Avaya Solution & Interoperability Test Lab

# 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.

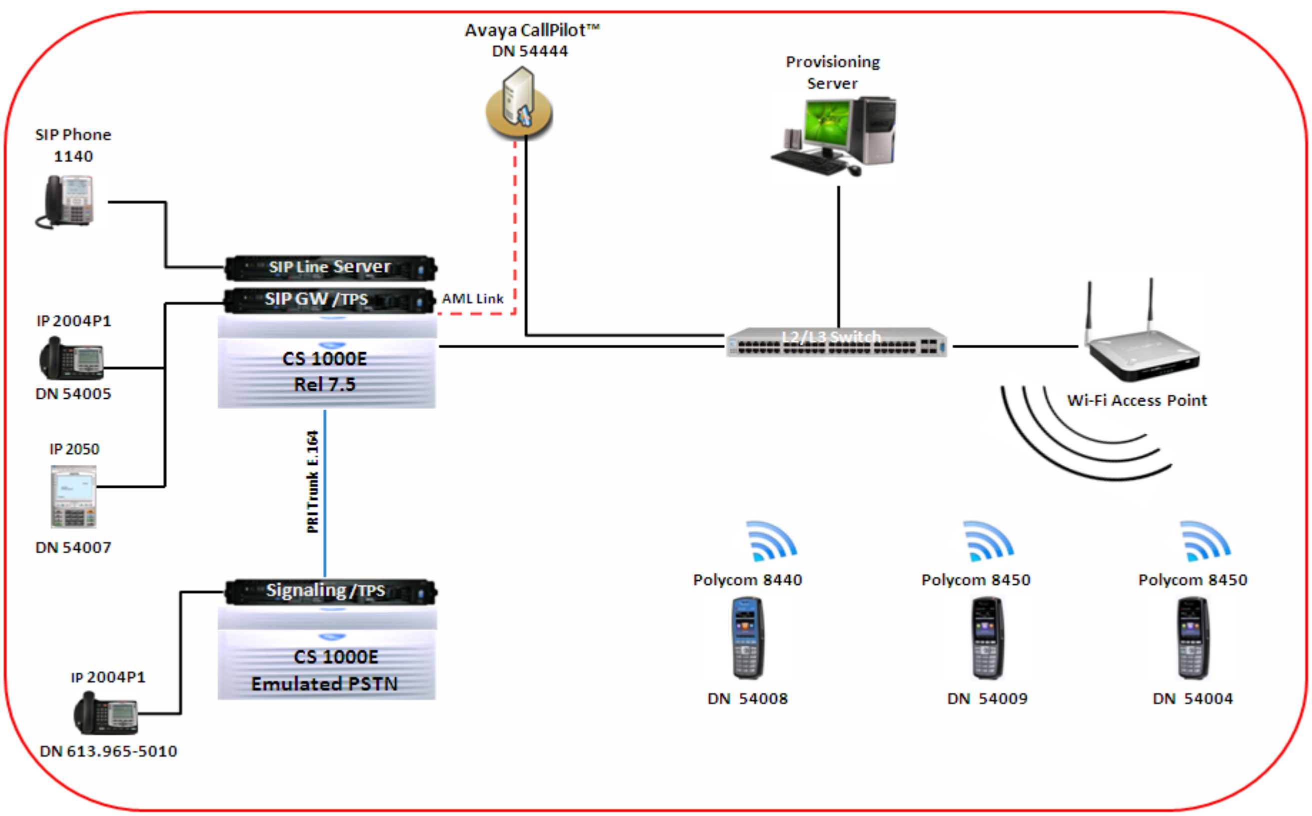

**Figure 1: Network Configuration Diagram**

# 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<br>Signaling Server (CPPM): 7.50.17 |
| Avaya CallPilot™ Messaging System | 5.0.1 |
| Avaya IP Soft Phone 2050<br>Avaya IP Phone 1140<br>Avaya IP Phone 2004P2<br>Avaya IP Phone 2002P2 | 3.04.0003<br>0625C6O<br>0692D93<br>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](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.

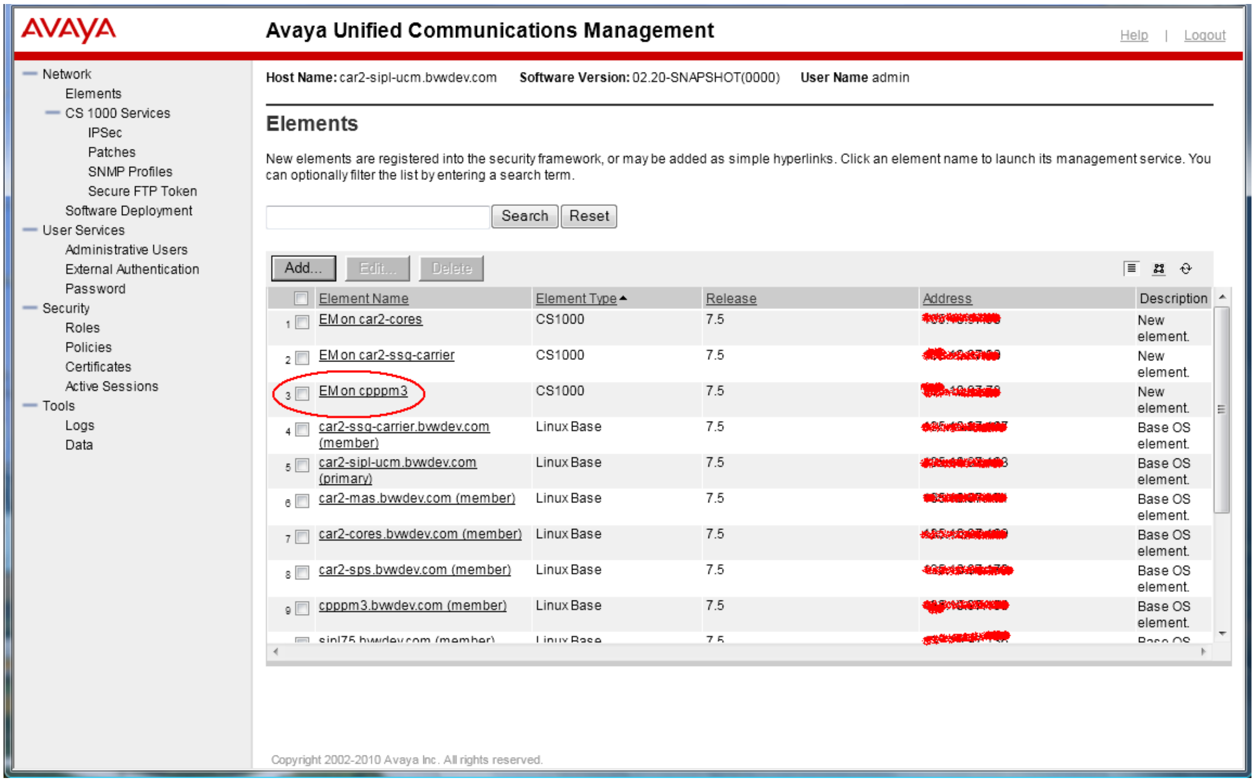


**Figure 2: The UCM Home Page of CS 1000 Release 7.5**

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.

