Avaya Solution & Interoperability Test Lab

Application notes for Polycom VVX 1500 SIP Telephones with Avaya Communication Server 1000 Release 6.0 – Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Communication Server 1000 SIP Line Release 6.0 and Polycom VVX 1500 SIP telephones. During the compliance testing, the Polycom VVX 1500 was able to register as a SIP client endpoint with the Communication Server 1000. The Polycom VVX 1500 SIP telephones were able to place and receive calls from Communication Server 1000 Release 6.0 non-SIP and SIP Line clients. The compliance tests focused on basic telephone features.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These application notes provide detail configurations of Avaya Communication Server 1000 SIP Line release 6.0 (hereafter referred to as CS1000) and the Polycom VVX 1500 SIP telephone release 3.3.1.0769 used during the compliance testing. The Polycom VVX 1500 was tested with the non-SIP and SIP clients of the CS1000 SIP line release 6.0. All the applicable telephony feature test cases of release 6.0 SIP line were executed on the Polycom VVX 1500 , where applicable, to ensure the interoperability with CS 1000.

1.1. Interoperability Compliance Testing

The focus of this testing was to verify that the Polycom VVX 1500 SIP telephone was able to interoperate with the CS1000 SIP line system. The following areas were tested:

- The Polycom VVX 1500 must be able to be installed in the same local VLAN network as the CS1000 successfully.
- Registration of the Polycom VVX 1500 SIP telephone to the CS1000 SIP Line Gateway.
- Calls establishment of Polycom VVX 1500 with CS1000 SIP and non-SIP telephones.
- Telephony features: Basic calls, conference, DTMF (dual tone multi frequency) 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.
- Codec negotiation – G.711 and G.729.
- Video Codec negotiation – H.623 and H.624.

1.2. Support

For technical support on Polycom VVX 1500 SIP endpoints, please contact Polycom, Inc technical support at website www.polycom.com or telephone: 1-888-248-4143

2. Reference Configuration

Figure 1 illustrates the test configuration used during the compliance testing between the Avaya CS1000 and the Polycom VVX 1500.

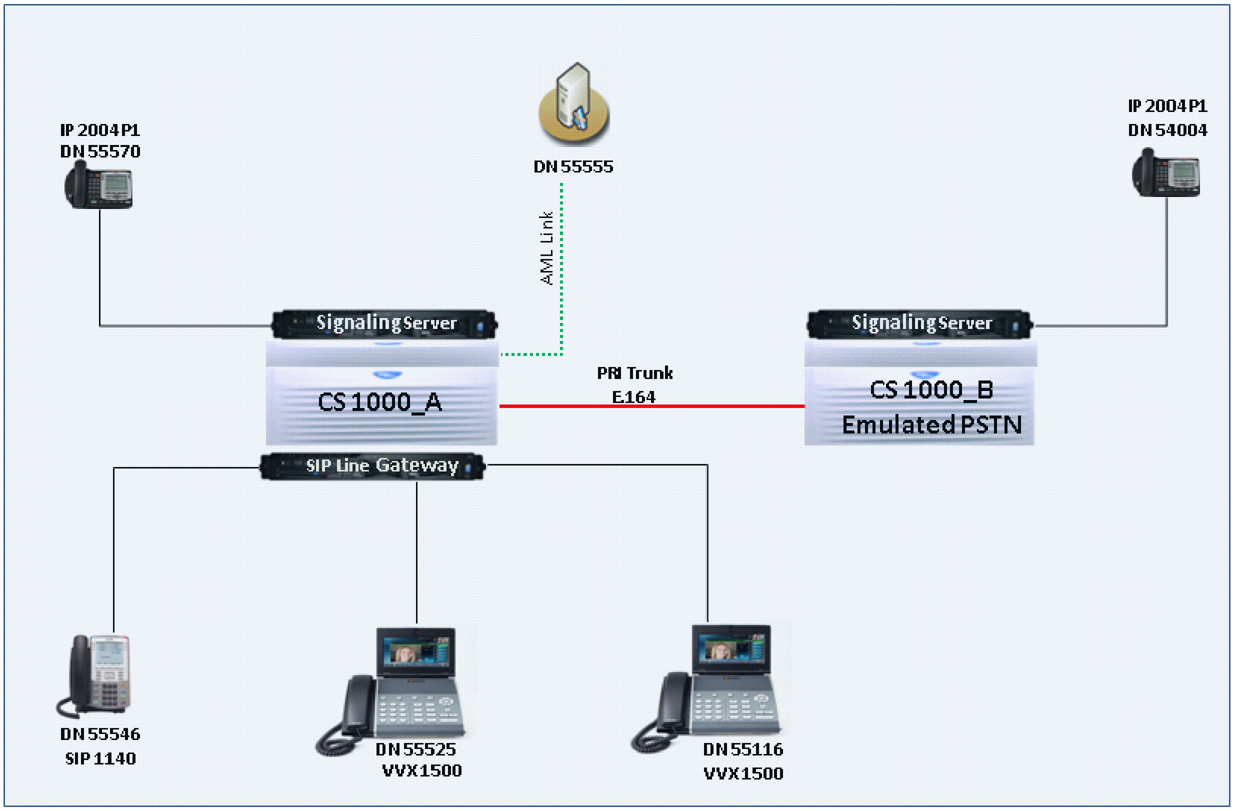

Figure 1 - Network Configuration Diagram

3. Equipment and Software Validated

System	Software Version
Avaya CS1000	• Call Server (CPPM): 6.00RJ • Signaling Server (CPPM): 6.00.18 • SIP Line Gateway (HP DL320)
Avaya voicemail system	• CallPilot 5.0 system
Avaya 1140 SIP client Avaya SIP soft-phones Avaya IP phones	• 02.02.16.00 • 2050PC: 3.02.0045
Polycom VVX 1500 SIP Software	• 3.3.1.0769
Polycom VVX 1500 BootROM	• 4.3.0.0460

4. 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 CS 1000 system. For detailed information, see section 9 [1].

4.1. Prerequisite

- A CS1000 server which has been:
    - Installed with CS 1000 Release 6.0 Linux Base.
    - Joined CS 1000 Release 6.0 Security Domain.
    - Deployed with SIP Line Application.

    For more information, see section 9 [6].

- Following packages are enabled in the keycode. 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 Number	Package Description	Package Type (New or Existing or Dependency)	Applicable Market
SIP_LINES	417	SIP Line Service package	New package	Global
FFC	139	Flexible Feature Codes	Existing package	Global
SIPL_NORTEL	415	Nortel SIP Line package	Existing package	--
SIPL_3RDPARTY	416	Third-Party SIP Line Package	Existing package	--

4.2. Login to Unified Communications Management (UCM) and Element Manager (EM)

- Using internet browser, 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.
- Login with the username/password which was defined during the primary security server configuration. For more information, see section 9 [8].

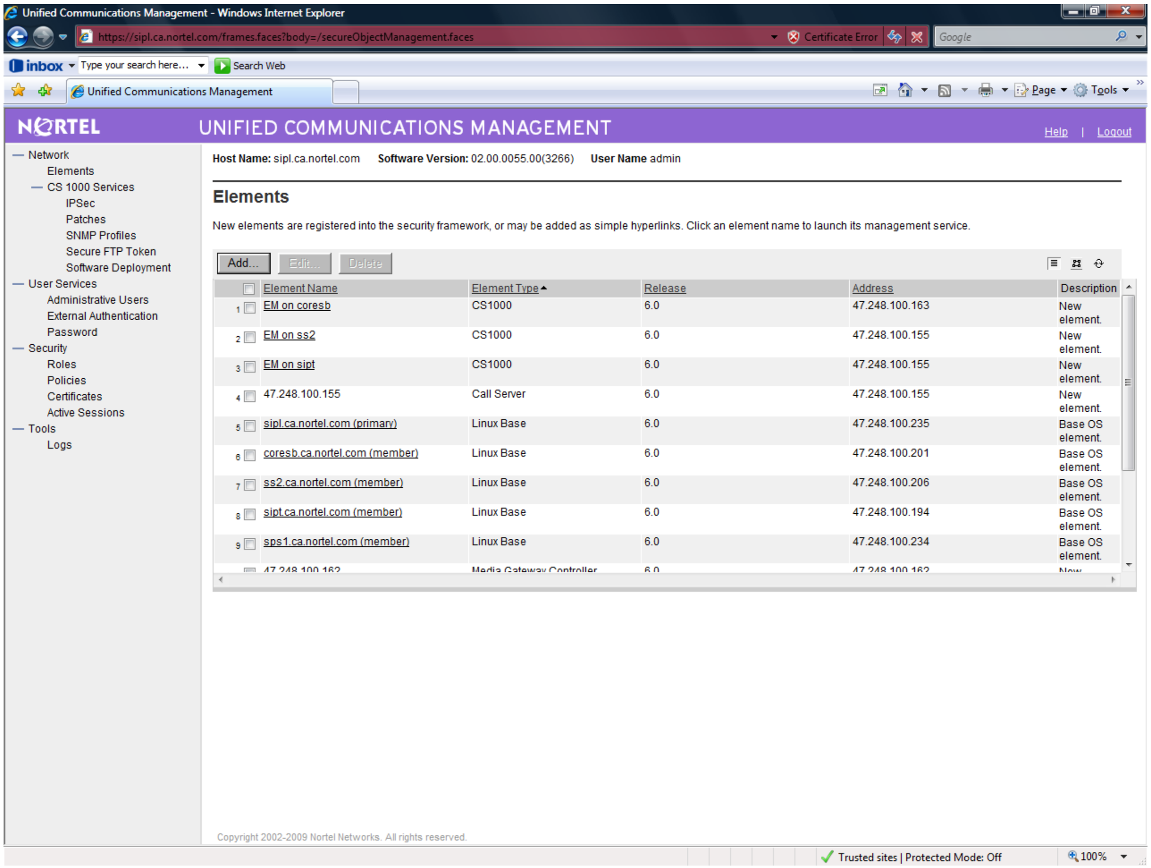

Figure 2 - UCM Home Page

- On the Unified Communications Management page, under the Element Name column, click on the server name to navigate to Element Manager for that server. The CS 1000 Element Manager page appears as show in **Figure 3** below.

