Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Escene Communication Technology ES620, ES410, ES320 and ES210 Enterprise IP Phones 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 Escene Communication Technology ES620, ES410, ES320 and ES210 Enterprise IP Phones 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 Escene Communication Technology ES620, ES410, ES320 and ES210 Enterprise IP Phones to interoperate with an Avaya SIP infrastructure consisting of Avaya Aura® Session Manager 6.1 and Avaya Aura® Communication Manager 6.0.1. Escene ES620, ES410, ES320 and ES210 belong to a series of business IP phones designed to replace traditional desktop office terminals in the enterprise.

2. General Test Approach and Test Results

To verify interoperability of Escene ES620, ES410, ES320 and ES210 IP Phones with Session Manager and Communication Manager, calls were made between Escene 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.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Successful registration of Escene ES620, ES410, ES320 and ES210 IP Phones with Session Manager.
- Calls between Escene telephones and Avaya SIP, H.323, and digital telephones.
- G.722, G.711 and G729A 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 using RFC2833, Message Waiting Indicator (MWI) subscription and feature access code dialing.
- PBX features including Multiple Call Appearances, Hold, Transfer, and Conference.
- Proper system recovery after an Escene 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 Escene Communication Technology can be obtained through the following:

- Phone: 400 090 5818 (China only), +86 20-28096798
- E-mail: support@escene.cn

3. Reference Configuration

The diagram illustrates an enterprise site with an Avaya SIP-based network, including S8800 Servers running Avaya Aura® System Manager and Session Manager, an S8800 Server running Communication Manager configured as an Evolution Server with a G450 Media Gateway, and Avaya SIP, H.323 and Digital endpoints. The enterprise site also contains four Escene SIP Phones used in the compliance testing. The Escene phones are registered with Session Manager and are configured as endpoint users on Session Manager and as stations on Communication Manager. The Windows 2003 Server provides DNS and DHCP service, and the downloading of phone firmware.

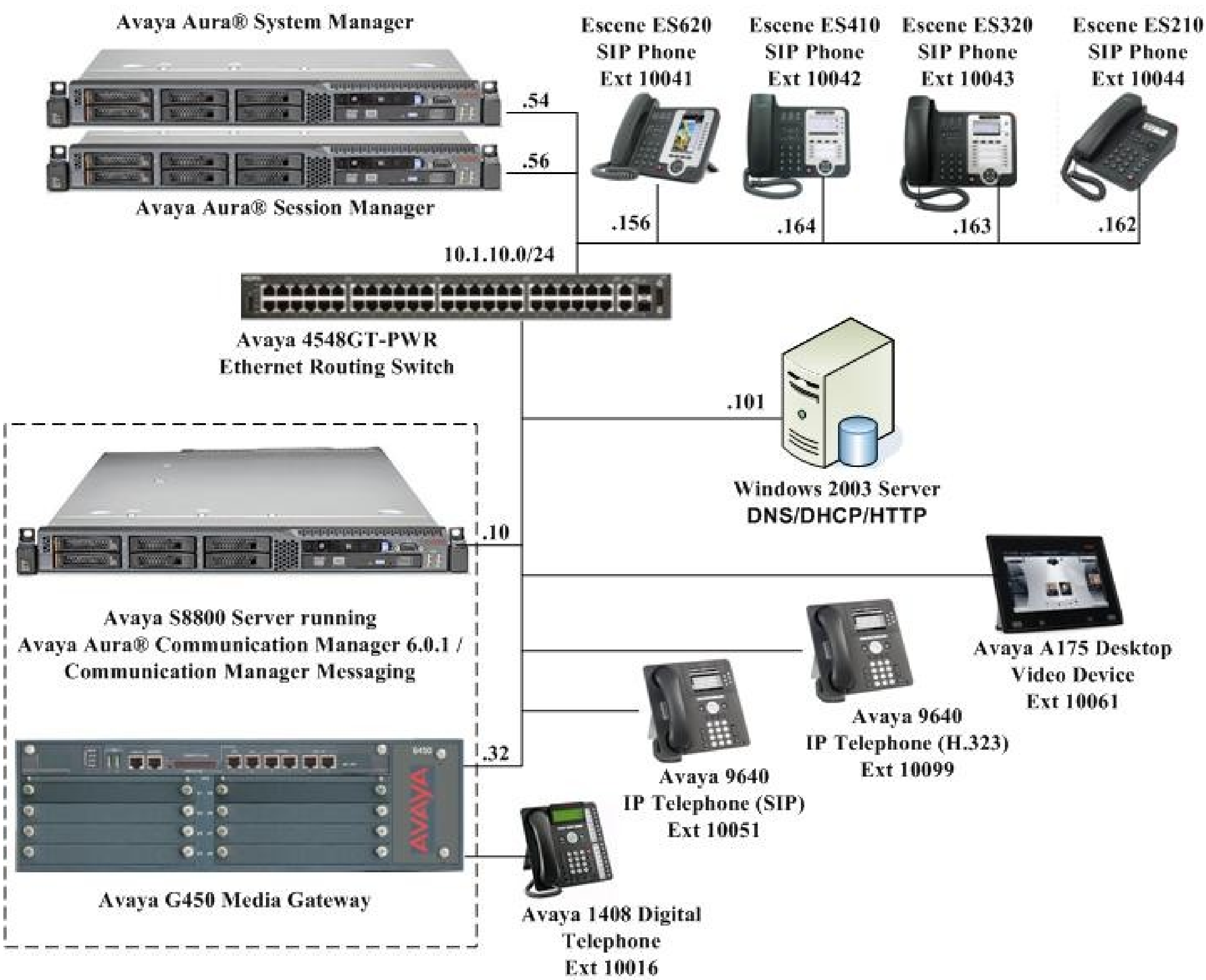


Figure 1: Escene SIP Phones with Avaya SIP Solution

Table 1 lists the extensions used for this testing.

