Avaya Solution & Interoperability Test Lab

Application Notes for the LifeSize Phone with Avaya Communication Manager and Avaya SIP Enablement Services – Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Communication Manager, Avaya SIP Enablement Services (SES), and LifeSize Communications Tabletop Conference Room Phones. The LifeSize Phone is a SIP-based VoIP tabletop High Definition phone intended for use in conference rooms and similar environments. During compliance testing, the LifeSize Phone successfully registered with Avaya SES, placed and received calls to and from SIP and non-SIP telephones, and established conference calls. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the Developer*Connection* Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe a solution comprised of Avaya Communication Manager 3.1.2, Avaya SIP Enablement Services 3.1.1, and LifeSize Phone. LifeSize Phone is a SIP-based VoIP tabletop High Definition phone intended for use in conference rooms and similar environments. Each LifeSize Phone supports two lines, and can bridge calls on the two lines to establish a 3-party conference.

Figure 1 illustrates a sample configuration consisting of Avaya S8710 Media Servers, an Avaya G650 Media Gateway, an Avaya SIP Enablement Services (SES) server, and the LifeSize Phone. Avaya Communication Manager is installed on the S8710 Media Servers. The solution described herein is also extensible to other Avaya Media Servers and Media Gateways. For completeness, Avaya 4600 Series SIP IP Telephones, Avaya 4600 Series H.323 IP Telephones, and Avaya 6400 and 8400 Series Digital Telephones, are included in **Figure 1** to demonstrate calls between the SIP-based LifeSize Phone and Avaya SIP, H.323, and digital phones. The analog PSTN phone is also included to demonstrate calls routed by Avaya Communication Manager between the LifeSize Phone and the PSTN.

The LifeSize Phone originates a call by sending a call request (SIP INVITE message) to the Avaya SES server. The Avaya SES server routes the call over a SIP trunk to Avaya Communication Manager for origination services. If the call is destined for another local SIP phone, such as another LifeSize Phone or an Avaya SIP phone, then Avaya Communication Manager routes the call back over the SIP trunk to the Avaya SES server, which in turn delivers the call to the destination SIP phone. Otherwise, Avaya Communication Manager routes the call to the PSTN, a local Avaya H.323, digital, or analog phone, an adjunct, a vector, a hunt group, etc., depending on the destination number. For a call arriving to Avaya Communication Manager that is destined for the LifeSize Phone, Avaya Communication Manager routes the call over the SIP trunk to the Avaya SES server, which in turn delivers the call to the LifeSize Phone. These Application Notes assume that the SIP trunk between Avaya Communication Manager and the Avaya SES server has already been configured. For details on configuring SIP trunks on Avaya Communication Manager and Avaya SES, consult [1] and [4].

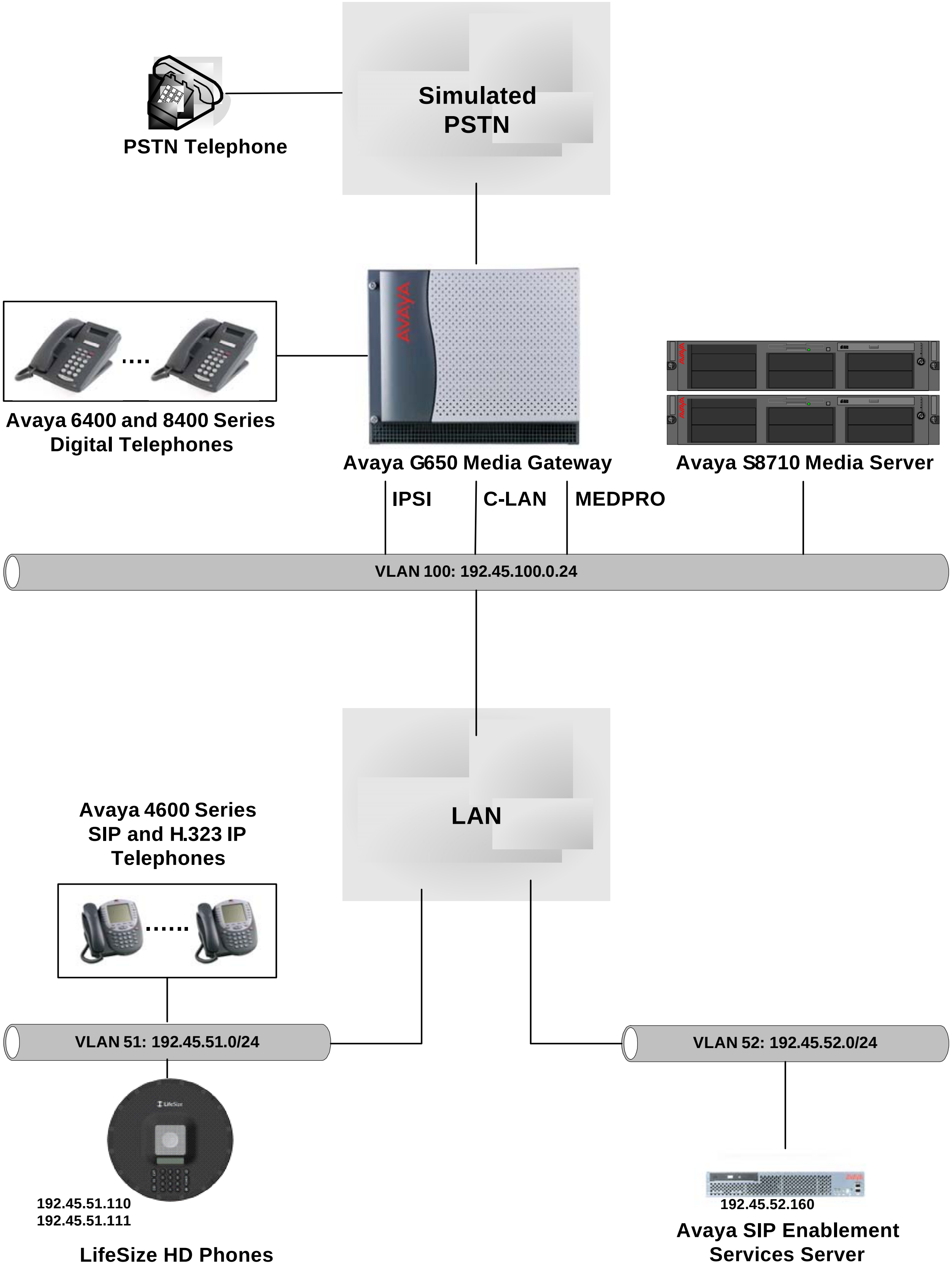


Figure 1: Sample configuration.

2. Equipment and Software Validated

The following equipment and software/firmware were used for the sample configuration provided:

Equipment	Software/Firmware
Avaya S8710 Media Server	Avaya Communication Manager 3.1.2 (R013x.01.2.632.1)
Avaya G650 Media Gateway	-
TN2312BP IP Server Interface	HW12 FW 31
TN799DP C-LAN Interface	HW01 FW 17
TN2302AP IP Media Processor	HW20 FW 112
Avaya SIP Enablement Services Server	SES 3.1.1(R03.1.1-03.1.114.0)
Avaya 4600 Series IP Telephones	2.3 (4602SW H.323) 2.5 (4625SW H.323) 2.2.3 (4610SW SIP)
Avaya 6400 and 8400 Series Digital Telephones	-
LifeSize Communications LifeSize Tabletop Conferencing Phones	2.1.4(9)
Analog Telephone	-

3. Configure Avaya Communication Manager

This section describes the steps for configuring IP codec sets and associating SIP phone numbers with off-PBX telephone stations in Avaya Communication Manager. The steps are performed from the Avaya Communication Manager System Access Terminal (SAT) interface. IP codec sets identify the codecs that may be used in calls involving VoIP endpoints. An off-PBX telephone is a phone that Avaya Communication Manager does not control, such as a cellular phone, a home phone, or a SIP phone. Avaya Communication Manager features and calling privileges, however, can be applied to an off-PBX telephone by associating a local, on-PBX, extension with the off-PBX telephone. This approach is taken for SIP phones that register with the Avaya SES server and intend to use Avaya Communication Manager for call origination and termination services. Specifically, an Administration WithOut Hardware (AWOH) on-PBX station is administered in Avaya Communication Manager and then associated with the phone number of the SIP phone. Similarly, on the Avaya SES server, the number of the SIP phone is administratively associated with the extension of the on-PBX station. Throughout the rest of this document, on-PBX stations associated with SIP phones in such a manner will be referred to as Outboard Proxy SIP (OPS) stations.

3.1. IP Codec Set

Enter the **change ip-codec-set c** command, where “c” is a number between 1 and 7, inclusive. Enter at least one of the codecs supported in the LifeSize Phone (see **Section 5 Step 6**). IP codec sets are specified in the **IP Network Region** forms to define which codecs may be used within and between network regions.

```
change ip-codec-set 2                                          Page   1 of   2

                          IP Codec Set

    Codec Set: 2

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.729AB           n         2        20
 2: G.711MU           n         2        20
 3:
 4:
 5:
 6:
 7:

     Media Encryption
 1: none
 2:
 3:
```

3.2. SIP Stations

This section describes the steps for administering OPS stations in Avaya Communication Manager and associating the OPS station extensions with the numbers of the LifeSize Phone.