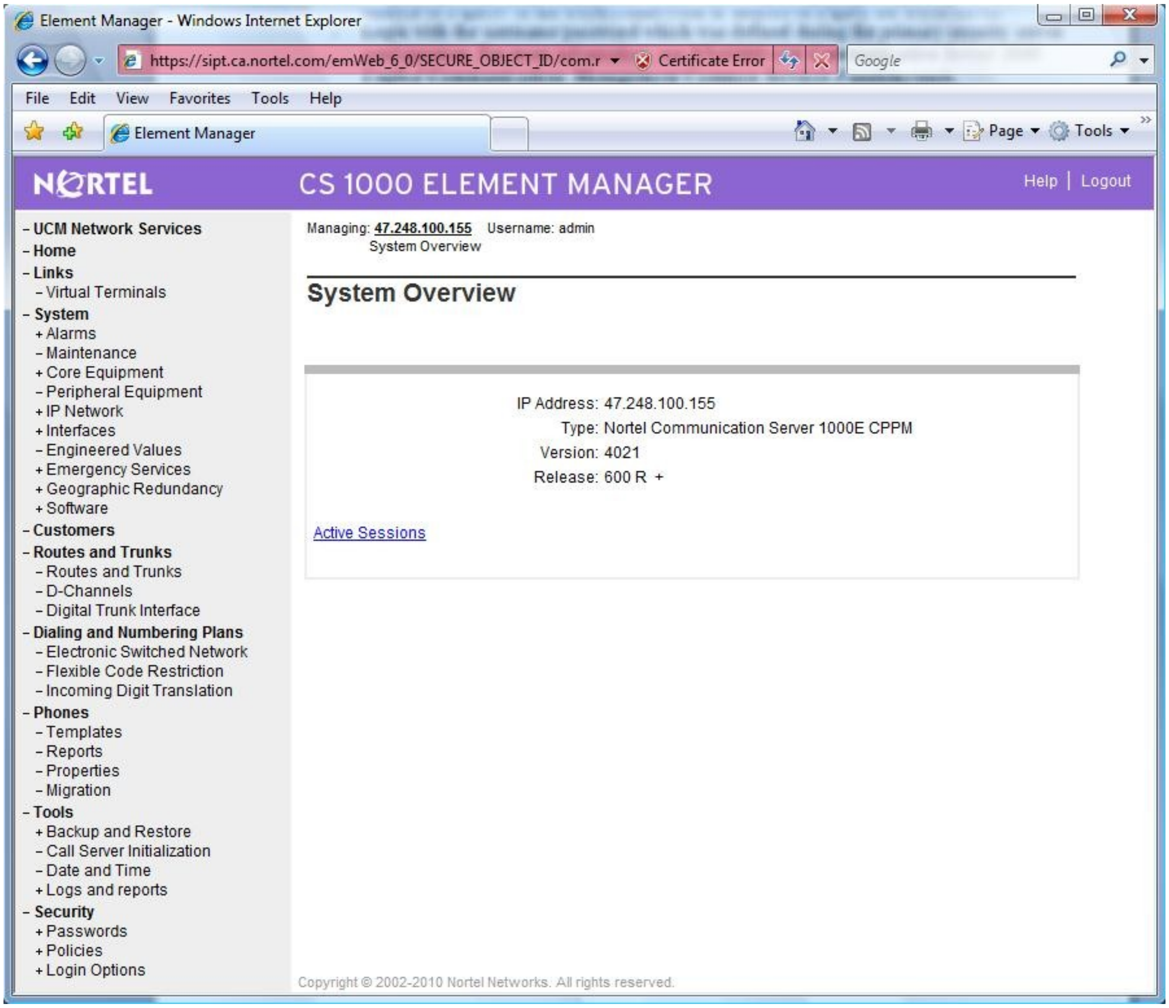

Figure 3 - CS 1000 EM Home Page

4.3. Enable SIP Line Service and Configure the Root Domain in Customer Data Block (CDB)

- 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 SIP Line **Root Domain** name in the ***Root Domain*** text box as shown in **Figure 4**.

Figure 4 - SIP Line Service in Customers Data Block

4.4. SIP Line Telephony Node Configuration

- On the EM page, navigate to ***System → IP Network → Nodes: Servers, Media Cards.***
- Click ***Add*** to add a new SIP Line Node to IP Telephony Nodes. To see the SIP Line node details, click on the SIP Line Node ID (not shown).
- Enter the following as show in **Figure 5**:
    - Enter Node ID in the ***Node ID*** text box.
    - Enter Call Server IP Address in the ***Call Server IP Address*** text box.
    - Enter Node IP Address in the ***Node IP Address*** text box.
    - Enter TLAN Subnet Mask in the ***Subnet Mask*** text box.
    - Enter ELAN Gateway IP Address in the ***Gateway IP Address*** text box.
    - Enter ELAN Subnet Mask in the ***Subnet Mask*** text box.
    - Check **SIP Line** check box to enable SIP Line for this Node.

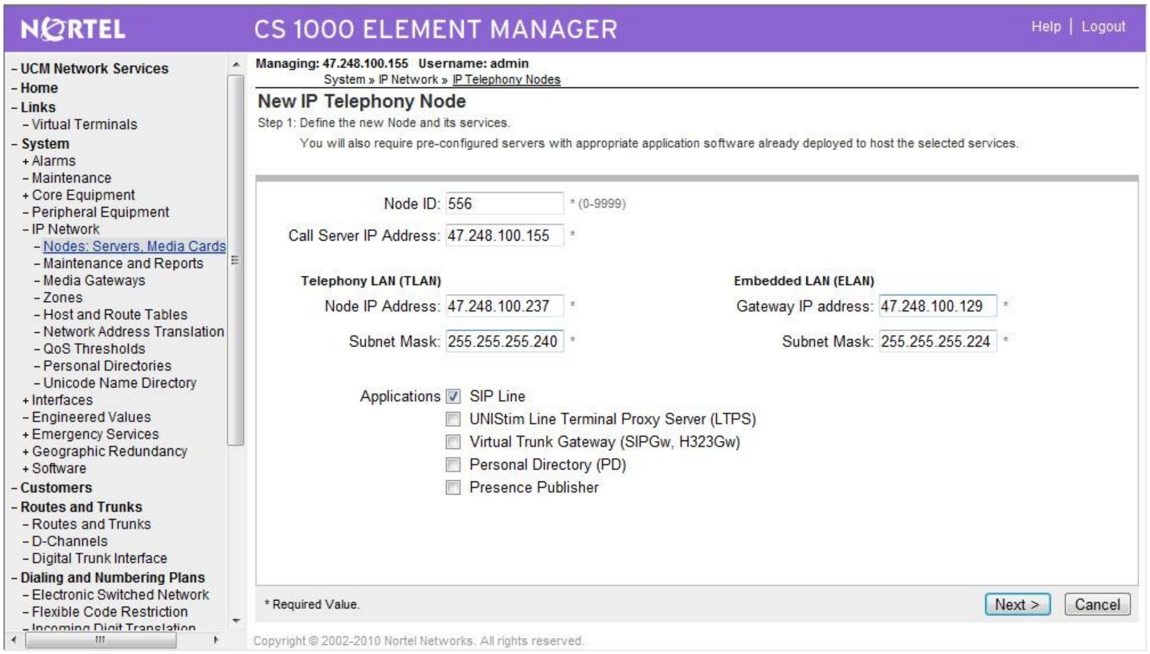

Figure 5 – IP Telephony Node

- Click ***Next.*** The page, New IP Telephony Note with Node ID, will appear as shown in **Figure 6**.
- On the Add Server page, from the ***drop down menu*** list, select the desired server to add to the node.
- Click ***Add***.
- Select the check box next to the newly added server, and click ***Make Leader***.

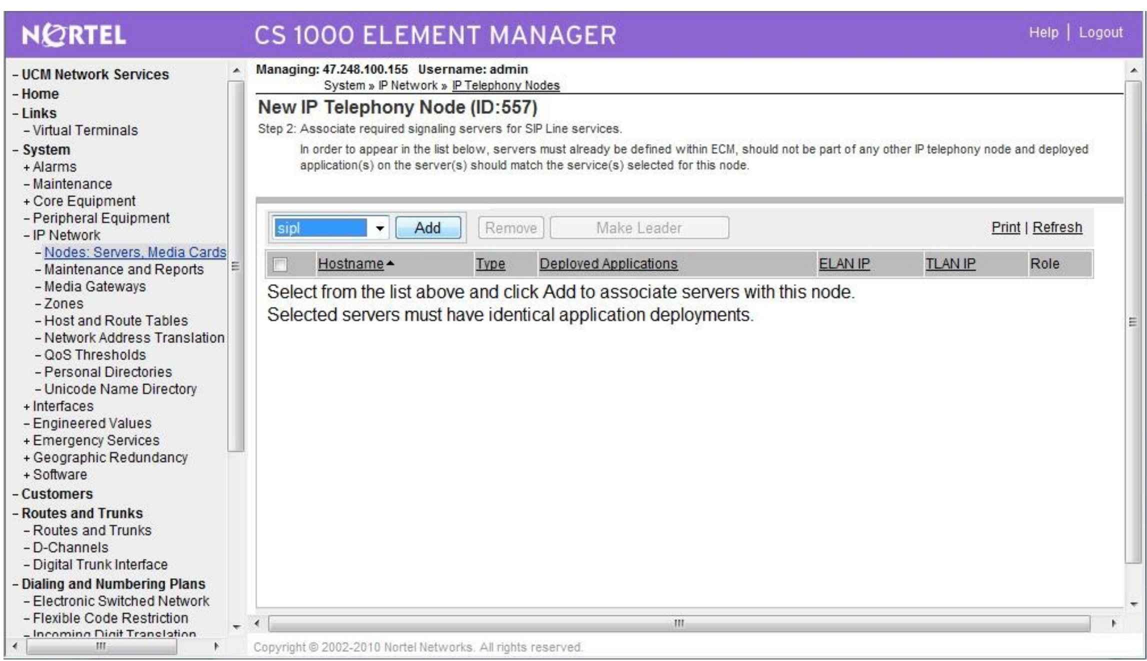

Figure 6 – IP Telephony Node – Add Server

- Click ***Next.*** The SIP Line Configuration Detail page appears as shown in **Figure 7**.
- Enter SIP Line domain name in ***SIP Domain name*** text box. This must be the same as the domain name configured in ***Customers***.

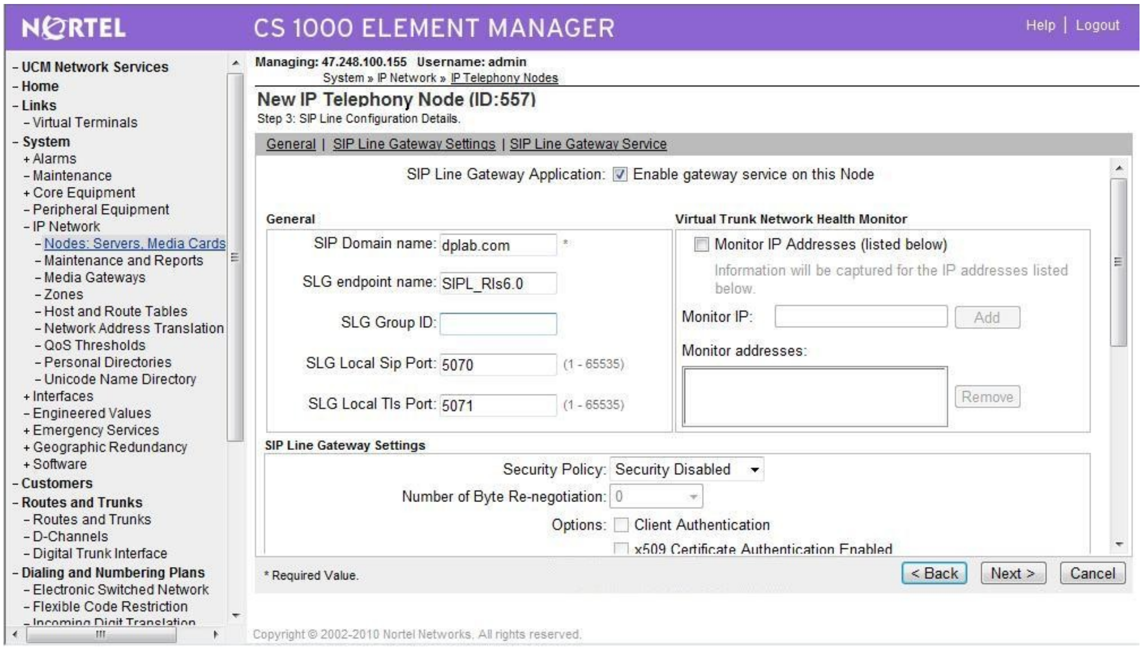

Figure 7 – SIP Line Node Details

- Under the ***SIP Line Gateway Services*** section, select ***MO*** from the ***SLG Role*** list.
- From the ***SLG Mode*** list, select ***S1/S2*** (SIP Proxy Server 1 and Server 2), see **Figure 8**.

