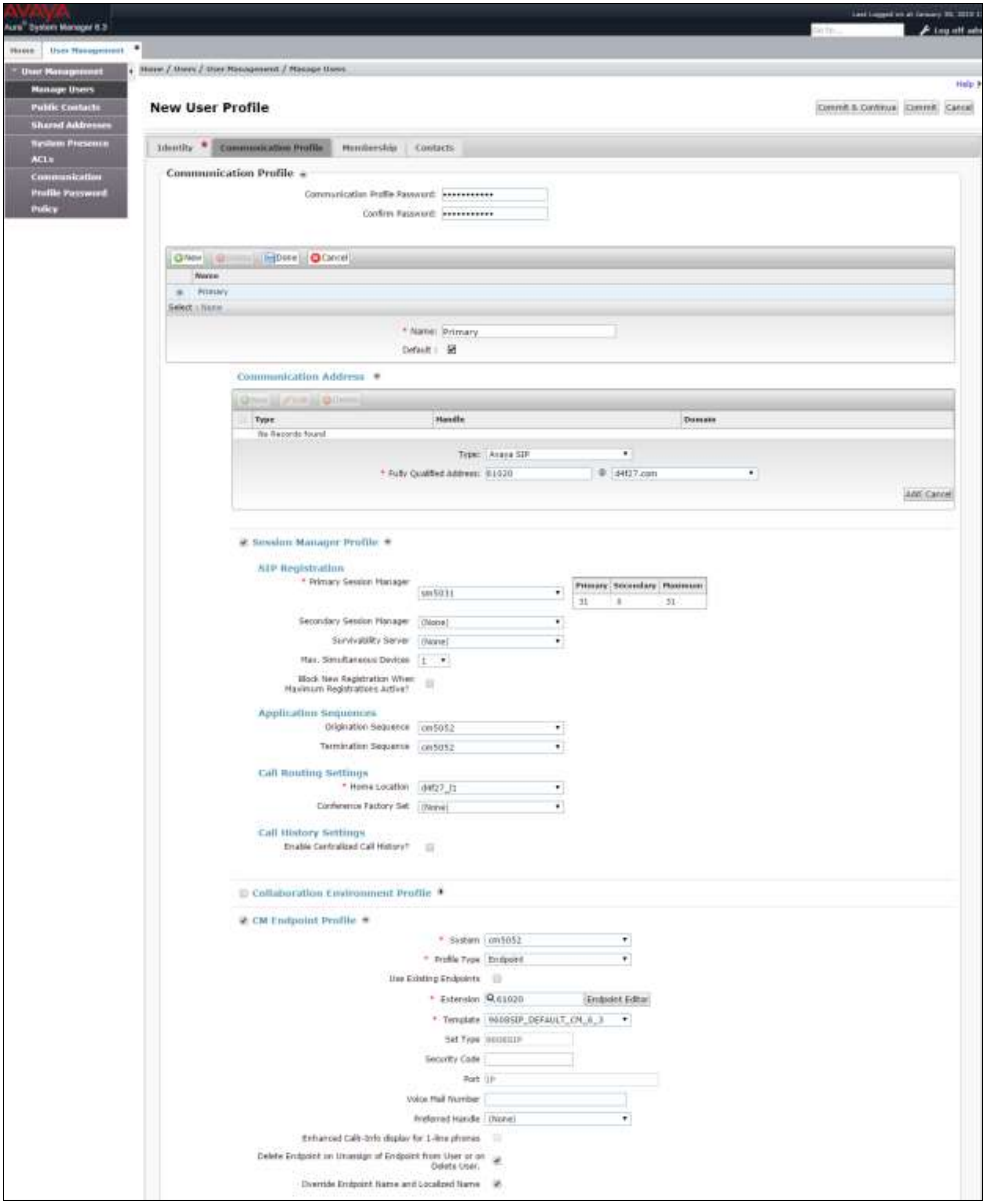
For **Type**, retain "Avaya SIP". For **Fully Qualified Address**, enter and select the SIP user extension and domain name from **Section 5.2.1**. Click **Add**.

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.



