



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring G-Tek/AEi Communications SKD-1203 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 SKD-1203 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 SKD-1203 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 SKD-1203 is an IP phones for the hospitality industry.

2. General Test Approach and Test Results

To verify interoperability of G-Tek/AEi SKD-1203 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. Calls were also made to and from the PSTN and various 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/AEi SKD-1203 SIP Phones with Session Manager.
- Calls between G-Tek/AEi telephones and Avaya SIP, H.323, analog, and digital telephones.
- 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: +1-650-552-9416
- 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 SKD-1203 SIP Phones, registered with Session Manager, were also used during the compliance test.

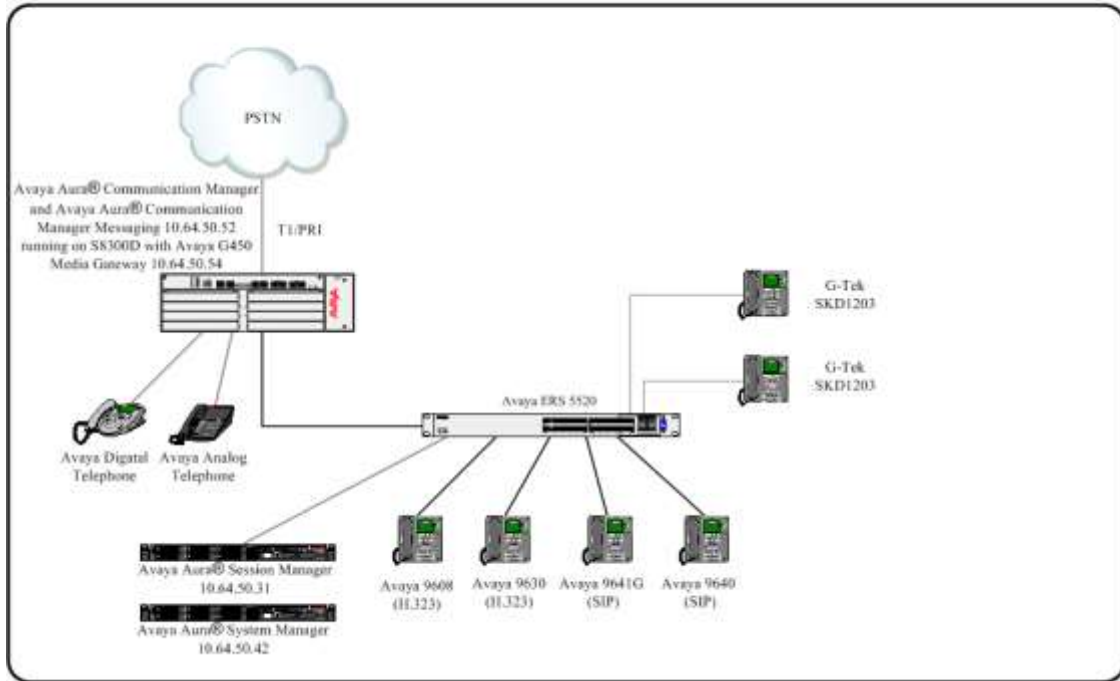


Figure 1: G-Tek SKD-1203 SIP Phone Reference Configuration

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 Products</i>	
SKD-1203	1811X.16.3.0F5