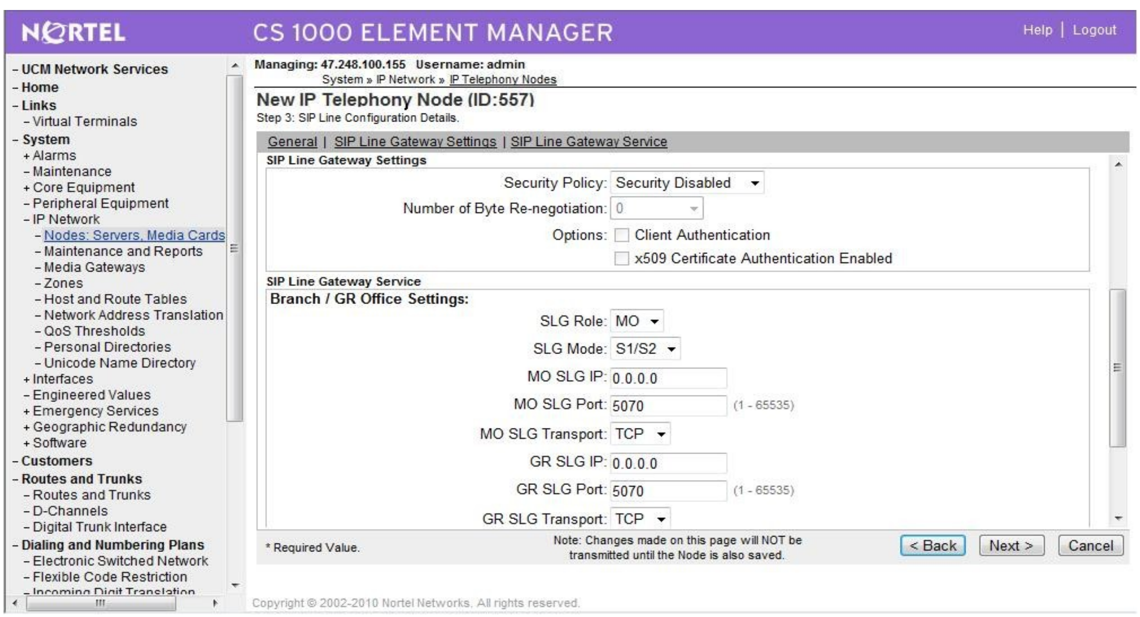

Figure 8 – SIP Line Node Details (cont.)

- Click *Next*. The ***Confirm new Node details*** page appears (not shown).
- Click ***Finish*** and wait for the configuration being saved. The ***Node Saved*** page appears, see **Figure 9**.

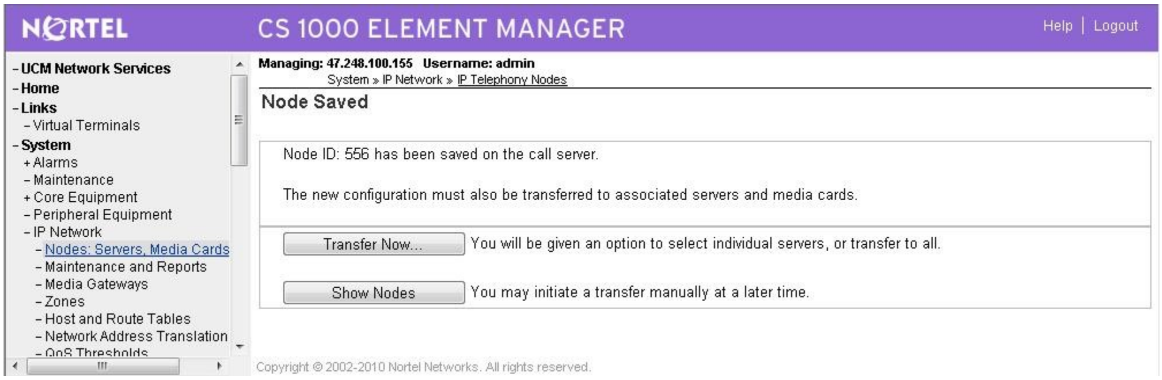

Figure 9 – Transfer Configuration

- Click ***Transfer Now.*** The ***Synchronize Configuration Files (Node ID 556) page*** appears.
- Select some or all of the node elements and then click ***Start Sync*** to transfer the configuration files to the selected servers, see **Figure 10**.

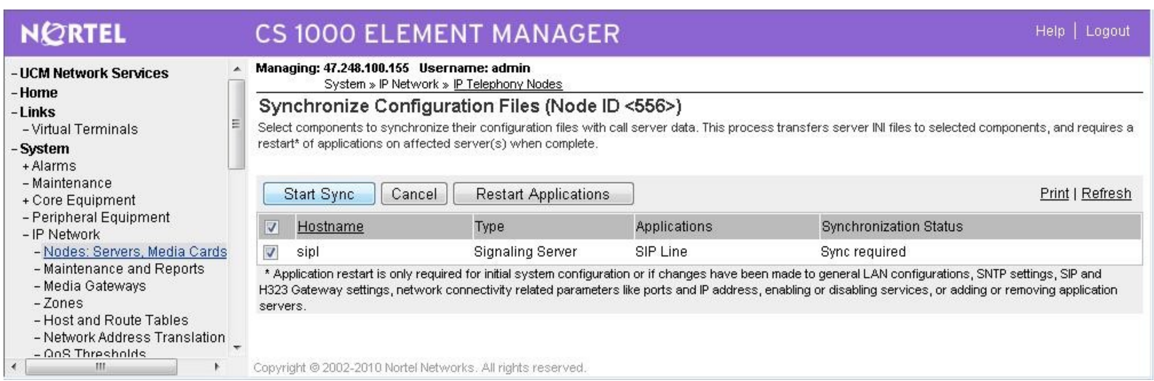

Figure 10 – Synchronize Configuration Files

4.5. D-Channel over IP Configuration

- On the EM page, on the left column menu navigate to ***Routes and Trunks → D-Channels.***
- Under the ***Configuration*** section, from the ***Choose a D-Channel Number*** list, select a D-Channel number, channel 30 in this configuration.
- Under the **Configuration** section, from the ***Type*** list, select ***DCH.***
- Click ***Add.***
- From the ***D channel Card Type (CTYP)*** list, select ***D-Channels is over IP (DCIP).***
- Click ***Add***.
- The ***D-Channels xx Property Configuration*** page appears as shown in **Figure 11**.
- From the ***Interface type for D-channel (IFC)*** list, select ***Meridian Meridian1 (SL1)***.
- Others fields are at default values.

- Click the ***Basic options (BSCOPT)*** link. The ***Basic options (BSCOPT)*** list expands.

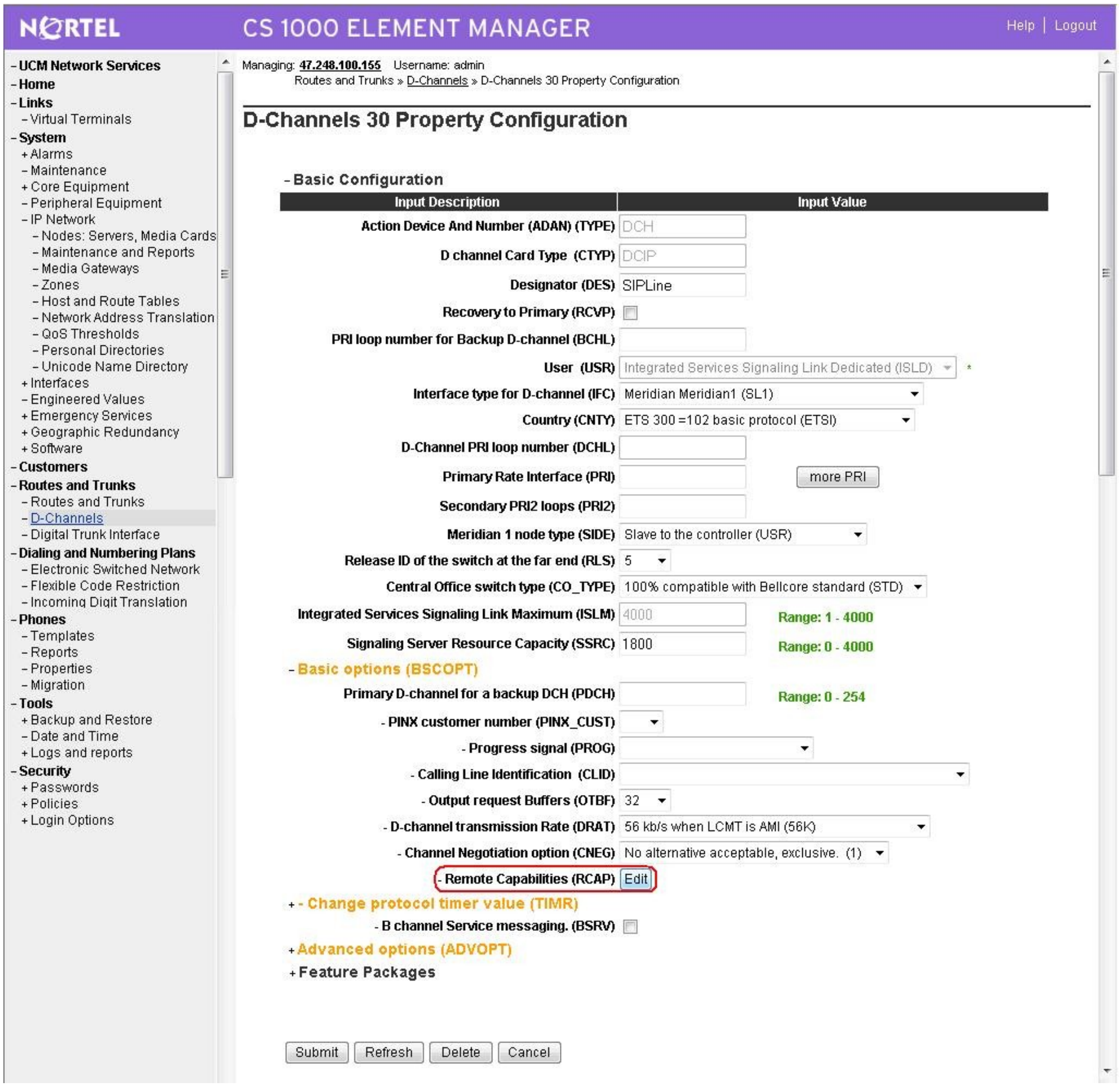

Figure 11 – SIP Line D-Channel Property Configuration

- Click ***Edit*** to configure ***Remote Capabilities (RCAP). The Remote Capabilities Configuration detail page*** will appear as shown in **Figure 12**.
- 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***.
- The ***D-Channel xx Property Configuration*** page reappears.

