



**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.

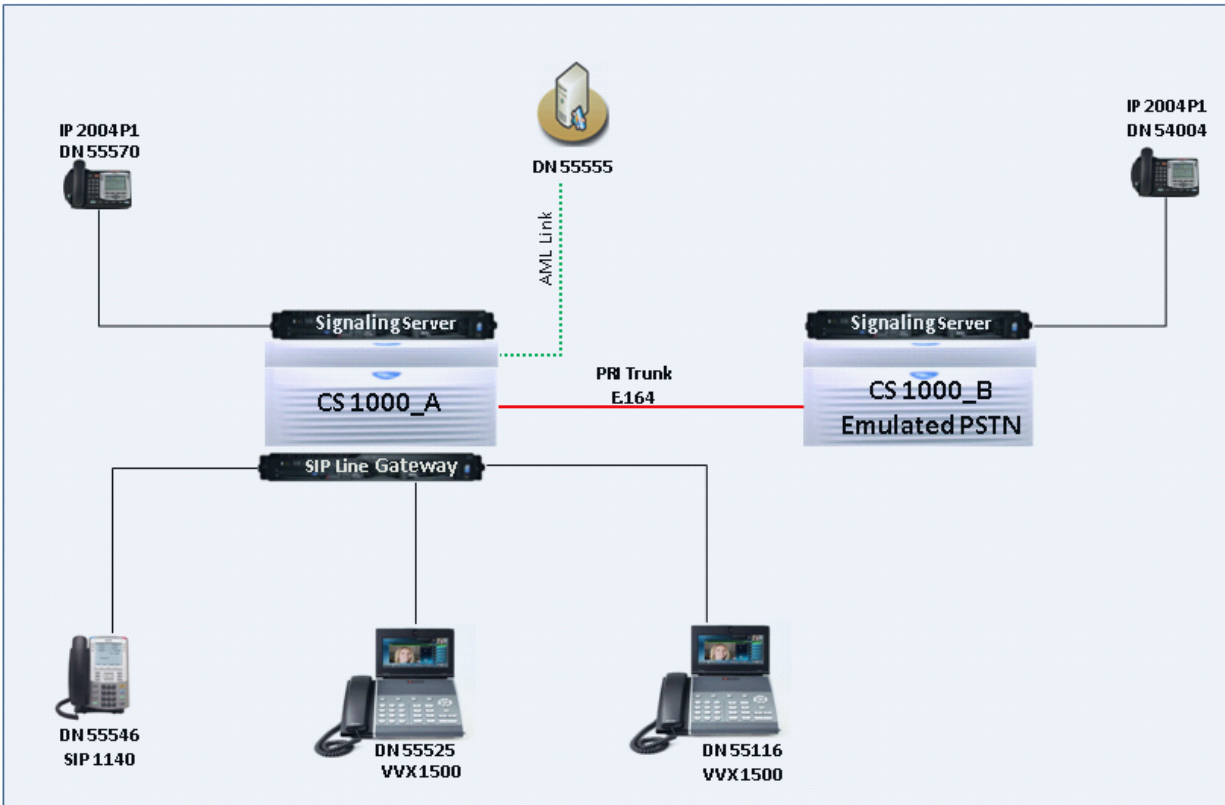


Figure 1 - Network Configuration Diagram

3. Equipment and Software Validated

System	Software Version
Avaya CS1000	<ul style="list-style-type: none"> Call Server (CPPM): 6.00RJ Signaling Server (CPPM): 6.00.18 SIP Line Gateway (HP DL320)
Avaya voicemail system	<ul style="list-style-type: none"> CallPilot 5.0 system
Avaya 1140 SIP client Avaya SIP soft-phones	<ul style="list-style-type: none"> 02.02.16.00
Avaya IP phones	<ul style="list-style-type: none"> 2050PC: 3.02.0045
Polycom VVX 1500 SIP Software	<ul style="list-style-type: none"> 3.3.1.0769
Polycom VVX 1500 BootROM	<ul style="list-style-type: none"> 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:
 - o Installed with CS 1000 Release 6.0 Linux Base.
 - o Joined CS 1000 Release 6.0 Security Domain.
 - o 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].

Unified Communications Management - Windows Internet Explorer

https://sipl.ca.nortel.com/frames.faces?body=/secureObjectManagement.faces

Unified Communications Management

NORTEL UNIFIED COMMUNICATIONS MANAGEMENT [Help](#) | [Logout](#)

Host Name: sipl.ca.nortel.com Software Version: 02.00.0055.00(3266) User Name admin

Elements

New elements are registered into the security framework, or may be added as simple hyperlinks. Click an element name to launch its management service.

