



Avaya Solution & Interoperability Test Lab

Application Notes for Polycom SoundStation IP 6000 and 7000 with Avaya IP Office – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Polycom SoundStation IP 6000 and 7000 to interoperate with Avaya IP Office. Polycom SoundStation IP 6000 and 7000 are SIP-based IP conference phones that integrate with Avaya IP Office as SIP endpoints.

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 configuration steps required for Polycom SoundStation IP 6000 and 7000 to interoperate with Avaya IP Office. Polycom SoundStation IP 6000 and 7000 are SIP-based IP conference phones that integrate with Avaya IP Office as SIP endpoints.

1.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing.

The feature testing included registration, basic call, display, hold/reconnect, conference, media shuffling, G.711, G.729, G.723, DTMF, do not disturb and call forwarding unconditional short code scenarios.

The serviceability testing focused on verifying the ability of Polycom SoundStation 6000 and 7000 to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet cable to the devices.

1.2. Support

Technical support on Polycom SoundStation IP 6000 and 7000 can be obtained through the following:

- **Phone:** (978) 292-5000
- **Web:** <http://www.polycom.com/support/index.html>

2. Reference Configuration

Figure 1 below shows the configuration used for the compliance testing.

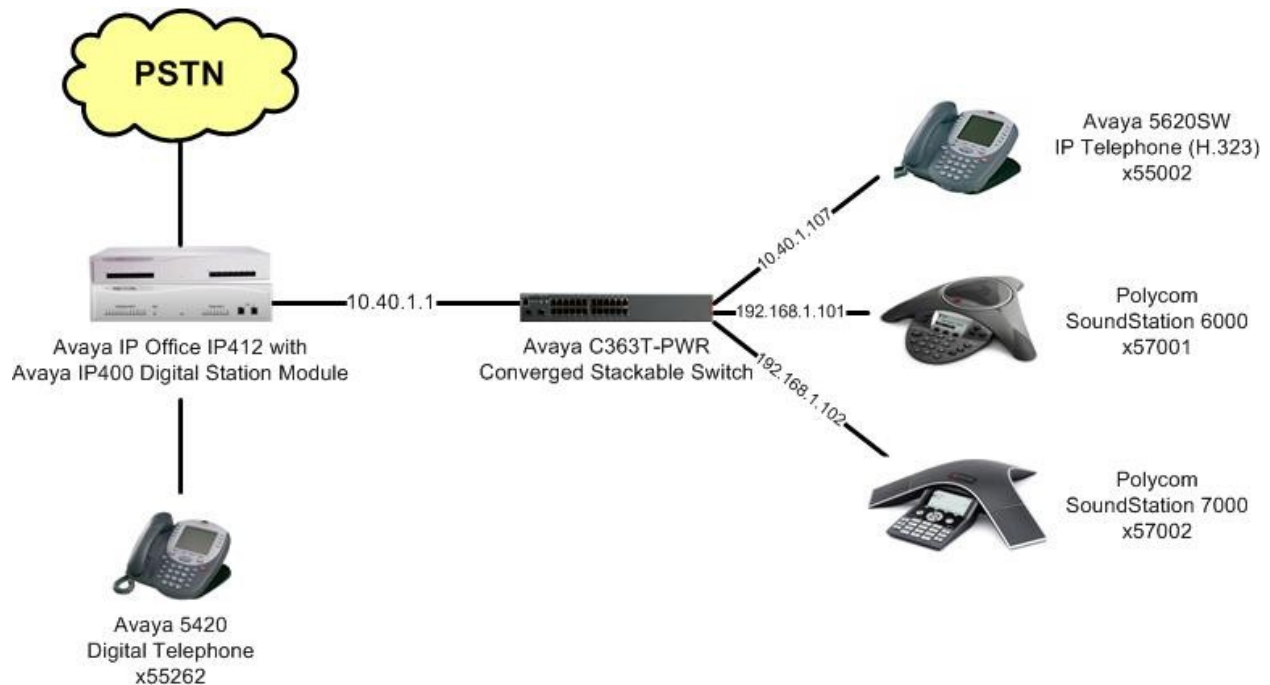


Figure 1: Polycom SoundStation IP 6000 and 7000 with Avaya IP Office

3. Equipment and Software Validated

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

Equipment	Software
Avaya IP Office IP412	5.0 (4)
Avaya 5610 IP Telephone	8.016
Avaya 5420 Digital Telephone	NA
Polycom SoundStation IP 6000	3.1.3.0439
Polycom SoundStation IP 7000	3.1.3.0439

