



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.

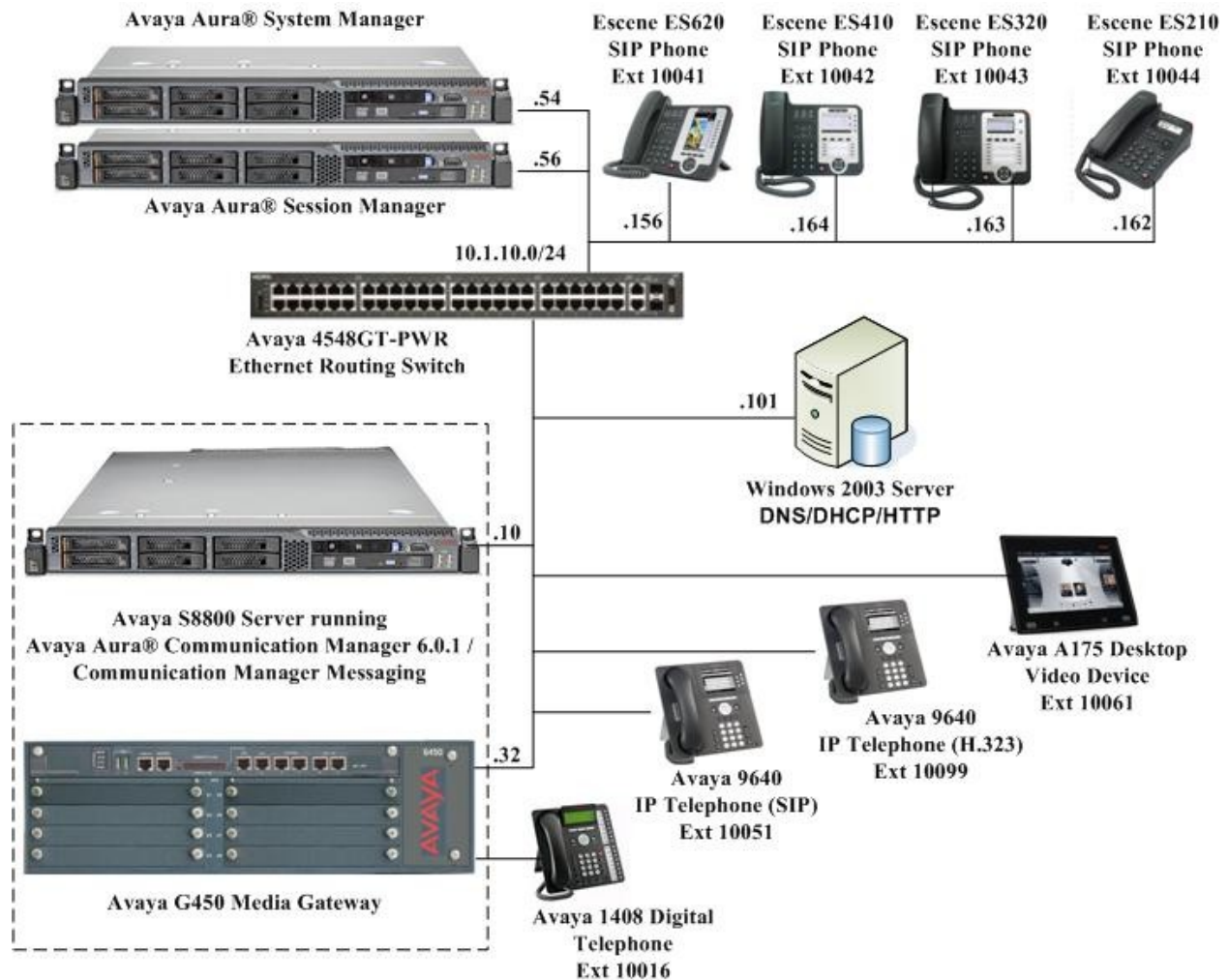


Figure 1: Escene SIP Phones with Avaya SIP Solution

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.

