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.

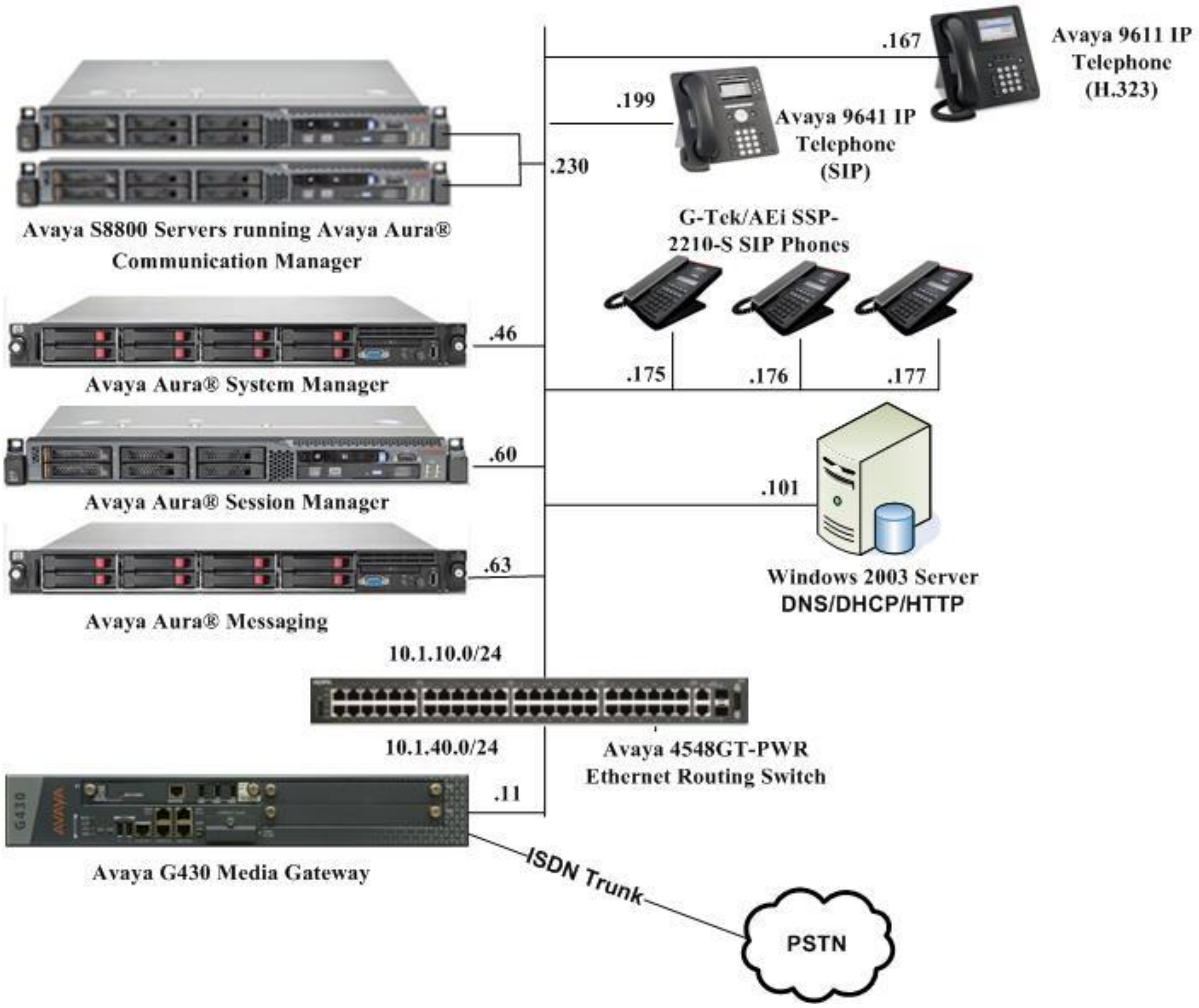


Figure 1: G-Tek/AEi SSP-2x10-S SIP Phones with Avaya SIP Solution

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

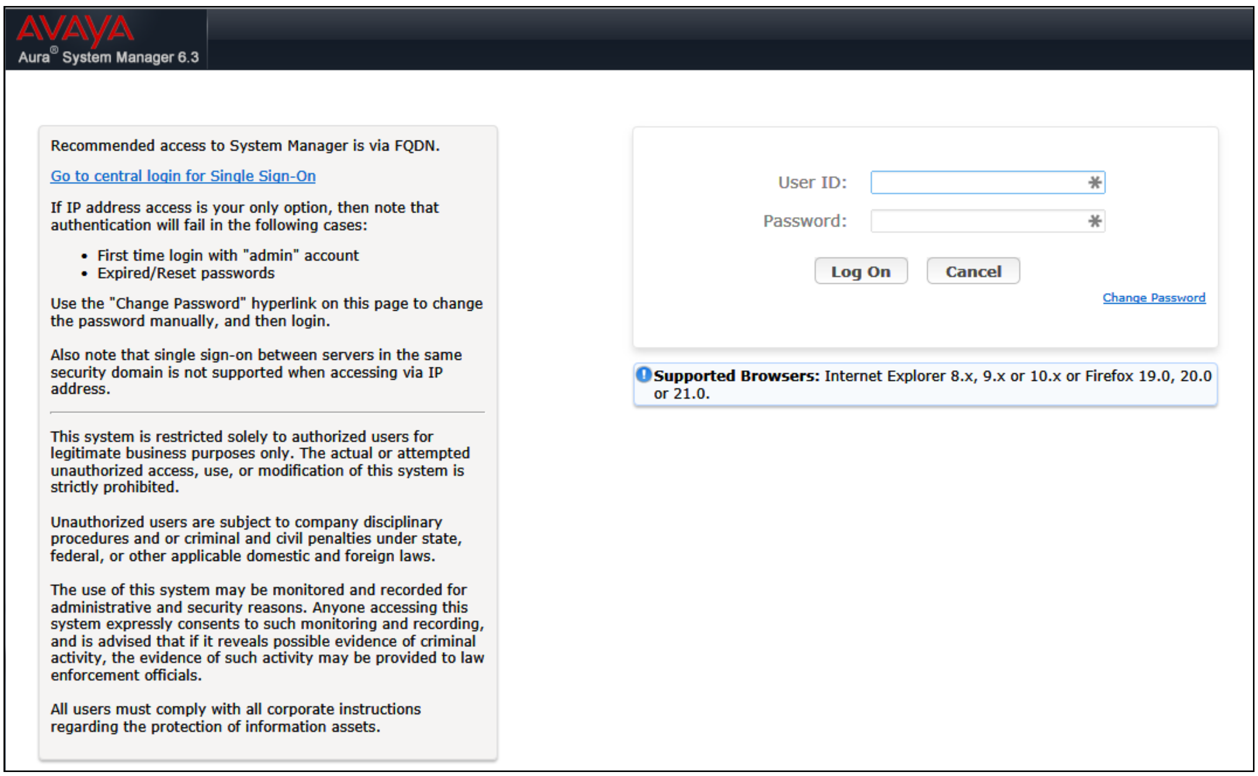
5. Configure Avaya Aura® Session Manager