Extension	Note
10099	Avaya 9640 IP Telephone (H.323)
10051	Avaya 9640 IP Telephone (SIP)
10016	Avaya 1408 Digital Telephone
10061	Avaya Desktop Video Device (ADVD) A175 (SIP)
10041 to 10044	Escene 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
Avaya S8800 Server	Avaya Aura® Communication Manager 6.0.1 (Service Pack 4 00.1.510.1-19100) / Avaya Aura® Communication Manager Messaging 6.0.1
Avaya G450 Media Gateway	31.20.0
Avaya S8800 Server	Avaya Aura® Session Manager 6.1 Service Pack 5
Avaya S8800 Server	Avaya Aura® System Manager 6.1 Service Pack 5
Avaya 9600 Series IP Telephones	2.6.4.0 (SIP) 3.1 SP2 (H.323)
Avaya 1408 Digital Telephone	-
Avaya Desktop Video Device	1.0.3
Avaya 4548GT-PWR Ethernet Routing Switch	V5.4.0.008
Escene ES620, ES410, ES320 and ES210 IP Phones	Software Version: V2.2.5.2-2503 Web Version: 2.9.3.2 Kernel Version: v2.2.9

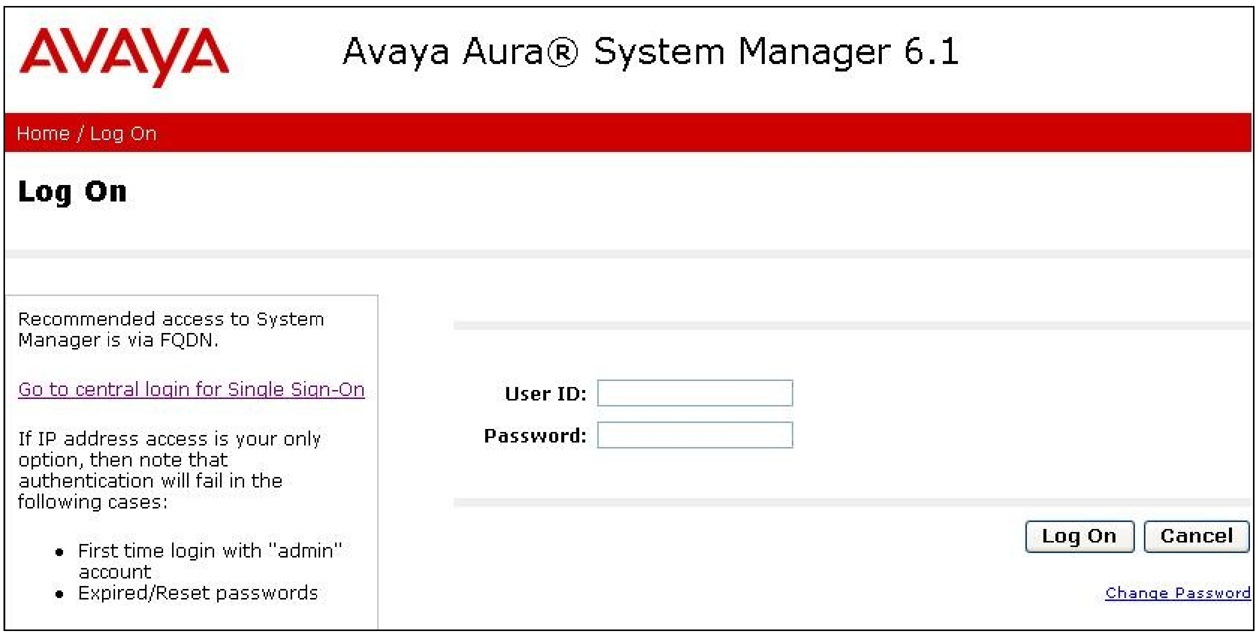
5. Configure Avaya Aura® Session Manager

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

- Launch System Manager
- Administer users

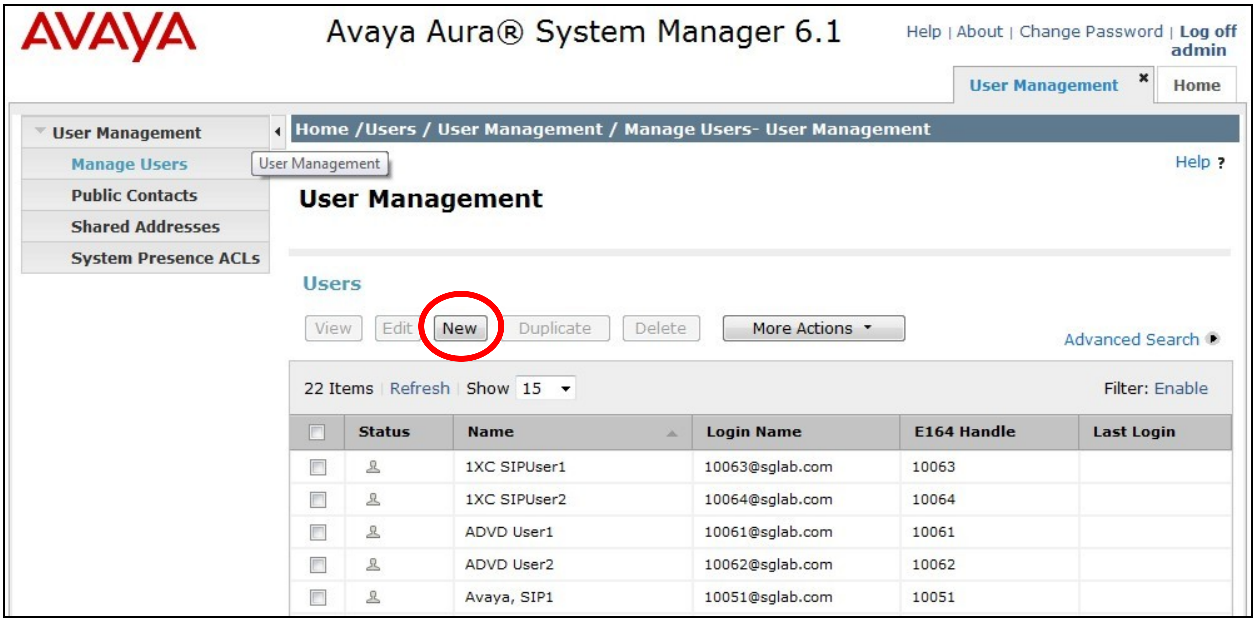
5.1. Launch System Manager

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.