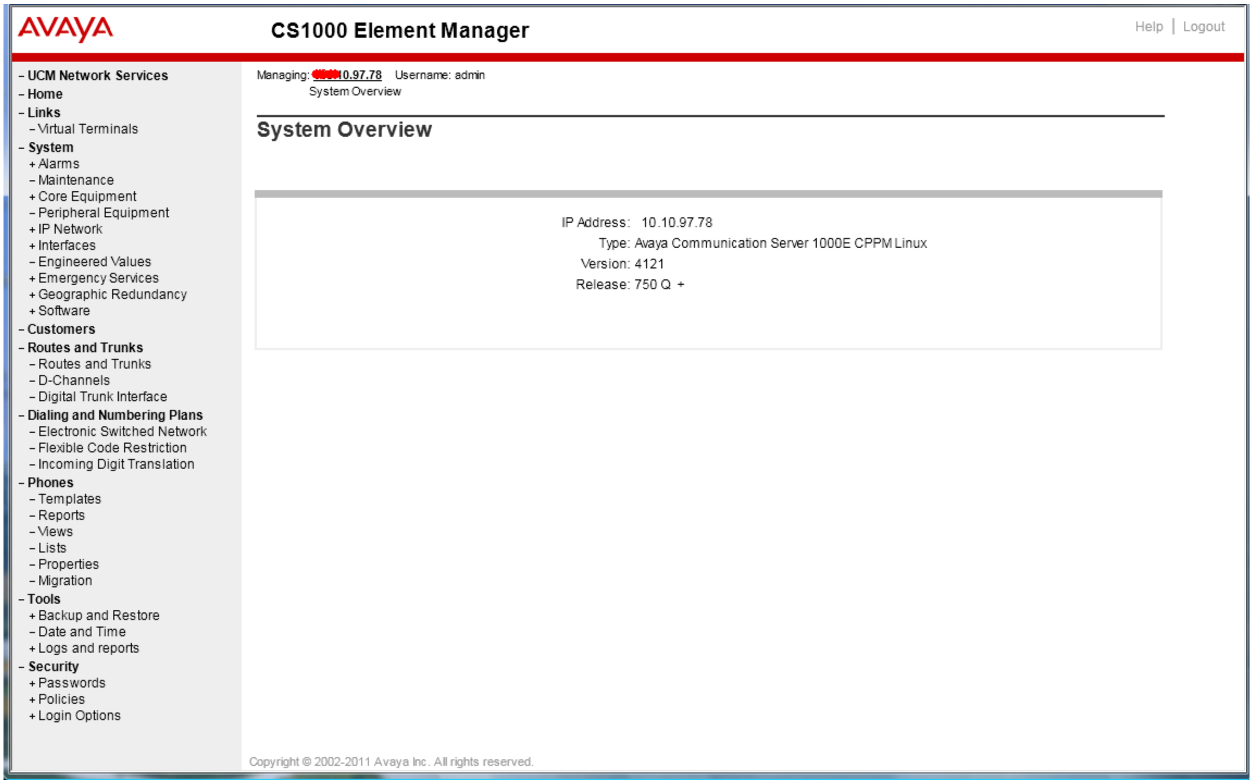


**Figure 3: CS 1000 Release 7.5 EM Home Page**

## 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**.

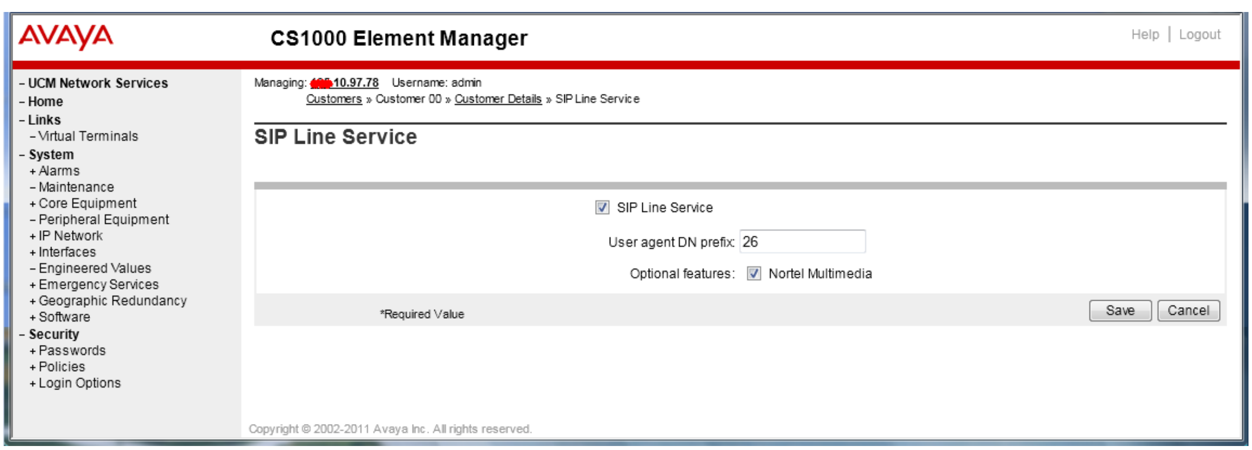


**Figure 4: SIP Line Service in Customers Data Block**

## 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.

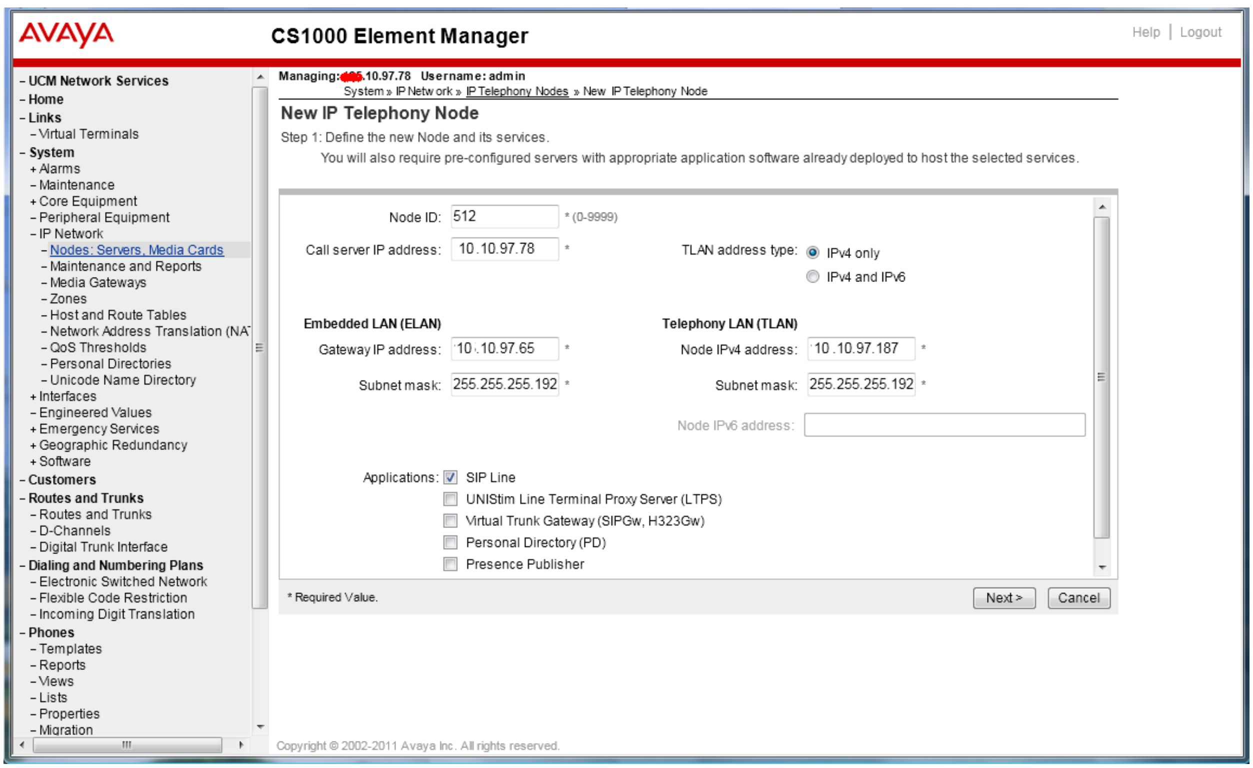


**Figure 5: Adding a New IP Telephony Node**

- 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).

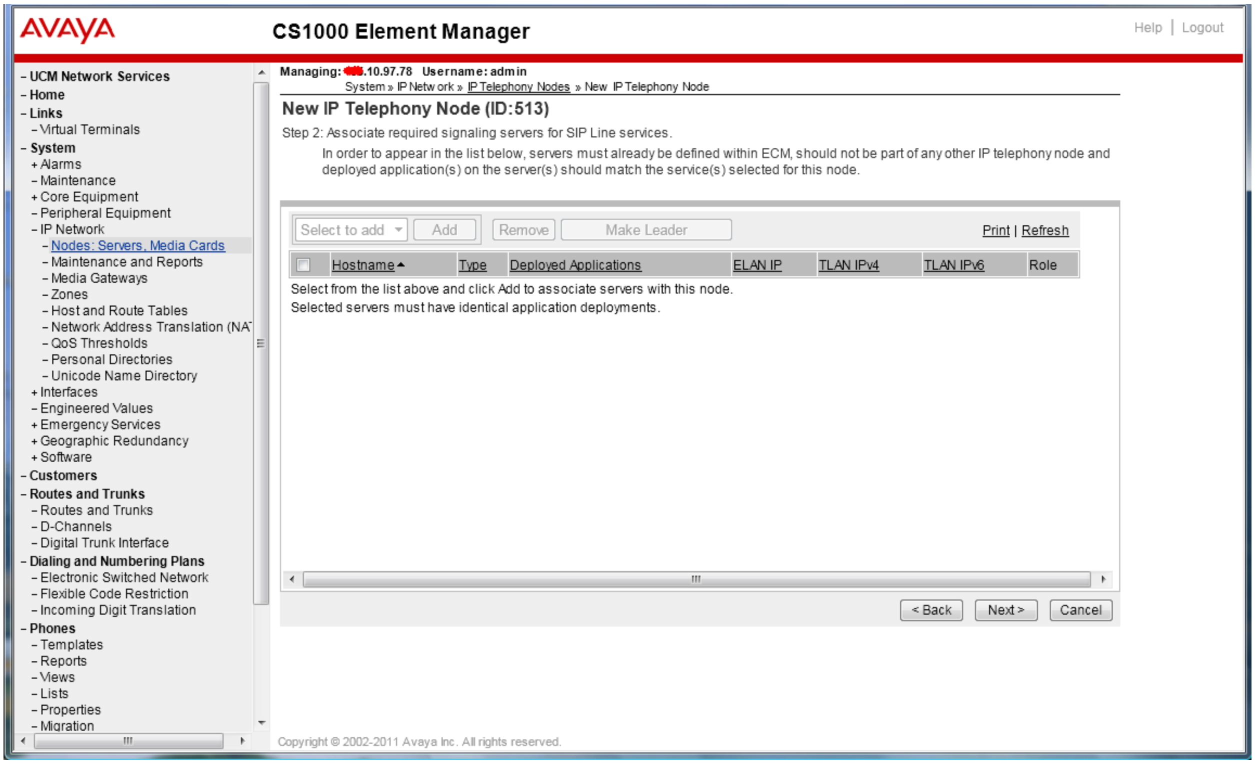


**Figure 6: Adding a New IP Telephony Node (cont)**

- 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 **sipl75.com**.