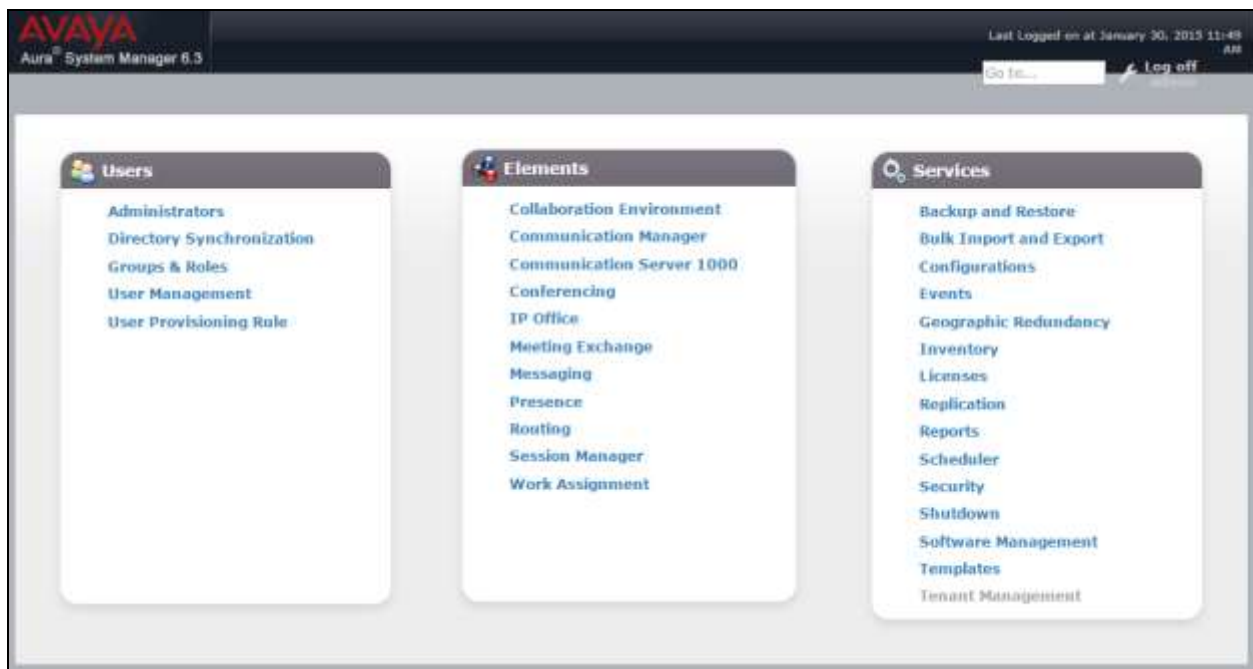
5. Configure Avaya Aura® Session Manager

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 shows the Avaya System Manager 6.3 User Management interface. The sidebar on the left contains the following menu items: Home, User Management, Manage Users, Public Contacts, Shared Addresses, System Presence, ACLs, Communication Profile Password, and Policy. The main content area is titled 'User Management' and includes a search bar and a table of users. The table has the following columns: Last Name, First Name, Display Name, Login Name, SIP Handle, and Last Login. The table contains 33 items, with 15 shown on this page. The users listed include Station, Analog, Analog Line 1, and Avaya agents.

Last Name	First Name	Display Name	Login Name	SIP Handle	Last Login
36101	Station	36101, Station	36101@4427.com	36101	
36102	Station	36102, Station	36102@4427.com	36102	
36103	Station	36103, Station	36103@4427.com	36103	
36104	Station	36104, Station	36104@4427.com	36104	
36105	Station	36105, Station	36105@4427.com	36105	
36106	Station	36106, Station	36106@4427.com	36106	
36107	Station	36107, Station	36107@4427.com	36107	
36200	Analog	36200, Analog	36200@4427.com	36200	
36201	Analog	36201, Analog	36201@4427.com	36201	
36301	Analog Line 1	36301, Analog Line 1	36301@4427.com	36301	
Avaya	ana-X Agent 1	Avaya, ana-X Agent 1	45001@br113.com	+9099445001	
Avaya	ana-X Agent 2	Avaya, ana-X Agent 2	45002@br113.com	+9099445002	
Avtec	Avtec Line1	Avtec Line1	61009@4427.com	61009	
Avtec	Avtec Line2	Avtec Line2	61010@4427.com	61010	
Avtec	Avtec Line3	Avtec Line3	61011@4427.com	61011	

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 SKD-1203 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 top navigation bar includes the Avaya logo, 'Aura System Manager 6.3', and a user session indicator 'Last logged on at January 30, 2015 11:48 AM' with a 'Log off admin' link. The left sidebar contains a menu with 'User Management' selected, and sub-items like 'Manage Users', 'Public Contacts', 'Shared Addresses', 'System Presence', 'ACLs', 'Communication', 'Profile Password', and 'Policy'. The main content area is titled 'New User Profile' and has a breadcrumb trail 'Home / Users / User Management / Manage Users'. There are three tabs: 'Identity' (selected), 'Communication Profile', and 'Membership'. Below the tabs is a 'User Provisioning Rule' section with a dropdown menu. The 'Identity' section contains several input 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' (61020@d4f27.com), and 'Authentication Type' (Basic). At the top right of the form area are buttons for 'Commit & Continue', 'Commit', and 'Cancel'.

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 SKD-1203.

