Avaya Solution & Interoperability Test Lab

Application Notes for Configuring G-Tek Electronics SGR-8x06-SMK SIP Phone with Avaya Aura® Session Manager 6.1 and Avaya Aura® Communication Manager 6.0.1 - Issue 1.0

Abstract

These Application Notes describe the configuration steps required for G-Tek Electronics SGR-8x06-SMK SIP Phone with DECT handset to interoperate with Avaya Aura® Session Manager 6.1 and Avaya Aura® Communication Manager 6.0.1.

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 Electronics SGR-8x06-SMK SIP Phone with DECT handset to interoperate with an Avaya SIP infrastructure consisting of Avaya Aura® Session Manager 6.1 and Avaya Aura® Communication Manager 6.0.1. G-Tek SGR-8x06-SMK is a series of business IP phones for the enterprise.

2. General Test Approach and Test Results

To verify interoperability of G-Tek SGR-8x06-SMKSIP Phone with Session Manager and Communication Manager, calls were made between G-Tek telephones and Avaya SIP, H.323 and Digital telephones using various codec settings and exercising common PBX features. The telephony features were 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 SGR-8x06-SMK SIP Phones with Session Manager.
- Calls between G-Tek telephones and Avaya SIP, H.323, and digital telephones.
- G.711 and 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.
- PBX features including Multiple Call Appearances, Hold, Transfer, and Conference.
- Proper system recovery after a G-Tek telephone restart and loss of IP connection.

2.2. Test Results

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

- When making changes in codec settings on the web interface while the phones are in talking state, call drops. There is no warning is given on web page.
- Data sheet does not indicate support of G.729 codec but the administration web interface and testing does indicate that this codec is supported.

2.3. Support

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

- Phone: +886-2-26962665 ext. 221
- E-mail: support@G-Tek.com.tw

3. Reference Configuration

The diagram illustrates an enterprise site with an Avaya SIP-based network, including Avaya Aura® Session Manager, an S8800 Server running Avaya Aura® Communication Manager with a G450 Media Gateway, and Avaya SIP, H.323 and Digital endpoints. The enterprise site also contains two G-Tek SGR-8x06-SMK SIP Phones used in the compliance testing. The G-Tek phones are registered with Session Manager and are configured as endpoint users.