4. Configure Avaya IP Office

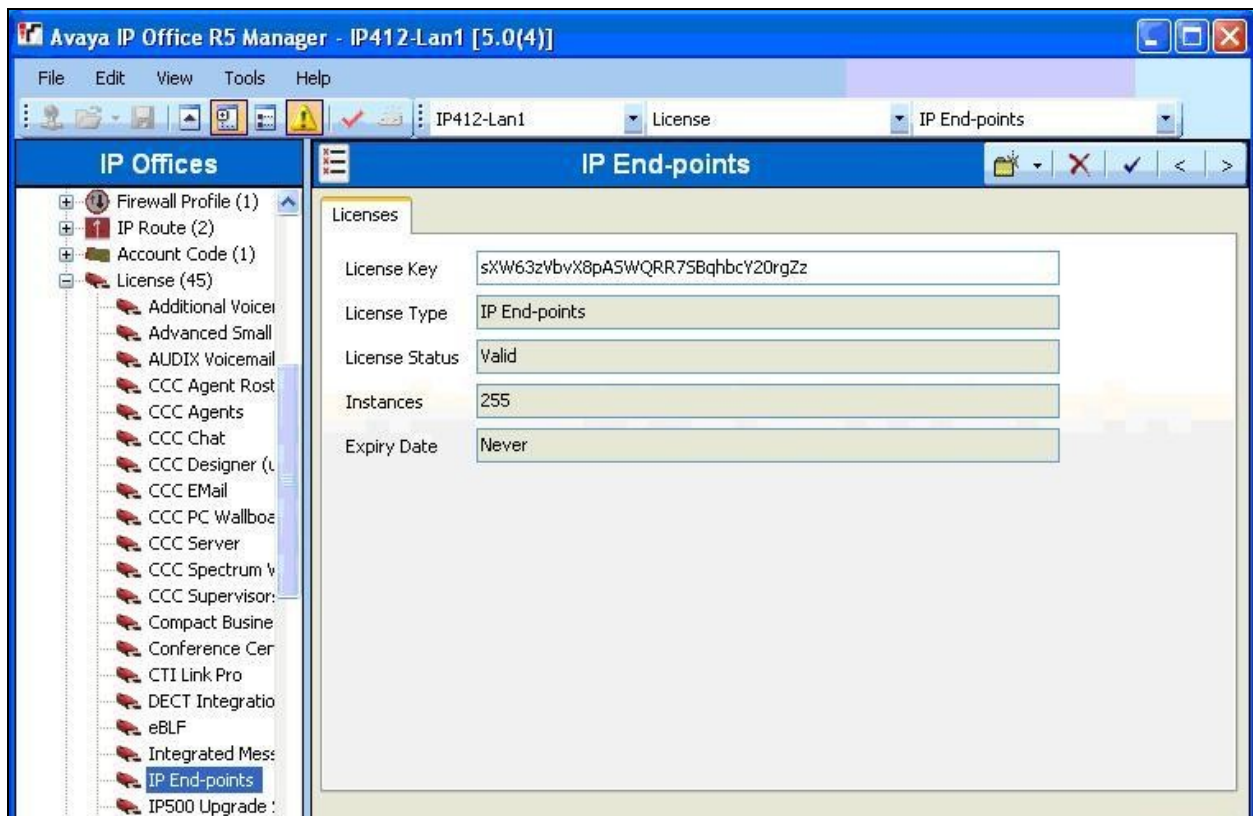
This section provides the procedures for configuring Avaya IP Office. The procedures include the following areas:

- Verify IP Office license
- Obtain LAN IP address
- Administer SIP registrar
- Administer SIP extensions
- Administer SIP users

4.1. Verify IP Office License

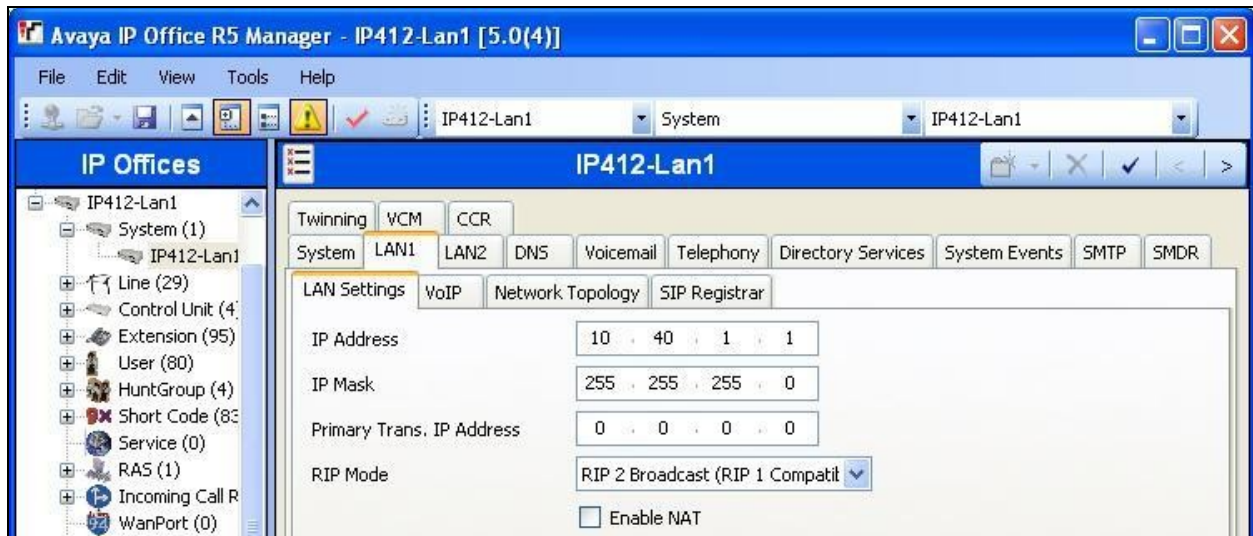
From a PC running the Avaya IP Office Manager application, select **Start > Programs > IP Office > Manager** to launch the Manager application. Select the proper IP Office system, and log in with the appropriate credentials.