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

- Launch System Manager
- Administer users

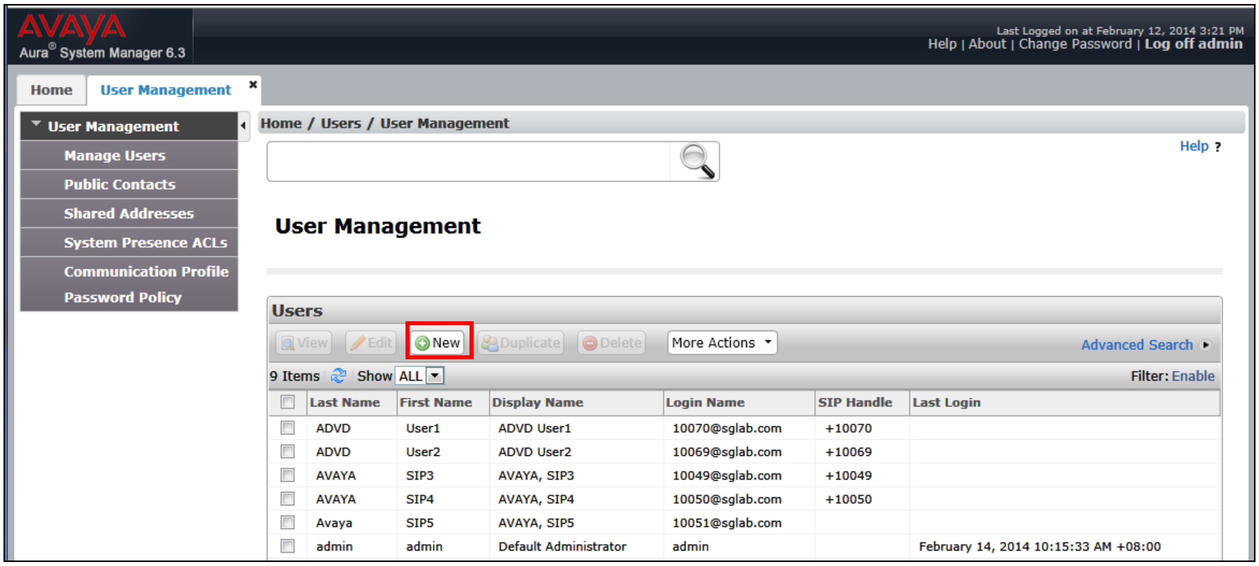
5.1. Launch System Manager

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.



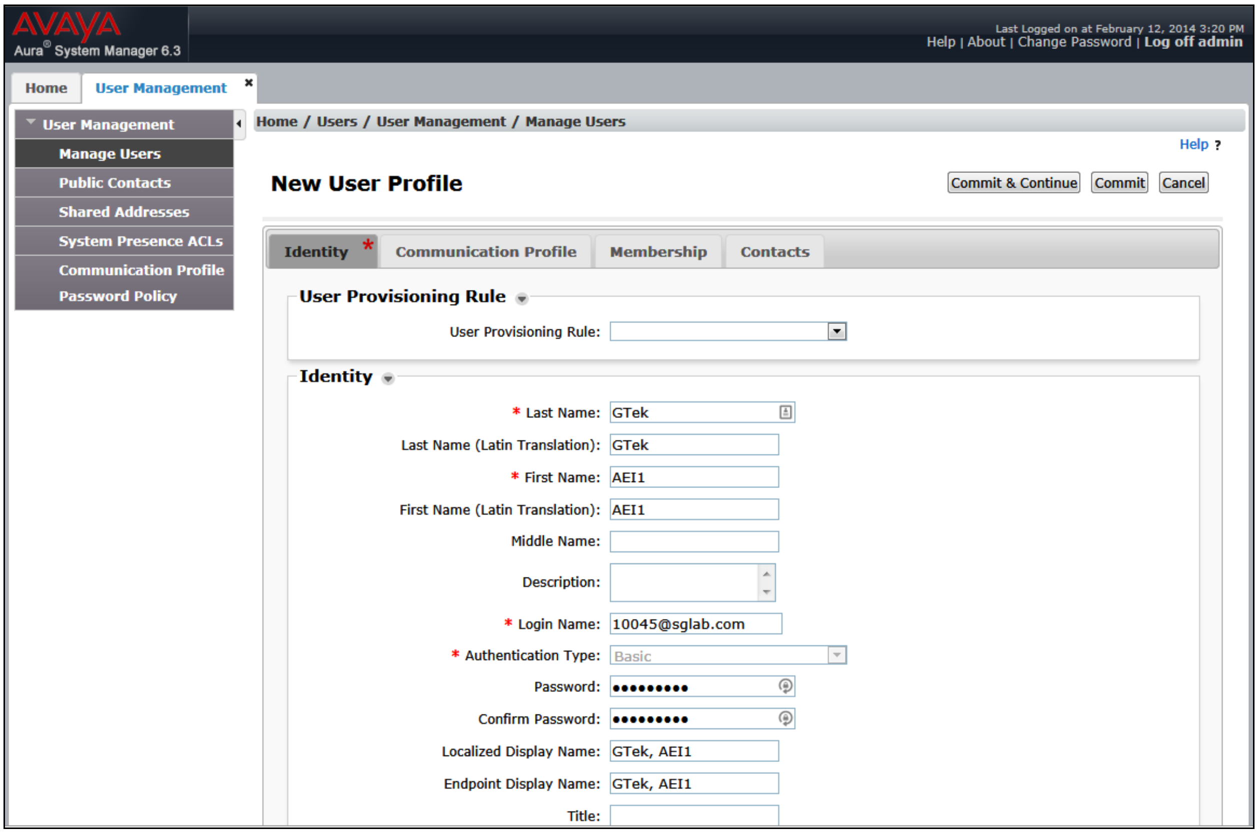
5.2. Administer Users

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.