Step	Description
1.	Enter the **display system-parameters customer-options** command. Verify that there are sufficient **Maximum Off-PBX Telephones – OPS** licenses. If not, contact an authorized Avaya account representative to obtain additional licenses.

```
display system-parameters customer-options                      Page   1 of  10
                                OPTIONAL FEATURES

     G3 Version: V13
       Location: 1                             RFA System ID (SID): 1
       Platform: 8                             RFA Module ID (MID): 1

                                                                USED
                                Platform Maximum Ports: 44000 908
                                      Maximum Stations: 36000 410
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 5     0
                    Maximum Off-PBX Telephones -   OPS: 200   50
                    Maximum Off-PBX Telephones - SCCAN: 0     0
```

Step	Description
2.	Enter the **add station s** command, where "**s**" is an available extension in the dial plan, to administer an OPS station. On Page 1 of the **station** form, set **Type** to "**6408D+**" and **Port** to "**X**", and enter a descriptive **Name**.

```
add station 54010                                               Page   1 of   4
                                     STATION

Extension: 54010                          Lock Messages? n          BCC: 0
     Type: 6408D+                         Security Code:             TN: 1
     Port: X                            Coverage Path 1:            COR: 1
     Name: SIP-54010                    Coverage Path 2:            COS: 1
                                        Hunt-to Station:

STATION OPTIONS
              Loss Group: 2             Personalized Ringing Pattern: 1
             Data Module? n                         Message Lamp Ext: 54010
            Speakerphone: 2-way                  Mute Button Enabled? y
        Display Language: english

                                                   Media Complex Ext:
                                                        IP SoftPhone? n
```

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>
Enter the change off-pbx-telephone station-mapping s command, where “s” is the extension of the OPS station configured in Step 2. On Page 1 of the off-pbx-telephone station-mapping form, configure the following:

Station Extension – Enter the extension of the OPS station.
Application – Set to “OPS”.
Phone Number – Enter the number that the LifeSize Phone will use for registration and call origination and termination. In the example below, the Phone Number is the same as the OPS Station Extension, but is not required to be the same.
Trunk Selection – Enter the number of the SIP trunk group connected to the Avaya SES server.
Configuration Set – Set to “1”, which during compliance testing used the default values of the off-pbx-telephone configuration-set form.

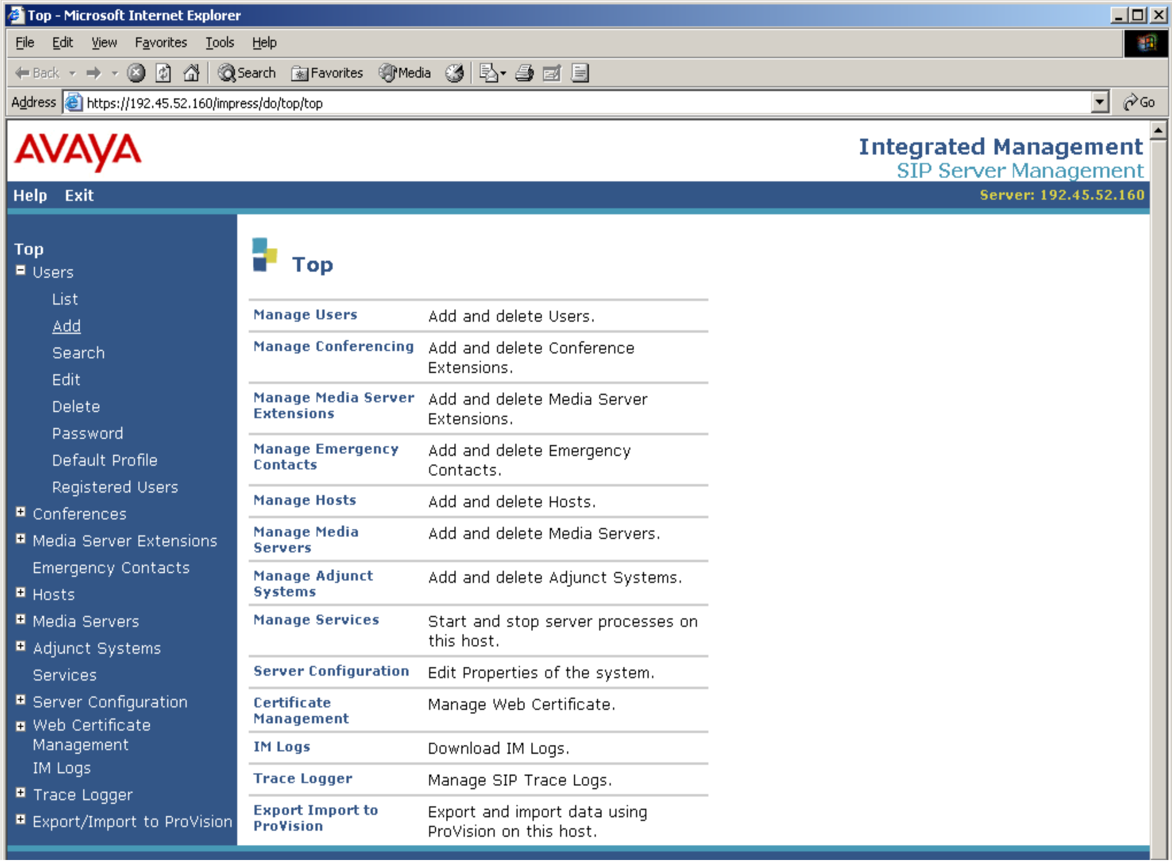
<pre>
change off-pbx-telephone station-mapping 54010   Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station   Application   Dial   Phone Number   Trunk   Configuration
 Extension   Prefix   Selection   Set
 54010   OPS   - 54010   10   1
</pre>
</td></tr>
<tr><td>4.</td><td>Repeat Steps 2 and 3 as necessary to administer OPS stations and associations for additional LifeSize Phones.</td></tr>
</table>

4. Configure Avaya SIP Enablement Services

This section describes the steps for creating SIP user accounts in Avaya SIP Enablement Services (SES) and associating the SIP users with an Avaya Communication Manager OPS station extension. The LifeSize Phone will register with Avaya SES using the SIP user accounts.

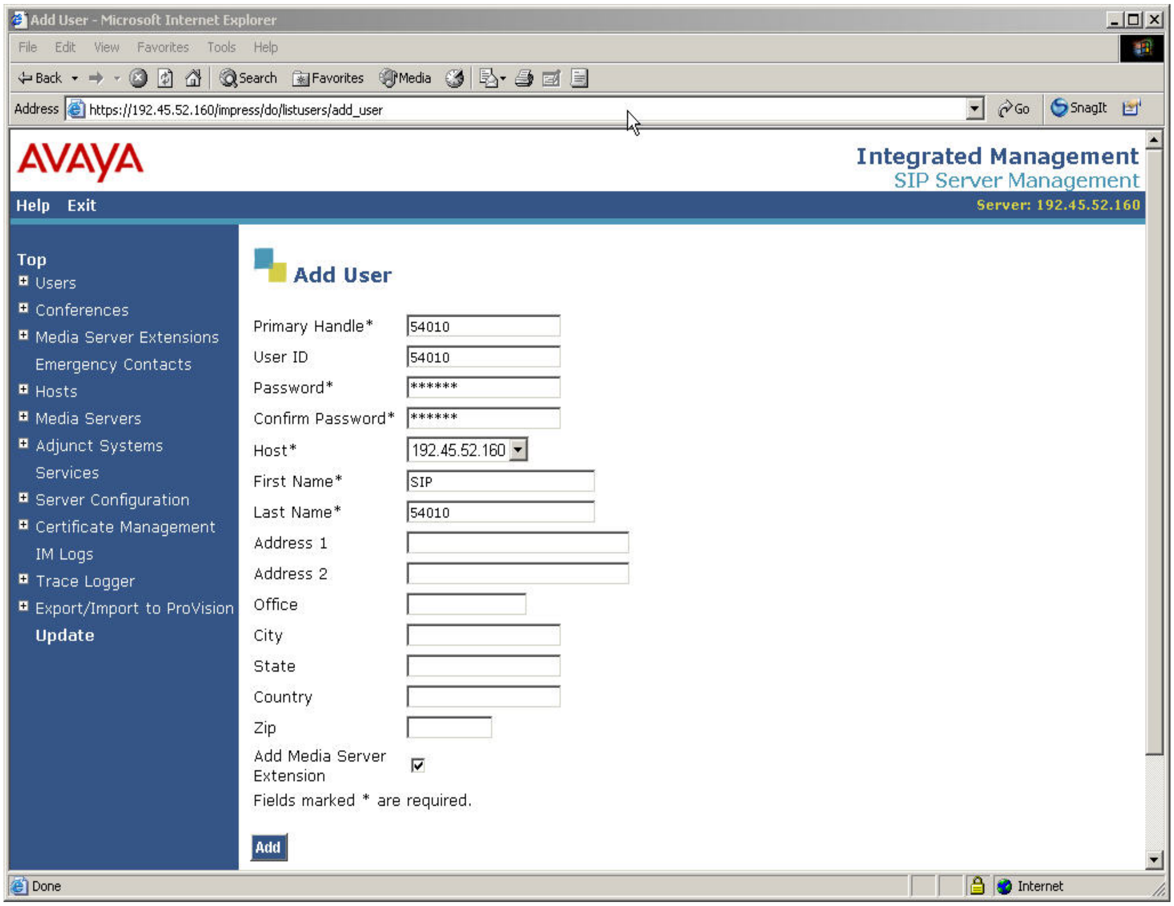
This section assumes that the necessary Avaya SES configuration steps for establishing a SIP trunk with Avaya Communication Manager have been completed. For further details, consult [4].