The **Avaya IP Office Manager** screen is displayed. From the configuration tree in the left pane, select **License > IP End-points** to display the **IP End-points** screen in the right pane. Verify that the **License Status** is “Valid”.



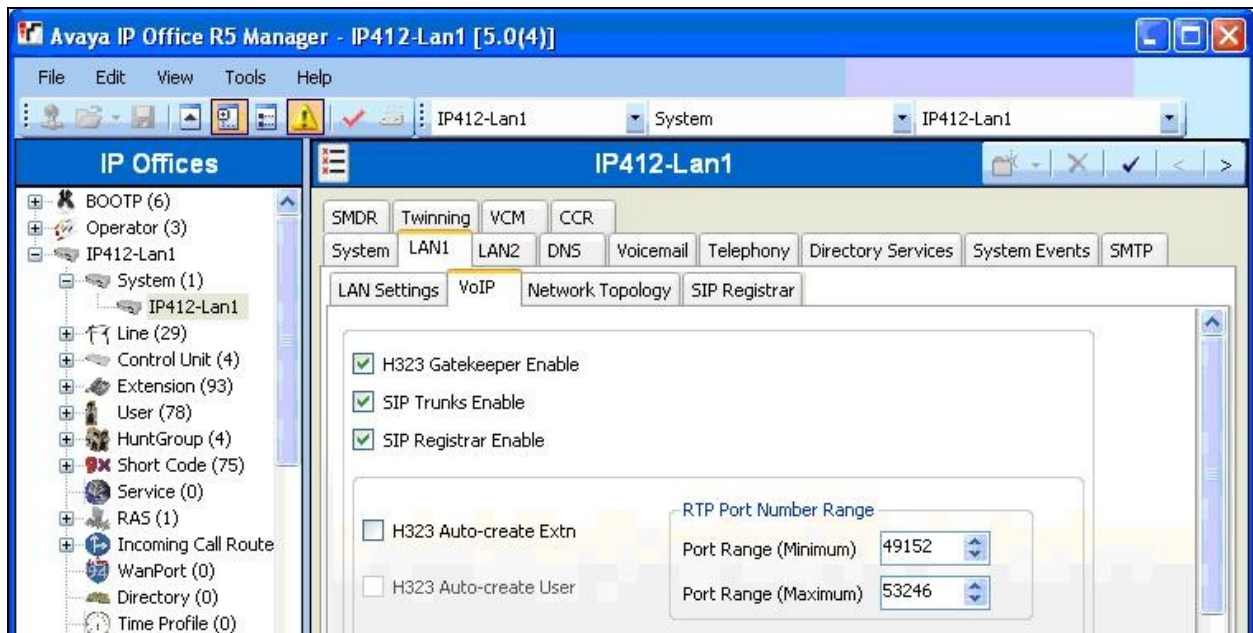
4.2. Obtain LAN IP Address

From the configuration tree in the left pane, select **System** to display the **IP412-Lan1** screen in the right pane. Select the **LAN1** tab, followed by the **LAN Settings** sub-tab in the right pane. Make a note of the **IP Address**, which will be used later to configure Polycom. Note that IP Office can support SIP extensions on the LAN1 and/or LAN2 interfaces, and the compliance testing used the LAN1 interface.

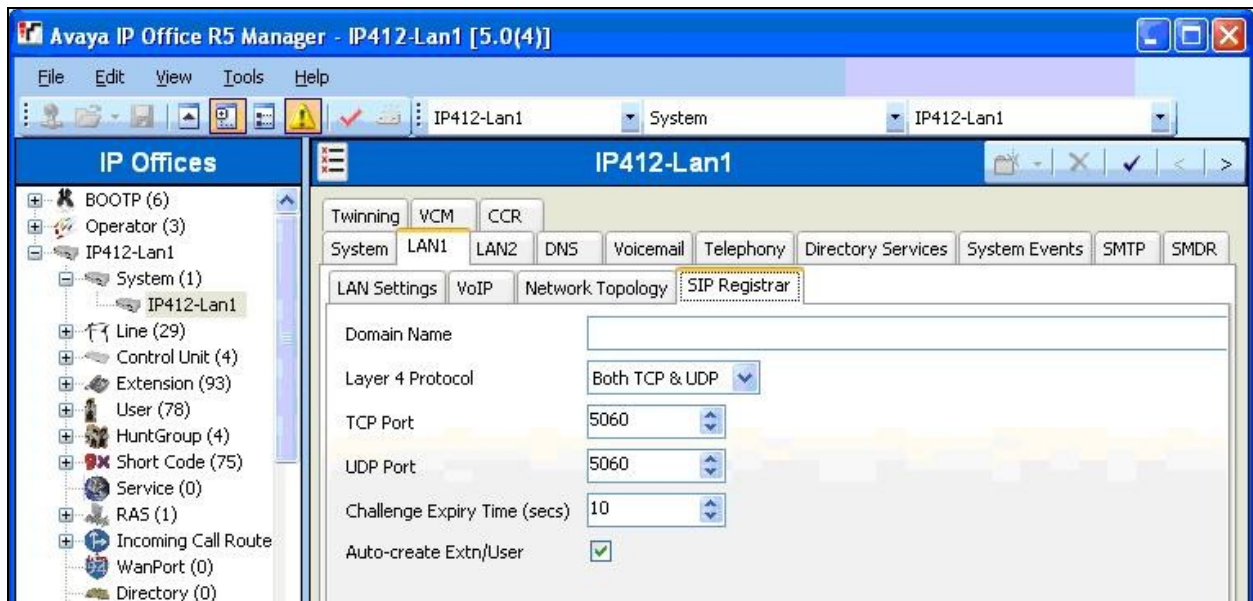


4.3. Administer SIP Registrar

Select the **VoIP** sub-tab. Make certain that **SIP Registrar Enable** is checked, as shown below.

