

Application Notes for Polycom SpectraLink 8440 SIP Telephone version 4.0.0.0282 with Avaya Communication Server 1000 Release 7.5 – Issue 1.1

Abstract

These Application Notes describe a solution comprised of Avaya Communication Server 1000 SIP Line Release 7.5 and Polycom SpectraLink 8440 SIP telephone. During the compliance testing, the Polycom SpectraLink 8440 was able to register as a SIP client endpoint with the Communication Server 1000 SIP Line gateway. The Polycom SpectraLink 8440 telephone was able to place and receive calls from the Communication Server 1000 Release 7.5 non-SIP and SIP Line clients. The compliance tests focused on basic telephone features.

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 provide detail configurations of Avaya Communication Server 1000 SIP Line release 7.5 (hereafter referred to as CS 1000) and the Polycom SpectraLink 8440 SIP telephone Version 4.0.0.0282 used during the compliance testing. The Polycom SpectraLink 8440 was tested with non-SIP and SIP clients using the CS1000 SIP line release 7.5. All the applicable telephony feature test cases of release 7.5 SIP line were executed on the Polycom SpectraLink 8440 , where applicable, to ensure that the interoperability with CS 1000.

2. General Test Approach and Test Results

The general test approach was to have the Polycom SpectraLink 8440 telephone to register to the CS1000 SIP line gateway successfully. From the CS1000 telephone clients/users to place a call to and from the Polycom SpectraLink 8440 telephone and to exercise other telephony features such as busy, hold, DTMF, MWI and codec negotiation

2.1. Interoperability Compliance Testing

The focus of this testing was to verify that the Polycom SpectraLink 8440 SIP telephone was able to interoperate with the CS 1000 SIP line system. The following areas were tested:

- Registration of the Polycom SpectraLink 8440 SIP telephone to the CS1000 SIP Line Gateway.
- Call establishment of Polycom SpectraLink 8440 with CS1000 SIP and non-SIP telephones.
- Telephony features: Basic calls, conference, transfer, DTMF (dual tone multi frequency) RFC2833 and INBAND transmission, voicemail with Message Waiting Indication (MWI) notification, busy, hold, speed dial, group call pickup, call waiting, ring again busy/no answer, multiple appearances Directory Number.
- PSTN calls over PRI trunk.
- Codec negotiation G.711 and G.729.

2.2. Test Results

The objectives outlined in the **Section 2.1** were verified. The following observations were made during the compliance testing:

- Avaya has not performed audio performance testing or reviewed the Polycom SpectraLink 8440 compliance to required industry standards.
- Polycom SpectraLink 8440 does not support DTMF via SIP INFO. DTMF defaults as INBAND. When using RFC2833 DTMF set these fields *tone.dtmf.rfc2833Control=*"1" and *tone.dtmf.rfc2833Payload=101* in the config file (sip.cfg).
- The Polycom 8440 Local Forward Busy feature which is set on the phone locally can be enabled but it will be not used for the busy call since when the 8440 phone is in busy status the Server Call Forward Busy feature of CS1000 SIPLine will take place before it

can be executed by the phone. It is recommended to set the call forward busy in the CS 1000 SIPLine server.

- It is highly recommended to disable the media security on the Call Server to avoid some unexpected behaviors such as one way audio from a call made from PSTN over a PRI trunk.
- There is a limitation for the local conference on the Polycom 8440 phone with Avaya UNIStim phone. When Polycom 8440 acts as the moderator of the conference and conferences with two Avaya UNIStim phones, the conference is successfully opened with 3-way audio. However, after the Polycom 8440 phone disconnects, the two Avaya UNIStim phones remain connected but have no speech path in between them. This issue doesn't happen with Avaya SIP 1140 phones and with other SIP phones.

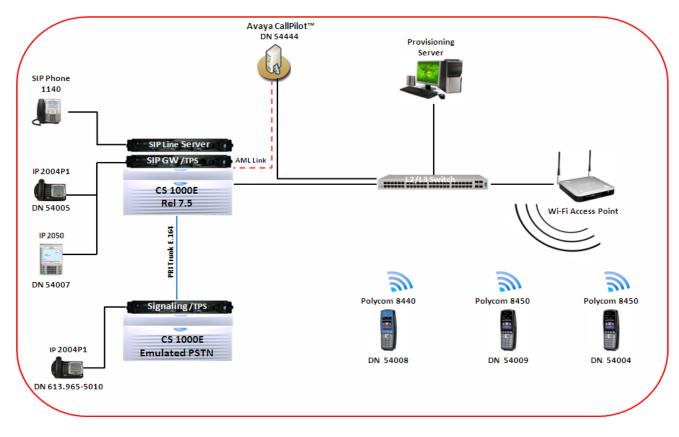
2.3. Support

For technical support for the Polycom SpectraLink 8440 SIP endpoints, please contact PolycomInc technical support as shown below:

1.800.POLYCOM or +1.925.924.6000 www.polycom.com

3. Reference Configuration

Figure 1 illustrates the test configuration used during the compliance testing between the Avaya CS1000 and the Polycom SpectraLink 8440.



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Figure 1: Network Configuration Diagram

4. Equipment and Software Validated

Equipment	Software Version
Avaya CS1000E	Call Server (CPPM): 7.50Q
	Signaling Server (CPPM): 7.50.17
Avaya CallPilot [™] Messaging System	5.0.1
Avaya IP Soft Phone 2050	3.04.0003
Avaya IP Phone 1140	0625C6O
Avaya IP Phone 2004P2	0692D93
Avaya IP Phone 2002P2	0604DC5
Avaya SIP 1140	02.02.21.00
Polycom SpectraLink 8440	4.0.0.0282
Polycom SpectraLink 8450	4.0.0.0282
Provisioning Server OS	Windows Vista x86

The following equipment and software was used during the lab testing:

5. Configure Avaya CS 1000 - SIP LINE

This section describes the steps to configure the Avaya CS1000 SIP Line using CS 1000 Element Manager. A command line interface (CLI) option is available to provision the SIP Line application on the CS 1000 system. For detailed information on how to configure and administer the CS 1000 SIP Line, please refer to the **Section 9 [1]**.

The following is the summary of tasks needs to be done for configuring the CS 1000 SIP Line:

- Log in to Unified Communications Management (UCM) and Element Manager (EM).
- Enable SIP Line Service and Configure the Root Domain.
- Create SIP Line Telephony Node.
- Create D-Channel for SIP Line.
- Create an Application Module Link (AML).
- Create a Value Added Server (VAS).
- Create a Virtual Trunk Zone.
- Create a Route Data Block (RDB).
- Create SIP Line Virtual Trunks.
- Create SIP Line phones.

5.1. Prerequisite

This document assumes that the CS1000 SIP Line server has been:

- Installed with CS 1000 Release 7.5 Linux Base.