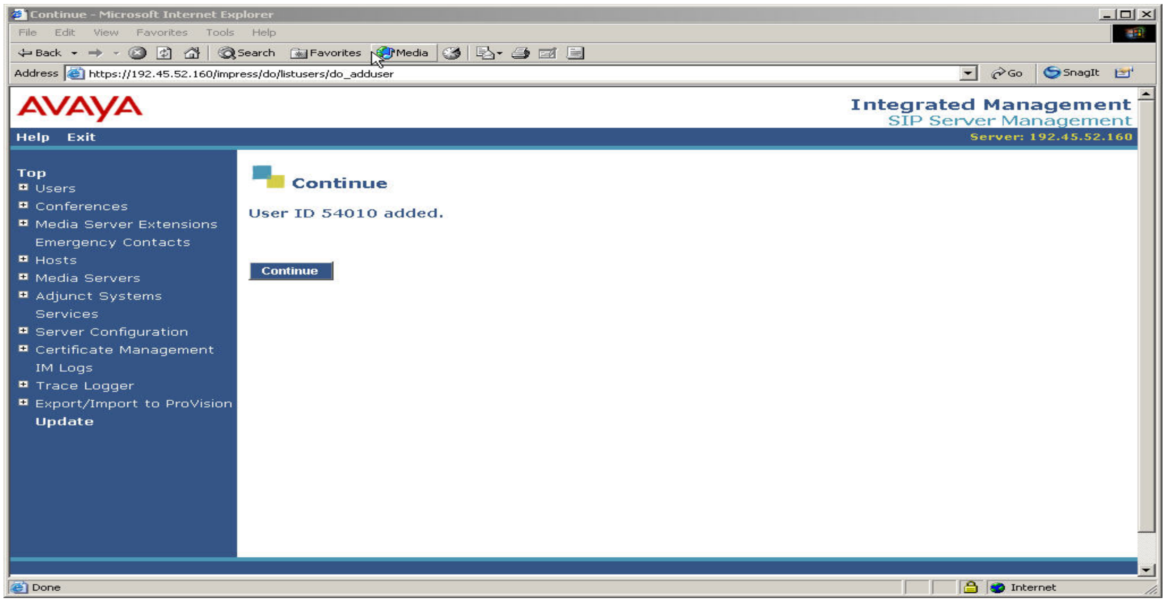
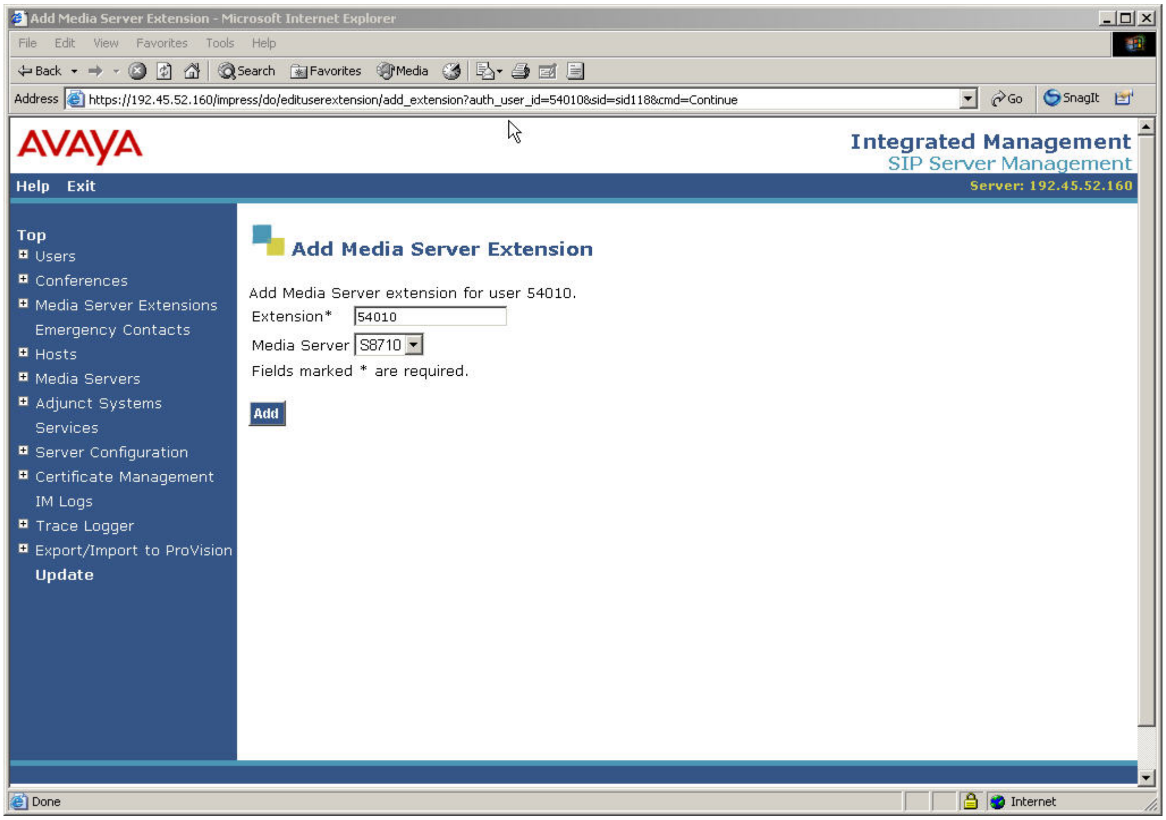
Step	Description
1.	Open a web browser, enter http://<IP address of Avaya SES server>/admin for the URL, and log in with the appropriate credentials. Click on the "**Launch Administration Web Interface**" link upon successful login.
2.	In the left pane of the SES Administration Web Interface, expand "**Users**" and click on "**Add**". 

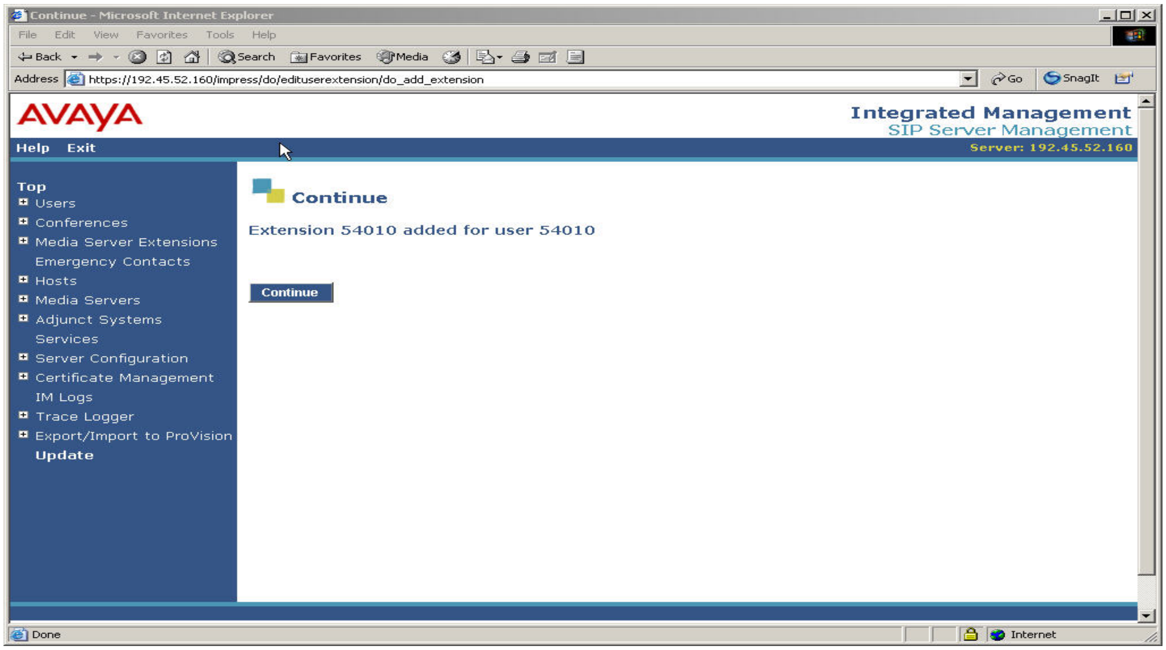
Address: https://192.45.52.160/impress/do/top/top