	Element Name	Element Type	Release	Address	Description
1	EM on coresb	CS1000	6.0	47.248.100.163	New element.
2	EM on ss2	CS1000	6.0	47.248.100.155	New element.
3	EM on sipd	CS1000	6.0	47.248.100.155	New element.
4	47.248.100.155	Call Server	6.0	47.248.100.155	New element.
5	sipl.ca.nortel.com (primary)	Linux Base	6.0	47.248.100.235	Base OS element.
6	coresb.ca.nortel.com (member)	Linux Base	6.0	47.248.100.201	Base OS element.
7	ss2.ca.nortel.com (member)	Linux Base	6.0	47.248.100.206	Base OS element.
8	sipt.ca.nortel.com (member)	Linux Base	6.0	47.248.100.194	Base OS element.
9	sps1.ca.nortel.com (member)	Linux Base	6.0	47.248.100.234	Base OS element.
4	47.248.100.162	Media Gateway Controller	6.0	47.248.100.162	New element.

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Trusted sites | Protected Mode: Off 100%

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.

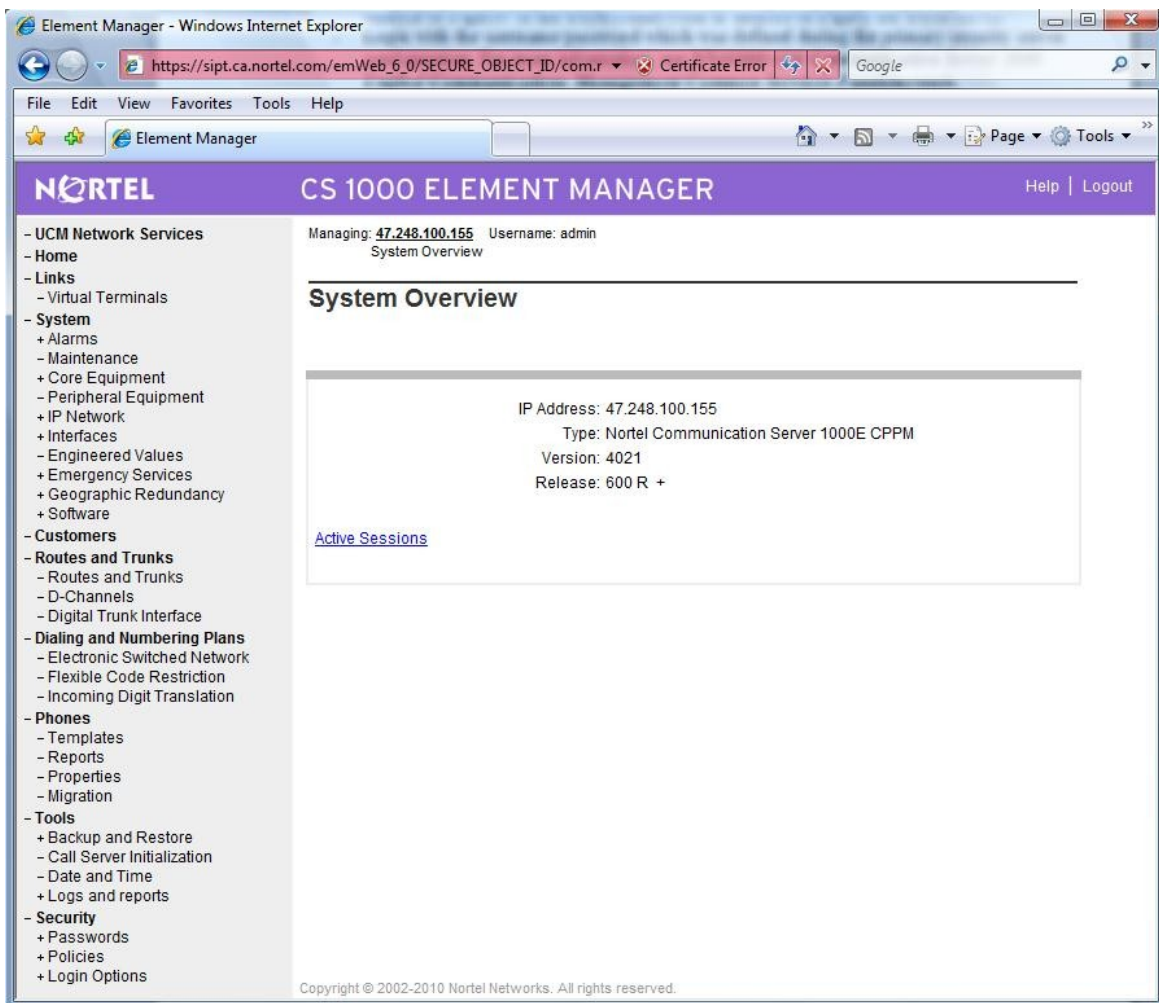


Figure 3 - CS 1000 EM Home Page

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**.

