Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Polycom SpectraLink 8400 Series SIP Telephone version 4.2.0.0197 with Avaya Communication Server 1000 Release 7.5 – Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Communication Server 1000 SIP Line Release 7.5 and Polycom SpectraLink 8400 Series SIP telephone. During the compliance testing, the Polycom SpectraLink 8400 was able to register as a SIP client endpoint with the Communication Server 1000 SIP Line gateway. The Polycom SpectraLink 8400 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 telephony 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 detailed configurations of Avaya Communication Server 1000 SIP Line release 7.5 (hereafter referred to as CS 1000) and the Polycom SpectraLink 8400 SIP telephone Version 4.2.0.0197. The Polycom SpectraLink 8400 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 8400, where applicable, to verify the interoperability with CS 1000.

2. General Test Approach and Test Results

The general test approach was to have the Polycom SpectraLink 8400 telephone register to the CS1000 SIP line gateway successfully. From the CS1000 telephone clients/users, place a call to and from the Polycom SpectraLink 8400 telephone and to exercise other telephony features such as busy, hold, DTMF, MWI and codec negotiation.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute a full product performance or feature testing performed by third party vendors, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a third party solution.

2.1. Interoperability Compliance Testing

The focus of this testing was to verify that the Polycom SpectraLink 8400 SIP telephone was able to interoperate with the CS 1000 SIP line system. The following areas were tested:

- Registration of the Polycom SpectraLink 8400 SIP telephone to the CS1000 SIP Line Gateway.
- Telephony features: Basic calls, conference, transfer, DTMF (dual tone multi frequency) RFC2833, SIP Info 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 ISDN/PRI trunk.
- Codec negotiation – G.711, G.729, and G.722.

2.2. Test Results

The objectives outlined in **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 8400 compliance to required industry standards.
- Polycom SpectraLink 8400 handsets are treated by the CS1000 as 3rd party SIP endpoints and use CS1000 3rd party SIP licenses.
- The Polycom SpectraLink 8440 local forward busy feature which is set on the phone locally can be enabled but it will not be used for the busy call when the 8400 phone is in busy status. The server call forward busy feature of CS1000 SIP Line will take place

before the local forward busy can be executed by the phone. For the call forward busy, use the forward busy on the CS1000 switch instead.

- Avaya Aura® Messaging system only supports DTMF RFC2833. To work properly with Avaya Aura® Messaging voicemail, the SpectraLink 8400 should be set to RFC2833 and not SIP INFO or INBAND.

2.3. Support

For technical support for the Polycom SpectraLink 8400 Series SIP phone, please contact Polycom Inc technical support as shown below:

- Phone: 1.800.POLYCOM or +1.925.924.6000
- Website: www.polycom.com

3. Reference Configuration

Figure 1 illustrates the test configuration used during the compliance testing between the Avaya Communication Server 1000 and the Polycom SpectraLink 8400. The SpectraLink 8400 phone registers to the CS1000 SIP Line server by going through the Wi-Fi access point that connects to the lab network. Avaya Aura® Session Manager was used for routing SIP calls between the CS1000 A and CS1000 B for test cases off-net. The PRI T1 trunk was configured to connect to the simulated PSTN for test cases off-net via PRI T1 trunk.

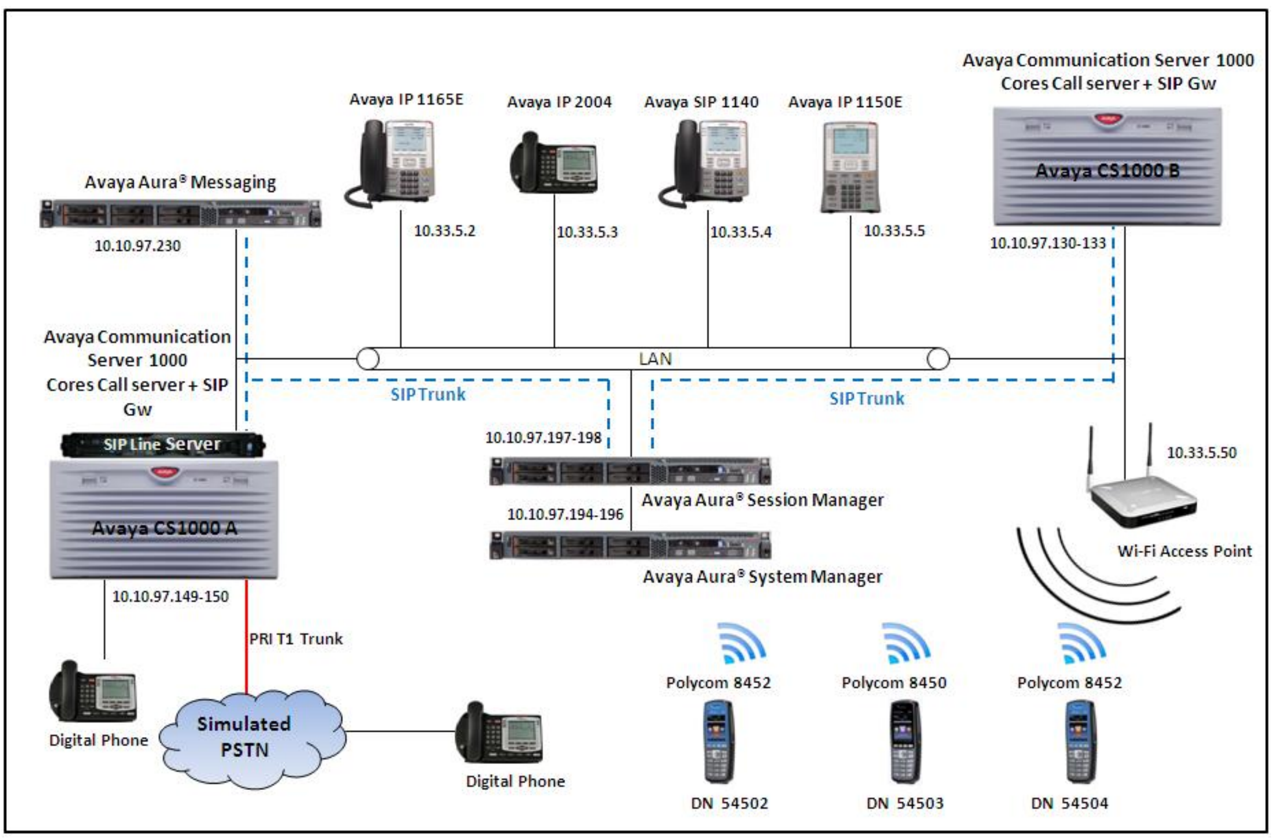


Figure 1: Test configuration diagram

4. Equipment and Software Validated

Equipment	Software
Avaya S8800 server running Avaya Aura® Session Manager Server	6.1 SP6 (Build No 6.1.6.0.616008)
Avaya S8800 server running Avaya Aura® System Manager Server	6.1 SP6 (Build No: 6.1.0.0.7345-6.1.5.606 Software Update Revision No: 6.1.10.1.1774)
Avaya S8800 server running Avaya Aura® Messaging Server	Avaya Aura® Messaging 6.1 SP2
Avaya Communication Server 1000E/CPPM	Avaya Communication Server Release 7.5 Q+ Deplist 1 (created: 2012-07-23) and Service Update 1 (Created: 2012-0708)
Avaya IP SIP Phone 1140E	4.3
Avaya IP Unistim Phone 1165E	0x25C8J
Avaya IP Unistim Phone 1150E	0x27C8J
Avaya IP Unistim Phone 2004	0604DCN
Polycom SpectraLink 8450	4.2.0.0197
Polycom SpectraLink 8452	4.2.0.0197

5. Configure Avaya Communication Server 1000

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 a summary of tasks required 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 local SIP 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.

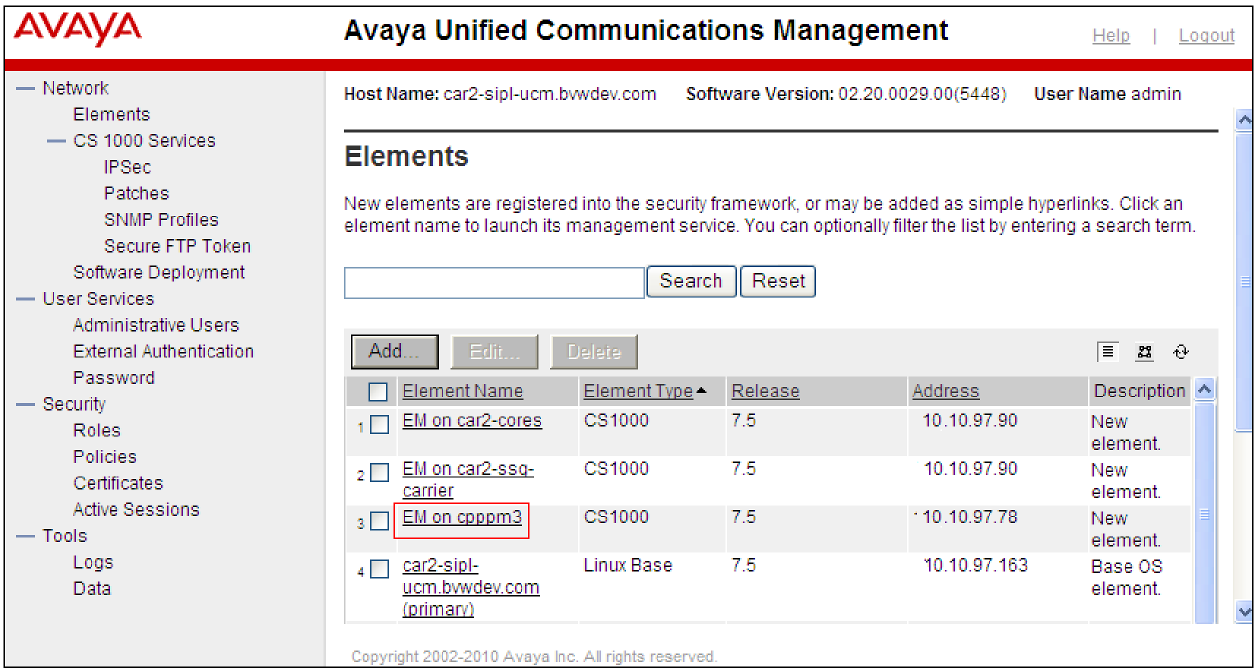
The following packages need to be enabled in the key code. If any of these features have not been enabled, please contact your Avaya account team or Avaya technical support at http://www.avaya.com.

