



Application Notes for Polycom SpectraLink 8450 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 8450 SIP telephone. During the compliance testing, the Polycom SpectraLink 8450 was able to register as a SIP client endpoint with the Communication Server 1000 SIP Line gateway. The Polycom SpectraLink 8450 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 8450 SIP telephone Version 4.0.0.0282 used during the compliance testing. The Polycom SpectraLink 8450 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 8450 , 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 8450 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 8450 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 8450 SIP telephone was able to interoperate with the CS 1000 SIP line system. The following areas were tested:

- Registration of the Polycom SpectraLink 8450 SIP telephone to the CS1000 SIP Line Gateway.
- Call establishment of Polycom SpectraLink 8450 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 8450 compliance to required industry standards.
- Polycom SpectraLink 8450 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 8450 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 8450 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 CS1000 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 8450 phone with Avaya UNISTim phone. When Polycom 8450 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 8450 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 phone or with other SIP phones.

2.3. Support

For technical support for the Polycom SpectraLink 8450 SIP endpoints, please contact Polycom Inc 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 8450.

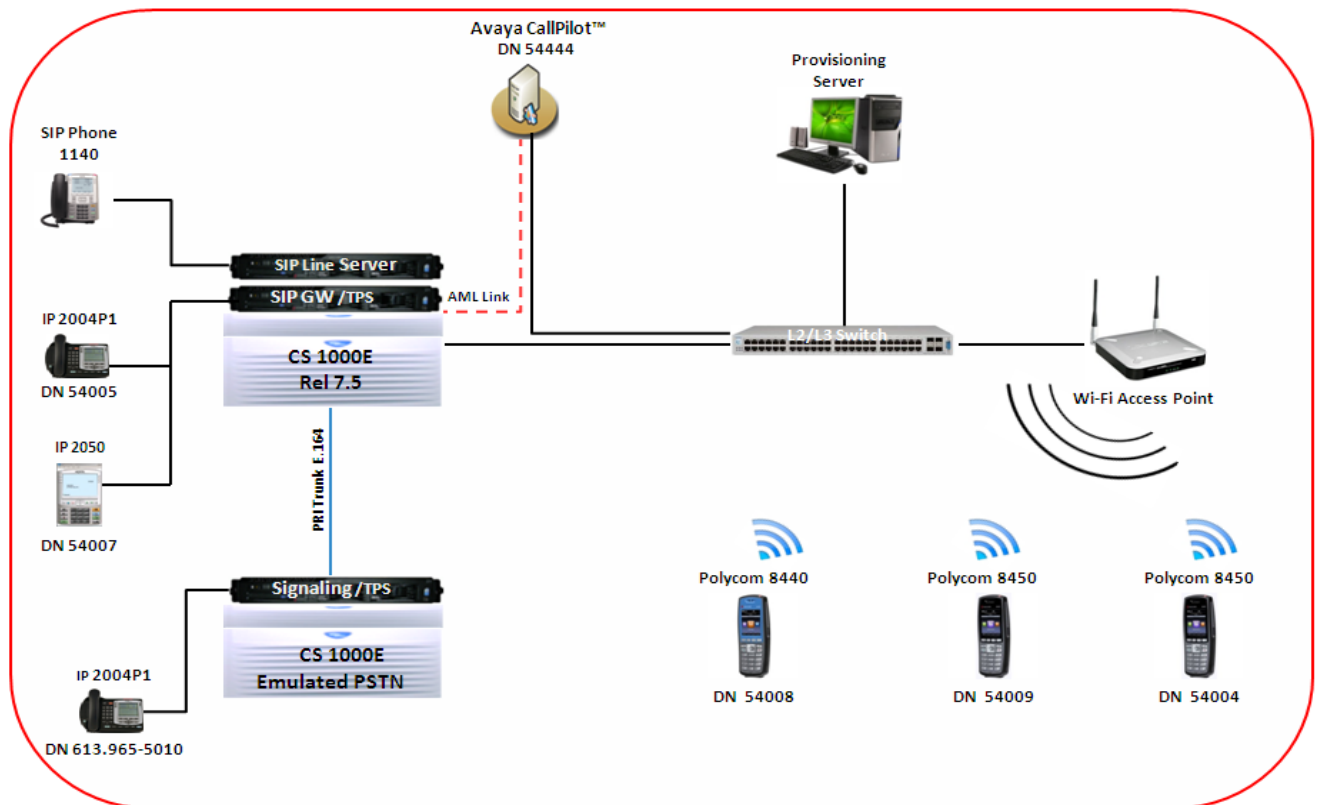


Figure 1: Network Configuration Diagram

4. Equipment and Software Validated

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

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

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 <http://www.avaya.com>.

Package Mnemonic	Package #	Descriptions	Package Type	Applicable market
SIP_LINES	417	SIP Line Service package	New package	Global
FFC	139	Flexible Feature Codes	Existing package	Global
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.

AVAYA

Avaya Unified Communications Management

[Help](#) | [Logout](#)

Network

Elements

CS 1000 Services

IPSec

Patches

SNMP Profiles

Secure FTP Token

Software Deployment

User Services

Administrative Users

External Authentication

Password

Security

Roles

Policies

Certificates

Active Sessions

Tools

Logs

Data

Host Name: car2-sipl-ucm.bvwdev.com

Software Version: 02.20-SNAPSHOT(0000)

User Name admin

Elements

New elements are registered into the security framework, or may be added as simple hyperlinks. Click an element name to launch its management service. You can optionally filter the list by entering a search term.

Add...

Edit...

Delete

<input type="checkbox"/>	Element Name	Element Type	Release	Address	Description
<input type="checkbox"/>	EM on car2-cores	CS1000	7.5	192.168.1.100	New element.
<input type="checkbox"/>	EM on car2-ssg-carrier	CS1000	7.5	192.168.1.101	New element.
<input checked="" type="checkbox"/>	EM on cpppm3	CS1000	7.5	192.168.1.102	New element.
<input type="checkbox"/>	car2-ssg-carrier.bvwdev.com (member)	Linux Base	7.5	192.168.1.103	Base OS element.
<input type="checkbox"/>	car2-sipl-ucm.bvwdev.com (primary)	Linux Base	7.5	192.168.1.104	Base OS element.
<input type="checkbox"/>	car2-mas.bvwdev.com (member)	Linux Base	7.5	192.168.1.105	Base OS element.
<input type="checkbox"/>	car2-cores.bvwdev.com (member)	Linux Base	7.5	192.168.1.106	Base OS element.
<input type="checkbox"/>	car2-sps.bvwdev.com (member)	Linux Base	7.5	192.168.1.107	Base OS element.
<input type="checkbox"/>	cpppm3.bvwdev.com (member)	Linux Base	7.5	192.168.1.108	Base OS element.
<input type="checkbox"/>	ein175.bvwdev.com (member)	Linux Base	7.5	192.168.1.109	Base OS element.

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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.