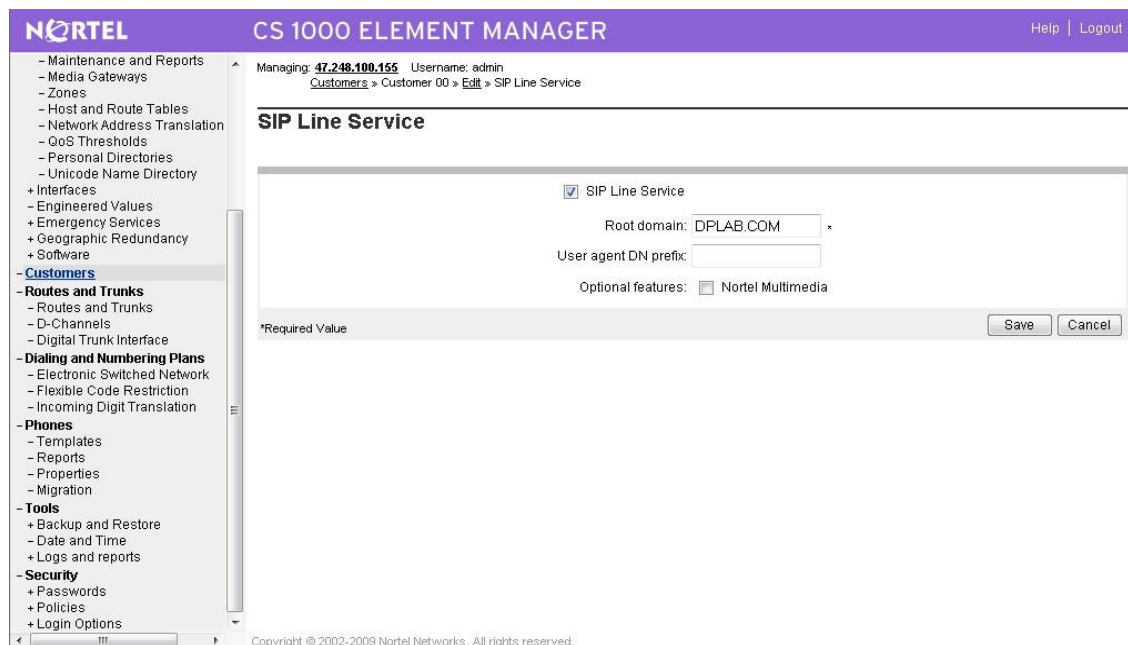


Figure 4 - SIP Line Service in Customers Data Block

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**:
 - o Enter Node ID in the **Node ID** text box.
 - o Enter Call Server IP Address in the **Call Server IP Address** text box.
 - o Enter Node IP Address in the **Node IP Address** text box.
 - o Enter TLAN Subnet Mask in the **Subnet Mask** text box.
 - o Enter ELAN Gateway IP Address in the **Gateway IP Address** text box.
 - o Enter ELAN Subnet Mask in the **Subnet Mask** text box.
 - o Check **SIP Line** check box to enable SIP Line for this Node.

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: 47.248.100.155 Username: admin
System » IP Network » IP Telephony Nodes

New IP Telephony Node

Step 1: Define the new Node and its services.
You will also require pre-configured servers with appropriate application software already deployed to host the selected services.

Node ID: 556 * (0-9999)

Call Server IP Address: 47.248.100.155 *

Telephony LAN (TLAN)
Node IP Address: 47.248.100.237 *
Subnet Mask: 255.255.255.240 *

Embedded LAN (ELAN)
Gateway IP address: 47.248.100.129 *
Subnet Mask: 255.255.255.224 *

Applications ☒ SIP Line
☐ UNiStim Line Terminal Proxy Server (LTPS)
☐ Virtual Trunk Gateway (SIPGw, H323Gw)
☐ Personal Directory (PD)
☐ Presence Publisher

* Required Value. Next > Cancel

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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**.

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: 47.248.100.155 Username: admin
System » IP Network » IP Telephony Nodes

New IP Telephony Node (ID:557)

Step 2: Associate required signaling servers for SIP Line services.
In order to appear in the list below, servers must already be defined within ECM, should not be part of any other IP telephony node and deployed application(s) on the server(s) should match the service(s) selected for this node.

sip1 Add Remove Make Leader Print Refresh

<input type="checkbox"/>	Hostname	Type	Deployed Applications	ELAN IP	TLAN IP	Role
Select from the list above and click Add to associate servers with this node. Selected servers must have identical application deployments.						

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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**.

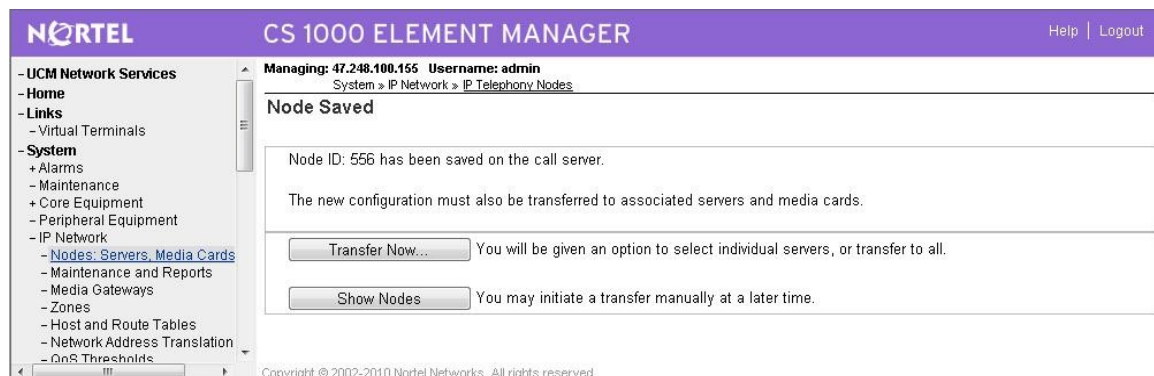


Figure 9 – Transfer Configuration

- 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**.