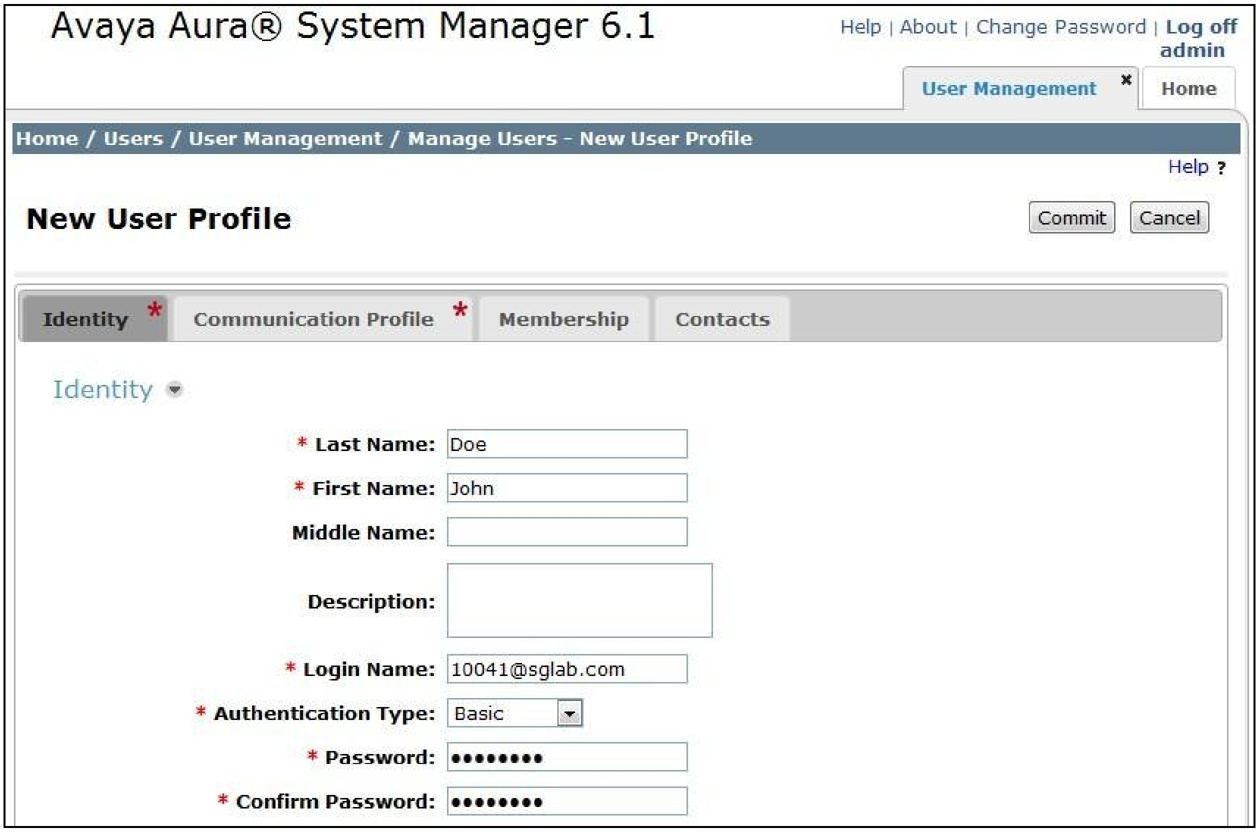
5.2. Administer Users

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.