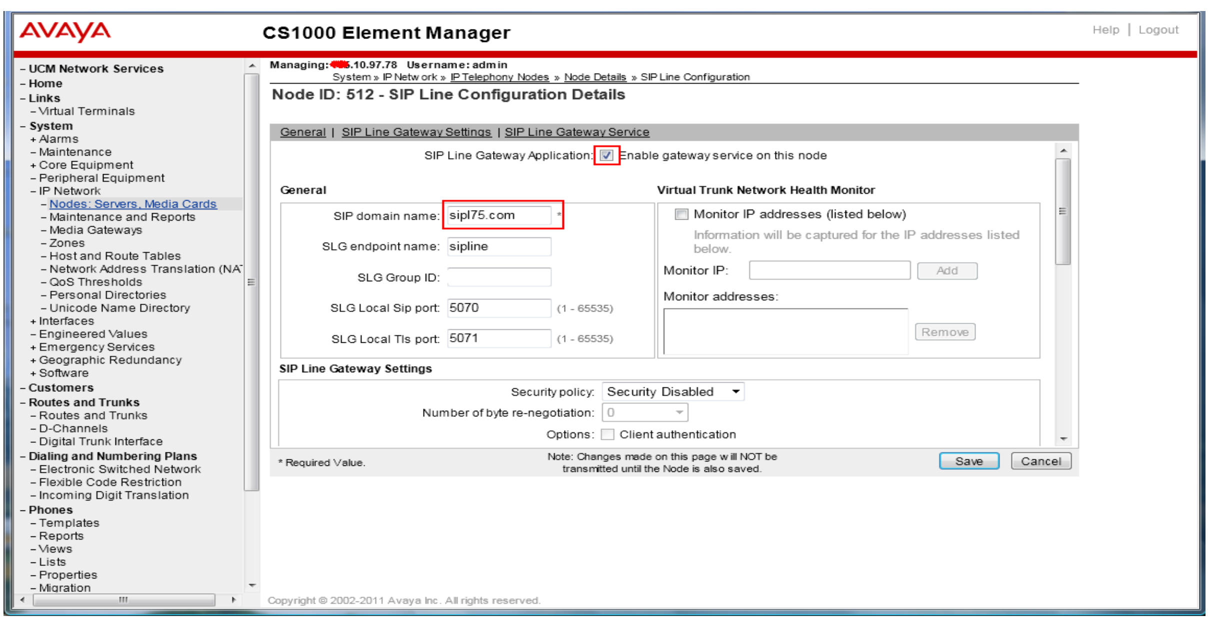


**Figure 7: Adding a new IP Telephony Node (cont)**

- 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**.

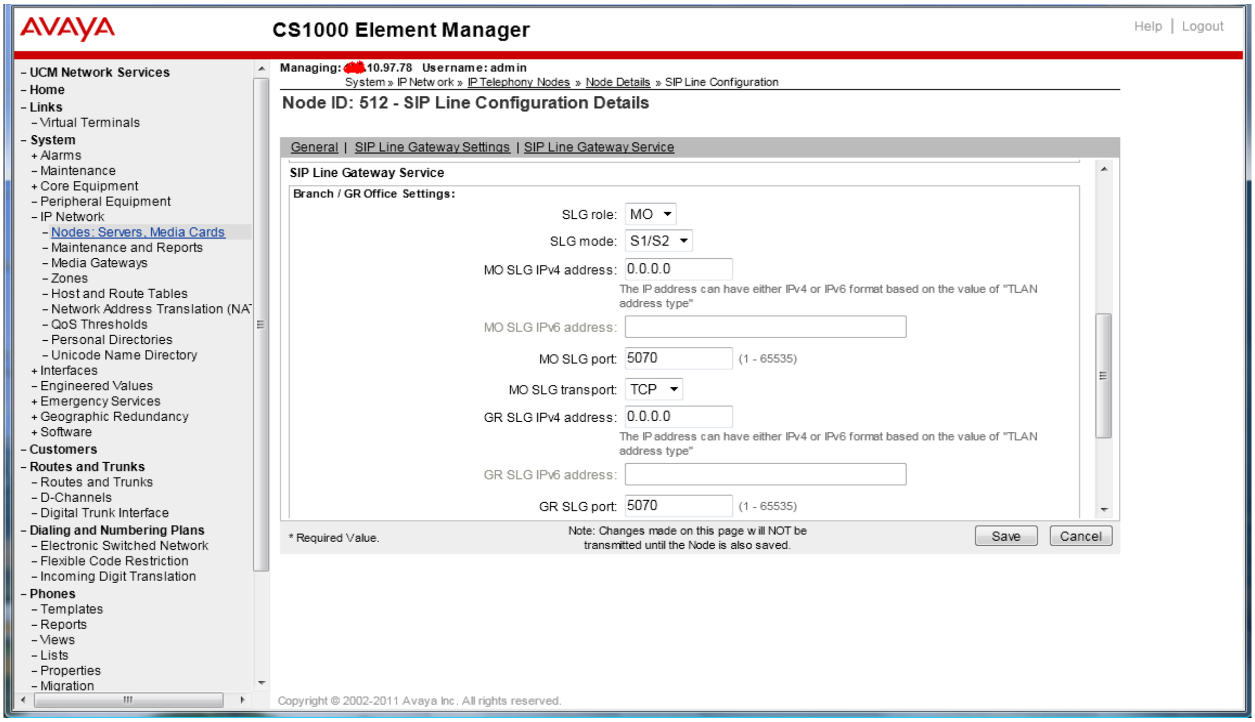


**Figure 8: Adding a new IP Telephony Node (cont)**

- 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**.

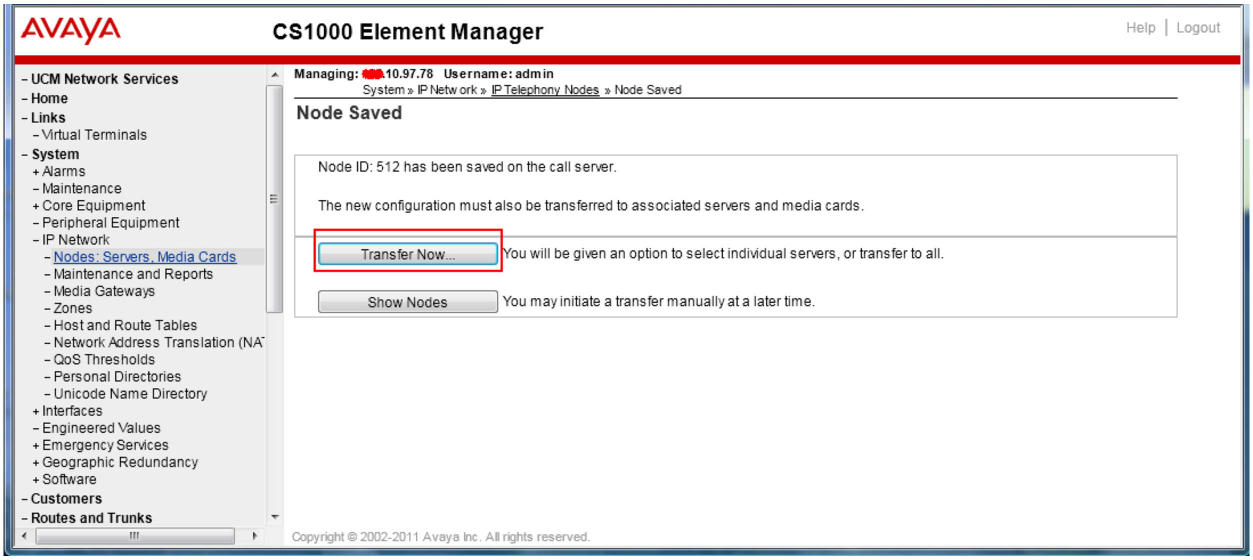


**Figure 9: Node Saved with Transfer Configuration**

- 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**.