- Joined CS 1000 Release 7.5 Security Domain.
- Deployed with SIP Line Application.

The following packages need to be enabled in the key code. If any of these features have not been enabled, please contact your Avaya account team or Avaya technical support at <u>http://www.avaya.com</u>.

Package Mnemonic	Package #	Descriptions	Package Type	Applicable market
SIP_LINES	417	SIP Line Service package	New package	Global
FFC	139	Flexible Feature Codes		
SIPL_AVAYA	415	Avaya SIP Line package	Existing package	Global
SIPL_3RDPARTY	416	Third-Party SIP Line Package	Existing package	Global

5.2. Log in to Unified Communications Management (UCM) and Element Manager (EM)

Use the Microsoft Internet Explorer browser to launch CS 1000 UCM web portal at http://<IP Address or FQDN> where <IP address or FQDN> is the UCM Framework IP address or FQDN for UCM server.

Log in with the username/password which was defined during the primary security server configuration, the UCM home page appears as shown in the **Figure 2** below.

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Figure 2: The UCM Home Page of CS 1000 Release 7.5

On the UCM home page, under the **Element Name** column, click on the EM name of CS 1000 system that needs to be configured, in this sample that is **cpppm3**. The CS 1000 Element Manager page appears as shown in **Figure 3** below.

Αναγα	CS1000 Element Manager	Help Logout
- UCM Network Services - Home - Links	Managing: <u>Utername:</u> admin System Overview	
- Virtual Terminals - System + Alarms - Maintenance + Core Equipment	System Overview	_
- Peripheral Equipment + IP Network + Interfaces - Engineered Values	IP Address: 10.10.97.78 Type: Avaya Communication Server 1000E CPPM Linux Version: 4121	
+ Emergency Services + Geographic Redundancy + Software - Customers	Release: 750 Q +	
- Routes and Trunks - Routes and Trunks - D-Channels - Digital Trunk Interface		
Dialing and Numbering Plans Electronic Switched Network Flexible Code Restriction Incoming Digit Translation		
- Phones - Templates - Reports - Views		
- Lists - Properties - Migration - Tools		
+ Backup and Restore - Date and Time + Logs and reports - Security		
+ Passwords + Policies + Login Options		
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Figure 3: CS 1000 Release 7.5 EM Home Page

5.3. Enable SIP Line Service in the Customer Data Block

On the EM page, navigate to **Customers** on the left column menu; select the customer number to be enabled with SIP Line Service (not shown).

- Enable SIP Line Service by clicking on the SIP Line Service check box.
- Enter the prefix number in the User agent DN prefix text box as shown in Figure 4.

Αναγα	CS1000 Element Manager	Help Logout
- UCM Network Services - Home - Links - Virtual Terminals - System + Alarms	Managing: <u>488-10.97.78</u> Username: admin <u>Customers</u> » Customer 00 » <u>Customer Details</u> » SIP Line Service SIP Line Service	
- Maintenance + Core Equipment - Peripheral Equipment + IP Network + Interfaces - Engineered Values + Emergency Services	 ✓ SIP Line Service User agent DN prefix 26 Optional features: ✓ Nortel Multimedia 	
Geographic Redundancy Software Soctivity Passwords Policies Login Options	"Required Value	Save Cancel
	Copyright © 2002-2011 Avaya Inc. All rights reserved.	

Figure 4: SIP Line Service in Customers Data Block

5.4. Add a new SIP Line Telephony Node

On the EM page, navigate to menu System \rightarrow IP Network \rightarrow Nodes: Servers, Media Cards. Click Add to add a new SIP Line Node to the IP Telephony Nodes. The new IP Telephony Node page appears as shown in Figure 5.

Enter the information as shown below:

- Node ID text box: 512 -> this is the node ID of SIP Line server.
- Call Server IP Address text box: 10.10.97.78.
- Node IP Address text box: 10.10.97.187 -> this is the IP address that SIP endpoint uses to register to.
- Subnet Mask text box: 255.255.255.192.
- Embedded LAN (ELAN) Gateway IP Address text box: 10.10.97.66.
- Embedded LAN (ELAN) Subnet Mask text box: 255.255.255.192.
- Check **SIP** Line check box to enable SIP Line for this Node.

Αναγα	CS1000 Element M	lanager				Help Logout				
- UCM Network Services			n New ID Telephone Made							
- Home		System » IP Network » IP Telephony Nodes » New IP Telephony Node								
- Links	New IP Telephony N	w IP Telephony Node								
- Virtual Terminals	Step 1: Define the new Node	o 1: Define the new Node and its services.								
- System	You will also require a	You will also require pre-configured servers with appropriate application software already deployed to host the selected services.								
+ Alarms										
- Maintenance										
+ Core Equipment	Node ID:	510	* (0-9999)							
 Peripheral Equipment IP Network 	Node ID:	512	-(0-9999)							
- Nodes: Servers. Media Cards	Call server IP address:	10 10 97 78	* TLAN address type:							
- Maintenance and Reports	Gan Schern address.	10.10.01.10		-						
- Media Gateways				IPv4 and IPv6						
- Zones										
- Host and Route Tables	Embedded LAN (ELAN)		Telephony LAN (TLAN)							
 Network Address Translation (NA[*]) 										
- QoS Thresholds	Gateway IP address:	10.10.97.65	* Node IPv4 address:	10.10.97.187 *						
 Personal Directories Unicode Name Directory 					=					
+ Interfaces	Subnet mask:	255.255.255.192	* Subnet mask:	255.255.255.192 *						
- Engineered Values										
+ Emergency Services			Node IPv6 address:							
+ Geographic Redundancy										
+ Software										
- Customers	Applications:	SIP Line								
- Routes and Trunks		UNIStim Line Te	erminal Proxy Server (LTPS)							
- Routes and Trunks			iteway (SIPGw, H323Gw)							
- D-Channels										
– Digital Trunk Interface		-								
- Dialing and Numbering Plans - Electronic Switched Network		Presence Publis	sher		Ψ.					
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- Phones										
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- Reports										
- Views										
- Lists										
- Properties										
- Migration	Copyright © 2002-2011 Avaya In	All rights reserved								
	Copyright @ 2002-2011 AVaya In	. All rights reserved.								

Figure 5: Adding a New IP Telephony Node

- Click on the **Next** button to go to next page. The page, New IP Telephony Node with Node ID, will appear as shown in **Figure 6**.
- On the Select to Add drop down menu list, select the desired server to add to the node.
- Click the Add button
- Select the check box next to the newly added server, and click **Make Leader** (not shown).

