Avaya Solution & Interoperability Test Lab

Application Notes for Configuring the Polycom SoundStation Duo running UC Software Release 4.0.2 with Avaya Communication Server 1000 Release 7.5 - Issue 1.0

Abstract

These Application Notes describe a solution for supporting interoperability between the Polycom SoundStation Duo conference telephone running UC software release 4.0.2 with Avaya Communication Server 1000 release 7.5. Emphasis of the testing was to verify voice calls of SoundStation Duo as a SIP endpoint registered to the Avaya Communication Server 1000 SIP line system.

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 1000 (hereafter referred to as CS1000) and the Polycom SoundStation Duo (hereafter referred to as Duo) used during the compliance testing. The Polycom Duo was tested with non-SIP and SIP telephones using CS1000 release 7.5. All the applicable telephony feature test cases of release 7.5 were executed on the Polycom Duo, where applicable, to ensure the interoperability with CS1000.

2. General Test Approach and Test Results

The general test approach was to have the Polycom Duo to register to the SIP line gateway of the CS1000. Calls were then placed from CS1000 telephone clients/users to and from the Polycom Duo. Other telephony features such as busy, hold, DTMF, transfer, conference and codec negotiation were also verified.

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 Duo SIP conference telephone was able to interoperate with the Avaya CS1000 SIP line system. The following areas were tested:

- Registration of the Duo to the CS1000 SIP line gateway.
- Call establishment of Duo with CS1000 telephones.
- Telephony features: Basic calls, conference, blind and consultative transfer, DTMF (dual tone multi frequency) RFC2833, leaving and retrieving voicemail message, busy, hold, call forward busy, call forward unconditional, call forward no answer, MWI (Message Waiting Indicator) and Do not Disturb (DND).
- Codec negotiation – G.711, G.729 and G722.
- Duo calls PSTN telephone via SIP trunk.

2.2. Test Results

The objectives outlined in **Section 2.1** were verified. The Duo was registered to CS1000 SIP line gateway successfully. Calls have been made between CS1000 telephones and Duo with clear voice path. Observation is that when performing 3 ways conference where Duo is a host of the conference and CS1000 telephones are the participant, as the Duo hangs up the 2 CS1000 participant telephones are not able to establish the voice path. This issue is on the CS1000. A work item wi01046703 is raised to track to resolution.

2.3. Support

Technical support for the Polycom SoundStation Duo conference phone can be obtained through Polycom global technical support:

- Phone: 1-888-248-4143 or 1-408-474-2067
- Web: http://support.polycom.com

3. Reference Configuration

Figure 1 illustrates the reference configuration used during compliance testing.

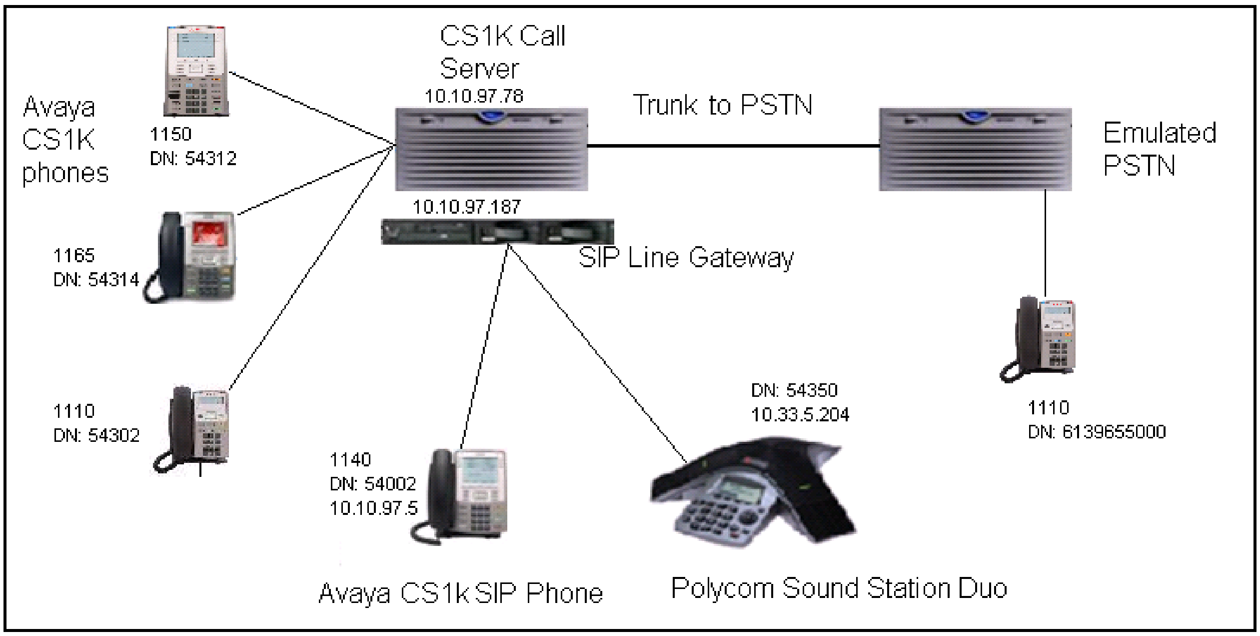


Figure 1: Network Configuration Diagram

4. Equipment and Software Validated

The following equipment and software/firmware were used for the reference configuration:

Equipment/Software	**Release/Version**
Avaya Communication Server 1000E Call Server Signaling Server SIP Line Gateway	7.50Q GA plus DEPLIST dated:2012-07-16 7.50.17 GA plus Service_Pack_Linux_7.50_17_20120713.ntl 7.50.17 GA plus Service_Pack_Linux_7.50_17_20120713.ntl
Avaya CS1000 IP Phones: 1110 1150 1165	 0623C8J 0626C8J 0626C8J
Avaya CS1000 SIP Phone: 1140	SIP1140
Polycom SoundStation Duo	4.0.2

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.

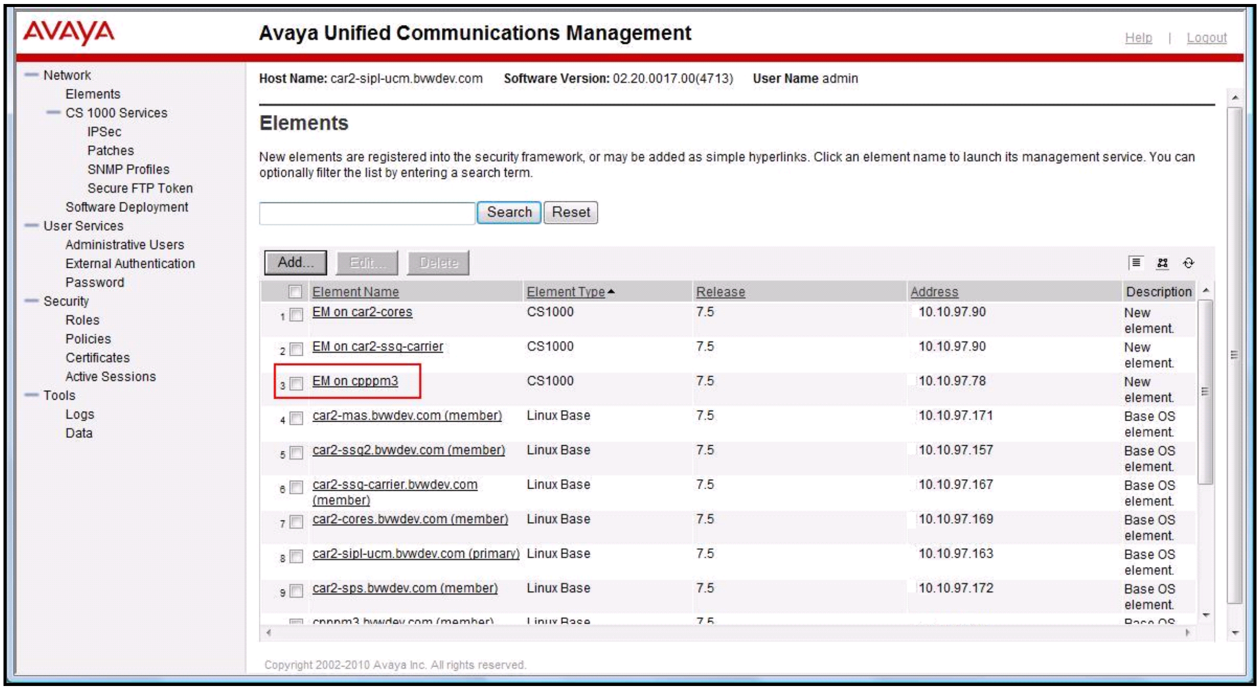
The following packages need to be enabled in the key code. If any of these features have not been enabled, please contact your Avaya account team or Avaya technical support at http://www.avaya.com.

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 a web browser to launch the Avaya CS1000 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. Login with the username/password which was defined during the primary security server configuration (not shown). For more information, see **[1].**

On the **Elements** page of Unified Communications Management, under the **Element Name** column, click the server name to navigate to Element Manager for that server.