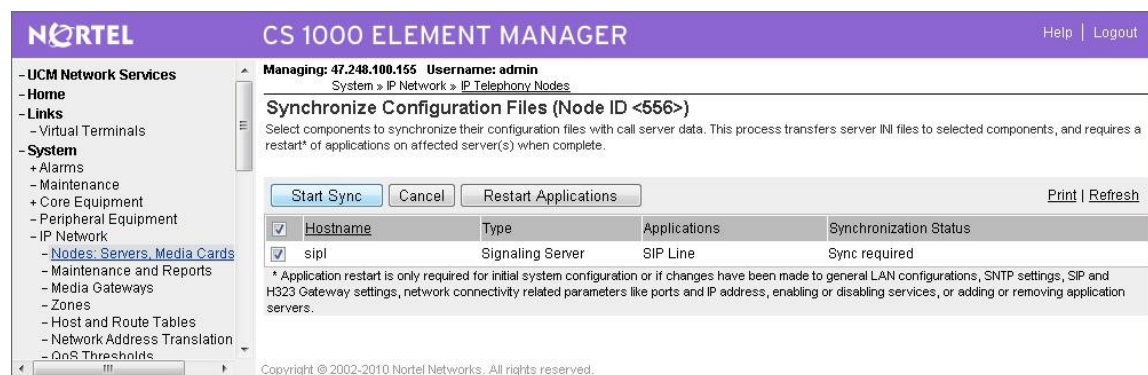


Figure 10 – Synchronize Configuration Files

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.

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: 47.248.100.155 Username: admin
Routes and Trunks » D-Channels » D-Channels 30 Property Configuration

D-Channels 30 Property Configuration

- Basic Configuration

Input Description	Input Value
Action Device And Number (ADAN) (TYPE)	DCH
D channel Card Type (CTYP)	DCIP
Designator (DES)	SLLine
Recovery to Primary (RCVP)	<input type="checkbox"/>
PRI loop number for Backup D-channel (BCHL)	
User (USR)	Integrated Services Signaling Link Dedicated (ISLD) *
Interface type for D-channel (IFC)	Meridian Meridian1 (SL1)
Country (CNTY)	ETS 300 =102 basic protocol (ETSI)
D-Channel PRI loop number (DCHL)	
Primary Rate Interface (PRI)	<input type="button" value="more PRI"/>
Secondary PRI2 loops (PRI2)	
Meridian 1 node type (SIDE)	Slave to the controller (USR)
Release ID of the switch at the far end (RLS)	5
Central Office switch type (CO_TYPE)	100% compatible with Bellcore standard (STD)
Integrated Services Signaling Link Maximum (ISLM)	4000 Range: 1 - 4000
Signaling Server Resource Capacity (SSRC)	1800 Range: 0 - 4000
- Basic options (BSCOPT)	
Primary D-channel for a backup DCH (PDCH)	Range: 0 - 254
- PINX customer number (PINX_CUST)	
- Progress signal (PROG)	
- Calling Line Identification (CLID)	
- Output request Buffers (OTBF)	32
- D-channel transmission Rate (DRAT)	56 kb/s when LCMT is AMI (56K)
- Channel Negotiation option (CNEG)	No alternative acceptable, exclusive. (1)
- Remote Capabilities (RCAP)	Edit
+ - Change protocol timer value (TIMR)	
- B channel Service messaging. (BSRV)	<input type="checkbox"/>
+ Advanced options (ADVOPT)	
+ Feature Packages	

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.

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: **47.248.100.155** Username: admin
 Routes and Trunks > D-Channels > D-Channels 30 Property Configuration > Remote Capabilities Configuration