Αναγα	CS1000 Element Manager	Help Logout
- UCM Network Services - Home - Links - Mitual Terminals - System + Alarms - Maintenance + Core Equipment	Managing: #1.10.97.78 Username: admin System > IP Network > IP Telephony Nodes > New IP Telephony Node New IP Telephony Node (ID:513) Step 2: Associate required signaling servers for SIP Line services. In order to appear in the list below, servers must already be defined within ECM, should not be part of any other IP telephony node and deployed application(s) on the server(s) should match the service(s) selected for this node.	
- Peripheral Equipment - IP Network - Nodes: Servers, Media Cards	Select to add Add Remove Make Leader Print Refresh	
- Maintenance and Reports	Hostname A Type Deployed Applications ELAN IP TLAN IPv4 TLAN IPv6 Role	
- Media Gateways - Zones - Host and Route Tables - Network Address Translation (NAT - QoS Thresholds - Personal Directories - Unicode Name Directory Interfaces - Engineered Values * Emergency Services * Geographic Redundancy * Software - Customers - Routes and Trunks - D-Channels - Digital Trunk Interface - Diating and Numbering Plans - Electronic Switched Network - Flexible Code Restriction	Select from the list above and click Add to associate servers with this node. Selected servers must have identical application deployments.	
- Incoming Digit Translation - Phones - Templates - Reports - Mews - Lists - Properties - Migration	< Back Next > Cancel	
< III >	Copyright © 2002-2011 Avaya Inc. All rights reserved.	

Figure 6: Adding a New IP Telephony Node (cont)

- Click on the **Next** button to go to next page. The **SIP Line Configuration Detail** page appears as shown in **Figure 7**.
- Enter SIP Line domain name in **SIP Domain name** text box, for example **sipl75.com**.

Αναγα	CS1000 Element Ma	anager		Help Logout
- UCM Network Services - Home - Uriks - Virtual Terminals - System + Narms - Maintenance + Core Equipment - Perioheral Equipment	Node ID: 512 - SIP Line	me:admin 19 Telephony Nodes > Node Details > e Configuration Details Settings SIP Line Gateway Servic Line Gateway Application	<u>ê</u>	_
- IP Network	General		Virtual Trunk Network Health Monitor	
 <u>Nodes: Servers, Media Cards</u> Maintenance and Reports Media Gateways 	SIP domain name:	sipl75.com *	Monitor IP addresses (listed below) Information will be captured for the IP addresses listed	
– Zones – Host and Route Tables – Network Address Translation (NA	SLG endpoint name:	sipline	Monitor IP: Add	
 – QoS Thresholds ≡ – Personal Directories 	SLG Group ID:		Monitor addresses:	
- Unicode Name Directory + Interfaces - Engineered Values	SLG Local Sip port		Remove	
+ Emergency Services + Geographic Redundancy + Software	SIP Line Gateway Settings			
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- Dialing and Numbering Plans - Electronic Switched Network - Flexible Code Restriction	* Required Value.		de on this page will NOT be Save Cancel	
- Incoming Digit Translation				
– Templates – Reports				
- Views - Lists				
- Lists - Properties				
- Migration				
4 III +	Copyright @ 2002-2011 Avaya Inc. /	All rights reserved.		

Figure 7: Adding a new IP Telephony Node (cont)

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- Under the SIP Line Gateway Services section, select MO from the SLG Role list.
- From the SLG Mode list, select S1/S2 (SIP Proxy Server 1 and Server 2), see Figure 8.

- UCM Network Services	Αναγα	CS1000 Element Manager	Help Logout
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- Zones - Host and Route Tables - Network Address Translation (NA' - QoS Thresholds - Personal Directory - Unicode Name Directory - Customers - Customers - Customers - Customers - Dichannels - Dichannels - Dichannels - Dichannels - Dipting and Numbering Plans - Electronic Switched Network - Flexible Code Restriction - Incoming Digit Translation - Phones - Templates - Required Value - Value - Templates - Reports - Templates - Reports - Vews		MO SLG IPv4 address: 0.0.0.0	
- Note and Rundle Tables - Note Address Translation (NA' - CoS Thresholds - Personal Directories - Unicode Name Directory - Emergency Services - Engineered Values - Emergency Services - Geographic Redundancy - Software - Routes and Trunks - Customers - Routes and Trunks - D-Channels - Dolling and Numbering Plans - Electronic Switched Network - Flexible Code Restriction - Incoming Digit Translation - Phones - Templates - Reports - Templates - Reports - Vews			
- Return R Joiness Translation (NA - QoS Thresholds - Personal Directories - Personal Directories - Unicode Name Directory Interfaces - Engineered Values Emergency Services Geographic Redundancy Software - Customers - Routes and Trunks - Dochannels - Dochannels - Dialing and Numbering Plans - Electronic Switched Network - Flexable Code Restriction - Incoming Digit Translation - Phones - Required Value *			
Personal Directories Unicode Name Directory Unicode Name Directory Unicode Name Directory Interfaces Emergency Services Geographic Redundancy Software Customers Customers Routes and Trunks D-Channels Dolgital Trunk Interface Daling and Numbering Plans Electronic Switched Network Flexible Code Restriction Incoming Digit Translation Phones Tempiates Reports			
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 Interfaces Engineered Values Emergency Services Geographic Redundancy Software Customers Routes and Trunks D-Channels Digital Trunk Interface Dialing and Numbering Plans Electronic Switched Network Flexable Code Restriction Incoming Digit Translation Phones Templates Reports We sol 		[272]	
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+ Geographic Redundancy + Software - Customers - Routes and Trunks - D-channels - Digital Trunk Interface - Dialing and Numbering Plans - Electronic Switched Network - Flexible Code Restriction - Incoming Digit Translation - Tempiates - Reports - Tempiates - Reports - Wews		MO SLG transport TCP 👻	
Software Customers Coutes and Trunks Routes and Trunks Dechannels Digital Trunk Interface Dialing and Numbering Plans Electronic Switched Network Flexible Code Restriction Incoming Digit Translation Phones Templates Required Value		GR SI G IDv/ address: 0.0.0.0	
- Customers - Customers - Routes and Trunks - Routes and Trunks - D-Channels - Dialing and Numbering Plans - Electronic Switched Network - Flexible Code Restriction - Incoming Digit Translation - Phones - Tempiates - Reports - Reports - Wews			
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- Routes and Trunks - D-Channels - D-Channels - Digital Trunk Interface - Digital Trunk Interface - Discussion - Electronic Switched Network - Flexible Code Restriction - Incoming Digit Translation - Phones - Templates - Reports - Wews - Wews			
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- Electronic Switched Network - Flexible Code Restriction - Incoming Digit Translation - Phones - Templates - Reports - Mews	- Digital Trunk Interface	GR SLG port: 5070 (1 - 65535)	
- Incoming Digit Translation - Phones - Templates - Reports - Views	- Electronic Switched Network		
- Phones - Templates - Reports - Views			
- Templates - Reports - Views			
- Reports - Views			
- Views			
	- Lists		
- Properties			
- Migration			
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Figure 8: Adding a new IP Telephony Node (cont)

- Click Next. The Confirm new Node details page appears (not shown).
- Click on the **Transfer Now** button and then The **Synchronize Configuration Files** (Node ID 512) page appears.
- Click Finish and wait for the configuration to be saved. The Node Saved page appears, see Figure 9.

