Avaya Solution & Interoperability Test Lab

Application Notes for Configuring the Polycom SoundStructure VoIP Interface running UC software release 4.0.2B with Avaya Communication Server 1000 Release 7.5 - Issue 1.0

Abstract

These Application Notes describe a solution for supporting interoperability between the Polycom SoundStructure VoIP interface card running UC software release 4.0.2B with Avaya Communication Server 1000 release 7.5. Emphasis of the testing was to verify voice calls of SoundStructure 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 detailed configurations of Avaya Communication 1000 (hereafter referred to as CS1000) and the Polycom SoundStructure VoIP Interface (hereafter referred to as SoundStructure) used during the compliance testing. SoundStructure 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 SoundStructure, where applicable, to ensure the interoperability with CS1000.

2. General Test Approach and Test Results

The general test approach was to have SoundStructure register to the SIP line gateway of CS1000. Calls were then placed from CS1000 telephone clients/users to and from SoundStructure. 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 SoundStructure was able to interoperate with Avaya CS1000 SIP line system. The following areas were tested:

- Registration of SoundStructure to the CS1000 SIP line gateway.
- Call establishment of SoundStructure 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 G.722.
- Incoming and Outgoing calls to SoundStructure from PSTN.

2.2. Test Results

The objectives outlined in **Section 2.1** were verified. SoundStructure was registered to CS1000 SIP line gateway successfully. Calls have been made between CS1000 telephones and SoundStructure with clear voice path. All executed test cases passed with the following observations,

- G.722 is not supported on Avaya IP (UNIStim) deskphone and therefore could not be tested however it was tested on Avaya IP (SIP) deskphone since it is supported here.
- When performing 3-way conference where SoundStructure is a host of the conference and CS1000 telephones are the participants, as SoundStructure hangs up, the 2 CS1000

participant telephones are not able to establish the voice path. This issue is being investigated by the CS1000 team.

- The Split feature of SoundStructure does not function as intended in this integration.
- Call Forward Unconditional has to be configured on the Web Administration page of SoundStructure where the Always Forward radio button has to be disabled and the number to be forwarded to has to be entered in the Always forward to Contact field. Also the timer in the No Answer Timeout field has to be set to 0.
- Call Forward No Answer (CFNA) has to be configured on CS1000 at the set level and not through the SoundStructure Web Administration page.
- Call Forward on Busy (CFB) has to be configured on CS1000 at the set level and not through the SoundStructure Web Administration page.

2.3. Support

Technical support for the Polycom SoundStructure installed audio system and VoIP Interface 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. The SoundStructure VoIP interface card is inserted into the SoundStructure C series box. The VoIP interface card registers to CS1000 as a third party SIP endpoint via the SIP Line Gateway. The SoundStructure Studio application is used to manage the configuration of the VoIP interface card.

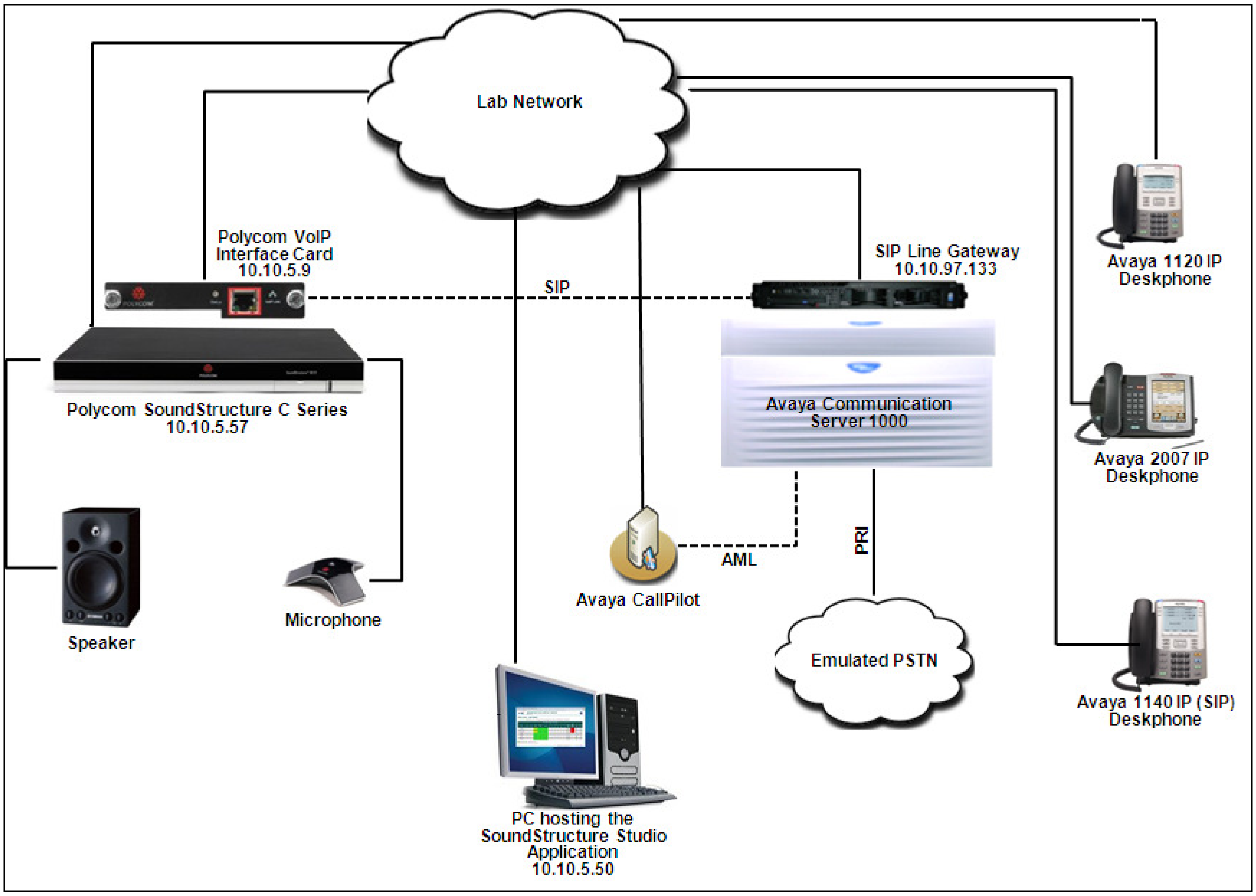


Figure 1: Network Configuration Diagram

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 SIP Line Gateway	7.50Q 7.50Q
Avaya Call Pilot	5.00.41
Avaya CS1000 IP Phones: 2007 1120	 0621C8L 0624C8L
Avaya CS1000 SIP Phone: 1140	 04.03.12
Polycom SoundStructure VoIP Interface Card	4.0.2.11307
Polycom SoundStructure C Series	1.6.1
Polycom SoundStructure Studio running on Windows XP Professional OS	1.8.0.13

Note: In general the C series firmware and Studio version do not affect the VoIP card, which runs software independently (and should work with prior and future versions of firmware and Studio that are compatible with VoIP card). It is therefore recommended to always use the latest available C series firmware and Studio version.

5. Configure Avaya CS1000 – SIP Line

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

The following is the summary of tasks that needs to be done for configuring the CS1000 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 CS 1000 SIP Line server has been:

- Installed with CS1000 Release 7.5 Linux Base.
- Joined CS1000 Release 7.5 Security Domain.
- Deployed with SIP Line Application.

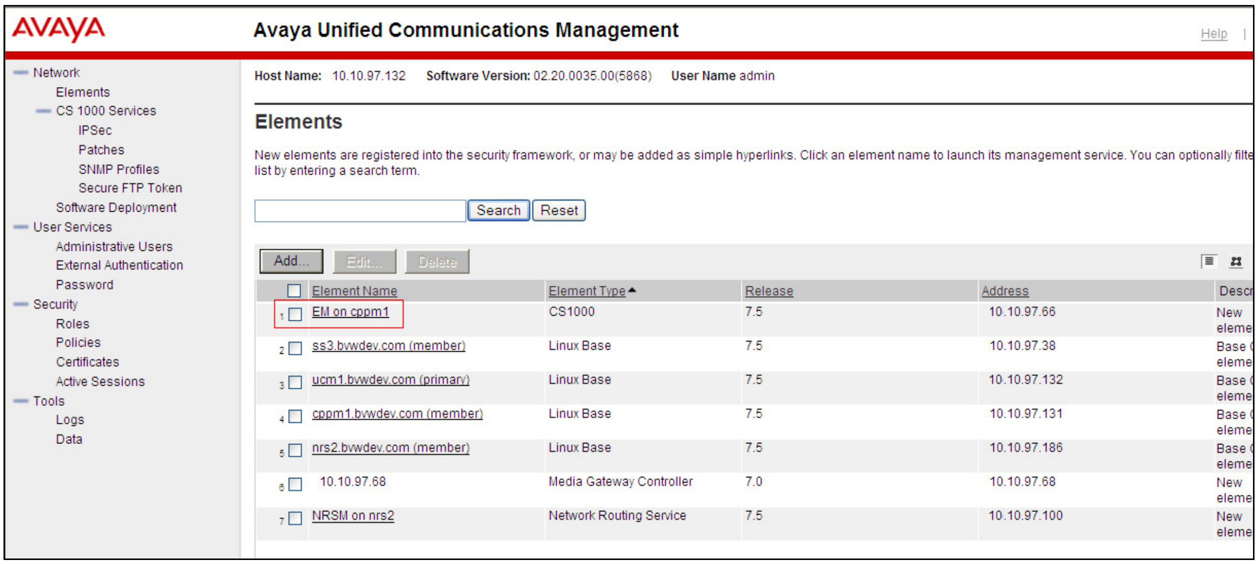
The following packages need to be enabled in the key code. If any of these features have not been enabled, please contact your Avaya account team or Avaya technical support at http://www.avaya.com.

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

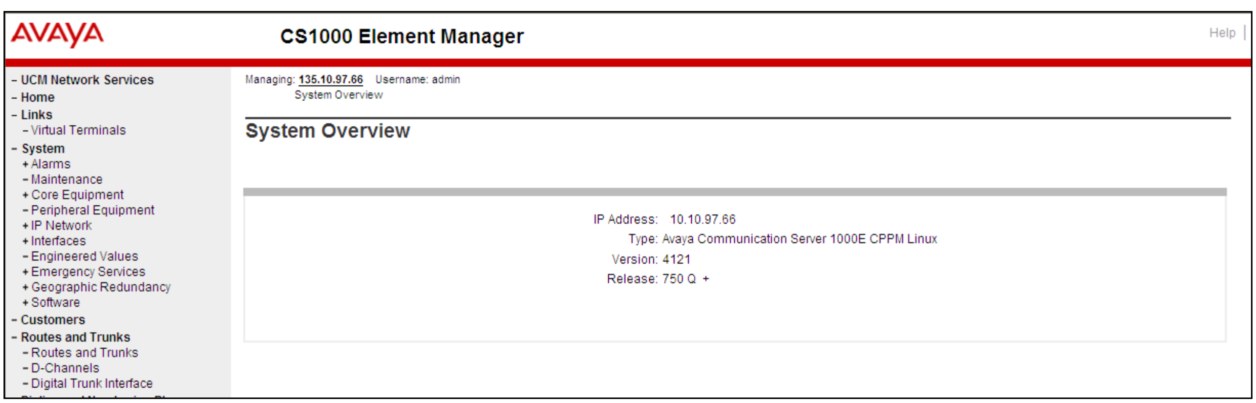
5.2. Log in to Unified Communications Management (UCM) and Element Manager (EM)

Use a web browser to launch 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 **Section 9[2].**

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.



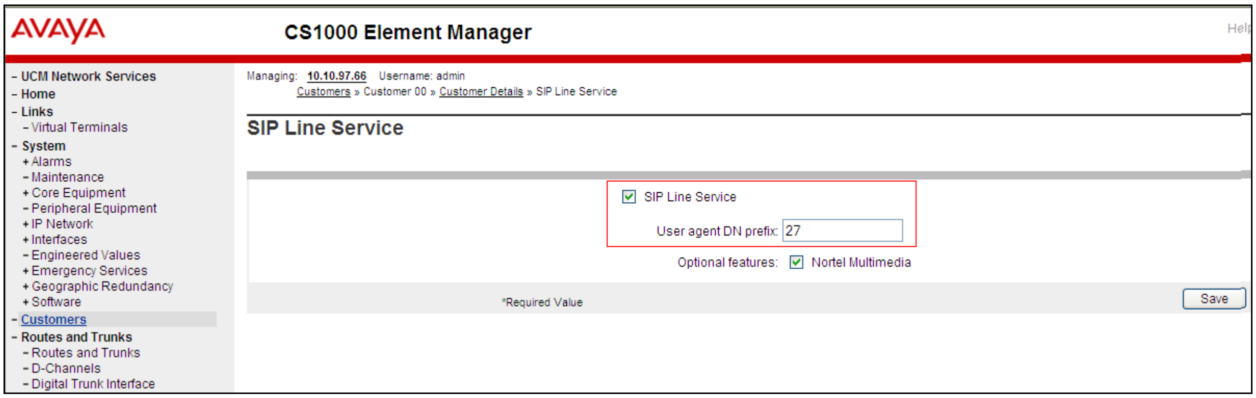
The Avaya CS1000 Element Manager (EM) page appears as shown.