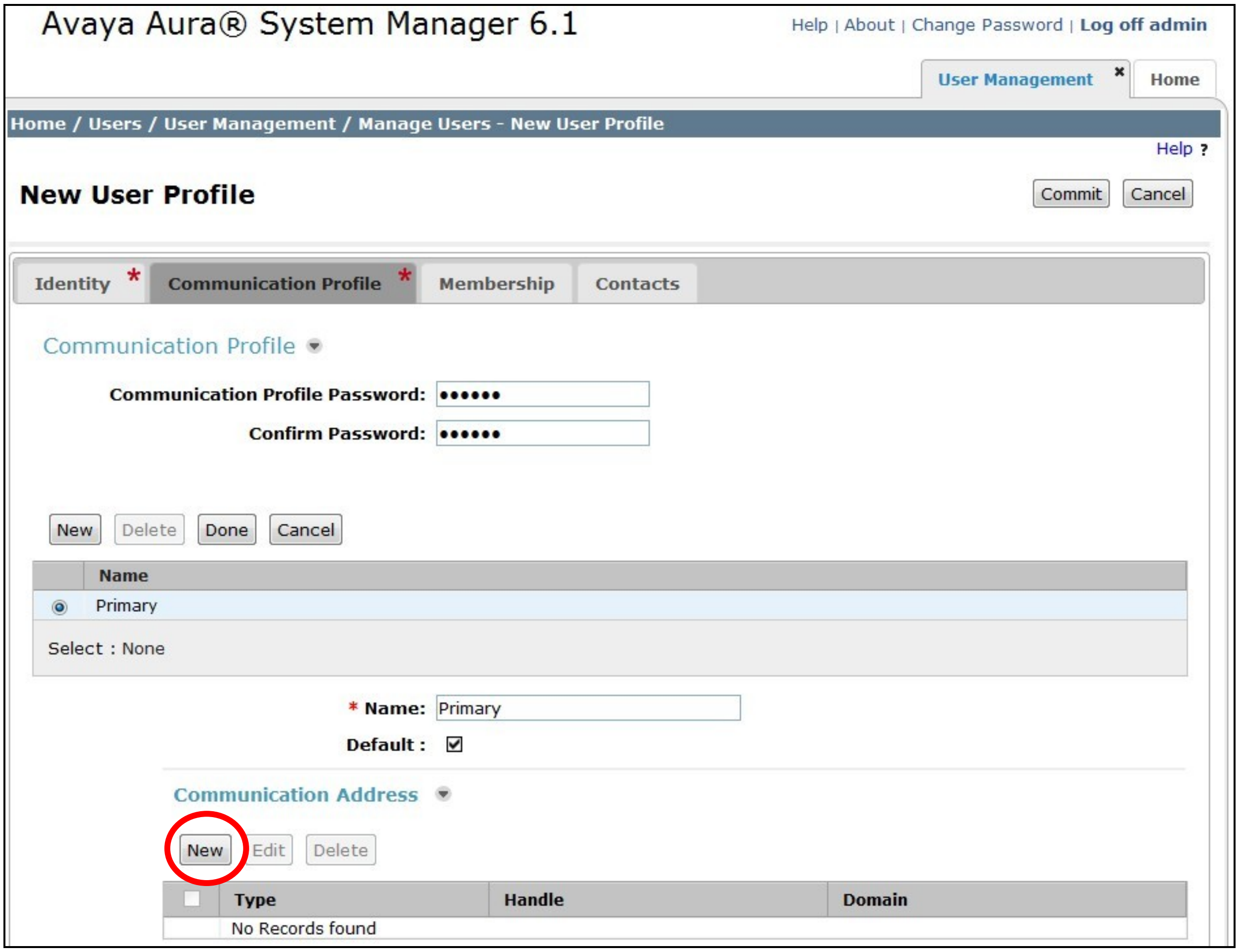
5.2.1. Identity

The **New User Profile** screen is displayed. Enter desired **Last Name** and **First Name**. For **Login Name**, enter **n@x**, where **n** is the first Escene user extension and **x** is the domain name. For the compliance testing, **sglab.com** was used. For **Password** and **Confirm Password**, enter the appropriate credentials for System Manager. Retain the default values in the remaining fields.

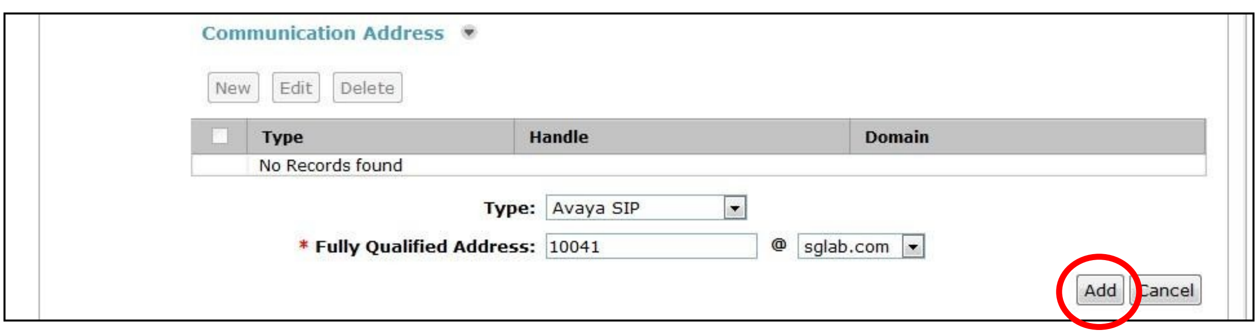


5.2.2. Communication Profile

Select the **Communication Profile** tab. For **Communication Profile Password** and **Confirm Password**, enter the desired password for the Escene phones to use for registration. Scroll down to the **Communication Address** sub-section, and click **New** to add a new address.