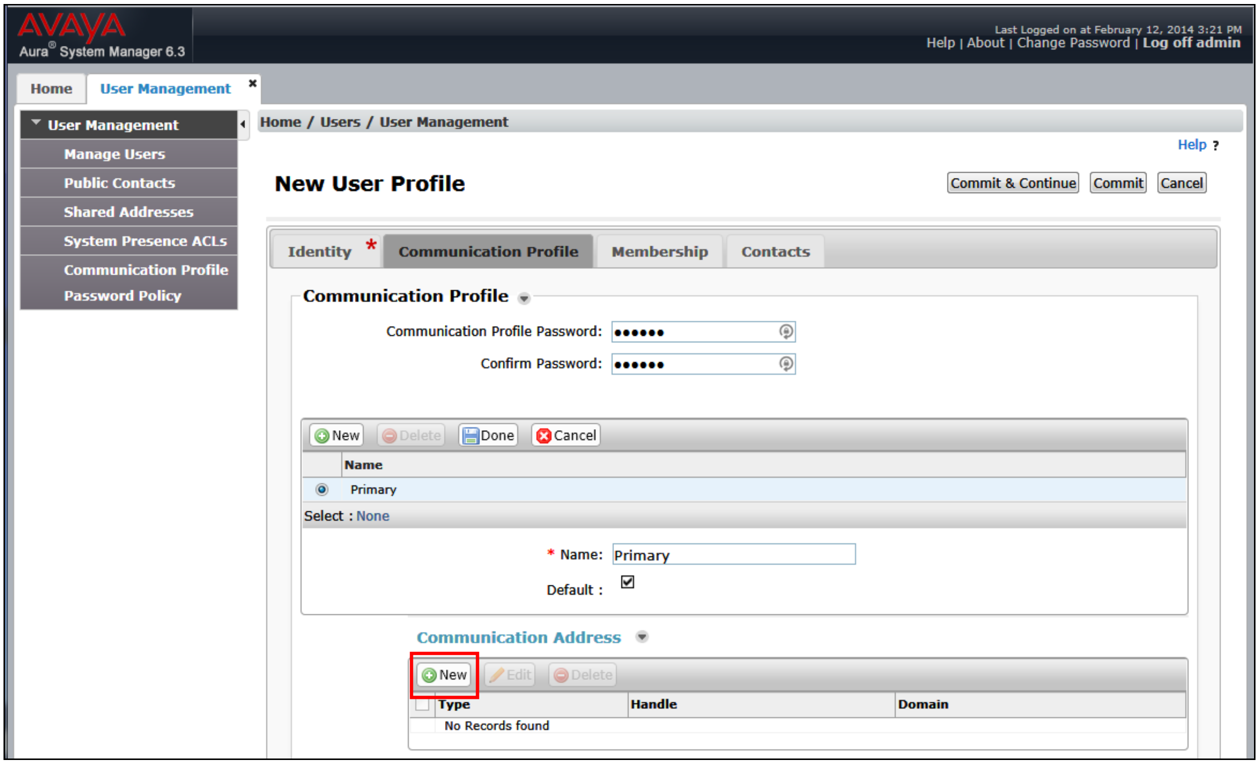
5.2.1. Identity

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.

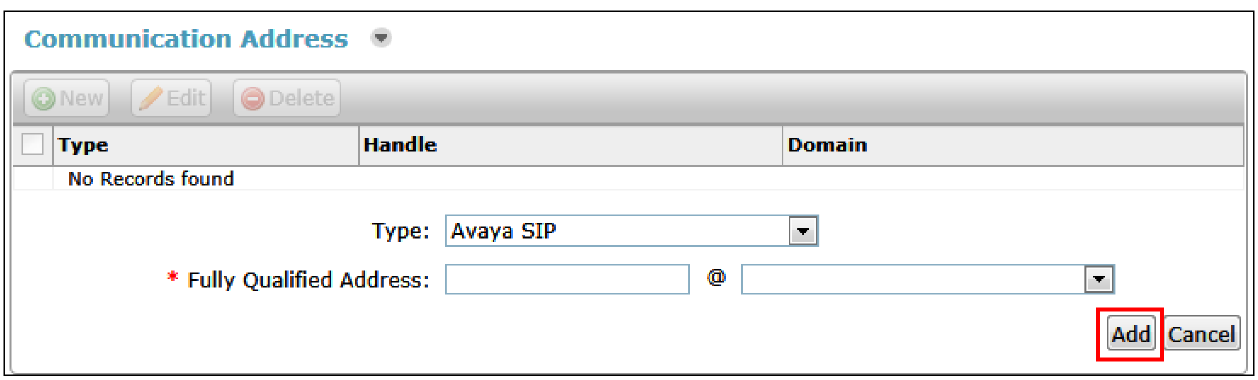


5.2.2. Communication Profile

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.