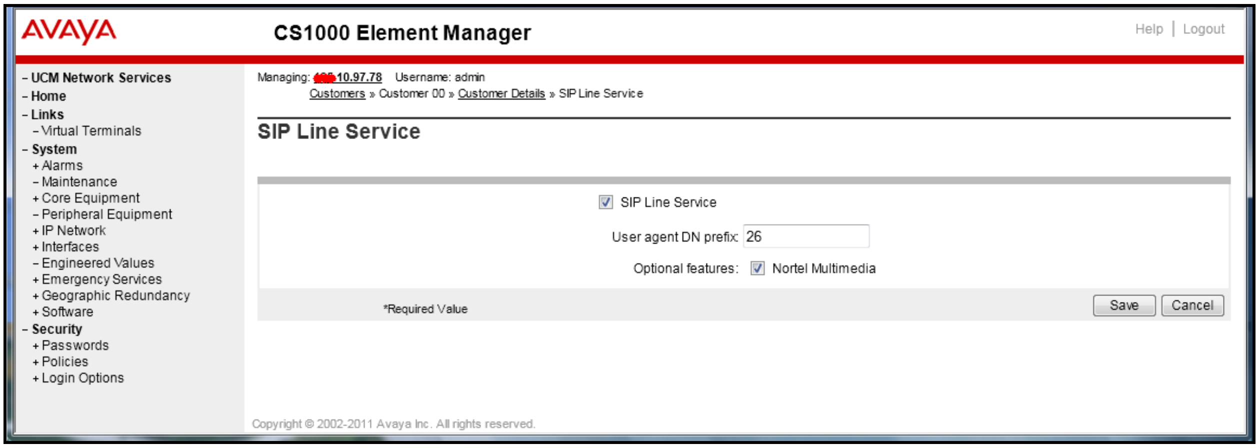


The Avaya CS1000 Element Manager (EM) page appears as shown.

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).

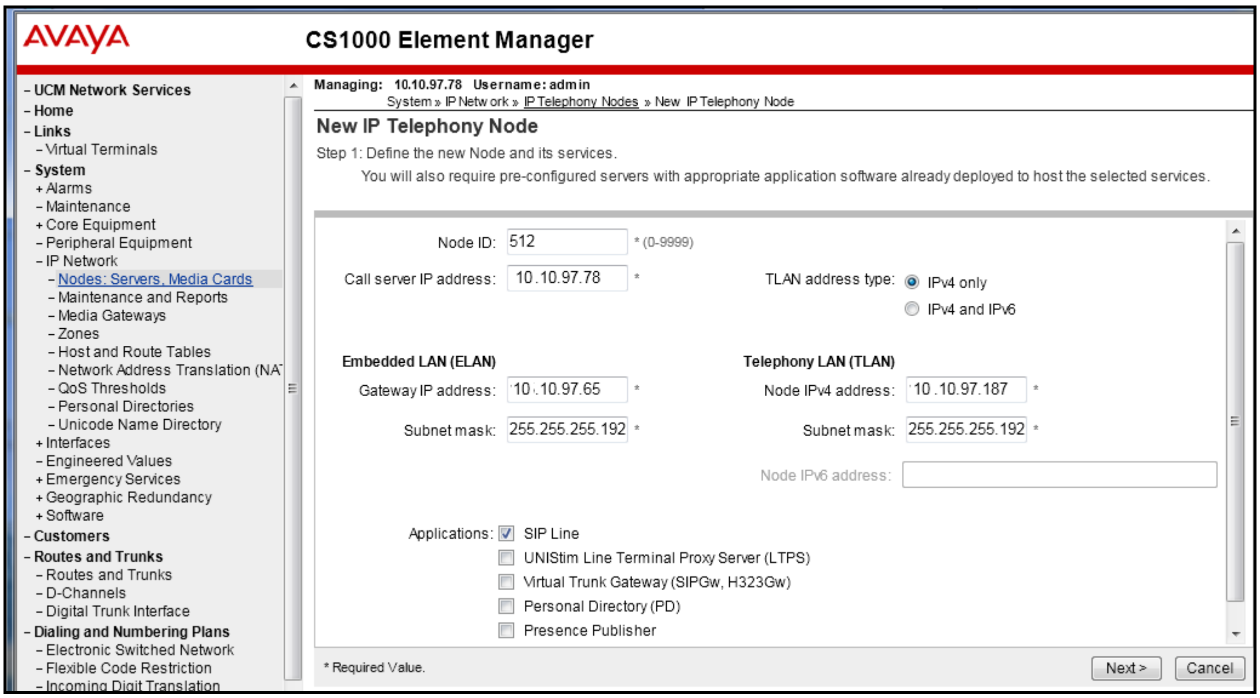
- Enable SIP Line Service by clicking on the **SIP Line Service** check box.
- Enter the prefix number in the **User agent DN prefix** text box.



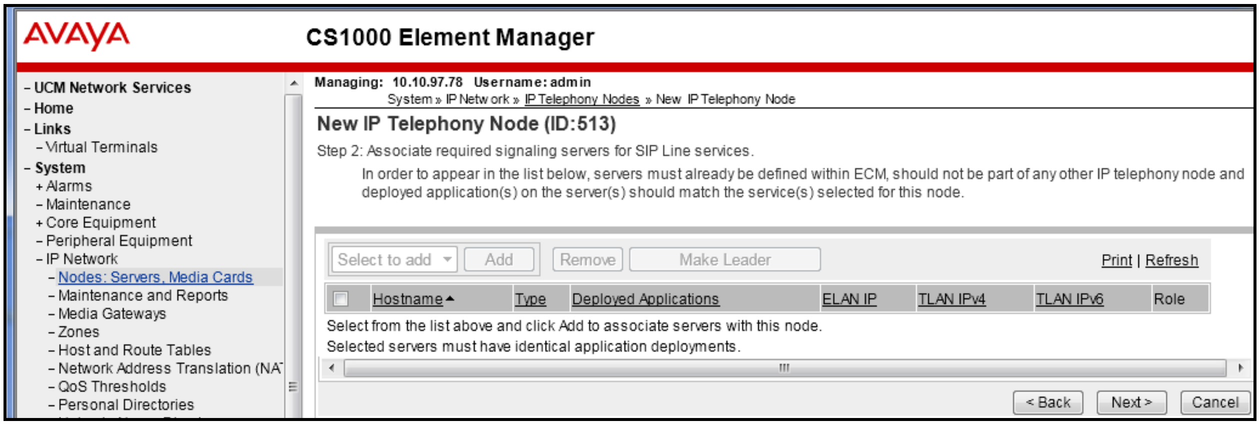
5.4. Add a new SIP Line Telephony Node

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 below. Enter the information as shown below:

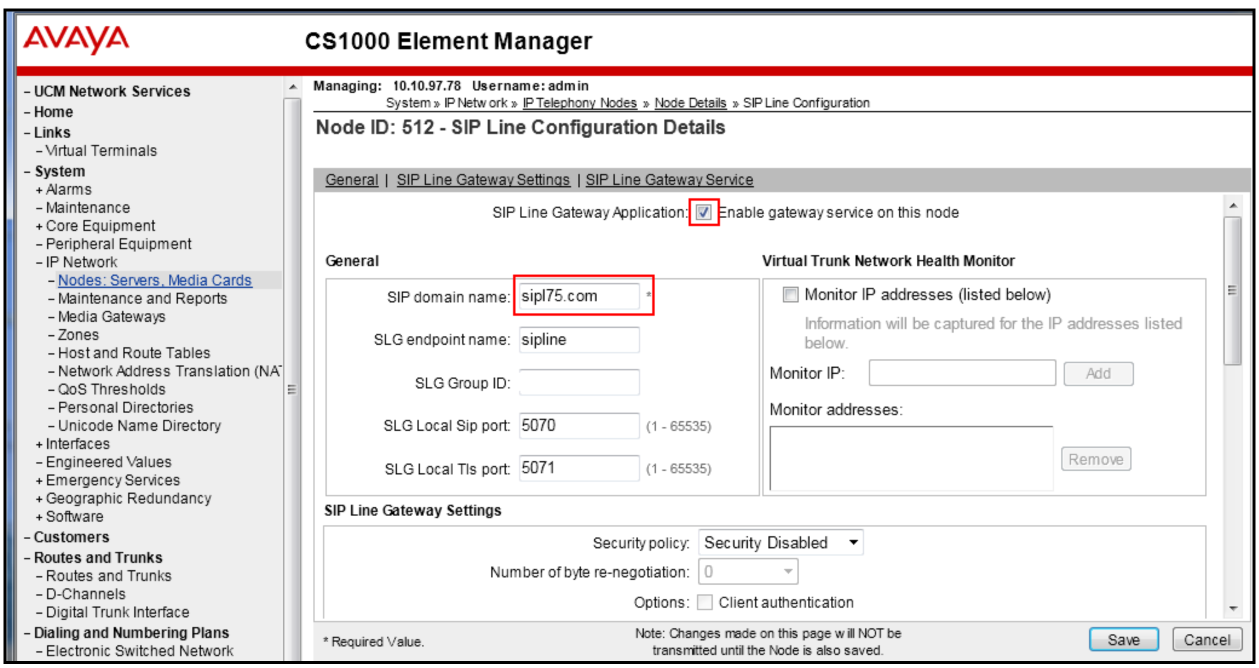
- **Node ID** text box: 512 → this is the node ID of SIP Line server.
- **Call Server IP Address** text box: 10.10.97.78.
- **Node IP Address** text box: 10.10.97.187 → this is the IP address that SIP endpoint uses to register to.
- **Subnet Mask** text box: 255.255.255.192.
- **Embedded LAN (ELAN) Gateway IP Address** text box: 10.10.97.65.
- **Embedded LAN (ELAN) Subnet Mask** text box: 255.255.255.192.
- Check **SIP Line** check box to enable SIP Line for this Node.