The screenshot shows the Avaya Aura® System Manager 6.1 login interface. At the top, the Avaya logo is on the left and the title 'Avaya Aura® System Manager 6.1' is on the right. Below the title bar is a red navigation bar with 'Home / Log On'. The main heading is 'Log On'. On the left, a box contains text: 'Recommended access to System Manager is via FQDN.' followed by a link 'Go to central login for Single Sign-On'. Below this, it states 'If IP address access is your only option, then note that authentication will fail in the following cases:' followed by a bulleted list: '• First time login with "admin" account' and '• Expired/Reset passwords'. On the right, there are input fields for 'User ID:' and 'Password:'. At the bottom right are 'Log On' and 'Cancel' buttons, and a link 'Change Password'.

AVAYA Avaya Aura® System Manager 6.1

Home / Log On

Log On

Recommended access to System Manager is via FQDN.

[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

User ID:

Password:

[Change Password](#)

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.

The screenshot shows the Avaya Aura System Manager 6.1 User Management interface. The top navigation bar includes the Avaya logo, the title 'Avaya Aura® System Manager 6.1', and links for 'Help | About | Change Password | Log off admin'. Below this is a breadcrumb trail: 'Home / Users / User Management / Manage Users- User Management'. The left sidebar contains a 'User Management' dropdown menu with options: 'Manage Users' (highlighted), 'Public Contacts', 'Shared Addresses', and 'System Presence ACLs'. The main content area is titled 'User Management' and features a 'Users' section with buttons for 'View', 'Edit', 'New' (circled in red), 'Duplicate', 'Delete', and 'More Actions'. Below these buttons is a table with 22 items, showing a list of users. The table has columns for 'Status', 'Name', 'Login Name', 'E164 Handle', and 'Last Login'. The 'New' button is circled in red to indicate where to click to add a new user.

Status	Name	Login Name	E164 Handle	Last Login
1XC SIPUser1	10063@sglab.com	10063		
1XC SIPUser2	10064@sglab.com	10064		
ADVD User1	10061@sglab.com	10061		
ADVD User2	10062@sglab.com	10062		
Avaya, SIP1	10051@sglab.com	10051		

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.

Avaya Aura® System Manager 6.1

Help | About | Change Password | [Log off admin](#)

User Management x Home

Home / Users / User Management / Manage Users - New User Profile

Help ?

New User Profile

Commit Cancel

Identity * Communication Profile * Membership Contacts

Identity ▾

* Last Name: Doe

* First Name: John

Middle Name:

Description:

* Login Name: 10041@sglab.com

* Authentication Type: Basic ▾

* Password: ●●●●●●

* Confirm Password: ●●●●●●

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.

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

User Management x Home

Home / Users / User Management / Manage Users - New User Profile

Help ?

New User Profile

Commit Cancel

Identity * Communication Profile * Membership Contacts

Communication Profile ▾

Communication Profile Password: ••••••

Confirm Password: ••••••

New Delete Done Cancel

Name
Primary

Select : None

* Name: Primary

Default : ☒

Communication Address ▾

New Edit Delete

Type	Handle	Domain
No Records found		

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**.

Communication Address ▾

New Edit Delete

Type	Handle	Domain
No Records found		

Type: Avaya SIP ▾

* Fully Qualified Address: 10041 @ sglab.com ▾

Add Cancel

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.

☒ **Session Manager Profile**

* **Primary Session Manager**

me1-sm

Primary	Secondary	Maximum
23	0	23

Secondary Session Manager

(None)

Primary	Secondary	Maximum

Origination Application Sequence

cm6-site1-app-seq

Termination Application Sequence

cm6-site1-app-seq

Survivability Server

(None)

* **Home Location**

Location1

☒ **Endpoint Profile**

* **System**

cm6-site1

* **Profile Type**

Endpoint

Use Existing Endpoints

☐

* **Extension**

10041

Endpoint Editor

* **Template**

DEFAULT_9630SIP_CM_6_0

Set Type

9630SIP

Security Code

* **Port**

IP

Voice Mail Number

Delete Endpoint on Unassign of Endpoint from User or on Delete User.