New User Profile

Communication Profile

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

Name: [Primary] (Name: Primary, Default: [checkbox])

Communication Address

Type	Handle	Device
No records found		

Type: Avaya SIP
Fully Qualified Address: 61020 @ 6407.com

Session Manager Profile

SIP Registration

Primary Session Manager: [w5031] (Primary Secondary Provision: [table])
Secondary Session Manager: [Name]
Survivability Server: [Name]
Max. Simultaneous Devices: [1]
Block New Registration When Maximum Registrations Active? [checkbox]

Application Sequences

Origination Sequence: [w5052]
Termination Sequence: [w5052]

Call Routing Settings

Home Location: [6407_13]
Conference Factory Set: [Name]

Call History Settings

Enable Centralized Call History? [checkbox]

Collaboration Environment Profile

CM Endpoint Profile

System: [cm3052]
Profile Type: Endpoint
Use Existing Endpoints: [checkbox]
Extension: [61020] (Endpoint Editor)
Template: [9688SP_DEFAULT_CM_6_3]
Set Type: [9688SP]
Security Code: [field]
Port: [JP]
Voice Mail Number: [field]
Preferred Handle: [Name]

Enhanced call info display for 3-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 SKD-1203 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:
```

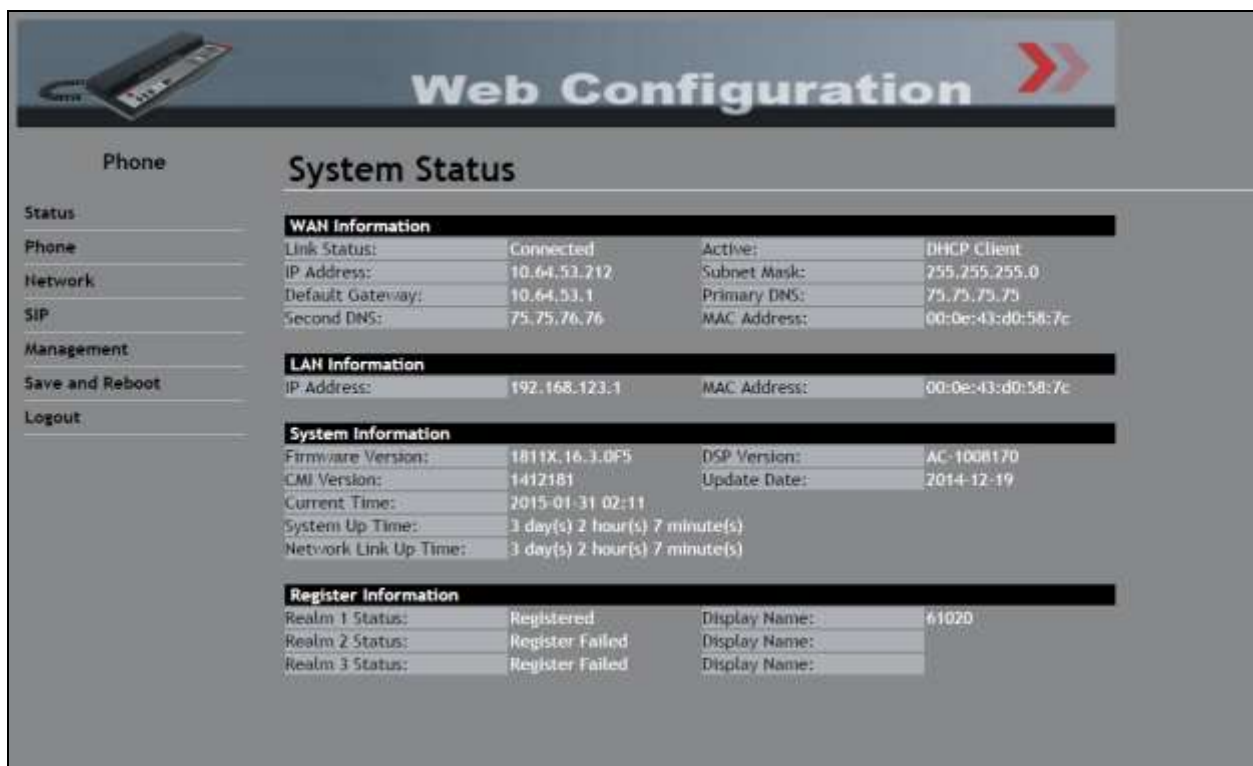
7. Configure G-Tek/AEi SKD-1203SIP Phone

This section provides the procedures for configuring G-Tek/AEi SKD-1203 SIP Phone. 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/AEi phone, into the address bar of web browser and log in using a valid account. The **System Status** screen is displayed.



The screenshot displays the 'Web Configuration' interface for a G-Tek/AEi SKD-1203 SIP Phone. The main heading is 'Web Configuration' with a red double arrow icon. Below this, there is a 'Phone' section with a navigation menu on the left containing: Status, Phone, Network, SIP, Management, Save and Reboot, and Logout. The main content area is titled 'System Status' and is divided into several sections:

- WAN Information:**

Link Status:	Connected	Active:	DHCP Client
IP Address:	10.64.53.212	Subnet Mask:	255.255.255.0
Default Gateway:	10.64.53.1	Primary DNS:	75.75.75.75
Second DNS:	75.75.76.76	MAC Address:	00:0e:43:d0:58:7c
- LAN Information:**

IP Address:	192.168.123.1	MAC Address:	00:0e:43:d0:58:7c