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

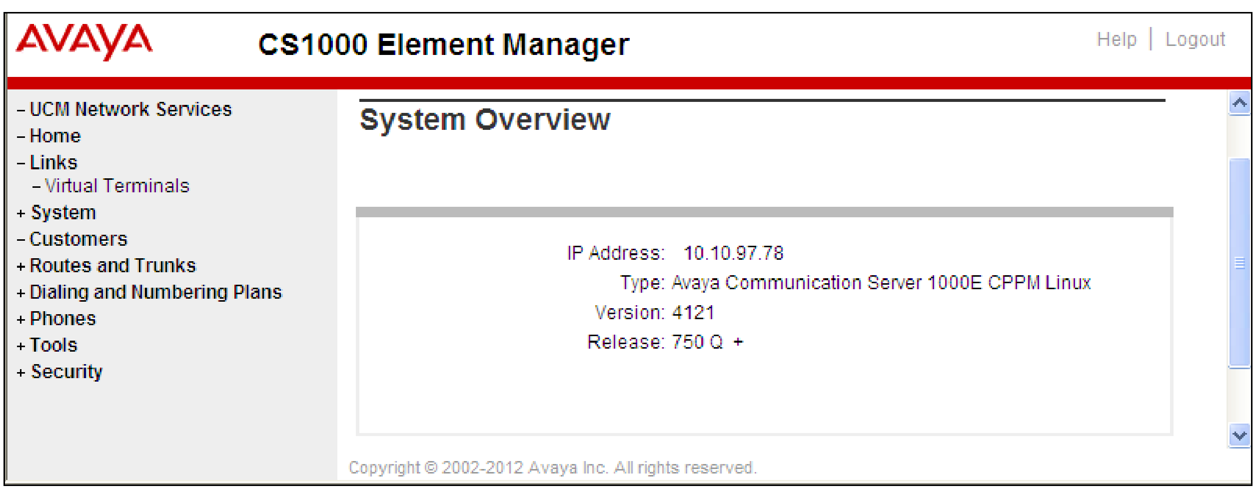
5.2. Log in to Unified Communications Management (UCM) and Element Manager (EM)

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 screen below. On the UCM home page, under the **Element Name** column, click on the Element Manager 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 below.

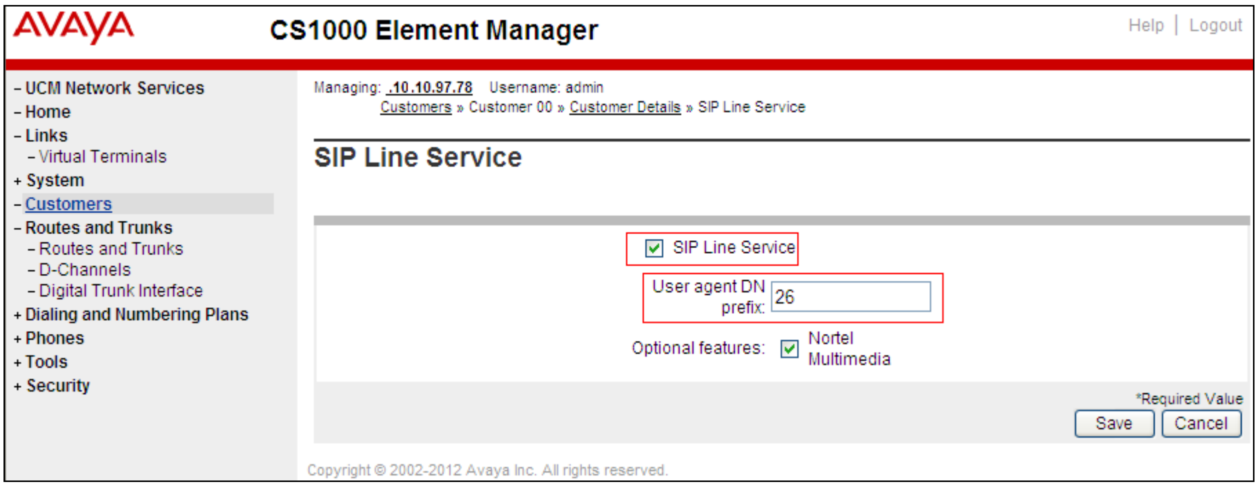


5.3. Enable SIP Line Service in the Customer Data Block

On the Element Manager page, navigate to **Customers** on the left menu. The list of Customer ID displays on the right, select the customer number (Customer 0) to be enabled with SIP Line Service (screen not shown). The screen below shows the SIP Line Service page.

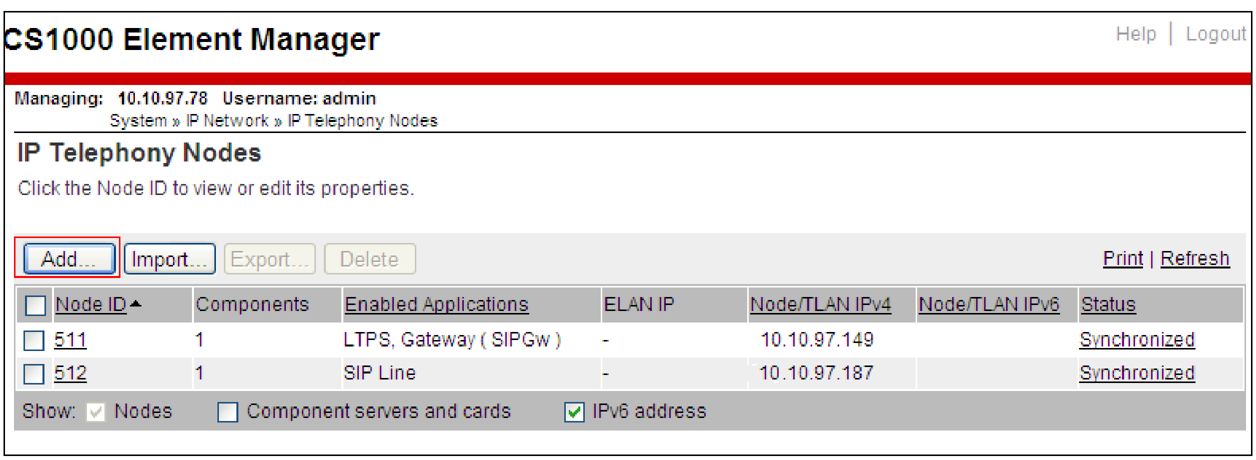
- Enable SIP Line Service by clicking on the **SIP Line Service** check box.
- Enter the prefix number in the **User agent DN prefix** text box, e.g., **26** as shown below.

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



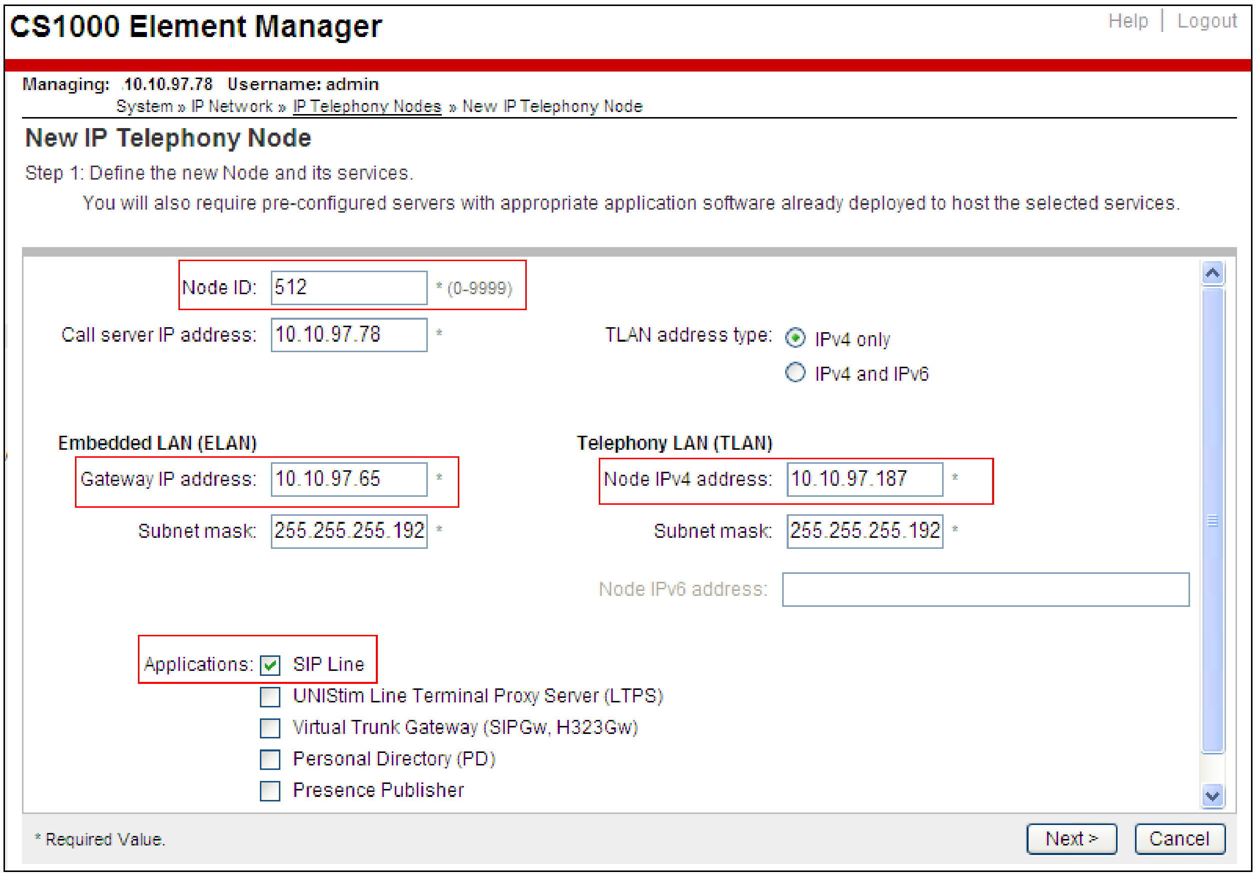
5.4. Add a new SIP Line Telephony Node

On the Element Manager page, navigate to menu **System → IP Network → Nodes: Servers, Media Cards**. The **IP Telephony Nodes** page is displayed as the screen below. Click **Add** button to add a new SIP Line Node to the IP Telephony Nodes.



The **new IP Telephony Node** page is displayed. Enter the information for each field shown below.

- **Node ID**: enter **512** which is the node ID of SIP Line server.
- **Telephony LAN (TLAN) Node IP Address**: enter **10.10.97.187** which is the Node IP address of SIP Line.
- **Embedded LAN (ELAN) Gateway IP Address**: enter **10.10.97.65** which is the gateway IP of Call server subnet.
- **Applications: SIP Line**: check the check box to enable SIP Line service for this Node.