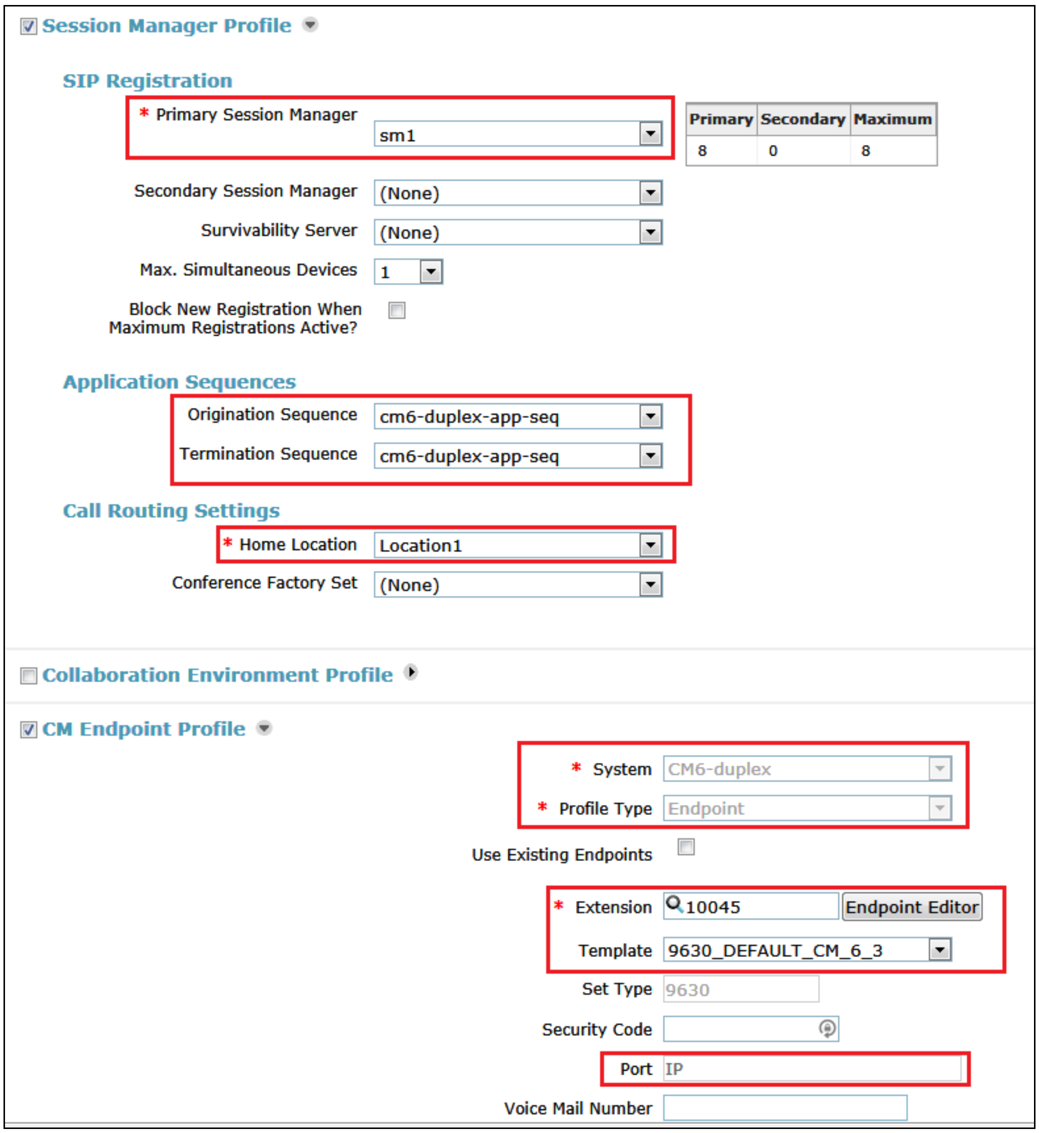


For **Type**, retain "Avaya SIP". For **Fully Qualified Address**, enter and select the SIP user extension and domain name from **Section** Error! Reference source not found.. Click **Add**.



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.