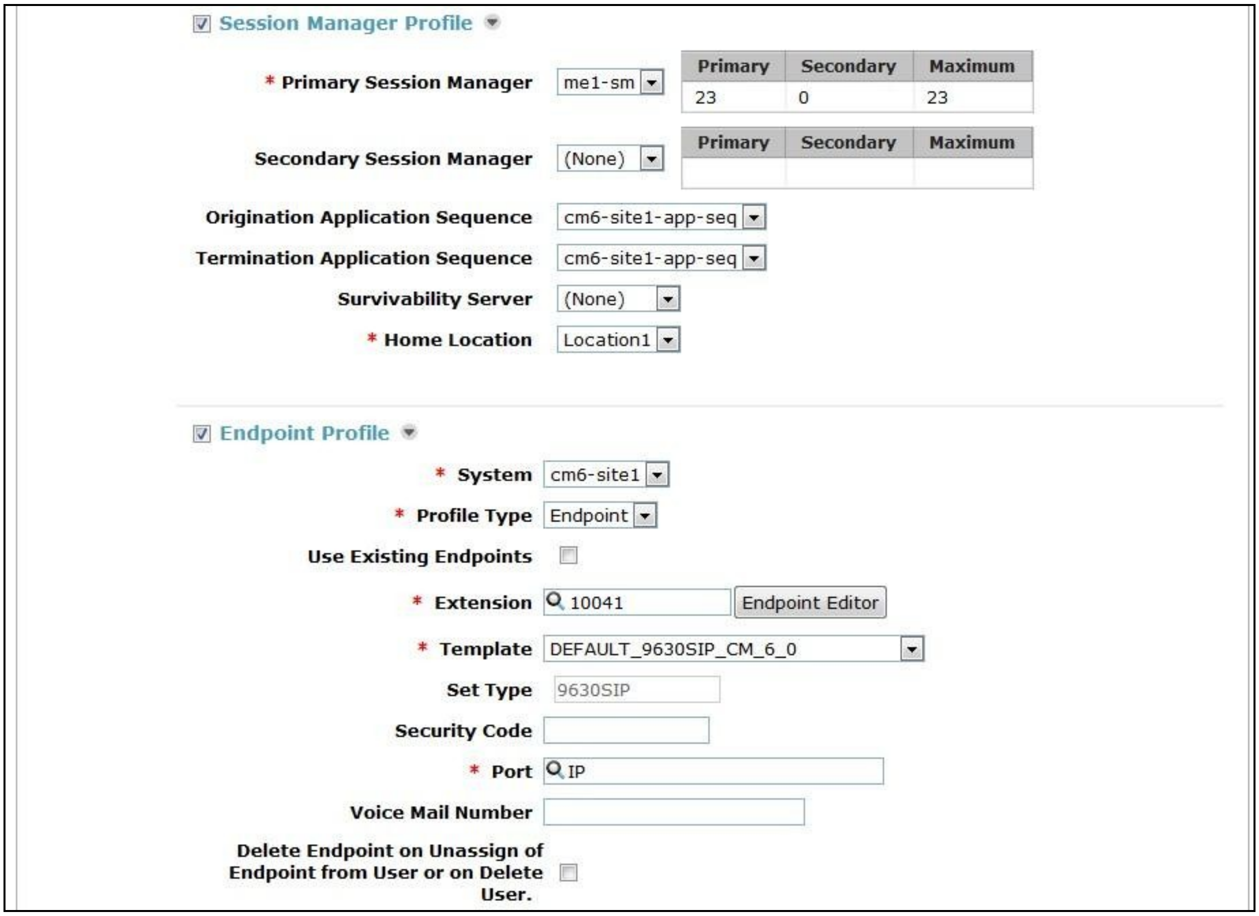


For **Type**, retain **Avaya SIP**. For **Fully Qualified Address**, enter and select the SIP user extension and domain name from **Section 5.2.1**. Click **Add**.

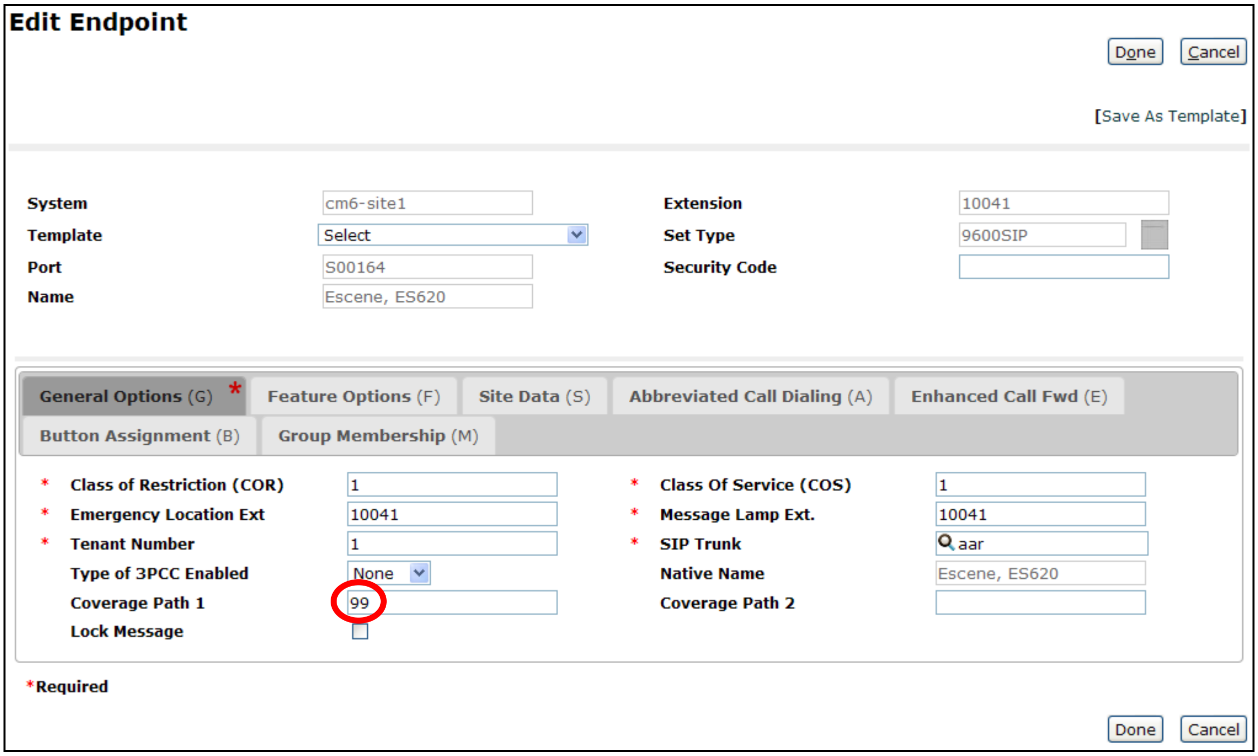


Scroll down to check and expand **Session Manager Profile**. For **Primary Session Manager**, **Origination Application Sequence**, **Termination Application Sequence**, and **Home Location**, select the values corresponding to the applicable Communication Manager. Retain the default values in the remaining 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. Click **Endpoint Editor** to configure the voicemail coverage path.



The **Edit Endpoint** screen is displayed. Enter the voicemail coverage path in the **Coverage Path 1** field. For the compliance testing, coverage path **99** was used. Click **Done** to return to the previous screen.