- Remote Capabilities Configuration

Input Description	Input Value
Basic rate interface (BRI)	<input type="checkbox"/>
Call completion on busy using integer value (CCBI)	<input type="checkbox"/>
Call completion on busy using object identifier (CCBO)	<input type="checkbox"/>
Call completion on busy for QSIG and EuroISDN BRI (CCBS)	<input type="checkbox"/>
Call completion on no response using integer value (CCNI)	<input type="checkbox"/>
Call completion on no response using object identifier (CCNO)	<input type="checkbox"/>
Call completion to no reply for QSIG and EuroISDN BRI (CCNR)	<input type="checkbox"/>
Network call park (CPK)	<input type="checkbox"/>
Connected line identification presentation (COLP)	<input type="checkbox"/>
Call transfer integer (CTI)	<input type="checkbox"/>
Call transfer object (CTO)	<input type="checkbox"/>
Diversion info. is sent using integer value (DV1I)	<input type="checkbox"/>
Diversion info. is sent using object identifier (DV1O)	<input type="checkbox"/>
Rerouting requests processed using integer value (DV2I)	<input type="checkbox"/>
Rerouting requests processed using object identifier (DV2O)	<input type="checkbox"/>
Diversion info. sent. rerouting requests processed (DV3I)	<input type="checkbox"/>
EuroISDN - div. info sent. rerouting req. processed (DV3O)	<input type="checkbox"/>
Call transfer notification and invocation to EuroISDN (ECTO)	<input type="checkbox"/>
Malicious call identification (MCID)	<input type="checkbox"/>
MCDN QSIG conversion (MQC)	<input type="checkbox"/>
Remote D-channel is on a MSDI card (MSL)	<input type="checkbox"/>
Message waiting interworking with DMS-100 (MWI)	<input checked="" type="checkbox"/>
Network access data (NAC)	<input type="checkbox"/>
Network call trace supported (NCT)	<input type="checkbox"/>
Network name display method 1 (ND1)	<input type="checkbox"/>
Network name display method 2 (ND2)	<input checked="" type="checkbox"/>
Network name display method 3 (ND3)	<input type="checkbox"/>
Name display - integer ID coding (NDI)	<input type="checkbox"/>
Name display - object ID coding (NDO)	<input type="checkbox"/>
Path replacement uses integer values (PRI)	<input type="checkbox"/>
Path replacement uses object identifier (PRO)	<input type="checkbox"/>
Release Link Trunks over IP (RLTI)	<input type="checkbox"/>
Remote virtual queuing (RVQ)	<input type="checkbox"/>
Trunk anti-tromboning operation (TAT)	<input type="checkbox"/>
User to user service 1 (UUS1)	<input type="checkbox"/>
NI-2 name display option. (NDS)	<input type="checkbox"/>
Message waiting indication using integer values (QMWI)	<input type="checkbox"/>
Message waiting indication using object identifier (QMWIO)	<input type="checkbox"/>
User to user signalling (UUI)	<input type="checkbox"/>

Return - Remote Capabilities Cancel

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.

The screenshot shows the 'New Application Module Link' configuration page in the Nortel CS 1000 Element Manager. The page has a purple header with the Nortel logo and 'CS 1000 ELEMENT MANAGER'. A sidebar on the left contains a tree view with categories like UCM Network Services, Home, Links, System, Customers, and Routes and Trunks. The 'Application Module Link' option is selected under the 'Links' category. The main content area is titled 'New Application Module Link' and contains the following fields:

- Managing: 47.248.100.155 Username: admin
- System » Interfaces » Application Module Link » New Application Module Link
- Port number: 32 (range 16 - 127)
- AML over ELAN
- Description: SIPLine
- ☐ Link control system parameters
- Maximum octets: 512 (per HDLC frame)
- Save and Cancel buttons at the bottom right.

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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.

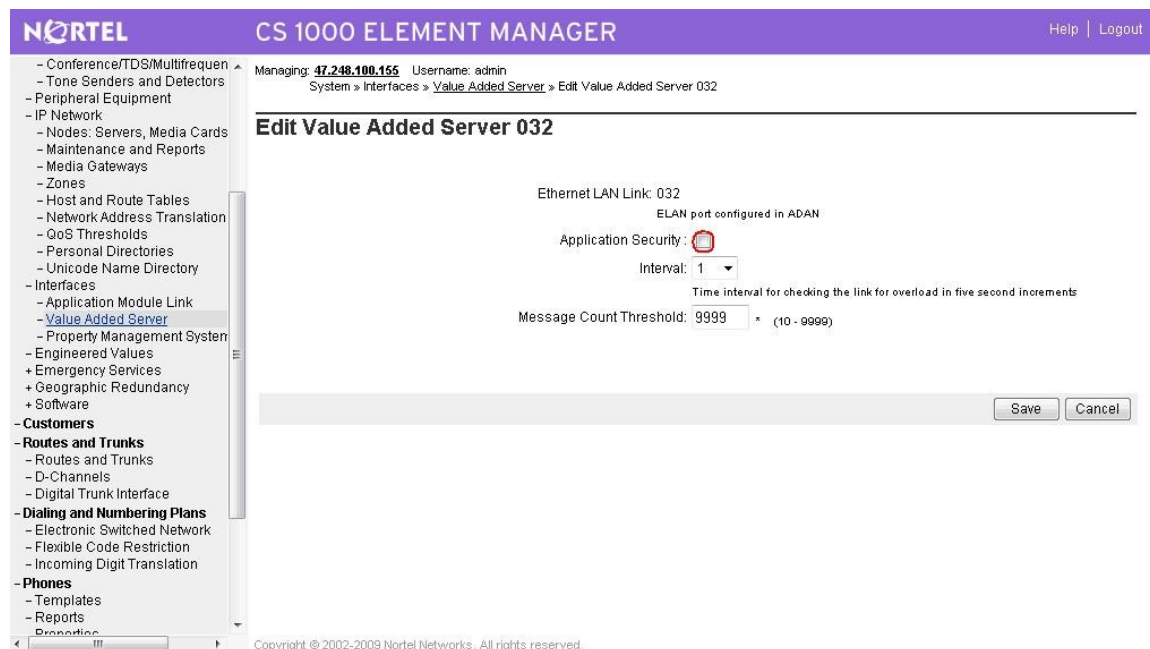


Figure 14 – Value Added Service for Application Module Link

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.