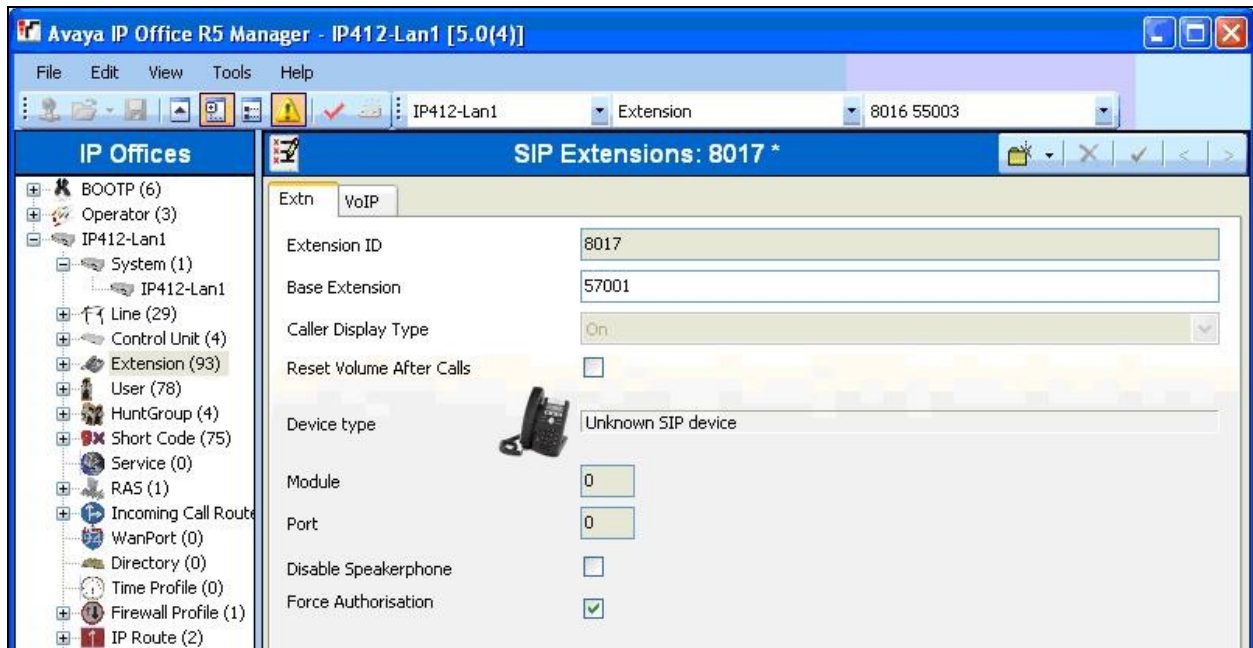


Select the **SIP Registrar** sub-tab, and enter a valid **Domain Name** for SIP endpoints to use for registration with IP Office. In the compliance testing, the **Domain Name** was left blank, so the SIP endpoints used the LAN IP address for registration.



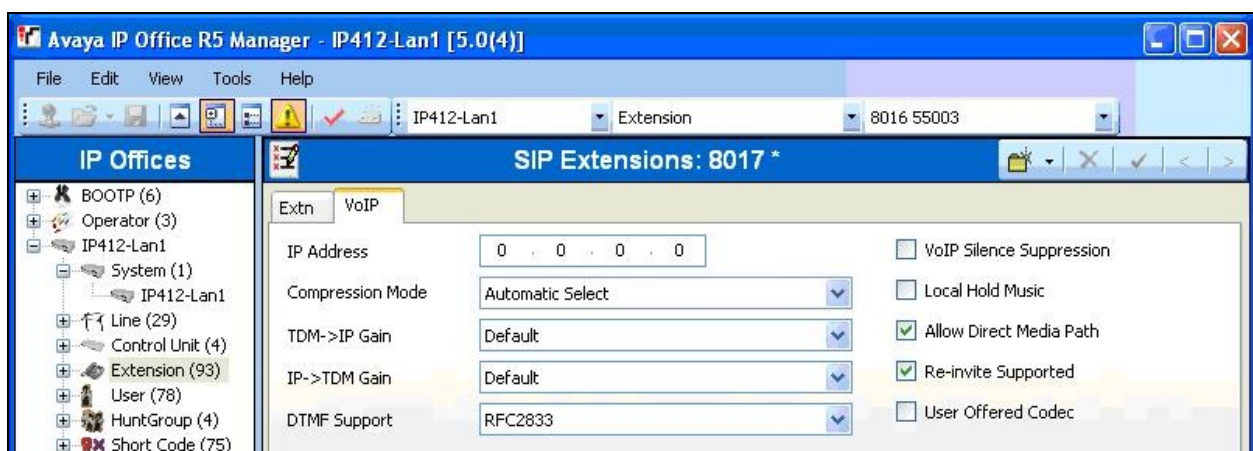
4.4. Administer SIP Extensions

From the configuration tree in the left pane, right-click on **Extension**, and select **New > SIP Extension** from the pop-up list to add a new SIP extension. Enter the desired digits for **Base Extension**, and retain the default check in the **Force Authorisation** field as shown below.