KP; Reviewed:
SPOC 6/20/2011

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Polycom8450CS1K

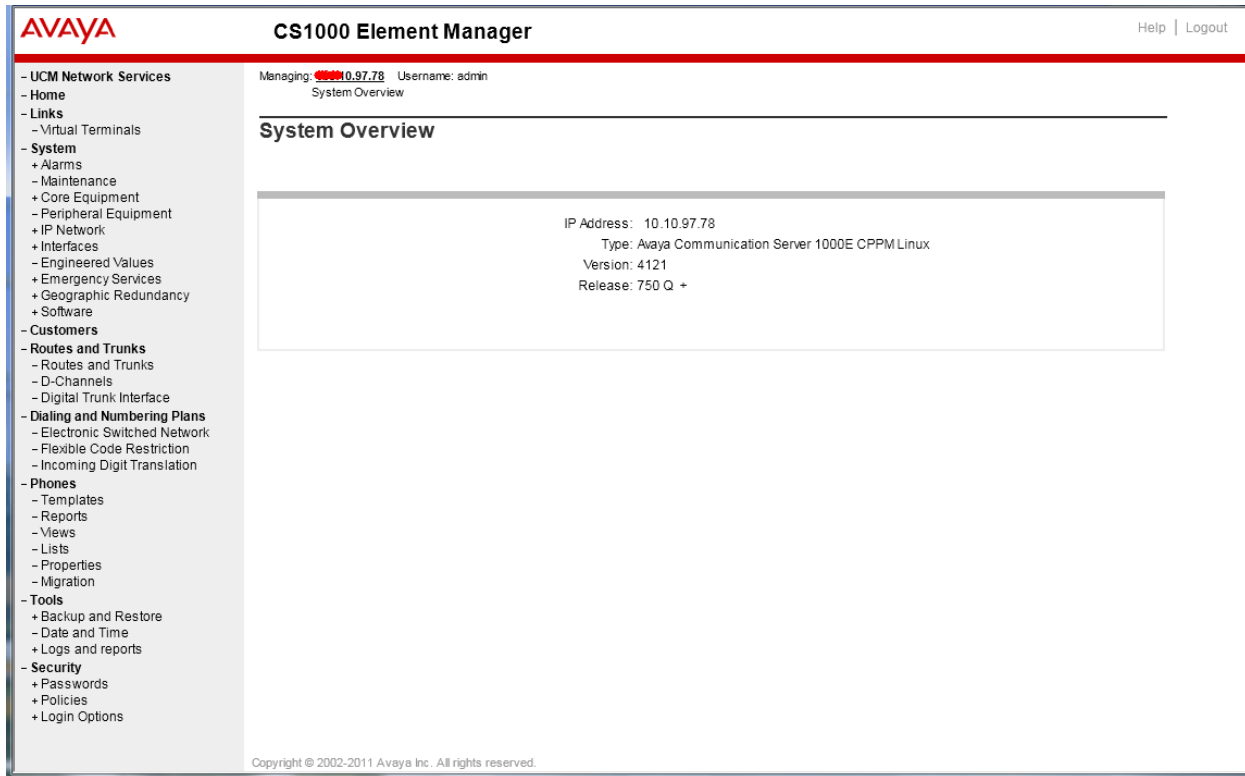


Figure 3: CS 1000 Release 7.5 EM Home Page

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**.

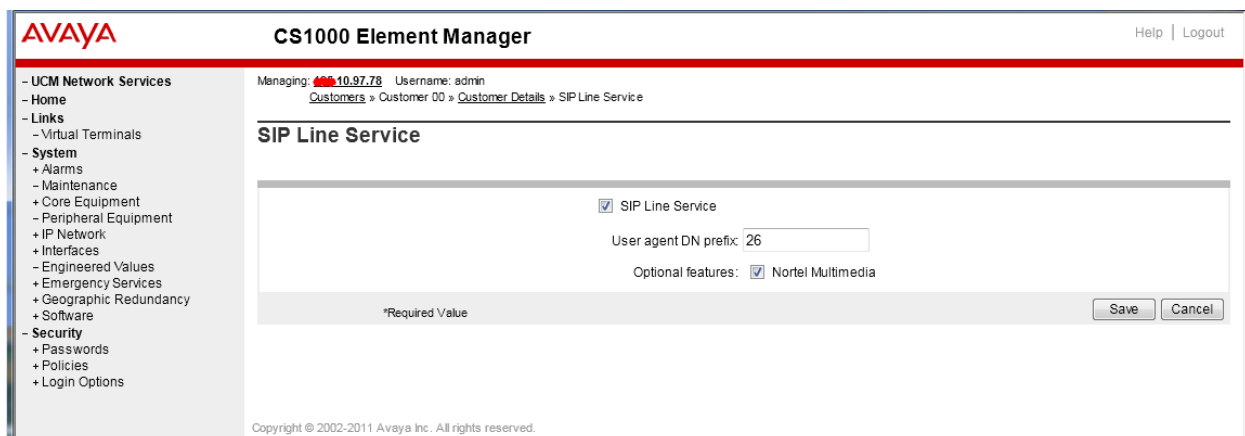


Figure 4: SIP Line Service in Customers Data Block

5.4. Add a new SIP Line Telephony Node

On the EM page, navigate to menu **System → IP Network → 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.

The screenshot shows the 'New IP Telephony Node' configuration page in the AVAYA CS1000 Element Manager. The page is titled 'New IP Telephony Node' and includes a step indicator 'Step 1: Define the new Node and its services.' Below this, a note states: 'You will also require pre-configured servers with appropriate application software already deployed to host the selected services.'

The configuration fields are as follows:

- Node ID:** 512 (required, range 0-9999)
- Call server IP address:** 10.10.97.78 (required)
- TLAN address type:** ☒ IPv4 only, ☐ IPv4 and IPv6
- Embedded LAN (ELAN):**
 - Gateway IP address:** 10.10.97.65 (required)
 - Subnet mask:** 255.255.255.192 (required)
- Telephony LAN (TLAN):**
 - Node IPv4 address:** 10.10.97.187 (required)
 - Subnet mask:** 255.255.255.192 (required)
 - Node IPv6 address:** (empty field)
- Applications:**
 - ☒ SIP Line
 - ☐ UNISTim Line Terminal Proxy Server (LTPS)
 - ☐ Virtual Trunk Gateway (SIPGw, H323Gw)
 - ☐ Personal Directory (PD)
 - ☐ Presence Publisher

At the bottom, there is a note '* Required Value.' and two buttons: 'Next >' and 'Cancel'.

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).