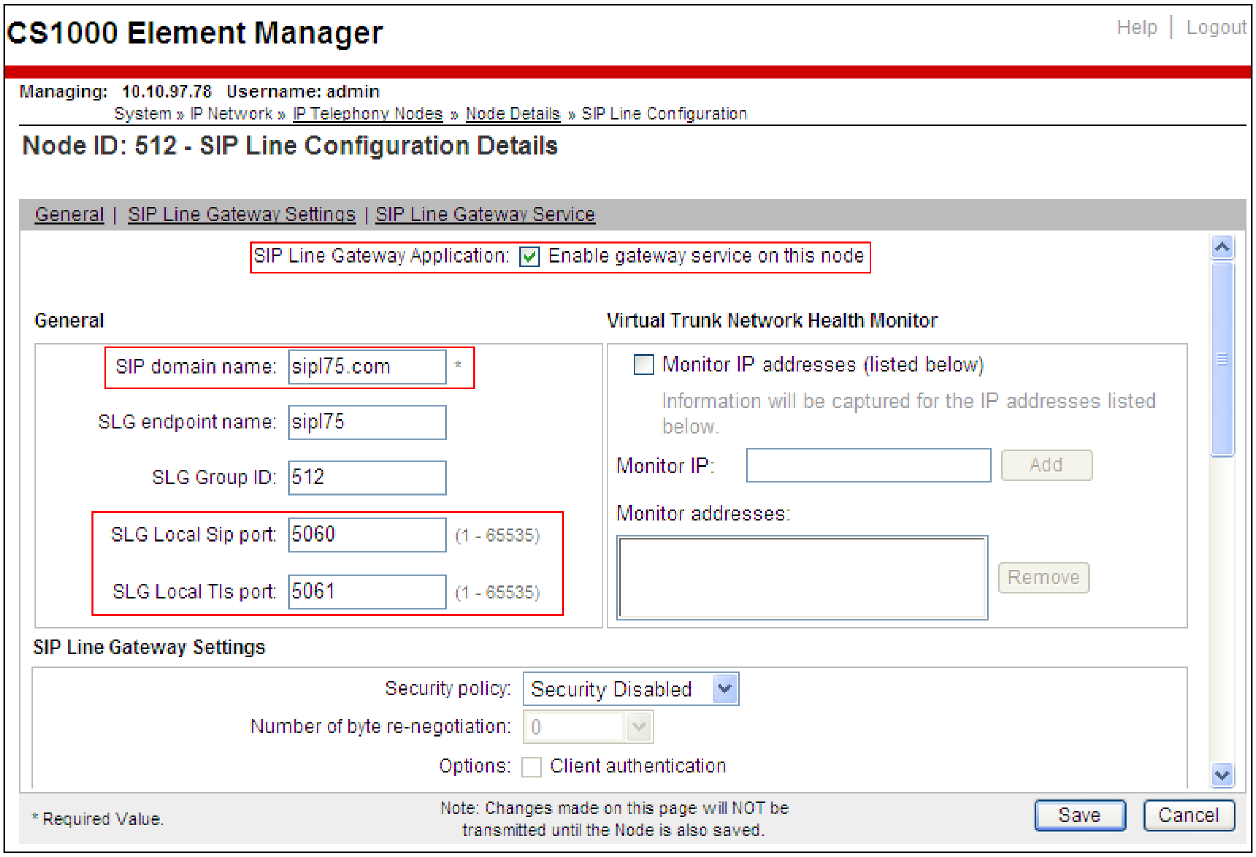
Click on the **Next** button to go to next page**.** The page, **New IP Telephony Node with Node ID** is displayed. On this page, in the **Select to Add** drop down menu list, select the desired server to add to the node. Click the **Add** button and select the check box next to the newly added server,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

Click on the **Next** button to go to next page**.** The **SIP Line Configuration Details** page is displayed as the screen below.

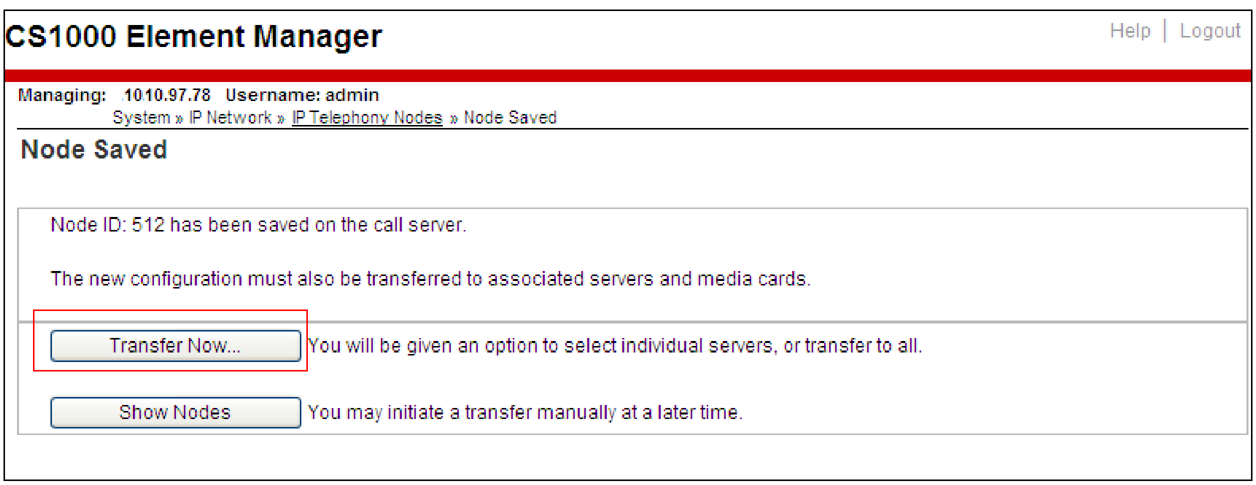
- **SIP Line Gateway Application**: Check on the check box **Enable gateway service on this node**.
- In the **General** section:
  - **SIP domain name**: enter the domain "**sipl75.com**".
  - **SLG Local Sip Port**: enter port "**5060**".

- **SLG Local Tls port**: enter the port "**5061**".
- Keep other sections as default.

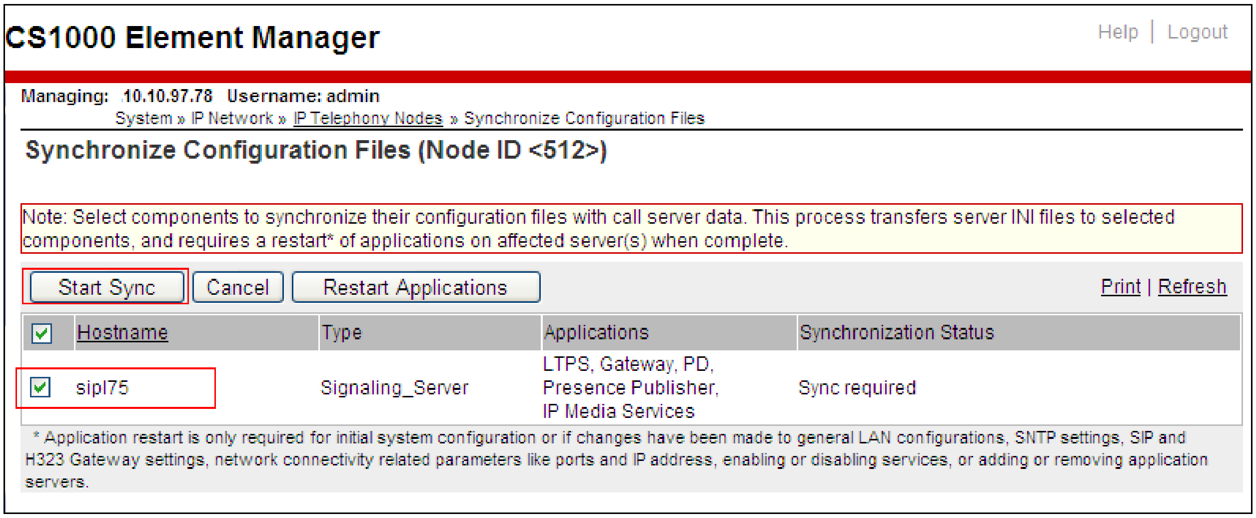
Click on the **Save** button to save the changes



Click **Next**. The **Confirm new Node details** page appears (screen not shown). Next click on the **Transfer Now** button in the **Node Saved** page as displayed in the screen below.



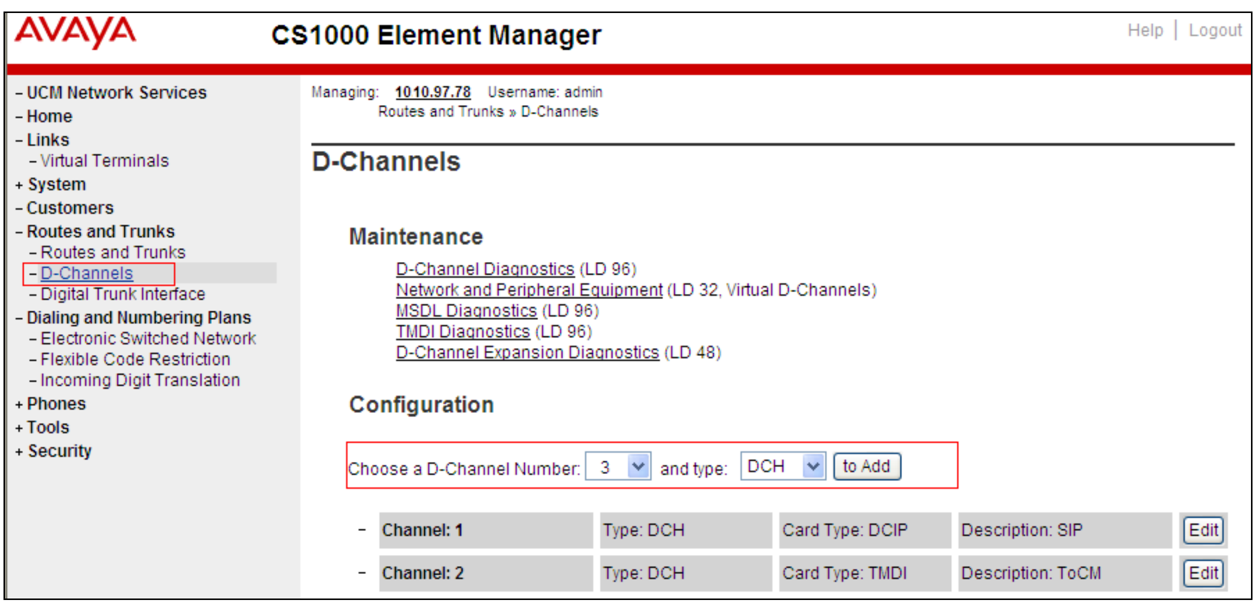
Click on the **Transfer Now** button, the **Synchronize Configuration Files (Node ID 512)** page is displayed. Select the SIP Line server that is associated with the changes and then click on the **Start Sync** button to transfer the configuration files to the selected servers as shown below.



<u>Note</u>: 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 the command: **appstart restart**.

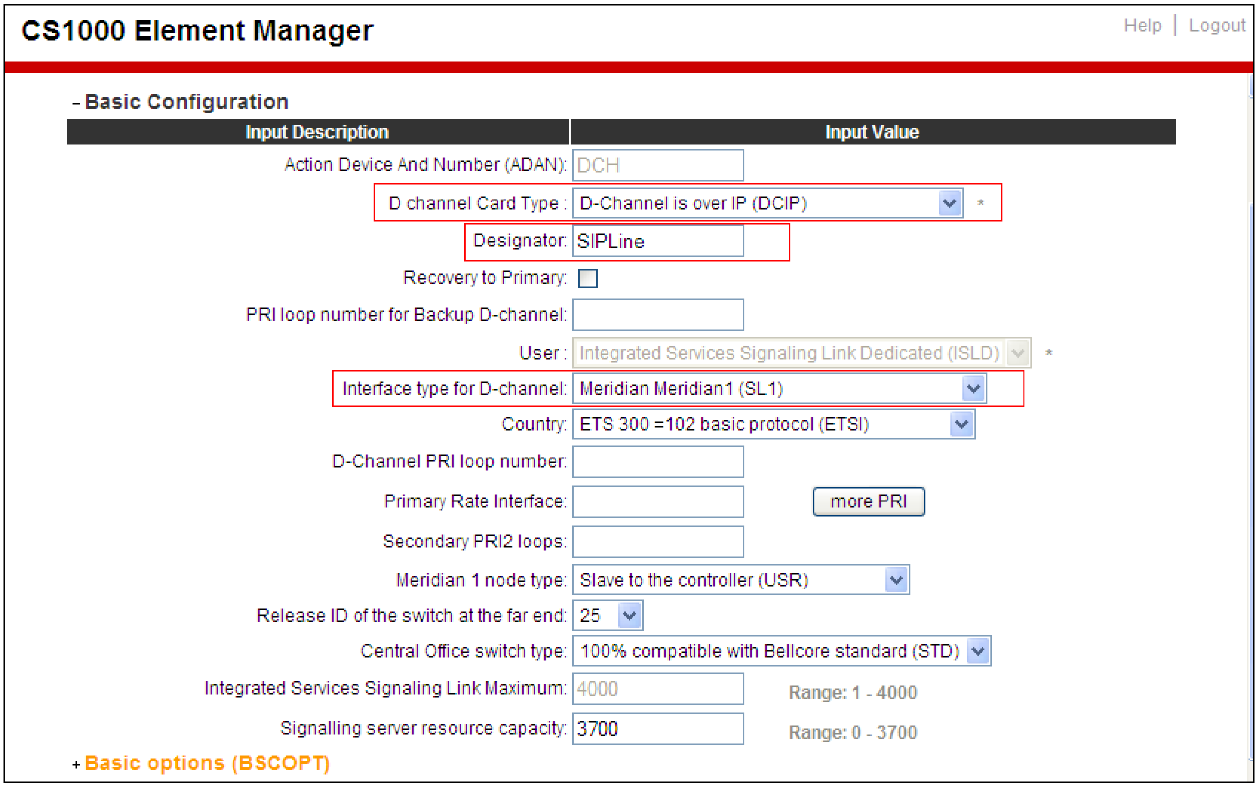
5.5. Create a D-Channel for SIP Line

On the Element Manager page, navigate to **Routes and Trunks → D-Channels**. The **D-Channels** page is displayed on the right, under the **Configuration** section as shown below, enter an available number in the **Choose a D-Channel Number** drop down menu, e.g., **3** and click on the "**to Add**" button.