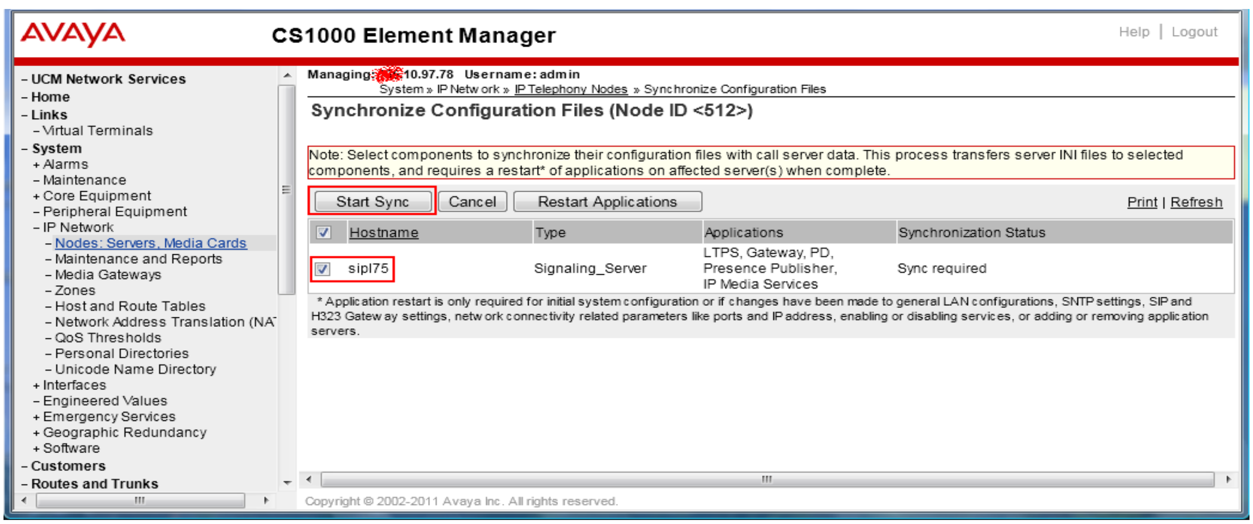


**Figure 10: Synchronize Configuration Files**

<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 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.

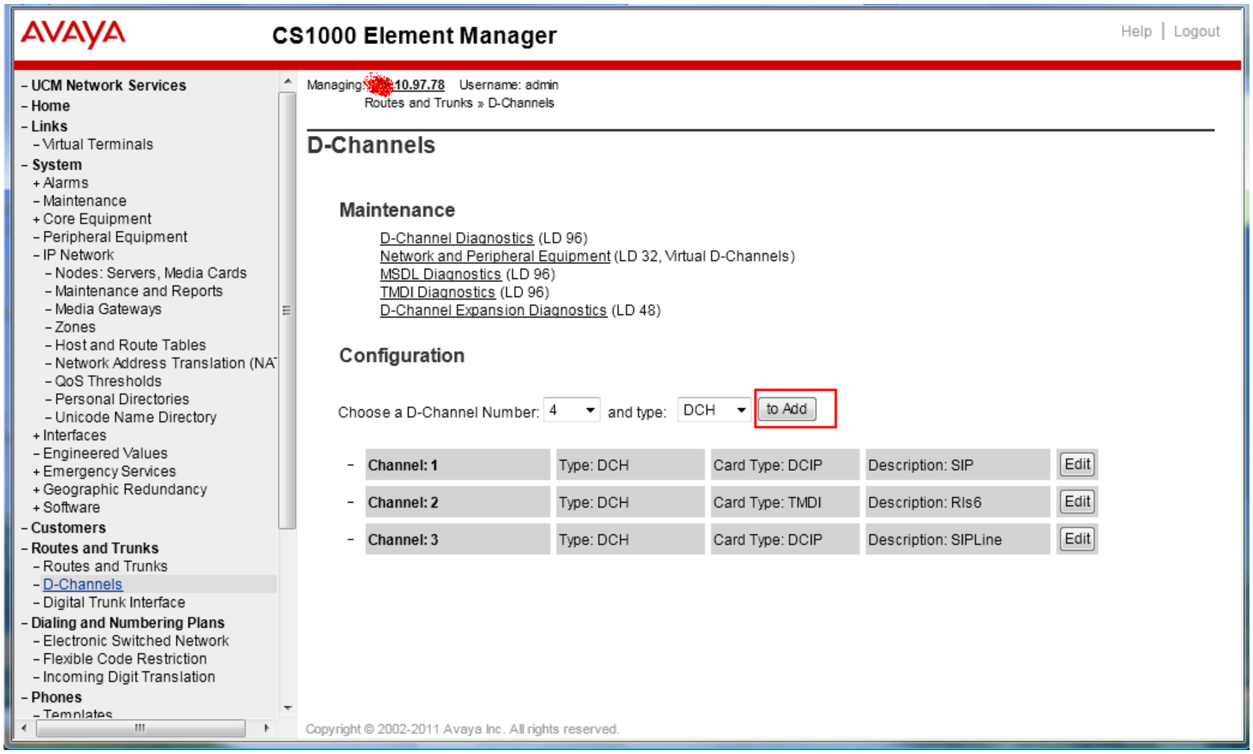


**Figure 11: D-Channels configuration page**

- 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.

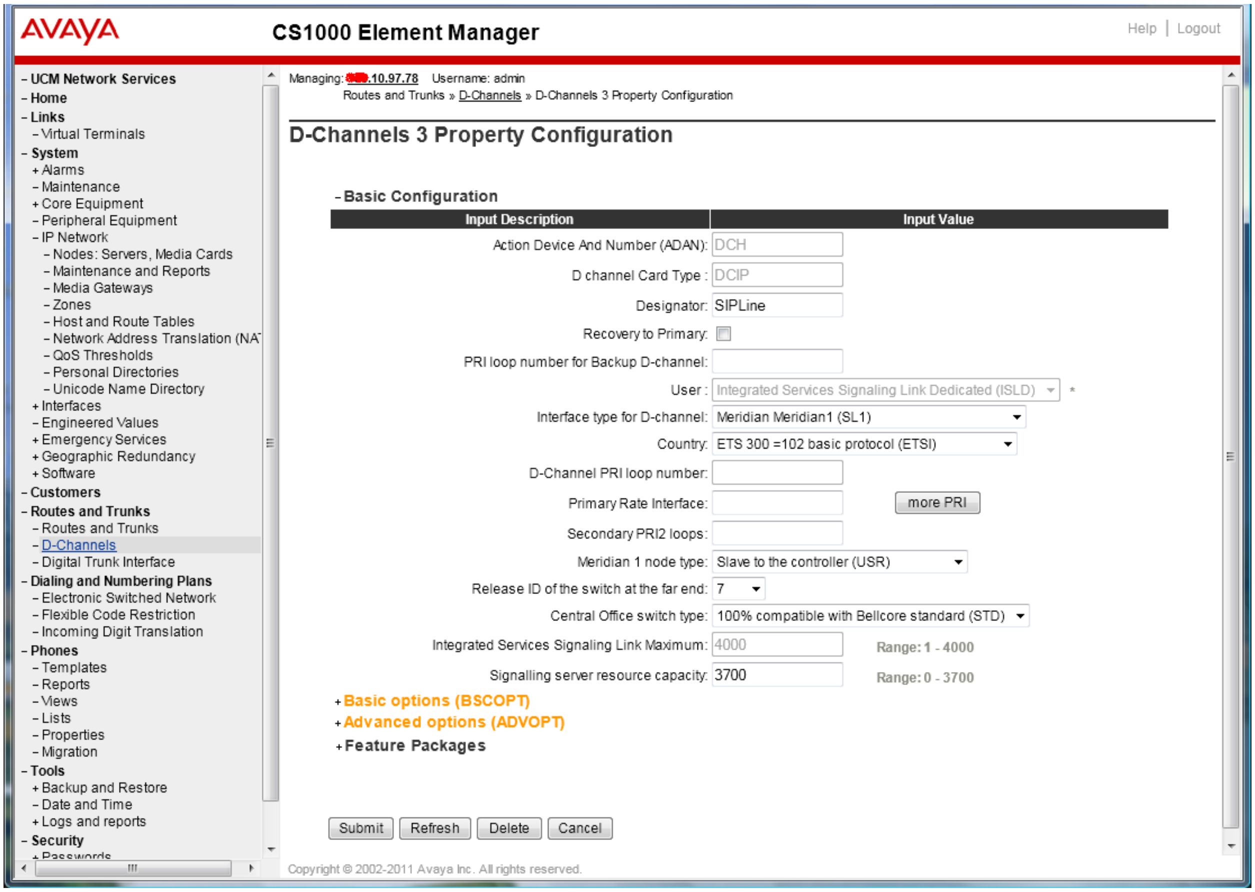


**Figure 12: SIP Line D-Channel Property Configuration**

- 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.