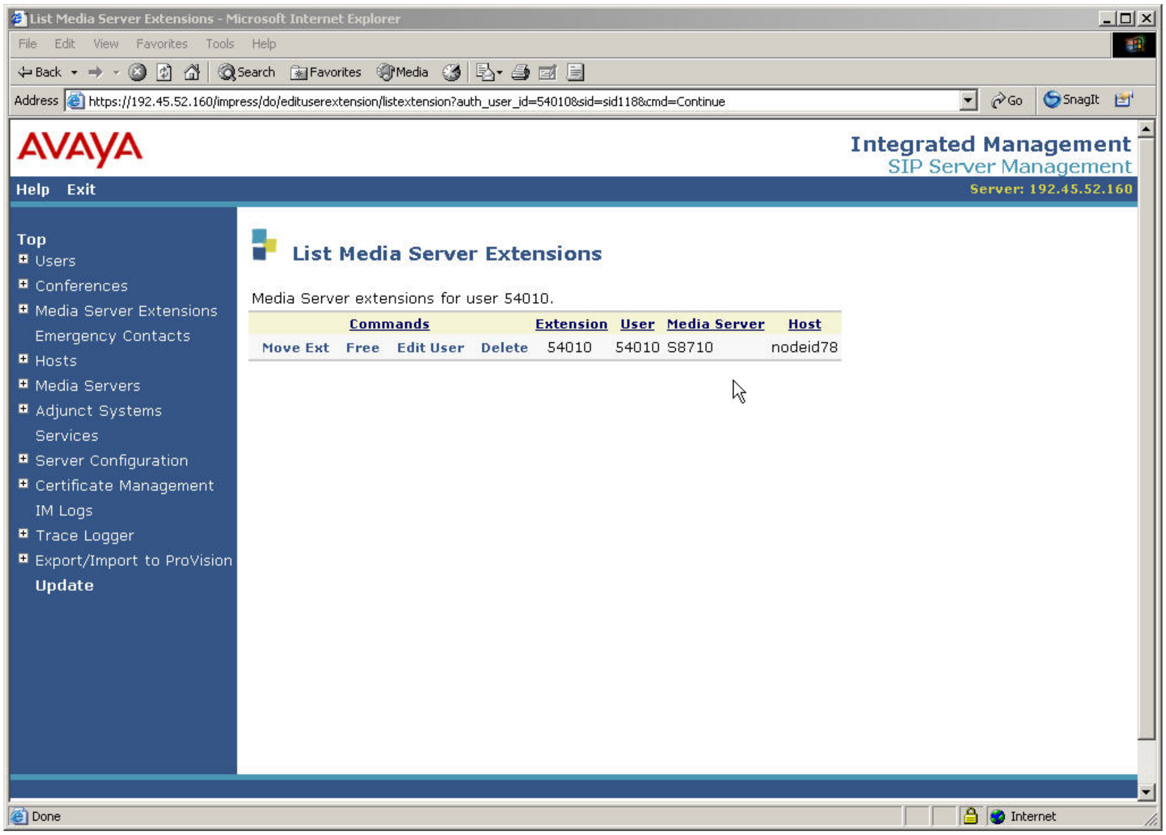
Server: 192.45.52.160

Manage Users	Add and delete Users.
Manage Conferencing	Add and delete Conference Extensions.
Manage Media Server Extensions	Add and delete Media Server Extensions.
Manage Emergency Contacts	Add and delete Emergency Contacts.
Manage Hosts	Add and delete Hosts.
Manage Media Servers	Add and delete Media Servers.
Manage Adjunct Systems	Add and delete Adjunct Systems.
Manage Services	Start and stop server processes on this host.
Server Configuration	Edit Properties of the system.
Certificate Management	Manage Web Certificate.
IM Logs	Download IM Logs.
Trace Logger	Manage SIP Trace Logs.
Export Import to ProVision	Export and import data using ProVision on this host.

Step	Description
3.	In the **Add User** page, configure the following: - **Primary Handle –** Enter the phone number of the LifeSize Phone. The number must match the **phone number** entered in **Section 3.2 Step 3**. - **Password** and **Confirm Password –** Specify a password that the LifeSize Phone must use to successfully register with Avaya SES. - **Host –** Select the IP address or FQDN of the Avaya SES server. - **First Name** and **Last Name –** Enter descriptive names. - Check the **Add Media Server Extension** checkbox. Click on "**Add**".  Address https://192.45.52.160/impress/do/listusers/add_user Server: 192.45.52.160 **Add User** Primary Handle* 54010 User ID 54010 Password* ****** Confirm Password* ****** Host* 192.45.52.160 First Name* SIP Last Name* 54010 Address 1 Address 2 Office City State Country Zip Add Media Server Extension ☑ Fields marked * are required.

Step	Description
4.	Click on "**Continue**".  Address: https://192.45.52.160/impress/do/listusers/do_adduser **Continue** User ID 54010 added.
5.	Enter the **Extension** of the corresponding Avaya Communication Manager OPS station configured in **Section 3.2 Step 3** and select the **Media Server** on which the OPS station is configured. Calls from this user will always be routed to the selected Avaya Communication Manager media server for origination services. Click on "**Add**".  Address: https://192.45.52.160/impress/do/edituserextension/add_extension?auth_user_id=54010&sid=sid118&cmd=Continue **Add Media Server Extension** Add Media Server extension for user 54010. Extension* 54010 Media Server S8710 Fields marked * are required. **Note:** Media Server (S8710) was previously configured on SES.

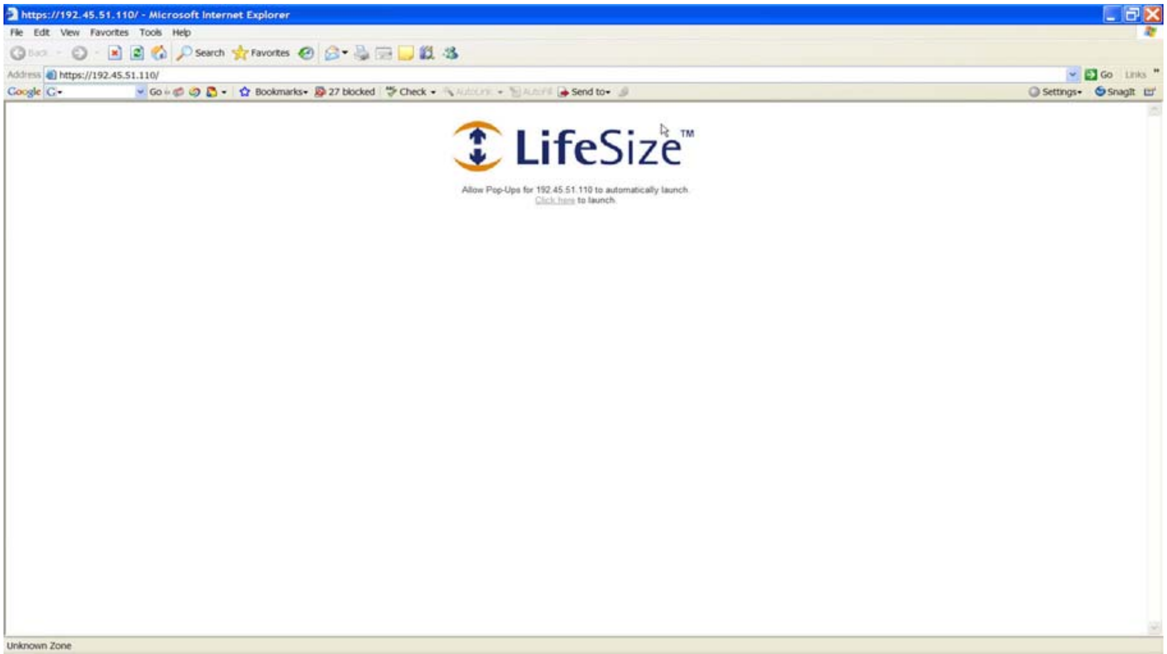
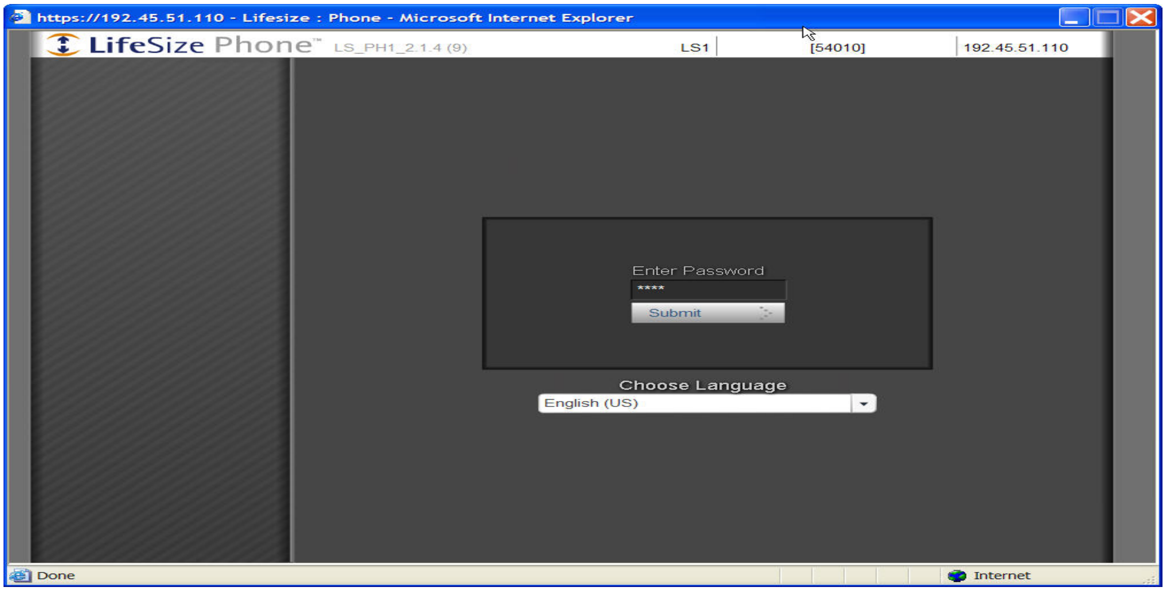
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>6.</td><td>Click on “Continue”.