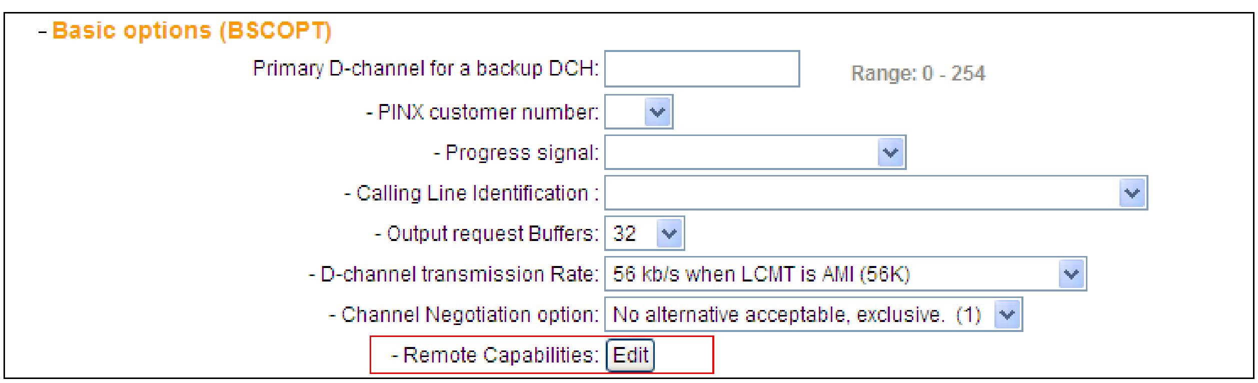


The **D-Channels 3 Property Configuration** page is displayed. In the **Basic Configuration** section:

- **D channel Card Type**: select **D-Channel is over IP (DCIP)**.
- **Designator**: enter a descriptive name, e.g., “**SIPLine**”.
- **Interface type for D-channel (IFC):** select **Meridian Meridian1 (SL1)**.
- Leave the other fields in the section at default values.

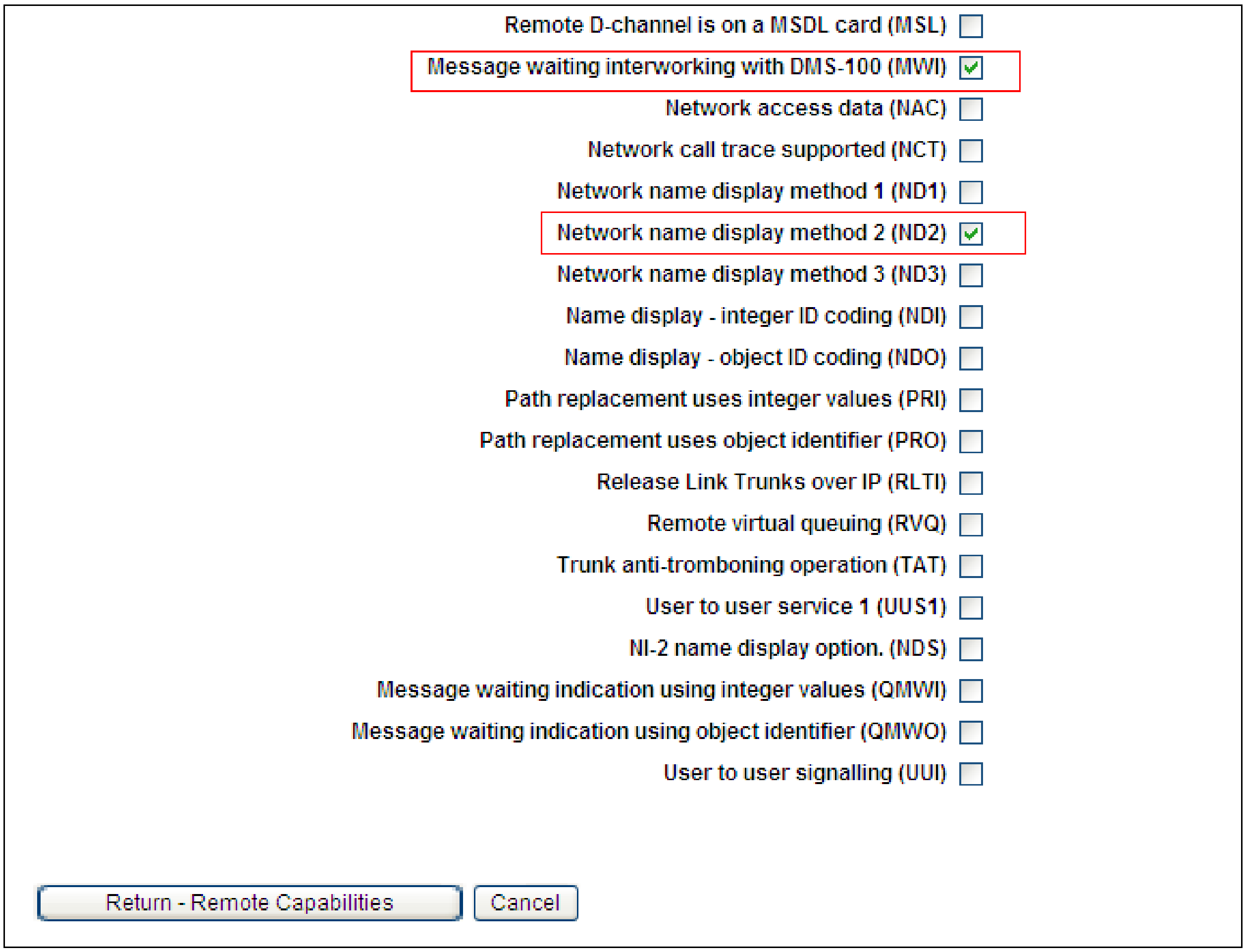


Click on the **Basic options (BSCOPT)** link to expand this section. The **Basic options (BSCOPT)** section is displayed as shown below. Click on **Edit** button to configure **Remote Capabilities (RCAP)**.



The **Remote Capabilities Configuration** page is displayed. Select the **Message waiting interworking with DMS-100 (MWI***)* and **Network name display method 2 (ND2)** check boxes. At the bottom of the **Remote Capabilities Configuration** page, click **Return - Remote Capabilities** button to return the **D-Channel 3 Property Configuration** page.

Note that the **Message Waiting Interworking with DMS-100 (MWI)** must be enabled to support voice mail notification on SIP Line endpoints and **Network Name Display Method 2 (ND2)** must be enabled to support name display between SIP Line endpoints.



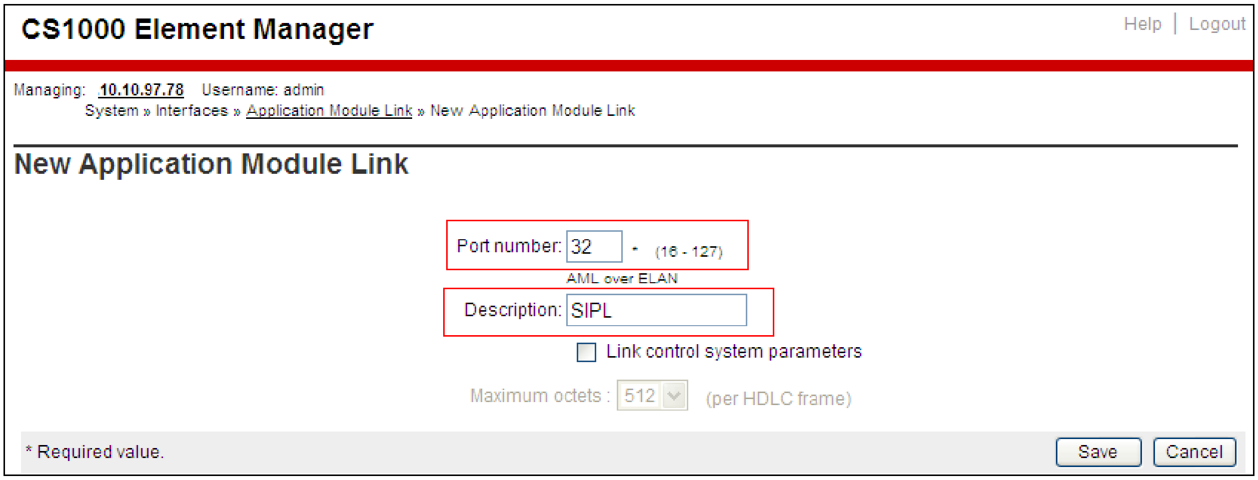
Leave the **Advance options (ADVOPT)** section at default.

Click on the **Submit** button at the bottom of the **D-Channel 3 Property Configuration** page to save changes and complete the creation of new D channel.

5.6. Create an Application Module Link (AML)

On the Element Manager page, navigate to **System → Interfaces → Application Module Link**. The **Application Module Link** page is displayed on the right (screen not shown), click on the **Add** button to add a new Application Module Link. The **New Application Module Link** page is displayed as below.

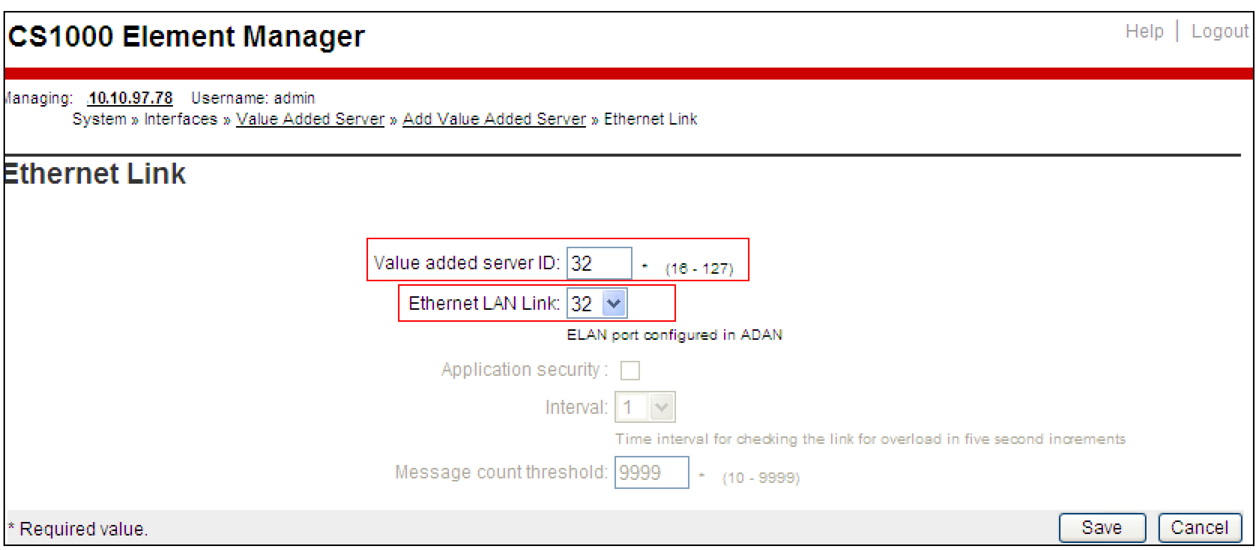
Enter an AML port number in the **Port number** text box, e.g., **32** and a descriptive name, e.g., "**SIPL**" in the **Description** ox. Note that The AML of SIP Line Service can use any port from 32 to 127. In this case, SIP Line Service is configured to use port **32**. Click on the **Save** button to complete the addition of the new AML link.