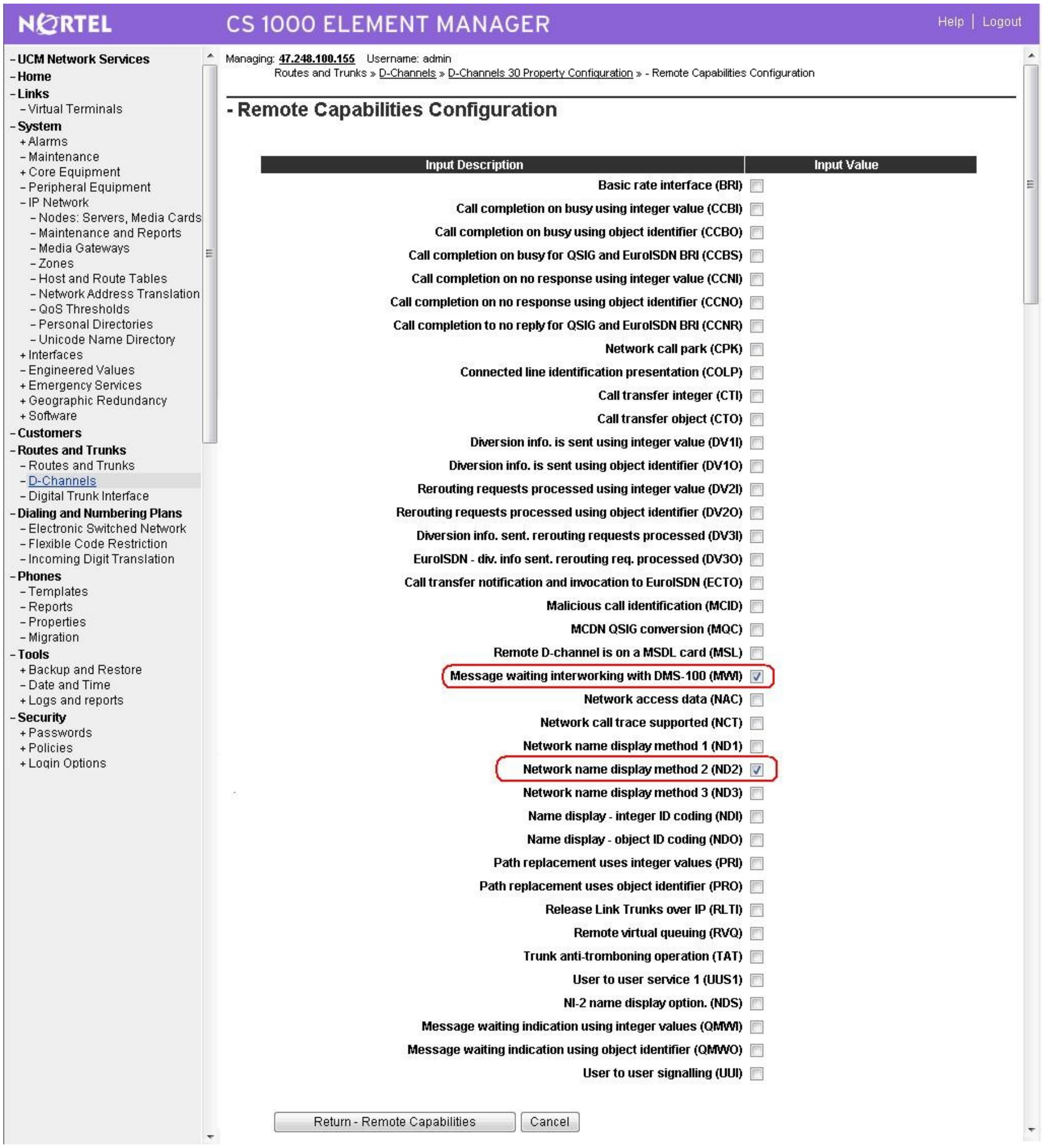

Figure 12 – SIP Line D-Channel RCAP Configuration Details

Message Waiting Interworking with DMS-100 (MWI) must be enabled to support voice mail notification on SIP Line endpoints.
Network Name Display Method 2 (ND2) must be enabled to support name display between SIP Line endpoints.
Others check boxes are left unchecked.

4.6. Application Module Link (AML) over Embedded LAN (ELAN) Configuration

- On the EM page, navigate to ***System → Interfaces → Application Module Link.***

- Click ***Add*** to add an Application Module Link. ***New Application Module Link*** page appears as shown in **Figure 13**.
- Enter AML port in the ***Port number*** text box. The SIP Line Service can use port 32 to port 127. In this case, SIP Line Service is configured to use port 32.
- Click ***Save*** to save the configuration.

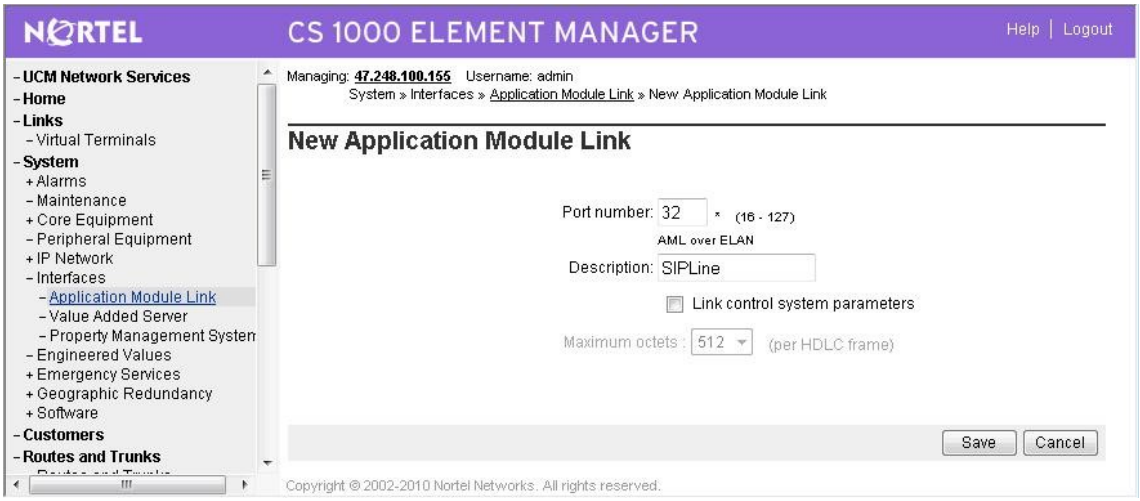

Figure 13 – Application Module Link Configuration

4.7. Value Added Server (VAS) Configuration

- On the EM page, navigate to ***System → Interfaces → Value Added Server.***
- Click ***Add*** to add new Value Added Server. The ***Add Value Added Server*** page appears.
- Click on the ***Ethernet LAN Link*** as shown in **Figure 14**.
- Enter the Ethernet LAN Link number in the ***Ethernet LAN Link*** text box.
- Ensure that the ***Application Security*** check box is unchecked.

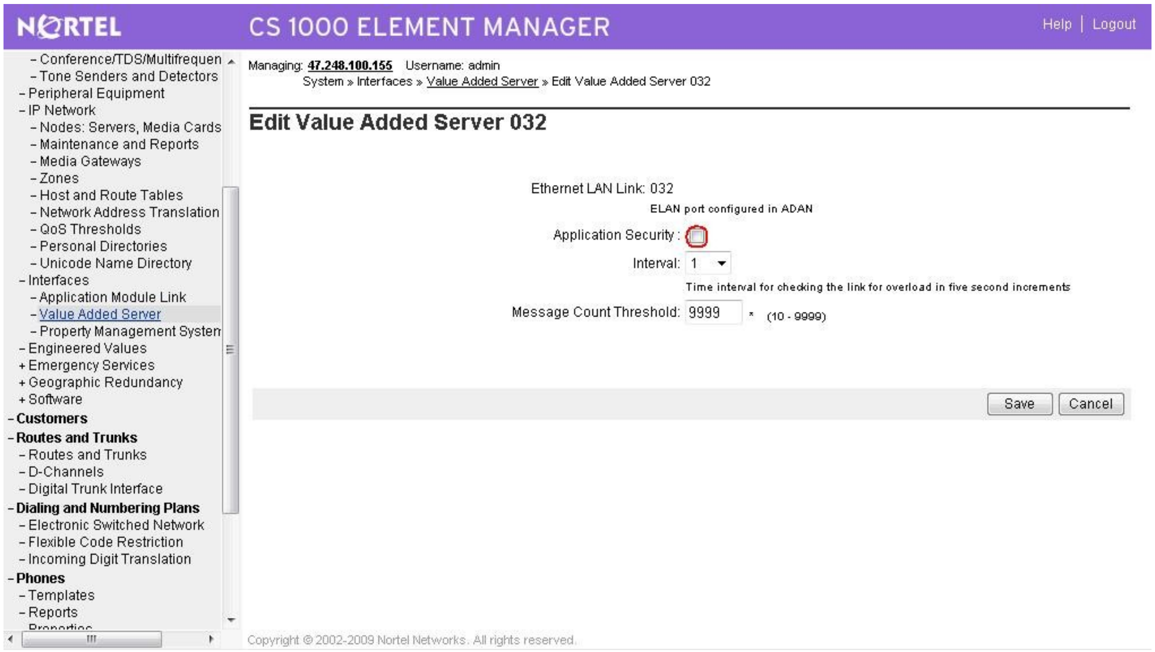

Figure 14 – Value Added Service for Application Module Link

4.8. Virtual Trunk Zone Configuration

- On the EM page, navigate to ***System* → *IP Network* → *Zones.***
- On the ***Zones*** page, select ***Bandwidth Zones***.
- On the ***Bandwidth Zones*** page, select a ***Bandwidth Zone number*** from the list, and click to ***Add*** (not shown).
- On the ***Zone Basic Property and Bandwidth Management*** page, set the zone properties based on bandwidth availability. It is recommended to set the ***Zone Strategy*** to ***BestQuality (BQ)*** as shown in **Figure 15**.
- From the ***Zone Intent (ZBRN)*** list, select ***VTRK (VTRK)***.
- Click ***Submit*** button at the bottom of the page to save and commit the changes.

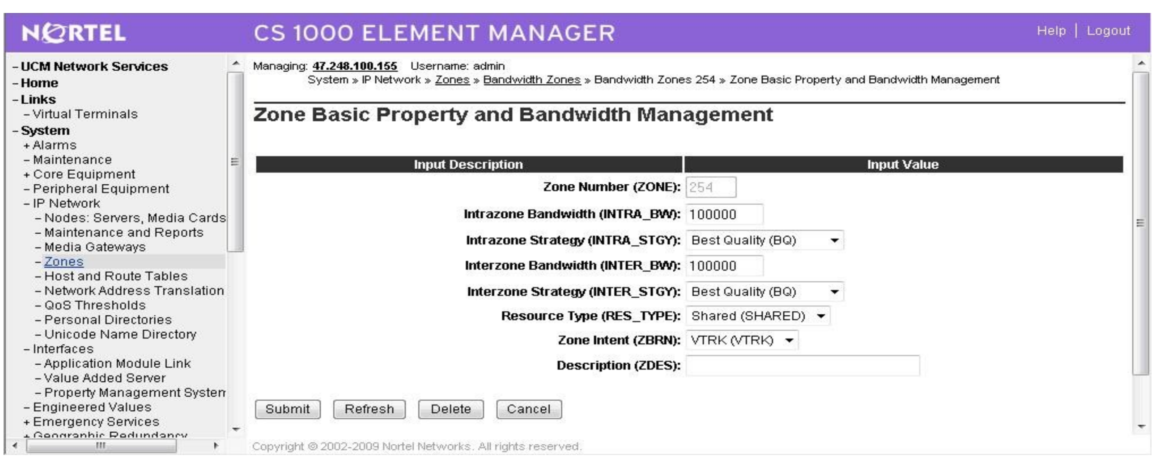

Figure 15 – Virtual Trunk Zone Configuration

4.9. SIP Line Route Data Block (RDB) Configuration

- On the EM page, navigate to ***Routes and Trunks → Routes and Trunks.***
- Click ***Add*** for the customer number.
- On the ***Customer xx, New Route Configuration*** page as shown in **Figure 16**.
- From the ***Route number (ROUT)*** list, select a route number.
- From the ***Trunk type (TKTP)*** list, select ***TIE trunk data block (TIE).***
- When Trunk Type (TKTP) is selected, the following options appear:
    - Trunk type M911P (M911P).
    - The route is for a virtual trunk route (VTRK).
    - Digital trunk route (DTRK).
    - Integrated services digital network option (ISDN).
- From the ***Incoming and outgoing trunk (ICOG)*** field, select ***Incoming and Outgoing (IAO)***.
- In the ***Access code for the trunk route (ACOD)*** field, enter the access code.
- Select ***The route is for virtual trunk route (VTRK)*** check box.
- In the ***Zone for codec selection and bandwidth management (ZONE)*** field, enter the zone number. (Use the same zone as configured in ***4.8 "Virtual Trunk Zone Configuration"***).
- In the ***Node ID of signaling server of this route (NODE)*** field, enter the node ID of the SIP Line Gateway.
- From the ***Protocol ID for the route (PCID)*** list, select ***SIP Line (SIPL).***
- Select the ***Integrated services digital network option (ISDN)*** check box.
- From the ***Mode of operation (MODE)*** list, select ***Route uses ISDN Signaling Link (ISLD)***.
- In the ***D channel number (DCH)*** field, enter the D-channel number.
- From the ***Interface type for route (IFC)*** list, select ***Meridian M1 (SL1)***.
- Ensure the ***Network calling name allowed (NCNA)*** check box is selected.
- Select the ***Trunk route optimization (TRO)*** check box. (Optional)
- ***Basic Route Options, Network Options, General Options, and Advanced Configurations*** sections set as default.
- Click ***Submit button to save the configuration changes***.