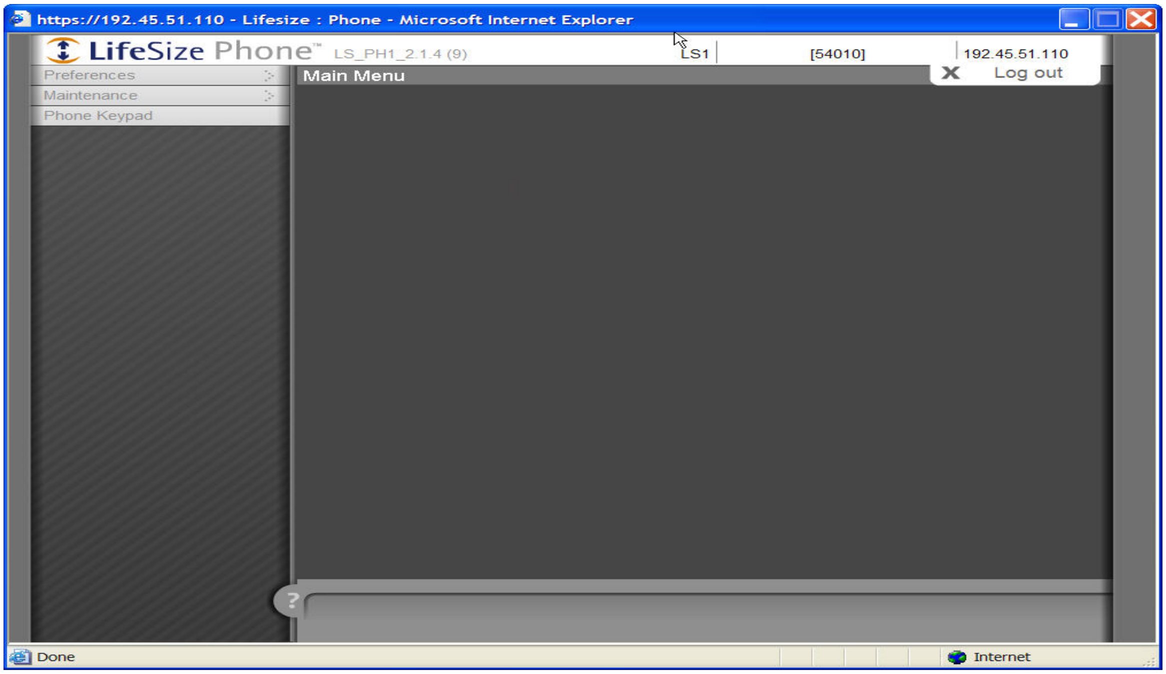
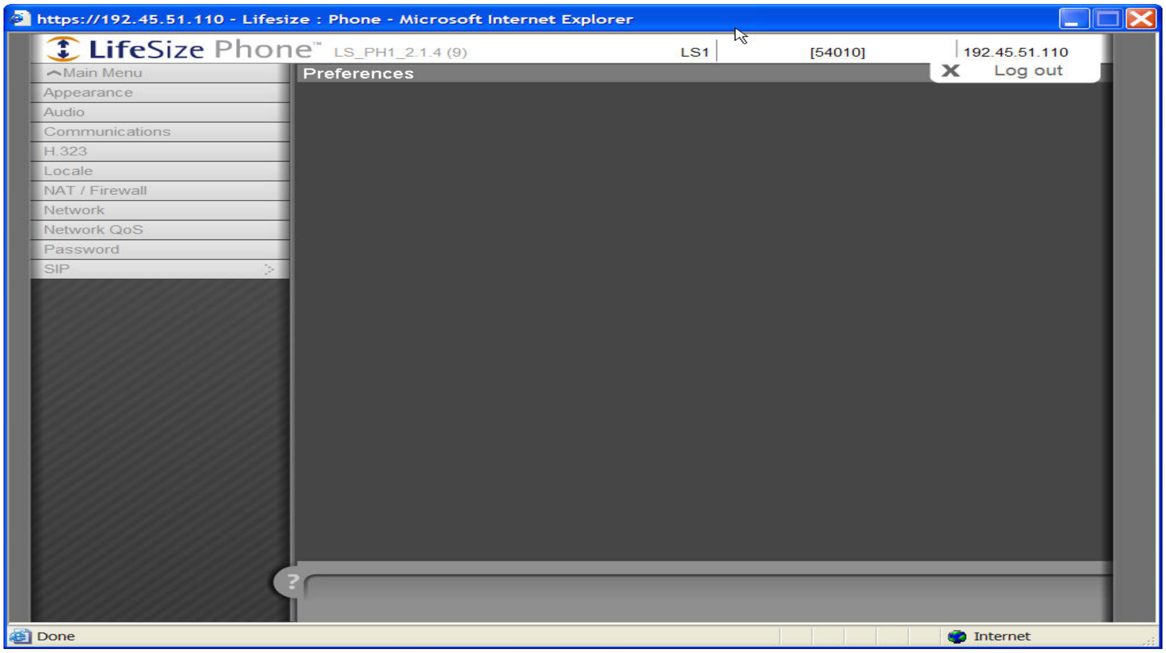
https://192.45.52.160/impress/do/edituserextension/do_add_extension
Continue
Extension 54010 added for user 54010</td></tr>
<tr><td>7.</td><td>Click on “Update” at the bottom of the left pane.


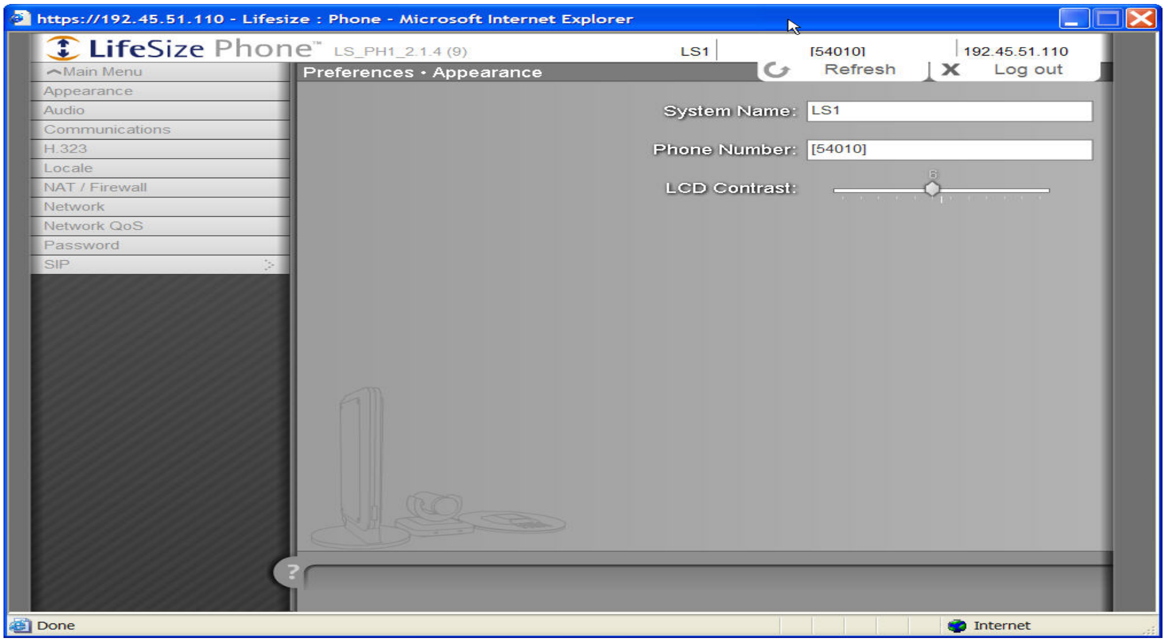
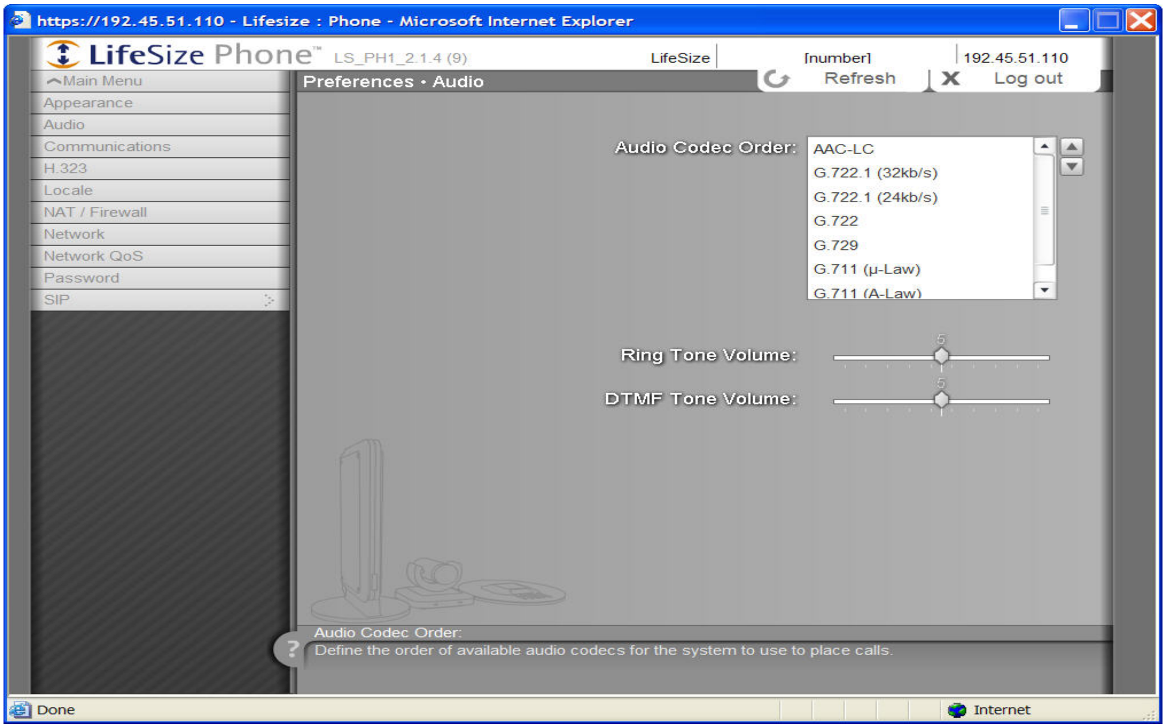
https://192.45.52.160/impress/do/edituserextension/listextension?auth_user_id=54010&sid=sid118&cmd=Continue
List Media Server Extensions
Media Server extensions for user 54010.
<table>
<tr><th colspan="4">Commands</th><th>Extension</th><th>User</th><th>Media Server</th><th>Host</th></tr>
<tr><td>Move Ext</td><td>Free</td><td>Edit User</td><td>Delete</td><td>54010</td><td>54010</td><td>S8710</td><td>nodeid78</td></tr>
</table></td></tr>
<tr><td>8.</td><td>Repeat Steps 2 – 7 as necessary to configure SIP users for additional LifeSize Phones.</td></tr>
</table>