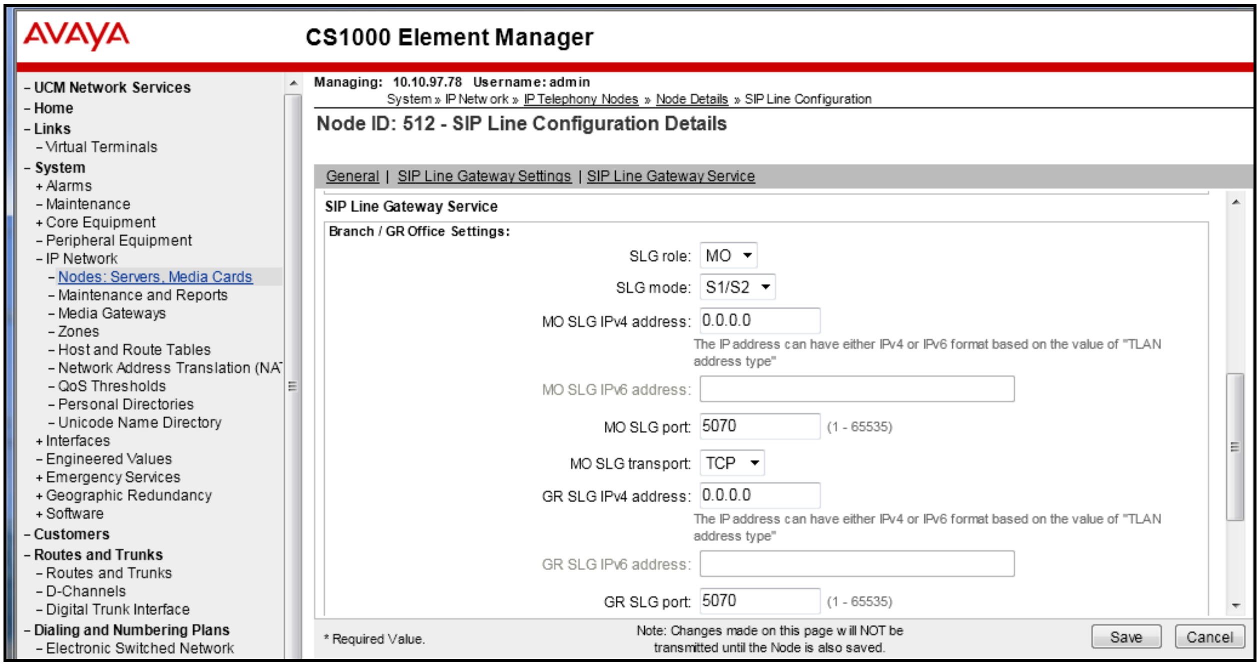
- Click on the **Next** button to go to next page**.** The page, **New IP Telephony Node** with Node ID, will appear as shown below.
- On the **Select to Add** drop down menu list, select the desired server to add to the node.
- Click the **Add** button
- Select the check box next to the newly added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



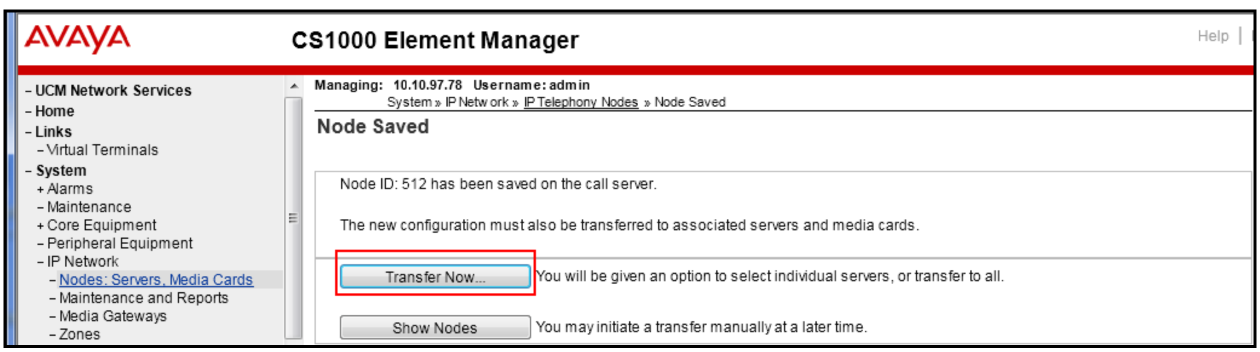
- Click on the **Next** button to go to next page**.** The **SIP Line Configuration Details** page appears.
- Enter SIP Line domain name in **SIP Domain name** text box, for example **sipl75.com**.
- Enable the **SIP Line Gateway Application** as show below.



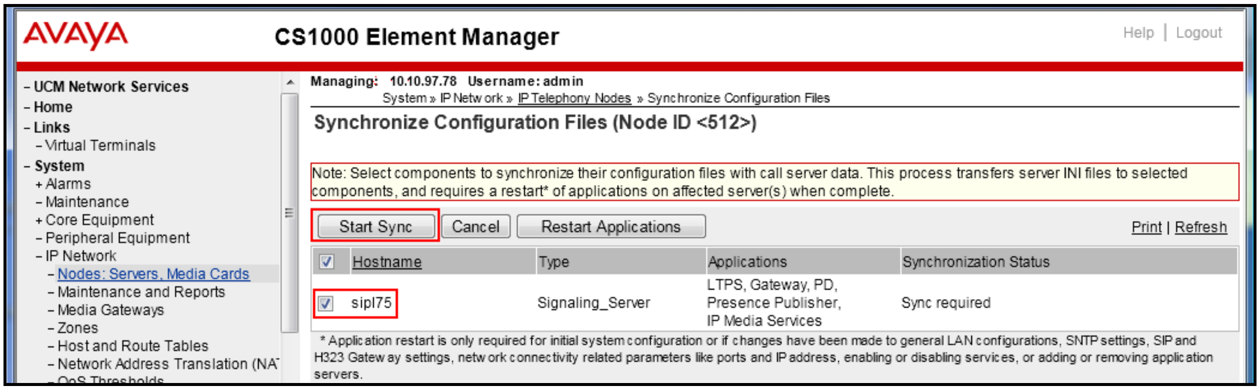
- Under the **SIP Line Gateway Services** section, select **MO** from the **SLG Role** list.
- From the **SLG Mode** list, select **S1/S2** (SIP Proxy Server 1 and Server 2).



- 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 as shown below.



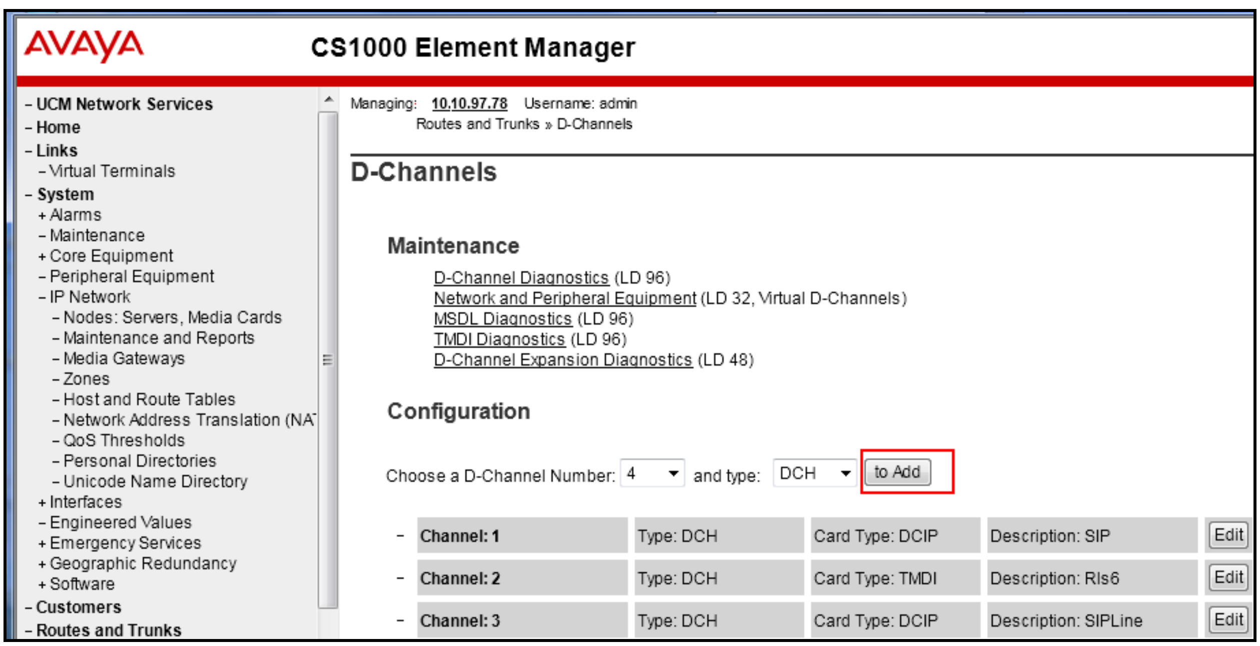
- Select the SIP Line server that associated with changes and then click on the **Start Sync** button to transfer the configuration files to the selected servers.