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:
```

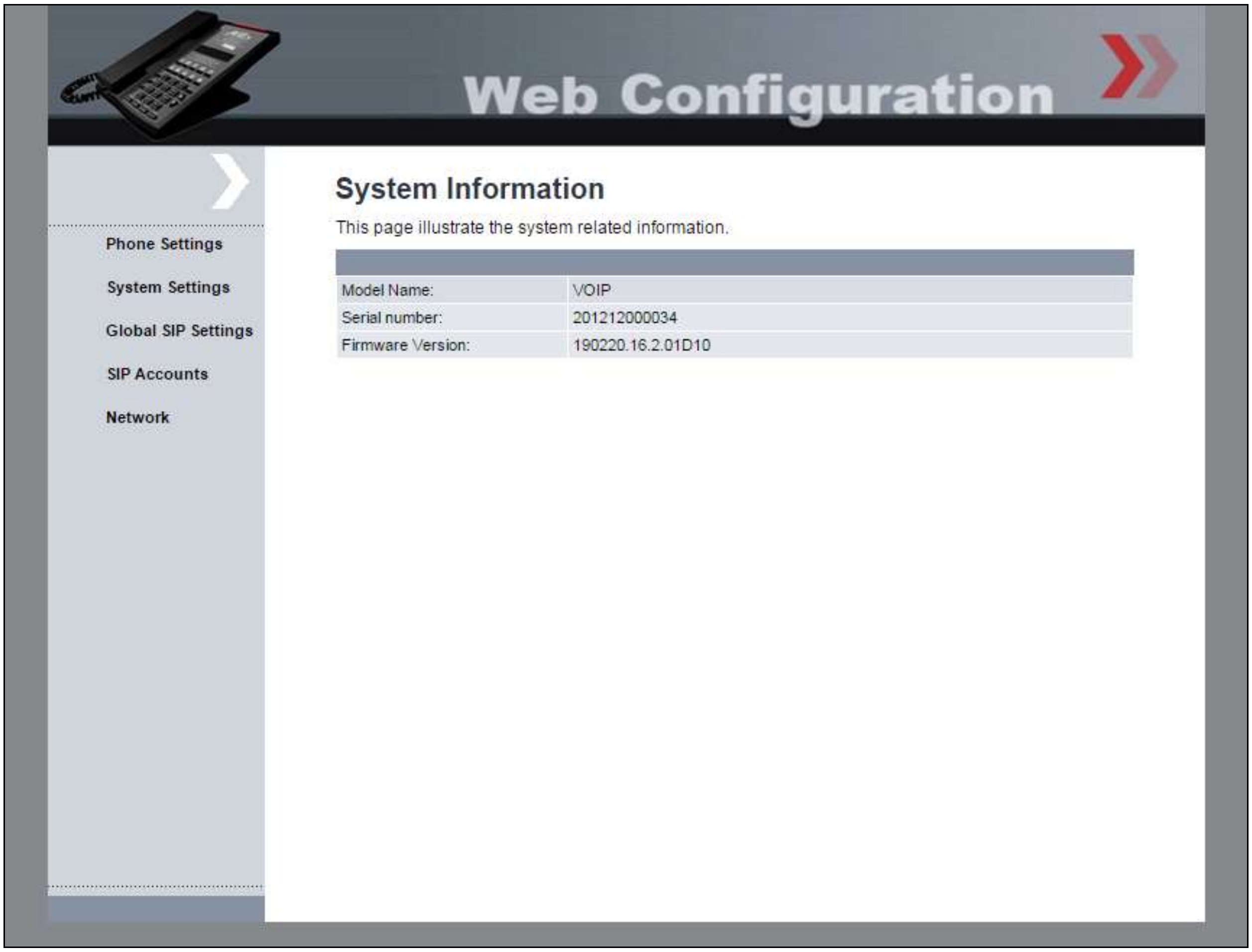
7. Configure G-Tek/AEi SSP-2210-SG SIP Phones

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