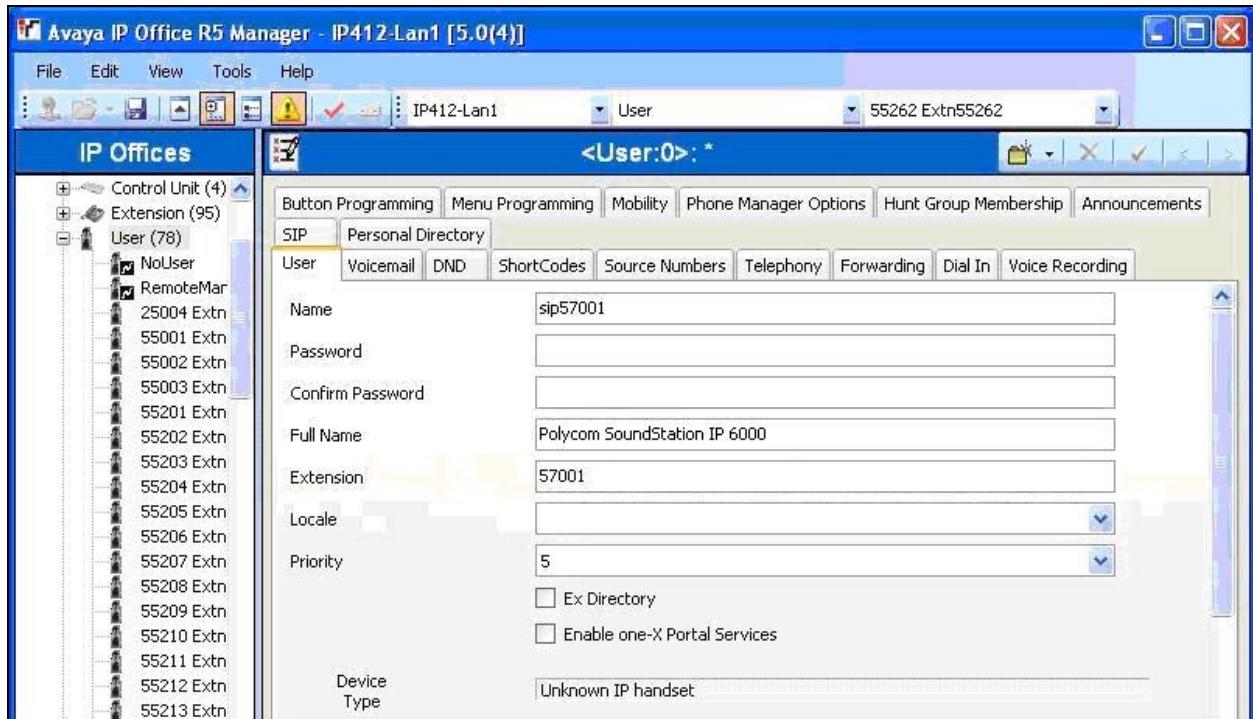
Select the **VoIP** tab, and retain the default values in all fields.

Repeat this section to add a new SIP extension for each Polycom SoundStation IP 6000 and 7000. In the compliance testing, two SIP extensions with base extensions of “57001” and “57002” were created.

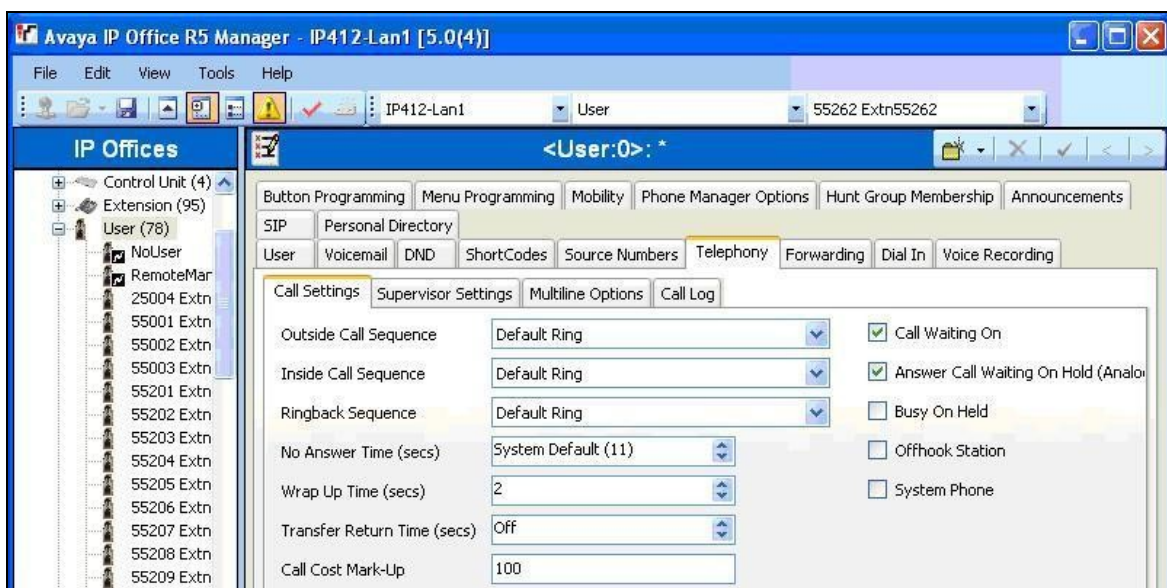


4.5. Administer SIP Users

From the configuration tree in the left pane, right-click on **User**, and select **New** from the pop-up list. Enter desired values for **Name** and **Full Name**. For **Extension**, enter the first SIP extension from **Section 4.4**.