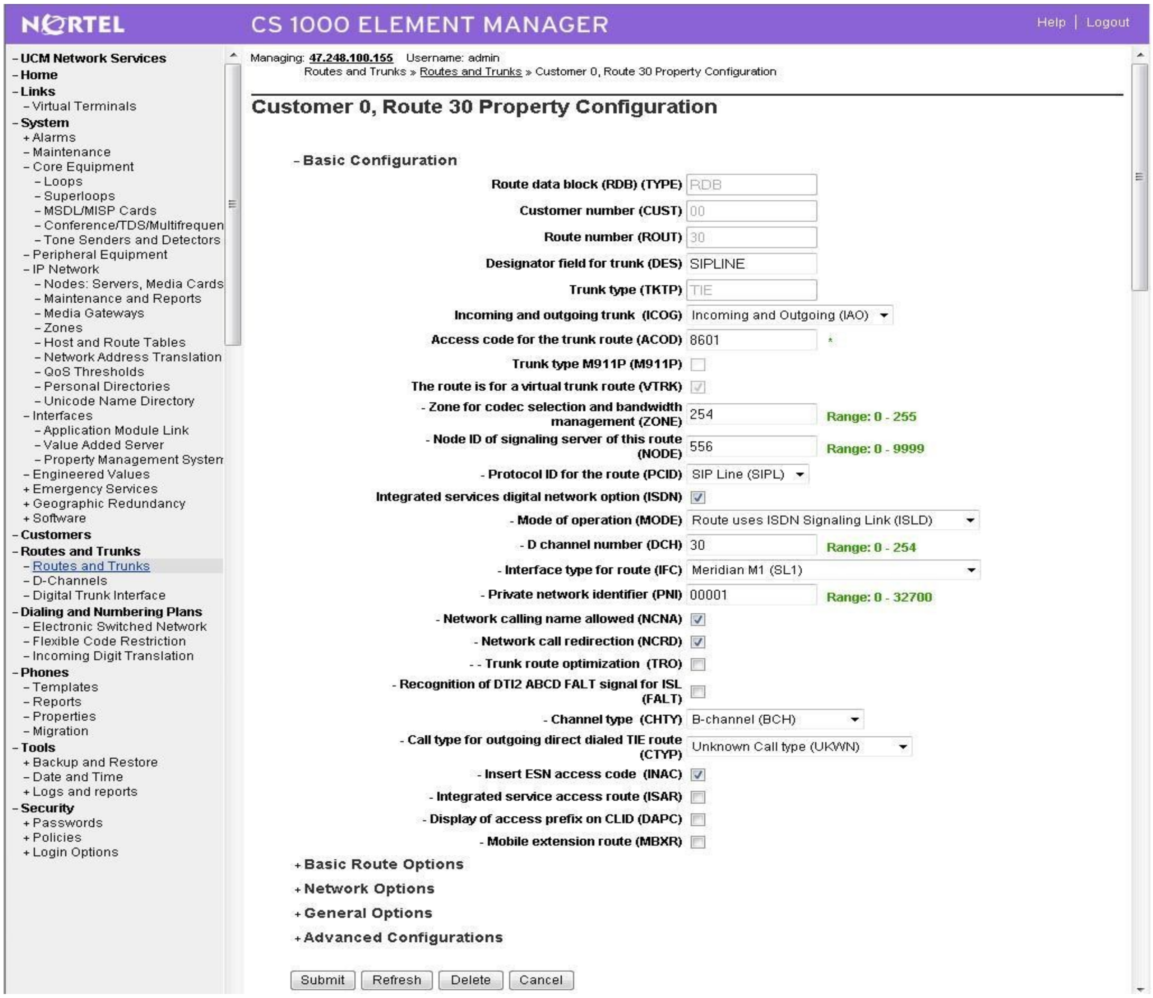

Figure 16 –SIP Line Route Configuration

Note: There is an outstanding issue with the CS 1000 Call Waiting feature which occurs when ***Network Call Redirection*** is enabled. If the Network Call Redirection feature is not required, uncheck the feature to make Call Waiting work.

4.10. SIP Line Virtual Trunk Configuration

- On the EM page, navigate to ***Routes and Trunks → Routes and Trunks.***
- Select the customer for which you are configuring Virtual Trunks.
- Click ***Add trunk associated with the route listing*** to add new trunk members.
- The ***Customer xx, Route yy, New Trunk Configuration*** Web page appears as show in **Figure 17**.
- Choose ***Multiple trunk input number (MTINPUT)*** if you are using more than one trunk.
- From the ***Trunk data block (TYPE)*** list, select ***IP Trunk (IPTI)***.
- In the ***Terminal Number (TN)*** field, enter a TN.
- Enter a ***Route number***, ***Member number (RTMB)***.
- Enter a ***Trunk Group Access Restriction (TGAR)*** value.

- In the ***Channel ID for this trunk (CHID)*** field, enter a ***channel ID*** (where the range is 1 to 382).
- To specify a ***Class of Service (CLS)*** for the trunk, click ***Edit***. The ***Class of Service Configuration*** Web page appears (not shown).
- Select a ***Class of Service***.
- Click ***Return Class of Service*** to return to the ***New Trunk Configuration*** Web page.
- Select ***Basic Configuration***. The ***Basic Configuration*** list expands.
- From the ***Start arrangement Incoming (STRI)*** list, select a value for the start arrangement for incoming calls.
- From the ***Start arrangement Outgoing (STRO)*** list, select a value for the start arrangement for outgoing calls.
- Select ***Advanced Trunk Configurations***. The ***Advanced Trunk Configurations*** list expands (not shown).
- Configure ***Network Class of Service group (NCOS).*** Please select the value between 0 - 99. The default value is 0.
- Click ***Save***.

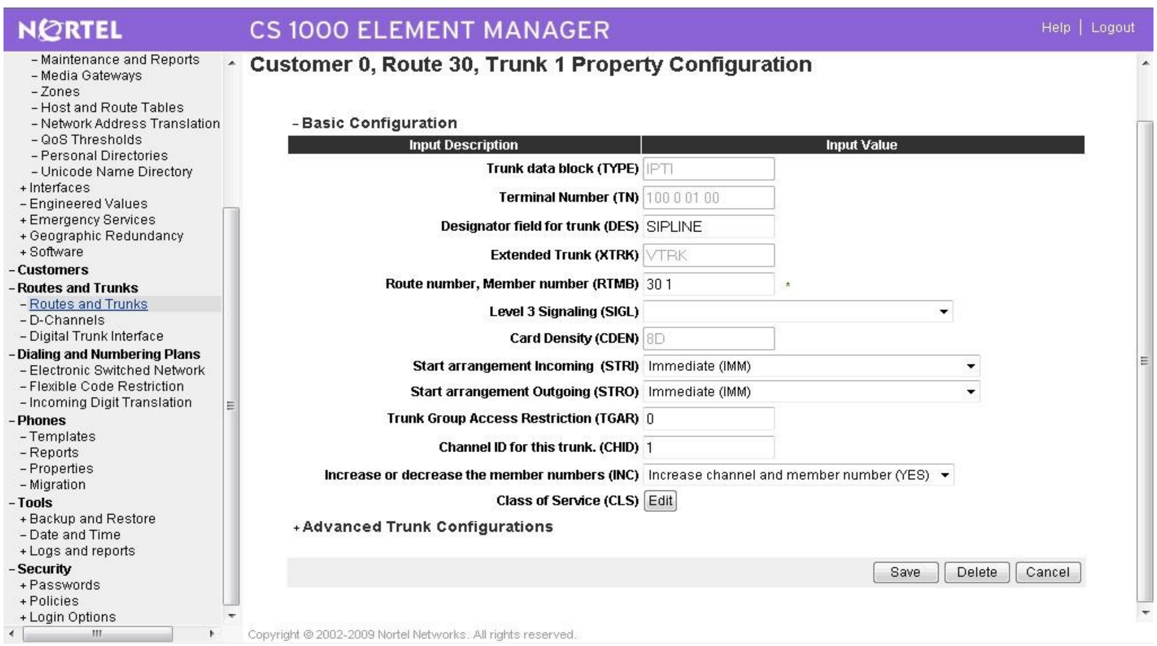

Figure 17 – SIP Line Trunk Configuration

4.11. SIP Line Phones Configuration

Following is a sample configuration for a Third Party SIP Line endpoint. Depending on supported features and service access level of the user, this configuration can be adjusted accordingly. The sample is using the Command Line Interface of CS1000. This can be done by logging in to the Call Server of CS1000 and using overlay 11 as shown below. The bold text values are the changes required while the rest are left at default values.

```
>LD 11

REQ: prt
TYPE: tnb
TN   100 1 11 5
DATE
PAGE
DES

DES TELE
TN   100 1 11 5  VIRTUAL
TYPE UEXT
CDEN 8D
CTYP XDLC
CUST 0
UXTY SIPL
MCCL YES
SIPN 0 ← Set this to 1 and set SIP3 to 0 if this TN is reserved for Avaya SIP Phones
SIP3 1 ← Set this to 1 and set SIPN to 0 if this TN is reserved for third party SIP Phones
FMCL 0
TLSV 0
SIPU 55116
NDID 556
SUPR NO
SUBR DFLT MWI RGA CWI MSB
UXID
NUID
NHTN
CFG_ZONE 001
CUR_ZONE 001
ERL
ECL  0
FDN  55555    ← If CLS FNA is equipped, call will be forwarded no answer to this number
TGAR 0
LDN  NO
NCOS 0
SGRP 0
RNPG 2        ← This field must be set first if call pickup is equipped (CLS PUA)
SCI  0
SSU
XLST
SCPW 1234
SFLT NO
CAC_MFC 0
CLS  UNR FBA WTA LPR PUA MTD FNA HTA TDD HFA CRPD
   MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
```

POD DSX VMD SLKD CCSD **SWD** LND **CNDA**
CFTD SFD MRD DDV **CNIA** CDCA MSID DAPA BFED RCBD
ICDD CDMD LLCN MCTD CLBD AUTU
GPUA DPUA **DNDA** CFXA 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 ICRD MCDD T87D MSNV FRA PKCH
CPND_LANG ENG
RCO 0
HUNT **55555** ← If CLS HTA/FBA is equipped, call will be forwarded busy to this number
LHK 0
PLEV 02
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY 00 **SCR 55116** 0 MARP
CPND
CPND_LANG ROMAN
NAME **POLYCOM 55116**
XPLN 13
DISPLAY_FMT FIRST, LAST
01 HOT U **2655116** MARP 0
02 SCU **0004** ← Speed Call User
03
04 **MSB** ← This key can be different than key 04 to enable Make Set Busy (MBS) feature
05
.
.
16
17 TRN
18 AO6
19 CFW 16 55116
20 RGA
21 PRK
22 RNP
23
24 PRS

25 CHG
26 CPN

4.12. PSTN Outside Trunk Configuration

Following is a sample configuration which was used during compliance testing. For more information about PRI Trunk Configuration, see Section 9[3].

4.12.1. Procedure summary

This procedure was applied for the CS 1000 system under test as per **Figure 1**. The provisioning is done using the Command Line Interface by logging in to the Call Server of the CS1000.

No.	Overlay	Action
1	LD 17	Adding a PRI card
2	LD 17	Adding a PRI D-Channel
3	LD 15	Defining a PRI customer
4	LD 16	Defining a PRI service route
5	LD 14	Defining service channels and PRI trunks
6	LD 73	Defining system timers and clock controller
7	LD 48	Enable TMDI or PRI MSDL card
8	LD 60	Enable Clock Controller
9	LD 60	Enable Digital trunk loop
10	LD 96	Enable D-channel

4.12.2. Adding a PRI card

The programming example below shows how to add a PRI card using LD 17. For all other fields not listed in the example press RETURN to use default values.

Prompt	Response	Description
REQ	CHG	Change data.
TYPE	CFN	Configuration data block.