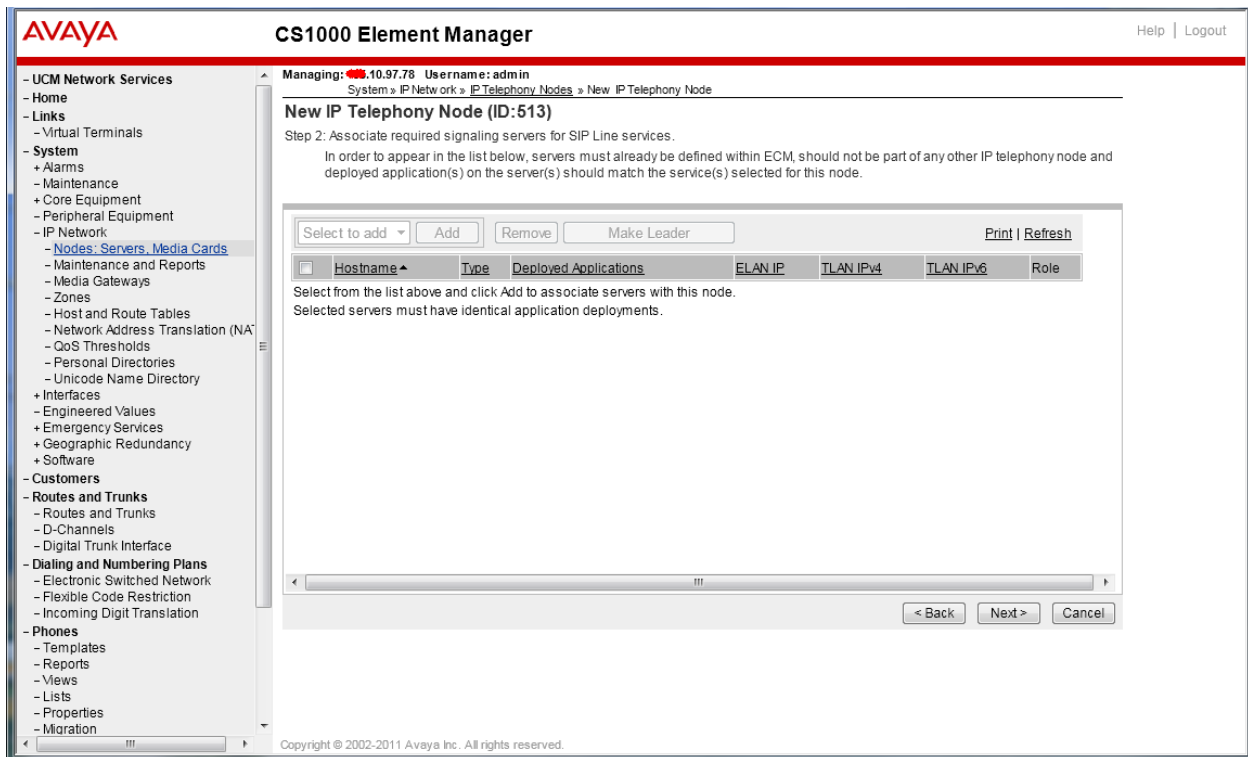


Figure 6: Adding a New IP Telephony Node (cont)

- 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 **sip175.com**.

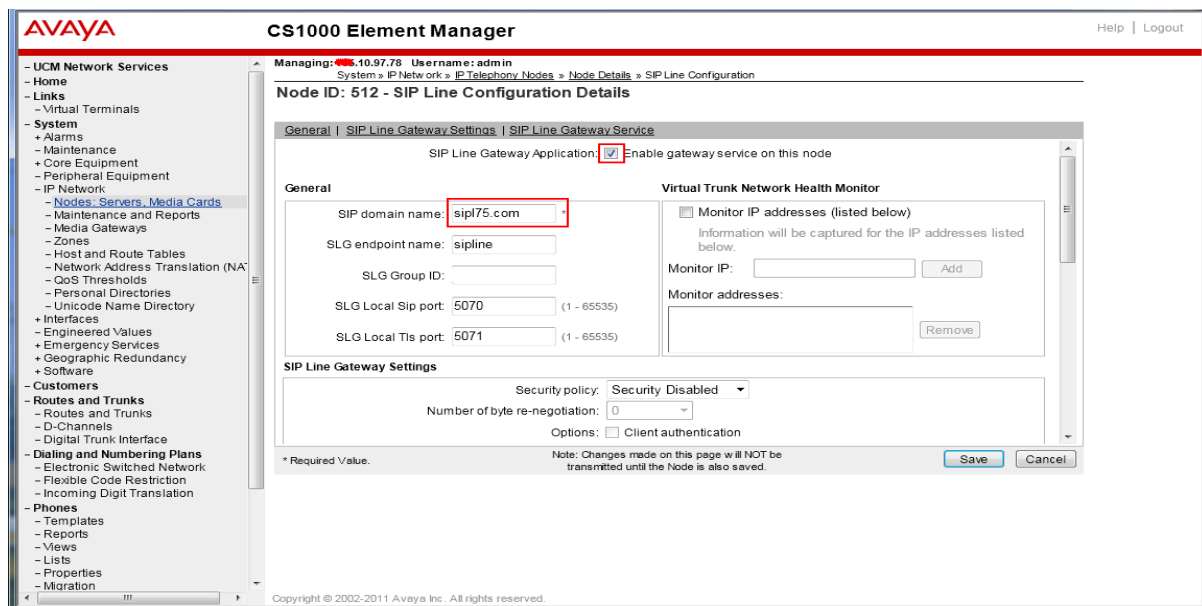


Figure 7: Adding a new IP Telephony Node (cont)

- 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**.

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**.

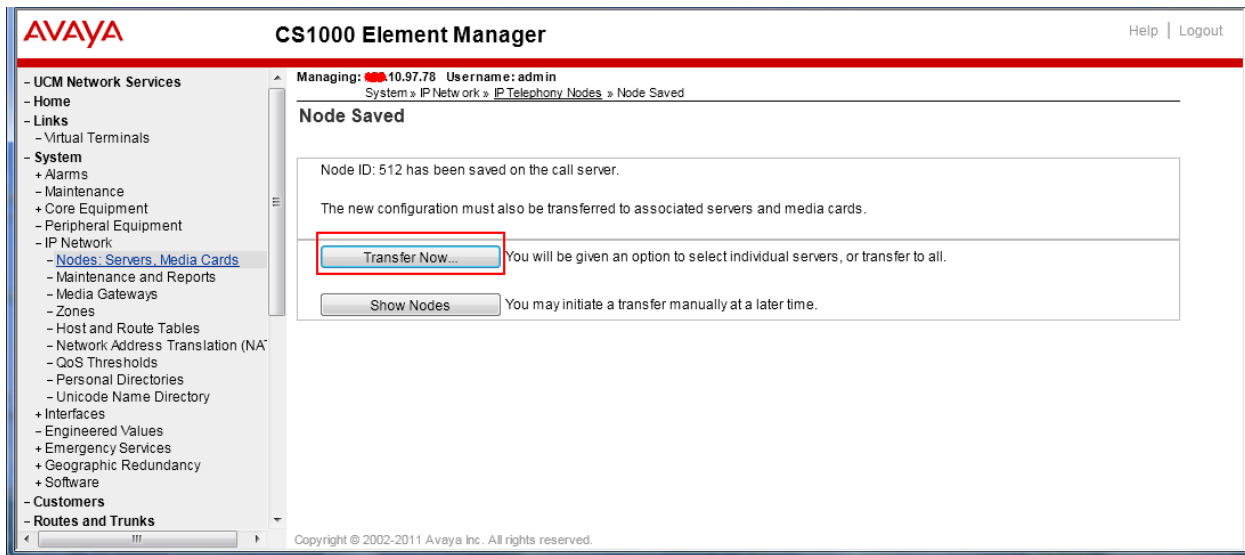


Figure 9: Node Saved with Transfer Configuration

- 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**.