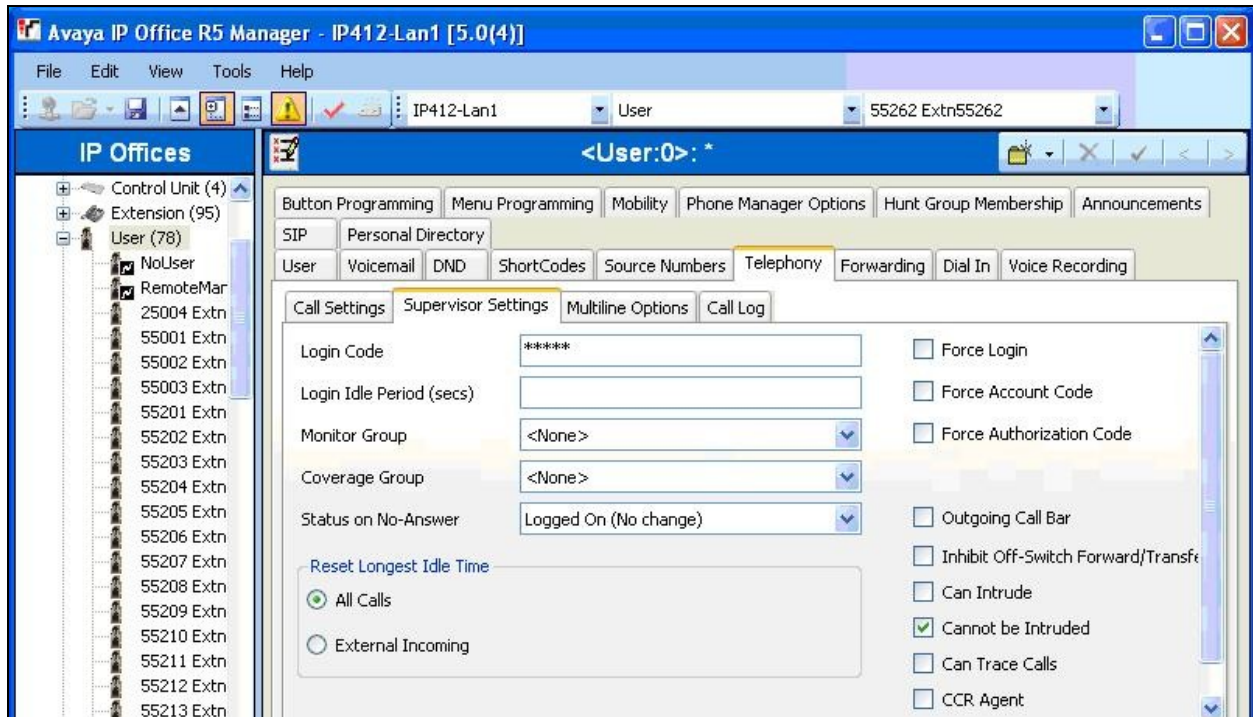


Select the **Telephony** tab, followed by the **Call Settings** sub-tab. Check the **Call Waiting On** field, as shown below.



Select the **Supervisor Settings** tab, and enter a desired **Login Code**.

Repeat this section to add a new user for each SIP extension from **Section 4.4**. In the compliance testing, two users with names of “sip57001” and “sip57002” were created.



5. Configure Polycom SoundStation IP 6000 and 7000

This section provides the procedures for configuring the Polycom SoundStation IP 6000 and 7000. The procedures include the following areas:

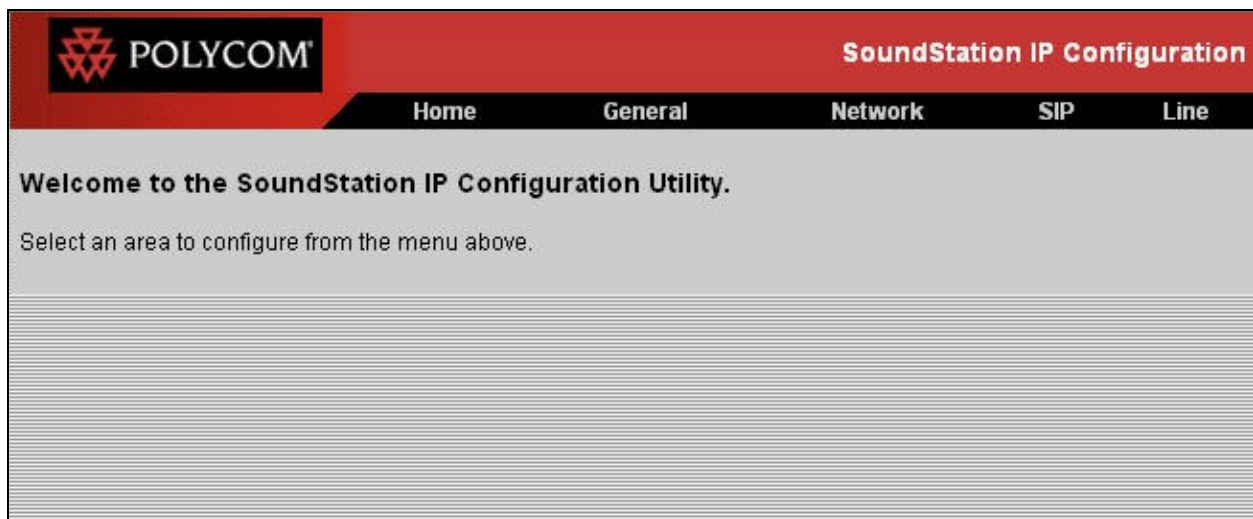
- Launch web interface
- Administer SIP
- Administer line