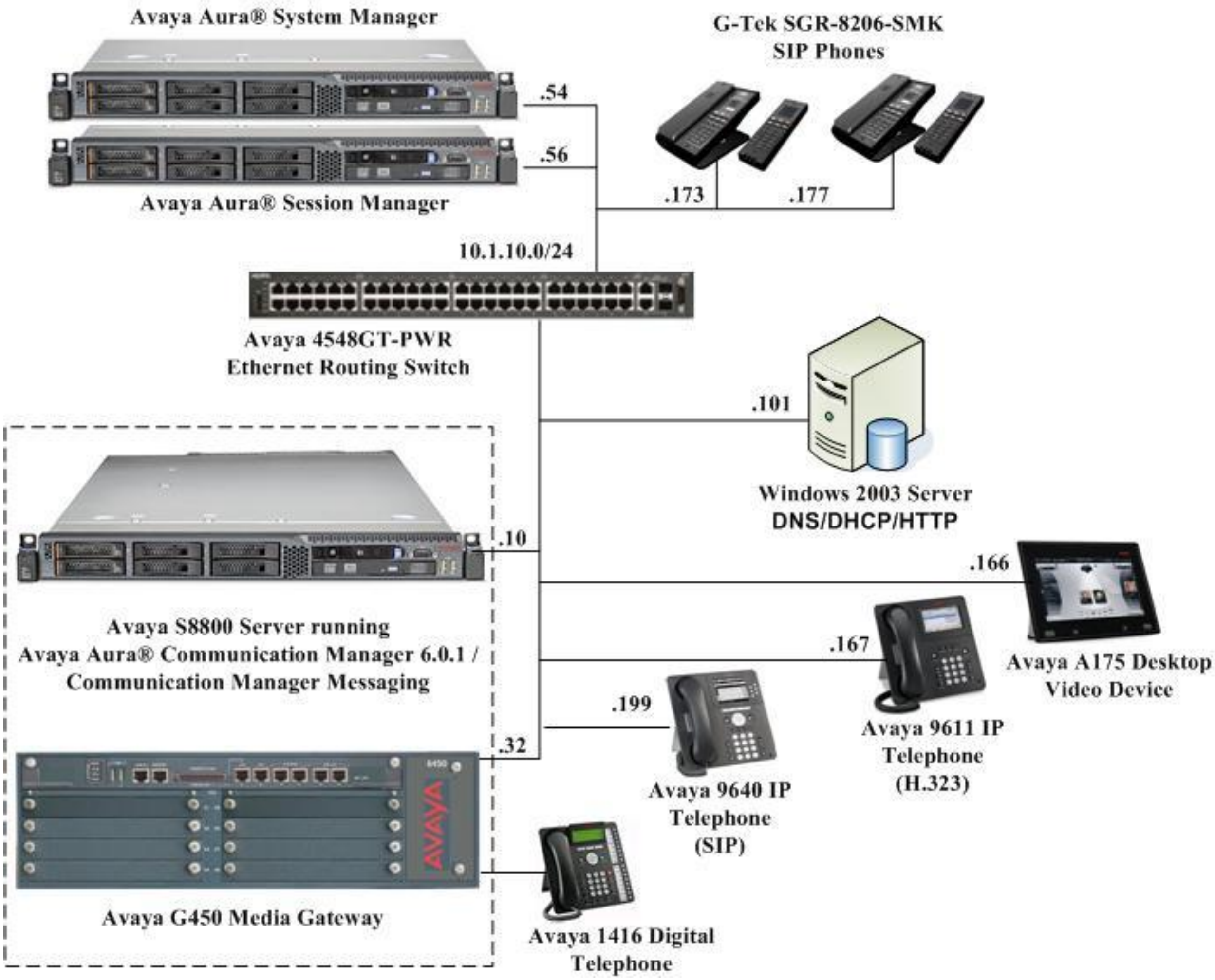


Figure 1: G-Tek SGR-8x06-SMK SIP Phones with Avaya SIP Solution

Table 1 lists the extensions used for this testing.

Extension	Note
10004	Avaya 9611 IP Telephone (H.323)
10051	Avaya 9640 IP Telephone (SIP)
481122	Avaya 1416 Digital Telephone
10062	Avaya Desktop Video Device (ADVD) (SIP)
10073 to 10074	G-Tek SGR-8x06-SMK 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 and Avaya Aura® Communication Messaging Manager on Avaya S8800 Server	R6.0.1 SP 8
Avaya G450 Media Gateway	31.20.0
Avaya Aura® Session Manager on Avaya S8800 Server	R6.1 SP 5
Avaya Aura® System Manager on Avaya S8800 Server	R6.1 SP 7
Avaya 9640IP Telephone (SIP)	2.6 SP5
Avaya 9611 IP Telephone (H.323)	6.0 SP5
Avaya 1408 Digital Telephone	-
Avaya A175 Desktop (SIP)	1.0.3
Avaya 4548GT-PWR Ethernet Routing Switch	V5.4.0.008
G-Tek SGR-8x06-SMK SIP Phones	Firmware Version: 2190X.16.1.03C Codec Version: Fri Mar 26 10:47:24 2004

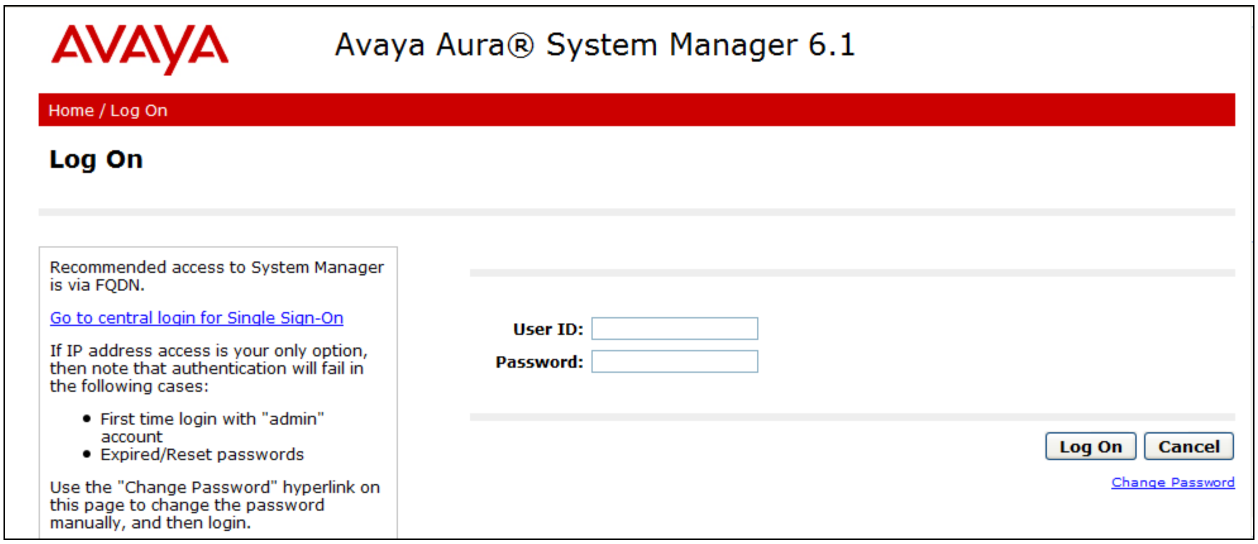
5. Configure Avaya Aura® Session Manager