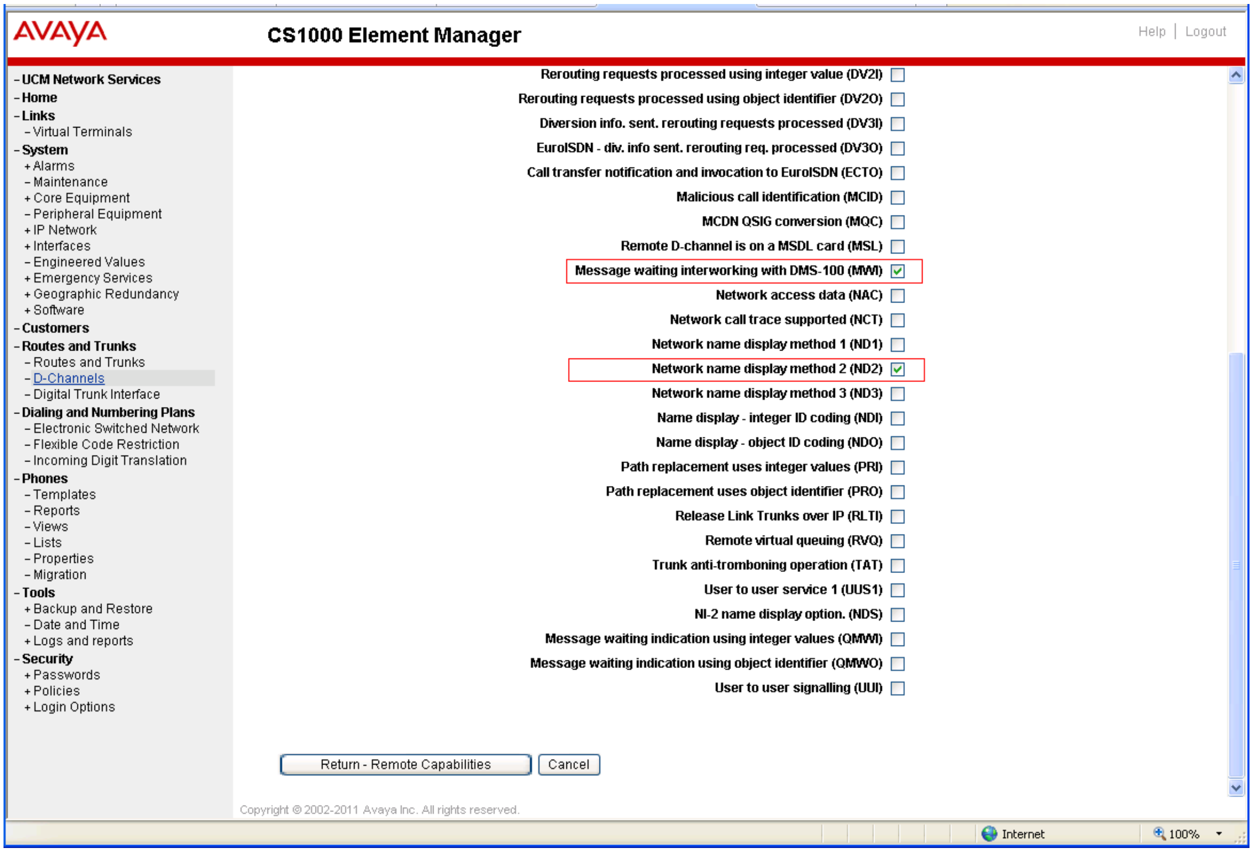


**Figure 13: SIP Line D-Channel RCAP Configuration Details**

- **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.