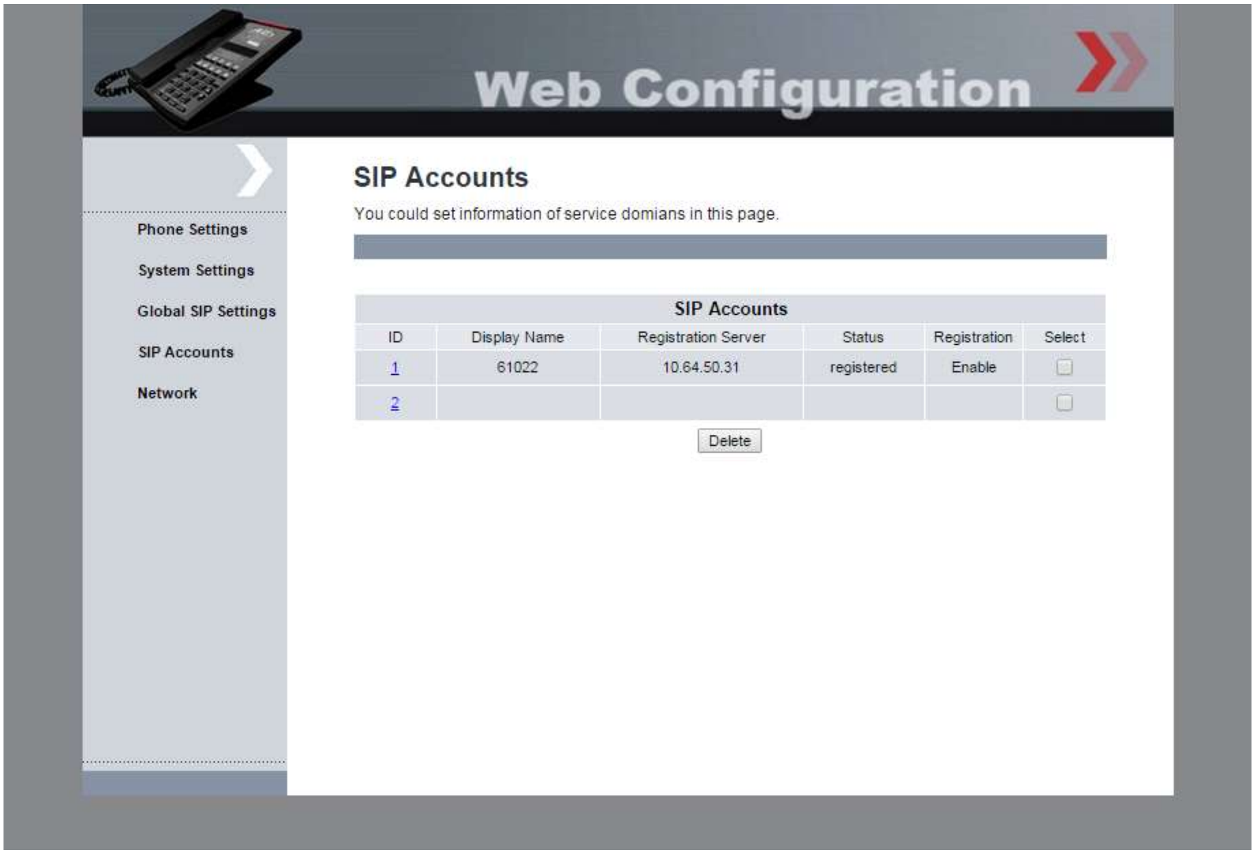
7.1. Access Web Interface

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.



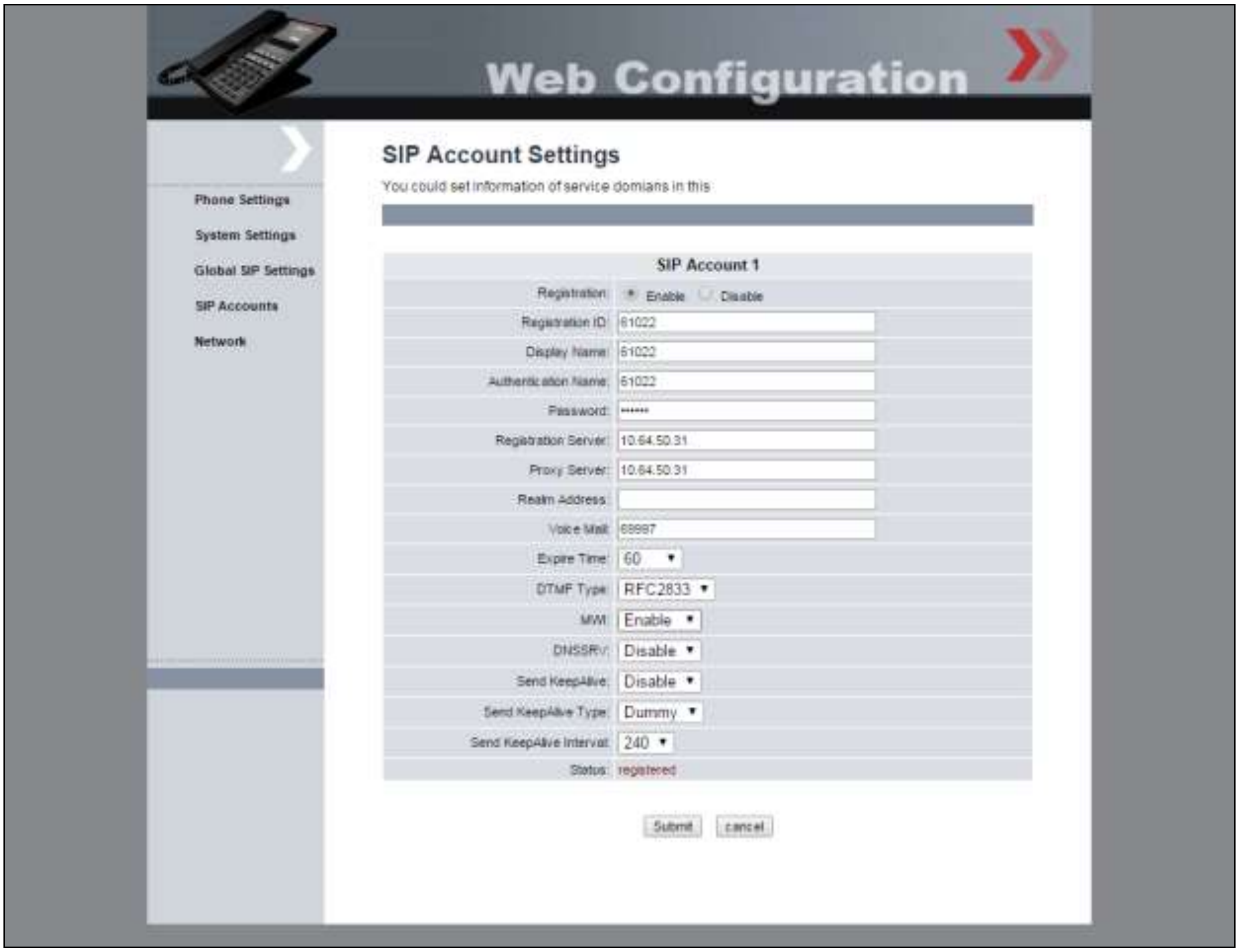
7.2. Configure SIP Account