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

- Launch System Manager
- Administer users

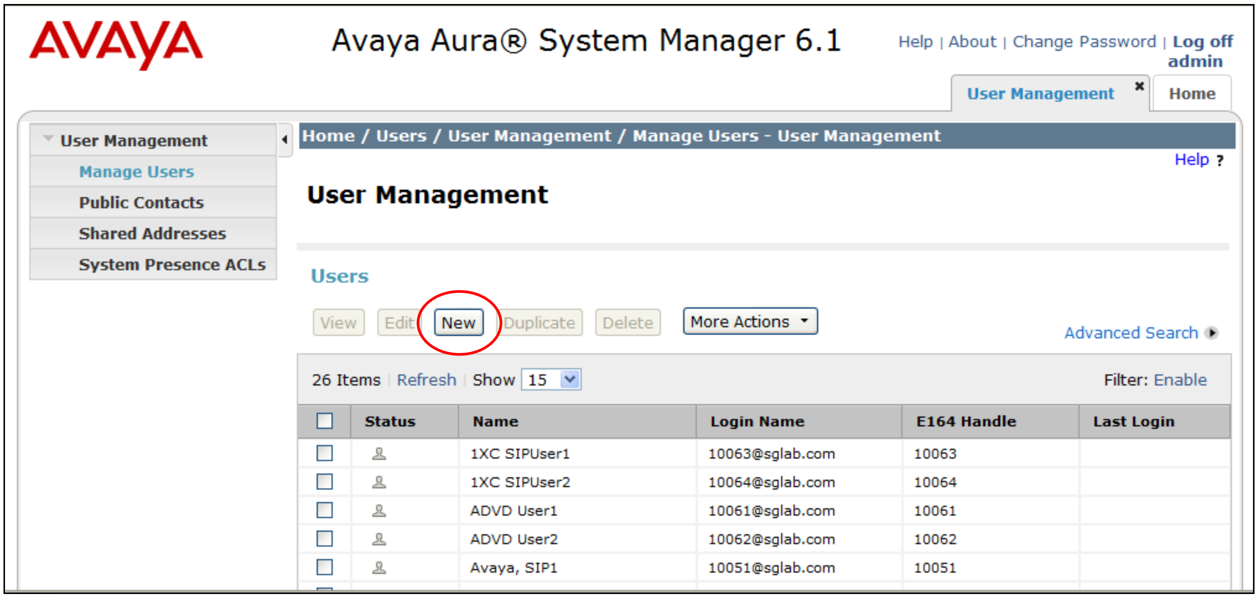
5.1. Launch System Manager

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.



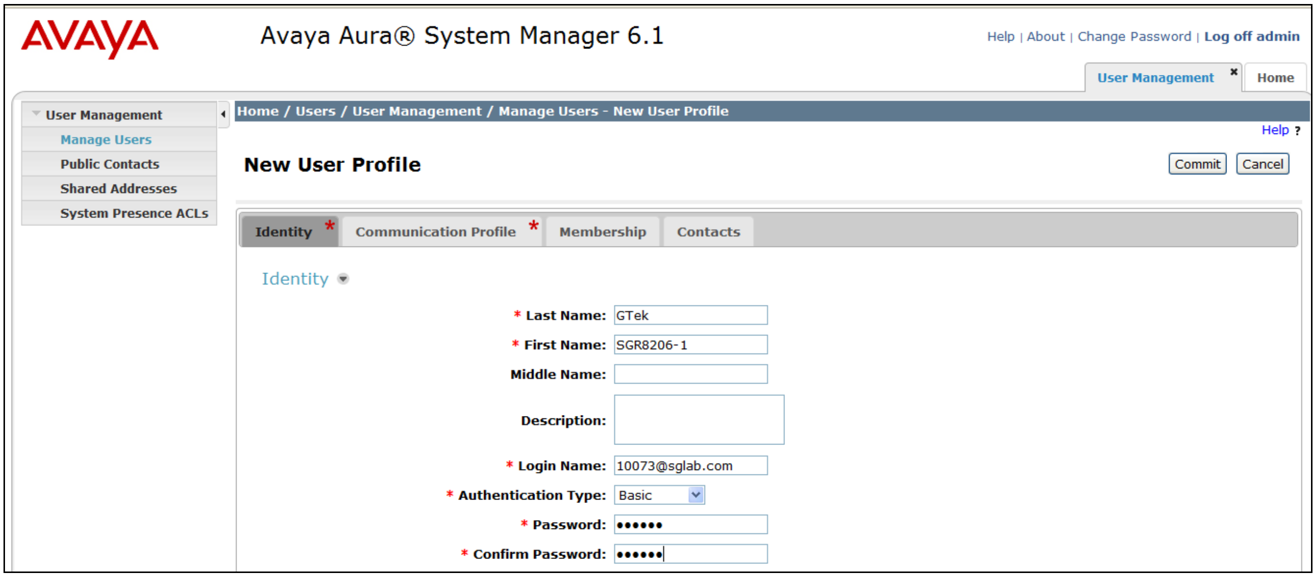
5.2. Administer Users

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.