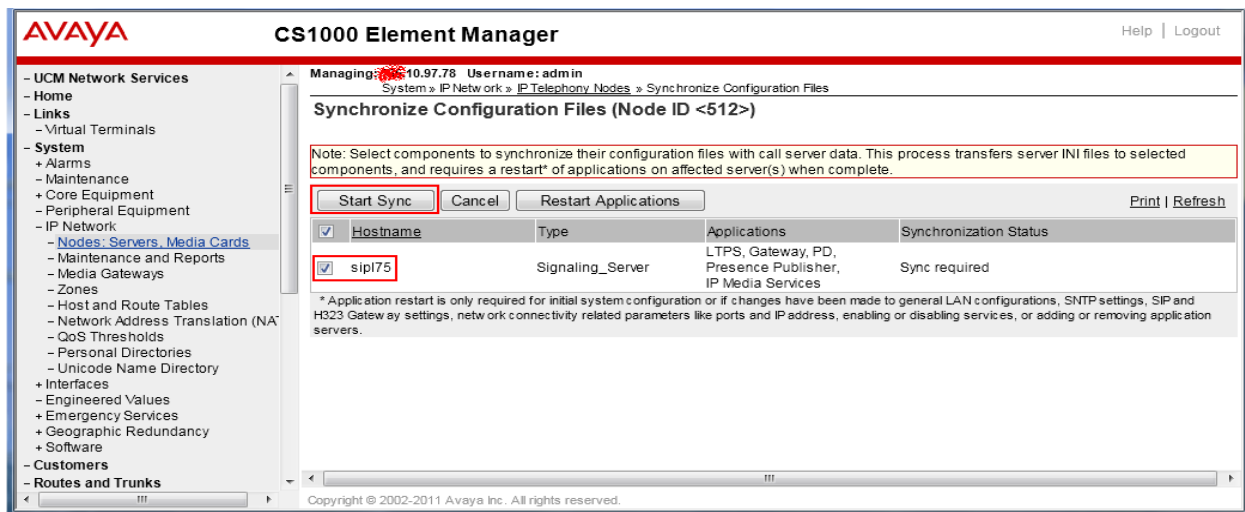


Figure 10: Synchronize Configuration Files

Note: The first time a new Telephony Node is added and transferred 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 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.

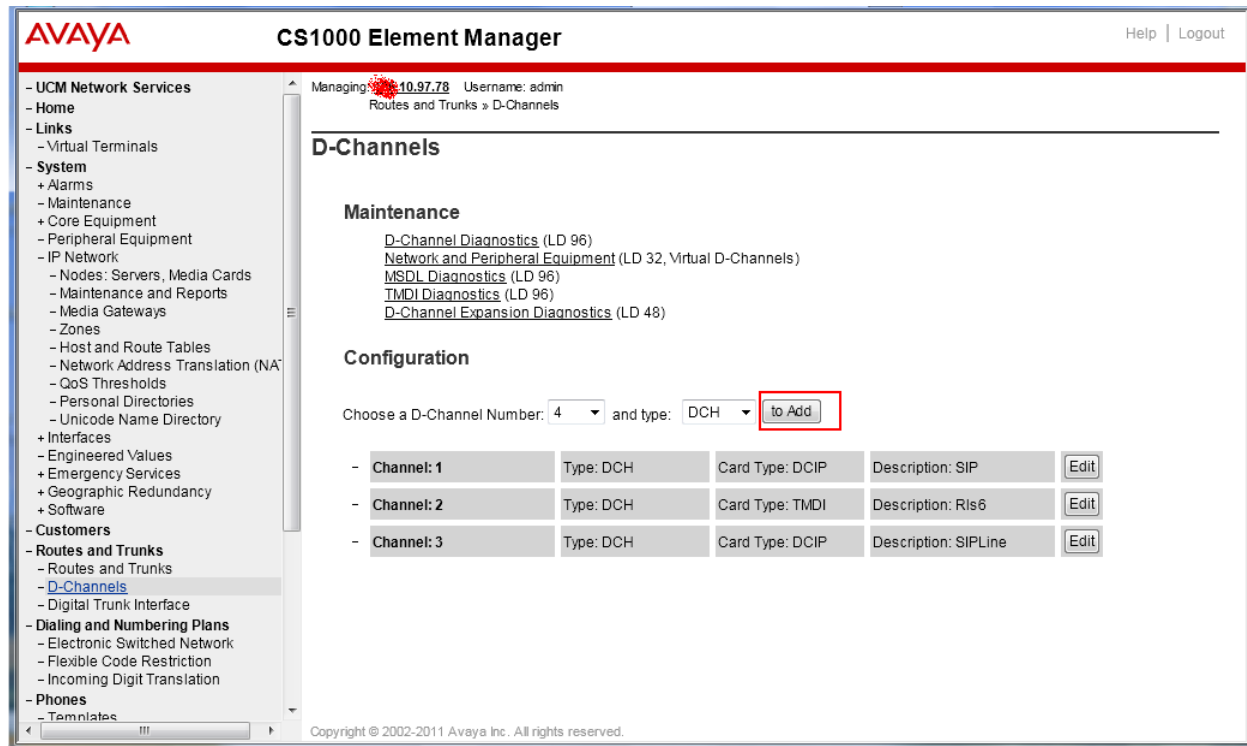


Figure 11: D-Channels configuration page

- 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.

AVAYA **CS1000 Element Manager** Help | Logout

Managing: 10.97.78 Username: admin
Routes and Trunks » D-Channels » D-Channels 3 Property Configuration

D-Channels 3 Property Configuration

- Basic Configuration

Input Description	Input Value
Action Device And Number (ADAN):	DCH
D channel Card Type:	DCIP
Designator:	SIPLine
Recovery to Primary:	<input type="checkbox"/>
PRI loop number for Backup D-channel:	
User:	Integrated Services Signaling Link Dedicated (ISLD) *
Interface type for D-channel:	Meridian Meridian1 (SL1)
Country:	ETS 300 =102 basic protocol (ETSI)
D-Channel PRI loop number:	
Primary Rate Interface:	<input type="button" value="more PRI"/>
Secondary PRI2 loops:	
Meridian 1 node type:	Slave to the controller (USR)
Release ID of the switch at the far end:	7
Central Office switch type:	100% compatible with Bellcore standard (STD)
Integrated Services Signaling Link Maximum:	4000 Range: 1 - 4000
Signalling server resource capacity:	3700 Range: 0 - 3700

[+ Basic options \(BSCOPT\)](#)
[+ Advanced options \(ADVOPT\)](#)
[+ Feature Packages](#)

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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.