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

Repeat **Section 5.2** to add a user for each Escene user. In the compliance testing, four users with extensions **10041** to **10044** 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 Escene 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   281
                                      Maximum Stations: 1000   173
                              Maximum XMOBILE Stations: 2400   0
                    Maximum Off-PBX Telephones - EC500: 250    0
                    Maximum Off-PBX Telephones -   OPS: 1000   32
                    Maximum Off-PBX Telephones - PBFMC: 250    0
                    Maximum Off-PBX Telephones - PVFMC: 250    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.722-64K, G.711MU, G.711A and G.729, which are the codecs tested.

```
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.722-64K                   2        20
 2: G.711MU          n          2        20
 3: G.711A           n          2        20
 4: G.729            n          2        20
```

7. Configure Escene SIP Phones

This section provides the procedures for configuring Escene SIP Phones. The procedures are the same for the different models (ES620, ES410, ES320 and ES210) used for the compliance testing. The procedures include the following areas:

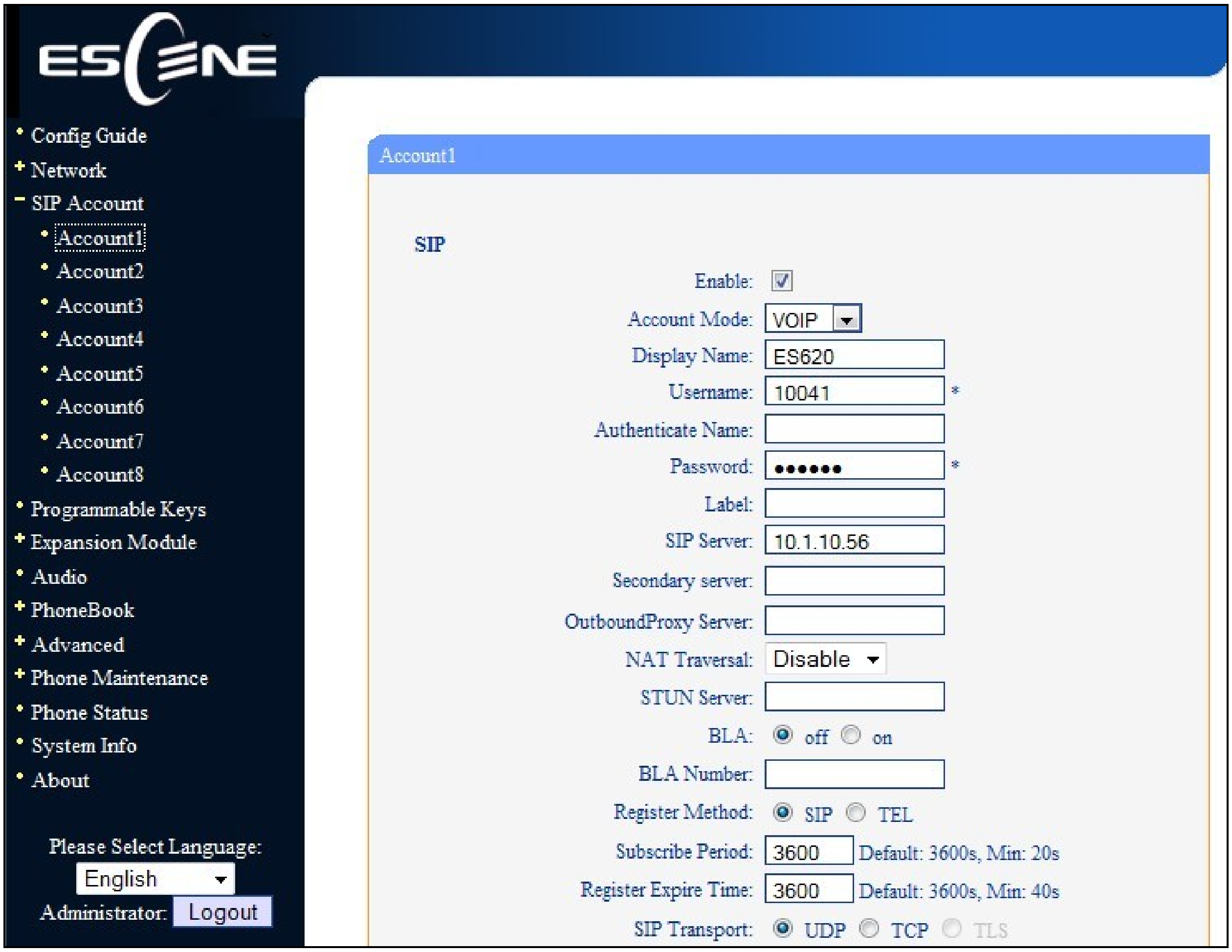
- Access Web Interface
- Configure SIP Account Settings
- Configure Audio Codecs
- Configure DTMF Setting

7.1. Access Web Interface

Enter http://<ip-addr>/, where **<ip-addr>** is the IP address of the Escene IP phone, into the address bar of web browser and log in using a valid account described in **Reference [3]**. The **IPPhone WEB Software** screen is displayed.