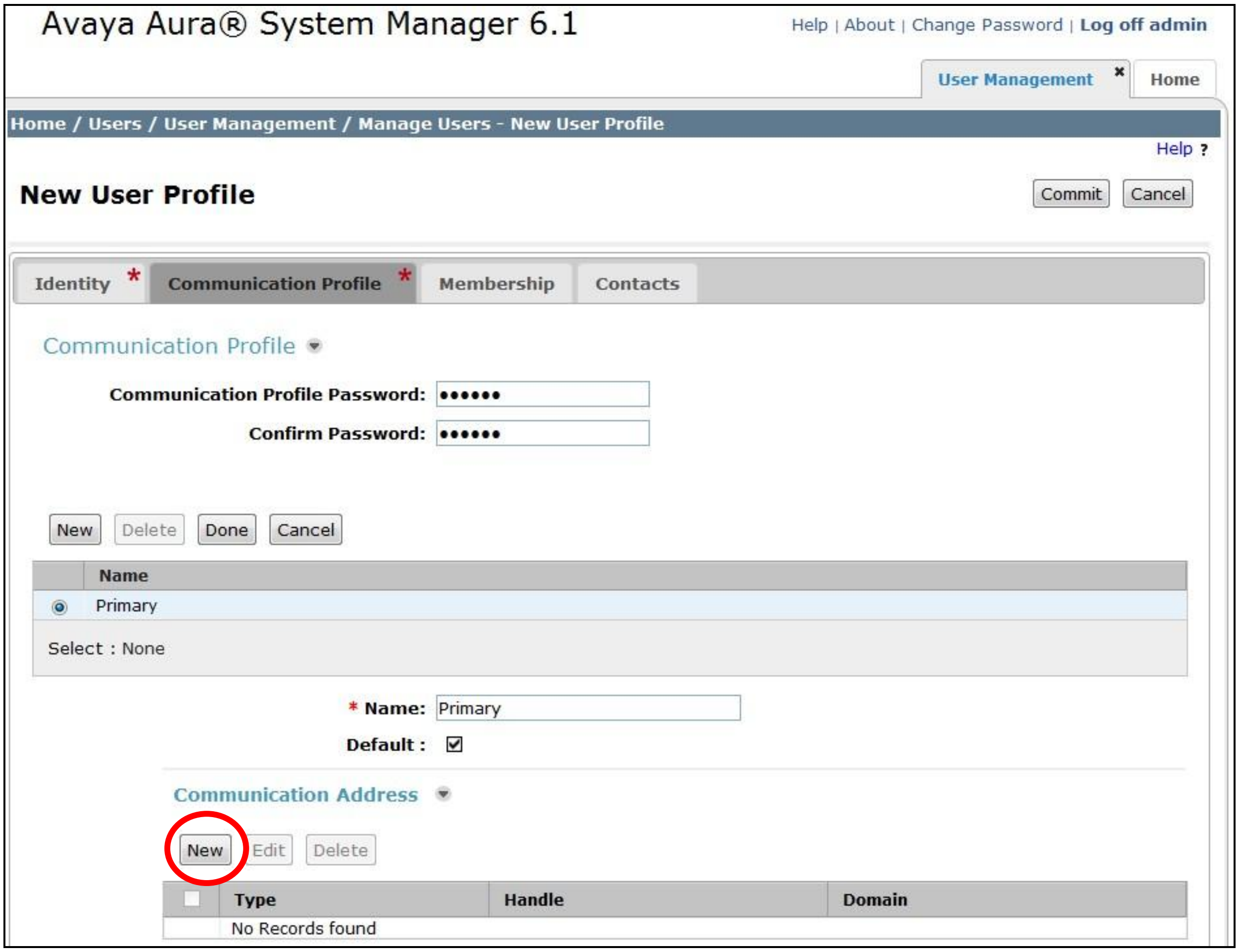
5.2.1. Identity

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 SGR-8x06 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. Retain the default values in the remaining fields.