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).

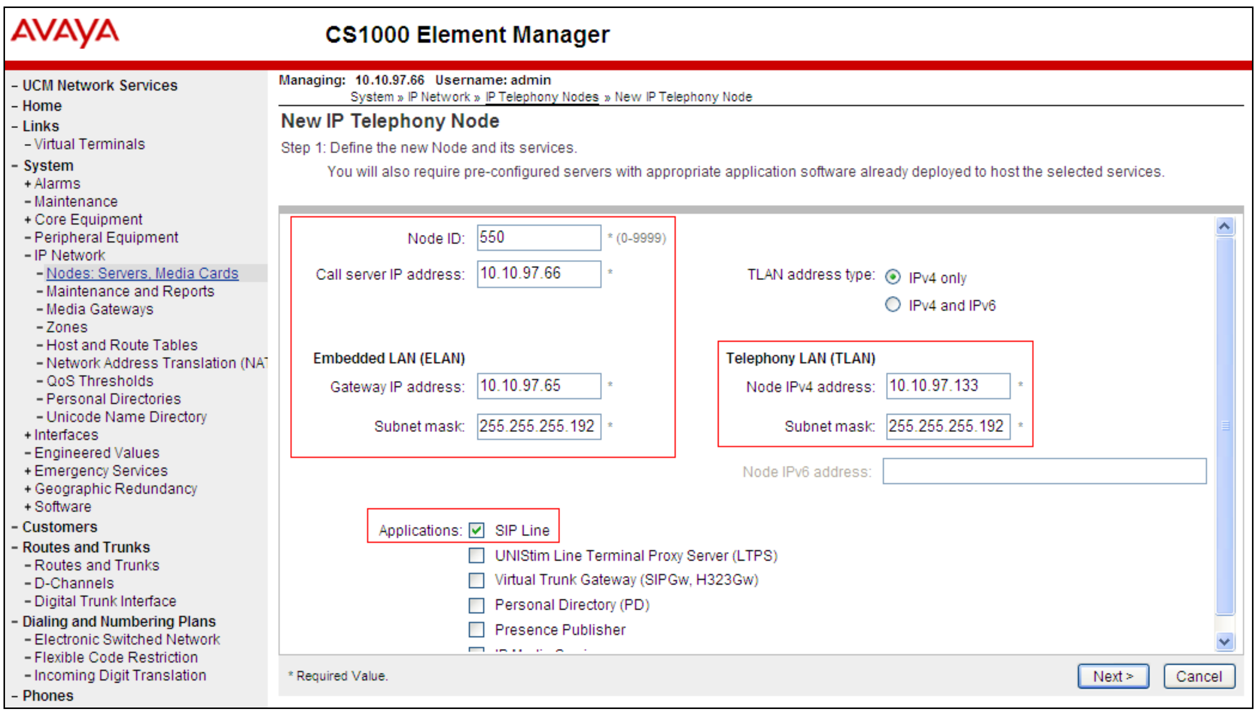
- Enable SIP Line Service by clicking on the **SIP Line Service** check box.
- Enter the prefix number in the **User agent DN prefix** text box. During compliance testing a value of **27** was used.
- Rest of the values remains at default.
- Click on **Save**.



5.4. Add a new SIP Line Telephony Node

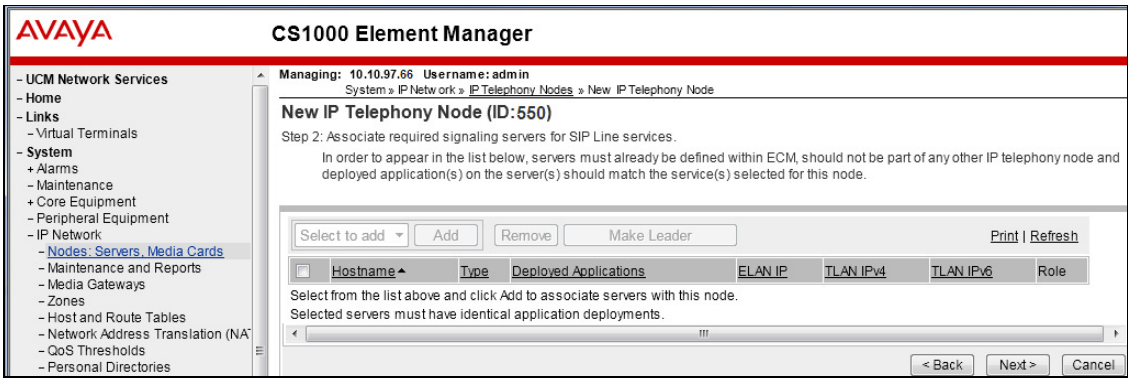
On the EM page, navigate to menu **System → IP Network → Nodes: Servers, Media Cards**. Click **Add** to add a new SIP Line Node to the IP Telephony Nodes. The new IP Telephony Node page appears as shown below. Enter the information as shown below:

- **Node ID** text box: 550; this is the node ID of SIP Line server.
- **Call Server IP Address** text box: 10.10.97.66
- **Node IP Address** text box: 10.10.97.133; 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.

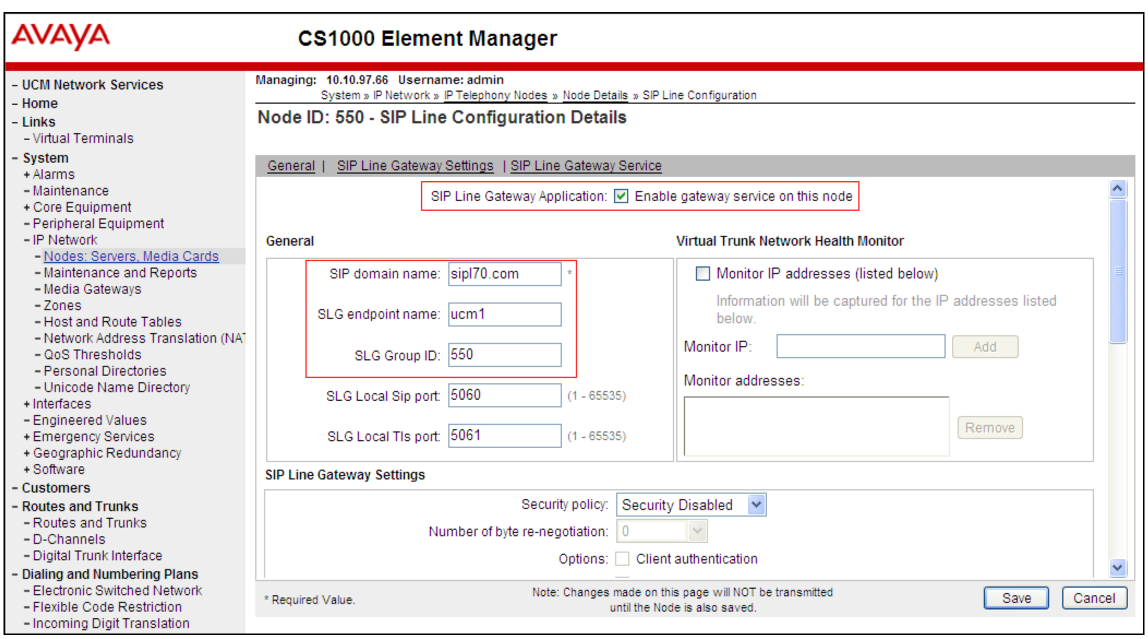
- On the **Select to Add** drop down menu list, select the desired server to add to the node (not shown).
- Click the **Add** button
- Select the check box next to the newly added server, and click **Make Leader** (not shown).
- Click on the **Next** button to go to next page



The **SIP Line Configuration Details** page appears as shown below. In the **General** section,

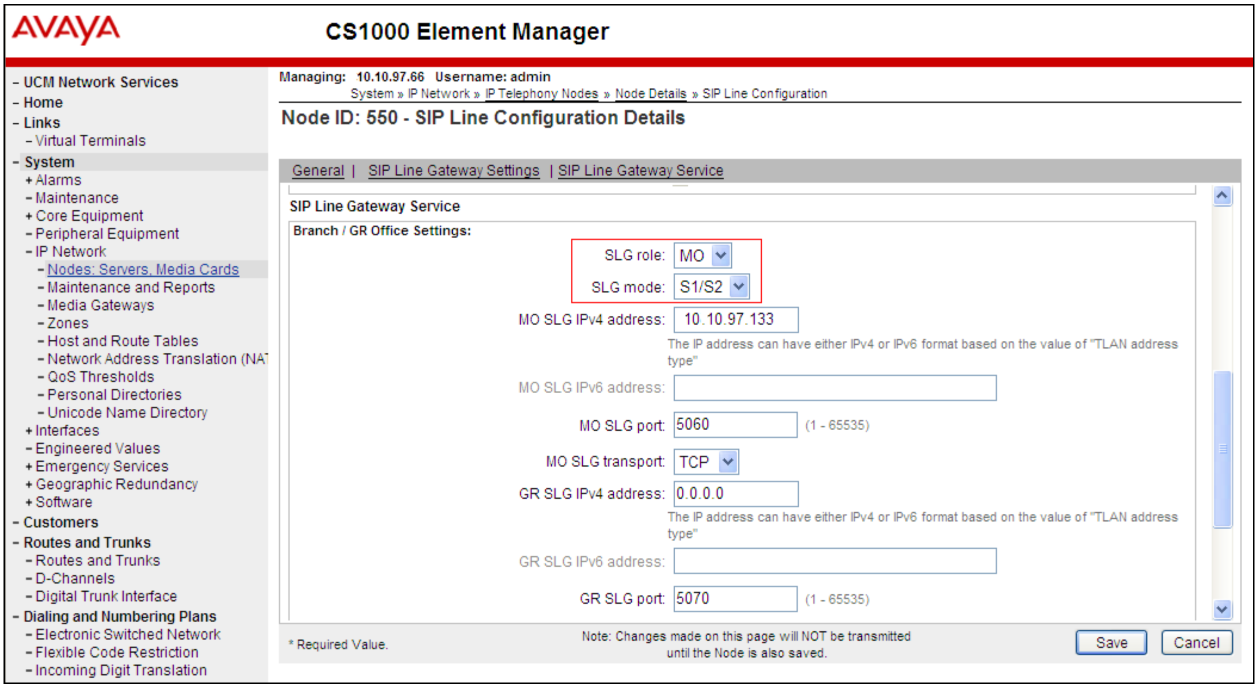
- Check the **Enable gateway service on this node** box.
- Enter SIP Line domain name in **SIP Domain name** text box, during compliance testing **sipl70.com** was used.
- Enter the **SLG endpoint name**. During compliance testing **ucm1** was used.
- Enter the **SLG Group ID**. During compliance testing **550** was used.
- Retain default values for all other fields.

<u>Note:</u> that SIP Line Gateway is configured by default to allow both UDP and TCP transport protocols.