Αναγα	CS1000 Element Manager	D Logout
– UCM Network Services – Home – Links – Mrtual Terminals – System	Managing: text 10.97.78 Username: admin System > IP Network > IP Telephony Nodes aved Node Saved Node Saved	
Aarms Aarms Maintenance Core Equipment Peripheral Equipment IP Network <u>Nodes: Servers, Media Cards</u> Maintenance and Reports Media Gateways	Node ID: 512 has been saved on the call server. The new configuration must also be transferred to associated servers and media cards. Transfer Now You will be given an option to select individual servers, or transfer to all.	
- Zones - Host and Route Tables - Network Address Translation (NA - QoS Thresholds - Personal Directories - Unicode Name Directory + Interfaces	Show Nodes You may initiate a transfer manually at a later time.	
- Engineered Values + Emergency Services + Geographic Redundancy + Software - Customers - Routes and Trunks < m +	▼ Copyright © 2002-2011 Avaya Inc. All rights reserved.	

Figure 9: Node Saved with Transfer Configuration

- Select the SIP Line server that associated with changes and then click on the **Start Sync** button to transfer the configuration files to the selected servers, see **Figure 10**.

Αναγα	CS	31000 Element Man	ager			Help Logout
- UCM Network Services - Home - Links - Virtual Terminals	-		» IP Telephony Nodes » Synchro			
- System + Alarms - Maintenance + Core Equipment - Peripheral Equipment	ш	Note: Select components to sy components, and requires a re Start Sync Cancel			This process transfers server INI ete.	files to selected Print Refresh
IP Network <u>Nodes: Servers. Media Cards</u> Maintenance and Reports Media Gateways Zones		Hostname sip175 Application restart is only requi	Type Signaling_Server	Applications LTPS, Gateway, PD, Presence Publisher, IP Media Services or if changes have been mar	Synchronization Status Sync required de to general LAN configurations, SN	TPsettings SIP and
Host and Route Tables Network Address Translation (N QoS Thresholds Personal Directories Unicode Name Directory Interfaces Engineered Values Emergency Services Geographic Redundancy Software	NA"				ing or disabling services, or adding o	
+ Sonware - Customers - Routes and Trunks		✓ Copyright © 2002-2011 Avaya Inc.	All rights reserved	III		

Figure 10: Synchronize Configuration Files

<u>Note</u>: The first time a new Telephony Node is added and transfered to the call server, the SIP Line services need to be restarted. To restart the SIP Line services, log in as administrator to the command line interface of the SIP Line server and issue the command: **appstart restart**.

5.5. Create a D-Channel for SIP Line

On the EM page, on the left column menu navigate to **Routes and Trunks -> D-Channels**. Under the **Configuration** section as shown in **Figure 11**, enter a number in the **Choose a D-Channel Number** field, and click on the **to Add** button.

AVAYA c	S1000 E	Element Manage	r			Help Logout
- UCM Network Services - Home - Links - Virtual Terminals - System - Alarms - Maintenance + Core Equipment - Peripheral Equipment - Peripheral Equipment - Nodes: Servers, Media Cards - Maintenance and Reports - Media Gateways - Zones - Host and Route Tables - Network Address Translation (NA ⁺ - QoS Thresholds	D-Cha Mai	to.97.78 Username: admi toutes and Trunks » D-Channels Innels D-Channel Diagnostics (L Network and Peripheral EC MSDL Diagnostics (LD 96 TMDI Diagnostics (LD 96) D-Channel Expansion Dia Infiguration	D 96) <u>uuipment</u> (LD 32, Virtual)	D-Channels)		
– Personal Directories – Unicode Name Directory + Interfaces	Choo	Choose a D-Channel Number: 4 v and type: DCH v to Add				
 Engineered Values + Emergency Services 		Channel: 1	Type: DCH	Card Type: DCIP	Description: SIP	Edit
+ Geographic Redundancy + Software		Channel: 2	Type: DCH	Card Type: TMDI	Description: RIs6	Edit
- Customers - Routes and Trunks - Routes and Trunks - D-Channels - Digital Trunk Interface - Dialing and Numbering Plans - Electronic Switched Network - Flexible Code Restriction	-	Channel: 3	Type: DCH	Card Type: DCIP	Description: SIPLine	Edit
- Incoming Digit Translation - Phones - Templates - T	Copyright @) 2002-2011 Avaya Inc. All right	is reserved.			

Figure 11: D-Channels configuration page

- The D-Channels xx Property Configuration page appears as shown in Figure 12.
- From the Interface type for D-channel (IFC) list, select Meridian Meridian1 (SL1).
- Leave the other fields at default values.

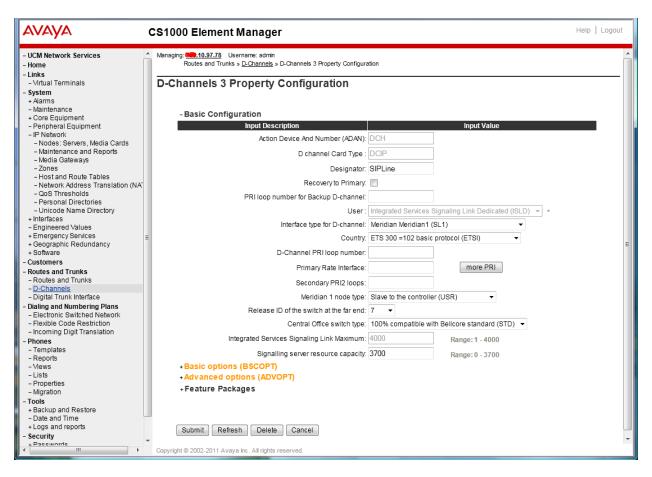


Figure 12: SIP Line D-Channel Property Configuration

- Click on the **Basic options (BSCOPT)** link. The **Basic options (BSCOPT)** list expands (not shown).
- Click on Edit to configure Remote Capabilities (RCAP). The Remote Capabilities Configuration detail page will appear as shown in Figure 13.
- Select the Message waiting interworking with DMS-100 (MWI) check box.
- Select the Network name display method 2 (ND2) check box.
- At the bottom of the **Remote Capabilities Configuration** page, click **Return Remote Capabilities** to return the **D-Channel xx Property Configuration** page.