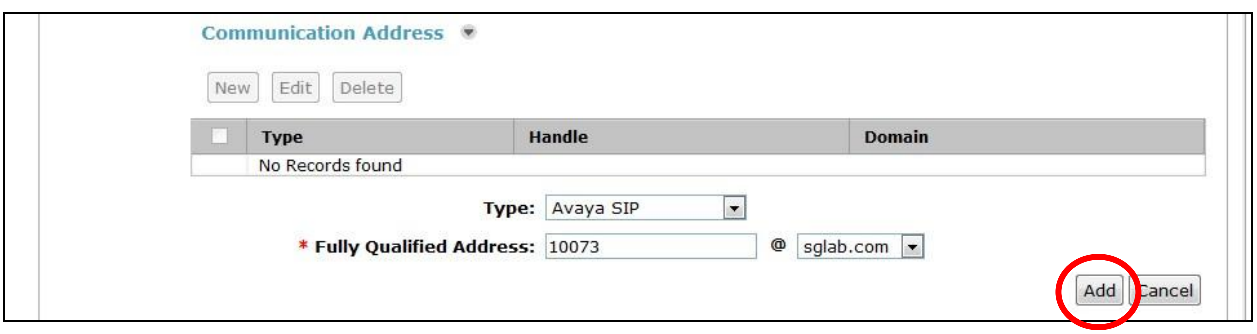


5.2.2. Communication Profile

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.

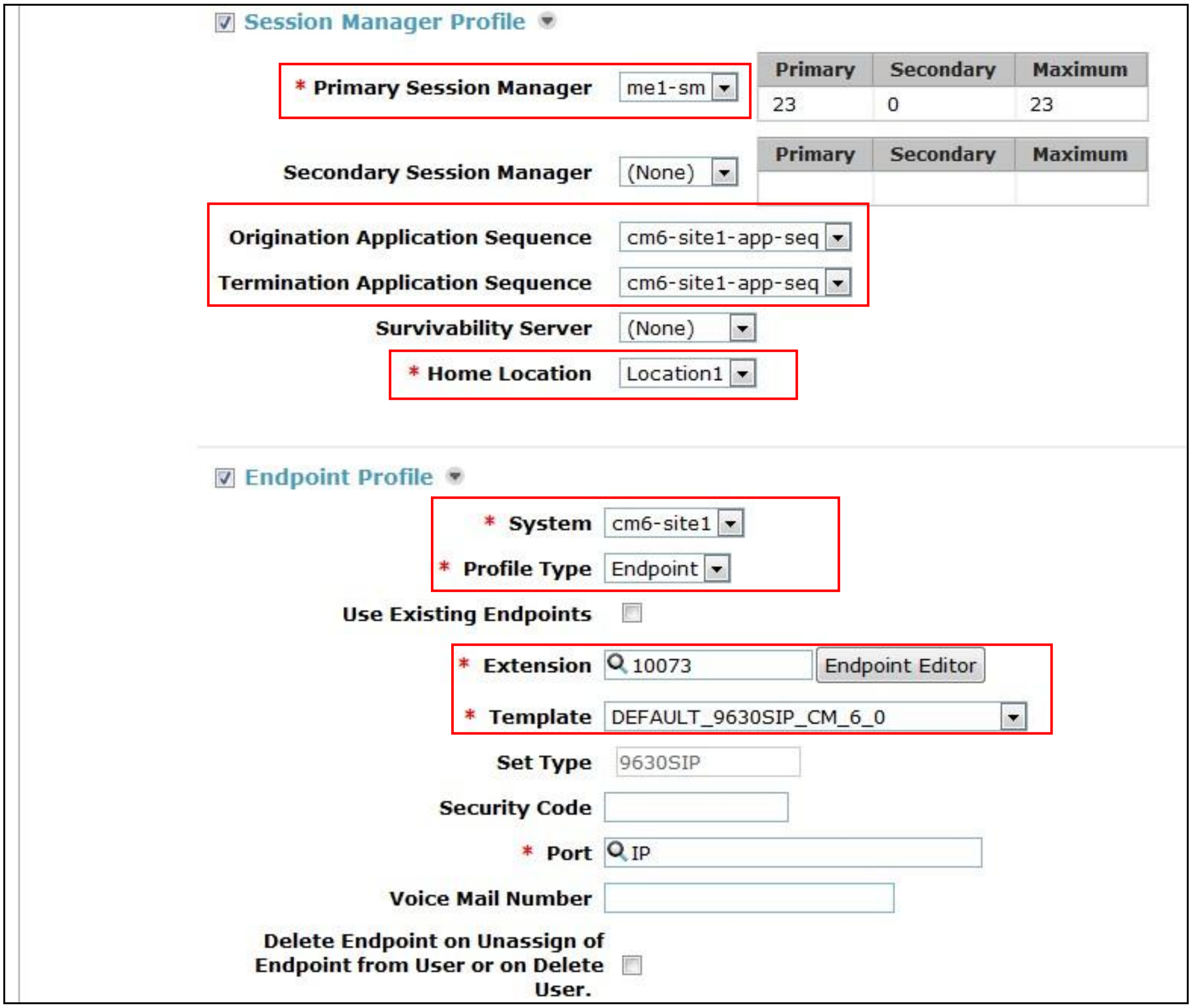


For **Type**, retain "Avaya SIP". For **Fully Qualified Address**, enter and select the SIP user extension and domain name from **Section 5.2.1**. Click **Add**.