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:
```

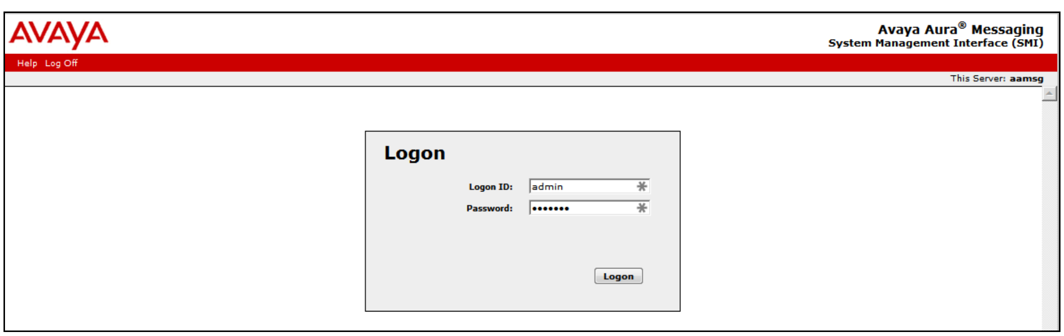
7. Configure Avaya Aura® Messaging

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:

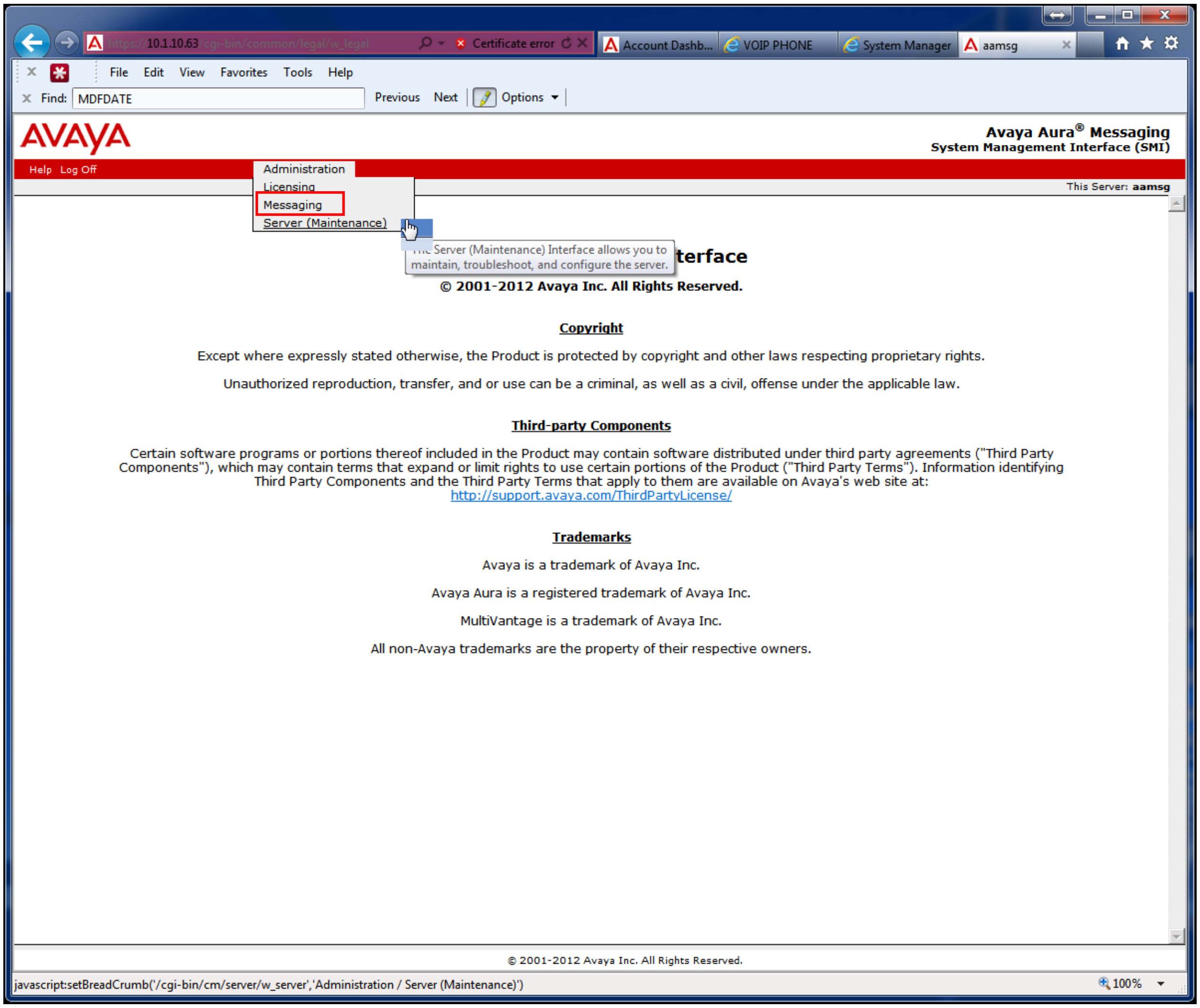
- Launch Messaging System Management Interface (SMI)
- Administer users

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).

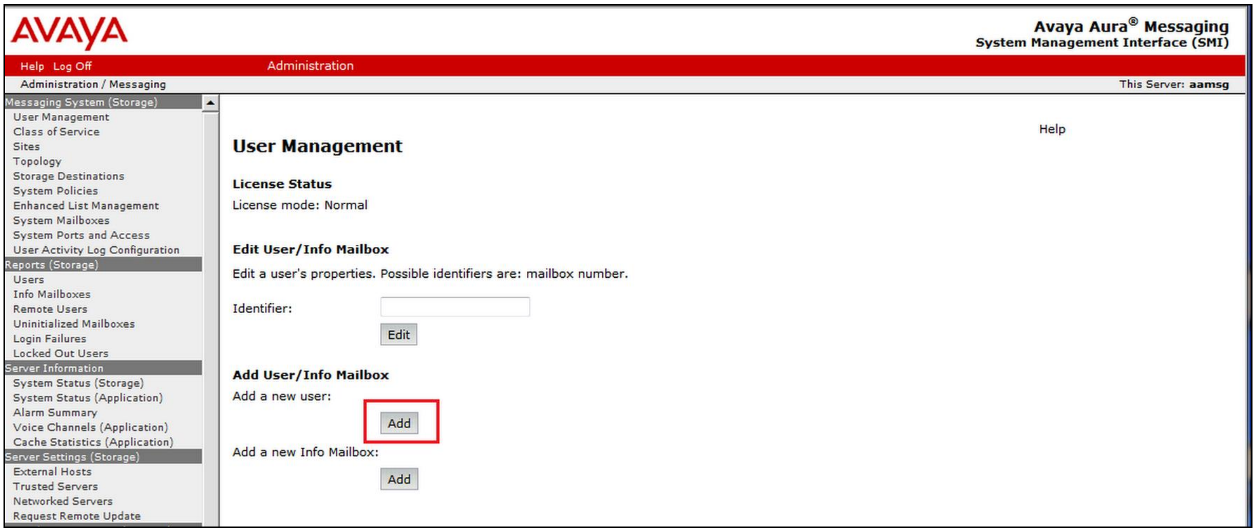


Select **Messaging** from the **Administration** drop down menu on top.