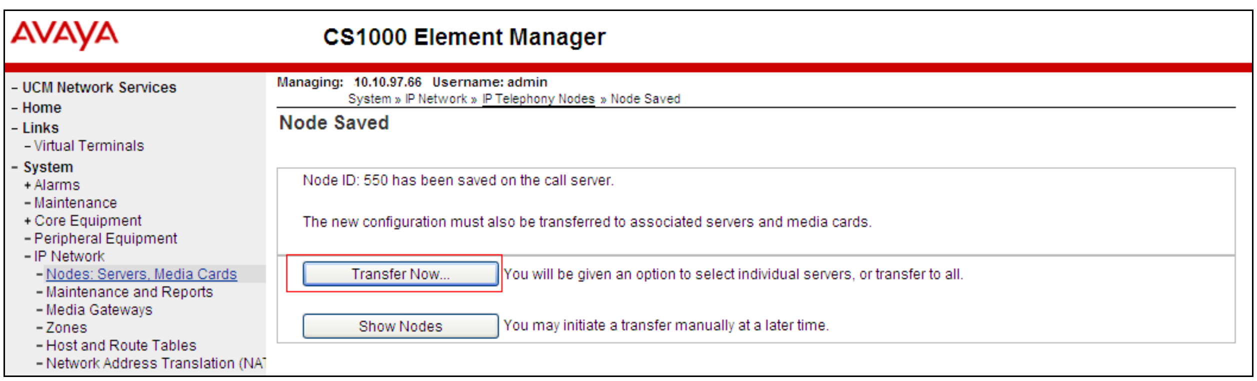


Under the **SIP Line Gateway Services** section,

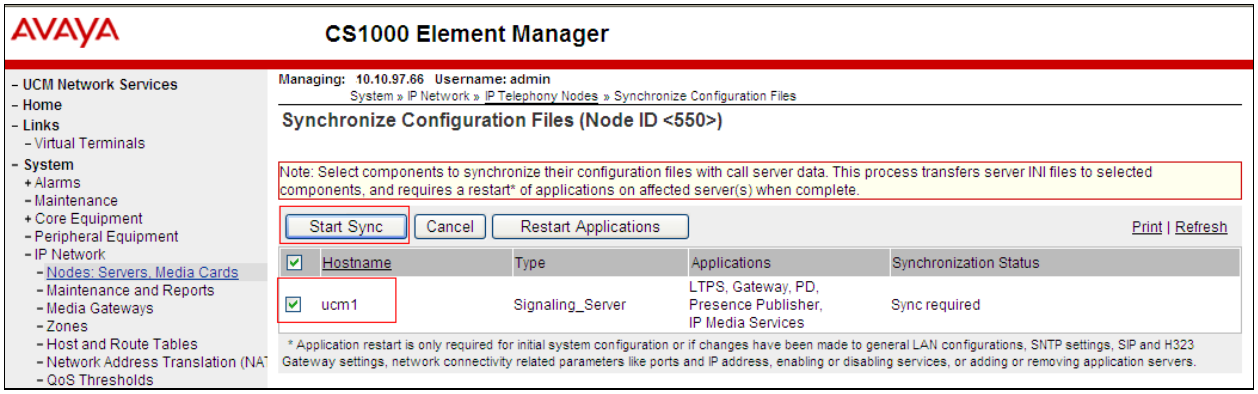
- Select **MO** from the **SLG Role** drop down menu.
- From the **SLG Mode** list, select **S1/S2** (SIP Proxy Server 1 and Server 2).
- Retain default values for all other fields.



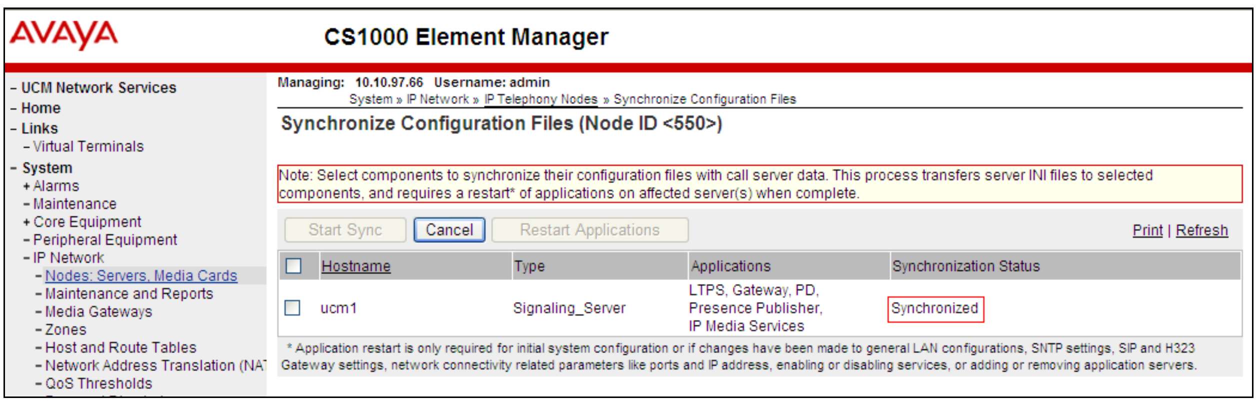
- Click **Next**. The **Confirm new Node details** page appears (not shown) and then **Save**.
- Click on the **Transfer Now** button as shown in the screen below.



- The **Synchronize Configuration Files (Node ID <550>)** page appears as shown below.
- Select the SIP Line server associated with the changes and then click on the **Start Sync** button to transfer the configuration files to the selected servers.



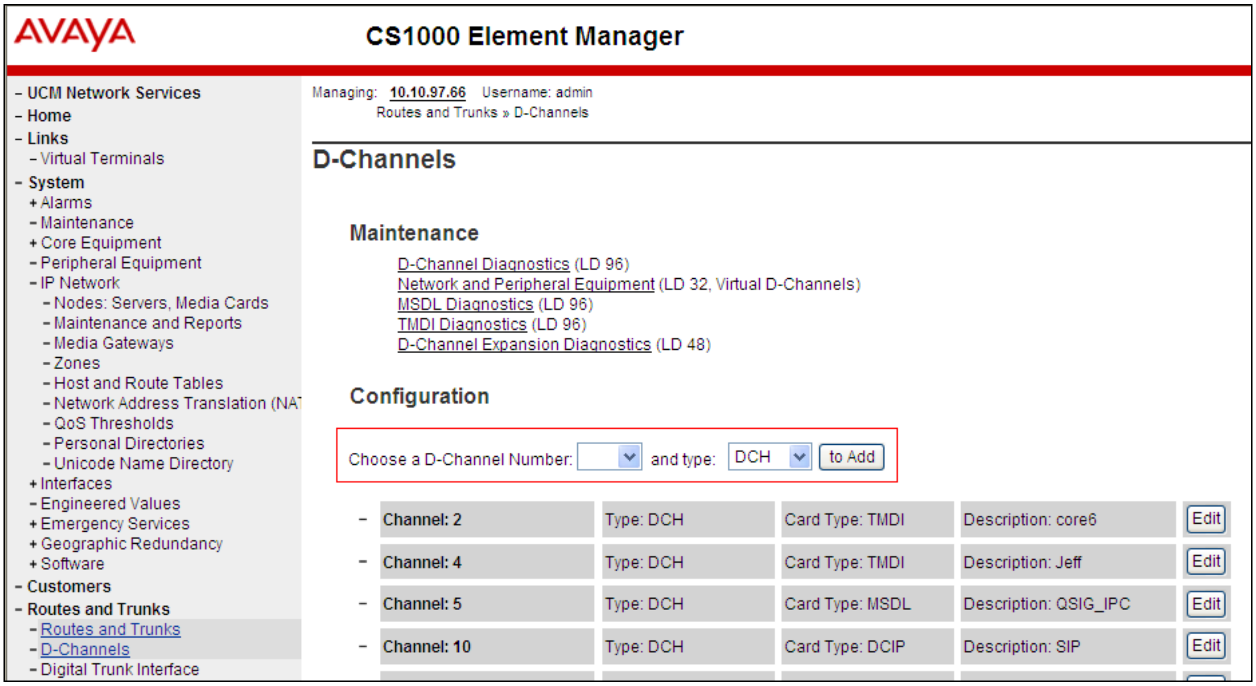
- Ensure that the synchronization is completed by checking the **Synchronization Status** column as shown below.



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**.

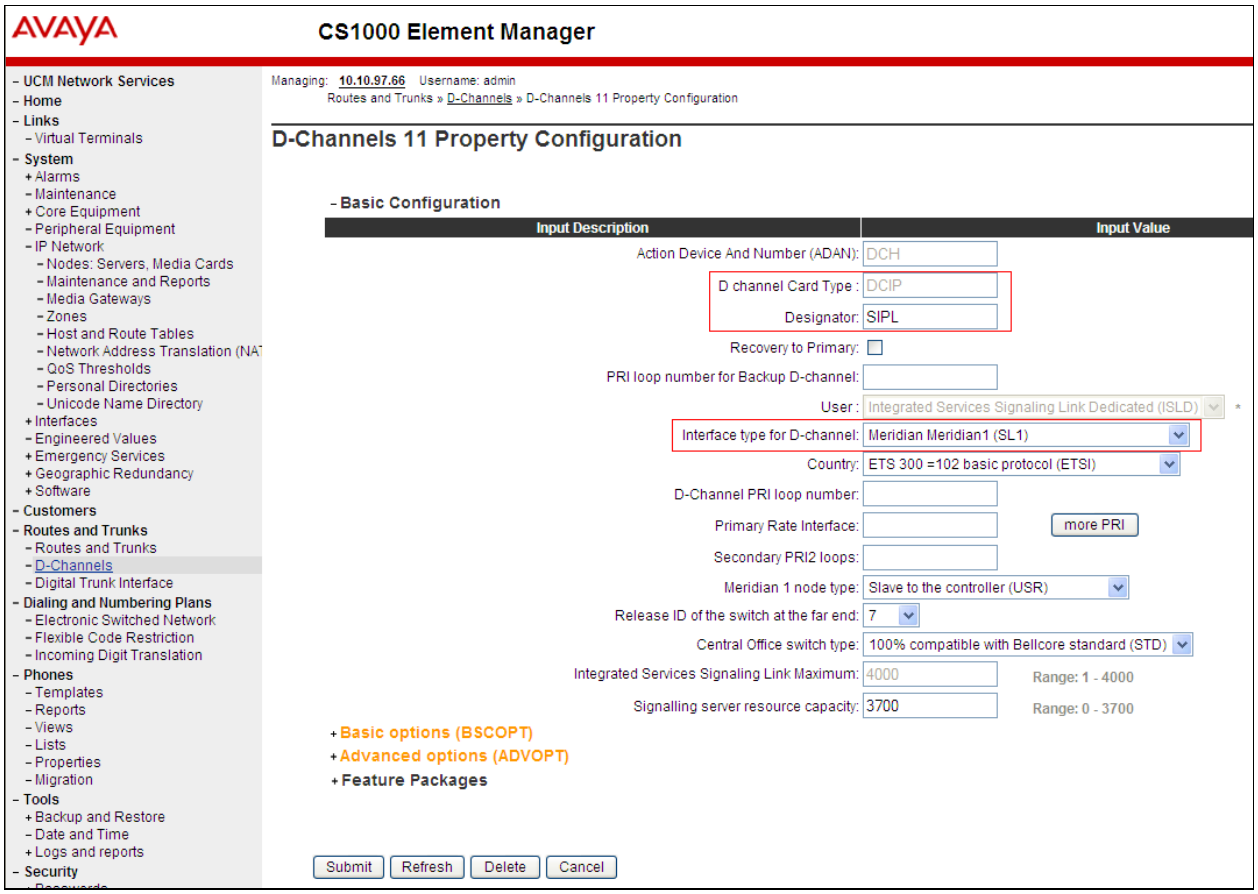
5.5. Create a D-Channel for SIP Line

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



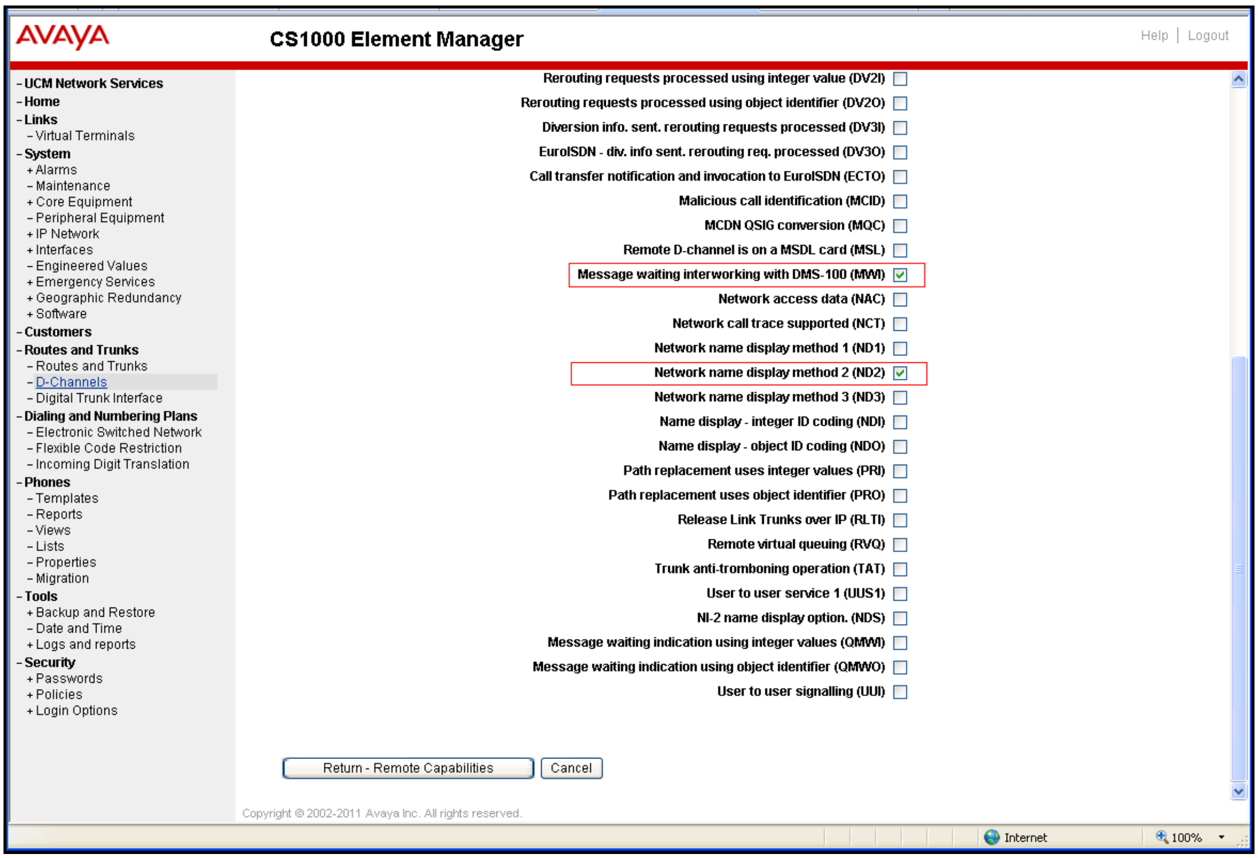
Screen below shows the **D-Channels xx Property Configuration** page for which was configured during compliance testing. Configure the **Basic Configuration** section as follows,

- In the **D channel Card Type** enter **DCIP**.
- Enter a valid name in the **Designator** field.
- From the **Interface type for D-channel** drop down menu, select **Meridian Meridian1 (SL1)**.
- Retain default values for rest of the fields.