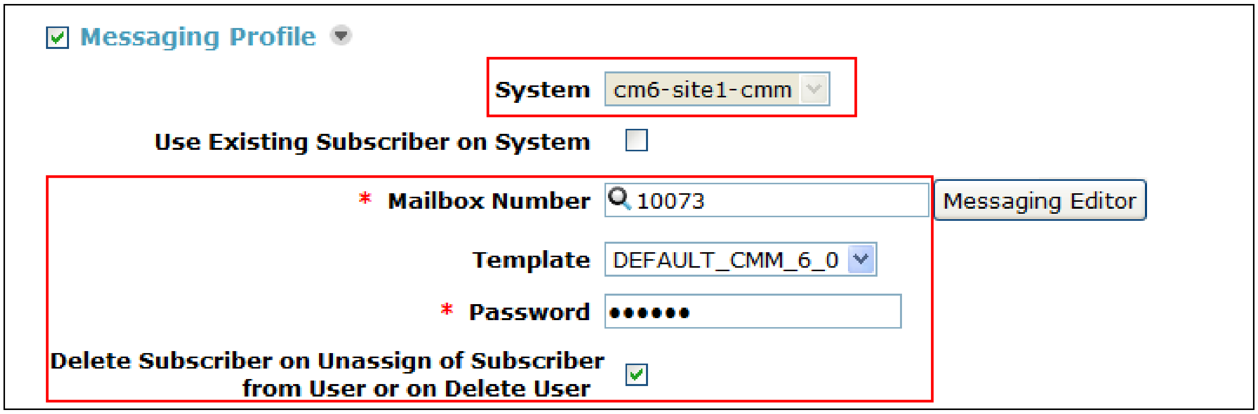


Scroll down to check and expand **Session Manager Profile**. For **Primary Session Manager**, **Origination Application Sequence**, **Termination Application Sequence**, and **Home Location**, select the values corresponding to the applicable Session Manager. Retain the default values in the remaining fields. These settings are configured during the initial setup of Session Manager.

Scroll down to check and expand **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 "**DEFAULT_9630SIP_CM_6_0**". For **Port**, select "**IP**". Retain the default values in the remaining fields.



Scroll down to check and expand on **Messaging Profile**. For **System**, select the value corresponding to the applicable Communication Messaging Manager. For **Mailbox Number**, select the same user extension number. For Template, select "**DEFAULT_CMM_6_0**". For **Password**, enter the appropriate mailbox password. Click "**Delete Subscriber on Unassign of Subscriber from User or on Delete User**" to select it.



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

Repeat **Section 5.2** to add a user for each G-Tek SGR-8X06 user. In the compliance testing, two users with extensions "10073" and "10074" 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 SGR-8x06 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 317
                                      Maximum Stations: 1000  215
                              Maximum XMOBILE Stations: 41000 0
                    Maximum Off-PBX Telephones - EC500: 1000  1
                    Maximum Off-PBX Telephones -   OPS: 1000  35
                    Maximum Off-PBX Telephones - PBFMC: 1000  0
                    Maximum Off-PBX Telephones - PVFMC: 1000  0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
                         Maximum Survivable Processors: 10    1




        (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, G.729 and G.729B.

```
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: G.729             n         2        20
 4: G.729B            n         2        20
 5:
 6:
 7:
```