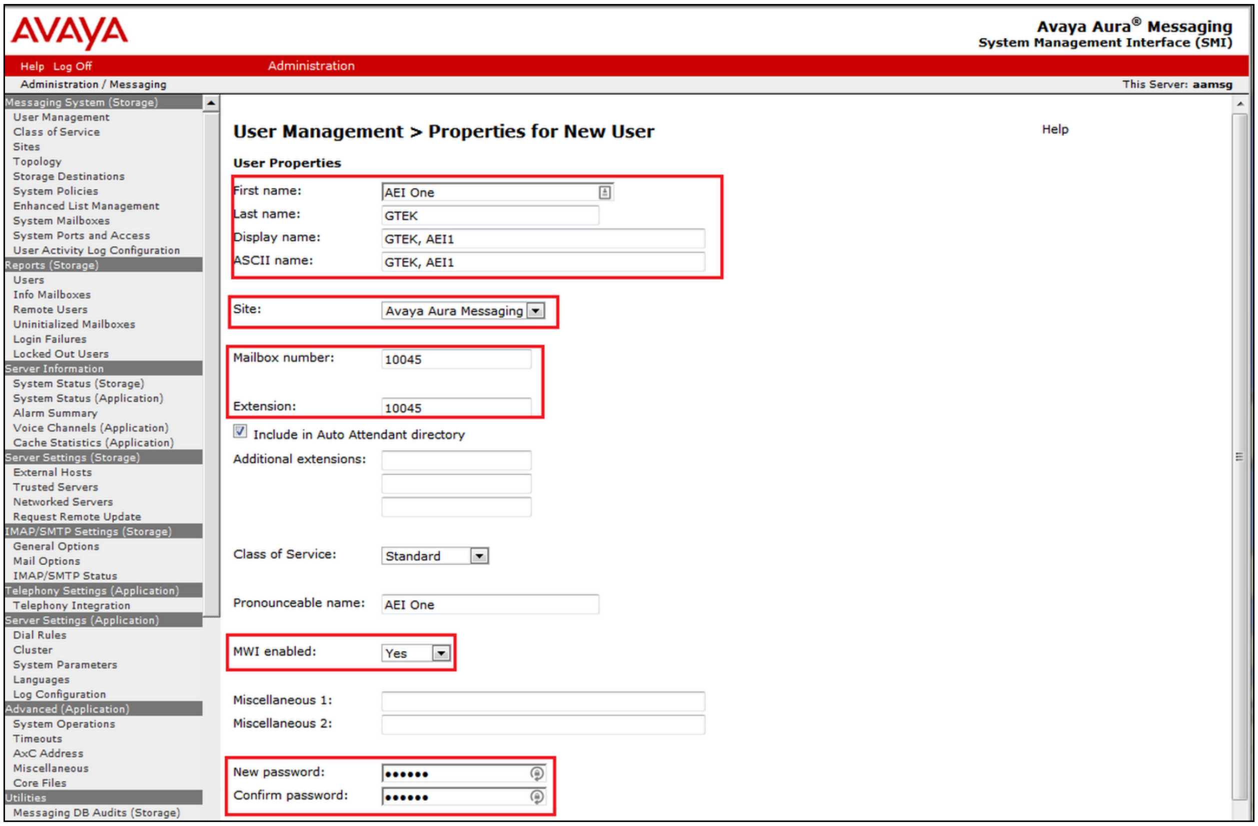


7.2. Administer Users

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



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.



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

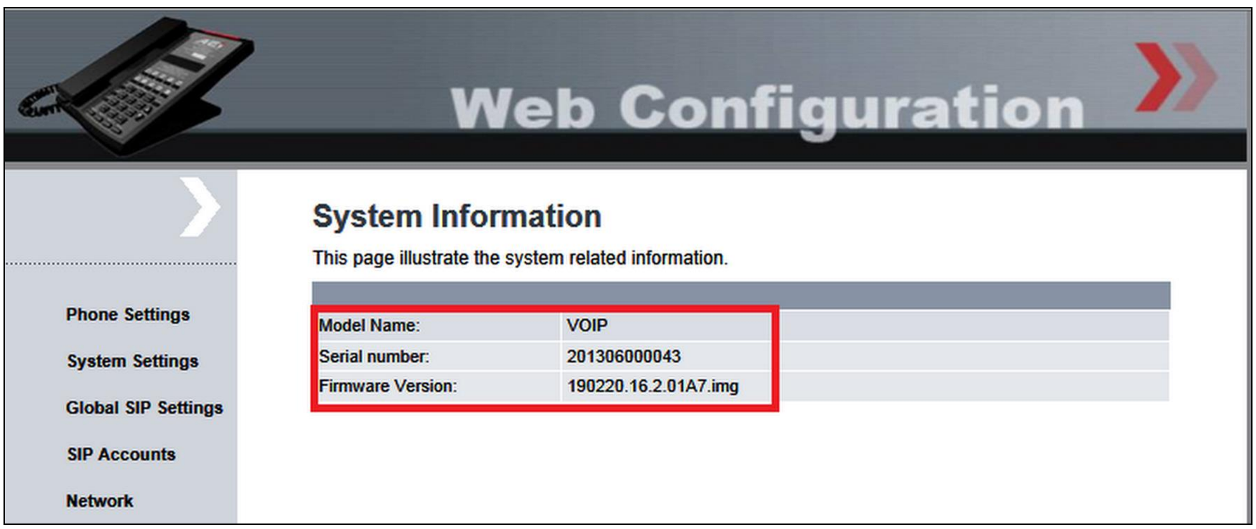
This section provides the procedures for configuring G-Tek/AEi SSP-2x10 SIP Phones on User 10045. The procedures include the following areas:

- 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.

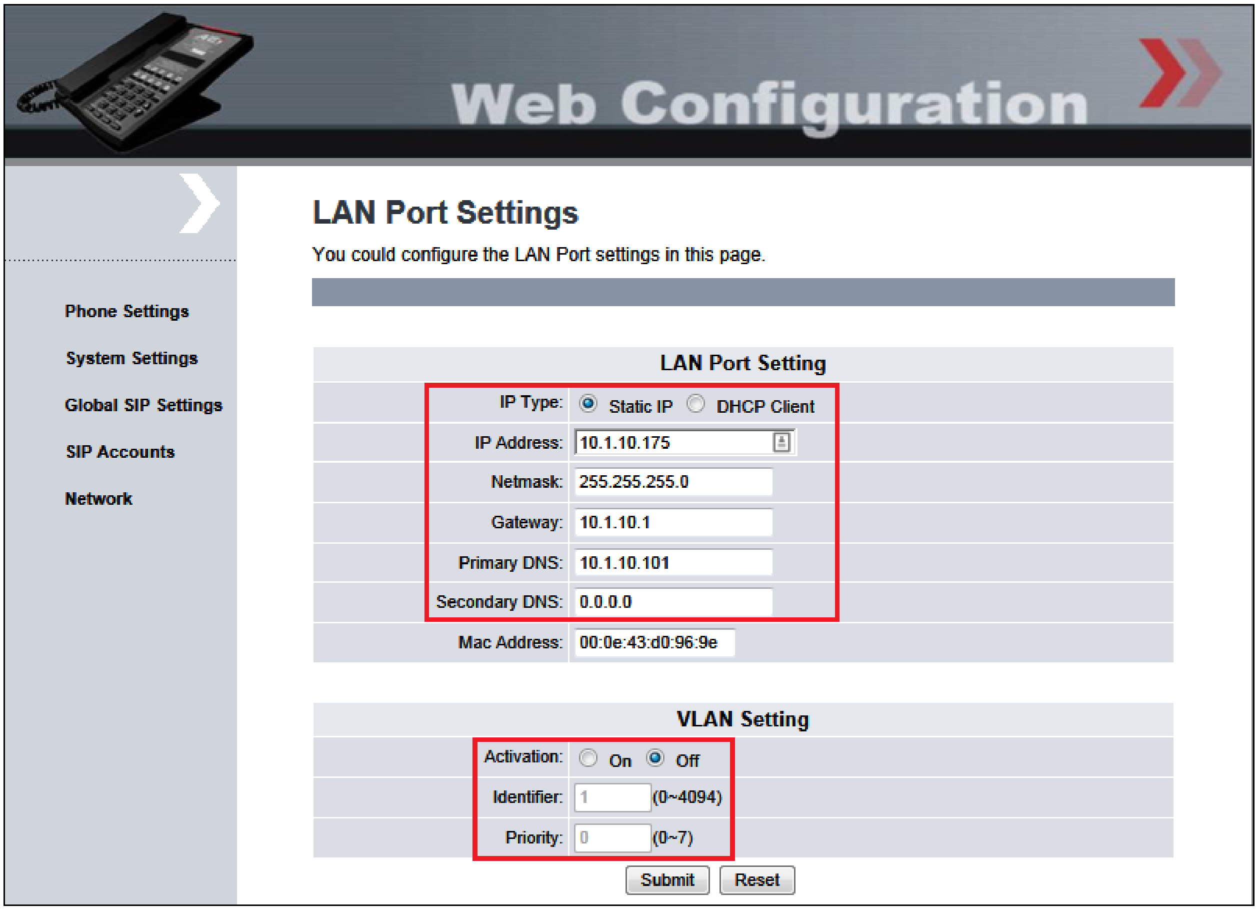