☐

Edit Endpoint

[Done] [Cancel]

[Save As Template]

System	<input type="text" value="cm6-site1"/>	Extension	<input type="text" value="10041"/>
Template	<input type="text" value="Select"/>	Set Type	<input type="text" value="9600SIP"/>
Port	<input type="text" value="S00164"/>	Security Code	<input type="text"/>
Name	<input type="text" value="Escene, ES620"/>		

General Options (G) * Feature Options (F) Site Data (S) Abbreviated Call Dialing (A) Enhanced Call Fwd (E)

Button Assignment (B) Group Membership (M)

* Class of Restriction (COR)	<input type="text" value="1"/>	* Class Of Service (COS)	<input type="text" value="1"/>
* Emergency Location Ext	<input type="text" value="10041"/>	* Message Lamp Ext.	<input type="text" value="10041"/>
* Tenant Number	<input type="text" value="1"/>	* SIP Trunk	<input type="text" value="aar"/>
Type of 3PCC Enabled	<input type="text" value="None"/>	Native Name	<input type="text" value="Escene, ES620"/>
Coverage Path 1	<input type="text" value="99"/>	Coverage Path 2	<input type="text"/>
Lock Message	<input type="checkbox"/>		

*Required

[Done] [Cancel]

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 Codec	Silence Suppression	Frames Per Pkt	Packet 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).

The screenshot shows the ES620 web interface for configuring SIP Account settings. The left sidebar contains a navigation menu with the following items: Config Guide, Network, SIP Account (expanded), Account1 (selected), Account2, Account3, Account4, Account5, Account6, Account7, Account8, Programmable Keys, Expansion Module, Audio, PhoneBook, Advanced, Phone Maintenance, Phone Status, System Info, and About. Below the menu is a language selection dropdown set to 'English' and an 'Administrator: Logout' link. The main content area is titled 'Account1' and 'SIP'. It contains the following configuration fields: 'Enable' (checked), 'Account Mode' (VOIP), 'Display Name' (ES620), 'Username' (10041), 'Authenticate Name' (empty), 'Password' (masked with dots), 'Label' (empty), 'SIP Server' (10.1.10.56), 'Secondary server' (empty), 'OutboundProxy Server' (empty), 'NAT Traversal' (Disable), 'STUN Server' (empty), 'BLA' (radio buttons for off and on, with 'off' selected), 'BLA Number' (empty), 'Register Method' (radio buttons for SIP and TEL, with 'SIP' selected), 'Subscribe Period' (3600, Default: 3600s, Min: 20s), 'Register Expire Time' (3600, Default: 3600s, Min: 40s), and 'SIP Transport' (radio buttons for UDP, TCP, and TLS, with 'UDP' selected).

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.

ES (Escene)

- Config Guide
- Network
- SIP Account
- Programmable Keys
- Expansion Module
- Audio
- PhoneBook
- Advanced
- Phone Maintenance
- Phone Status
- System Info
- About

Please Select Language: **English**

Administrator: **Logout**

Audio

Tone

Dial Tone: **DialTone 2** Ring Volume(0~9): **3**

Output Volume (1~9)

Handset Volume: **5**

SpeakerPhone Volume: **4**

Headset volume: **5**

Input Volume (1~7)

Handset Mic Volume: **3**

SpeakerPhone Mic Volume: **3**

Headset Mic Volume: **3**

Voice Codec

Payload Length: **20** ms High Rate of G723.1: ☒

Other

VAD: ☐ Echo Suppression Mode: ☐

Ring

Ring Type: **Ring1** **Delete**

Uploading Ring Tone

Browse...

Upload **Cancel**

((Please upload a ring tone with G711 audio coding, and the size must less than 300k.))

Audio Codecs :

enableCode

Up **Down** **<<** **>>**

G722
G711A
G711U
G729A

G723 **disableCode**

Submit

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**.

EScene

- Config Guide
- Network
- SIP Account
- Programmable Keys
- Expansion Module
- Audio
- PhoneBook
- Advanced
 - Phone Setting
 - VLAN Setting

Phone Setting

Basic

Called No AnswerTime: ☒ 30 s (Min:20, Max:99)

DTMF: ☒ RFC 2833 ☐ Inband ☐ SIP Info ☐ Auto

Pound Send Method: ☒ # ☐ %23

RFC 2833 PayLoad:

BackLight: ☐ off ☐ Always On ☒ timer s (Min:1, Max:255)

PSTN Setting

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Avaya Aura® Session Manager, Avaya Aura® Communication 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.