AVAYA	CS1000 Element Manager	Help Logout
- UCM Network Services	Rerouting requests processed using integer value (DV2I)	~
- Home	Rerouting requests processed using object identifier (DV20)	_
- Links	Diversion info. sent. rerouting requests processed (DV3I)	
– Virtual Terminals – System	EuroISDN - div. info sent. rerouting reg. processed (DV30)	
+ Alarms	Call transfer notification and invocation to EuroISDN (ECTO)	
– Maintenance + Core Equipment	Malicious call identification (MCID)	
– Peripheral Equipment	MCDN QSIG conversion (MQC)	
+ IP Network + Interfaces	Remote D-channel is on a MSDL card (MSL)	
- Engineered Values	Message waiting interworking with DMS-100 (MWI)	
+ Emergency Services + Geographic Redundancy	Network access data (NAC)	
+ Software	Network call trace supported (NCT)	
- Customers	Network name display method 1 (ND1)	
 Routes and Trunks Routes and Trunks 	Network name display method 2 (ND2)	
- <u>D-Channels</u>	Network name display method 3 (ND3)	
 Digital Trunk Interface Dialing and Numbering Plans 		
- Electronic Switched Network	Name display - integer ID coding (NDI)	
 Flexible Code Restriction Incoming Digit Translation 	Name display - object ID coding (NDO)	
- Phones	Path replacement uses integer values (PRI)	
– Templates – Reports	Path replacement uses object identifier (PRO)	
- Views	Release Link Trunks over IP (RLTI)	
- Lists	Remote virtual queuing (RVQ)	
– Properties – Migration	Trunk anti-tromboning operation (TAT) 📃	
- Tools	User to user service 1 (UUS1)	
+ Backup and Restore - Date and Time	NI-2 name display option. (NDS) 📃	
+ Logs and reports	Message waiting indication using integer values (QMWI) 📃	
- Security + Passwords	Message waiting indication using object identifier (QMWO) 📃	
+ Policies + Login Options	User to user signalling (UUI)	
	Return - Remote Capabilities Cancel	~
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Figure 13: SIP Line D-Channel RCAP Configuration Details

- Message Waiting Interworking with DMS-100 (MWI) must be enabled to support voice mail notification on SIP Line endpoints.
- Network Name Display Method 2 (ND2) must be enabled to support name display between SIP Line endpoints.
- Other check boxes are left unchecked.

Click on the **Submit** button of the D-Channel Property Configuration page to save changes.

5.6. Create an Application Module Link (AML)

On the EM page, navigate to **System -> Interfaces -> Application Module Link**, click on the **Add** button to add a new Application Module Link (not shown). The **New Application Module Link** page appears as shown in **Figure 14**.

Enter an AML port number in the **Port number** text box. The AML of SIP Line Service can use a port from 32 to 127. In this case, SIP Line Service is configured to use port 33.

Click on the Save button to complete adding the AML link, and to save the configuration.

(C) (C) (C) https://cpppm3.bvwde	v.com/em/Web_6-0/SEC 🔎 👻 Certificate er 🗟 C 🗙 🦉 Element Manager 🛛 🗙 😳
Αναγα	CS1000 Element Manager Help Logout
- UCM Network Services - Home - Links - Virtual Terminals	Managing: <u>135.10.97.78</u> Username: admin System » Interfaces » <u>Application Module Link</u> » New Application Module Link New Application Module Link
- System + Alarms - Maintenance + Core Equipment - Peripheral Equipment - IP Network - Nodes: Servers, Media Cards	E Port number 33 • (18 - 127) AML over ELAN Description: For SIPLine
Maintenance and Reports Media Gateways Zones Host and Route Tables Network Address Translation (NA QoS Thresholds Personal Directories	Link control system parameters Maximum octets : 512 - (per HDLC frame)
 Unicode Name Directory Interfaces 	* Required value. Save Cancel
 Application Module Link Value Added Server Property Management System Engineered Values 	· · ·
• <u> </u>	Copyright © 2002-2011 Avaya Inc. All rights reserved.

Figure 14: Adding a new AML

5.7. Create a Value Added Server (VAS)

On the EM page, navigate to System -> Interfaces -> Value Added Server and click on the Add button to add a new VAS.

The Value Added Server page appears (not shown), in this page, select the Ethernet Link link and the Ethernet Link page appears as shown in Figure 15.

Enter a number in the Value added server ID field, in this example 33 was used. In the Ethernet LAN Link drop down list, select the AML number of ELAN that was created in the Section 5.6.

Leave other fields as default values and click on the **Save** button to complete adding the **VAS** and save the configuration.

A ttps://cpppm3.bvwdev	com ′emWeb_6-0/S P → S Certificate e S C × 2 Element Manager ×
Αναγα	CS1000 Element Manager Help Logout
– Virtual Terminals – System + Alarms – Maintenance	 Managing: <u>strand</u>: Username: admin System » Interfaces » <u>Value Added Server</u> » <u>Add Value Added Server</u> » Ethernet Link
+ Core Equipment - Peripheral Equipment - Peripheral Equipment - IP Network - Nodes: Servers, Media Cards - Media Cardes - Media Cateways - Zones - Host and Route Tables - Network Address Translation (NA* - CoS Thresholds - Personal Directories - Unicode Name Directory - Interfaces - Application Module Link - Yalue Addred Server - Property Management System - Engineered Values	Ethernet Link Value added server ID: 33 • (18 - 127) Ethernet LAN Link 33 • ELAN port configured in ADAN Application security : Interval: 1 • Time interval for checking the link for overload in five second increments Message count threshold: 9999 • (10 - 9999)
+ Emergency Services + Geographic Redundancy + Software	* Required value. Save Cancel
- Customers	Copyright © 2002-2011 Avaya Inc. All rights reserved.

Figure 15: Adding a new Value Added Service for the AML

5.8. Create a Virtual Trunk Zone

On the EM page, navigate to menu **System -> IP Network -> Zones**. The **Zones** page appears on the right, in this page select **Bandwidth Zones** link.

On the **Bandwidth Zones** page, click on the **Add** button, the **Zone Basic Property and Bandwidth Management** page appears as shown in **Figure 16**.

Enter a zone number in the **Zone Number (Zone)** field and in the **Zone Intent (ZBRN)** drop down menu select **VTRK (VTRK)**.

Leave other fields as default values and click on the **Save** button to complete adding the Zone.

<u>Note</u>: Repeat the step above to create another zone for the SIP Line phone; however remember to select **MO**, instead of VTRK in the field **Zone Intent**.