5.7. Create a Value Added Server (VAS)

On the Element Manager home page, navigate to **System → Interfaces → Value Added Server**. The **Value Added Server** page is displayed on the right, click on the **Add** button. The **Add Value Added Server** page is displayed; select the link **Ethernet LAN Link**.

The **Ethernet Link** page is displayed as shown below. Enter a number in the **Value added server ID** field, e.g., **32** and in the **Ethernet LAN Link** drop down list, select the AML number of ELAN that was created in **Section 5.6**. Leave the other fields as default values and click on the **Save** button to complete the addition of the new **VAS**.

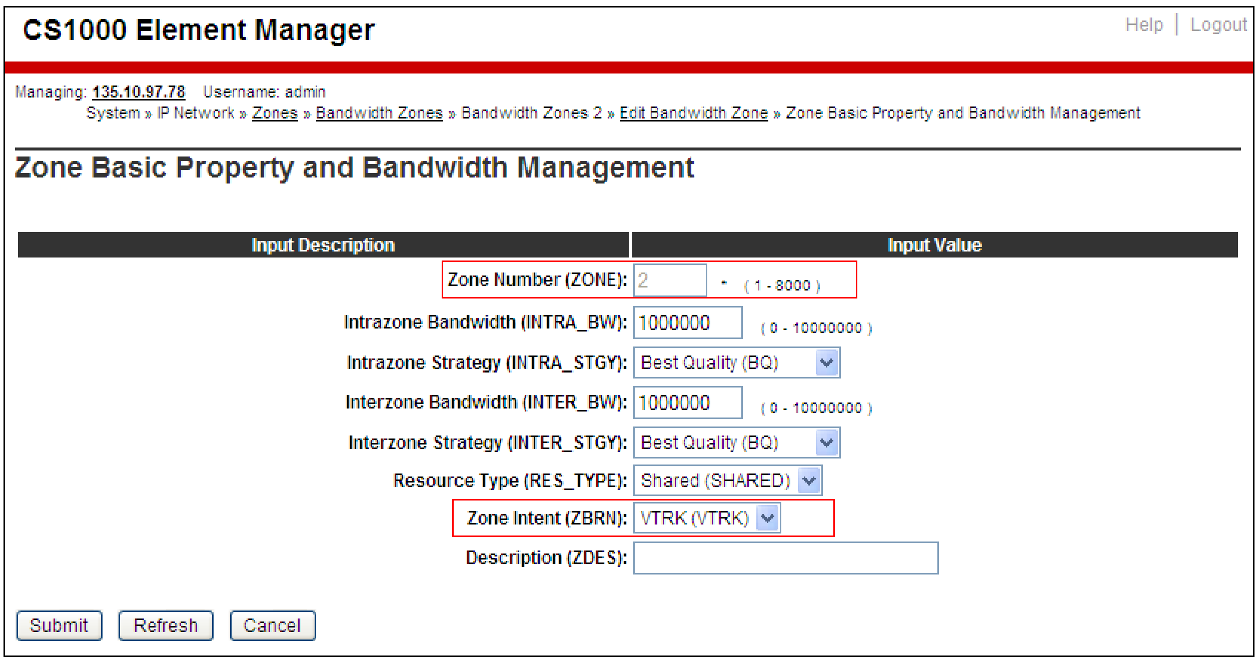


5.8. Create a Virtual Trunk Zone

On the Element Manager home page, navigate to menu **System → IP Network → Zones**. The **Zones** page is displayed 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 is displayed as shown the screen below.

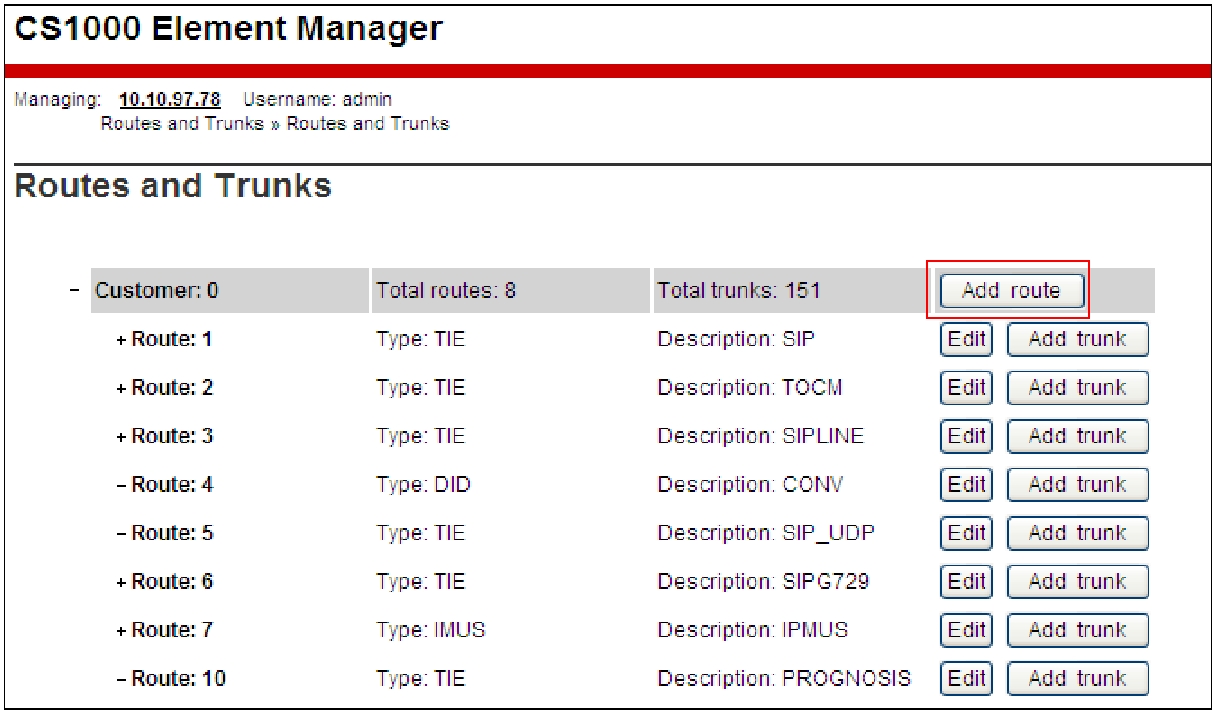
Enter a zone number in the **Zone Number (Zone)** field and in the **Zone Intent (ZBRN)** drop down menu select **VTRK (VTRK)**. Leave other fields as default values and click on the **Save** button to complete adding the Zone.

Repeat the procedure above to create another zone for the SIP Line phone; however remember to select **MO**, instead of **VTRK** in the field **Zone Intent**.

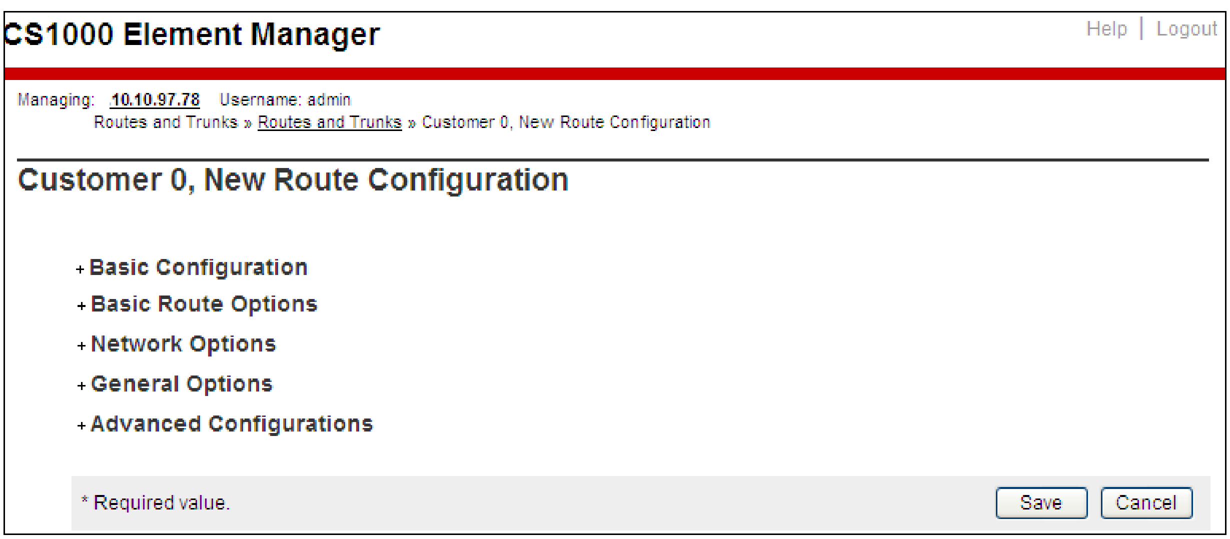


5.9. Create a SIP Line Route Data Block (RDB)

On the Element Manager home page, navigate to the menu **Routes and Trunks → Routes and Trunks**. The **Routes and Trunks** page is displayed on the right. In this page, click on the **Add route** button next to the customer number that the route will belong to.