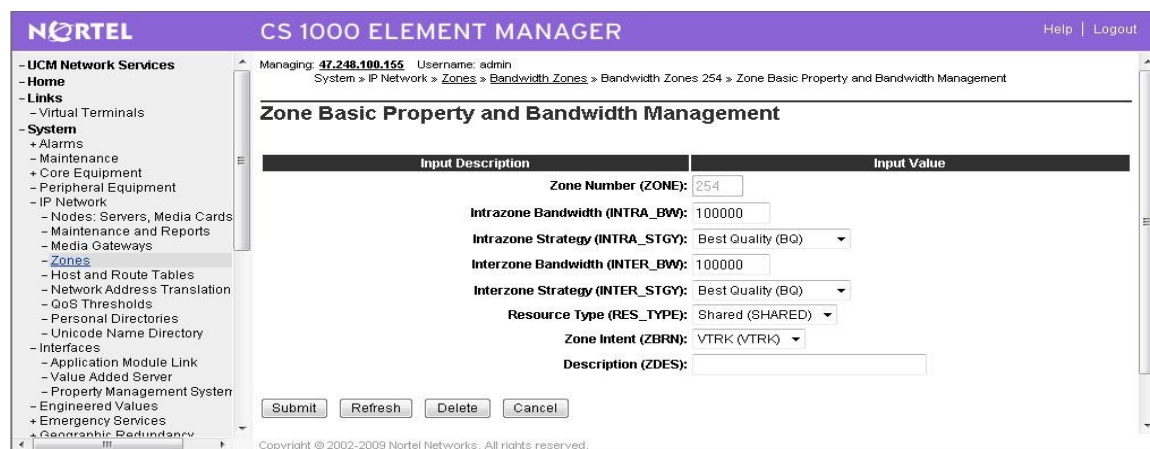


Figure 15 – Virtual Trunk Zone Configuration

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*.

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: **47.248.100.155** Username: admin
Routes and Trunks > Routes and Trunks > Customer 0, Route 30 Property Configuration

Customer 0, Route 30 Property Configuration

- Basic Configuration

Route data block (RDB) (TYPE)
 Customer number (CUST)
 Route number (ROUT)
 Designator field for trunk (DES)
 Trunk type (TKTP)
 Incoming and outgoing trunk (ICOG) Incoming and Outgoing (IAO)
 Access code for the trunk route (ACOD) Range: 0 - 255
 Trunk type M911P (M911P) ☐
 The route is for a virtual trunk route (VTRK) ☒
 Zone for codec selection and bandwidth management (ZONE) Range: 0 - 255
 Node ID of signaling server of this route (NODE) Range: 0 - 9999
 Protocol ID for the route (PCID)
 Integrated services digital network option (ISDN) ☒
 Mode of operation (MODE)
 D channel number (DCH) Range: 0 - 254
 Interface type for route (IFC)
 Private network identifier (PNI) Range: 0 - 32700
 Network calling name allowed (NCNA) ☒
 Network call redirection (NCRD) ☒
 Trunk route optimization (TRO) ☐
 Recognition of DTI2 ABCD FALT signal for ISL (FALT) ☐
 Channel type (CHTY)
 Call type for outgoing direct dialed TIE route (CTYP)
 Insert ESN access code (INAC) ☒
 Integrated service access route (ISAR) ☐
 Display of access prefix on CLID (DAPC) ☐
 Mobile extension route (MBXR) ☐

+ Basic Route Options
+ Network Options
+ General Options
+ Advanced Configurations

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**.

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Customer 0, Route 30, Trunk 1 Property Configuration

- Basic Configuration

Input Description	Input Value
Trunk data block (TYPE)	IPTr
Terminal Number (TN)	100 0 01 00
Designator field for trunk (DES)	SIPLINE
Extended Trunk (XTRK)	VTRK
Route number, Member number (RTMB)	30 1 *
Level 3 Signaling (SIGL)	[Dropdown]
Card Density (CDEN)	8D
Start arrangement Incoming (STRI)	Immediate (IMM) [Dropdown]
Start arrangement Outgoing (STRO)	Immediate (IMM) [Dropdown]
Trunk Group Access Restriction (TGAR)	0
Channel ID for this trunk (CHID)	1
Increase or decrease the member numbers (INC)	Increase channel and member number (YES) [Dropdown]
Class of Service (CLS)	[Edit]

+ Advanced Trunk Configurations

Save Delete Cancel

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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 RCB
 ICDD CDMD LLCN MCTD CLBD AUTU
 GPU A DPU A **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 CCB
 RTDD RBDD RBHD PGND FLXD FTTC DNDY DNO3
 MCBN
 FDS
 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

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.