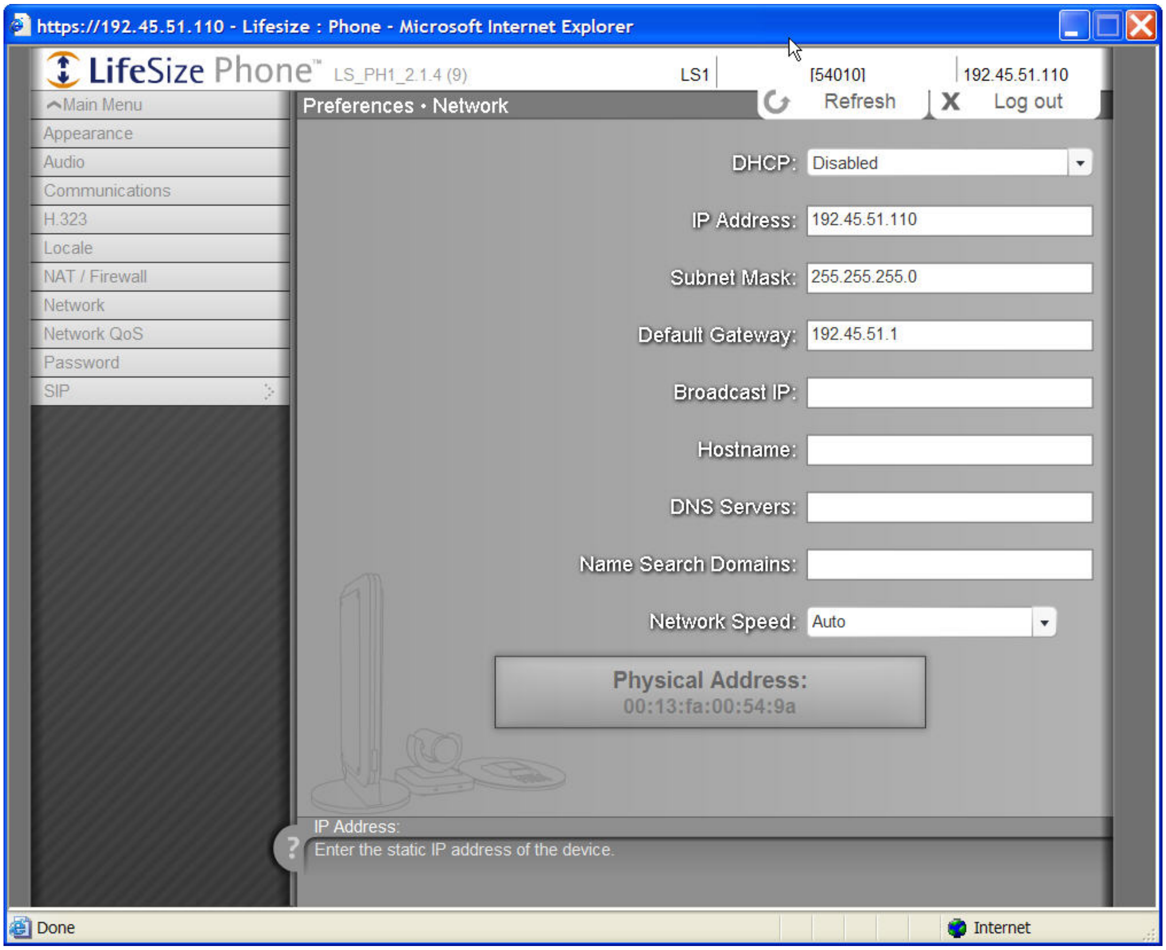
5. Configure the LifeSize Phone

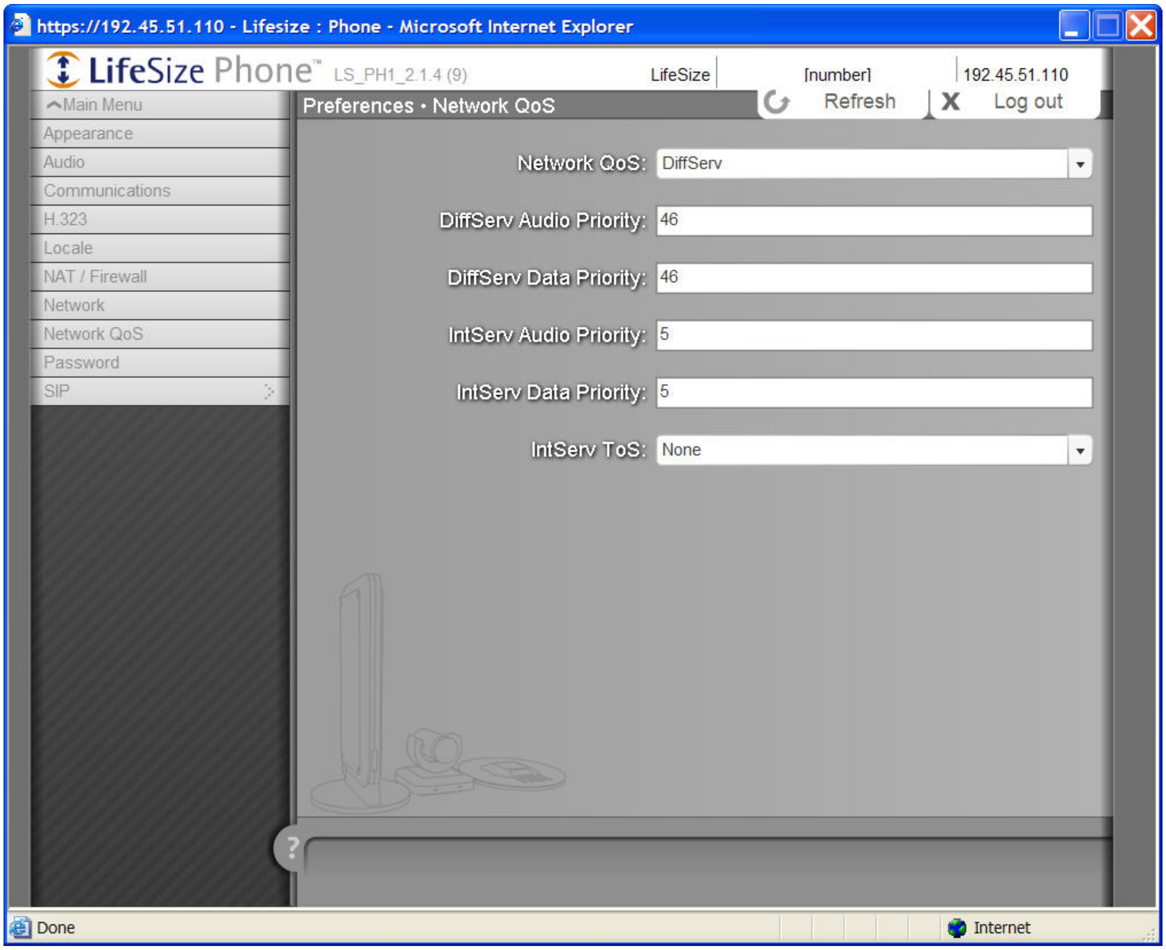
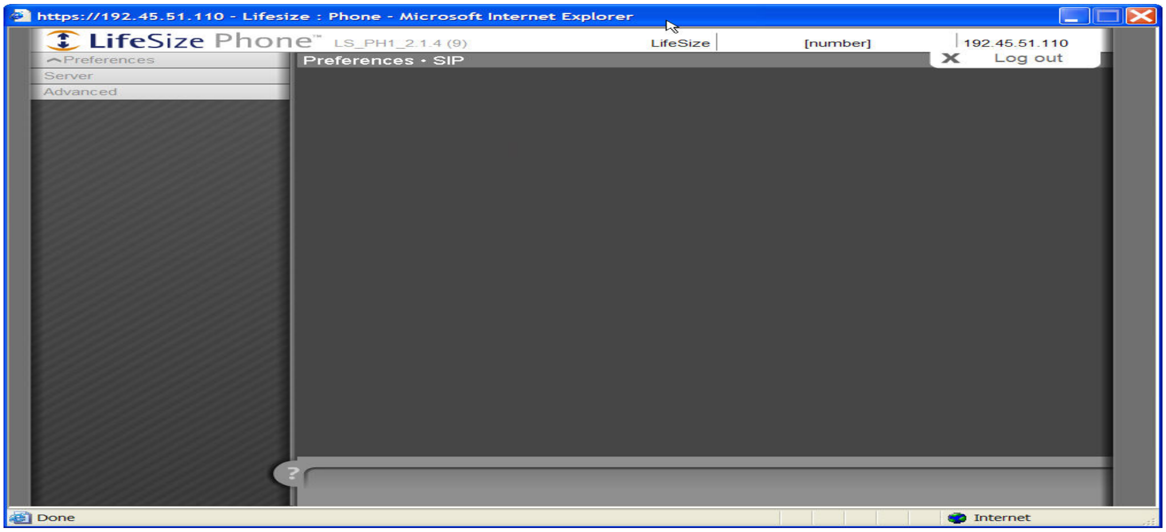
This section describes the steps for configuring the LifeSize Phone. This section assumes that the LifeSize Phone has an IP address already configured.