CEQU	YES	Changes to common equipment.
DLOP	10	Digital Trunk Interface Loop
MG_CARD	4 0 1	MG card assigned to superloop
MODE	PRI	Mode of operation
TMDI	YES	Card is TMDI card
TRSH	0	Threshold

4.12.3. Adding a PRI D-channel

The programming example below shows how to add a PRI D-channel using LD 17. For all other fields not listed in the example press RETURN to use default values.

Prompt	Response	Description
REQ	CHG	Change existing data
TYPE	CFN	Configuration data block.
ADAN	NEW DCH 10	Add a primary D-channel (any unused SDI port.) xx = 1-9 for Option 11C main cabinet, 11-19 for IP expansion cabinet 1, 21-29 for IP expansion cabinet 2, 31-39 for IP expansion cabinet 3, and 41-49 for IP expansion cabinet 4. Xx = 11-14, 21-24, 31-34, 41-44 of the first, second, third and fourth Media Gateway, respectively.
CTYP	TMDI	Card type where: MSDL = The NTBK51BA Downloadable D-Channel Daughterboard. TMDI = TMDI (NTRB21) card.
DES	T1_QSIG	Designator field.
USR	PRI	D-channel is for ISDN PRI only. ***Note:*** 2.0 Mb only supports PRI or SHA user
IFC	ISGF	Interface type.
DCHL	10	PRI card number carries the D-channel. Must match entry made for the "CDNO" associated with the "DCHI" prompt above.

		Where: xx = 1-9 for Option 11C main cabinet, 11-19 for IP expansion cabinet 1, 21-29 for IP expansion cabinet 2, 31-39 for IP expansion cabinet 3, and 41-49 for IP expansion cabinet 4. xx = 11-14, 21-24, 31-34, 41-44 of the first, second, third and fourth Media Gateway, respectively.
SIDE	NET	NET = network, the controlling switch (applied for CS 1000 PSTN simulator USR = slave to the controller (applied for CS 1000 system under test)
RLS	6	Software release of far-end. This is the current software release of the far-end. If the far-end has an incompatible release of software, it prevents the sending of application messages, for example, 'Network Ring Again.
RCAP	CCBI CCNI PRI DV3I CTI QMWI	Remote Capabilities.
PR_TRIGS PR_TRIGS PR_TRIGS PR_TRIGS	DIV 2 3 CNG 2 3 CON 2 3 CTR2 2 3	Path Replacement Triggers

4.12.4. Defining a PRI customer

The programming example below shows how to define a PRI customer using LD 15. For all other fields not listed in the example press RETURN to use default values.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CDB	Customer data block.
CUST	0	Customer number.
ISDN	YES	Customer is equipped with ISDN.

4.12.5. Defining a PRI service route

The programming example below shows how to add a PRI service route using LD 16. For all other fields not listed in the example press RETURN to use default values.

Prompt	Response	Description
REQ	NEW	Create new data
TYPE	RDB	Route data block
CUST	0	Customer number
ROUT	10	Route number
DES	T1_QSIG	Designator field for trunk
TKTP	TIE	Trunk type
DTRK	YES	Digital trunk route
ISDN	YES	ISDN option
MODE	PRI	Route used for PRI only
PNI	1	Customer private network identifier. Is the same as the CDB PNI at far-end.
IFC	ISGF	Interface type.
ICOG	IAO	Incoming and outgoing
ACOD	8010	Trunk access code

4.12.6. Defining service channels and PRI trunks

The programming example below shows how to create service channels and PRI trunks using LD 14. For all other fields not listed in the example press RETURN to use default values.

Prompt	Response	Description
REQ	NEW 23	Create 23 new trunks
TYPE	Tie	Trunk type
TN	10 1	Loop (card) and channel number for digital trunks
PCML	MU	System PCM law.

DES	T1_QSIG	Designator field for trunk
CUST	0	Customer number
RTMB	10 1	Service route number and trunk member number
CLS	UNR DTN	Trunk Class Of Service

5. Configure Polycom VVX 1500

This section describes how to access the Polycom VVX 1500 SIP endpoint web interface and configure the Polycom VVX 1500 for testing.

5.1. SIP Registration

This section shows how to log in to the main SIP telephone registration.

In the web browser address field, enter the Polycom VVX 1500 IP address. The Polycom VVX 1500 login page will appear as shown in **Figure 18**. Enter the username, **Polycom,** and its default password, **456**.

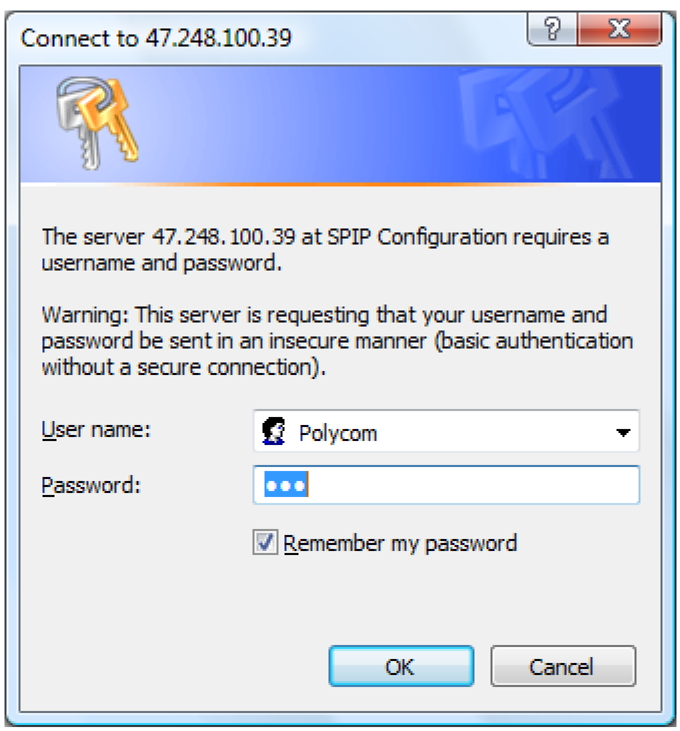

Figure 18 – Login Screen

Click the OK button, the main VXX Configuration webpage screen appears as in **Figure 19** below.

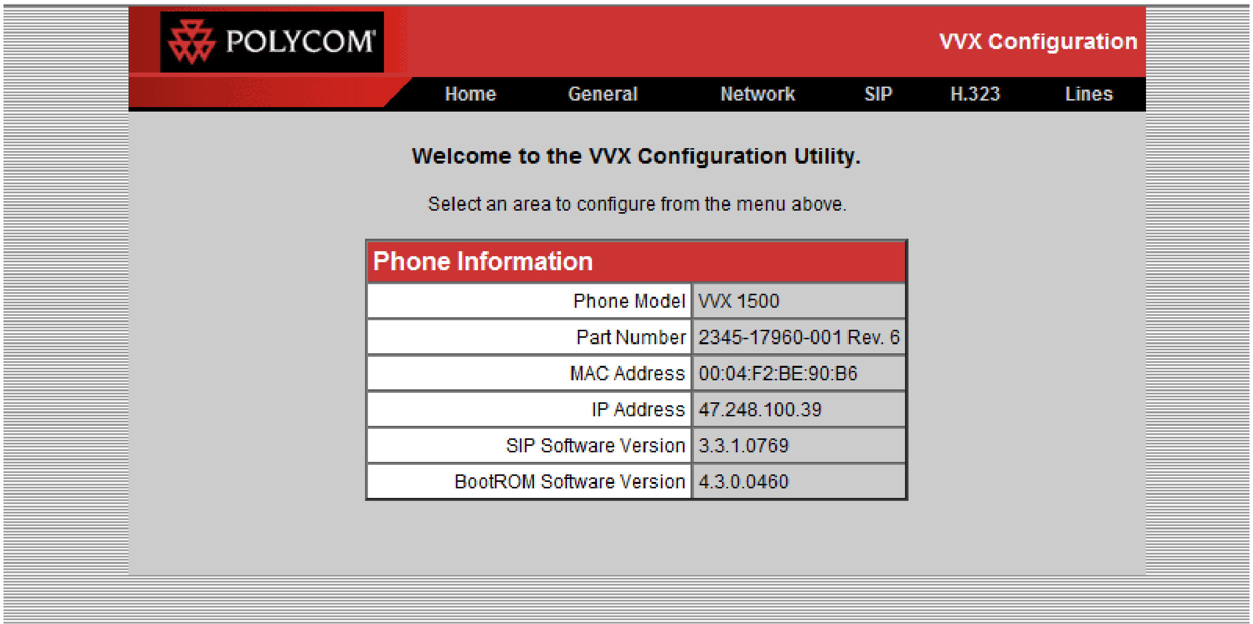

Figure 19 - Main Configuration Screen

5.2. Configure the SIP Lines

This section shows how to configure the SIP telephone to register with the CS1000 SIP Line.

In the main configuration screen (see **Figure 19**), click on the Lines menu on the right hand of menu bar.
Provide the SIP information as shown in **Figure 20** below. Click on the Submit button.

Figure 20 - SIP Configuration Screen

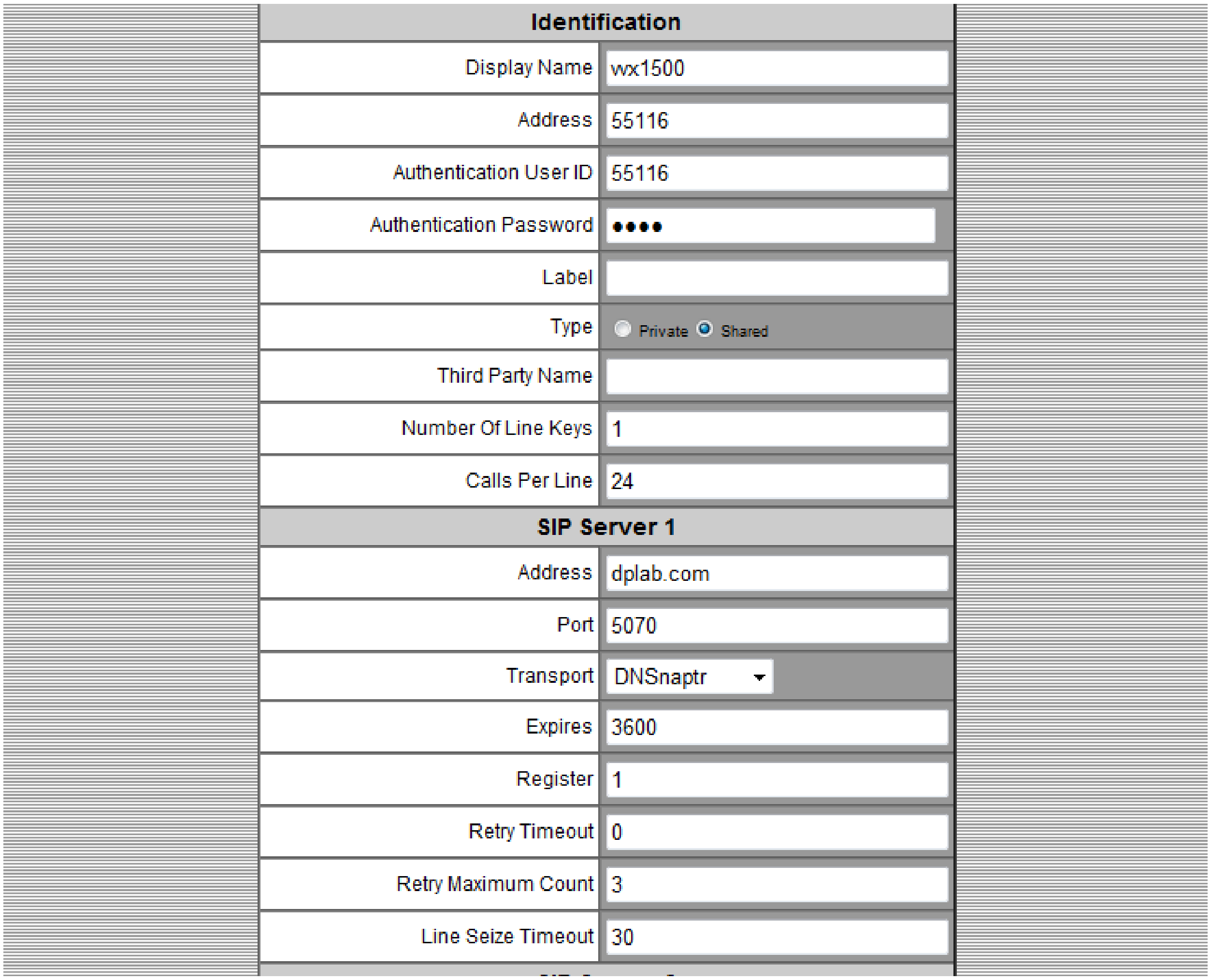

Figure 20b – SIP Configuration Screen (Cont)

In the main configuration screen, click on the SIP menu in the menu bar.
Provide SIP information as shown in **Figure 21** below and click on the **Submit** button.