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Session Manager x Home

Home / Elements / Session Manager / System Status / User Registrations - User Registrations

Help ?

User Registrations

Select rows to send notifications to AST devices. Click on Details column for complete registration status.

AST Device Notifications: As of 4:38 PM

Customize

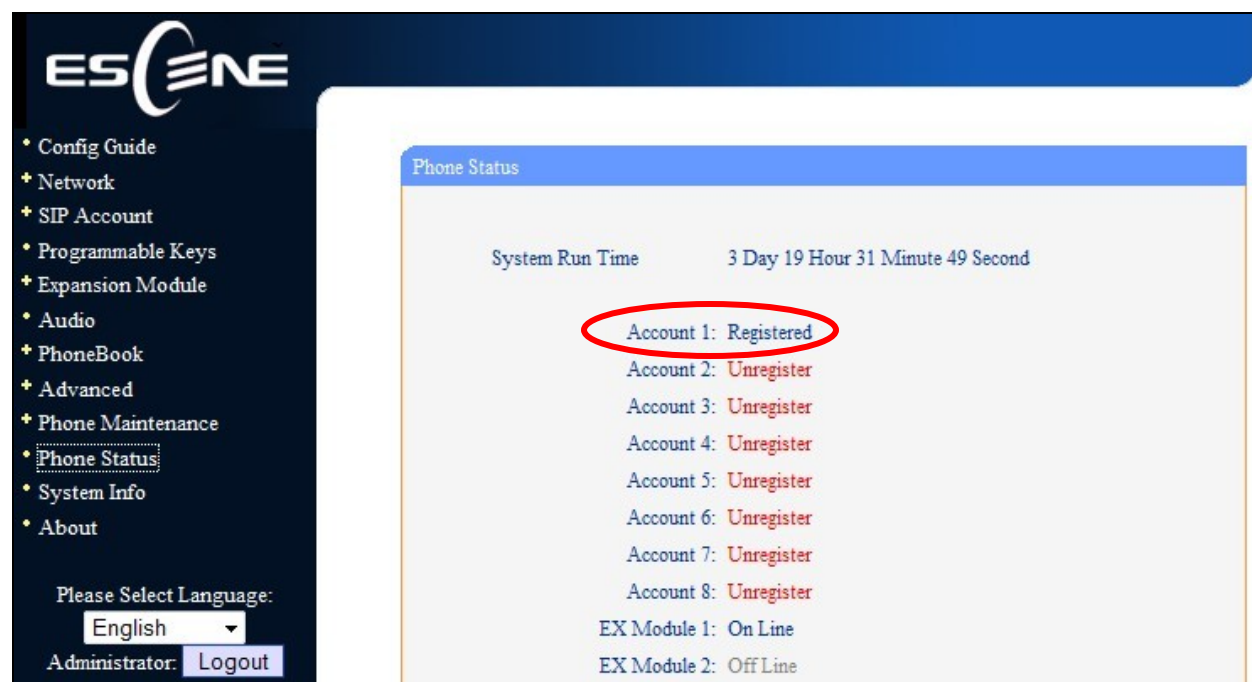
Advanced Search

4 Items Found Show ALL

Filter: Disable, Apply, Clear

	Details	Address	Login Name	First Name	Last Name	Location	IP Address	AST Device	Registered		
									Prim	Sec	Surv
<input type="checkbox"/>	► Show	10041@sglab.com	10041@sglab.com	ES620	Escene	Location1	10.1.10.161:5060	<input type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	► Show	10042@sglab.com	10042@sglab.com	ES410	Escene	Location1	10.1.10.156:5060	<input type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	► Show	10043@sglab.com	10043@sglab.com	ES320	Escene	Location1	10.1.10.163:5060	<input type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	► Show	10044@sglab.com	10044@sglab.com	ES210	Escene	Location1	10.1.10.162:5060	<input type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>

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*.

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