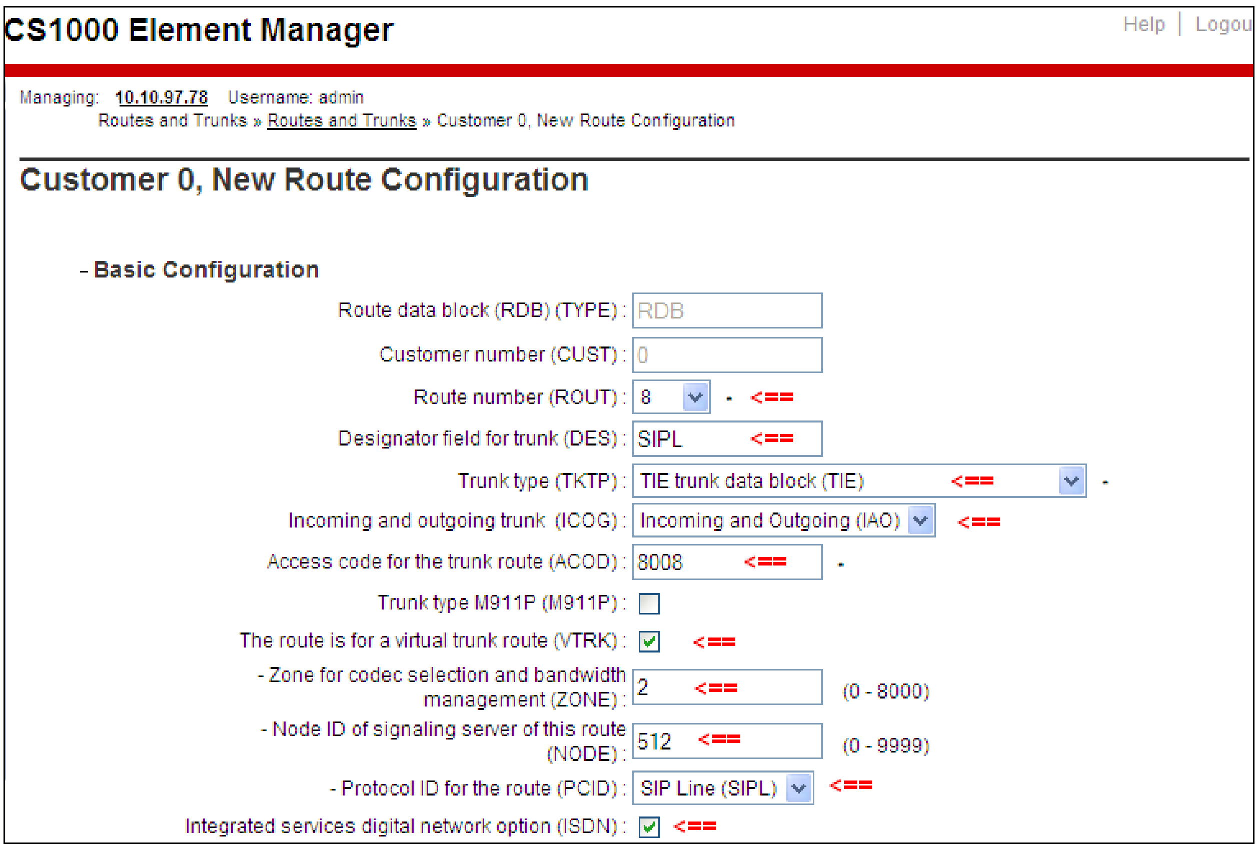


The **Customer ID, New Route Configuration** page is displayed. There are 5 sections in the new route configuration page.



Expand the **Basic Configuration** section, and enter values as shown in the two screens below.

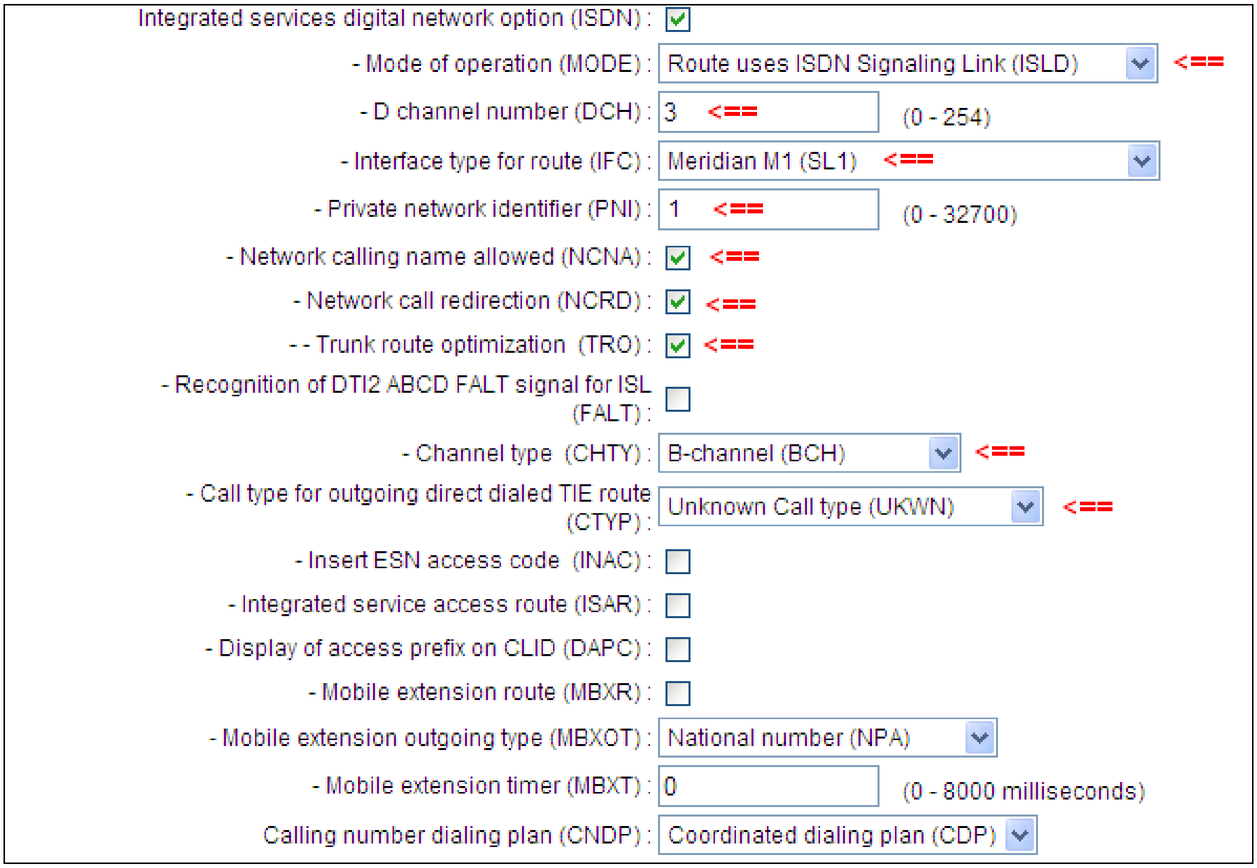
- **Route Number (ROUT)**: select an available number in the list, e.g., **8**.
- **Designator field for trunk (DES)**: enter a descriptive name, e.g. **SIPL**.
- **Trunk type (TKTP)**: select **TIE trunk data block (TIE)**.
- **Incoming and Outgoing trunk (ICOG)**: select **Incoming and Outgoing (IAO)**.
- **Access Code for Trunk group (ACOD)**: enter a number for ACOD, for example 8008. Note that this number has to follow the dialing plan rule.
- **The route is for a virtual trunk route (VTRK)**: check the checkbox.
- **Zone for codec selection and bandwidth management (ZONE)**: enter **2** which is the Virtual trunk zone number created in **Section 5.8**.
- **Node ID of signaling server of this route (NODE)**: enter **512** which is the node ID of the SIP Line configured in **Section 5.4**.
- **Protocol ID for the route (PCID)**: select **SIP Line (SIPL)** in the list.
- **Integrated services digital network option (ISDN)**: check the check box.



- **Mode of operation (MODE)**: select **Route uses ISDN Signaling Link (ISLD)**.
- **D channel number (DCH)**: enter **3** which is the D-channel number created in the **Section 5.5**.
- **Interface type for route (IFC)**: select **Meridian M1 (SL1)**.
- **Network calling name allowed (NCNA)**: check the check box.
- **Channel type (CHTY)**: B-channel (BCH).
- **Trunk route optimization (TRO)**: checked.

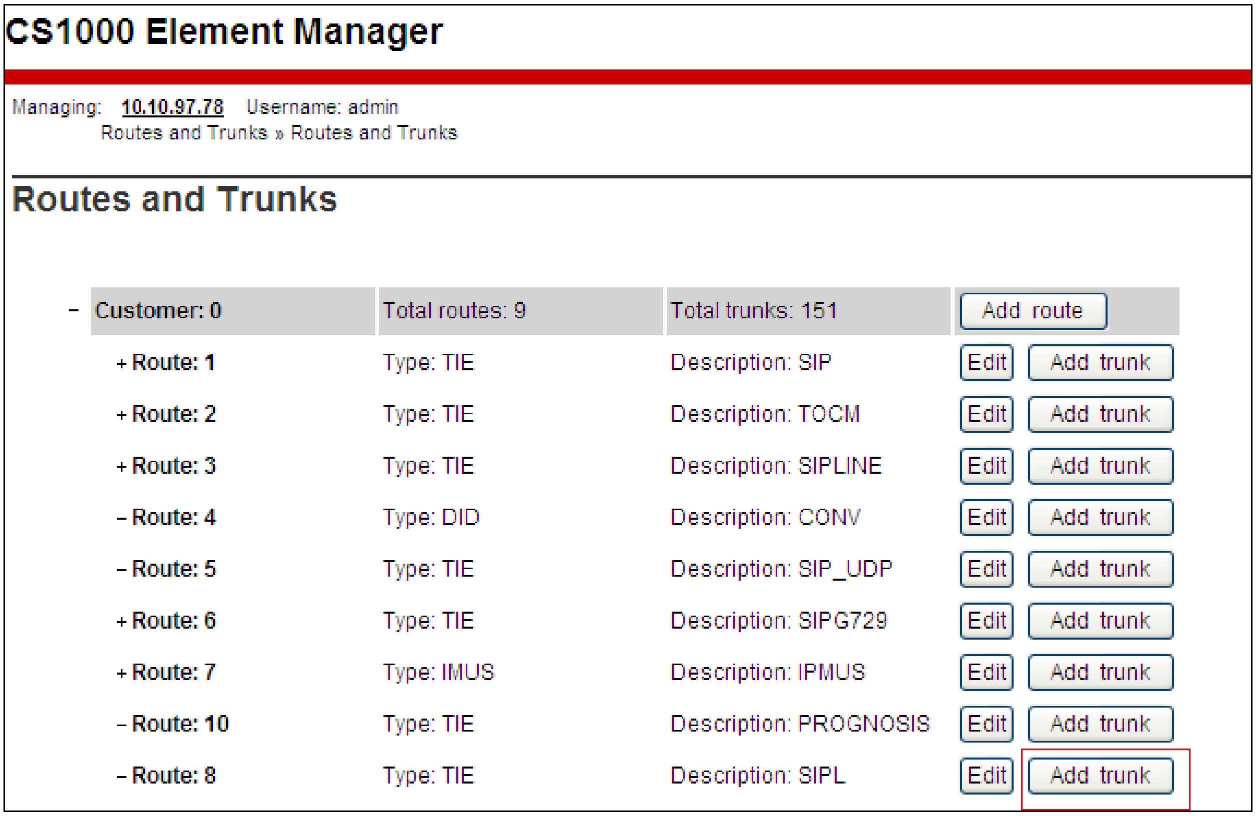
- **Call type for outgoing direct dialed TIE route (CTYP)**: select **Unknown Call type (UKWN)**.
- **Calling Number dialing plan (CNDP)**: select **Coordinated dialing plan (CDP)**.

Leave default values for The **Basic Route Options, Network Options, General Options, and Advanced Configurations** sections. Click the **Submit** button to complete the addition of new route and save configuration.