Click on the **Basic options (BSCOPT)** link. The **Basic options (BSCOPT)** section expands (not shown). Click on **Edit** to configure **Remote Capabilities (RCAP)** (not shown). The **Remote Capabilities Configuration** page will appear as shown below.

- Select the **Message waiting interworking with DMS-100 (MWI***)* check box. **Message Waiting Interworking with DMS-100 (MWI)** must be enabled to support voice mail notification on SIP Line endpoints.
- Select the **Network name display method 2 (ND2)** check box. **Network Name Display Method 2 (ND2)** must be enabled to support name display between SIP Line endpoints.
- Retain default values for all other fields.
- At the bottom of the **Remote Capabilities Configuration** page, click **Return - Remote Capabilities** to return the **D-Channel xx Property Configuration** page.

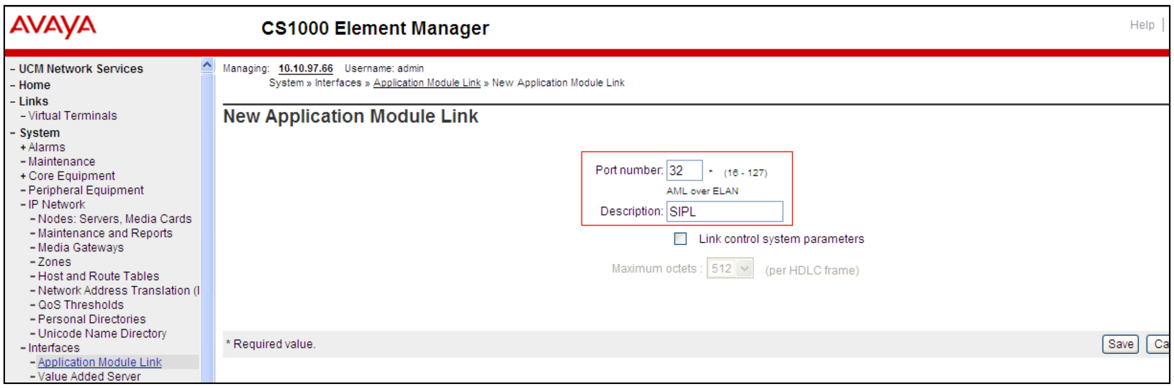


Click on the **Submit** button (not shown) of the D-Channel Property Configuration page to save changes.

5.6. Create an Application Module Link (AML)

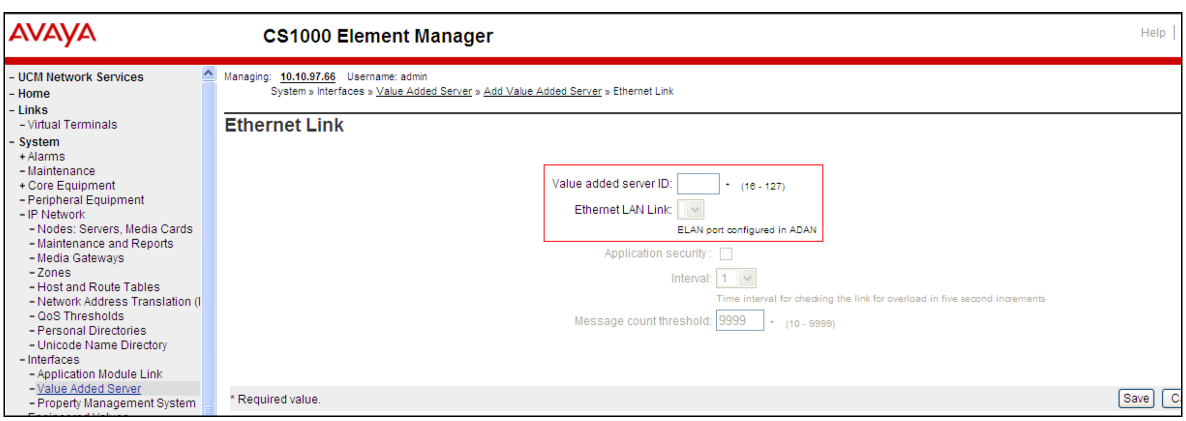
On the EM page, navigate to **System → Interfaces → Application Module Link**, click on the **Add** button to add a new Application Module Link (not shown). The **New Application Module Link** page appears as shown below.

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 **32**. Enter a valid entry in the **Description** field.
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 (not shown).
The **Value Added Server** page appears (not shown), in this page, select the **Ethernet LAN Link** (not shown) option from this page and the **Ethernet Link** page appears as shown below.
Enter a number in the **Value added server ID** field; during compliance testing **32** 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 at default values and click on the **Save** button to complete adding the **VAS** and save the configuration.

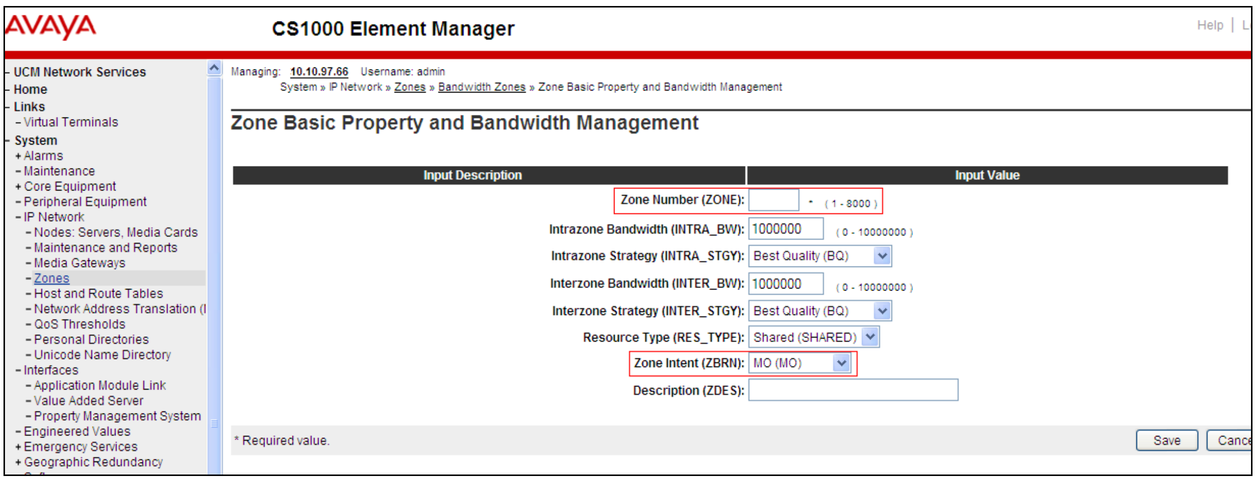


5.8. Create a Virtual Trunk Zone

On the EM page, navigate to menu **System → IP Network → Zones**. The **Zones** page appears on the right (not shown), in this page select **Bandwidth Zones** link.

On the **Bandwidth Zones** page, click on the **Add** button (not shown), 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 at default values and click on the **Save** button to complete adding the Zone.

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



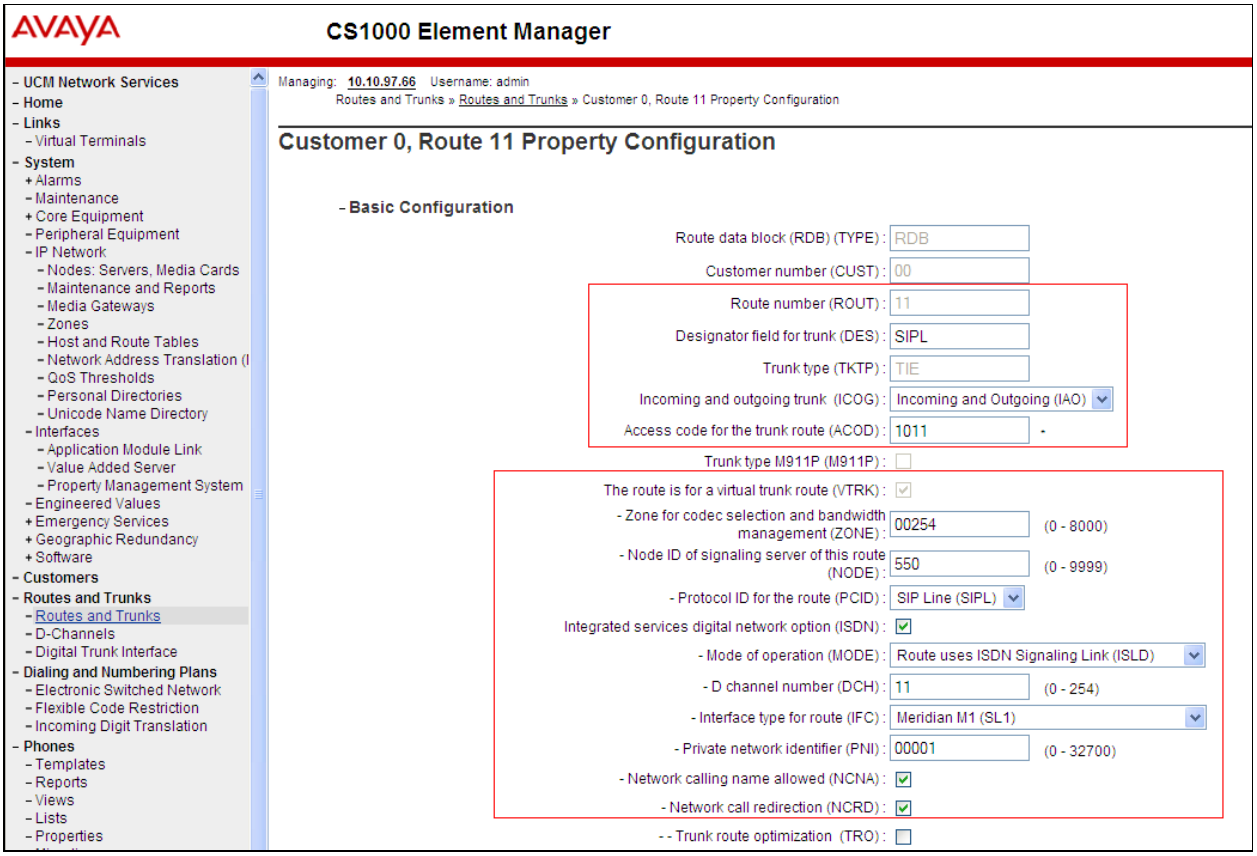
5.9. Create a SIP Line Route Data Block (RDB)

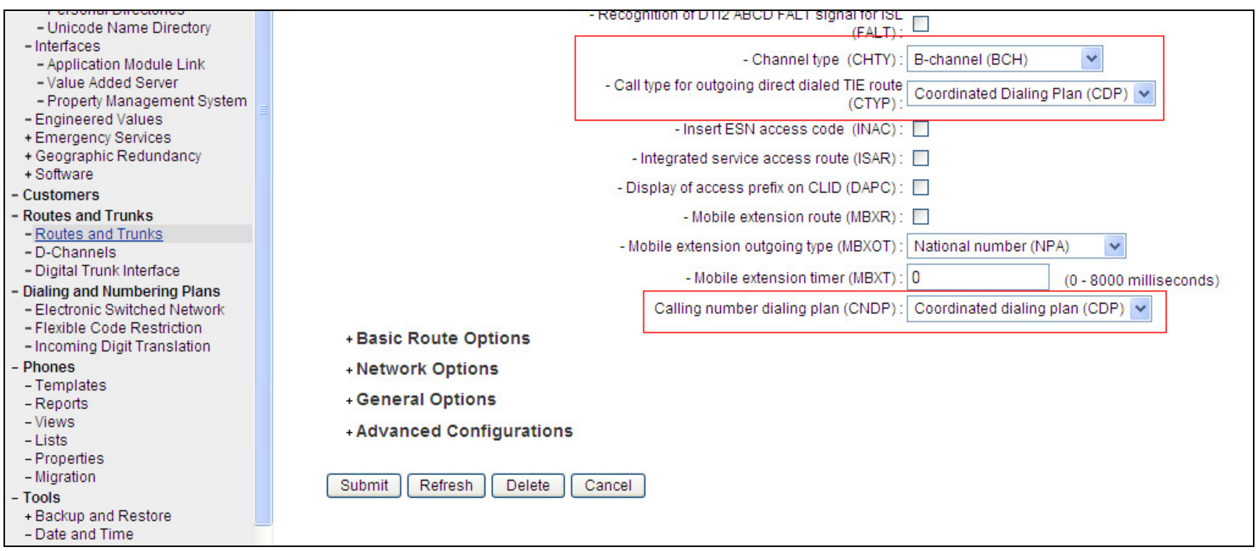
On the EM page, navigate to the menu **Routes and Trunks → Routes and Trunks**; the **Routes and Trunks** page appears (not shown). In this page, click on the **Add route** button (not shown) next to the customer number that the route will belong to.