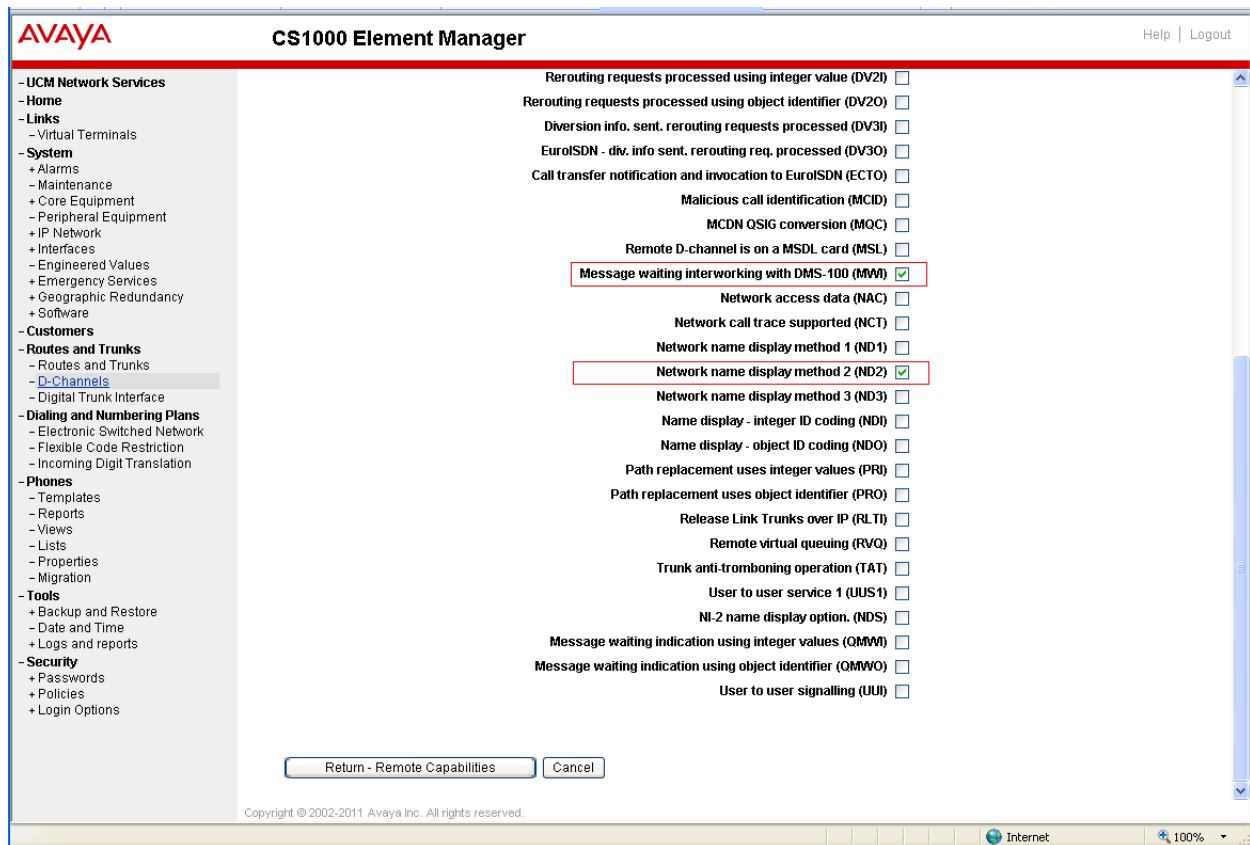


Figure 13: SIP Line D-Channel RCAP Configuration Details

- **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.

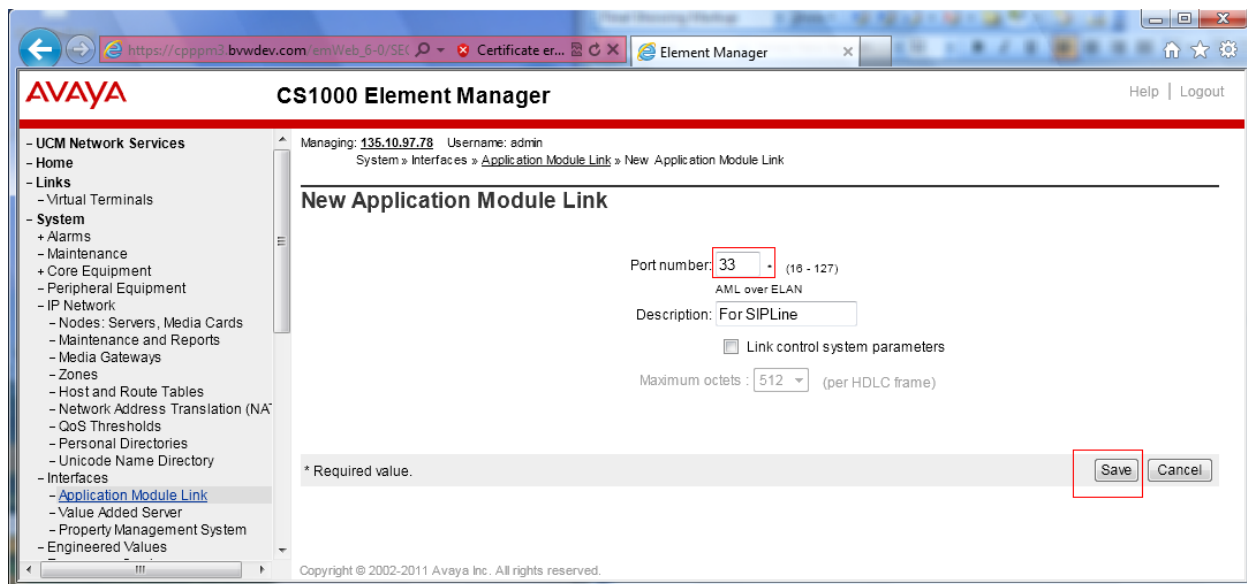


Figure 14: Adding a new AML

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.

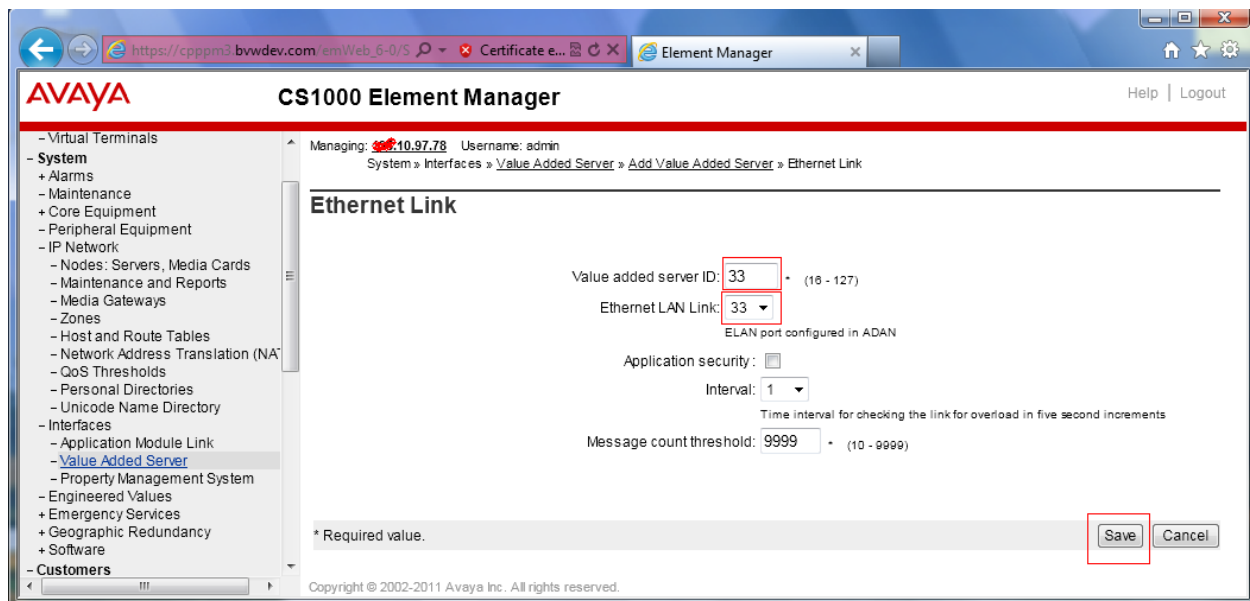


Figure 15: Adding a new Value Added Service for the AML

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.