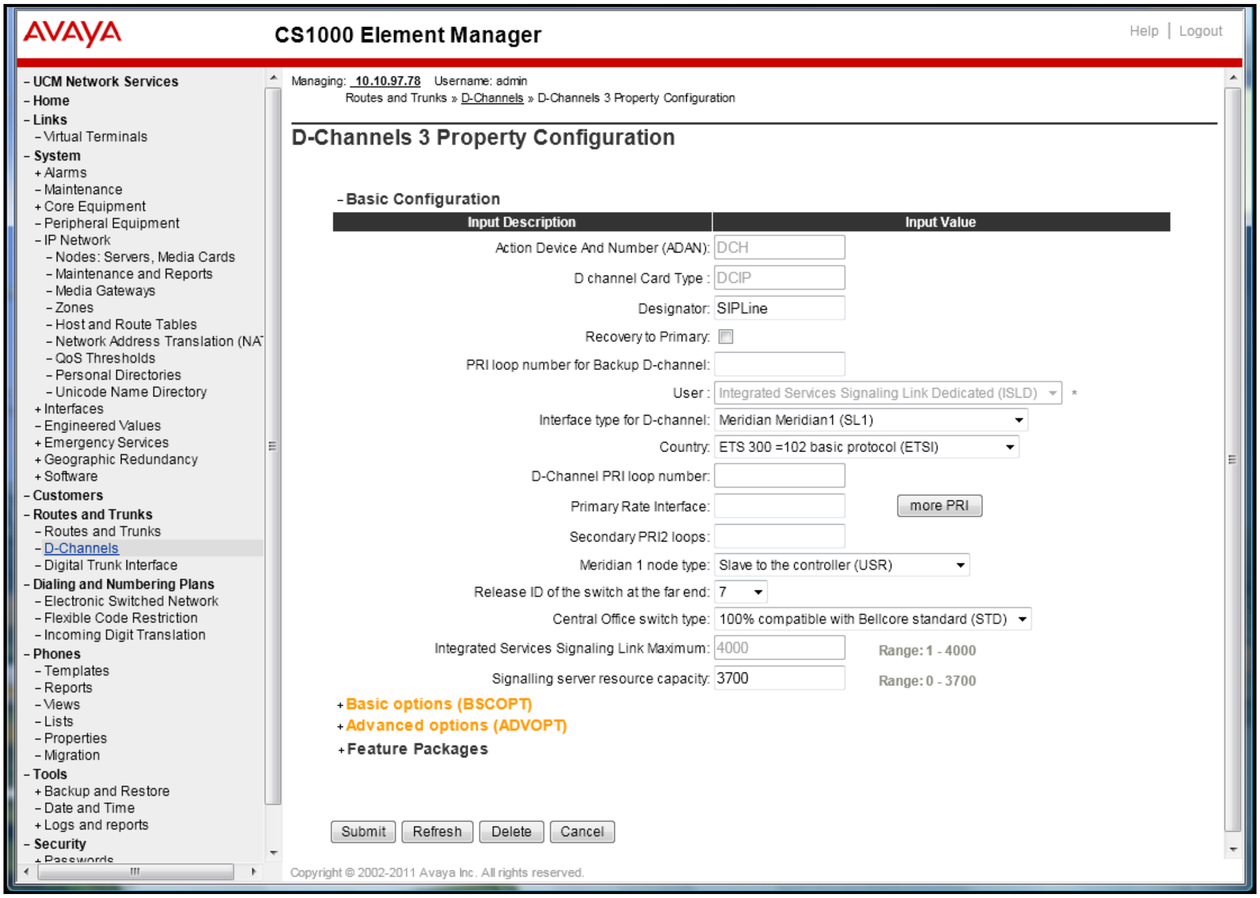
Note: The first time a new Telephony Node is added and transferred to the call server, the SIP Line services need to be restarted. To restart the SIP Line services, log in as administrator to the command line interface of the SIP Line server and issue command: **appstart restart**.

5.5. Create a D-Channel for SIP Line

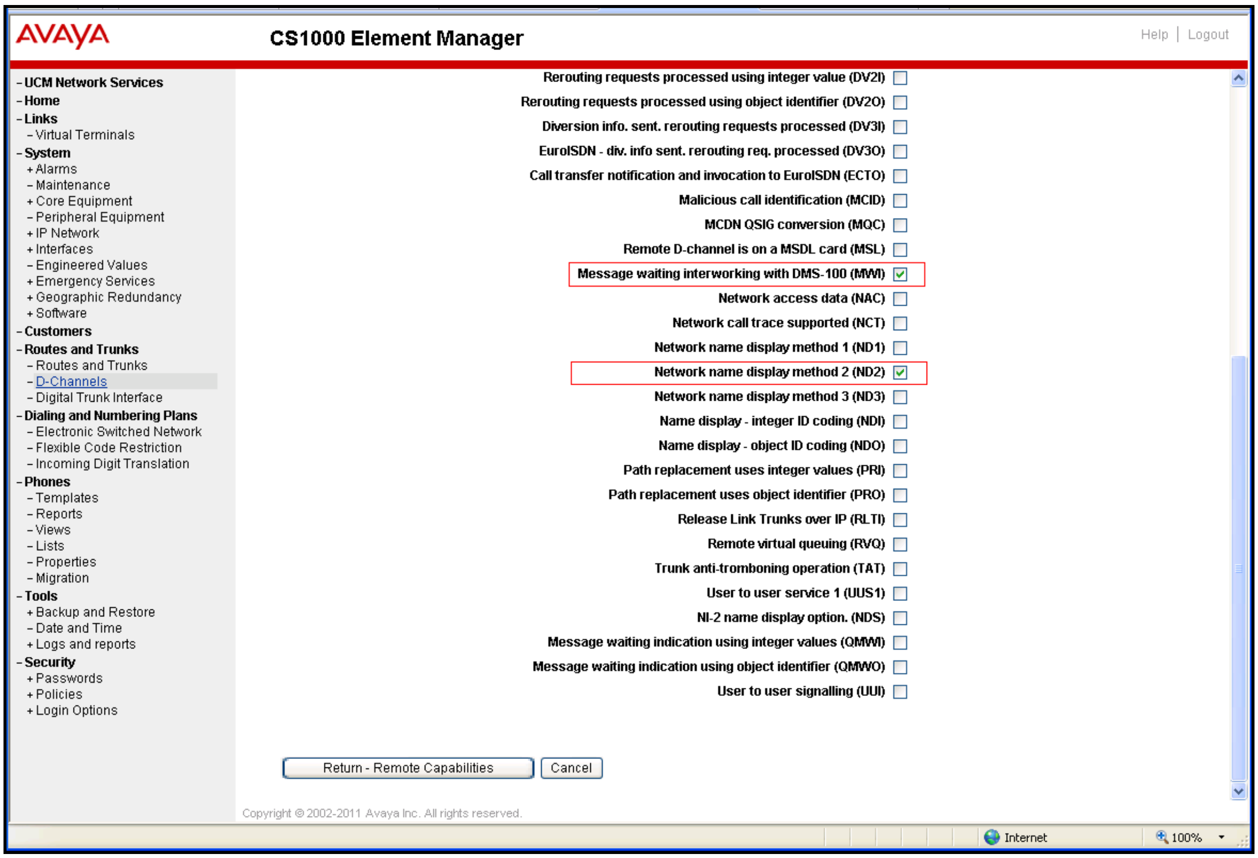
On the EM page, on the left column menu navigate to **Routes and Trunks → D-Channels**. Under the **Configuration** section as shown below, enter a number in the **Choose a D-Channel Number** field, and click on the **to Add** button.



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



- 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 below.
- 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.
- **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.

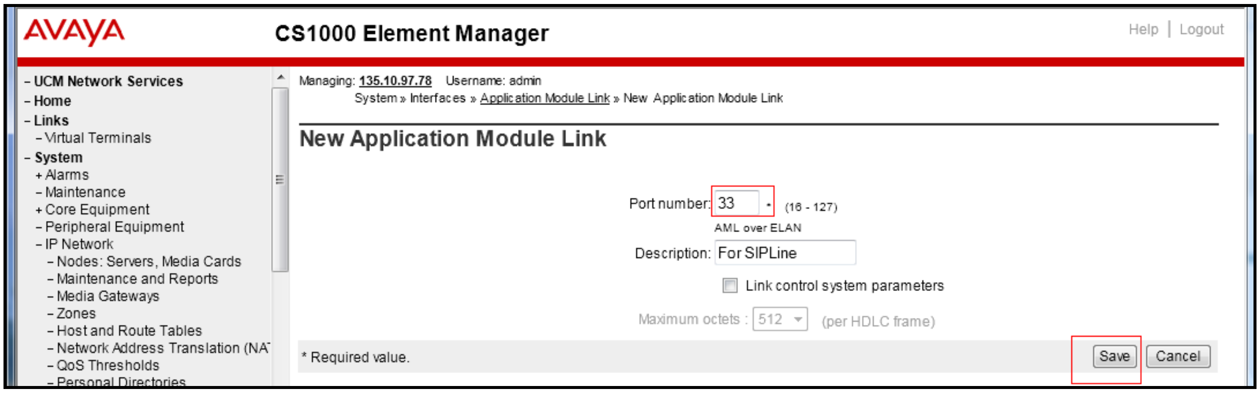


Click on the **Submit** button (not shown) of the **D-Channel Property Configuration** page to save changes.