		<p>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.</p> <p>xx = 11-14, 21-24, 31-34, 41-44 of the first, second, third and fourth Media Gateway, respectively.</p>
SIDE	NET	<p>NET = network, the controlling switch (applied for CS 1000 PSTN simulator)</p> <p>USR = slave to the controller (applied for CS 1000 system under test)</p>
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	DIV 2 3	Path Replacement Triggers
PR_TRIGS	CNG 2 3	
PR_TRIGS	CON 2 3	
PR_TRIGS	CTR2 2 3	

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**.

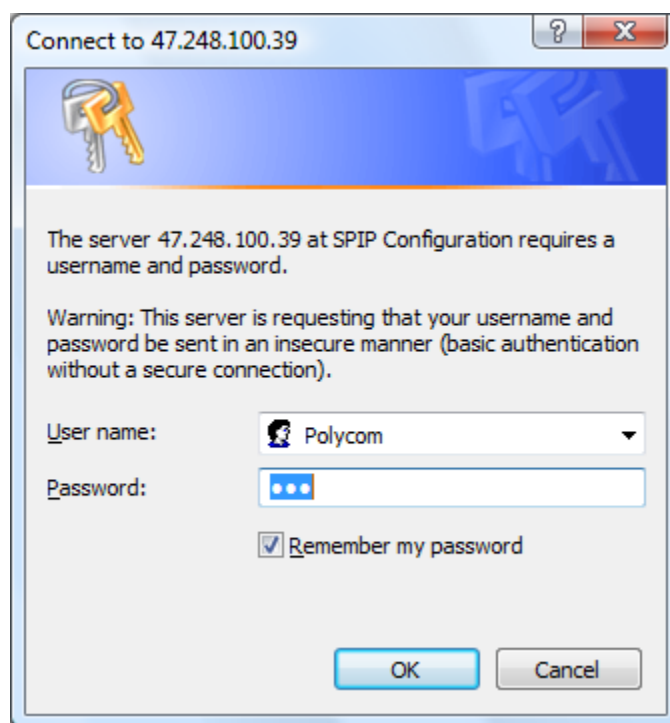


Figure 18 – Login Screen

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

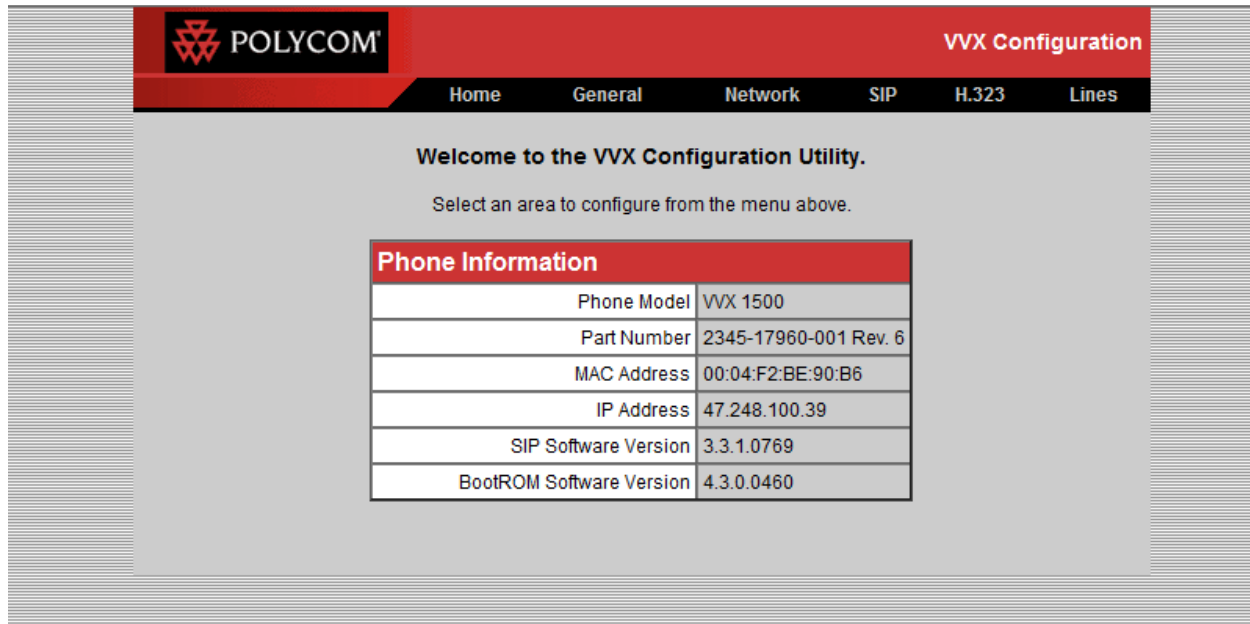


Figure 19 - Main Configuration Screen

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.