8.1. Access Web Interface

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.



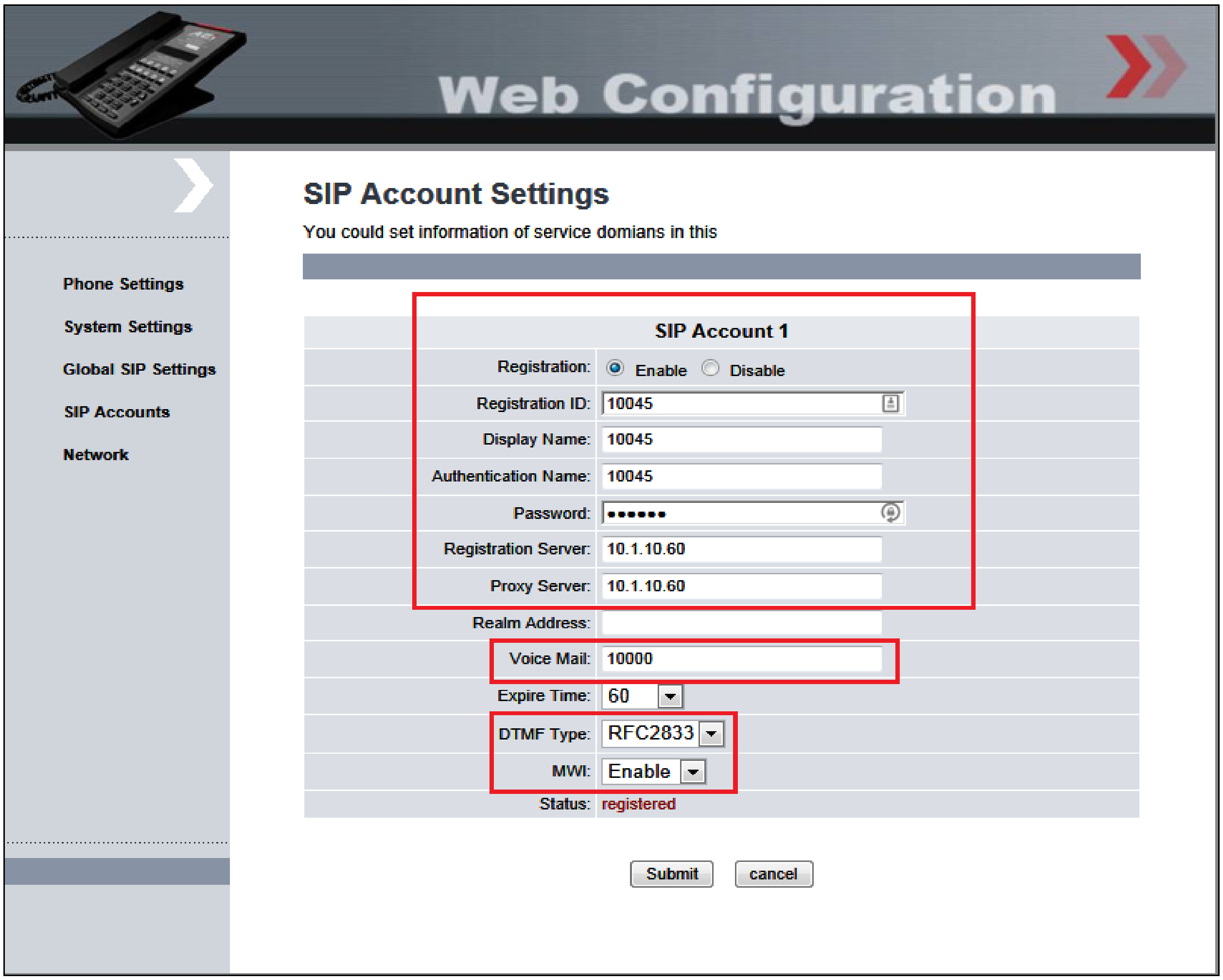
8.2. Configure LAN Port Settings

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.



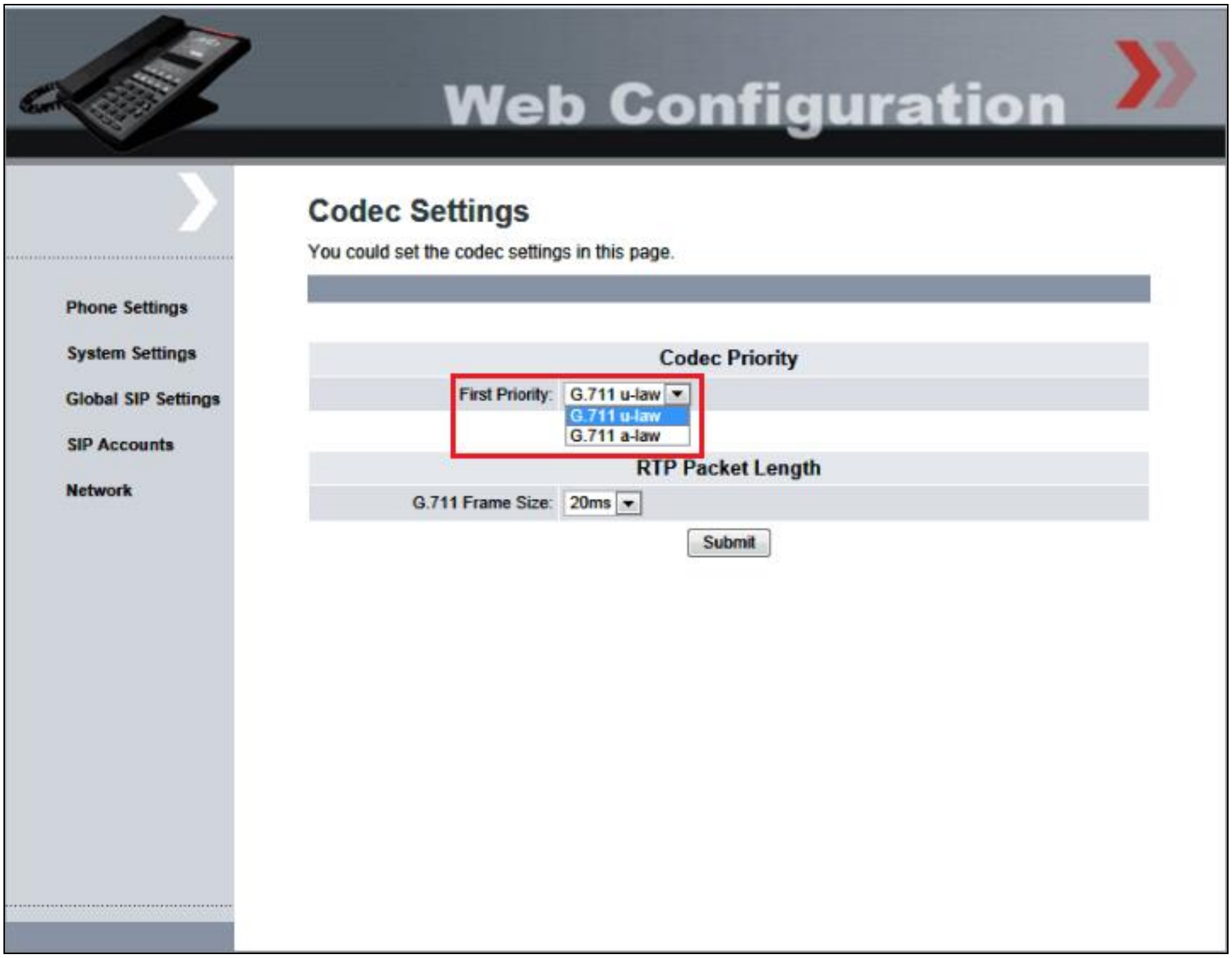
8.3. Configure SIP Account Settings

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.



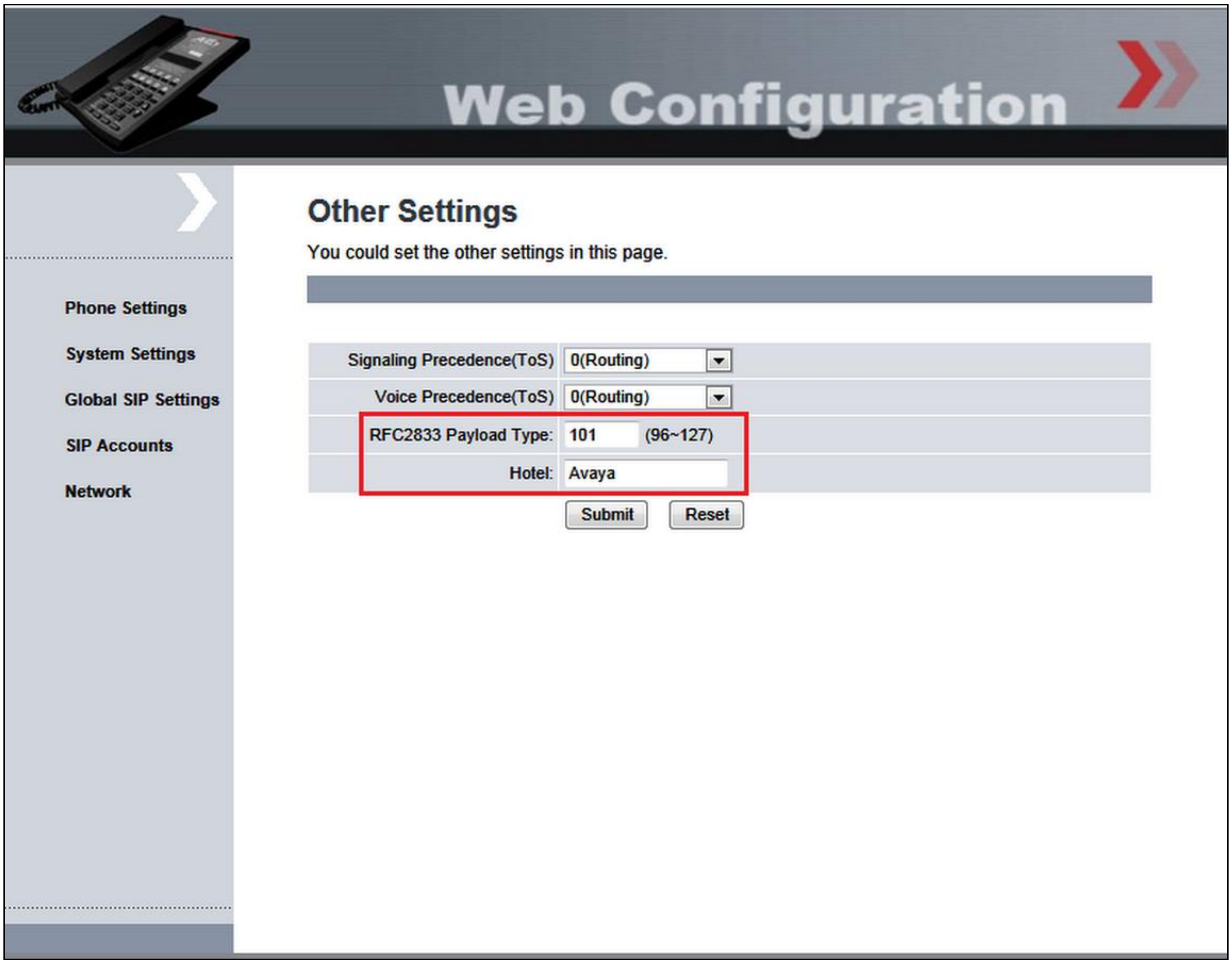
8.4. Configure Codec Settings

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