5.6. Create an Application Module Link (AML)

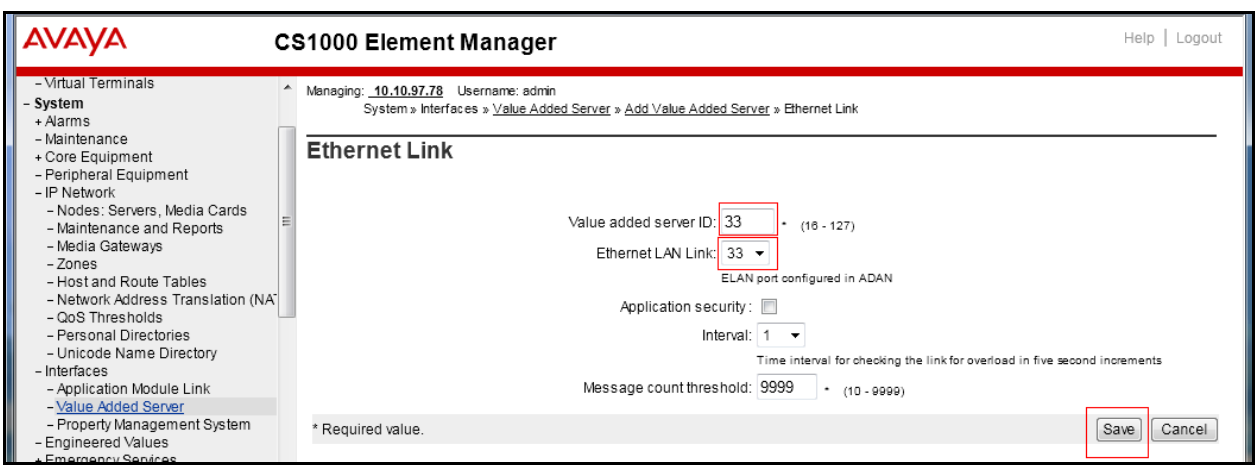
On the EM page, navigate to **System → Interfaces → Application Module Link**, click on the **Add** button to add a new Application Module Link (not shown). The **New Application Module Link** page appears as shown bellow.

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 **Save** to complete adding the AML link, and to save the configuration.



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 below.
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.

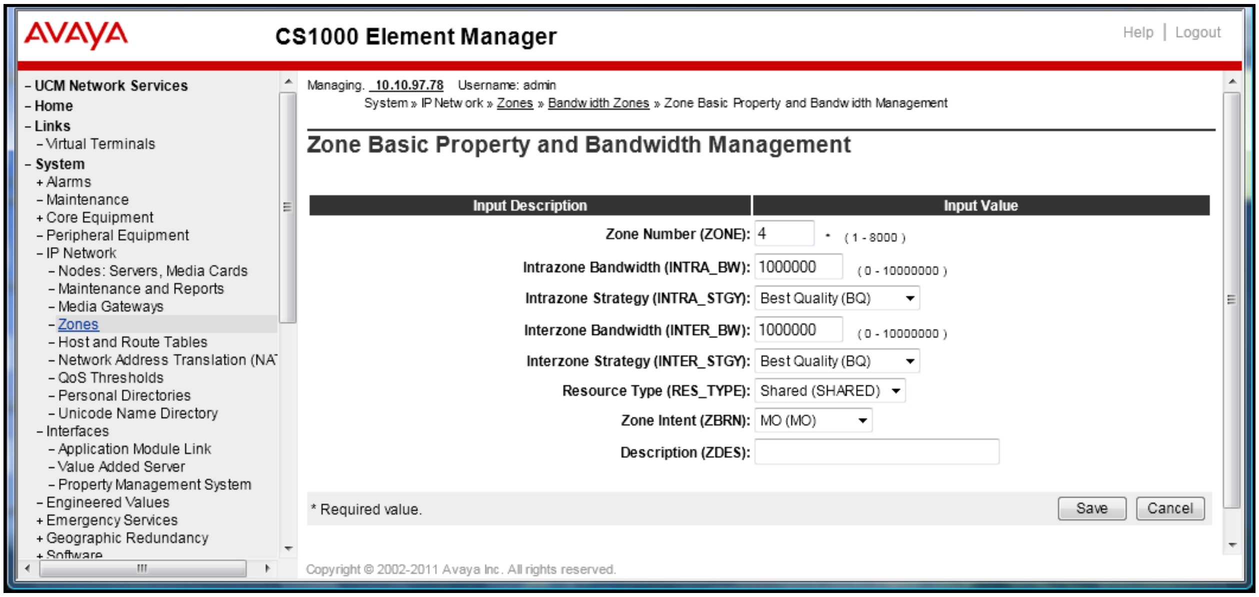


5.8. Create a Virtual Trunk Zone

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 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.

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**.



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

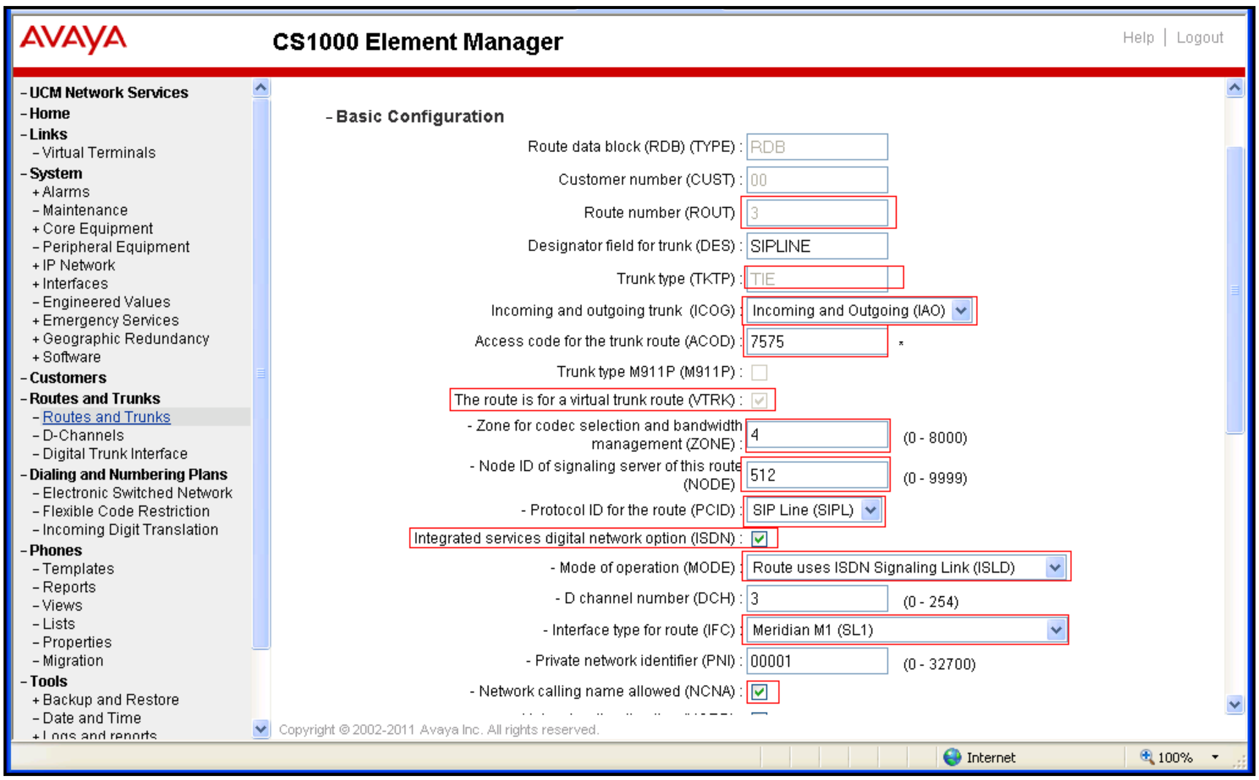
On the EM page, navigate to the menu **Routes and Trunks → Routes and Trunks**; the **Routes and Trunks** page appears (not shown). In this page, click on the **Add route** button next to the customer number that the route will belong to.

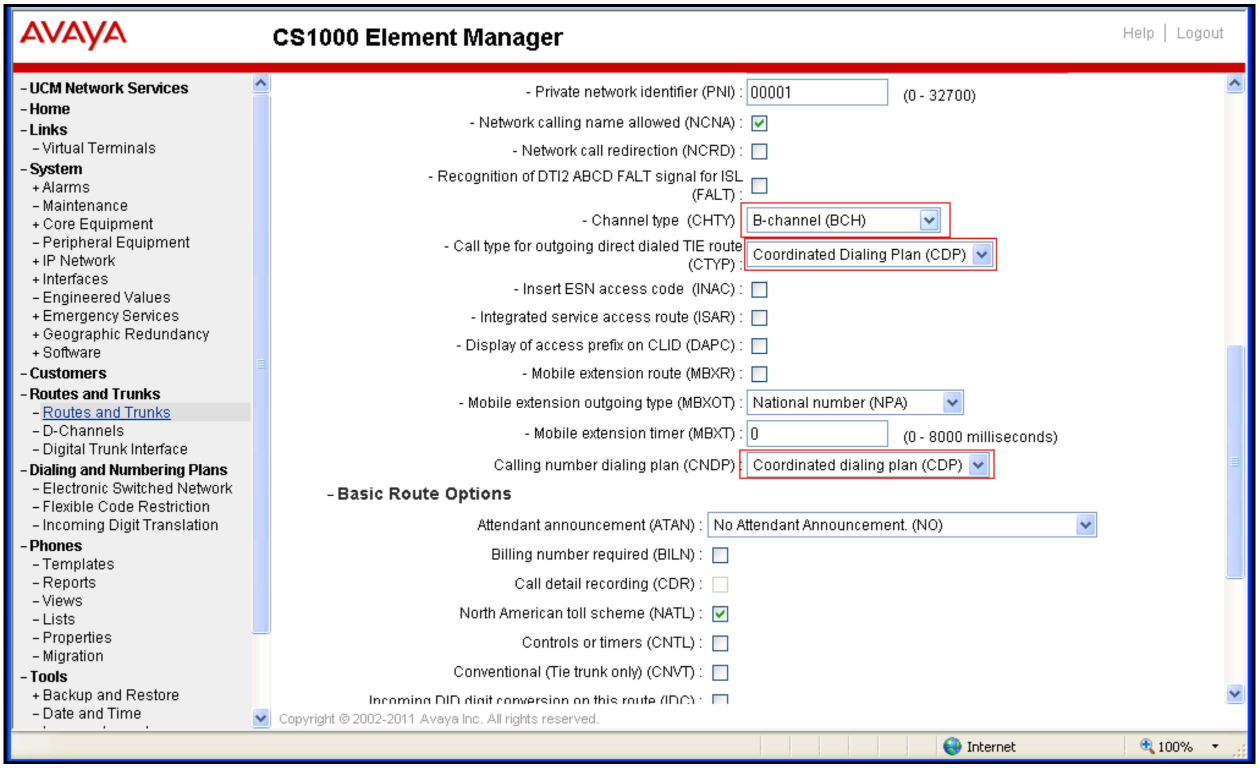
The **Customer ID, New Route Configuration** page appears, expand the **Basic Configuration** tab, and enter values below and as shown in next two figures.