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)**: 11; this is the value used during compliance testing.
- **Designator field for trunk (DES)**: Enter a descriptive name.
- **Trunk type(TKTP)**: TIE
- **Incoming and Outgoing trunk (ICOG)**: Incoming and Outgoing (IAO)
- **Access Code for Trunk group (ACOD)**: 1011; this is the value used during compliance testing.
- **The route is for a virtual trunk route (VTRK)**: Checked.
- **Zone for codec selection and bandwidth management (ZONE*)***: 254, this is the Virtual trunk zone number that was created in **Section 5.8**.
- **Node ID of signaling server of this route (NODE)**: 550; 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)**: 11; the D-channel number that was created in **Section 5.5**.
- **Interface type for route (IFC)**: Meridian M1 (SL1).
- **Private network identifier (PNI): 0000**1; this is the value used during compliance testing.
- **Network calling name allowed (NCNA)**: Checked.
- **Network call redirection (NCRD)**: Checked
- **Channel type (CHTP)**: B-channel (BCH).
- **Call type for outgoing direct dialed TIE route (CTYP)**: Coordinated Dialing Plan (CDP).
- **Calling Number dialing plan (CNDP)**:
- Coordinated Dialing Plan (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.


AVAYA
CS1000 Element Manager
- UCM Network Services
- Home
- Links
- Virtual Terminals
- System
+ Alarms
- Maintenance
+ Core Equipment
- Peripheral Equipment
- IP Network
- Nodes: Servers, Media Cards
- Maintenance and Reports
- Media Gateways
- Zones
- Host and Route Tables
- Network Address Translation (I
- QoS Thresholds
- Personal Directories
- Unicode Name Directory
- Interfaces
- Application Module Link
- Value Added Server
- Property Management System
- Engineered Values
+ Emergency Services
+ Geographic Redundancy
+ Software
- Customers
- Routes and Trunks
- Routes and Trunks
- D-Channels
- Digital Trunk Interface
- Dialing and Numbering Plans
- Electronic Switched Network
- Flexible Code Restriction
- Incoming Digit Translation
- Phones
- Templates
- Reports
- Views
- Lists
- Properties
Managing: 10.10.97.66 Username: admin
Routes and Trunks » Routes and Trunks » Customer 0, Route 11 Property Configuration
Customer 0, Route 11 Property Configuration
- Basic Configuration
Route data block (RDB) (TYPE) : RDB
Customer number (CUST) : 00
Route number (ROUT) : 11
Designator field for trunk (DES) : SIPL
Trunk type (TKTP) : TIE
Incoming and outgoing trunk (ICOG) : Incoming and Outgoing (IAO)
Access code for the trunk route (ACOD) : 1011
Trunk type M911P (M911P) :
The route is for a virtual trunk route (VTRK) :
- Zone for codec selection and bandwidth management (ZONE) : 00254 (0 - 8000)
- Node ID of signaling server of this route (NODE) : 550 (0 - 9999)
- Protocol ID for the route (PCID) : SIP Line (SIPL)
Integrated services digital network option (ISDN) :
- Mode of operation (MODE) : Route uses ISDN Signaling Link (ISLD)
- D channel number (DCH) : 11 (0 - 254)
- Interface type for route (IFC) : Meridian M1 (SL1)
- Private network identifier (PNI) : 00001 (0 - 32700)
- Network calling name allowed (NCNA) :
- Network call redirection (NCRD) :
- - Trunk route optimization (TRO) :


- Unicode Name Directory
- Interfaces
- Application Module Link
- Value Added Server
- Property Management System
- Engineered Values
+ Emergency Services
+ Geographic Redundancy
+ Software
- Customers
- Routes and Trunks
- Routes and Trunks
- D-Channels
- Digital Trunk Interface
- Dialing and Numbering Plans
- Electronic Switched Network
- Flexible Code Restriction
- Incoming Digit Translation
- Phones
- Templates
- Reports
- Views
- Lists
- Properties
- Migration
- Tools
+ Backup and Restore
- Date and Time
- Recognition of DTI2 ABCD FALT signal for ISL (FALT) :
- Channel type (CHTY) : B-channel (BCH)
- Call type for outgoing direct dialed TIE route (CTYP) : Coordinated Dialing Plan (CDP)
- Insert ESN access code (INAC) :
- Integrated service access route (ISAR) :
- Display of access prefix on CLID (DAPC) :
- Mobile extension route (MBXR) :
- Mobile extension outgoing type (MBXOT) : National number (NPA)
- Mobile extension timer (MBXT) : 0 (0 - 8000 milliseconds)
Calling number dialing plan (CNDP) : Coordinated dialing plan (CDP)
+ Basic Route Options
+ Network Options
+ General Options
+ Advanced Configurations
Submit
Refresh
Delete
Cancel

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.

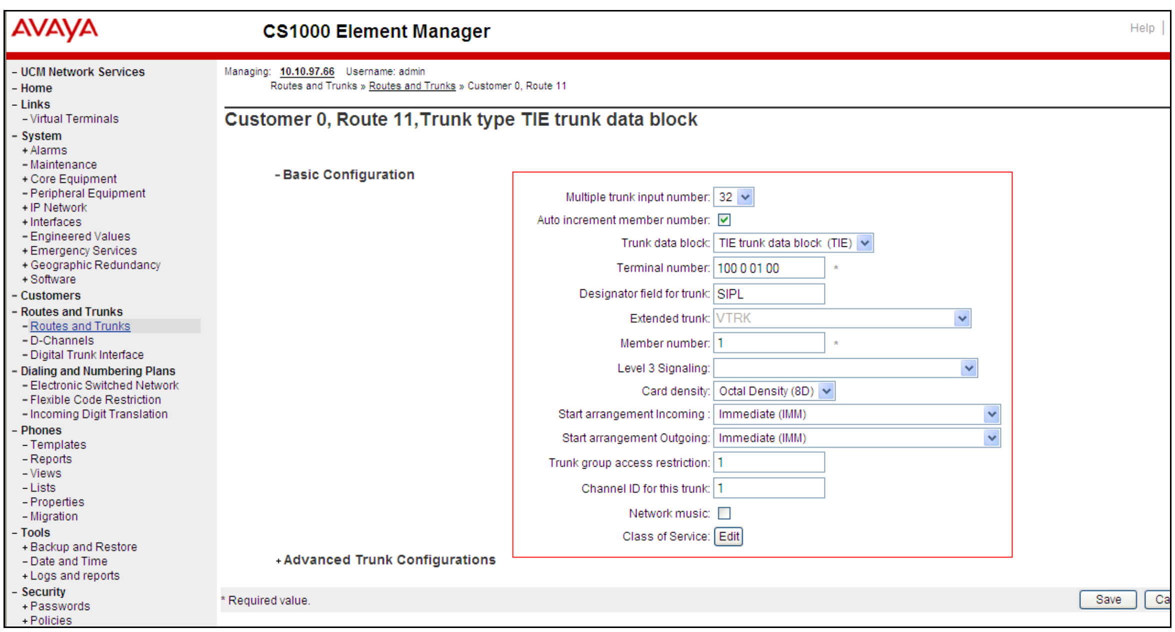
The **Customer ID, Route ID, and Trunk type TIE trunk data block** page appears as shown below, enter values for fields as shown below:

- **Multiple trunk input number (MTINPUT)**: 32; create 32 trunks.
- **Auto increment member number**: Checked.
- **Trunk data block:** TIE trunk data block (TIE).
- **Terminal Number (TN)**: Enter an available range. 100 0 01 00 was used during compliance testing.
- **Designator field for trunk**: Enter a descriptive name.
- **Extended trunk**: VTRK.
- **Member number**: 1; 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**: 1.
- **Channel ID for this trunk**: 1; this ID should be the same with the ID of Member Number.

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

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

Retain default values for all other fields 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.

Screen below shows a print out of the already configured SIP phone. 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.

```
TYPE TNB
TN   100 1 11 06  → Terminal number on which the set is configured.
DATE
PAGE
DES
DES  POLY   → Description of the phone.
TN   100 1 11 06  VIRTUAL
TYPE UEXT   → Universal Extension type is used for SIP phone.
.
.
UXTY SIPL   → Universal Extension type is SIP Line type.
MCCL YES
SIPN 0
SIP3 1      → Value needs to be 1 for 3rd party SIP phone.
FMCL 0
TLSV 0
SIPU 58117  → SIP phone user ID.
NDID 550    → SIP Line node ID.
.
.
NHTN
CFG_ZONE 00001    → SIP Line zone configured on.
CUR_ZONE 00001
MRT
.
.
VSIT NO
FDN  58888  → Forward DN.
TGAR 1
LDN  NO
NCOS 7      → Network Class of Service. Seven is the highest value.
.
.
XLST
SCPW 1234   → SIP phone user password.
SFLT NO
CAC_MFC 0
CLS  UNR FBA WTA LPR MTD FNA HTA TDD HFD CRPD
     MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
     POD SLKD CCSD SWA LND CNDD
     CFTD SFA MRD DDV CNID CDCA MSID DAPA BFED RCBD
     …

CPND_LANG ENG
```

```
RCO  0
HUNT 58000  → Hunt DN
LHK  0
.
.
DNDR 0
KEY  00 SCR 58117 0     MARP  → Extension number for the SIP phone
        CPND
          CPND_LANG ROMAN
            NAME Polycom, SS  → CLID information
            XPLN 13
            DISPLAY_FMT FIRST,LAST
     01 HOT U 2758117 MARP 0
     02
     03
```

6. Configure Polycom SoundStructure VOIP Interface Card

This section describes how to set up and configure the SoundStructure VoIP interface card using SoundStructure Studio application. The VoIP interface card is inserted into the SoundStructure C16 box. A speaker, microphone and an Ethernet connection is hooked up to the SoundStructure C16 box.

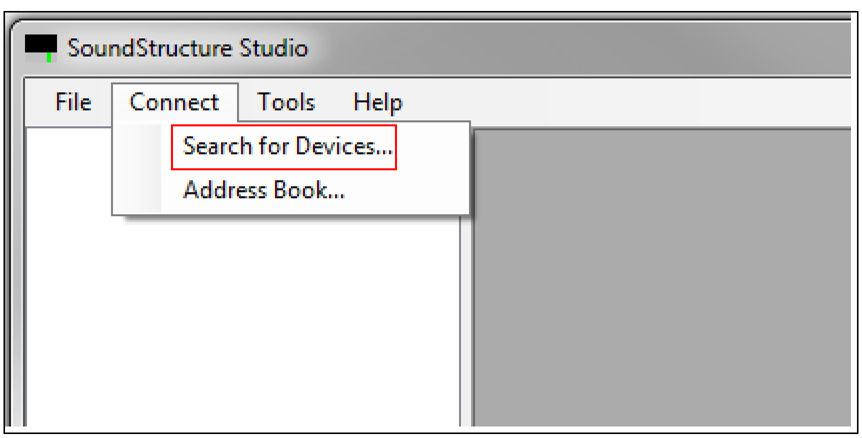
6.1. Installing the SoundStructure Studio Application

SoundStructure Studio Software 1.8.0 must be downloaded from the SoundStructure support site at the following link. (If a later version of Studio software and companion version of SoundStructure Firmware is available it is recommended for use and will be compatible with the VoIP Interface Card.)
http://support.polycom.com/PolycomService/support/us/support/voice/soundstructure/c_series.html
After download, install on the PC that will be used to manage the SoundStructure VoIP card and perform testing and configuration changes. Ensure that the PC running the SoundStructure Studio application is on the same subnet as the SoundStructure VoIP interface card.