AVAYA	CS1000 Element Manager	Help Logout
- UCM Network Services - Home - Links - Virtual Terminals - System + Alarms - Maintenance	Managing: 10.97.78 Username: admin System » IP Netw ork » Zones » Bandwidth Zones » Zone Basic Property and Bandwidth Management Zone Basic Property and Bandwidth Management Input Description Input Value	
Core Equipment Core Equipment Peripheral Equipment IP Network Nodes: Servers, Media Cards Maintenance and Reports Media Gateways Zones Host and Route Tables Network Address Translation (NA CoS Thresholds Personal Directories Unicode Name Directory Interfaces Application Module Link Value Added Server	Zone Number (ZONE): 4 (1-8000) Intrazone Bandwidth (INTRA_BW): 1000000 (0-10000000) Intrazone Strategy (INTRA_STGY): Best Quality (BQ) Interzone Bandwidth (INTER_BW): 1000000 (0-10000000) Interzone Strategy (INTER_STGY): Best Quality (BQ) Resource Type (RES_TYPE): Shared (SHARED) Zone Intent (ZBRN): MO (MO) Description (ZDES):	E
Property Management System Engineered Values Emergency Services Geographic Redundancy Software III	* Required value. Save	Cancel

Figure 16: Adding a new Zone for Virtual Trunk

5.9. Create a SIP Line Route Data Block (RDB)

On the EM page, navigate to the menu **Routes and Trunks** -> **Routes and Trunks**; the **Routes and Trunks** page appears (not shown). In this page, click on the **Add route** button next to the customer number that the route will belong to.

The Customer ID, New Route Configuration page appears, expand the Basic Configuration tab, and enter values below and as shown in Figure 17 and 18.

- Route Number (ROUT): 3
- Trunk type(TKTP): TIE
- Incoming and Outgoing trunk (ICOG): IAO
- Access Code for Trunk group (ACOD): enter a number for ACOD, for example 757.
- The route is for a virtual trunk route (VTRK): Checked.
- Zone for codec selection and bandwidth management (ZONE): 4, this is the Virtual trunk zone number that created in the Section 4.8.
- Node ID of signaling server of this route (NODE): 512, this is the node ID of the SIP Line.
- Protocol ID for the route (PCID): SIP Line (SIPL).
- Integrated services digital network option (ISDN): checked.
- Mode of operation (MODE): Route uses ISDN Signaling Link (ISLD).
- D channel number (DCH): 4, the D-channel number that was created in the Section 4.5.
- Interface type for route (IFC): Meridian M1 (SL1).
- Network calling name allowed (NCNA): checked.
- Channel type (CHTP): B-channel (BCH).
- Call type for outgoing direct dialed TIE route (CTYP): CDP.
- Calling Number dialing plan (CNDP): CDP.

Leave default values for The Basic Route Options, Network Options, General Options, and Advanced Configurations sections.

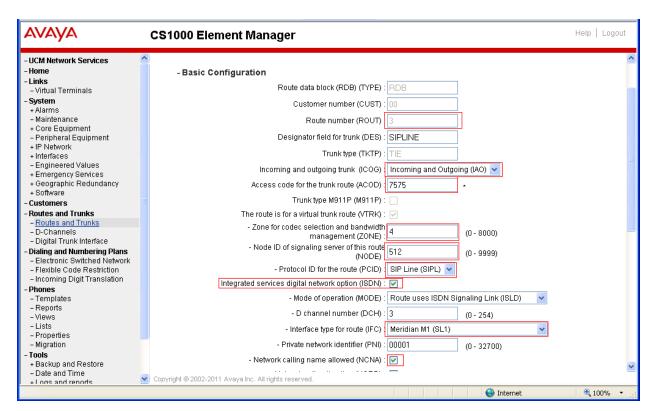


Figure 117: SIP Line Route Configuration

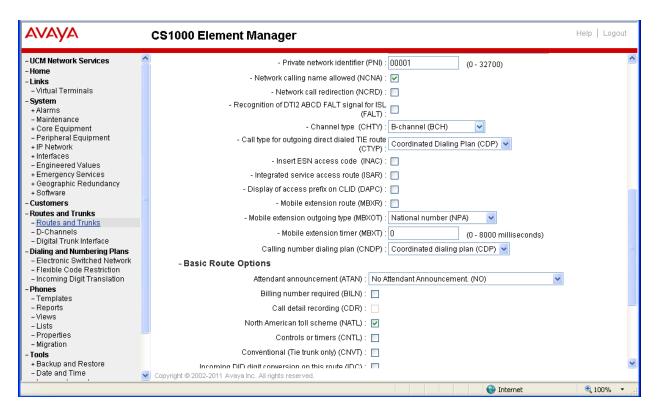


Figure 18: SIP Line Route Configuration (cont)

5.10. Create SIP Line Virtual Trunks

On the EM page, navigate to **Routes and Trunks** -> **Routes and Trunks** and select the **Add route** button beside to the route was created in the **Section 5.9** above to create new trunks.

The Customer ID, Route ID, and Trunk type TIE trunk data block page appears as shown in Figure 19, enter values for fields as shown below:

- Multiple trunk input number (MTINPUT): 32 -> create 32 trunks.
- Auto increment member number: checked.
- Trunk data block (TYPE): IP Trunk (IPTI).
- Terminal Number (TN): 100 0 2 0 -> enter the first TN of a range TN.
- **Member number**: 33, this is ID of trunk, just enter the first ID for first trunk, next ID will be automatically created and incremented.
- Start arrangement Incoming: Immediate (IMM).
- Start arrangement Outgoing: Immediate (IMM).
- Trunk Group Access Restriction (TGAR): 1.
- Channel ID for this trunk: 33, this ID should be the same with the ID of Member Number.

Click on the **Class of Service** button and assign following class of services (not shown):

- Media security: Media Security Never (MSNV).
- Restriction level: Unrestricted.

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Leave other fields at default values and click on the **Return Class of Service** button to return to the **Trunk type TIE trunk data block** page.

AVAYA	CS1000 Element Manager	Logout
- UCM Network Services - Home - Links	Managing: <u>#36 10.97.78</u> Username: admin Routes and Trunks » <u>Routes and Trunks</u> » Customer 0, Route 3	^
- Virtual Terminals - System	Customer 0, Route 3,Trunk type TIE trunk data block	
+ Alarms - Maintenance + Core Equipment	-Basic Configuration	
 Peripheral Equipment + IP Network 	Multiple trunk input number: 32 Range: 2 - 3700	
+ Interfaces	Auto increment member number: 📝	
 Engineered Values Emergency Services 	Trunk data block: IP Trunk (IPTI)	
+ Geographic Redundancy + Software	Terminal number: 100 0 2 0 *	
- Customers	Designator field for trunk: SIPLINE	
- Routes and Trunks	Extended trunk: VTRK	=
- D-Channels - Digital Trunk Interface	Member number: 33 *	
- Dialing and Numbering Plans	Level 3 Signaling:	
 Electronic Switched Network Flexible Code Restriction 	Card density: Octal Density (8D) 🔻	
– Incoming Digit Translation	Start arrangement Incoming : Immediate (IMM) -	
- Phones - Templates	Start arrangement Outgoing: Immediate (IMM)	
- Reports	Trunk group access restriction:	
- Lists	Channel ID for this trunk: 33	
– Properties – Migration	Class of Service: Edit	
- Tools	+ Advanced Trunk Configurations	
+ Backup and Restore - Date and Time		
+ Logs and reports	* Required value. Save C	Cancel 👻
- Security	 Copyright © 2002-2011 Avaya Inc. All rights reserved. 	