- **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 7575.
- **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 5.8**.

- **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 5.5**.
- **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 **Submit** to complete adding the route and save configuration.





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 below, enter values for fields as shown below:

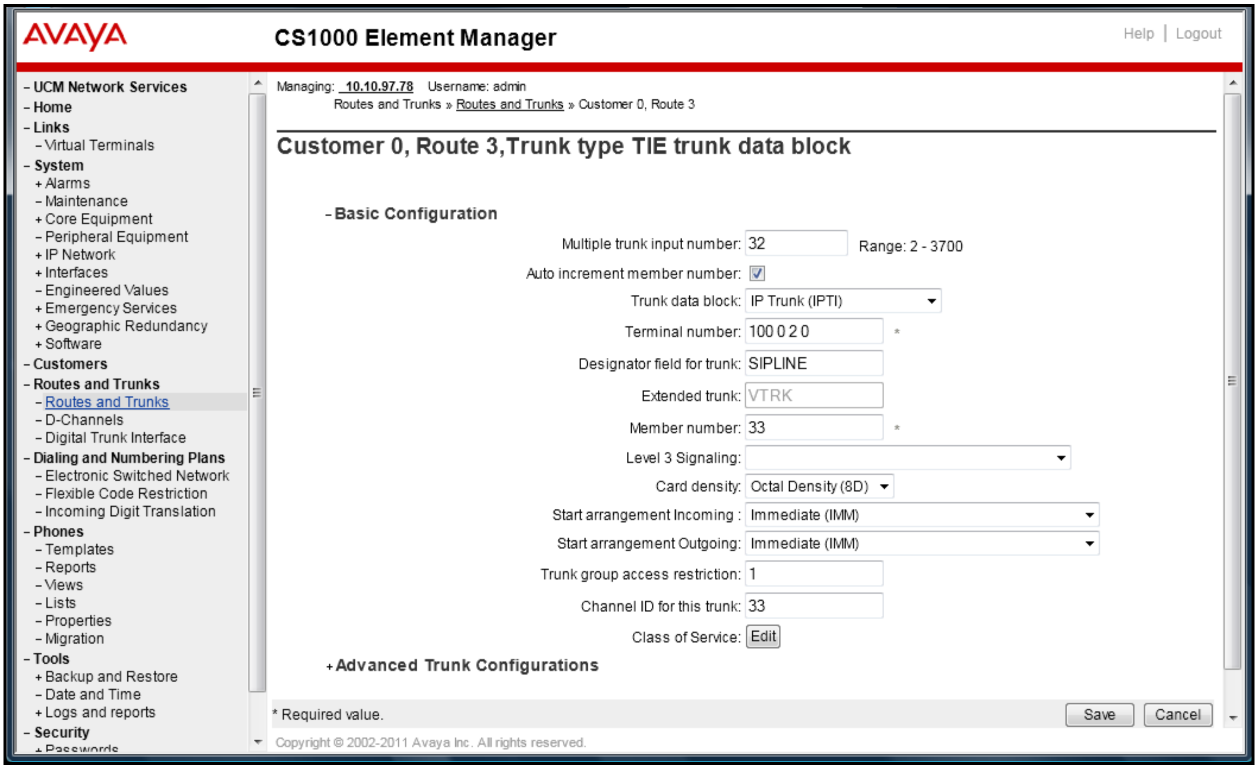
- **Multiple trunk input number (MTINPUT)**: 32 → create 32 trunks.
- **Auto increment member number**: checked.
- **Trunk data block**: IP Trunk (IPTI).
- **Terminal Number** (TN): 100 0 2 0 → enter the first TN of a range TN.
- **Member number**: 33, this is ID of trunk, just enter the first ID for 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** 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 **Save** to complete adding 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.

LD20
…..
REQ: new
TYPE: UEXT → Universal extension type for SIP line phone
TN 104 0 0 10
…
DES POLY1 → Phone Description
CUST 0
UXTY SIPL → Universal extension is SIP line type
MCCL YES
SIPN 0
SIP3 1
FMCL 0
TLSV 0
SIPU 54350 → SIP phone user extenstion DN
NDID 512 → Node ID of SIP line system
SUPR NO
….
NHTN
ZONE 1 → SIP line zone configured
MRT
….
FDN 54443 → Forward no answer to this DN, class of server FNA should be enable
TGAR 1
LDN NO
NCOS 7 → Network class server, 7 is highest level
SGRP 0
…..
XLST
SCPW 1234 → Password authentication for SIP user 54350
SFLT NO
CAC_MFC 0
CLS CTD **FBA** WTA LPR MTD **FNA HTA** TDD HFD CRPD MWA
….
CPND_LANG ENG
RCO 0
HUNT 54443 → Forward busy to this DN , need to enable class of server FBA and HTA
KEY 00 SCR 54350 0 MARP → Key 00 is SIP phone DN
CPND
CPND_LANG ROMAN
NAME Poly1 → SIP line phone display name
XPLN 13
DISPLAY_FMT FIRST,LAST
01 HOT U 2654350 MARP 0
…..

6. Configure Polycom SoundStation Duo SIP interface

This section describes how to set up the Duo network interface, to access the Polycom Duo SIP endpoint web interface and to configure the Polycom Duo for testing.

6.1. Determine the IP address used by the Polycom Duo

This section shows how to configure network IP address used by the Polycom Duo.

On the Polycom Duo (not shown), push the '**Menu**' button and navigate to **Settings → Advanced.** Enter phone administrator password, **456** and press **Enter**. Continue to navigate to **Admin Settings → Network Configuration → Ethernet Menu**.
In this example configuration, the following parameters are used as below. Others are left at default.

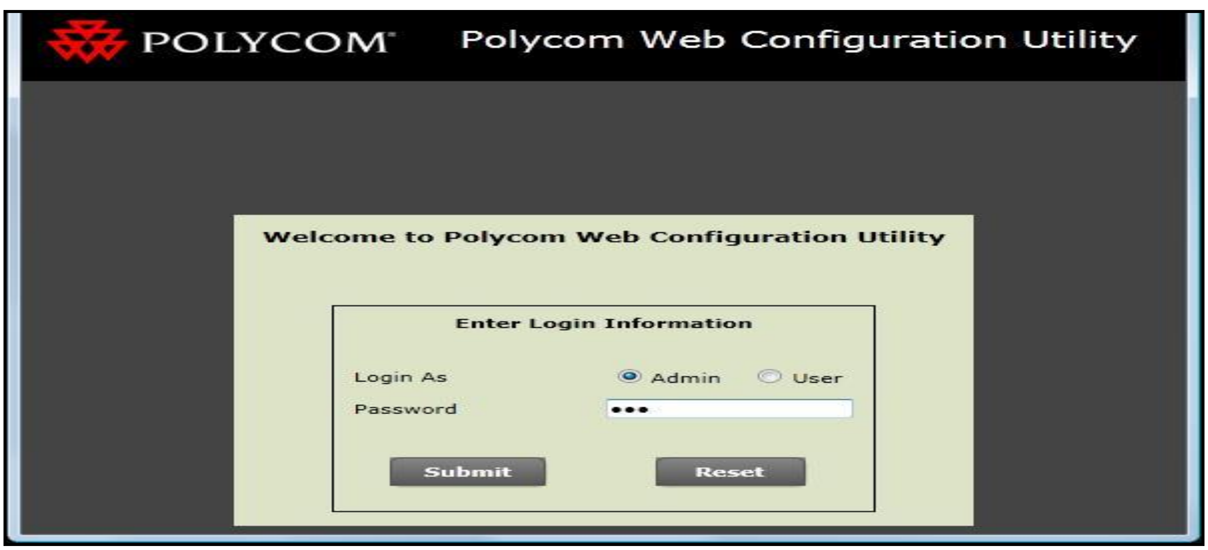
- **DHCP**: Disable
- **IP Address**: 010.033.005.204
- **Subnet Mask**: 255.255.255.000
- **IP Gateway**: 010.033.005.001

Click **Save** and reboot the Duo.

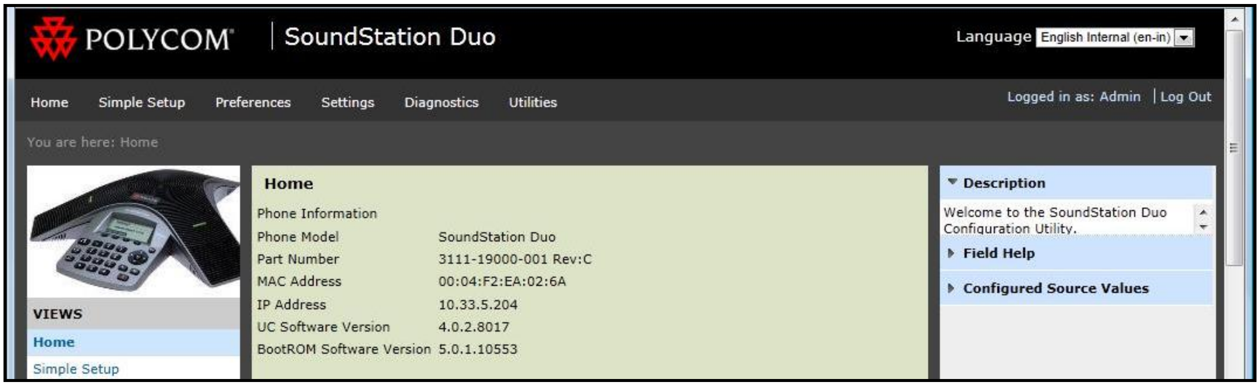
6.2. Login to Polycom Duo SIP Web Browser

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

Open the web browser, and in the address field enter the Polycom Duo IP address as format **http://10.33.5.204** and the Polycom Duo login page will appear as shown below. Enter default password, **456**.