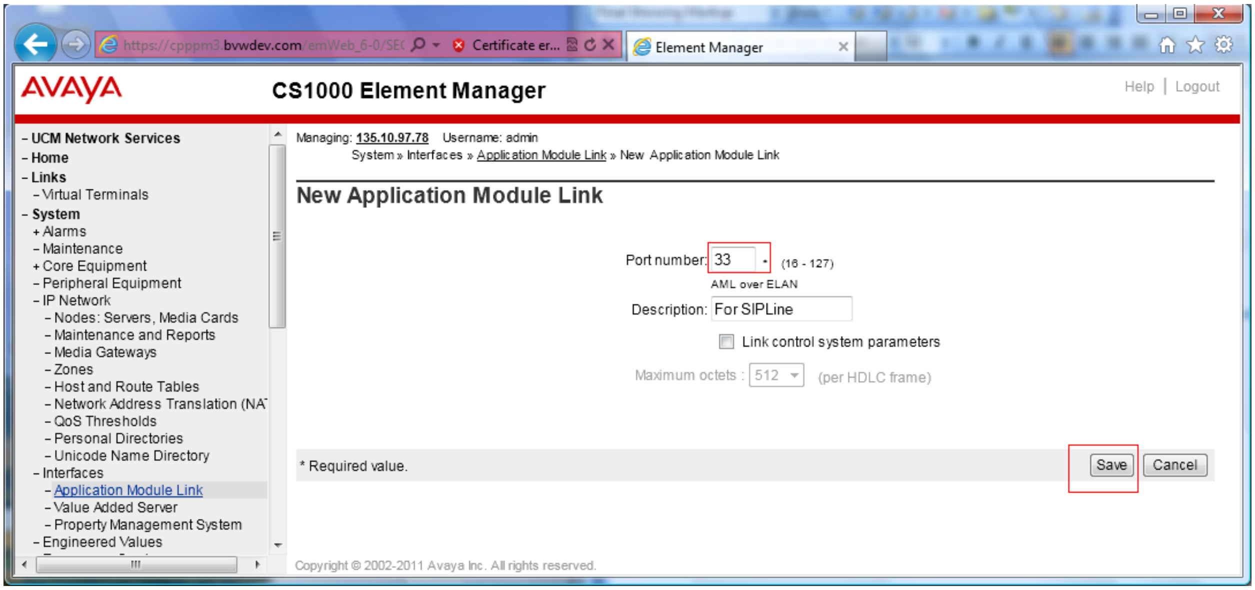


**Figure 14: Adding a new AML**

## 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.

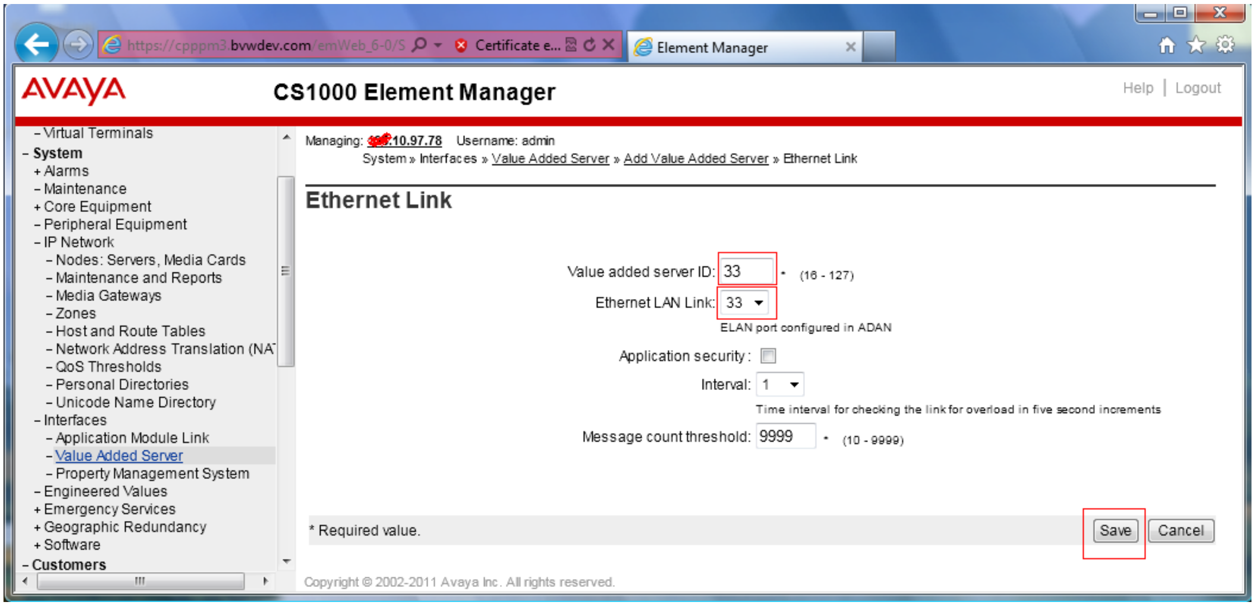


**Figure 15: Adding a new Value Added Service for the AML**

## 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**.

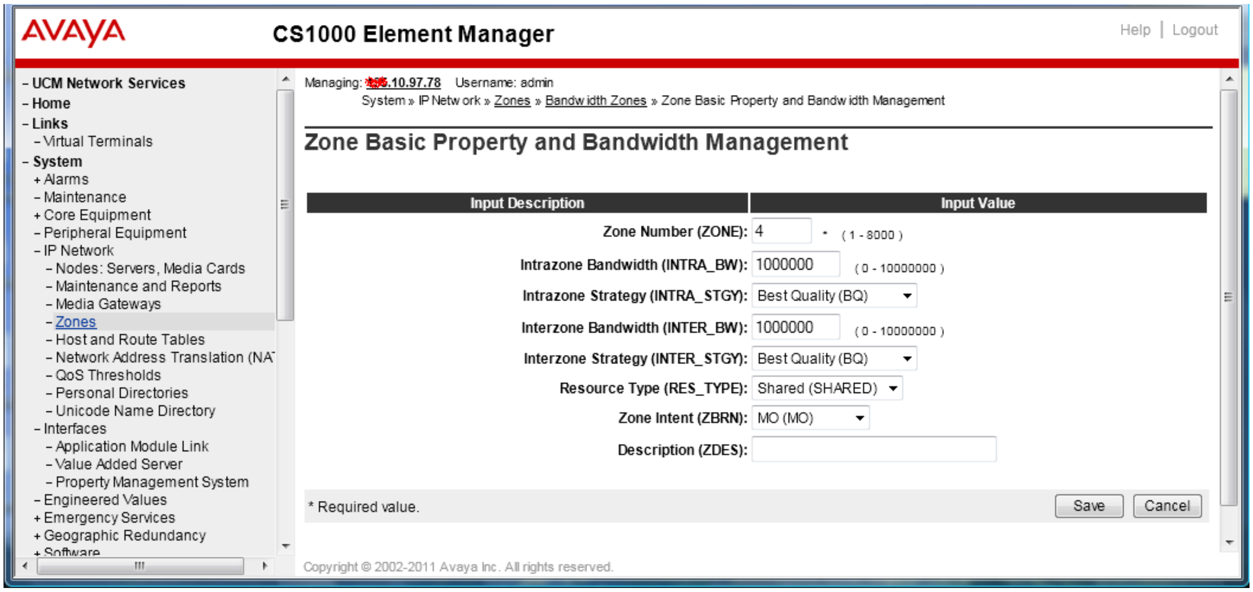


**Figure 16: Adding a new Zone for Virtual Trunk**

## 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.

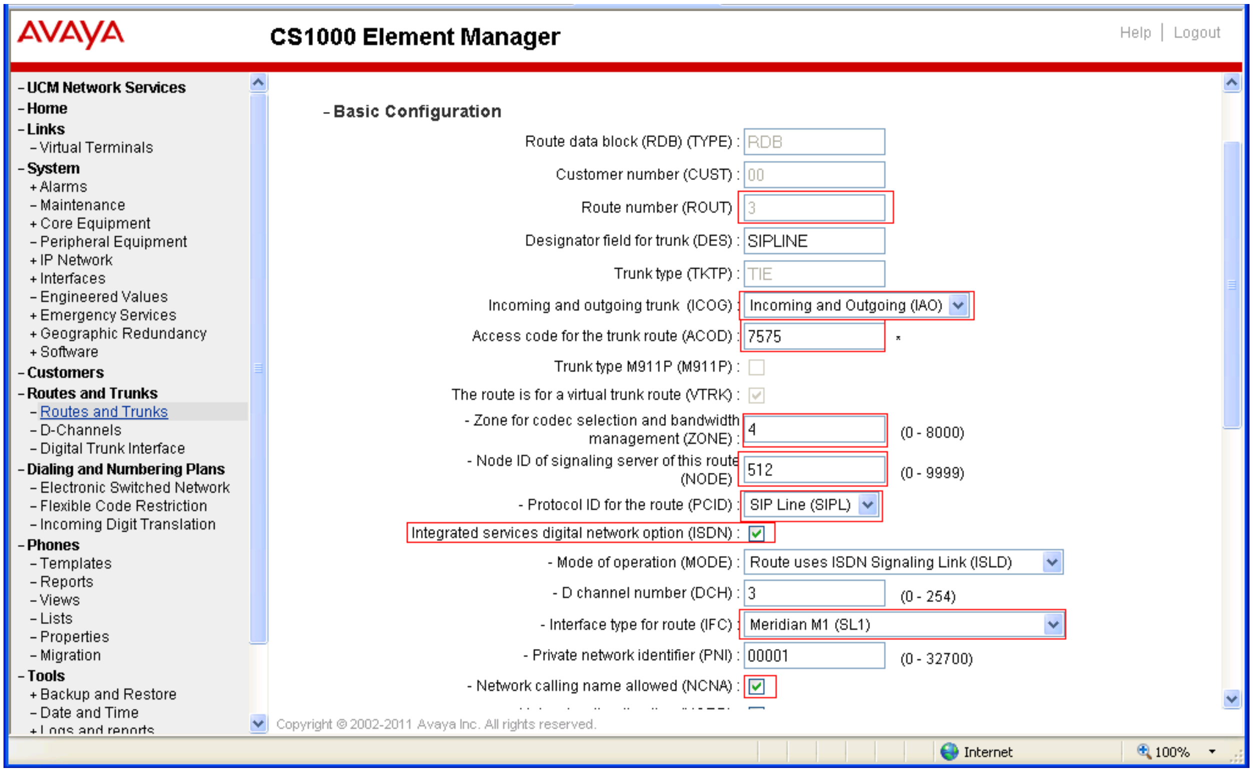


**Figure 117: SIP Line Route Configuration**

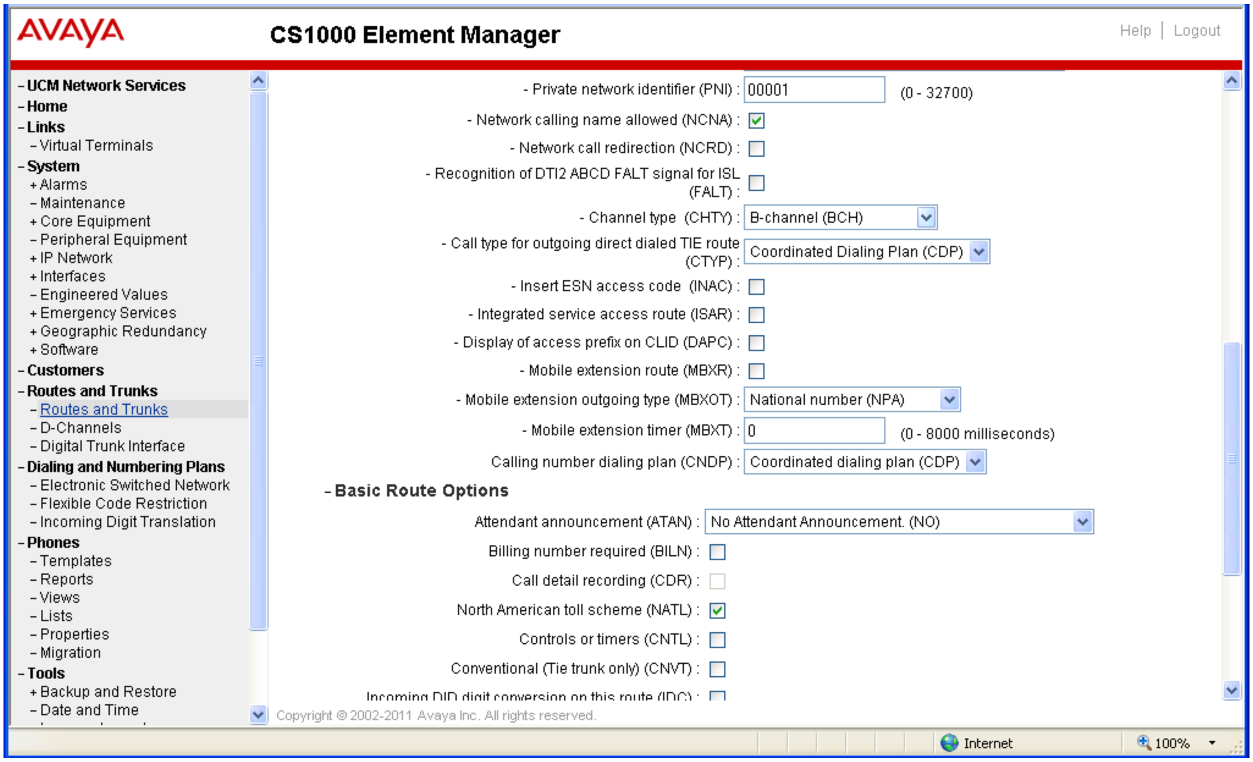


**Figure 18: SIP Line Route Configuration (cont)**

## 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.

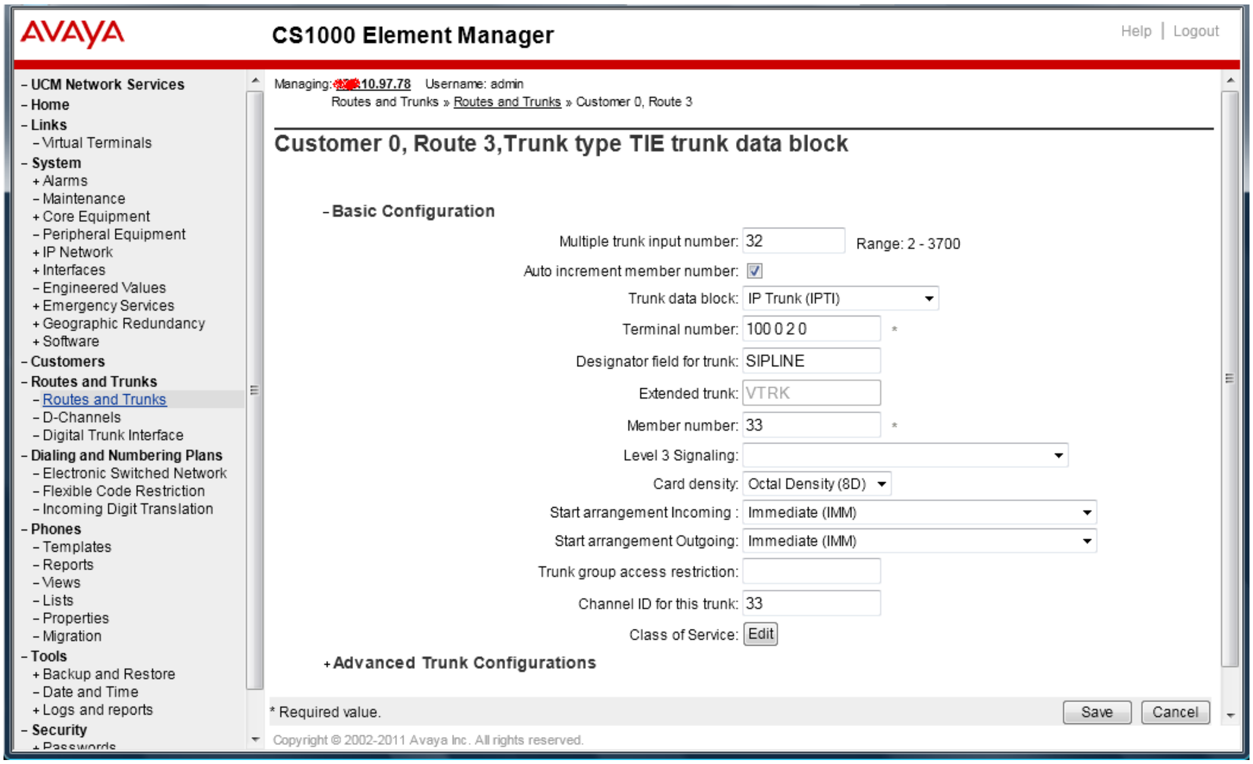