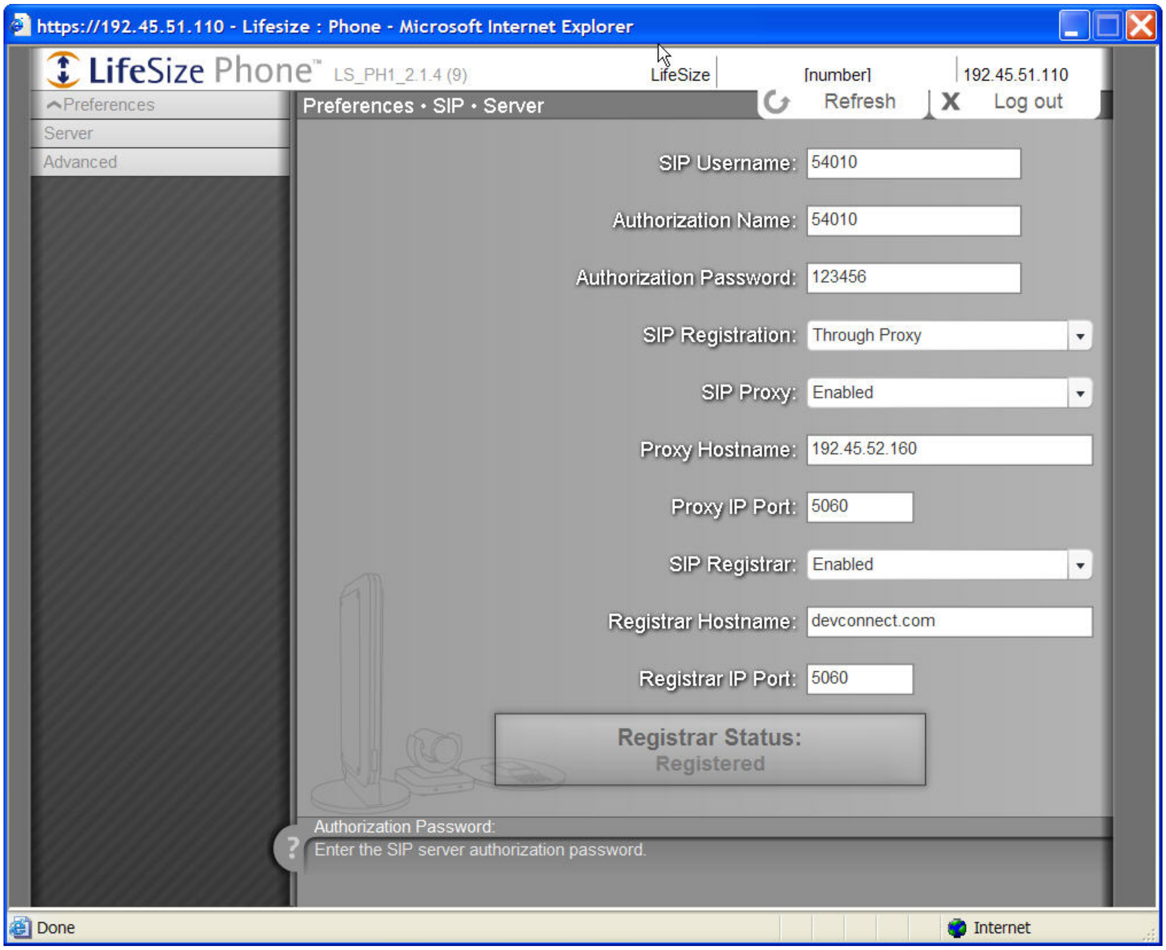
Step	Description
1.	Open a web browser, enter http://a.b.c.d for the URL, where a.b.c.d is the IP address of the LifeSize Phone. Click on **"click here"** to proceed to the configuration screens.  Allow Pop-Ups for 192.45.51.110 to automatically launch. Click here to launch. **Note:** Make sure Macromedia flash player is installed and the pop-up blocker is disabled.
2.	Enter the password and click on "**Submit**".  Enter Password **** Choose Language English (US)

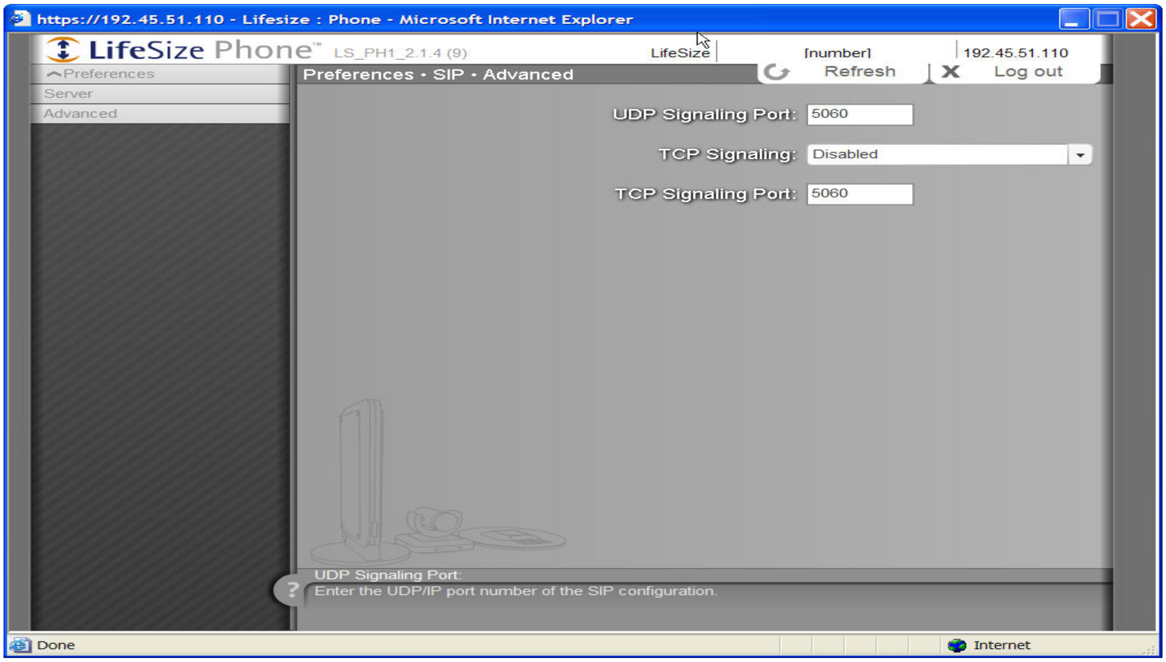
Step	Description
3.	From **Main Menu**, Click on "**Preferences**". 
4.	From the **Preferences** screen, click on **"Appearance"**. 

Step	Description
5.	On the **Preferences-Appearance** screen, enter the **System Name** and **Phone Number** as desired to display on the phone window.  System Name: LS1 Phone Number: [54010] LCD Contrast: 6
6.	On the **Preferred - Audio** screen, change the order of Preferences by changing the codecs position on the list if necessary. Codecs are listed in higher to lower Preferences order.  Audio Codec Order: AAC-LC, G.722.1 (32kb/s), G.722.1 (24kb/s), G.722, G.729, G.711 (μ-Law), G.711 (A-Law) Ring Tone Volume: 5 DTMF Tone Volume: 5 Audio Codec Order: Define the order of available audio codecs for the system to use to place calls.

Step	Description
7.	On **Preferences – Network** screen, enter the following fields: • **IP Address** of the phone • **Subnet Mask** • **Default Gateway**  DHCP: Disabled IP Address: 192.45.51.110 Subnet Mask: 255.255.255.0 Default Gateway: 192.45.51.1 Broadcast IP: Hostname: DNS Servers: Name Search Domains: Network Speed: Auto Physical Address: 00:13:fa:00:54:9a IP Address: Enter the static IP address of the device.