Prior to the configuration, follow the procedures in [4] to manually set or obtain the IP address of the SoundStation 6000 and 7000. Note that the interface and procedures for configuring the SoundStation IP 6000 and 7000 are exactly the same, therefore follow this section for both SoundStation IP 6000 and 7000.

5.1. Launch Web Interface

Access the SoundStation IP web-based interface by using the URL “http://ip-address” in an Internet browser window, where “ip-address” is the IP address of the SoundStation IP 6000 or 7000. Log in with the appropriate credentials.

The **SoundStation IP Configuration** screen is displayed, as shown below. Select **SIP**.



5.2. Administer SIP

The **SIP Configuration Parameters** screen is displayed. In the **Outbound Proxy** and **Server 1** sections, enter the following values for the specified fields, and retain the default values for the remaining fields. Scroll down the screen and click the **Submit** button (not shown below) associated with the **Servers** section.

- **Address:** The LAN IP address from **Section 4.2**.
- **Transport:** “UDPonly”

SIP Configuration Parameters:	
Servers	Local Settings
Servers	
Outbound Proxy	
Address	10.40.1.1
Port	5060
Transport	UDPonly
Server 1	
Address	10.40.1.1
Port	5060
Transport	UDPonly
Expires	
Register	1
Retry Time Out	0
Retry Max Count	0
Line Seize Time Out	30
Server 2	

Scroll down the screen to display the **Local Settings** section. Modify **Digitmap** to match the dial plan configuration on Avaya IP Office. In the compliance testing, the value “*x.T|#x.T|x.T” was used to allow for dial strings prefixed with digits, “*”, or “#”. For more information on local settings provisioning, see [4].

Retain the default values for the remaining fields, and click **Submit**.

Local Settings	
Local SIP Port	<input type="text"/>
Calls Per Line Key	<input type="text"/>
New SDP Type	<input type="radio"/> Enabled <input checked="" type="radio"/> Disabled
LCS Support	<input type="radio"/> Enabled <input checked="" type="radio"/> Disabled
Non Standard Line Seize	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
Digitmap	*x.T #x.T x.T <input type="button" value="up"/> <input type="button" value="down"/>
Digitmap Timeout	3 3 3 3 3
Remove End-Of-Dial Marker	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
Digitmap Impossible Match	<input type="text"/>
top	<input type="button" value="Submit"/>

5.3. Administer Line

Select **Line** from the top menu to display the **Line Parameters** screen. In the **Identification** section, enter the following values for the specified fields, and retain the default values for the remaining fields. Scroll down the screen and click **Submit** (not shown below).

- **Address:** The SIP base extension from **Section 4.4**.
- **Auth User ID:** The SIP user name from **Section 4.5**.
- **Auth Password:** The SIP user login code from **Section 4.5**.
- **Label:** A desired string for the phone display.

The screenshot shows the Polycom SoundStation IP Configuration web interface. The top navigation bar includes the Polycom logo and the title "SoundStation IP Configuration". Below the navigation bar, the "Line" tab is selected. The main content area is titled "Line Parameters:" and shows "Line 1" selected. The configuration form for Line 1 is displayed, with the "Identification" section highlighted. The fields in the Identification section are: Display Name (empty), Address (57001), Auth User ID (sip57001), Auth Password (masked with dots), Label (x57001), Type (radio buttons for Private and Shared, with Private selected), Third Party Name (empty), Num Line Keys (empty), and Calls Per Line (empty). The "Server 1" section is partially visible at the bottom.

Line 1	
Identification	
Display Name	
Address	57001
Auth User ID	sip57001
Auth Password	•••••
Label	x57001
Type	<input checked="" type="radio"/> Private <input type="radio"/> Shared
Third Party Name	
Num Line Keys	
Calls Per Line	
Server 1	

6. General Test Approach and Test Results

The feature test cases were performed manually. Calls were manually established between Polycom SoundStation IP 6000 and/or 7000 with the Avaya H.323, Avaya Digital, or PSTN endpoints, and call controls such as hold and conference were performed from the Polycom SoundStation IP 6000 and/or 7000.

The serviceability test cases were performed manually by disconnecting and reconnecting the LAN cables to the Polycom SoundStation IP 6000 and 7000.

All test cases were executed.