Note: 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

Managing: 10.97.78 Username: admin
System » IP Network » Zones » Bandwidth Zones » Zone Basic Property and Bandwidth Management

Zone Basic Property and Bandwidth Management

Input Description	Input Value
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):	

* Required value.

Save Cancel

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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.

Click the **Submit** button to complete adding the route and save configuration.

The screenshot displays the AVAYA CS1000 Element Manager web interface. The left-hand navigation menu is expanded to show the 'Routes and Trunks' section, with 'Routes and Trunks' selected. The main configuration area is titled '- Basic Configuration' and contains the following fields and options:

- Route data block (RDB) (TYPE): RDB
- Customer number (CUST): 00
- Route number (ROUT): 3
- Designator field for trunk (DES): SIPLINE
- Trunk type (TKTP): TIE
- Incoming and outgoing trunk (ICOG): Incoming and Outgoing (IAO) (dropdown)
- Access code for the trunk route (ACOD): 7575
- Trunk type M911P (M911P): ☐
- The route is for a virtual trunk route (VTRK): ☒
- Zone for codec selection and bandwidth management (ZONE): 4 (0 - 8000)
- Node ID of signaling server of this route (NODE): 512 (0 - 9999)
- Protocol ID for the route (PCID): SIP Line (SIPL) (dropdown)
- Integrated services digital network option (ISDN): ☒
- Mode of operation (MODE): Route uses ISDN Signaling Link (ISLD) (dropdown)
- D channel number (DCH): 3 (0 - 254)
- Interface type for route (IFC): Meridian M1 (SL1) (dropdown)
- Private network identifier (PNI): 00001 (0 - 32700)
- Network calling name allowed (NCNA): ☒

The bottom of the interface shows a copyright notice: 'Copyright © 2002-2011 Avaya Inc. All rights reserved.' and a status bar with 'Internet' and '100%'.

Figure 117: SIP Line Route Configuration

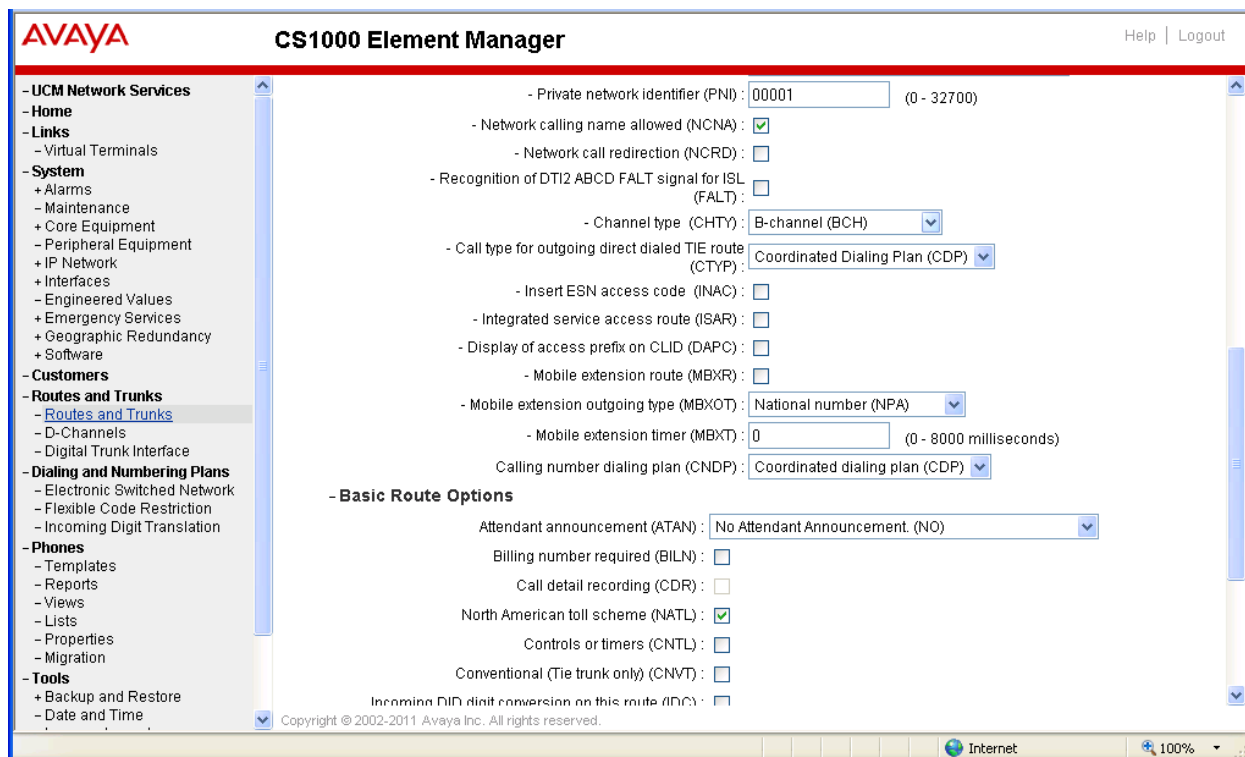


Figure 18: SIP Line Route Configuration (cont)

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.

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.