Step	Description
8.	On the **Preferences - Network QoS** enter **DiffServ Audio Priority.**  Preferences • Network QoS Network QoS: DiffServ DiffServ Audio Priority: 46 DiffServ Data Priority: 46 IntServ Audio Priority: 5 IntServ Data Priority: 5 IntServ ToS: None **Note:** LifeSize Phone does not support DiffServ QoS for signaling packets.
9.	From the **Preferences – SIP Menu,** click on **"Server".** 

Step	Description
10.	In the **Preferences – SIP – Server** screen, set the following values: • **SIP Username** to the value entered in **Primary Handle** field in Section 4, step 3 • **Authorization Name** to the value entered in **User ID** field in Section 4, step 3 • **Authorization Password** to the value entered in **Password** field in Section 4, step 3 • **SIP Registration** to the **“Through Proxy”** • **SIP Proxy** to the value entered in **“Enabled”** • **Proxy Hostname** to the value entered in **Host** field in Section 4, step 3 • **SIP Registrar** to the value entered in **“Enabled”** • **Registrar Hostname** to Domain name of the SIP Server  SIP Username: 54010 Authorization Name: 54010 Authorization Password: 123456 SIP Registration: Through Proxy SIP Proxy: Enabled Proxy Hostname: 192.45.52.160 Proxy IP Port: 5060 SIP Registrar: Enabled Registrar Hostname: devconnect.com Registrar IP Port: 5060 Registrar Status: Registered Authorization Password: Enter the SIP server authorization password.

Step	Description
11.	On the **Preferences-SIP-Advanced** screen, enter the **UDP Signaling Port** and disable the **TCP Signaling**.  Preferences · SIP · Advanced UDP Signaling Port: 5060 TCP Signaling: Disabled TCP Signaling Port: 5060 UDP Signaling Port: Enter the UDP/IP port number of the SIP configuration.

6. Interoperability Compliance Testing

The focus of the interoperability compliance testing was primarily on verifying call establishment on the LifeSize Phone. LifeSize Phone operations such as dialing methods (manual, re-dial, and phone book), hold, mute, and conference, and LifeSize Phone interactions with Avaya SIP Enablement Services (SES), Avaya Communication Manager, and Avaya SIP, H.323, and digital phones.

6.1. General Test Approach

The general test approach was to place calls to and from the LifeSize Phone and exercise basic phone operations on the LifeSize Phone. The main objectives were to verify that:

- The LifeSize Phone successfully registers with Avaya SES.
- The LifeSize Phone successfully establishes calls with Avaya SIP, H.323, and digital phones attached to Avaya SES or Avaya Communication Manager.
- The LifeSize Phone successfully establishes calls with PSTN phones through Avaya Communication Manager.
- The LifeSize Phone successfully handles concurrent calls on its two lines.
- The LifeSize Phone successfully negotiates the right codec.
- The LifeSize Phone successfully shuffles for VOIP calls.
- The LifeSize Phone successfully transmits DTMF during a call.
- The LifeSize Phone successfully handles layer-3 (DiffServ) QoS for Audio.

For serviceability testing, failures such as cable pulls and hardware resets were applied. For performance testing, a conference call involving two LifeSize Phones and two Avaya phones was formed as follows. A call was established between an Avaya phone and a LifeSize Phone. The LifeSize Phone then used its second line to establish a call with another LifeSize Phone, and bridged the two lines together, forming a 3-party conference. The second LifeSize Phone then used its second line to establish a call with another Avaya phone, and bridged its two lines together, effectively forming a 4-party conference.

6.2. Test Results

The test objectives of Section 6.1 were verified. For serviceability testing, the LifeSize Phone operated properly after recovering from failures such as cable disconnects, and resets of the LifeSize Phone, the Avaya SES server, and Avaya Communication Manager. For performance testing, the conference call was successfully maintained for approximately two hours.

The following observations were made during testing:

- LifeSize Phone does not support de-registration but when the phone is rebooted, it automatically re-registers with SES.
- LifeSize Phone shuffles properly but if the call is initiated by the LifeSize Phone, it requires an extra INVITE to complete the shuffle.
- LifeSize Phone operates with UDP as the SIP transport protocol.
- LifeSize Phone does not support hold and transfer features.

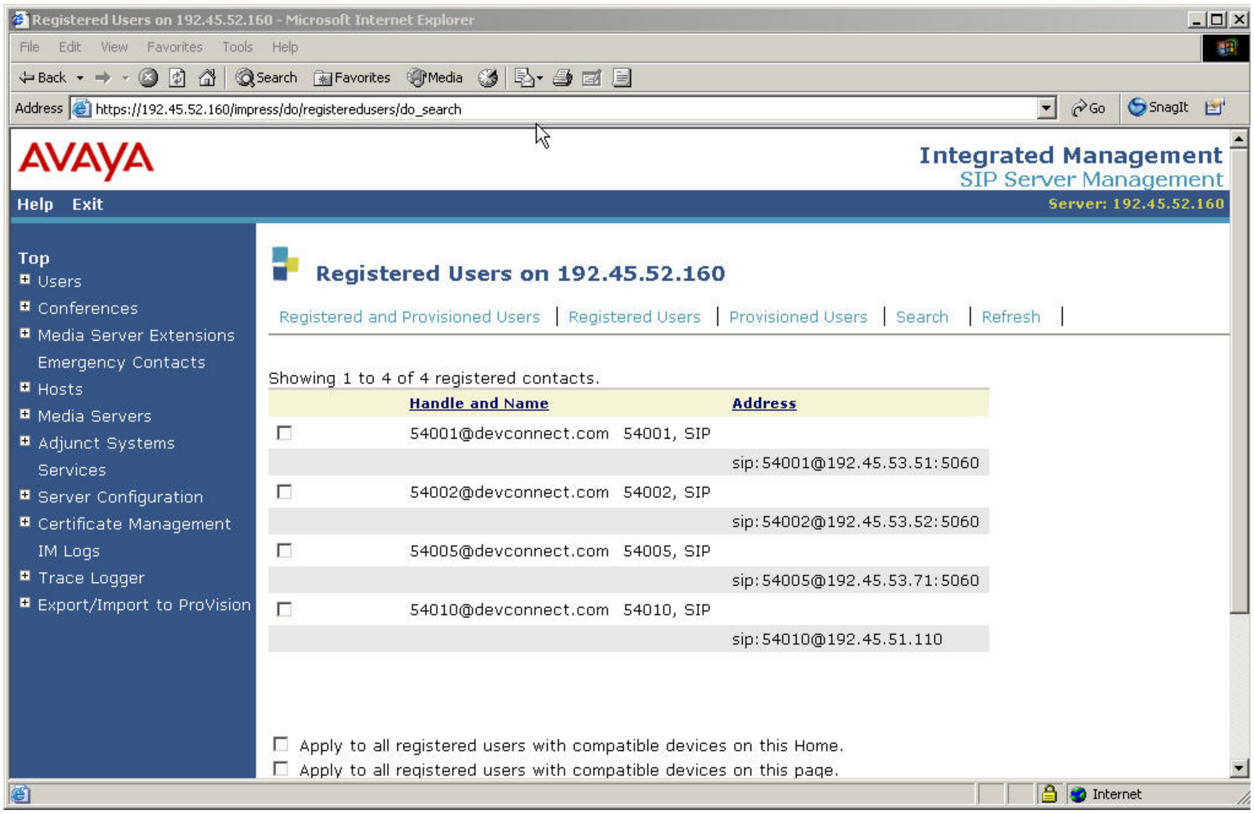
- LifeSize Phone supports 3-party conference; however, when the third party is added on by the LifeSize Phone, the second party is not put on hold.

LifeSize Communications expects to resolve the above observations in future releases. Contact LifeSize Communications (www.lifesize.com) for further updates.

7. Verification Steps

The following steps may be used to verify the configuration:

- Verify that the LifeSize Phone successfully register with the Avaya SES server by following the **Users -> Registered Users** links on the SES Administration Web Interface.



Showing 1 to 4 of 4 registered contacts.

	Handle and Name	Address
☐	54001@devconnect.com 54001, SIP	
		sip:54001@192.45.53.51:5060
☐	54002@devconnect.com 54002, SIP	
		sip:54002@192.45.53.52:5060
☐	54005@devconnect.com 54005, SIP	
		sip:54005@192.45.53.71:5060
☐	54010@devconnect.com 54010, SIP	
		sip:54010@192.45.51.110

☐ Apply to all registered users with compatible devices on this Home.
☐ Apply to all registered users with compatible devices on this page.

- Place calls to and from the LifeSize Phone and verify that the calls are successfully established with two-way talk path.

8. Support

For technical support on LifeSize Communications phones, consult the support pages at http://www.lifesize.com/support or contact LifeSize Communications technical support at:

- Phone: 1-877-LIFESIZE
- E-mail: support@lifesize.com

9. Conclusion

These Application Notes described a compliance-tested solution comprised of Avaya Communication Manager 3.1.2, Avaya SIP Enablement Services (SES) 3.1, and the LifeSize Communications Tabletop Conference Room Phone. LifeSize Phone is a SIP-based VoIP tabletop High Definition phone intended for use in conference rooms and similar environments. During compliance testing, the LifeSize Phone successfully registered with Avaya SES, placed and received calls to and from SIP and non-SIP telephones, and established 3-party conference calls.

10. Additional References

Product documentation for Avaya products may be found at http://support.avaya.com.

[1] *Administrator Guide for Avaya Communication Manager*, Issue 2.1, May 2006, Document Number 03-300509
[2] *Administration for Network Connectivity for Avaya Communication Manager*, Issue 11, February 2006, Document Number 555-233-504
[3] *SIP Support in Release 3.1 of Avaya Communication Manager*, Issue 6, February 2006, Document Number 555-245-206
[4] *Installing and Administering SIP Enablement Services R3.1.1*, Issue 2.0, August 2006, Document Number 03-600768

Product documentation for LifeSize Communications products may be found at http://www.lifesize.com.

[5] LifeSize *Product Information*, January 2006, Part No. 132-00003-002 Revision 2

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 Developer*Connection* Program at devconnect@avaya.com.