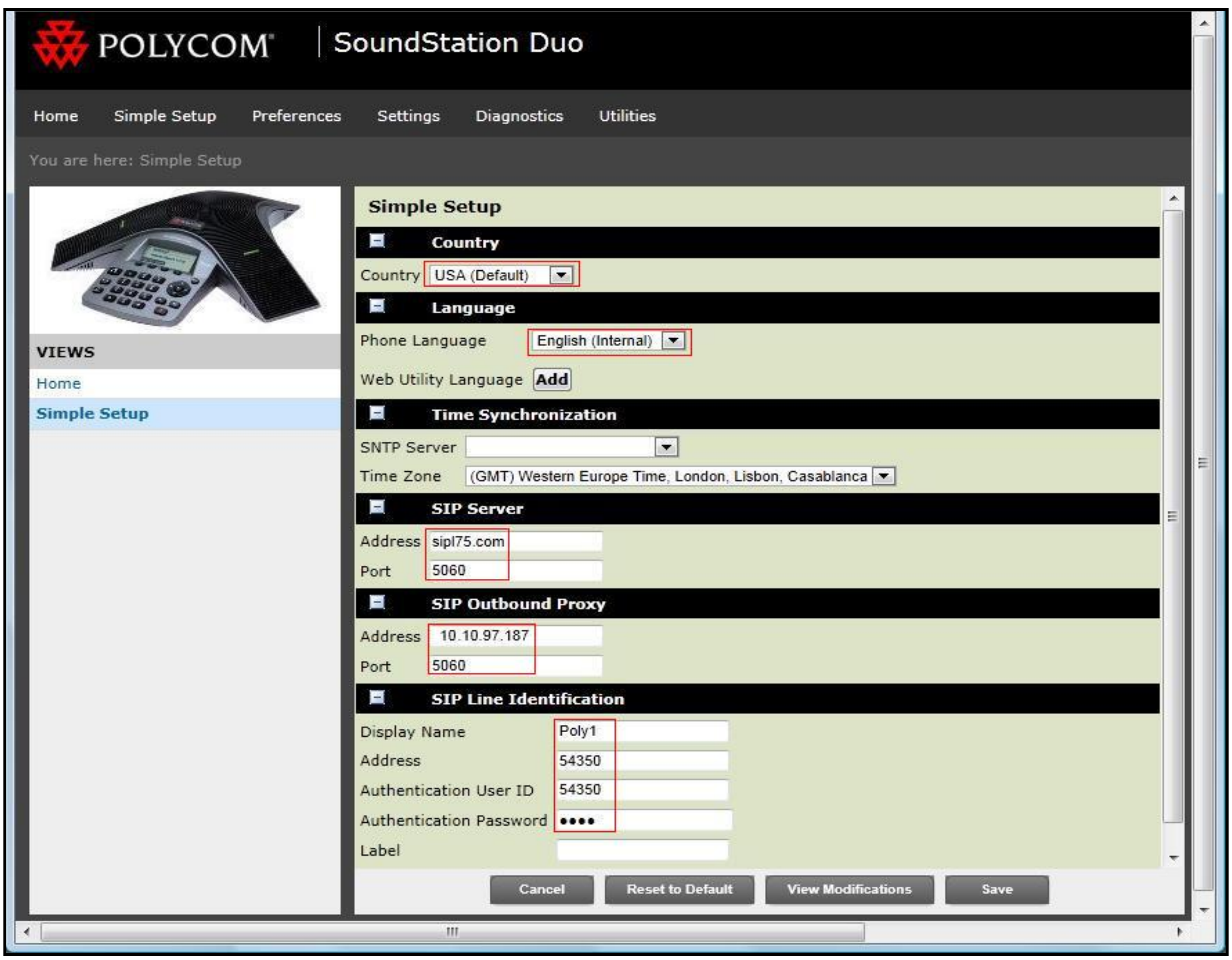


Click **Submit**, the homepage of Polycom Duo appears.



6.3. Configure the Lines for Polycom Duo

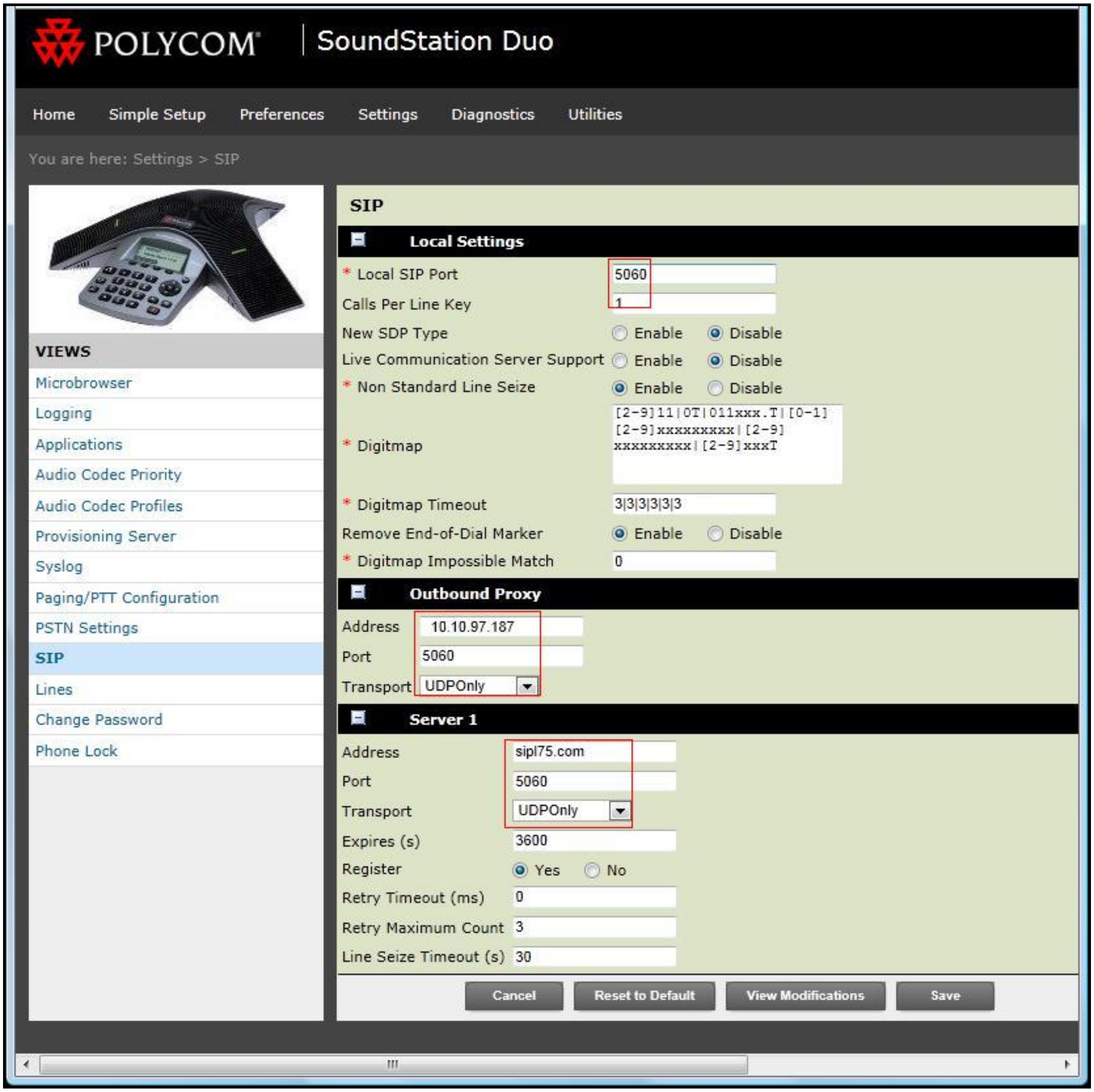
This section shows how to configure the Polycom Duo to register with CS1000.
On the homepage of configuration screen, click on the **Simple Setup** menu, the **Simple Setup** page appears as shown below. Enter values as highlighted in red-box and others are left at default. Click **Save**.



6.4. SIP Settings

This section shows how to set SIP parameters for Polycom Duo.