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



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:**

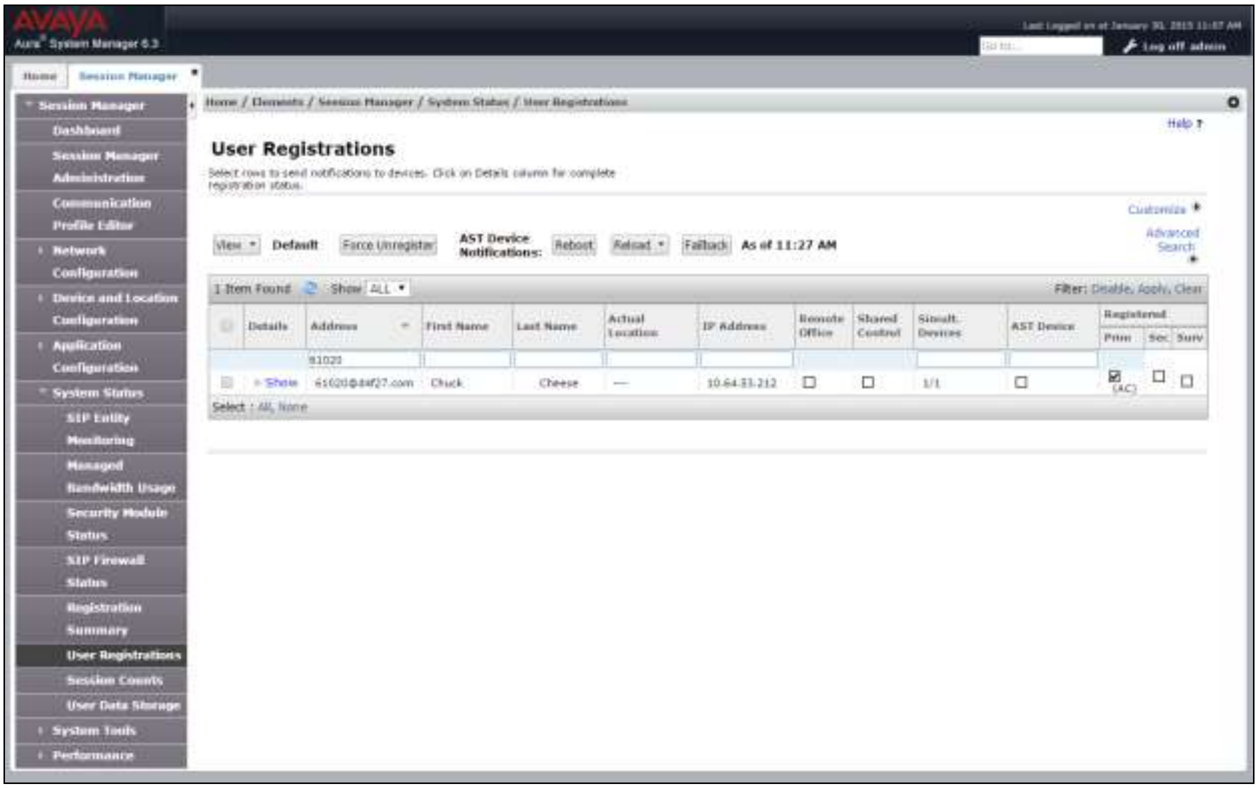
Click **Submit**, and **Status:** should display **Registered**.