6.2. Discovering and Connecting to SoundStructure System on Network

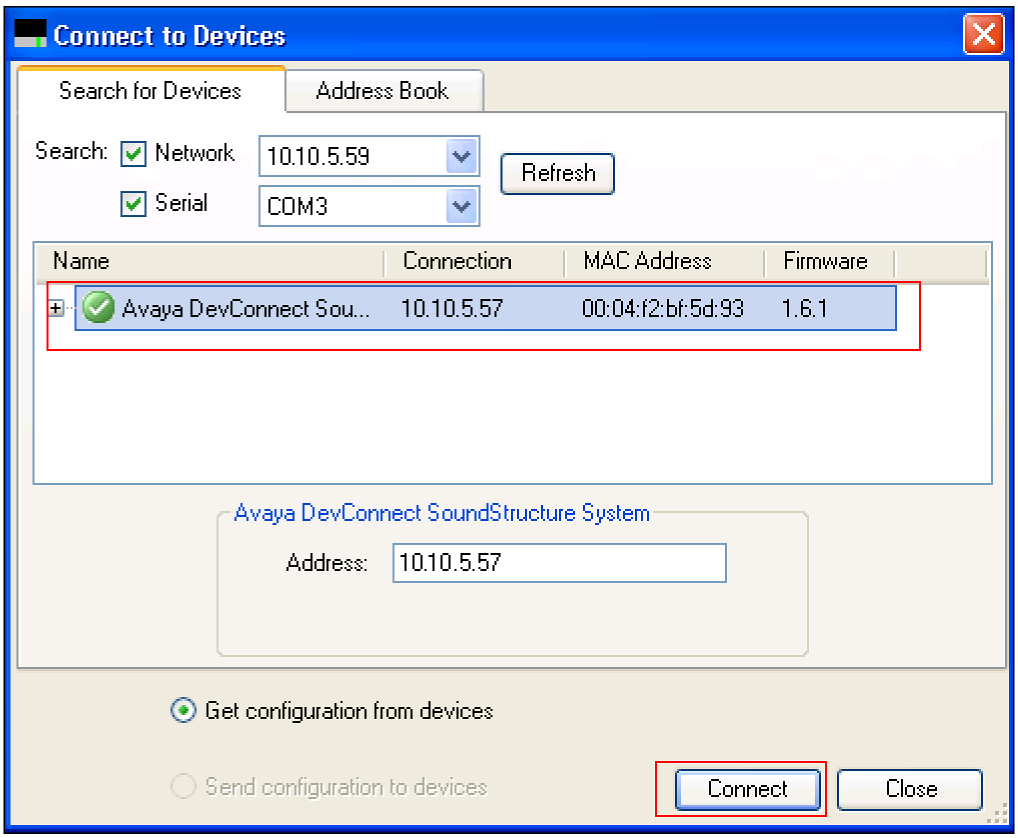
This section explains the steps to discover and connect to the SoundStructure system on the network. Launch the SoundStructure Studio application from the PC it was installed on.

Navigate to **Connect → Search for Devices** as shown in the screen below.

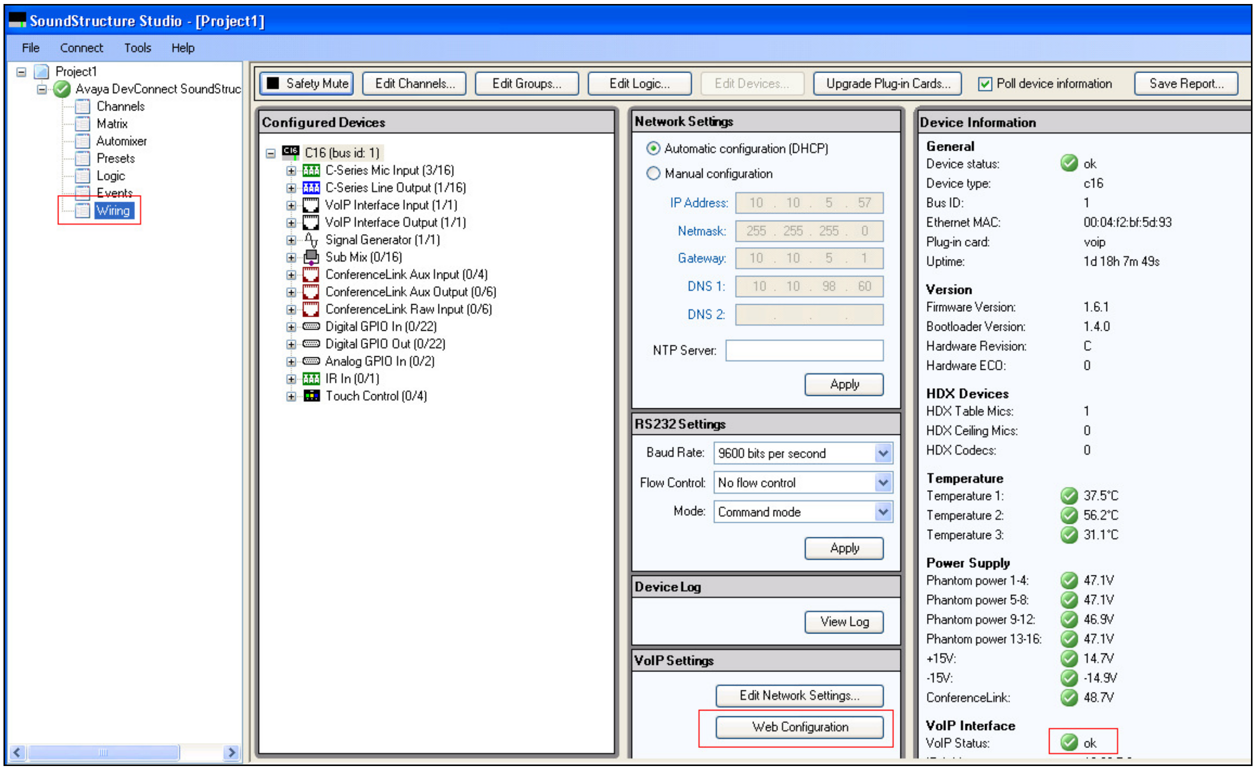


After the System has been discovered it will be displayed in the **Search for Devices** tab as shown in the screen below. Select the item, and click on the **Connect** button. Wait briefly for the **Receiving File** step to complete (not shown).

In case the system does not show up after search, ensure that the PC searching for the device is on the same subnet (not running on Wi-Fi, etc). If for some reason the SoundStructure is not showing up after repeated attempts, try to discover the SoundStructure after connecting it directly via Ethernet cable to the PC (bypassing the network). If problems persist, try a factory reset (instructions for factory reset can be found in SoundStructure Design Guide and in Firmware Release Notes).



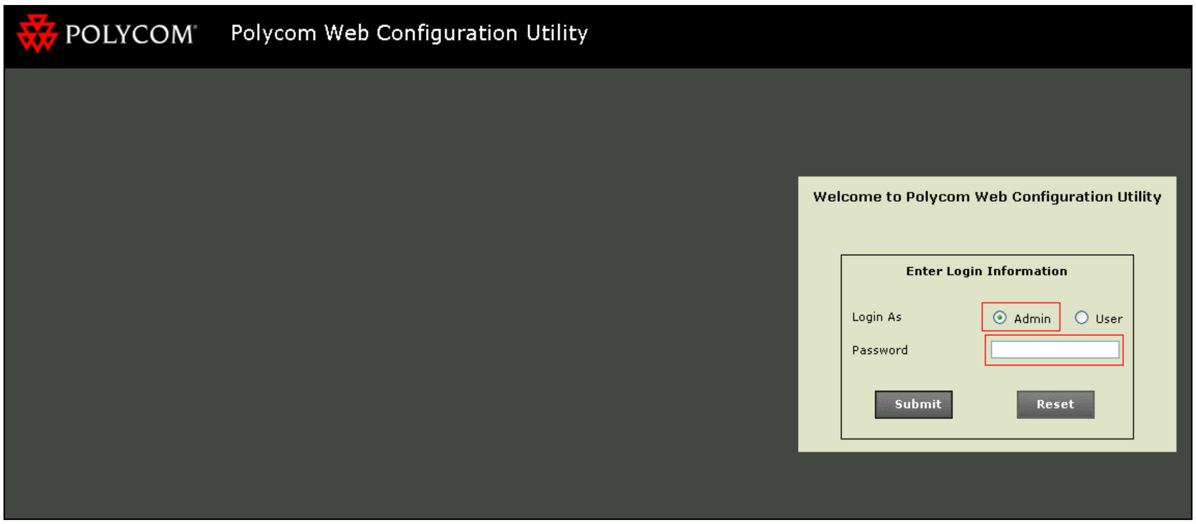
Click on the **Wiring** link to view wiring page with VoIP Card Status and links to the phone Web Configuration interface. Ensure that the **VoIP Status** under the **VoIP Interface** section is **OK**. Click on the **Web Configuration** button to launch the **Polycom Web Configuration Utility** as shown in **Section 6.3**.



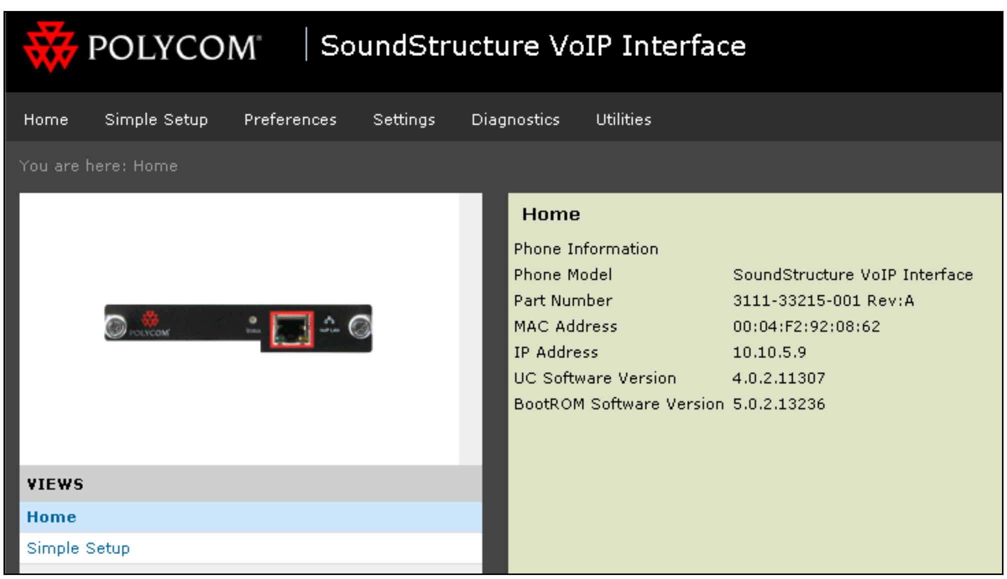
6.3. Polycom Web Configuration Utility

This section shows how to log in to the home page of Polycom Web Configuration Utility that is required to configure the SoundStructure VoIP interface card.

Click the **Web Configuration** button in the SoundStructure Studio as shown in **Section 6.2**. Alternatively, type the IP address displayed in VoIP Interface section of SoundStructure Studio into the URL address bar of a web browser. The web configuration utility login interface will be displayed as shown below. Select the **Admin** radio button and type in the default password of **456**.



Click **Submit**, the homepage of the Polycom SoundStructure VoIP Interface is seen as shown below.