**Figure 19: Adding virtual trunks for SIP Line Trunk**

## 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.
CUST 0
```

**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**.

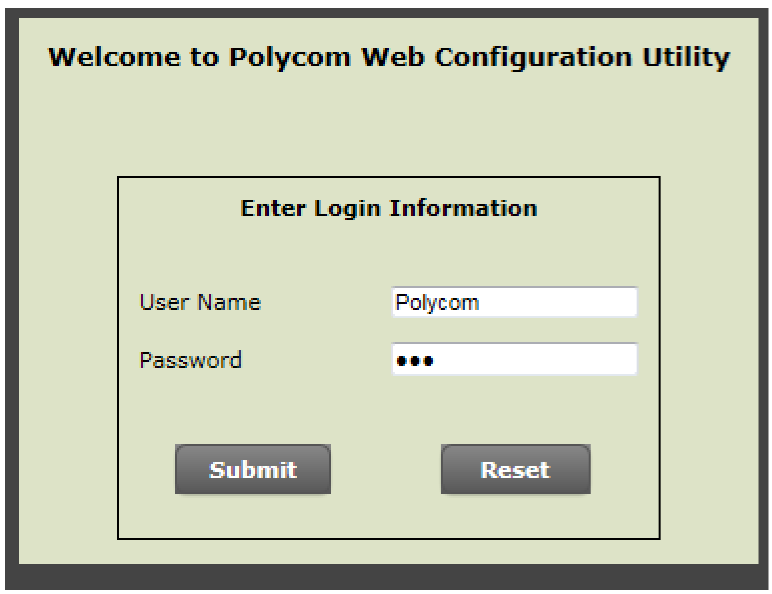


**Figure 20: Polycom SpectraLink 8450 Login Screen**

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

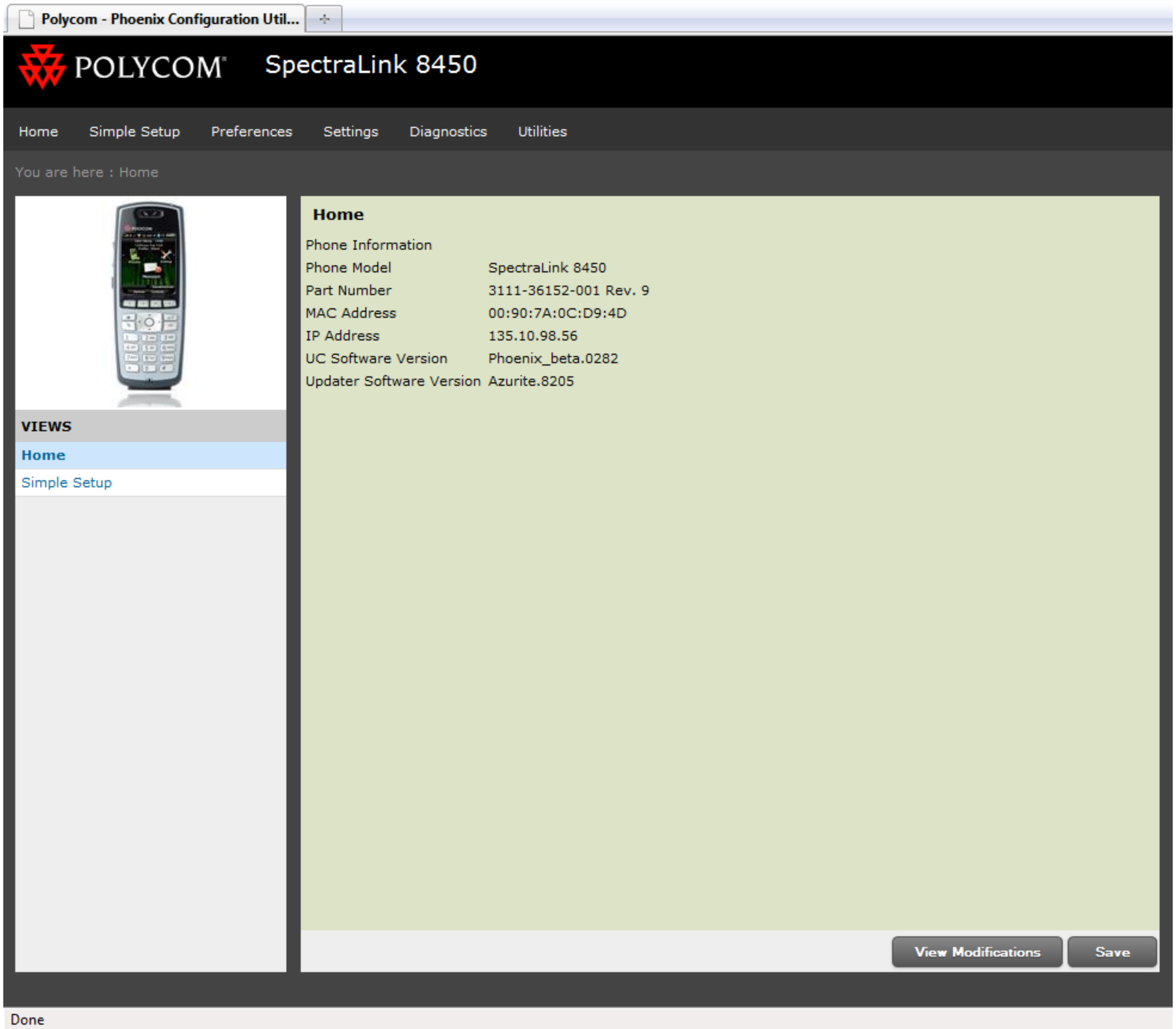


**Figure 21: Home page of Polycom SpectraLink 8450 telephone**

## 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@sipl75.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.

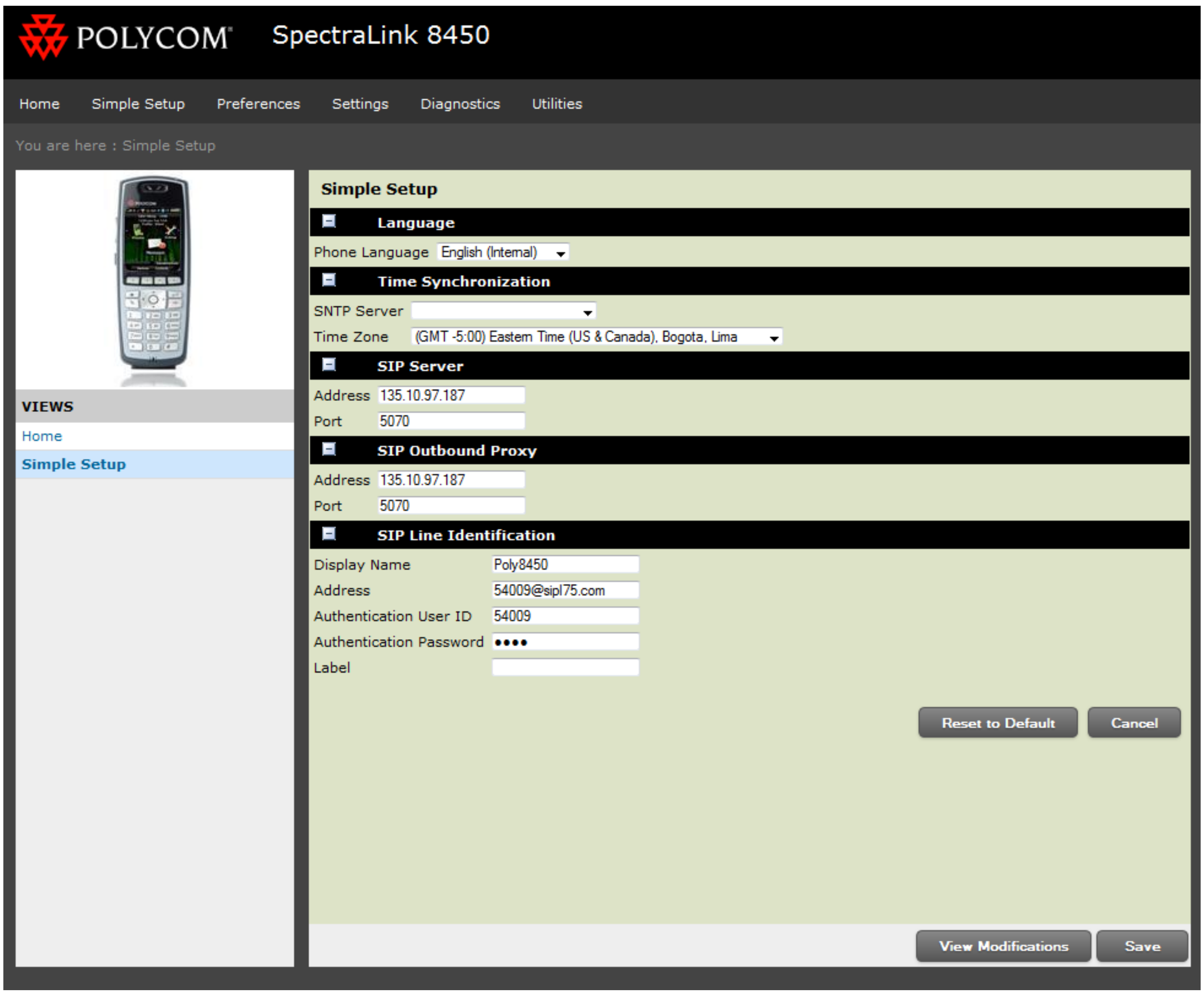