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- System Information:**

Firmware Version:	1811X.16.3.0P5	DSP Version:	AC-1008170
CMU Version:	1412181	Update Date:	2014-12-19
Current Time:	2015-01-31 02:11		
System Up Time:	3 day(s) 2 hour(s) 7 minute(s)		
Network Link Up Time:	3 day(s) 2 hour(s) 7 minute(s)		
- Register Information:**

Realm 1 Status:	Registered	Display Name:	61020
Realm 2 Status:	Register Failed	Display Name:	
Realm 3 Status:	Register Failed	Display Name:	

7.2. Configure SIP Account and DTMF Settings

Select **SIP → Service** from the left menu. Set the **Realm Active:** field to **Enable**. Enter username configured on Session Manager for **Display Name:**, **Phone Number:**, and **Authentication ID:**. Provide user password for **Authentication Password:**. In the **Domain Server** and **Proxy Server** fields enter the IP Address of Session Manager's signaling interface. Set **Subscribe for MWI:** to **Enable**.



The screenshot shows the 'Web Configuration' interface for a phone. The main heading is 'Service Domain Setting'. On the left, there is a navigation menu with the following items: Status, Phone, Network, SIP, Service, Codecs, Advanced, Management, Save and Reboot, and Logout. The 'Service' menu item is selected. The main content area contains the following configuration fields:

Realm:	1 ▼
Realm Active:	Enable ▼
Display Name:	61020
Phone Number:	61020
Authentication ID:	61020
Authentication Password:	*****
Domain Server:	10.64.50.31
Proxy Server:	10.64.50.31
Outbound Proxy:	
Subscribe for MWI :	Enable ▼

At the bottom of the configuration area, there are two buttons: 'Submit' and 'Reset'.

7.3. Configure Audio Codecs

Select **SIP** → **Codecs** from the left menu. In the **Codec Settings** section, verify that the required codes are moved to the Enable Codes box and are prioritized accordingly.

Click **Submit** to continue.

Web Configuration

Phone

Codecs Setting

Disable Codes

Enable Codes

G.711 u-law
G.711 a-law
G.723
G.729

Move Up Down

G.711 and G.729: 20 ms
G.723: 30 ms
G.723 5.3K: Disable
Silence Suppression (VAD): Disable
Echo Canceller: Enable

Codec Type	ID Value
RFC 2833:	Default 101 (95-127)

Submit Reset

7.4. Reboot after configuration

Select **Save and Reboot** from the left menu to reboot the phone after setting the various parameters.

Click the **Save** button.