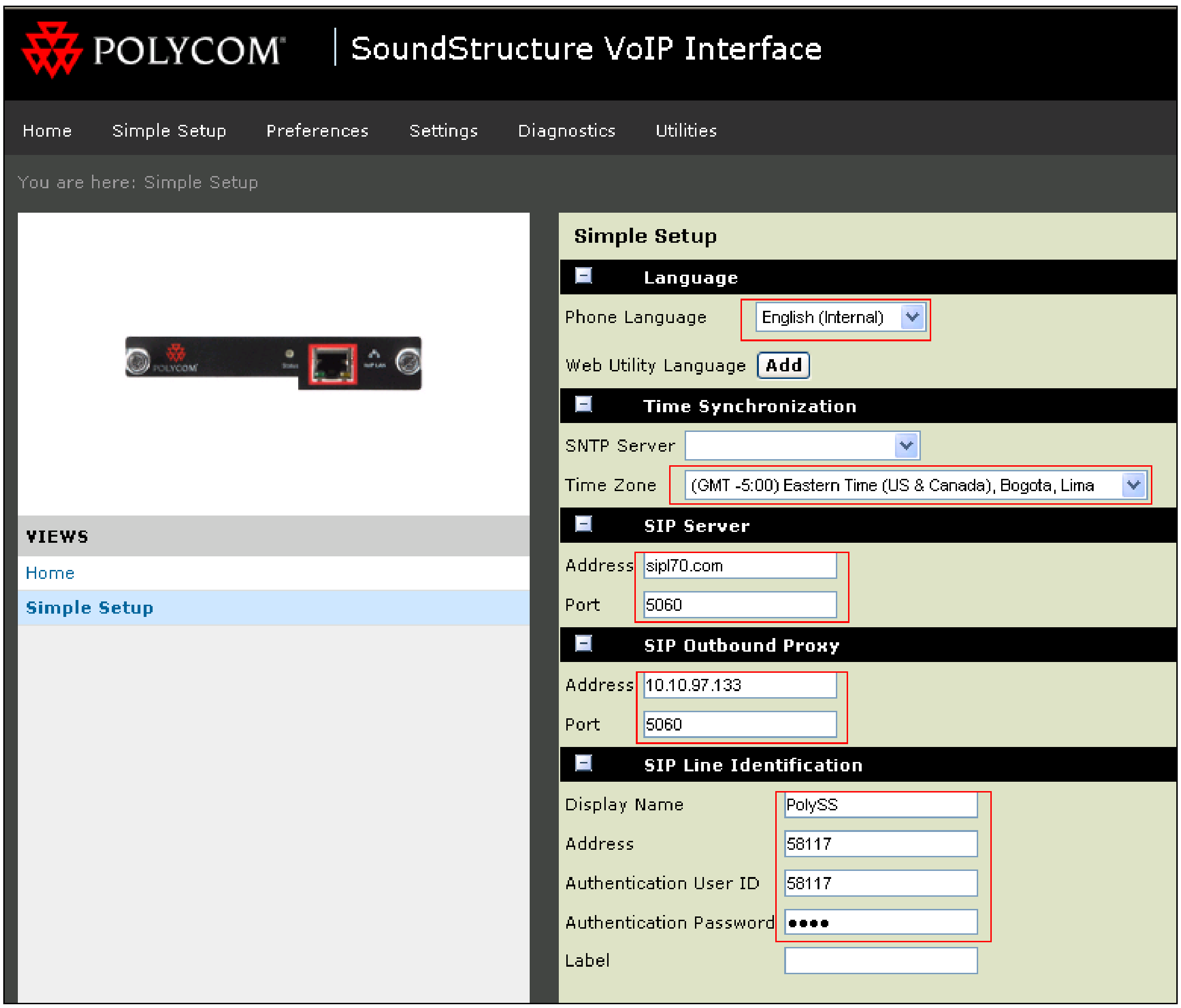
6.4. Configure the Lines for Polycom SoundStructure

This section shows how to configure the Polycom SoundStructure to register with CS1000 SIP Line Gateway.

On the homepage of configuration screen, click on the **Simple Setup** menu, the **Simple Setup** page appears as shown below. Enter the following values,

- **Phone Language**: English (internal)
- **Time Zone**: Select an appropriate one for the region.
- Under **SIP Server** section, **Address**: sipl70.com and **Port**: 5060; configured in **Section 5.4**.
- Under **SIP Outbound Proxy** section, **Address**: 10.10.97.133 and **Port**: 5060; configured in **Section 5.4**.
- Under the **SIP Line Identification** section, **Display Name**: an appropriate name, **Address**: 58117, **Authentication User ID**: 58117 and **Authentication Password**: 1234; configured in **Section 5.11**.

Click on **Save** (not shown).



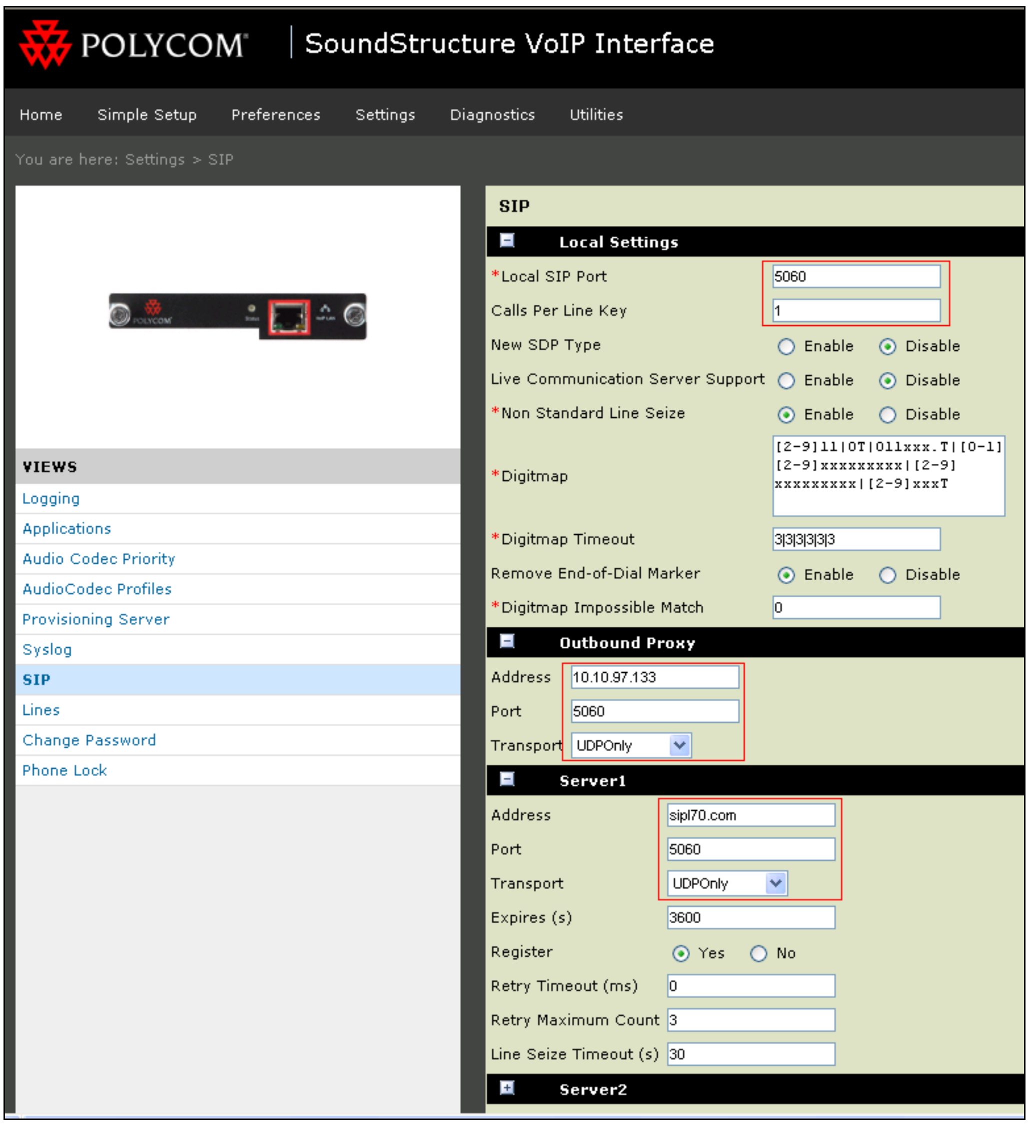
6.5. SIP Settings

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

On the homepage of Polycom SoundStructure, navigate to menu **Settings → SIP** (not shown), **SIP** screen is shown below. Enter the following values and retain rest at default.

- Under the **Local Settings** section, **Local SIP Port**: 5060; configured in **Section 5.4**. **Calls Per Line Key**: 1
- Under the **Outbound Proxy** section, **Address**: 10.10.97.133 and **Port**: 5060; configured in **Section 5.4.Transport**: UDPOnly.
- Under the **Server1** section, **Address**: sipl70.com and **Port**: 5060; configured in **Section 5.4.Transport**: UDPOnly.

Click on **Save** (not shown).



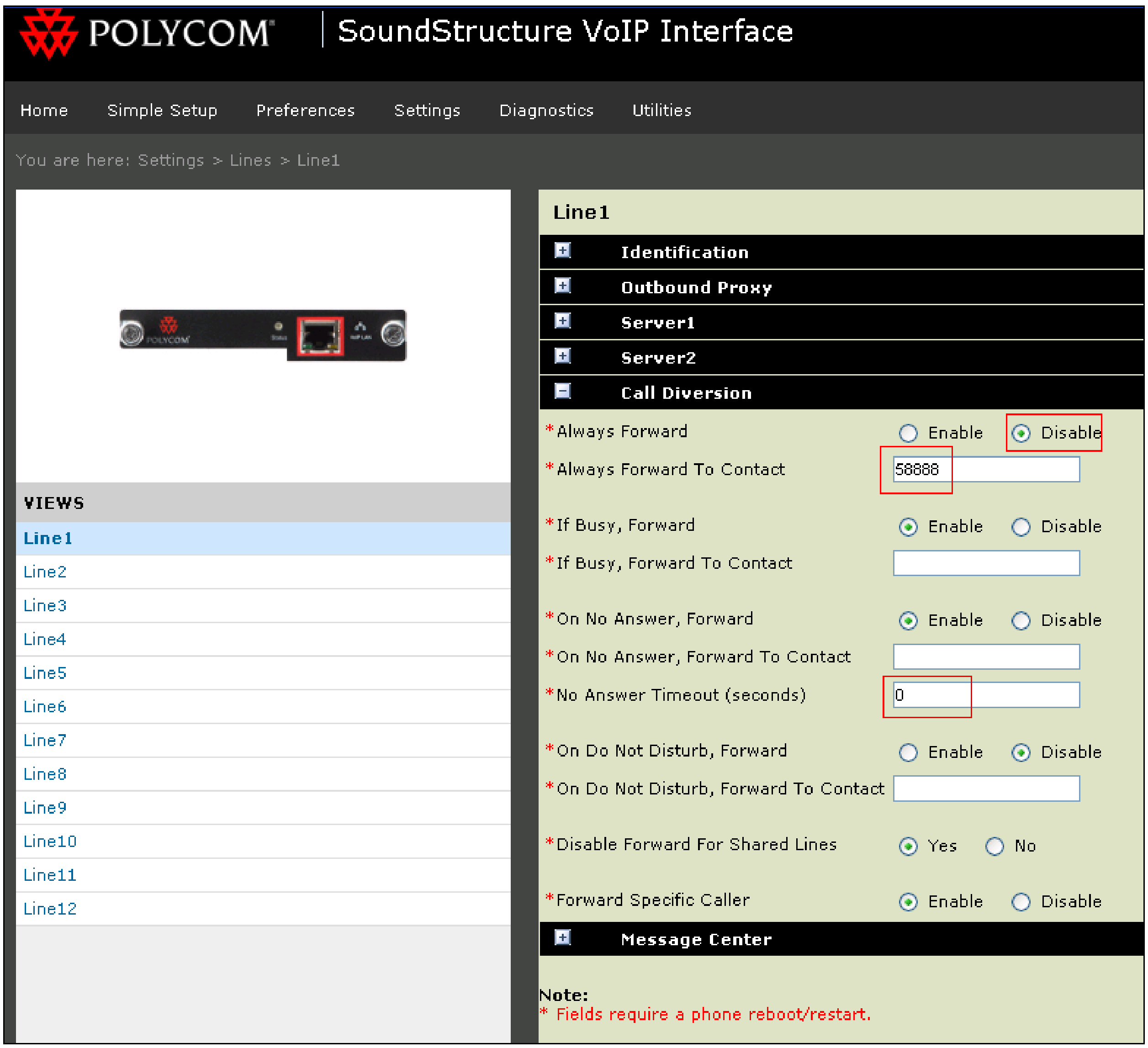
6.6. Local Call Forward Settings

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

On the homepage of Polycom SoundStructure, navigate to menu **Settings → Lines** (not shown). **Line1** screen is shown below. Enter the following values and retain rest at default.