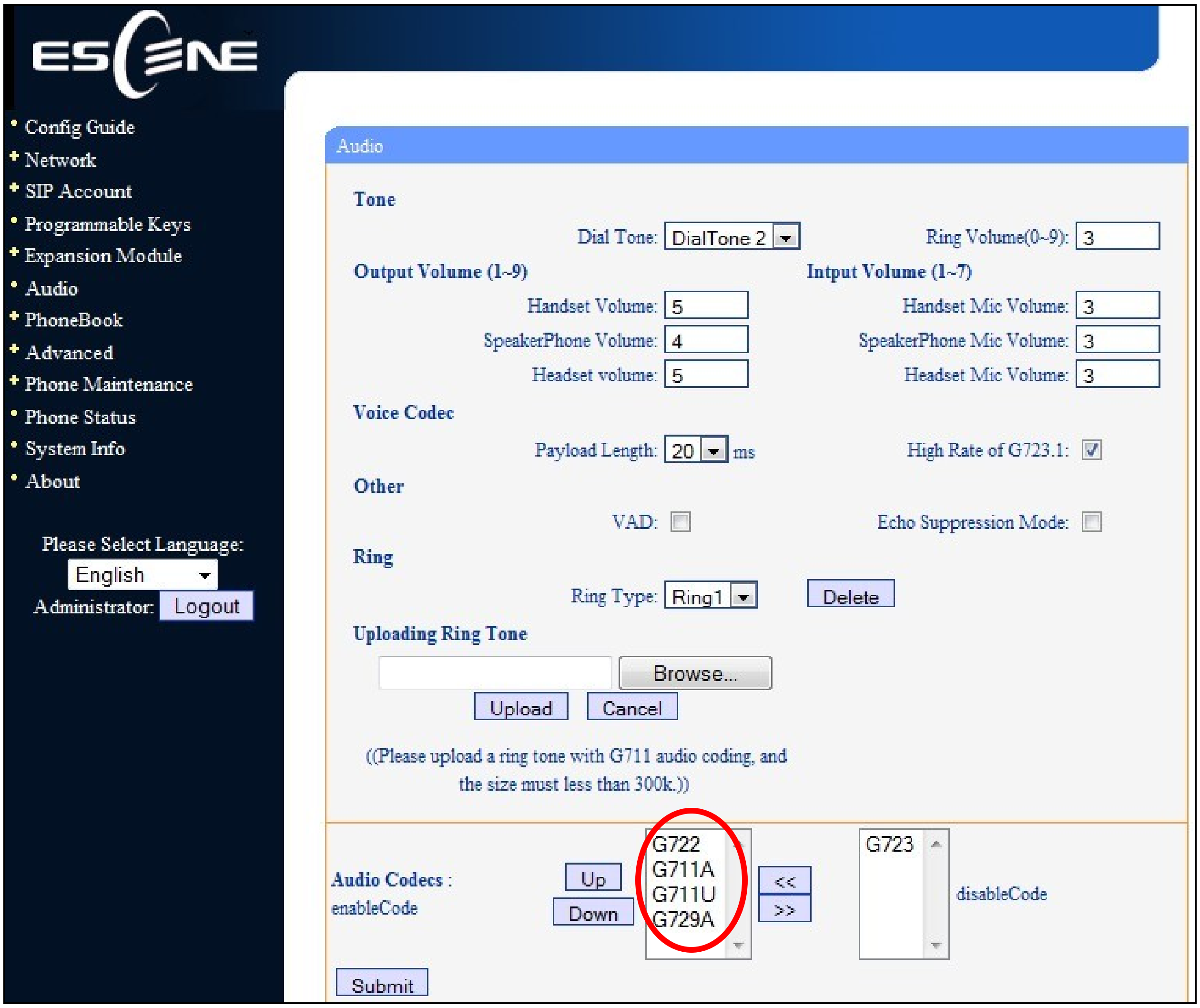
7.2. Configure SIP Account Settings

Select **SIP Account → Account1** from the left menu. Check the **Enable** field and select **VOIP** for **Account Mode**. Enter the IP address of Session Manager signaling interface as shown in **Figure 1** for the **SIP Server** field. For the fields **Username** and **Password**, enter the account details as shown below to match the user settings in Session Manager added in **Section 5.2**. The **Password** is the **Communication Profile Password** in **Section 5.2.2**. Retain the default values for the rest of fields. Click **Submit** to continue (not shown).



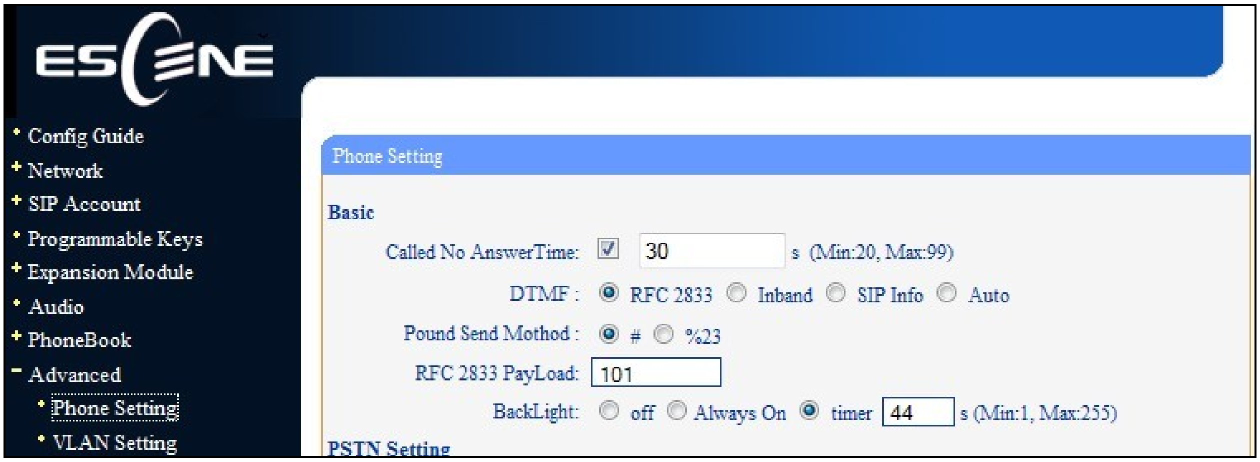
7.3. Configure Audio Codecs

Select **Audio** from the left menu. In the **Audio Codecs** section, enable and prioritize the audio codecs as shown below according to customer requirements.