8.5. Configure SIP Settings

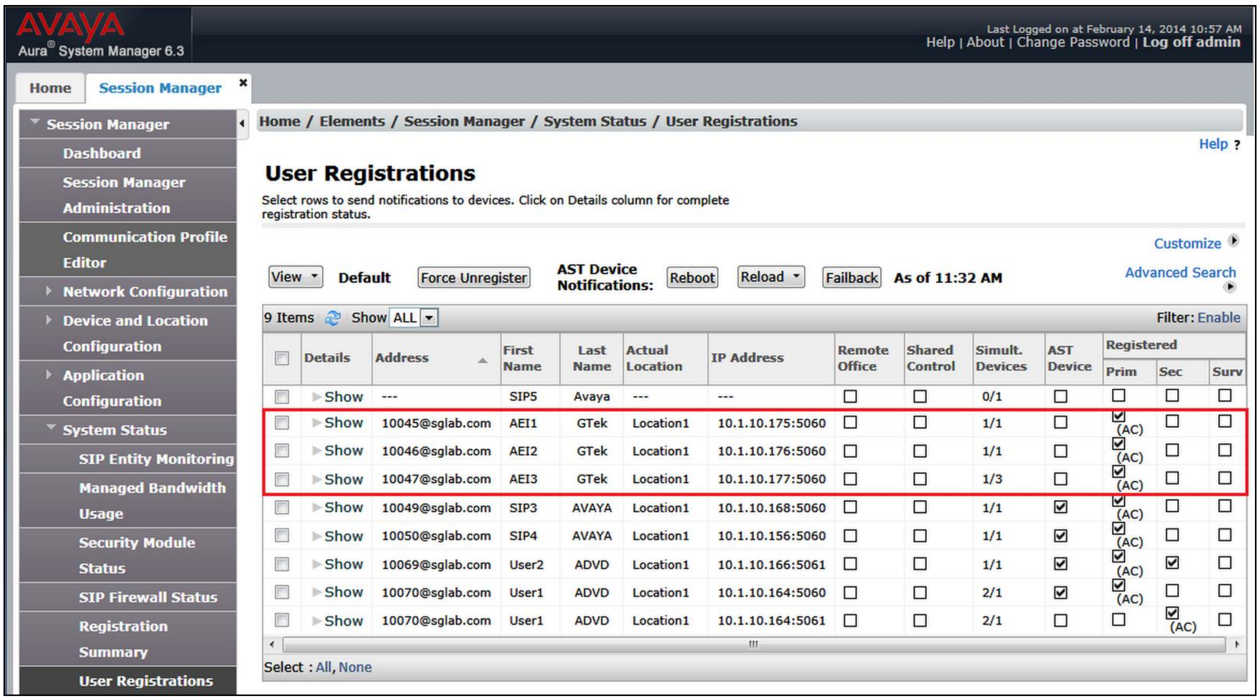
Select **Global SIP Settings → Other Settings** from the left menu. Check that **RFC2833 Payload Type** is set to 101. Enter the appropriate name for the **Hotel**. Click **Submit** to continue.



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.



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 r**egistered**.

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*

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.