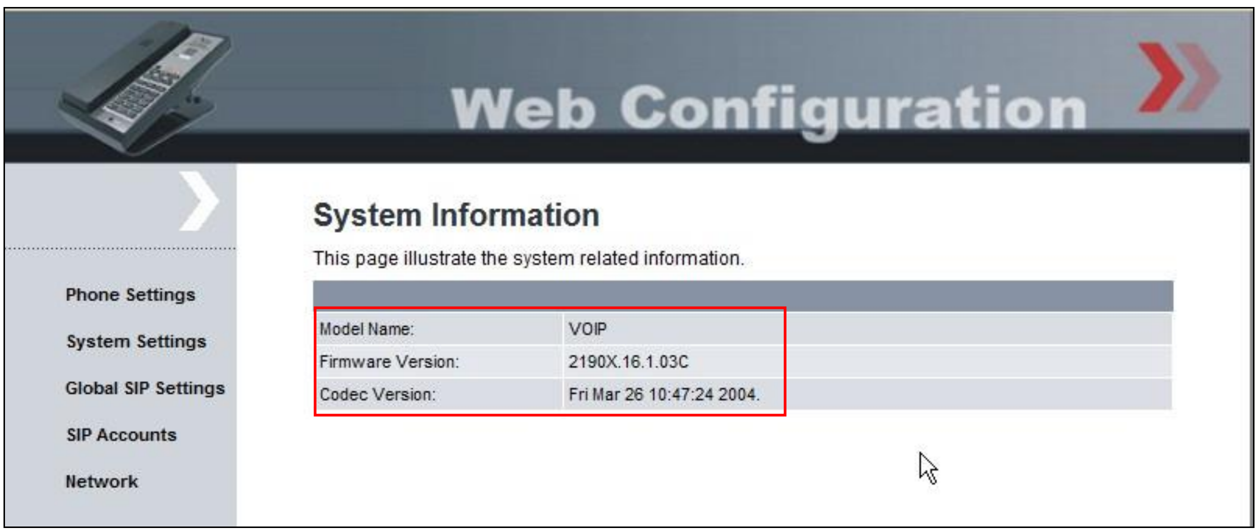
7. Configure G-Tek SGR-8x06 SIP Phones

This section provides the procedures for configuring G-Tek SGR-8x06 SIP Phones. The procedures include the following areas:

- Access Web Interface
- Configure SIP Account and DTMF Settings
- Configure Audio Codecs
- Configure Tone Settings
- Reboot after configuration

7.1. Access Web Interface

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

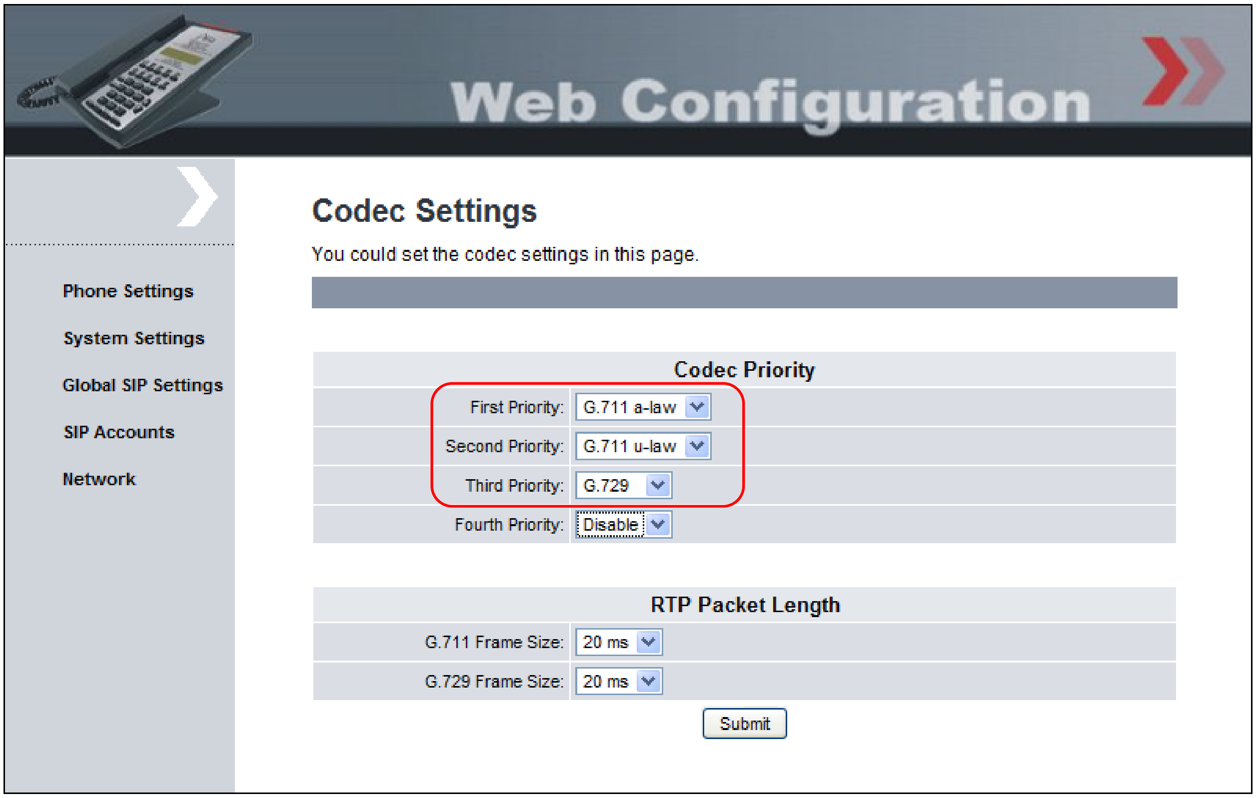


7.2. Configure SIP Account and DTMF Settings

Select **SIP Accounts** from the left menu. Set the **Active** field 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 and handset. Check the default **DTMF Type** settings is **RFC2833**. Click **Submit** to continue.

7.3. Configure Audio Codecs

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



7.4. Configure Tone Settings

Select **Phone Settings → Tone Settings** from the left menu. Select "**musiconhold.snd"** for **Music On Hold** in order to provide music while the call is on hold. Set **Call Waiting Tone** to **Enable**. Click **Submit** to continue.