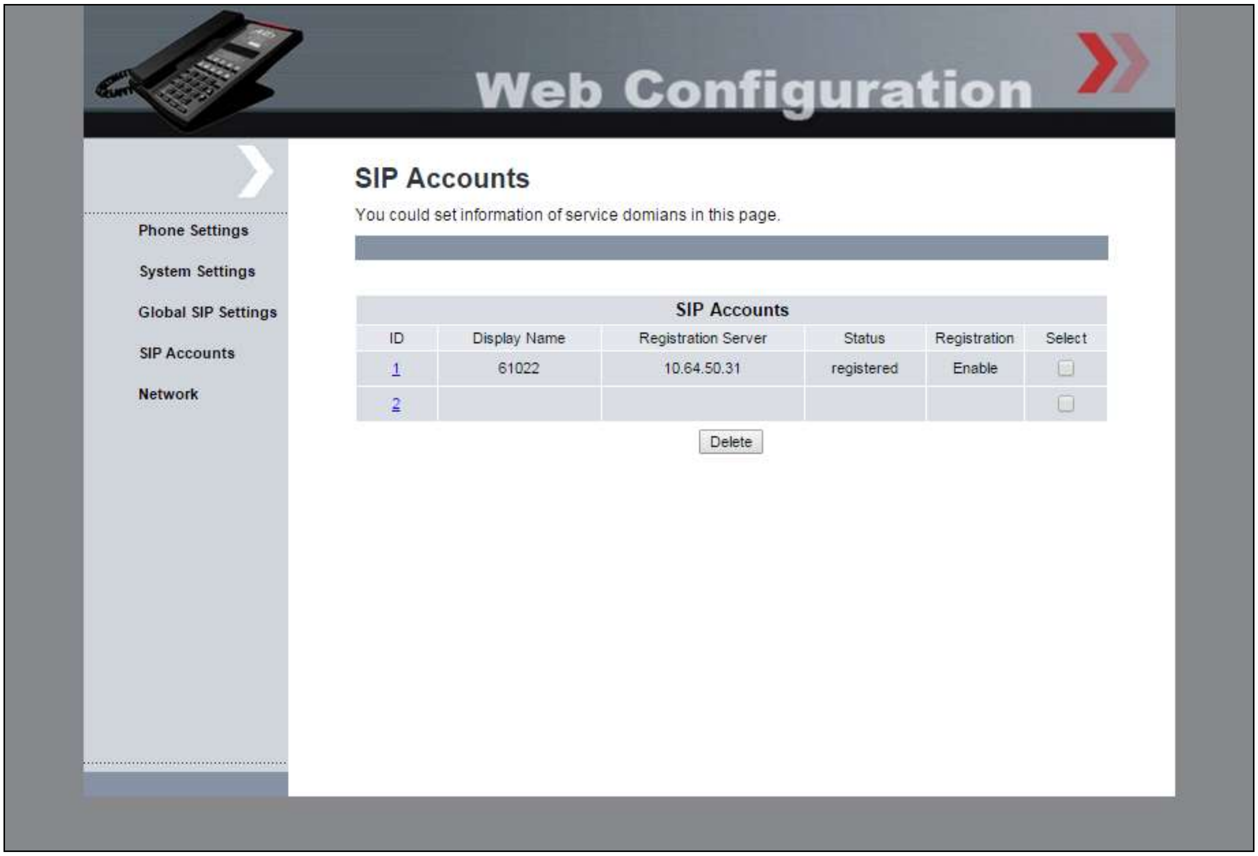
8. Verification Steps

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.



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.**



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

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.