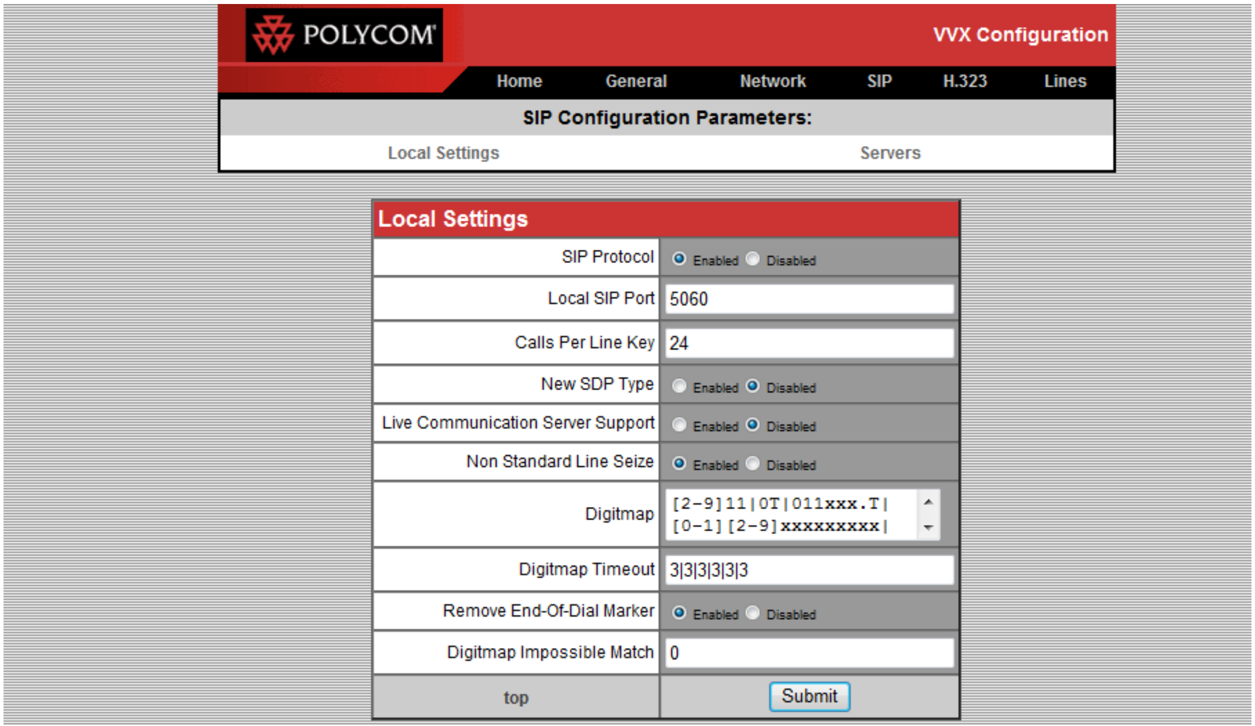

Figure 21 – SIP Configuration Parameters

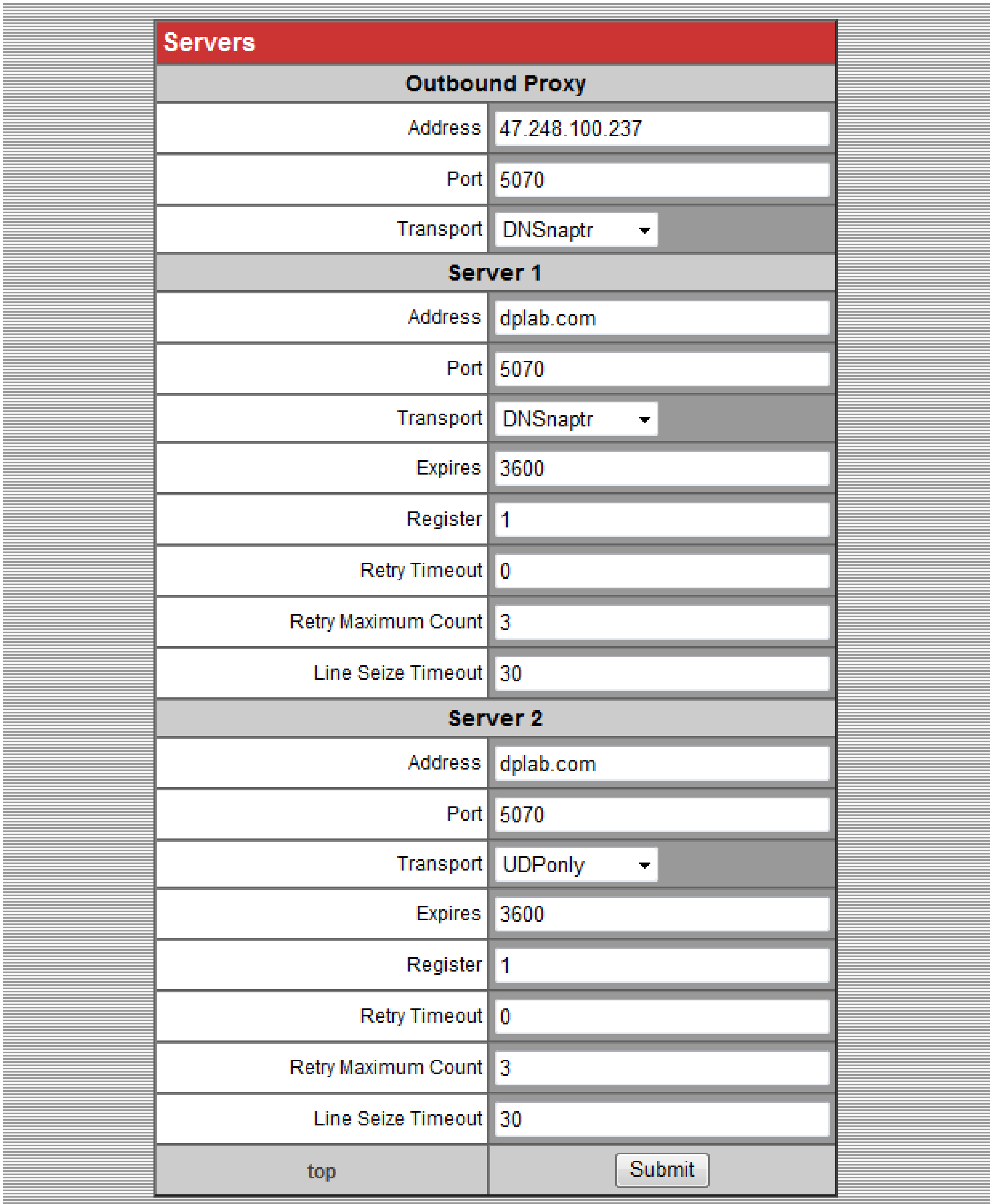

Figure 21b – SIP Configuration Parameters (Cont)

5.3. Local Call Forward Settings

This section shows how to set "Local Call Forward" on the SIP telephone.

On the **Figure 20** above, click on "Lines" menu to go Lines Parameters setting. Scroll down to the Call Diversion section as shown in **Figure 22** below.

The "Call Forward No Answer" ringing timeout settings established here should be less than the timeout setting on the CS 1000 Call Server in order to make the "Local Call Forward No Answer" take effect.

The "Call Forward Always" settings on the CS 1000 Call Server must be OFF in order to make the "Local Call Forward Always" take effect.

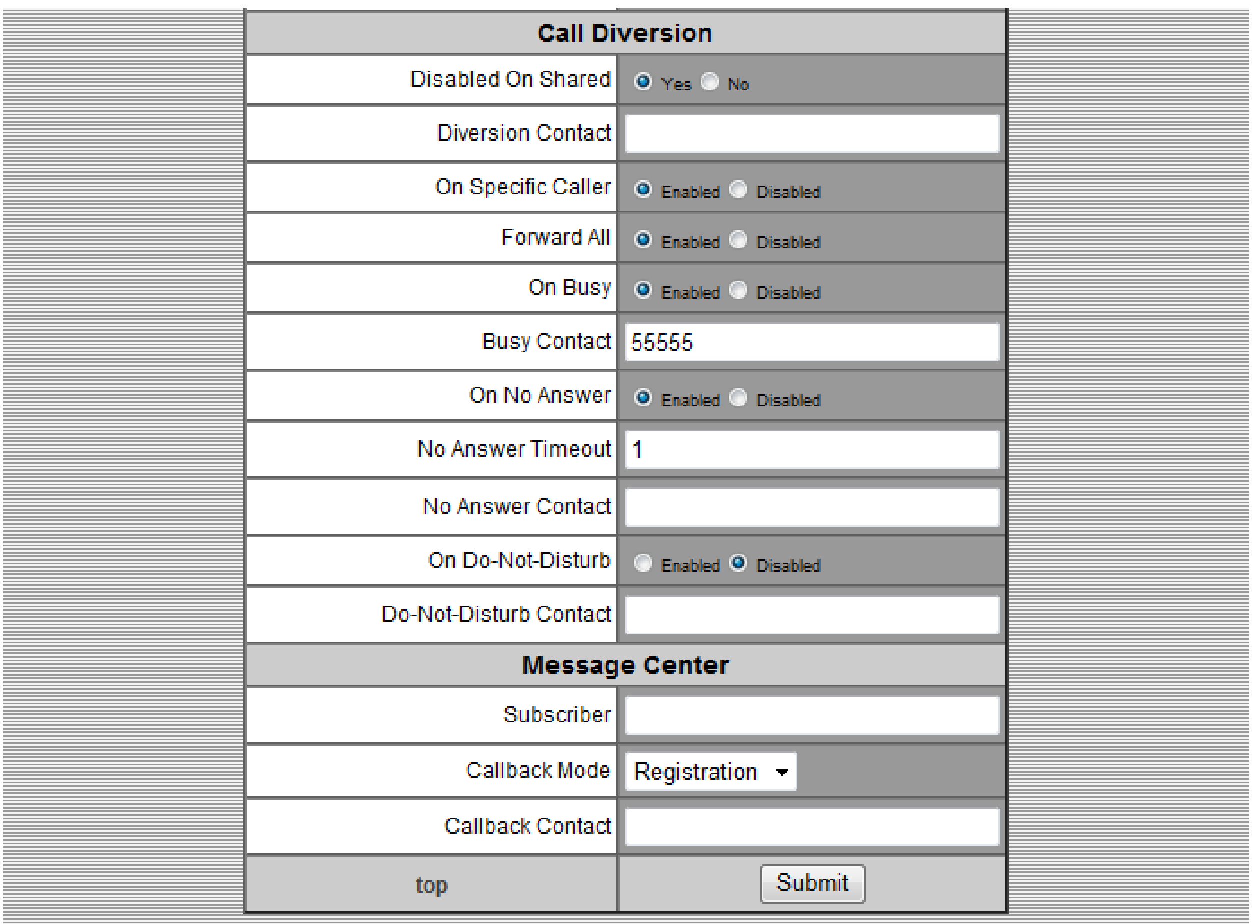

Figure 22- Local Call Forward Settings

5.4. Codec settings

This section describes how to configure DTMF and Codec settings in the Polycom VVX 1500 SIP telephone.