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 104 0 0 1

DES POLY1 -> Description of Phone.
CUST0
```

UXTY SIPL -> Universal extension type is SIP Line
MCCL YES
 SIPN 0
SIP3 1 -> 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 8450

This section describes how to access the Polycom SpectraLink 8450 SIP endpoint web interface and configure the Polycom 8450 for testing. For more information on how to configure the Polycom SpectraLink 8450 phone connected to the Access Point Wi-Fi router, please refer to the document in the **Section 9**.

6.1. Login Polycom SpectraLink 8450

This section shows how to log in to the home page of Polycom SpectraLink 8450 to manage and configure the 8450 phone.

Open the web browser, and in the address field enter the Polycom SpectraLink 8450 IP address: <http://ipaddress> and the Polycom SpectraLink 8450 login page will appear as shown in **Figure 20**. Enter the username, **Polycom**, and its default password, **456**.

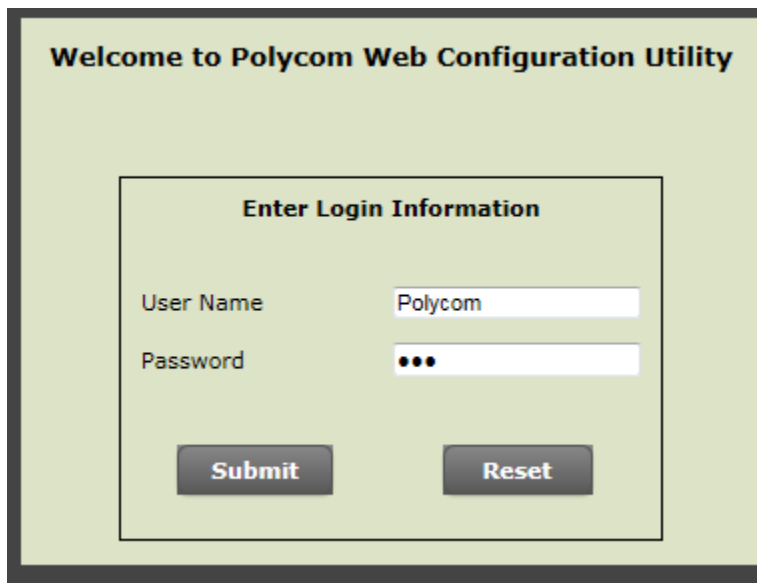
The image shows a web browser window displaying the Polycom Web Configuration Utility login screen. The page has a light green background with a dark border. At the top, it says "Welcome to Polycom Web Configuration Utility". Below this is a white box with a black border titled "Enter Login Information". Inside this box, there are two input fields: "User Name" with the text "Polycom" entered, and "Password" with three black dots entered. Below the input fields are two buttons: "Submit" and "Reset".

Figure 20: Polycom SpectraLink 8450 Login Screen

Click the **Submit** button, the homepage of Polycom SpectraLink 8450 appears as in **Figure 21** below.

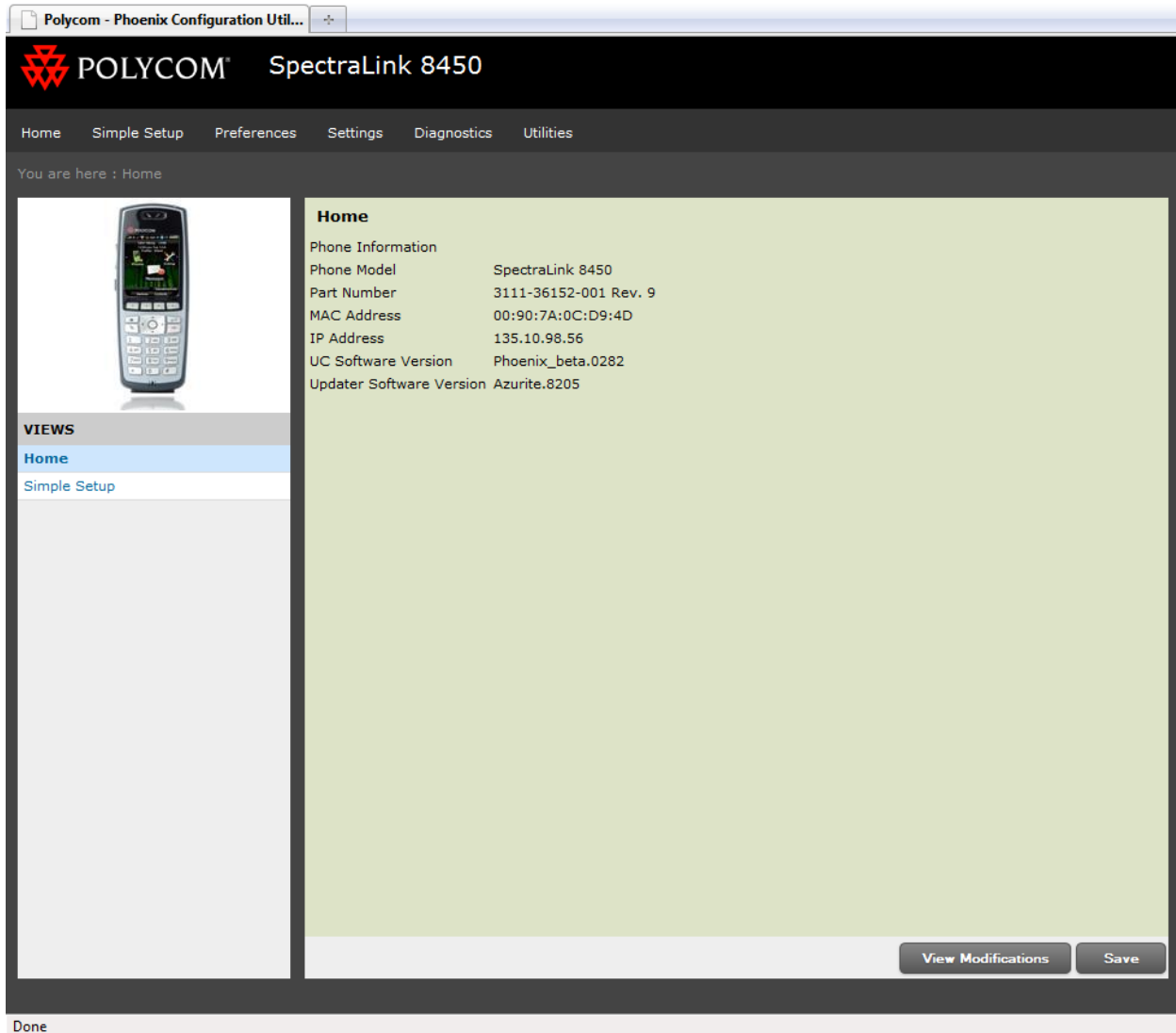


Figure 21: Home page of Polycom SpectraLink 8450 telephone

6.2. Configure the Lines for Polycom SpectraLink 8450

This section shows how to configure the Polcom 8450 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 8450
 - **Address:** 54008@sip175.com
 - **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 8450

Home Simple Setup Preferences Settings Diagnostics Utilities

You are here : Simple Setup

Simple Setup

Language
Phone Language: English (Internal)

Time Synchronization
SNTP Server: [Dropdown]
Time Zone: (GMT -5:00) Eastern Time (US & Canada), Bogota, Lima

SIP Server
Address: 135.10.97.187
Port: 5070

SIP Outbound Proxy
Address: 135.10.97.187
Port: 5070

SIP Line Identification
Display Name: Poly8450
Address: 54009@sip175.com
Authentication User ID: 54009
Authentication Password: [Masked]
Label: [Empty]

Reset to Default Cancel

View Modifications Save

Figure 22: Simple Setup for Polycom 8450 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 8450 telephone.

On the homepage of Polycom 8450 (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**.

Note: The “Server Call Forward Always” setting for the Polycom 8450 on the CS 1000 Call Server must be OFF in order to make the “Local Call Forward Always” on the Polycom 8450 take effect.

The screenshot displays the Polycom SpectraLink 8450 web interface. The top navigation bar includes links for Home, Simple Setup, Preferences, Settings, Diagnostics, and Utilities. The breadcrumb trail indicates the current location: Settings -> Lines -> Line 1. On the left, a 'VIEWS' sidebar lists Line 1 through Line 6, with Line 1 selected. The main content area is titled 'Line 1' and contains several expandable sections: Identification, Outbound Proxy, Server 1, Server 2, and Call Diversion. The 'Call Diversion' section is expanded, showing various settings with radio buttons for 'Enabled' or 'Disabled' and text input fields for contact numbers. The settings include: 'Disabled On Shared' (radio buttons), 'Diversion Contact' (text field), 'On Specific Caller' (radio buttons), 'Forward All' (radio buttons), 'On Busy' (radio buttons), 'Busy Contact' (text field with '54444'), 'On No Answer' (radio buttons), 'No Answer Timeout' (text field with '55'), 'No Answer Contact' (text field with '54008'), 'On Do-Not-Disturb' (radio buttons), and 'Do-Not-Disturb Contact' (text field). Below these settings is a 'Message Center' section. A red 'Note' states: '* Fields require phone reboot/restart'. At the bottom right, there are four buttons: 'Reset to Default', 'Cancel', 'View Modifications', and 'Save'.