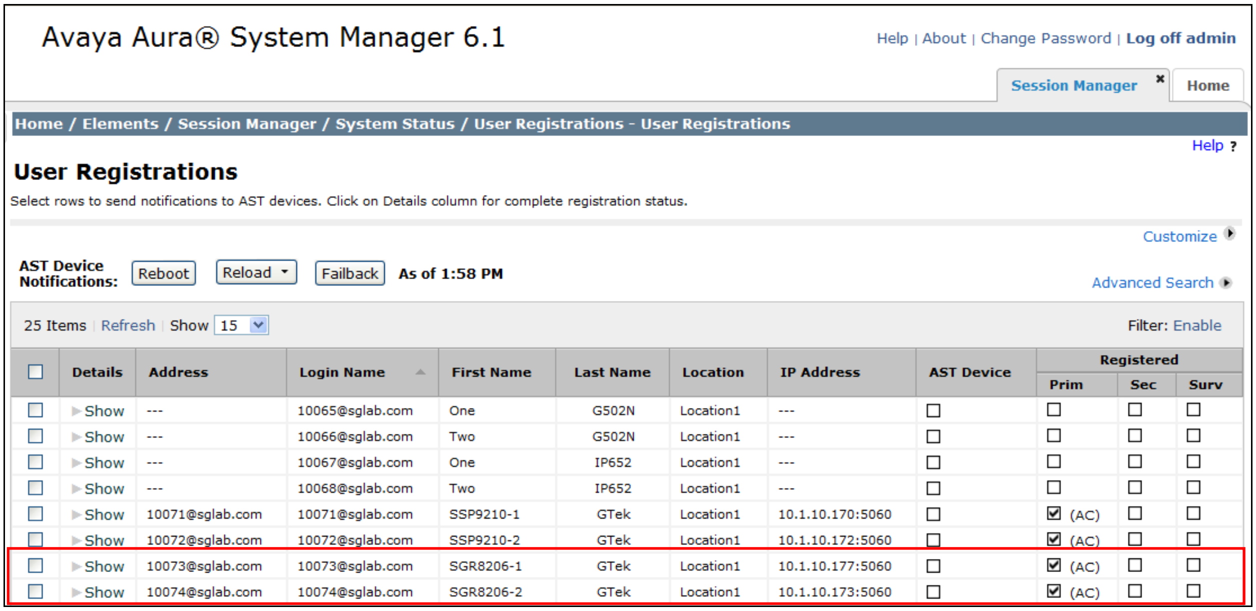
7.5. Reboot after configuration

Select **System Settings → Reboot** from the left menu to reboot the phone after setting the various parameters for the phones.

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Session Manager and G-Tek SGR-8x06 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.



	Details	Address	Login Name	First Name	Last Name	Location	IP Address	AST Device	Registered Prim	Registered Sec	Registered Surv
☐	Show	---	10065@sglab.com	One	G502N	Location1	---	☐	☐	☐	☐
☐	Show	---	10066@sglab.com	Two	G502N	Location1	---	☐	☐	☐	☐
☐	Show	---	10067@sglab.com	One	IP652	Location1	---	☐	☐	☐	☐
☐	Show	---	10068@sglab.com	Two	IP652	Location1	---	☐	☐	☐	☐
☐	Show	10071@sglab.com	10071@sglab.com	SSP9210-1	GTek	Location1	10.1.10.170:5060	☐	☑ (AC)	☐	☐
☐	Show	10072@sglab.com	10072@sglab.com	SSP9210-2	GTek	Location1	10.1.10.172:5060	☐	☑ (AC)	☐	☐
☐	Show	10073@sglab.com	10073@sglab.com	SGR8206-1	GTek	Location1	10.1.10.177:5060	☐	☑ (AC)	☐	☐
☐	Show	10074@sglab.com	10074@sglab.com	SGR8206-2	GTek	Location1	10.1.10.173:5060	☐	☑ (AC)	☐	☐

From the web interface of the G-Tek SGR-8x06 phone, click **SIP Accounts** from the left menu. Verify that the **Status** field shows as r**egistered.**

9. Conclusion

These Application Notes describe the configuration steps required for G-Tek Electronics SGR-8x06-SMK SIP Phones to successfully interoperate with Avaya Aura® Session Manager 6.1 and Avaya Aura® Communication Manager 6.0.1. 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*, Release 6.0, Doc ID 03-300509, June 2010.
[2] *Administering Avaya Aura® Session Manager*, Release 6.1, Doc ID 03-603324, Issue 1, November 2010.
[3] *Implementing Avaya Aura®* Communication *Manager Messaging,* Release 6.0, Doc ID 18-603644, June 2010
[4] *Administrator Guide for SSP9210-SM/SAX-8210P/SGR8210-SMK*

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.