POLYCOM

VVX Configuration

HomeGeneralNetworkSIPH.323Lines

Line Parameters:

Line 1Line 2Line 3Line 4Line 5Line 6

Line 1

SIP Settings

SIP Protocol
☒ Enabled
☐ Disabled

H.323 Settings

H.323 Protocol
☐ Enabled
☒ Disabled

Gatekeeper Address

Port

Registration Time

Identification

Display Name

Address

Authentication User ID

Authentication Password

Label

Type
☐ Private
☒ Shared

Third Party Name

Number Of Line Keys

Calls Per Line

Figure 20 - SIP Configuration Screen

Identification	
Display Name	wx1500
Address	55116
Authentication User ID	55116
Authentication Password	••••
Label	
Type	<input type="radio"/> Private <input checked="" type="radio"/> Shared
Third Party Name	
Number Of Line Keys	1
Calls Per Line	24
SIP Server 1	
Address	dplab.com
Port	5070
Transport	DNSnaptr ▼
Expires	3600
Register	1
Retry Timeout	0
Retry Maximum Count	3
Line Seize Timeout	30

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.

POLYCOM

VVX Configuration

Home
General
Network
SIP
H.323
Lines

SIP Configuration Parameters:

Local Settings
Servers

Local Settings

SIP Protocol	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
Local SIP Port	5060
Calls Per Line Key	24
New SDP Type	<input type="radio"/> Enabled <input checked="" type="radio"/> Disabled
Live Communication Server Support	<input type="radio"/> Enabled <input checked="" type="radio"/> Disabled
Non Standard Line Seize	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
Digitmap	<div>[2-9]11 0T 011xxx.T </div> <div>[0-1][2-9]xxxxxxxx </div>
Digitmap Timeout	3 3 3 3 3
Remove End-Of-Dial Marker	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
Digitmap Impossible Match	0
top	Submit

Figure 21 – SIP Configuration Parameters

Servers	
Outbound Proxy	
Address	47.248.100.237
Port	5070
Transport	DNSnaptr ▼
Server 1	
Address	dplab.com
Port	5070
Transport	DNSnaptr ▼
Expires	3600
Register	1
Retry Timeout	0
Retry Maximum Count	3
Line Seize Timeout	30
Server 2	
Address	dplab.com
Port	5070
Transport	UDPonly ▼
Expires	3600
Register	1
Retry Timeout	0
Retry Maximum Count	3
Line Seize Timeout	30
top	Submit

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.

Call Diversion	
Disabled On Shared	<input checked="" type="radio"/> Yes <input type="radio"/> No
Diversion Contact	<input type="text"/>
On Specific Caller	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
Forward All	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
On Busy	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
Busy Contact	<input type="text" value="55555"/>
On No Answer	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
No Answer Timeout	<input type="text" value="1"/>
No Answer Contact	<input type="text"/>
On Do-Not-Disturb	<input type="radio"/> Enabled <input checked="" type="radio"/> Disabled
Do-Not-Disturb Contact	<input type="text"/>
Message Center	
Subscriber	<input type="text"/>
Callback Mode	Registration ▼
Callback Contact	<input type="text"/>
top	<input type="button" value="Submit"/>

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**.

The screenshot displays the Polycom VVX Configuration web interface. At the top, there is a red header bar with the Polycom logo on the left and 'VVX Configuration' on the right. Below the header is a navigation menu with tabs: Home, General, Network, SIP, H.323, and Lines. The 'General' tab is selected. Under the 'General' tab, there is a section titled 'General Configuration Parameters:' which contains a grid of links: User Preferences, Time, Audio Processing, Video Processing, Background, Sampled Audio, Microbrowser, and Logging. The 'Audio Processing' link is highlighted. Below this grid, the 'Audio Processing' configuration page is shown. It has a red header 'Audio Processing' and a sub-header 'Codec Preferences'. The main content is a table with two columns: the codec name and a dropdown menu for selection. The first three rows are highlighted in white, indicating they are selected or active: G.711Mu (set to 1), G.711A (set to 2), and G.729AB (set to 3). All other codecs are set to 'Not Used'.

Codec Preferences	
G.711Mu	1
G.711A	2
G.729AB	3
iLBC 13.33kbps	Not Used
iLBC 15.2kbps	Not Used
G.722	Not Used
G.722.1 16kbps	Not Used
G.722.1 24kbps	Not Used
G.722.1 32kbps	Not Used
G.722.1C 24kbps	Not Used
G.722.1C 32kbps	Not Used
G.722.1C 48kbps	Not Used
Siren14 24kbps	Not Used
Siren14 32kbps	Not Used
Siren14 48kbps	Not Used
Siren22 32kbps	Not Used

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 unhold 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      = PolycomV VX-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

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