Line 1	
Call Diversion	
* Disabled On Shared	<input checked="" type="radio"/> Yes <input type="radio"/> No
* Diversion Contact	<input type="text"/>
* On Specific Caller	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
* Forward All	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
* On Busy	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
* Busy Contact	<input type="text" value="54444"/>
* On No Answer	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
* No Answer Timeout	<input type="text" value="55"/>
* No Answer Contact	<input type="text" value="54008"/>
* On Do-Not-Disturb	<input type="radio"/> Enabled <input checked="" type="radio"/> Disabled
* Do-Not-Disturb Contact	<input type="text"/>

Note:
* Fields require phone reboot/restart

Reset to Default Cancel View Modifications Save

Figure 23: Call Diversion section of Polycom 8450

6.4. Codec settings

This section shows how to set the Codec on the Polycom SpectraLink 8450 phone. The compliance testing has been done on both codecs, G711 and G729.

On the homepage of Polycom SpectraLink 8450 (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.

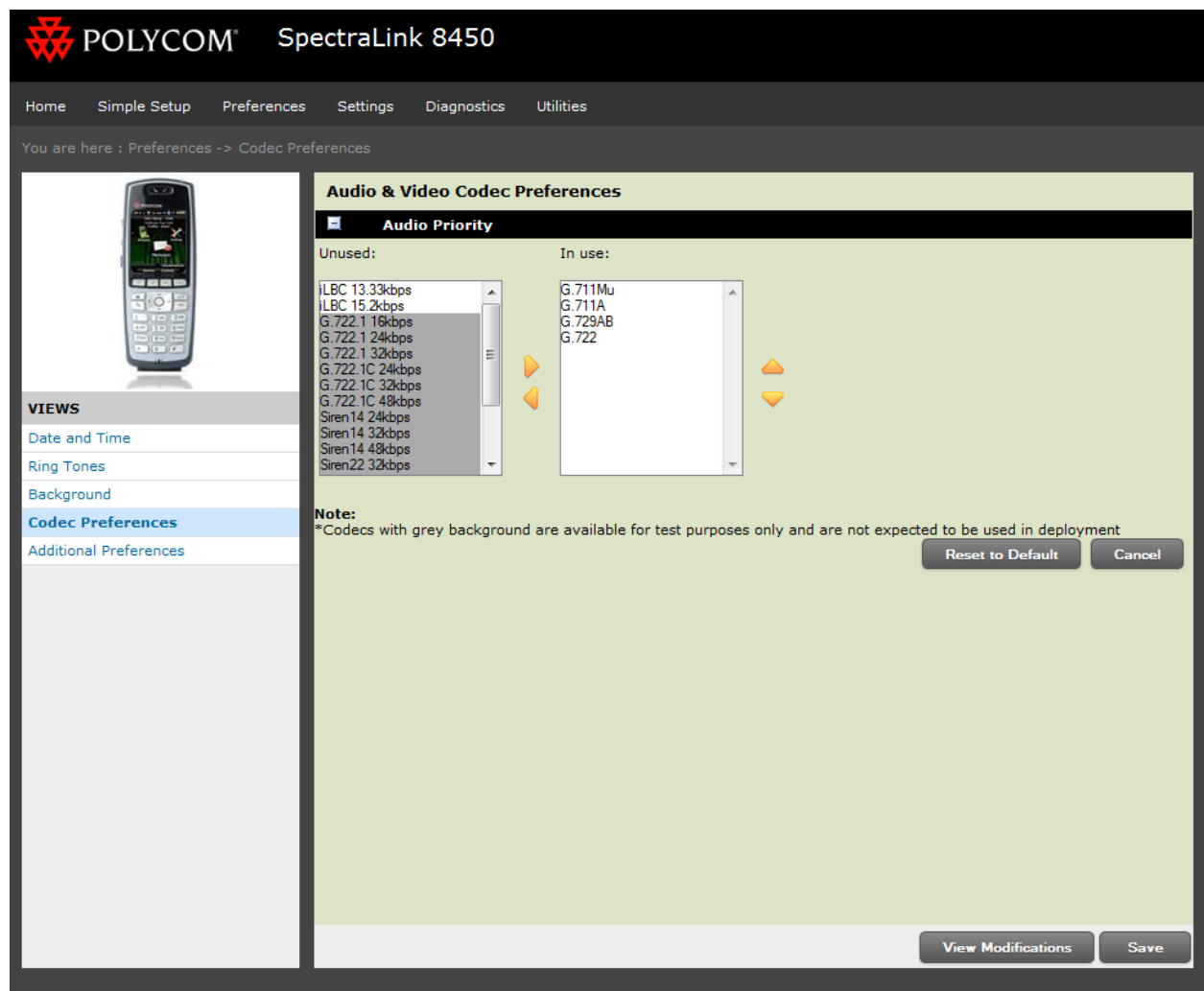


Figure 24: Audio & Video Codec Preferences

7. Verification Steps

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

- Verify that the Poycom SpectraLink 8450 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@sip1 ~]$ slgSetShowByUID 54009
=== VTRK ===
UserID          AuthId          TN          Clients  Calls
SetHandle  Pos ID      SIPL Type
-----
-----
          54009          54008      104-00-00-03          1          0
0x8fc4cf8          SIP Lines
      StatusFlags = Registered Controlled KeyMapDwld SSD
      FeatureMask =
      CallProcStatus = 0

      Current Client = 0, Total Clients = 1

      == Client 0 ==
      IPv4:Port:Trans = 10.10.98.57:5060:udp
      Type            = SIP3
      UserAgent        = PolycomSpectraLink-SL_8450-
UA/Phoenix_beta.0282
      x-nt-guid        = 267d228547c1562399f1f743a2971fb5
      RegDescrip        =
      RegStatus         = 1
      PbxReason         = OK
      SipCode           = 200
      hTransc           = (nil)
      Expire            = 3600
      Nonce             = f56a9946ba497bde7eb445efb518f4f1
      NonceCount        = 2
      hTimer            = 0x8f64e60
      TimeRemain        = 1338
      Stale             = 0
      Outbound          = 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     54009
      1   126    0     2654009
      2    9     0
      3   29     0
      4   22     0
      5    2     0     54334
```

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

```

== Subscription Info ==
Subscription Event = None
Subscription Handle = (nil)
SubscribeFlag = 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 3
IDLE REGISTERED 00

```

- Place a call from and to Polycom SpectraLink 8450 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 8450 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:

<https://support.avaya.com/css/Products/>

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](#)

Avaya Communication Sever 1000 Co-resident Call Server and Signaling Server
Fundamentals

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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