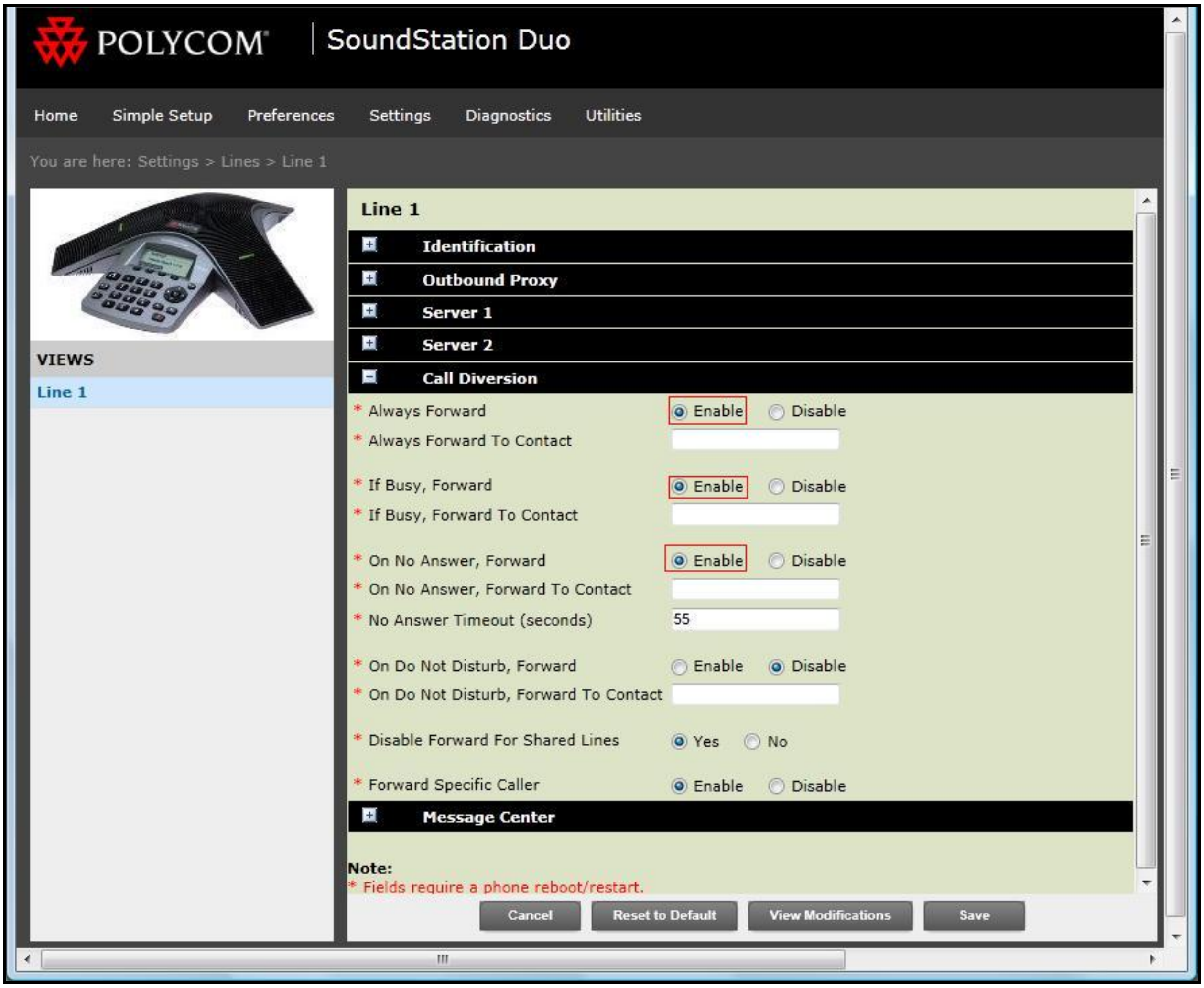
On the homepage of Polycom Duo, navigate to menu **Settings → SIP**, **SIP** page appears. Enter values as highlighted in red-box and others are left at default. Click **Save**.



6.5. Local Call Forward Settings

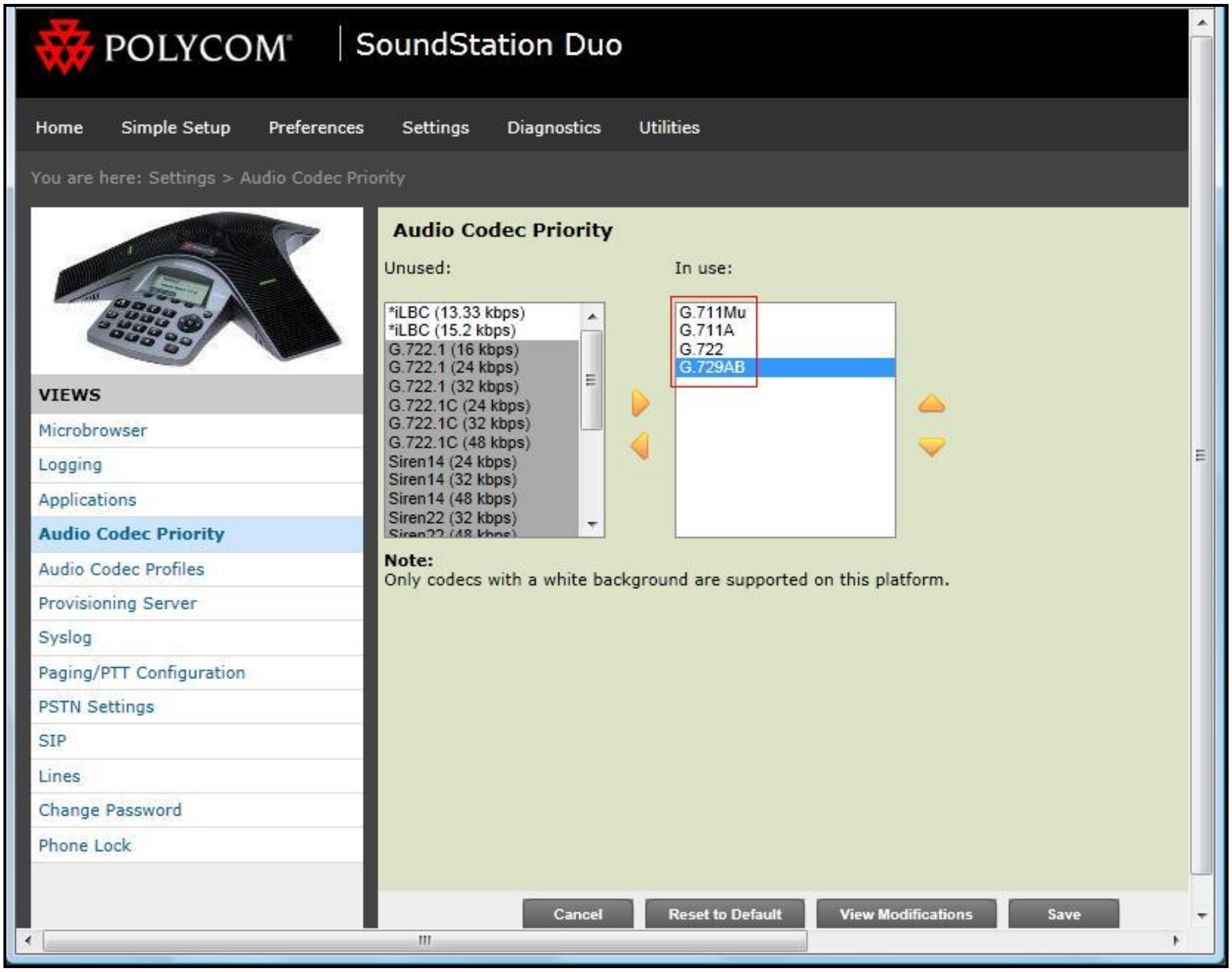
This section shows how to set up call forward settings for Polycom Duo.

On the homepage of Polycom Duo, navigate to menu **Settings → Lines** page appears. Enable values as highlighted in red-box and others are left at default. Click **Save**.



6.6. Codec Settings

On the homepage of Polycom Duo, navigate to menu **Settings → Audio Codec Priority**. Select the codec list as shown below. Click **Save**.



7. Verification Steps

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

Step1

Verify that the Poycom Soundstation Duo registers successfully with the CS 1000 SIP Line Gateway server by using the CS 1000 Linux command line.
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@sipl75 ~]$ slgSetShowByUID 54350

=== VTRK ===
UserID           AuthId      TN              Clients  Calls  SetHandle  Pos
ID    SIPL Type
---------------  ----------  --------------  -------  -----  ---------  ----
---   ---------
          54350       54350     104-00-00-10        1      0  0xa2be010
SIP Lines
        StatusFlags = Registered Controlled KeyMapDwld SSD
        FeatureMask =
        CallProcStatus = 0

        Current Client = 0, Total Clients = 1

         == Client 0 ==
         IPv4:Port:Trans = 10.33.5.204:5060:udp
         Type            = SIP3
         UserAgent       = PolycomSoundStation-SS_Duo-UA/4.0.2.8017
         x-nt-guid       = 04ce39958fd7849bdee075e58fd5ce1f
         RegDescrip      =
         RegStatus       = 1
         PbxReason       = OK
         SipCode         = 200
         hTransc         = (nil)
         Expire          = 3600
         Nonce           = 03bf57d83ed58c1456a877737ec0eccd
         NonceCount      = 5
         hTimer          = 0xa237d78
         TimeRemain      = 1775
         Stale           = 0
         Outbound        = 0
         ClientGUID      = 0
         MSec CLS        = MSNV (MSEC-Never)
         Contact         = sip:54350@10.33.5.204
         KeyNum          = 255
         AutoAnswer      = NO

        Key  Func  Lamp  Label
        0    3     0     54350
```

```
1    126   0     2654350
3    29    0
4    9     0
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
```

Step 2

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

These Application Notes illustrate the procedures necessary for configuring the Polycom Duo to interoperate with the Avaya CS1000. All feature functionality test cases described in **Section 2.2** were passed.

9. Additional References

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

[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

Product documentation for the Polycom Soundstation Duo products may be found at:
http://www.polycom.com

[2] Polycom SpectraLink 8400 Series Documents:
Administrator's Guide for the Polycom® UC Software
http://support.polycom.com/PolycomService/support/us/support/voice/soundstation_ip_series/soundstationduo.html

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.