5.4.1. Codec Settings

This section shows how to set Codec on the SIP telephone. Testing has been done on both codecs; G711 and G729. The example below only shows how to set them for testing G711, but it can be done the same way for G729.

In the main configuration screen (see **Figure 20**), click on the "General" menu, and then click on Audio Processing tab from the list, select the codec setting as shown in **Figure 23**.

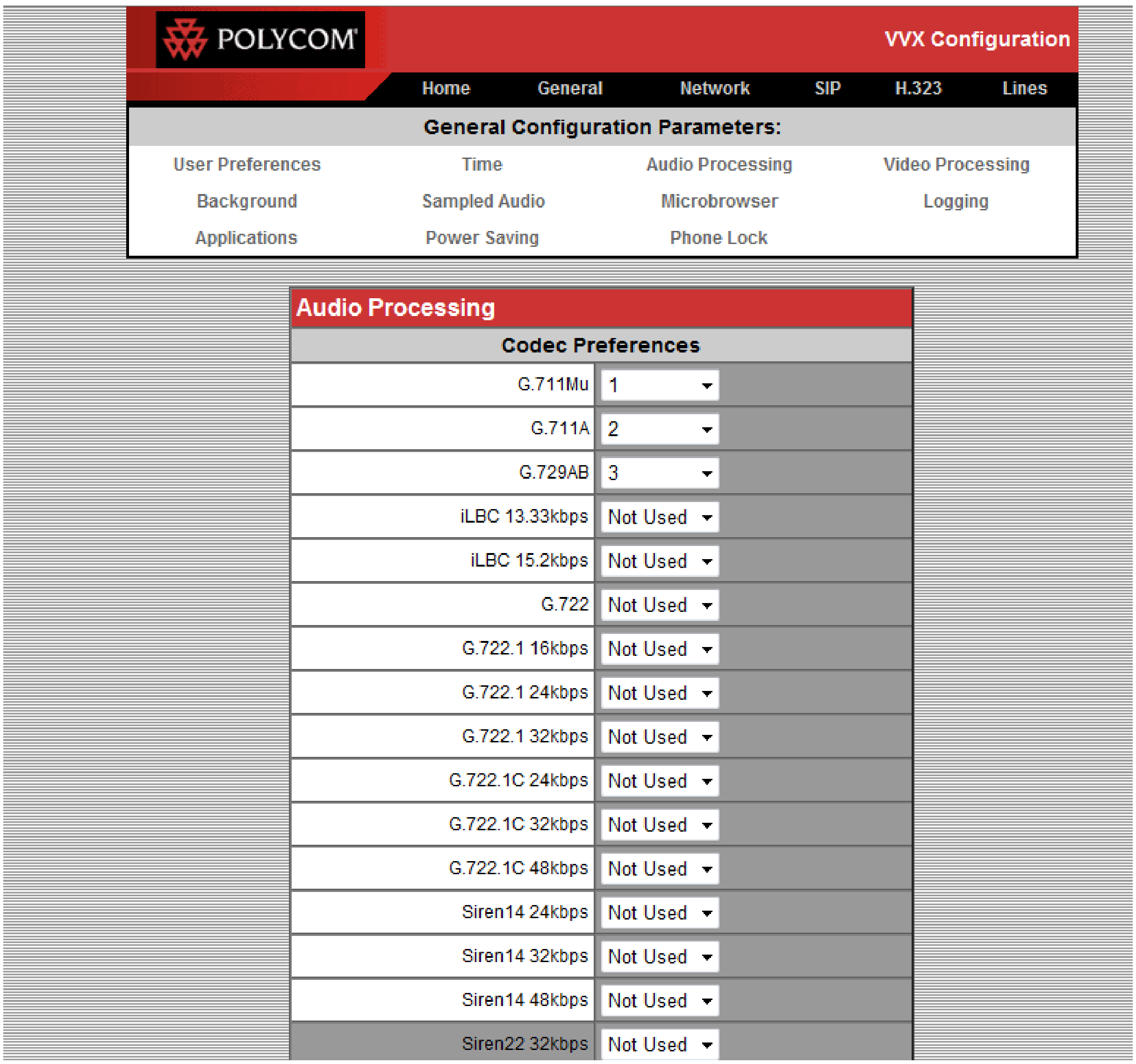

Figure 23 - Codec Preferences

6. General Test Approach and Test Results

The focus of this interoperability compliance testing was primarily to verify the call establishment between the Polycom VVX 1500 and the CS1000 telephones. Other call features, such as busy, hold, DTMF, MWI and codec negotiation, were exercised.

6.1. General test approach

The general test approach was to have one of the CS1000 telephone clients/users to place a call to and from the Polycom VVX 1500 telephone and to exercise other telephony features. The main objectives were to verify the Polycom VVX 1500 successfully performed the following:

- Registration of Polycom VVX 1500 telephones to the CS1000.
- Call establishment from Polycom VVX 1500 telephones with Avaya CS1000 SIP and non SIP telephones/clients.
- Call establishment from Polycom VVX 1500 telephones with emulated PSTN telephones.
- Basic call operation: DTMF transmission, voicemail with MWI notification, busy, hold, call waiting, second call.
- Advanced CS 1000 Call Server features: speed dial, group call pickup, ring again busy/no answer, call park/retrieve, call forward (busy/all call/no answer), conference and multiple appearances DN.
- Codec negotiation.

6.2. Test Results

The objectives outlined in section 6.1 were verified. The following observations were made during the compliance testing:

- Avaya has not performed audio performance testing or reviewed the Polycom VVX1500 compliance to required industry standards
- Enable Network Call Redirection (NCRD) in CS1000 Call Server SIP Line Route will cause an issue with Call Waiting.
- VVX1500 does not support DTMF via SIP INFO, DTMF default as RFC2833. For using IN-BAND DTMF, *set tone.dtmf.rfc2833Control*="0" on config file. (sip.cfg)
- During a call where video is enabled, if the VVX1500 phone is put on hold, when un-hold to resume the call, video screen display on VVX1500 is dropped. As a workaround, configure the phone to use the RTCP video fast update by setting video.*forceRtcpVideoCodecControl* ="1" in the config file (default is "0")
- It is recommended to use the FTP protocol to do an upgrade for Polycom VVX 1500 from version 3.2.2 to 3.3.1.

7. Verification Steps

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

- Verify that the SIP telephone registers successfully with the CS 1000 SIP Line Gateway server and Call Server by using CS 1000 Linux command line and CS 1000 Call Server overlay LD 32.
  - Log in to the sipline server using Avaya account.
  - Issue command "slgSetShowByUID [userID]" where userID is SIP Line user's ID being checked.

```
[admin@sipl ~]$ slgSetShowByUID 55116

=== VTRK ===
UserID           TN               Clients  Calls  SetHandle
---------------  ---------------  -------  -----  ---------
          55116     100-01-11-05        1      0  0xb5f06cc0
        StatusFlags = Registered Controlled KeyMapDwld SSD
        FeatureMask =
        CallProcStatus = -1

        Current Client = 0, Total Clients = 1

         == Client 0 ==
         IP:Port:Trans = 47.248.100.39:5060:udp
         Type          = SIP3
         UserAgent     = PolycomVVX-VVX_1500-UA/3.3.1.0769
         x-nt-guid     = 4d5292e739d5bf962f6e3069270b50a8
         RegDescrip    =
         RegStatus     = 1
         PbxReason     = OK
         SipCode       = 200
         Expire        = 3600
         Contact       = sip:55116@47.248.100.39:5060
         Nonce         = 07d4dc76098837ab648dd775172cab8b
         NonceCount    = 2
         hTimer        = 0xa39cec8
         TimeRemain    = 1596
         Stale         = 0
         Outbound      = 0
         ClientGUID    = 0
         Appearances   = 0

  ScrKey  Lamp  KeyDN Appearance

        Key  Func  Lamp  Label
        0    3     0     55116
        1    126   0     2655116
        2    21    0
        3    29    0
        4    22    0
        5    23    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
```

- Log in to the call server using admin account.
- Load overlay 32 and then issue command "stat [TN]" where TN is the SIP Line user's TN being checked

```
>ld 32
NPR000
.stat 100 1 11 5
IDLE REGISTERED 00
```

- Place a call from and to Polycom VVX 1500 telephone and verify that the call is established with 2-way speech path.
- During the call, use a pcap tool (ethereal/wireshark) at the SIPLine 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 6.1**, with some exceptions outlined in **Section 6.2**. The Polycom VVX 1500 version 3.3.1.0769 is considered in compliance with Avaya CS1000 SIP Line System Release 6.0.

9. Additional References

Product documentation for Avaya products may be found at:
http://support.nortel.com/go/main.jsp

[1] *Communication Server 1000 SIP Line Fundamental, Release 6.0, Revision 01.08, February 2010, Document Number NN43001-508*

[2] *Communication Server 1000E Maintenance, Release 6.0, Revision 03.16, January 2010, Document Number NN43041-700*

[3] *Communication Server 1000 ISDN Primary Rate Interface Installation and Commissioning, Revision 01.03, August 2007, Document Number NN43001-301*

[4] *Troubleshooting Guide for Distributors, Release 6.0, Revision 02.02, December 2009, Document Number NN43001-730*

[5] *Communication Server 1000E Installation and Commissioning, Release 6.0, Revision 03.06, February 2010, Document Number NN43041-310*

[6] *Communication Server 1000E Software Upgrades, Revision 03.12, February 2010, Document Number NN43041-458*

[7] *Communication Server 1000E Linux Platform Base and Applications Installation and Commissioning, Revision 03.10, February 09, 2009, Document Number NN43001-315*

[8] *Communication Server 1000 Unified Communications Management Common Services Fundamentals, Revision: 03.04, September 28, 2009, Document Number NN43001-116*

10. Appendix

This section is to help a user provisioning the clock synchronization of a network of CS1000 systems used under test. This will create the PRI trunk synchronization between 2 CS1000 under test in **Figure 1**; Main and Emulated PSTN systems. In this example, the emulated PSTN will have a clock controller card. Therefore, these provisioning steps below will only apply to the emulated PSTN CS1000 system only. The steps below can be accomplished by login to the Call Server and using command line interface on overlay 73.

Defining system timers and clock controller parameters

The programming example below shows how to define system timers and clock controller parameters using LD 73. For all other fields not listed in the example press RETURN to use default values.

Prompt	Response	Description
REQ	CHG	Change data.
TYPE	DDB	Digital Data Block
MGCLK	4 0 1	Card slot number for Media Gateway 4 0
PREF	1	Card number of PRI/DTI/SILC or DTI2/PRI2/SILC containing the primary clock reference.
SREF	1	Card number of PRI/DTI/SILC or DTI2/PRI2/SILC containing the primary clock reference.

Enabling T1 QSIG Service

Enable TMDI card

The example below shows how to enable TMDI card using LD 48.

>ld 48
LNK000
.enl tmdi 4 0 1

OK

Enable Clock Controller

The example below shows how to enable clock controller using LD 60.

>ld 60
DTI000
.enl cc 4 0
.OK

Enable PRI loop

The example below shows how to enable PRI loop using LD 60.

>ld 60
DTI000
.enll 10

OK
.

Enable D-Channel

The D-Channel may not automatically come up. The example below shows how to enable PRI D-channel using LD 96.

>ld 96
DCH000
.enl dch 10
.
DCH: 10 EST CONFIRM TIME: 19:38:44 30/09/2009

DCH 10 UIPE_OMSG CC_RESTART_REQ REF 00000000 CH 0 TOD 19:38:44 CK E0DAF978
TYPE: ALL CHANNEL

DCH 10 UIPE_IMSG CC_RESTART_CONF REF 00008000 TOD 19:38:44 CK E0DAF9C2
TYPE: ALL CHANNEL

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.