**Figure 22: Simple Setup for Polycom 8450 Phone**

## 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 Contac**t 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 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.

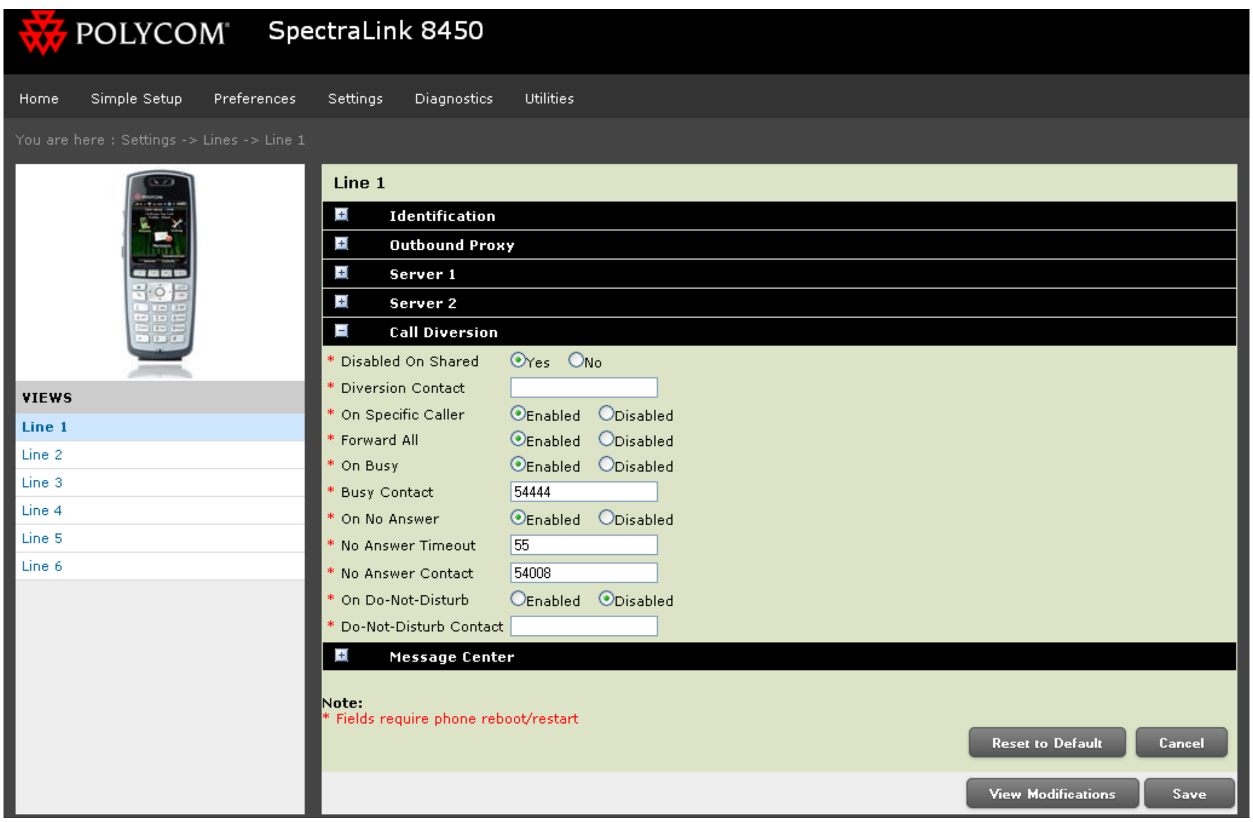


**Figure 23: Call Diversion section of Polycom 8450**

## 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.

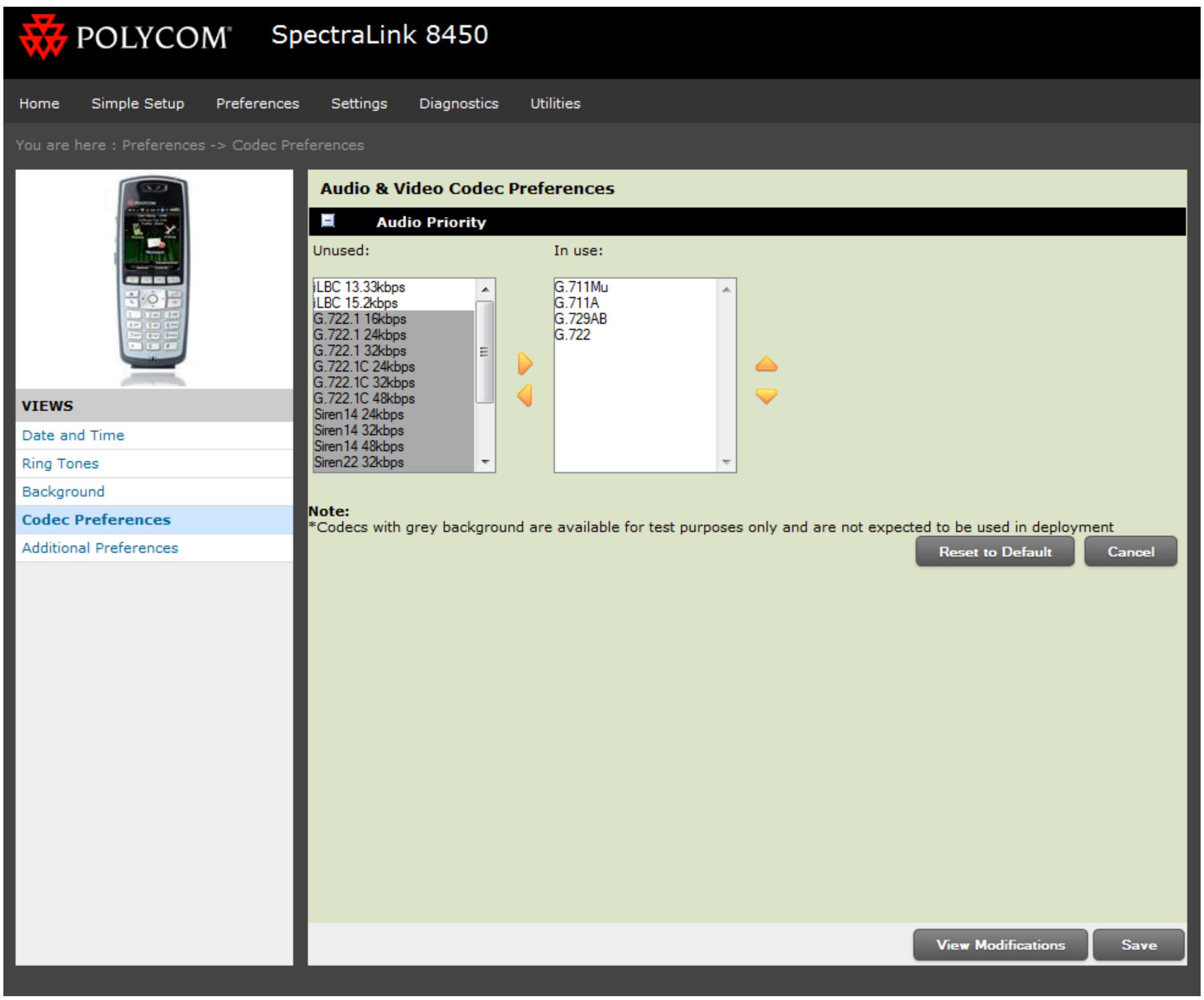


**Figure 24: Audio & Video Codec Preferences**

# 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@sipl ~]$ 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

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.