The following are the observations from the compliance testing:

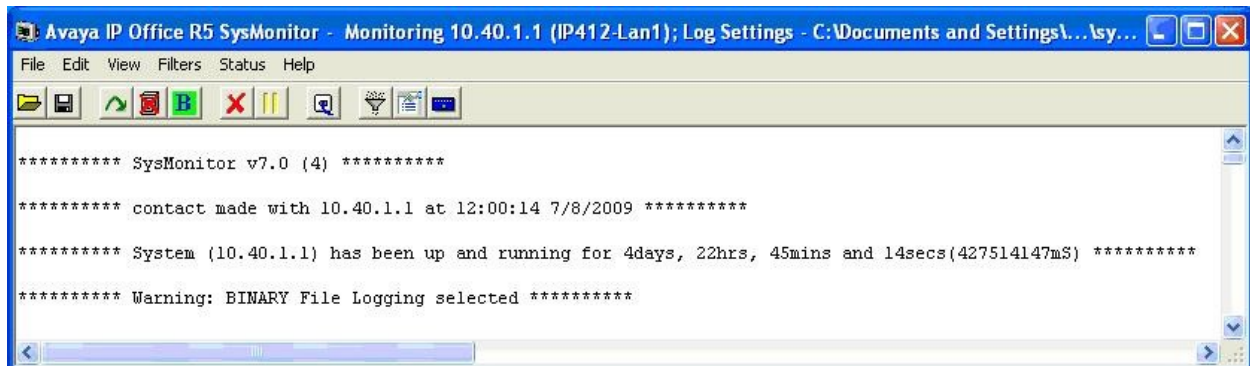
- When the LAN cable is disconnected from an active Polycom SoundStation IP 6000 for more than 30 seconds and then reconnected, the SoundStation IP 6000 becomes inoperable until a manual reboot is performed.
- Avaya IP Office can successfully activate Do Not Disturb, but sends back 503 Service Unavailable for the activation request.

7. Verification Steps

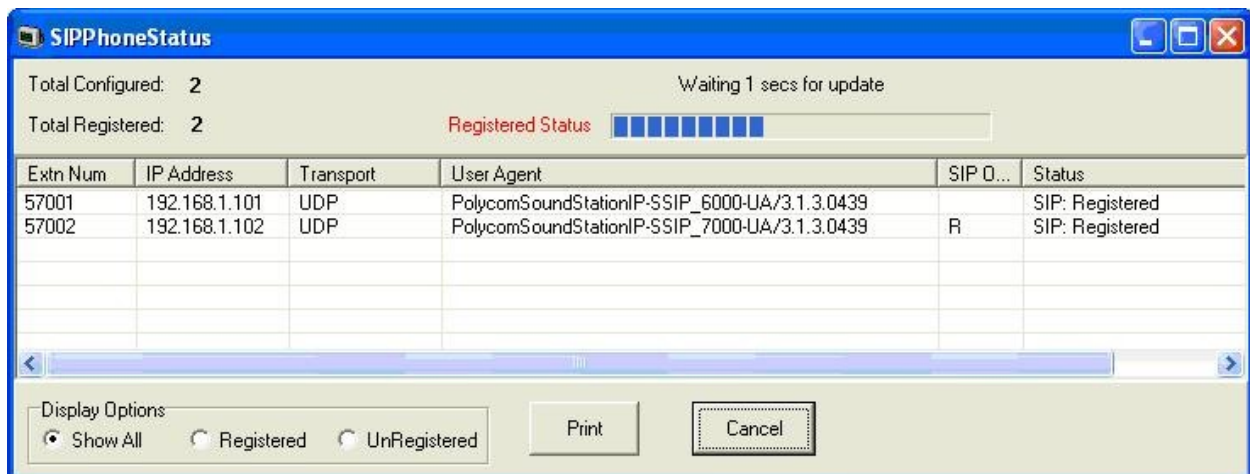
This section provides the tests that can be performed to verify proper configuration of Avaya IP Office and Polycom SoundStation IP 6000 and 7000.

7.1. Verify Avaya IP Office

From a PC running the Avaya IP Office Monitor application, select **Start > Programs > IP Office > Monitor** to launch the application. The **Avaya IP Office R5 SysMonitor** screen is displayed, as shown below. Select **Status > SIP Phone Status** from the top menu.



The **SIPPhoneStatus** screen is displayed. Verify that there is an entry for each SIP extension from **Section 4.4**, and that the **User Agent** is “PolycomSoundStationIP”, and the **Status** is “SIP: Registered”, as shown below.



8. Conclusion

These Application Notes describe the configuration steps required for Polycom SoundStation IP 6000 and 7000 to successfully interoperate with Avaya IP Office. All feature and serviceability test cases were completed with observations noted in **Section 6**.

9. Additional References

This section references the product documentation relevant to these Application Notes.

1. *IP Office 5.0 Documentation CD*, August 2009, available at <http://support.avaya.com>.
2. *User Guide for the Polycom SoundStation IP 6000 Phone*, SIP 3.1, August 2008, available at <http://polycom.com/support>.
3. *User Guide for the Polycom SoundStation IP 7000 Phone*, SIP 3.1.1, November 2008, available at <http://polycom.com/support>.
4. *Administrator's Guide for the Polycom SoundPoint IP / SoundStation IP Family*, SIP 3.1, August 2008, available at <http://polycom.com/support>.

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