Click on the Save button to complete adding virtual trunks for SIP Line.

Figure 19: Adding virtual trunks for SIP Line Trunk

5.11. Create a SIP Line Phone

To create a SIP Line phone on the Call Server, log in as administrator using the command line interface (CLI) and issue the overlay (LD) 11/20 as shown below.

The bold fields must be properly inputted as they are configured on the Call server, for other fields hit enter to leave it at default values.

```
LD20
PT0000
REQ:new
TYPE: UEXT -> Universal extension type for SIP Line phone
TN 104001
DES POLY1 -> Description of Phone.
CUST 0
```

UXTY SIPL -> Universal extension type is SIP Line MCCL YES SIPN 0 SIP31 -> For SIP phone third party, enter 1 in this field FMCL TLSV SIPU 54008 -> SIP phone username NDID 512 -> Node ID of SIP Line SUPR SUBR UXID NUID NHTN ZONE 3 -> Zone for SIP Line phone. MRT ERL ECL VSIT FDN 54002 -> Forward No Answer to this DN, need to enable class of service FNA TGAR 1 LDN NCOS 7 -> Network Class of Service, 7 is highest level. SGRP RNPG SCI SSU XLST SCPW 1234 → Password to log in to SIP Line username 54008 SFLT CAC MFC CLS FNA FBA HTA MWA DNDA CNDA CFXA -> class of service. RCO HUNT 54444 -> Forward busy to this DN, need to enable class of service FBA and HTA PLEV KEY 00 SCR 54008 0 MARP -> Key 0 is DN of SIP phone. CPND new CPND_LANG ROMAN NAME Poly 8440 -> Display name of SIP Phone. XPLN 13 DISPLAY_FMT FIRST,LAST 01 HOT U 2654008 MARP 0 -> Key 1 Hot U with prefix + DN 02 CWT -> Call Waiting key 03 MSB -> Make Set busy key 04 SCU 0000 -> Speech call dial key

6. Configure Polycom SpectraLink 8440

This section describes how to access the Polycom SpectraLink 8440 SIP endpoint web interface and configure the Polycom 8440 for testing. For more information on how to configure the Polycom SpectraLink 8440 phone connected to the Access Point Wi-Fi router, please refer to the document in the **Section 9**.

6.1. Login Polycom SpectraLink 8440

This section shows how to log in to the home page of Polycom SpectraLink 8440 to manage and configure the 8440 phone.

Open the web browser, and in the address field enter the Polycom SpectraLink 8440 IP address: <u>http://ipaddress</u> and the Polycom SpectraLink 8440 login page will appear as shown in **Figure 20**. Enter the username, **Polycom**, and its default password, **456**.

Welcome to Polycom Web Configuration Utility				
	Enter Login	Information		
	User Name	Polycom		
	Password	•••		
	Submit	Reset		

Figure 20: Polycom SpectraLink 8440 Login Screen

Click the **Submit** button, the homepage of Polycom SpectraLink 8440 appears as in **Figure 21** below.

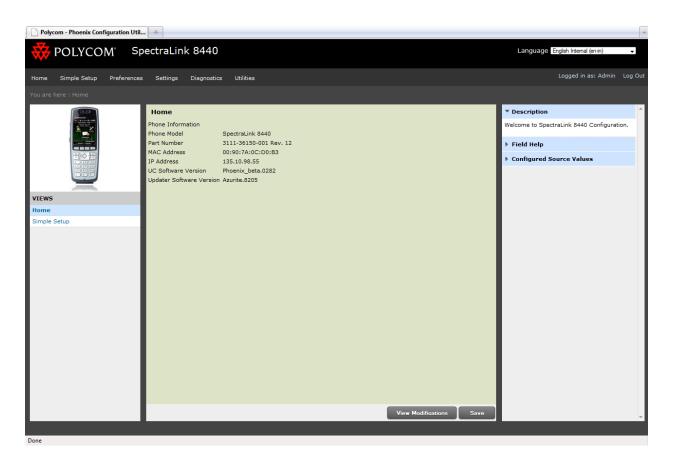


Figure 21: Home page of Polycom SpectraLink 8440 telephone

6.2. Configure the Lines for Polycom SpectraLink 8440

This section shows how to configure the Polcom 8440 telephone to register with the CS1000 SIP Line gateway.

On the homepage of the configuration screen (see Figure 21), click on the Simple Setup menu, the Simple Setup page appears as shown in Figure 22. Enter the values as shown below:

- Language: select English (Internal) in the Phone Language drop down menu.
- Time Synchronization: select time zone for phone, for example (GMT-5) Eastern Time (US and Canada).
- SIP Server:
 - Address: 10.10.97.187 -> this is IP address of CS 1000 SIP Line server.
 - **Port:** 5070
- SIP Outbound proxy:
 - Address: 10.10.97.187 -> Use the same IP as the CS 1000 SIP Line server.
 - Port: 5070
 - SIP Line Identification:
 - **Display Name**: Poly 8440

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- Address: <u>54008@sip175.com</u>
- Authentication User ID: 54008 -> this user ID was configured in the field SIPU when creating TN of SIP Line phone in the Section 5.11
- Authentication Password: 1234 -> this password was configured in the field SCPW when creating TN for SIP Line phone in the Section 5.1
- Label:

Click on the **Save** button to save changes.

POLYCOM SpectraLink 8440			
Home Simple Setup Preferences	Settings Diagnostics Utilities		
	Simple Setup Language Phone Language English (Internal) Time Synchronization SNTP Server		
VIEWS	Time Zone (GMT -5:00) Eastern Time (US & Canada), Bogota, Lima - SIP Server Address 10.10.97.187		
Home	Port 5070 SIP Outbound Proxy		
Simple Setup	Address 10.10.97.187 Port 5070 SIP Line Identification Display Name Poly 8440 Address 54008@sipl75.com Authentication User ID 54008 Authentication Password •••• Label Reset to Default Cancel		
	View Modifications Save		

Figure 22: Simple Setup for Polycom 8440 Phone

6.3. Local Call Forward Settings

This section shows how to set "Local Call Forward" such as Call Forward All calls, Call forward busy and Call Forward No Answer on the Polycom SpectraLink 8440 telephone.

On the homepage of Polycom 8440 (see Figure 21), navigate to menu Setting -> Lines -> Call Diversion, the Call Diversion section appears as shown in Figure 23.