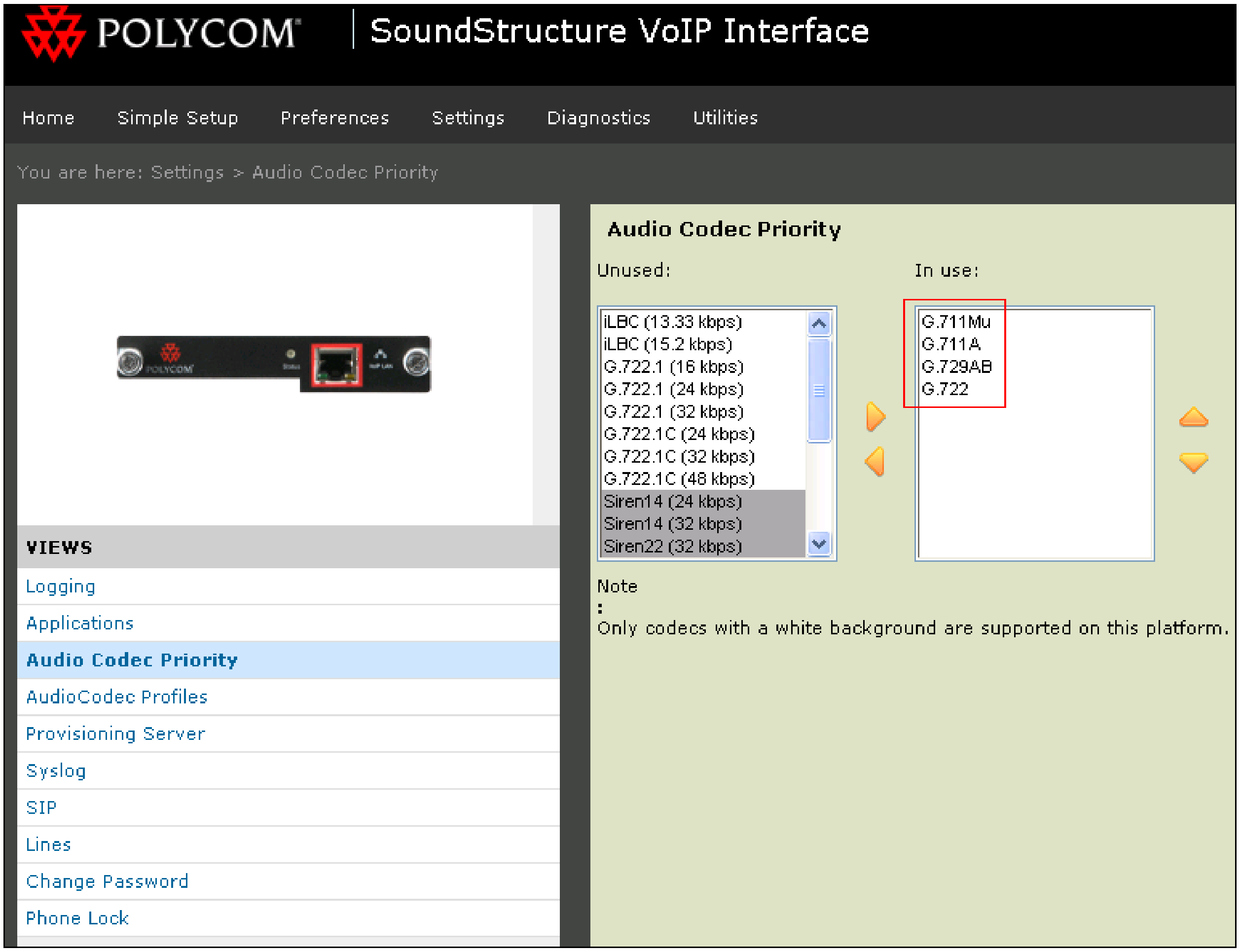
- Under the **Call Diversion** section, **Always Forward**: Disable and configure an appropriate Directory Number (DN) for the **Always Forward To Contact** field.
- **No Answer Timeout (seconds)**: 0

Click on **Save** (not shown).



6.7. Codec Settings

On the homepage of Polycom SoundStructure, navigate to menu **Settings → Audio Codec Priority** (not shown). Select the codec list as shown below. Click **Save** (not shown).



7. Verification Steps

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

Step1

Verify that the Poycom SoundStructure VoIP interface card 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 a valid account.
Issue command **slgSetShowByUID [userID]** where userID is SIP Line user's ID being checked.

```
[admin@ucm1 ~]$ slgSetShowByUID 58117

=== VTRK ===
UserID          AuthId     TN              Clients  Calls  SetHandle  Pos
ID
SIPL Type
---------------  ----------  ---------------  -------  -----  ---------  ----
---
---------
          58117       58117     100-01-11-06        1      0  0xab501ae0
SIP Lines
        StatusFlags = Registered Controlled KeyMapDwld SSD
        FeatureMask =
        CallProcStatus = 0

        Current Client = 0, Total Clients = 1

         == Client 0 ==
         IPv4:Port:Trans = 10.10.5.9:5060:udp
         Type           = SIP3
         UserAgent      = PolycomSoundStructure-SSTR_VOIP-UA/4.0.2.11307
         x-nt-guid      = 55efcea238c45260cb1db8c837b6a273
         RegDescrip     =
         RegStatus      = 1
         PbxReason      = OK
         SipCode        = 200
         hTransc        = (nil)
         Expire         = 3600
         Nonce          = 7b633ac65572f8e5030dad273995f043
         NonceCount     = 2
         hTimer         = 0xabb54108
         TimeRemain     = 3412
         Stale          = 0
         Outbound       = 0
         ClientGUID     = 0
         MSec CLS       = MSNV (MSEC-Never)
         Contact        = sip:58117@10.10.5.9:5060
         KeyNum         = 255
         AutoAnswer     = NO
```

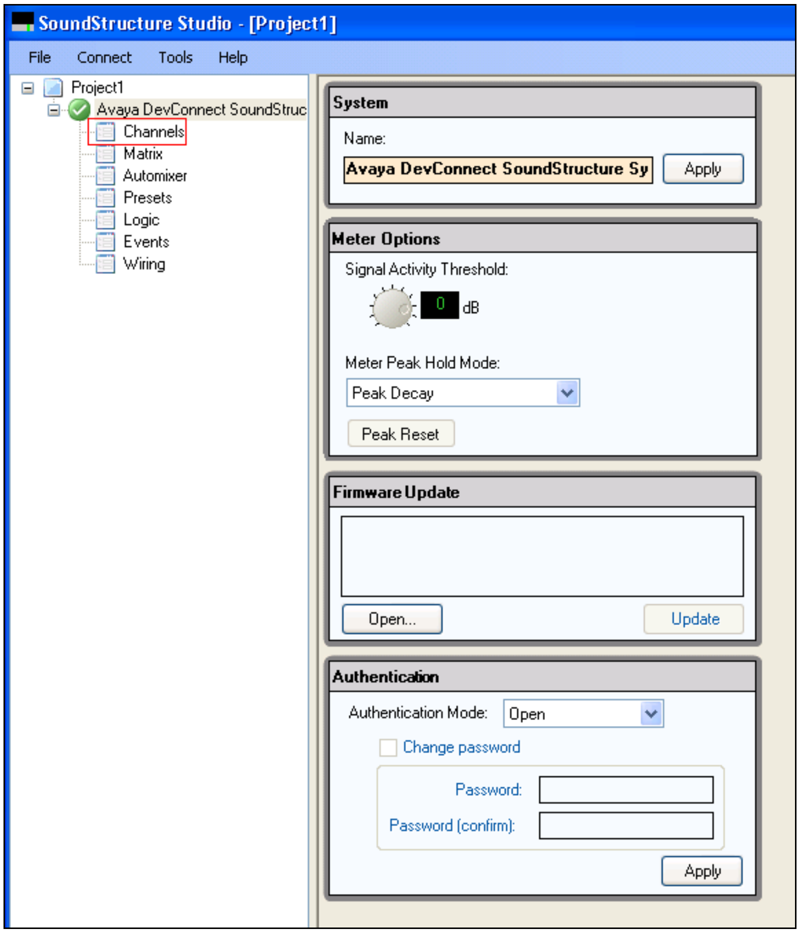
```
Key  Func  Lamp  Label
0    3     0     58117
1    126   0     2758117
5    9     0
17   16    0
18   18    0
19   27    0
20   19    0
21   52    0
22   25    0
24   11    0
25   30    0
26   31    0


 == Subscription Info ==
 Subscription Event  = None
 Subscription Handle = (nil)
 SubscribeFlag = 0
```

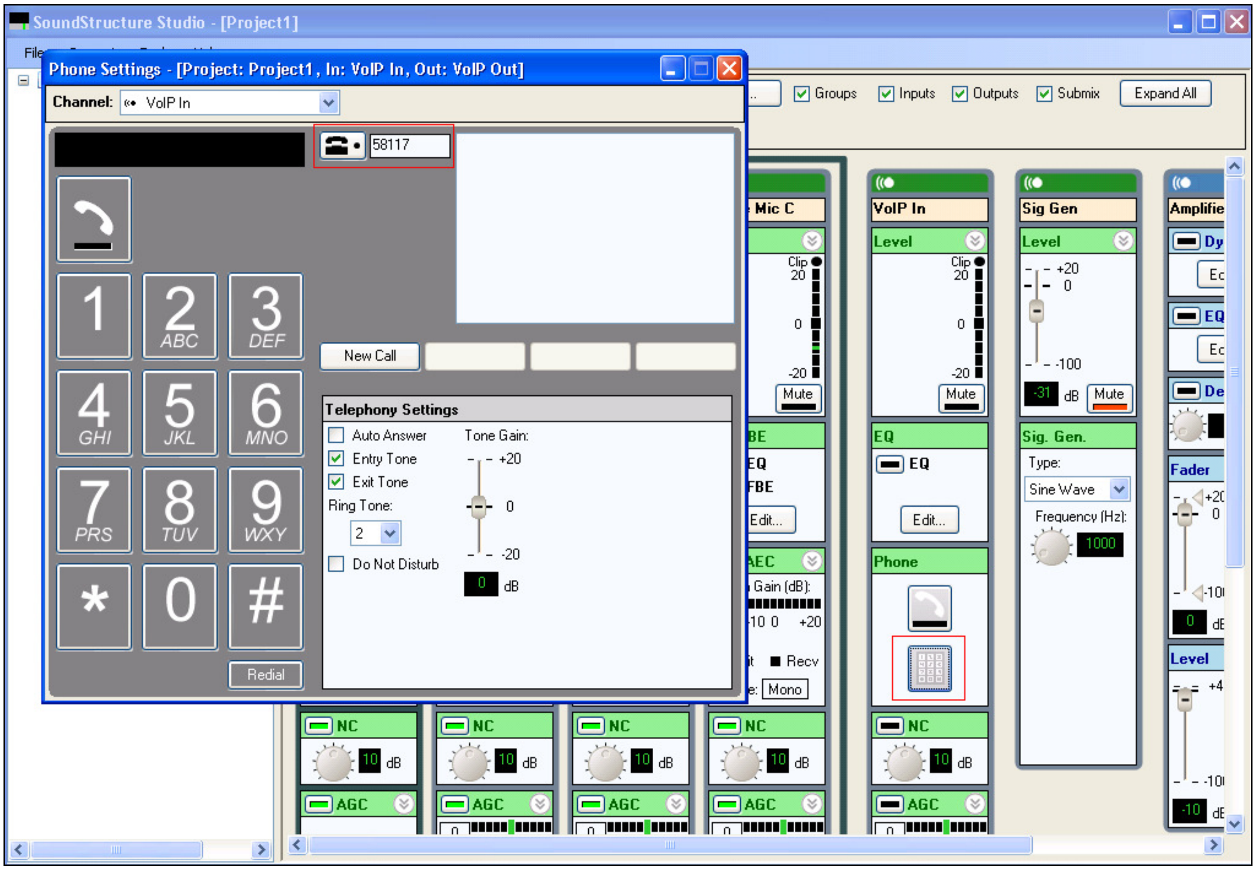
Note: If a set is not registered, no data is returned for the command slgSetShowByUID

Step 2

Click on **Channels** link in the SoundStructure Studio application as shown in the screen below.



The **Channels** page is shown below. Open the dialer by clicking on the dialpad image seen under the **Phone** section. A solid black telephone image is seen with the DN when the SoundStructure VoIP interface card is registered to the CS 1000 SIP Line Gateway.



Place a call from and to the SoundStructure using the above shown dialer 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 to make sure that all SIP request/response messages are correct.

8. Conclusion

These Application Notes illustrate the procedures necessary for configuring the Polycom SoundStructure to interoperate with the Avaya Communication Server 1000. All feature functionality test cases described in **Section 2.1** were passed along with the observations noted in **Section 2.2**.

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 SoundStructure products may be found at:
http://support.polycom.com

[1] *Communication Server 1000E Installation and Commissioning*, April 2012, Release 7.5, NN46041-310
[2] *SIP Line Fundamentals - Avaya Communication Server 1000,* January 2013, Release 7.5, NN43001-508.
[3] *Element Manager System Reference – Administration - Avaya Communication Server 1000,* August 2012, Release 7.5, NN43001-632.
[4] *Co-resident Call Server and Signaling Server Fundamentals - Avaya Communication Sever 1000,* August 2012, Release 7.5, NN43001-509.
[5] *Unified Communications Management Common Services Fundamentals - Avaya Communication Server 1000*, February 2013, Release 7.5, NN43001-116.
[6] *ISDN Primary Rate Interface Installation and Commissioning - Avaya Communication Server 1000,* March 2011, Release 7.5, NN43001-318.

[7] Polycom SoundStructure Documents:
Polycom® SoundStructure Design Guide
http://support.polycom.com/PolycomService/support/us/support/voice/soundstructure/c_series.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.