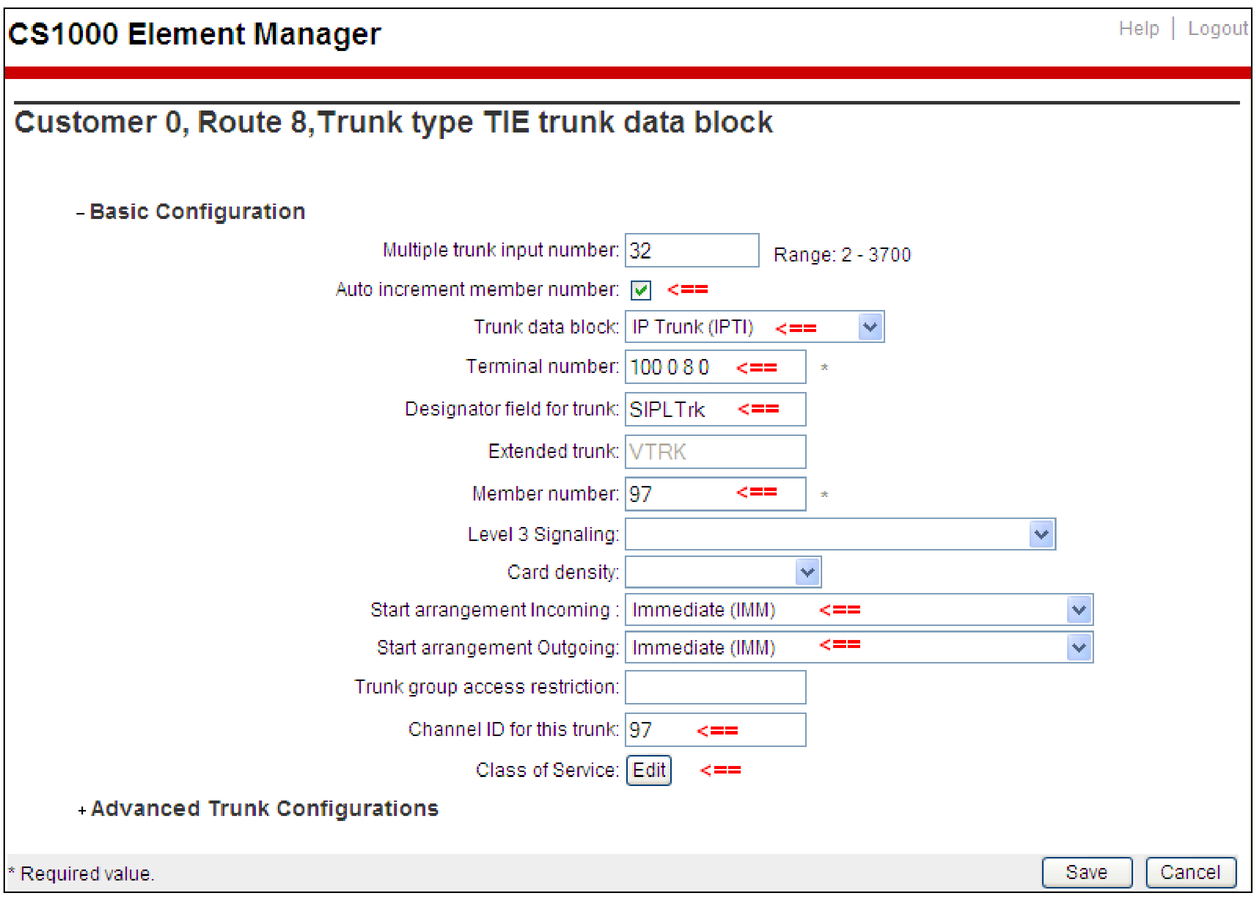
5.10. Create Virtual Trunks for SIP Line Route

On the Element Manager home page, navigate to **Routes and Trunks → Routes and Trunks**. The **Routes and Trunks** page is displayed on the right, select the **Add trunk** button beside the route **8** that was created in the **Section 5.9** above to create new trunks.



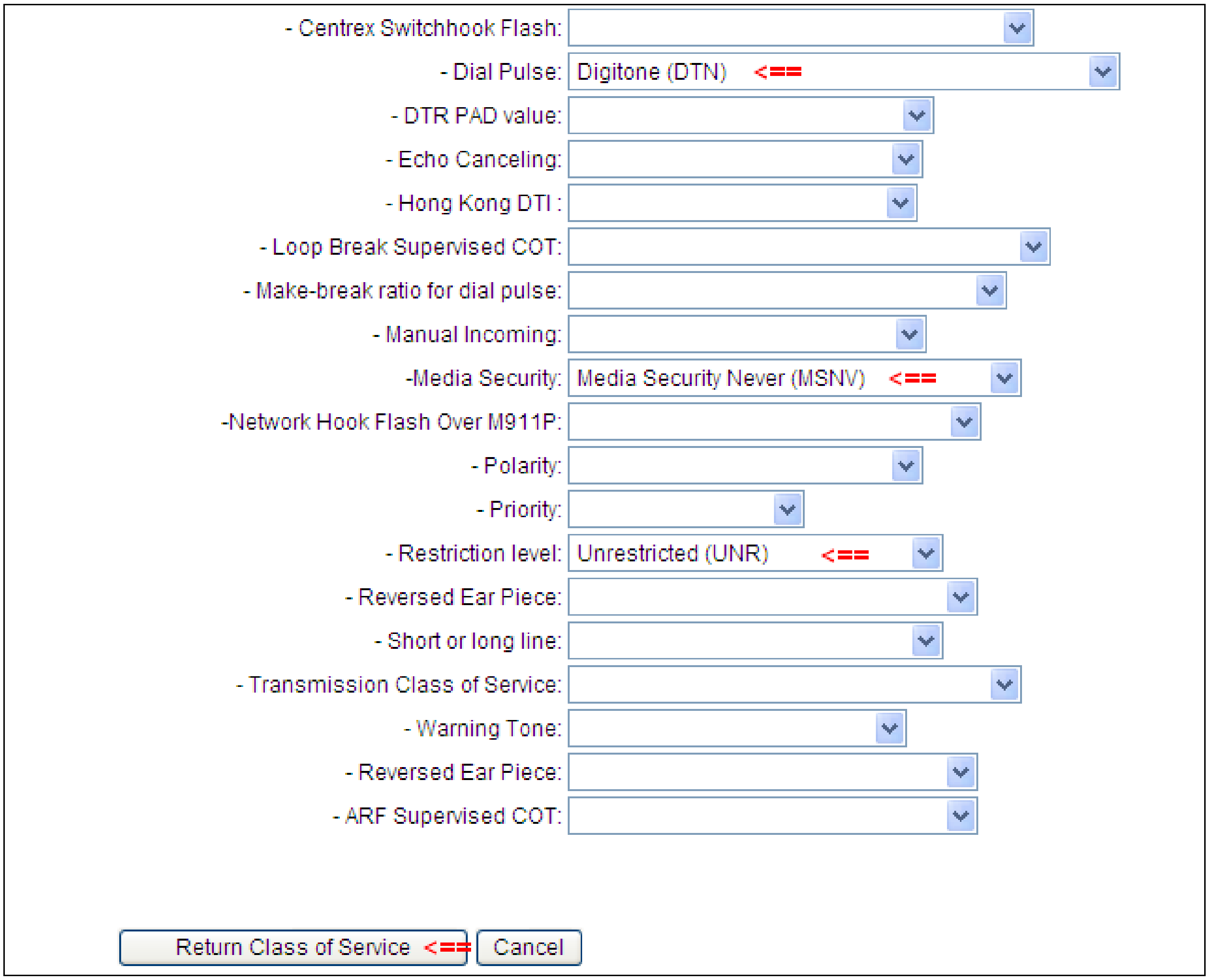
The **Customer 0, Route 8, Trunk type TIE trunk data block** page is displayed. Enter values for fields as shown below:

- **Multiple trunk input number (MTINPUT)**: enter **32** to create 32 trunks.
- **Auto increment member number**: checked. The trunks are created incrementally.
- **Trunk data block (TYPE)**: select **IP Trunk (IPTI**).
- **Terminal Number (TN)**: **100 0 8 0**. Enter the first Terminal Number in a range of Terminal number.
- **Designator field for trunk**: enter a descriptive name, e.g. "**SIPL Trk**".
- **Member number**: enter **97**. This is the ID of the trunk, just enter the first ID for the first trunk, next ID will be automatically created and incremented.
- **Start arrangement Incoming**: select **Immediate (IMM)**.
- **Start arrangement Outgoing**: select **Immediate (IMM)**.
- **Channel ID for this trunk**: **97**, this channel ID should be the same as the ID of Member Number and it has to be a unique number in the same type of trunk.



Click on the **Class of Service** button and assign following class of services as shown the screen below:

- **Dial Pulse** select **Digitone (DTN)**.
- **Media security**: select **Media Security Never (MSNV)**.
- **Restriction level**: select **Unrestricted (UNR)**.
- Leave other class of services at default values and click on the **Return Class of Service** button to return to the **Trunk type TIE trunk data block** page**.**



Leave the **Advance Trunk Configurations** section at default values and click on the **Save** button to complete the addition of new virtual trunks for SIP Line.

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.

```
LD 20
Req prt
TYPE: uext
TN   104 0 1 2
DES  POLY1
TN   104 0 01 02 → Terminal number of Universal Extension of SIP Line phone
TYPE UEXT
CDEN 8D
CTYP XDLC
CUST 0
UXTY SIPL → Type of UXTY is SIP Line
MCCL YES
SIPN 0
SIP3 1   → 3rd SIP endpoint is enabled
FMCL 0
TLSV 0
SIPU 54502 → SIP user which is used in the SIP endpoint for registration
NDID 512   → The node ID of SIP Line.
SUPR NO
UXID
NUID
NHTN
CFG_ZONE 00001 → Zone for SIP endpoint configured as MO
CUR_ZONE 00001
MRT
ERL
ECL  0
VSIT NO
FDN
TGAR 1
LDN  NO
NCOS 0
SGRP 0
RNPG 0
SCI  0
SSU
XLST
SCPW 1234 → The password to be used for registration along with SIP user
SFLT NO
CAC_MFC 0
CLS  CTD FBA WTA LPR MTD FNA HTD TDD HFD CRPD → Depend on feature cls enabled
     MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
     POD SLKD CCSD SWD LND CNDA
     CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
     ICDD CDMD LLCN MCTD CLBD AUTU
     GPUD DPUD DNDA CFXD ARHD CLTD ASCD
     CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
     UDI RCC HBTD AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
     DRDD EXR0
     USMD USRD ULAD CCBD RTDD RBDD RBHD PGND FLXD FTTC DNDY DNO3 MCBN
     FDSD NOVD VOLA VOUD CDMR PRED RECD MCDD T87D SBMD ELMD
     MSNV FRA  PKCH MWTD DVLD CROD ELCD
CPND_LANG ENG
RCO  0
```

```
HUNT
PLEV 02
PUID
UPWD
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY  00 SCR 54502 0     MARP → The main directory number of SIP endpoint
        CPND
          CPND_LANG ROMAN
            NAME Poly1 54502
            XPLN 13
            DISPLAY_FMT FIRST,LAST
     01 HOT U 2654502 MARP 0 → The Hot U with the prefix 26 configured in
adding SIP Line server.
     02 MSB → MSB key is used for Make Set busy feature on SIP endpoint
     03 CWT → CWT key is used for Call Waiting feature on SIP endpoint
     04
     05
     06
     07
     08
     09
     10
     11
     12
     13
     14
     15
     16
     17 TRN
     18 AO6
     19 CFW 16
     20 RGA
     21 PRK
     22 RNP
     23
     24 PRS
     25 CHG
     26 CPN
     27
```

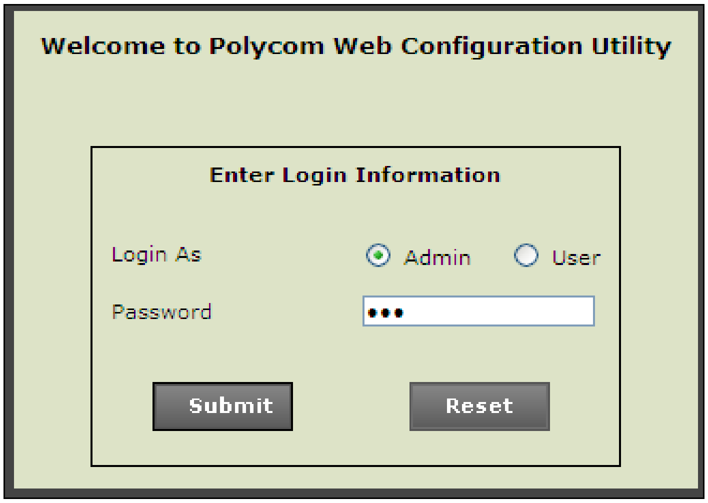
6. Configure Polycom SpectraLink 8400

This section describes how to access the Polycom SpectraLink 8400 SIP endpoint web interface and configure the Polycom 8400 for testing. For more information on how to configure the Polycom SpectraLink 8400 phone connected to the Access Point Wi-Fi router, please refer to the references in **Section 9**.