- To set the Forward All Calls, select the Enable option button in the line Forward All.
- To set the Forward Busy, select the **Enable** option button in the line **On Busy** and enter a forward DN on the **Busy Contact** box.
- To set the Call Forward No Answer, select the **Enable** option button in the line **On No Answer.**

<u>Note</u>: The "Server Call Forward Always" setting for the Polycom 8440 on the CS 1000 Call Server must be OFF in order to make the "Local Call Forward Always" on the Polycom 8440 take effect.

POLYCOM SpectraLink	3440
Home Simple Setup Preferences Settings D	agnostics Utilities
Line 1	
Ider	ification
Det Cout	ound Proxy
Server Server	er 1
	er 2
	liversion
* Disabled Or	
VIEWS * Diversion C	
Line 1	
Line 2 * Forward All	 ⊙Enabled ⊙Enabled ○Enabled
Line 3 * Busy Conta	
Line 4 * On No Answ	
Line 5 * No Answer	
Line 6 * No Answer	ontact
* On Do-Not-	isturb OEnabled ODisabled
* Do-Not-Dist	rb Contact
Mes	age Center
N ote: * Fields requir	phone reboot/restart Reset to Default Cancel
	View Modifications Save

Figure 23: Call Diversion section of Polycom 8440

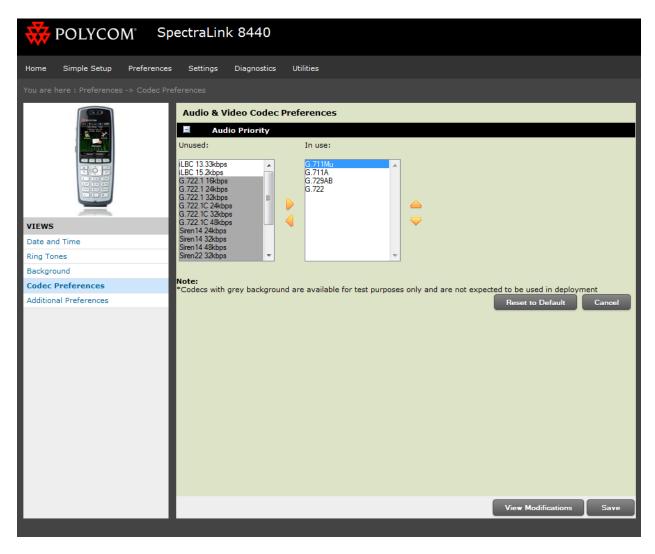
6.4. Codec settings

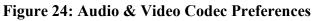
This section shows how to set the Codec on the Polycom SpectraLink 8440 phone. The compliance testing has been done on both codecs, G711 and G729.

On the homepage of Polycom SpectraLink 8440 (see Figure 21), navigate to menu Preference - > Codec Preferences, the Audio and Video Codec Preferences page appears as shown in Figure 24.

The list of audio Codecs that are being used appear under the **In use** column. To use the codec G711 as the first choice, move it up to the top of the **In Use** list, repeat the same for other codecs if it needs to be the first choice.

Click on the Save button to save changes.





7. Verification Steps

This section includes some steps that can be followed to verify the configuration.

- Verify that the Poycom SpectraLink 8440 telephone registers successfully with the CS 1000 SIP Line Gateway server and Call Server by using the CS 1000 Linux command line and CS 1000 Call Server overlay LD 32.
 - Log in to the SIP Line server as an administrator by using Avaya account.
 - Issue command "slgSetShowByUID [userID]" where userID is SIP Line user's ID being checked.

```
[admin@sipl ~]$ slgSetShowByUID 54008
=== VTRK ===
UserID AuthId TN Clients Calls
SetHandle Pos ID SIPL Type
----- ---- ----- ----- ----- -----
----- -----
             54008
                          54008 104-00-00-01 1 0
0x8fc4cf8 SIP Lines
          StatusFlags = Registered Controlled KeyMapDwld SSD
           FeatureMask =
           CallProcStatus = 0
            Current Client = 0, Total Clients = 1
             == Client 0 ==
             IPv4:Port:Trans = 10.10.98.55:5060:udp
            Type = SIP3
UserAgent = PolycomSpectraLink-SL_8440-
UA/Phoenix_beta.0282
            x-nt-guid = 267d228547c1562399f1f743a2971fb5
RegDescrip =

      RegDescrip
      =

      RegStatus
      =

      PbxReason
      =

      OK
      SipCode

      SipCode
      =

      200
      hTransc

      hTransc
      =

      (nil)
      =

      Expire
      =

      3600
      Nonce

      NonceCount
      =

      1
      =

      0
      =

      1
      =

      0
      =

             ClientGUID = 0

MSec CLS = MSNV (MSEC-Never)

Contact = sip:54008@135.10.98.55:5060

KeyNum = 255

AutoAnswer = NO
            Key Func Lamp Label
            0 3 0 54008
1 126 0 2654008
            2
                 9 0
            3 29 0
            4 22 0
5 2 0 54334
```

25 30 0 26 31 0	17 18 19 20 21 22 24	16 18 27 19 52 25 11	0 0 0 0 0 0
	25	30	0
	26	31	0

- Log in to the call server using the admin account.
- Load overlay 32 and then issue command "stat [TN]" where TN is the SIP Line user's TN being checked

```
>ld 32
NPR000
.stat 104 0 0 1
IDLE REGISTERED 00
```

- Place a call from and to Polycom SpectraLink 8440 telephone and verify that the call is established with 2-way speech path.
- During the call, use a pcap tool (ethereal/wireshark) at the SIP Line Gateway and clients to make sure that all SIP request/response messages are correct.

8. Conclusion

All of the executed test cases have passed and met the objectives outlined in the Section 2.1, with some exceptions outlined in Section 2.2. The Polycom SpectraLink 8440 version 4.0.0.0282 is considered to be in compliance with Avaya CS 1000 SIP Line System Release 7.5.

9. Additional References

Product documentation for the Avaya CS 1000 products may be found at: <u>https://support.avaya.com/css/Products/</u>

Product documentation for the Polycom SpectraLink 8400 series products may be found at: http://www.polycom.com

[1] Avaya CS1000 Documents:

Avaya Communication Server 1000E Installation and Commissioning Avaya Communication Server 1000 SIP Line Fundamental, Release 7.5 Avaya Communication Server 1000 Element Manager System Reference – Administration

KP; Reviewed:	Solution & Interoperability Test Lab Application Notes
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Avaya Communication Server 1000 Unified Communications Management Common Services Fundamentals.

Avaya Communication Server 1000 ISDN Primary Rate Interface Installation and Commissioning

[2] Polycom SpectraLink 8400 Series Documents:

Administrator's Guide for the Polycom® UC Software Polycom SpectraLink 8400 Series Wireless Handset User Guide Polycom® SpectraLink® 8400 Series Wireless Telephone Deployment Guide

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