7.4. Configure DTMF Setting

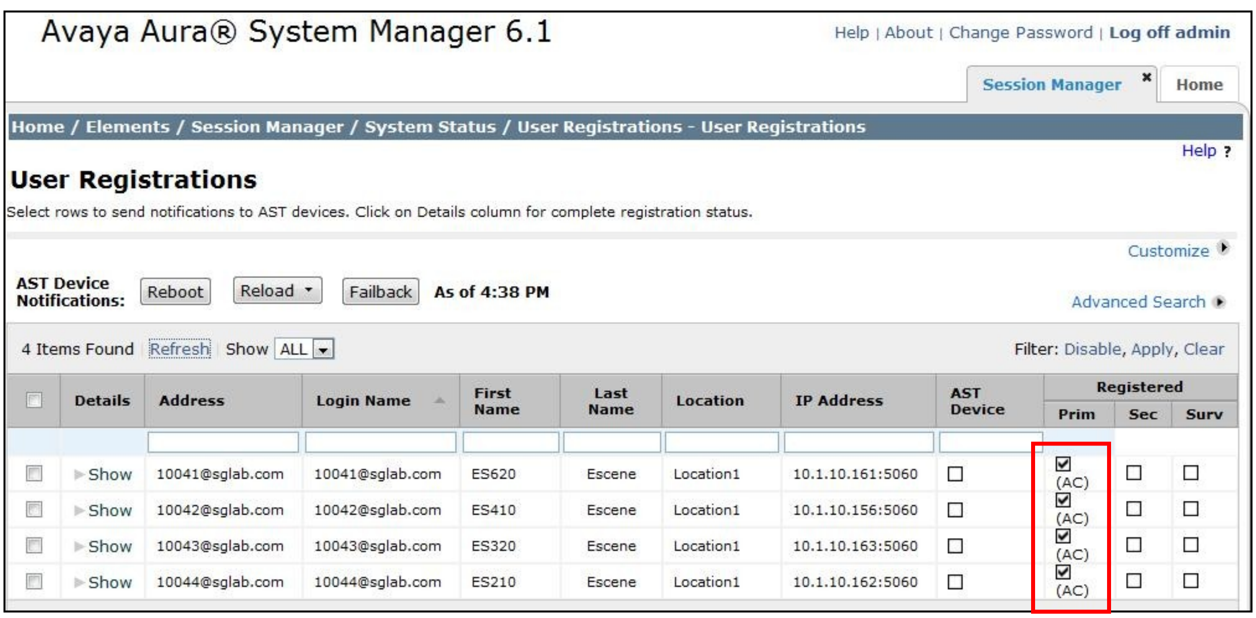
Select **Advanced → Phone Setting** from the left menu. Select **RFC 2833** for DTMF. The **RFC 2833 PayLoad** was left at the default value of **101**.



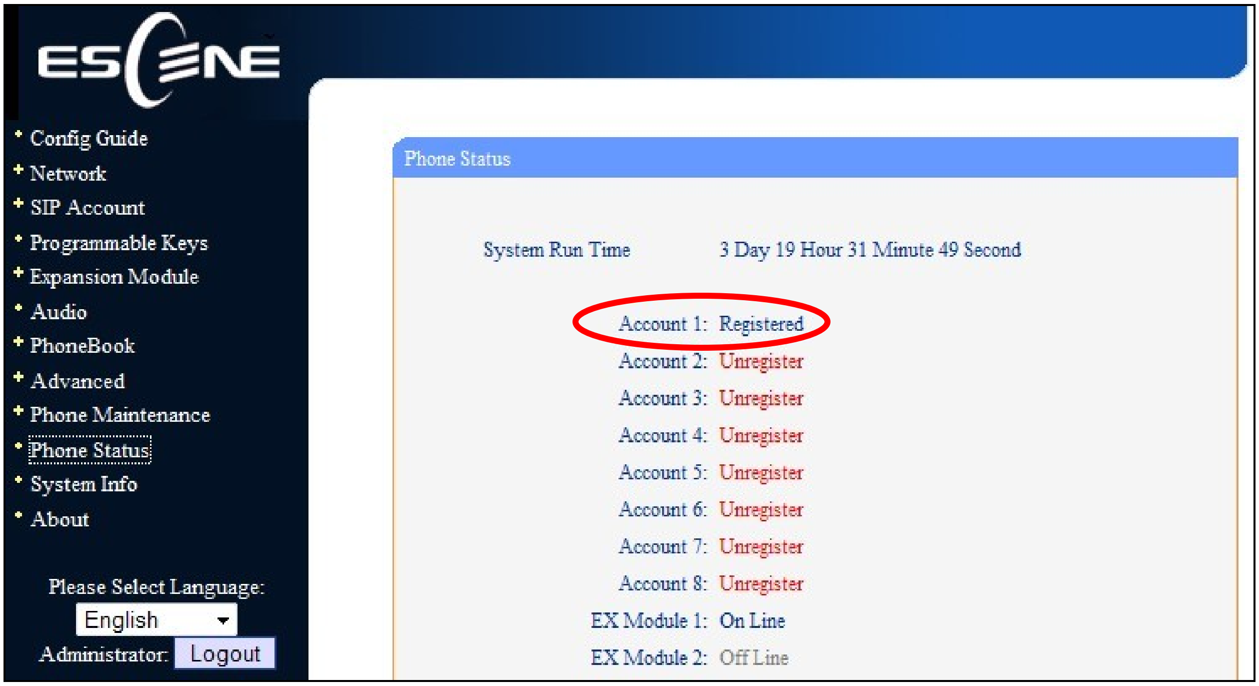
8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Avaya Aura® Session Manager, Avaya Aura® Commuication Manager and Escene 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 Escene phone, click **Phone Status** from the left menu. Verify that the **Account 1** field shows as **Registered**.



9. Conclusion

These Application Notes describe the configuration steps required for Escene Communication Technology ES620, ES410, ES320 and ES210 Enterprise IP 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, Issue 6.0, June 2010.
[2] *Administering Avaya Aura® Session Manager*, Release 6.1, Doc ID 03-603324, Issue 1.1, November 2010.

Escene product information is available at http://www.escene.cn/en/.

[3] *Escene ES620 IP Phone User Manual*.

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.