6.1. Login Polycom SpectraLink 8440

This section shows how to log in to the home page of Polycom SpectraLink 8400 to manage and configure the 8400 phone.

Open the web browser, and in the address box enter the Polycom SpectraLink 8440 IP address: http://ipaddress and the Polycom SpectraLink 8400 login page will appear as shown the screen below. Select the username, **Admin,** and enter its default password, **456** in the Password box. Click the **Submit** button to enter to the Polycom 8400 management page.



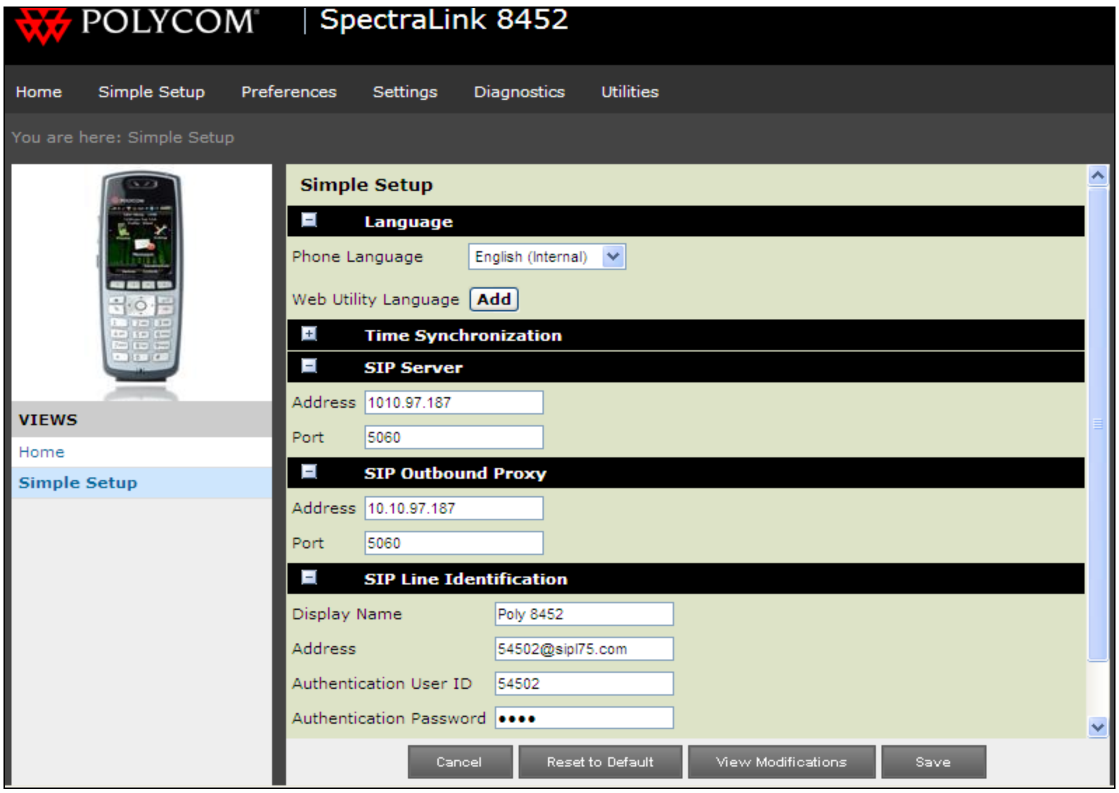
The screen below shows the home page of Polycom 8452 phone which is one of the models in the 8400 series phone.

6.2. Register Polycom SpectraLink 8400 to CS1000 SIP Line

This section shows how to configure the Polycom 8400 telephone to register with the CS1000 SIP Line gateway. On the homepage of the configuration screen, navigate to menu **Simple Setup**, the **Simple Setup** page is displayed as shown in the screen below. Enter the values as shown below:

- **SIP Server:**
  - **Address**: enter **10.10.97.187** → which is node IP address of CS 1000 SIP Line server.
  - **Port:** enter **5060** → which is local sip port of CS 1000 SIP Line.
- **SIP Outbound proxy:**
  - **Address**: enter **10.10.97.187** →Use the same Node IP address of SIP Line server.
  - **Port**: 5060
- **SIP Line Identification:**
  - **Display Name**: enter a descriptive name, e.g., **Poly 8452**.
  - **Address**: enter 54502@sipl75.com → The address should be like DN@SIPdomain.
  - **Authentication User ID**: enter **54502**→ This user ID that is configured in the field **SIPU** of Terminal Number of SIP Line phone in the **Section 5.11**
  - **Authentication Password**: enter **1234** → This password that is configured in the field **SCPW** of UEXT Terminal Number for SIP Line phone in the **Section 5.11**

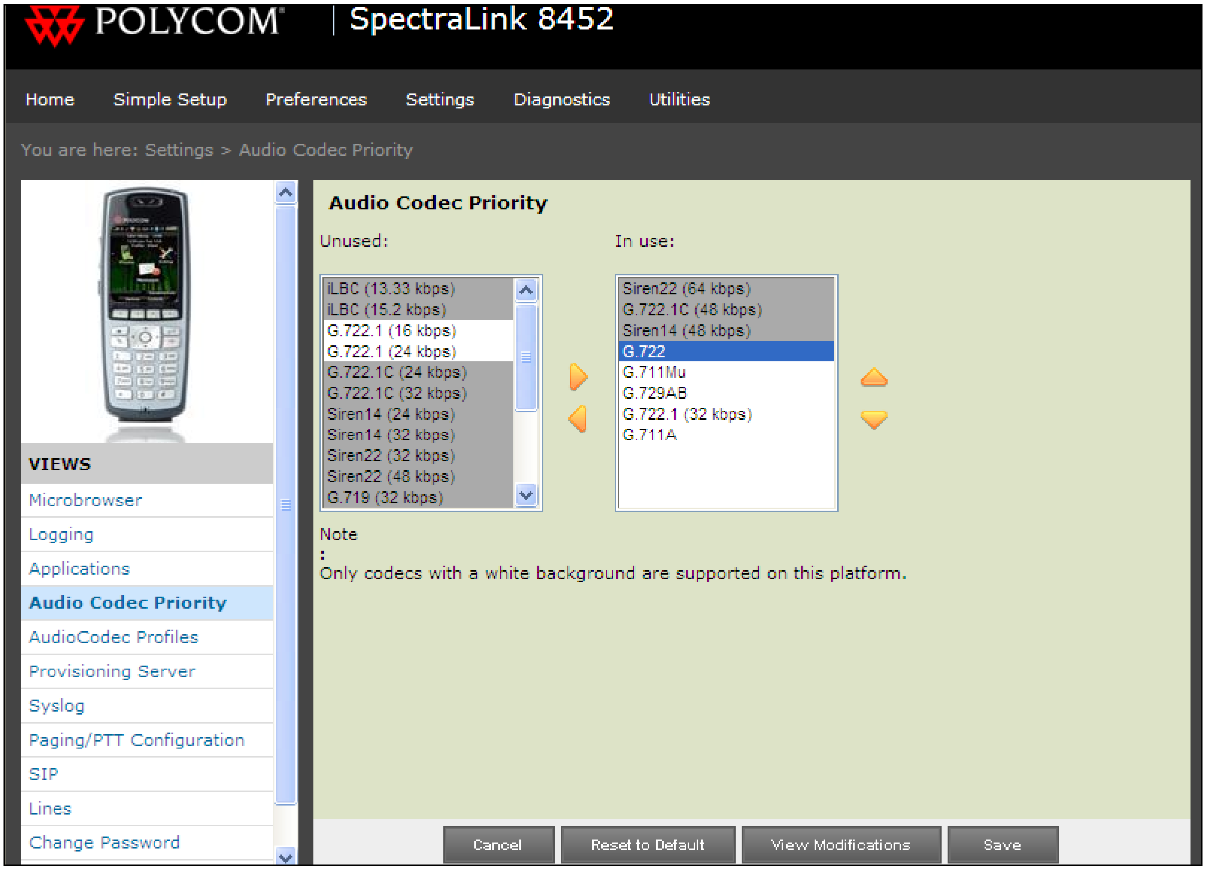
Click on the **Save** button to save changes. Note that the phone needs to be rebooted for the changes to take effect.



6.3. Configure Codec settings

This section shows how to set the codec on the Polycom SpectraLink 8400 phone. The compliance testing has been done on three codecs: G.722, G711 Mu law and G729.

On the homepage of Polycom SpectraLink 8400, navigate to menu **Settings → Audio Codec Priority**, the **Audio Codec Priority** page is displayed as shown below. The list of audio codecs being used appear under the **In use** column. To use the codec G722 as the first choice, move it up to the top of the **In Use** list, repeat the same procedure for other codecs. 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 Polycom 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 the Avaya account.
- Issue command “slgSetShowByUID [userID]” where userID is SIP Line user’s ID being checked.

```
[admin@sipl75 ~]$ slgSetShowByUID 54503

=== VTRK ===
UserID           AuthId      TN               Clients  Calls  SetHandle  Pos ID    SIPL Type
---------------  ----------  ---------------  -------  -----  ---------  -------   ---------
          54503       54503     104-00-01-03        1      0  0xa2b56f0         SIP Lines
        StatusFlags = Registered Controlled KeyMapDwld SSD
        FeatureMask =
        CallProcStatus = 0

        Current Client = 0, Total Clients = 1

         == Client 0 ==
         IPv4:Port:Trans = 10.33.5.52:5060:udp
         Type           = SIP3
         UserAgent      = SpectraLink-SL_8452-UA/4.2.0.0197
         x-nt-guid      = 78cce9a0bdf57a5aa3453a93a2a82992
         RegDescrip     =
         RegStatus      = 1
         PbxReason      = OK
         SipCode        = 200
         hTransc        = (nil)
         Expire         = 3600
         Nonce          = d1ae08b2812486874f88726da96ad309
         NonceCount     = 7
         hTimer         = 0xa2251c8
         TimeRemain     = 347
         Stale          = 0
         Outbound       = 0
         ClientGUID     = 0
         MSec CLS       = MSNV (MSEC-Never)
         Contact        = sip:54503@10.33.5.52
         KeyNum         = 255
         AutoAnswer     = NO

        Key  Func  Lamp  Label
        0    3     0     54503
        1    126   0     2654503
        3    3     0     54505
        4    2     0     54506
        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 the command “stat [TN]” where TN is the SIP Line user’s TN being checked

```
>ld 32
NPR000
.stat 104 0 1 3
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 **Section 2.1**, with some exceptions outlined in **Section 2.2**. The Polycom SpectraLink 8400 Series SIP phone version 4.2.0.0197 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.