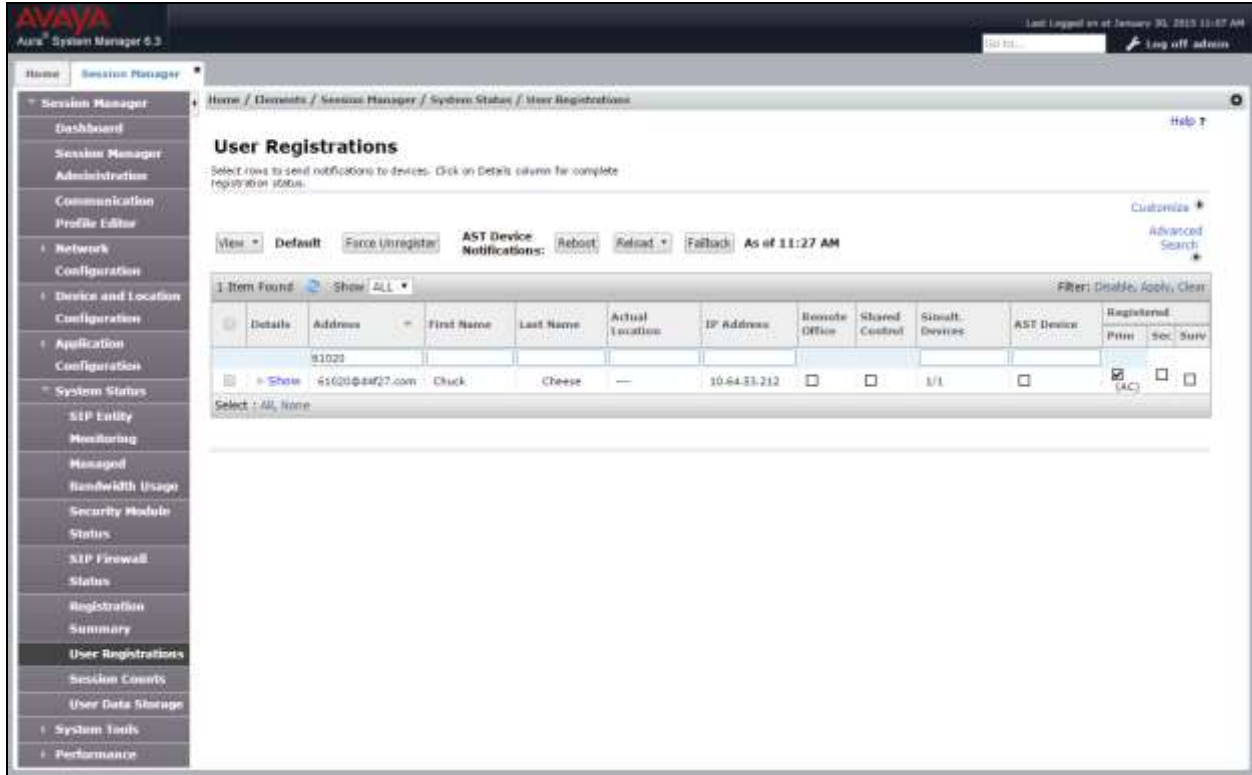
The system will reboot automatically after updates are complete.



8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Session Manager and G-Tek/AEi SKD-1203 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 web interface. The left-hand navigation menu is expanded to show the 'System Status' section, with 'User Registrations' selected. The main content area shows the 'User Registrations' page with 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 10.64.53.212, with a checkmark in the 'Registered Prim' column and '(SAC)' in parentheses next to it.

Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered
										Prim Sec Surv
Show	516214@2427.com	Chuck	Cheese	---	10.64.53.212	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/> (SAC) <input type="checkbox"/> <input type="checkbox"/>

From the web interface of the G-Tek/AEi SKD-1203 phone, click **Status** from the left menu. Verify that the **Realm 1 Status:** field shows **Registered**.

The screenshot displays the 'Web Configuration' interface for a G-Tek/AEi SKD-1203 phone. The main heading is 'System Status'. On the left, there is a navigation menu with options: Phone, Status, Phone, Network, SIP, Management, Save and Reboot, and Logout. The main content area is divided into several sections:

- WAN Information:**

Link Status:	Connected	Active:	DHCP Client
IP Address:	10.64.53.212	Subnet Mask:	255.255.255.0
Default Gateway:	10.64.53.1	Primary DNS:	75.75.75.75
Second DNS:	75.75.76.76	MAC Address:	00:0e:43:d0:58:7c
- LAN Information:**

IP Address:	192.168.123.1	MAC Address:	00:0e:43:d0:58:7c
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- System Information:**

Firmware Version:	1811X.16.3.0F5	DSP Version:	AC-1008170
CMI Version:	1412181	Update Date:	2014-12-19
Current Time:	2015-01-31 02:38		
System Up Time:	0 day(s) 0 hour(s) 6 minute(s)		
Network Link Up Time:	0 day(s) 0 hour(s) 6 minute(s)		
- Register Information:**

Realm 1 Status:	Registered	Display Name:	61020
Realm 2 Status:	Register Failed	Display Name:	
Realm 3 Status:	Register Failed	Display Name:	

9. Conclusion

These Application Notes describe the configuration steps required for G-Tek/AEi Communications SKD-1203 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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