



## **Avaya Solution & Interoperability Test Lab**

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# **Application Notes for Configuring the Polycom VVX 500/600 running UC software release 5.0.0.7403 with Avaya Communication Server 1000 Release 7.6 - Issue 1.0**

## **Abstract**

These Application Notes describe a solution for supporting interoperability between the Polycom VVX 500/600 running UC software release 5.0.0.7403 with Avaya Communication Server 1000 release 7.6. Emphasis of the testing was to verify voice calls of VVX 500/600 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 VVX 500/600 (hereafter referred to as VVX 500/600) used during the compliance testing. VVX 500/600 was tested with non-SIP and SIP telephones using CS1000 release 7.6. All the applicable telephony feature test cases of release 7.6 were executed on VVX 500/600, where applicable, to ensure the interoperability with CS1000.

## 2. General Test Approach and Test Results

The general test approach was to have VVX 500/600 register to the SIP line gateway of CS1000. Calls were then placed from CS1000 telephone clients/users to and from VVX 500/600. Other telephony features such as busy, hold, DTMF, transfer, conference, video (where applicable) 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 VVX 500/600 was able to interoperate with Avaya CS1000 SIP line system. The following areas were tested:

- Registration of VVX 500/600 to the CS1000 SIP line gateway.
- Call establishment of VVX 500/600 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 VVX 500/600 from PSTN.
- Video call between two VVX 500/600 phones.

**Note:** Based on the micro-processor type, VVX 500 and VVX 600 belong to the same family and therefore the test results of VVX 500 also holds good for VVX 600. During compliance testing, only VVX 500 was tested.

## 2.2. Test Results

The objectives outlined in **Section 2.1** were verified. VVX 500/600 was registered to CS1000 SIP line gateway successfully. Calls have been made between CS1000 telephones and VVX 500/600 with clear voice path. All executed test cases passed with the following observations,

- When performing 3-way conference where VVX 500/600 is a host of the conference and CS1000 telephones are the participants, as VVX 500/600 hangs up, the 2 CS1000 participant telephones are not able to establish the voice path. This issue is being investigated by the CS1000 team.
- Call Forward on Busy (CFB) has to be configured on CS1000 at the set level and not through the Polycom Web Configuration Utility. However Call Forward Unconditional (CFU) and Call Forward No Answer (CFNA) can be configured using the Polycom Web Configuration Utility.

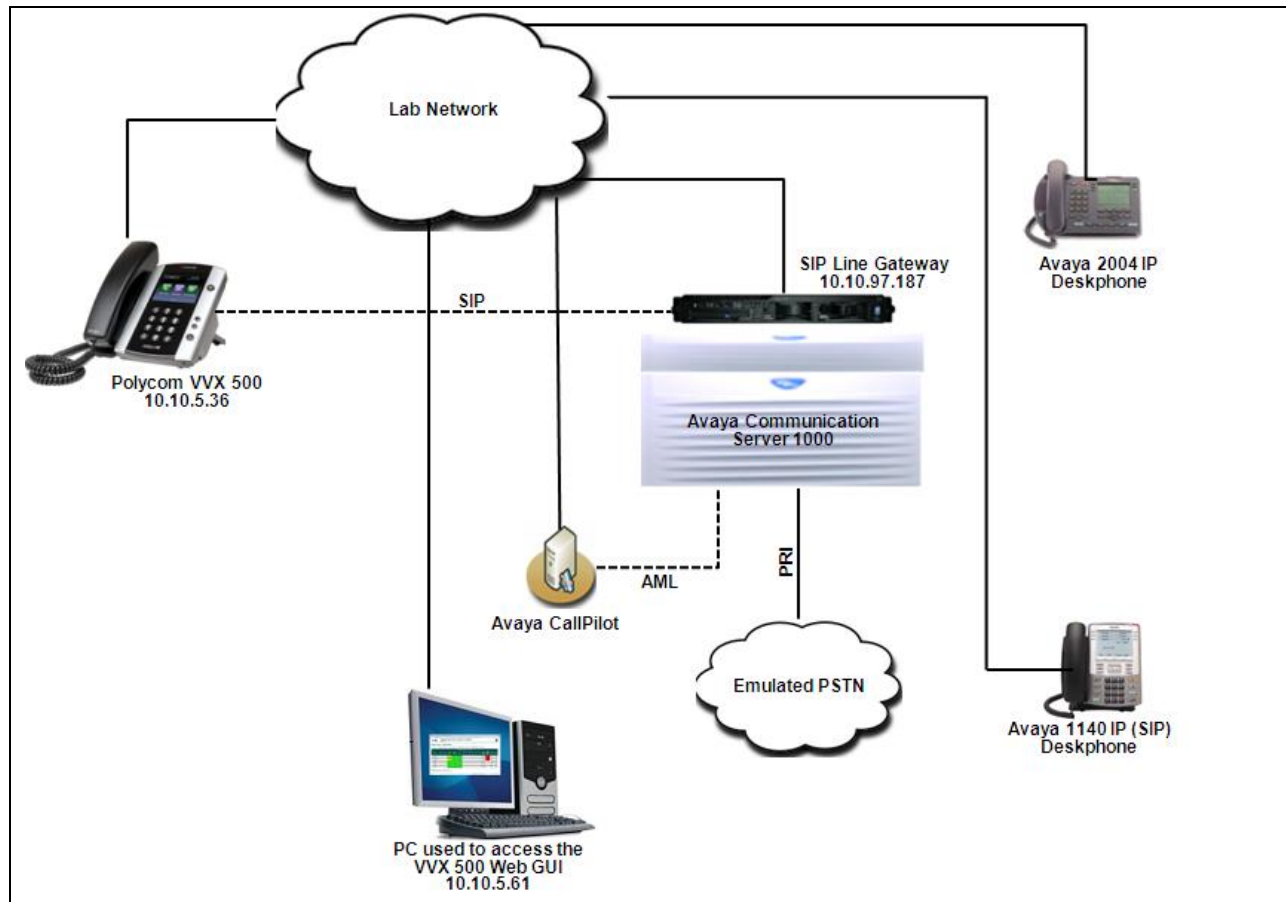
## 2.3. Support

Technical support for the Polycom VVX 500/600 phones 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 VVX 500/600 registers to CS1000 as a third party SIP endpoint via the SIP Line Gateway. Polycom Web Configuration Utility is used to manage the configuration of the VVX 500/600 phone.



**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	7.65P
Call Server	7.65.16
SIP Line Gateway	
Avaya Call Pilot	5.00.41
Avaya CS1000 IP (UNISTim) Phones:	
2007	0621C8L
2004P1	0602B76
Avaya CS1000 IP (SIP) Phone:	
1140	04.03.12
Polycom UC Software for VVX 500	5.0.0.7403
Polycom Web Configuration Utility	Windows XP Professional OS

## 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**.

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 7.6 Linux Base.
- Joined CS1000 Release 7.6 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

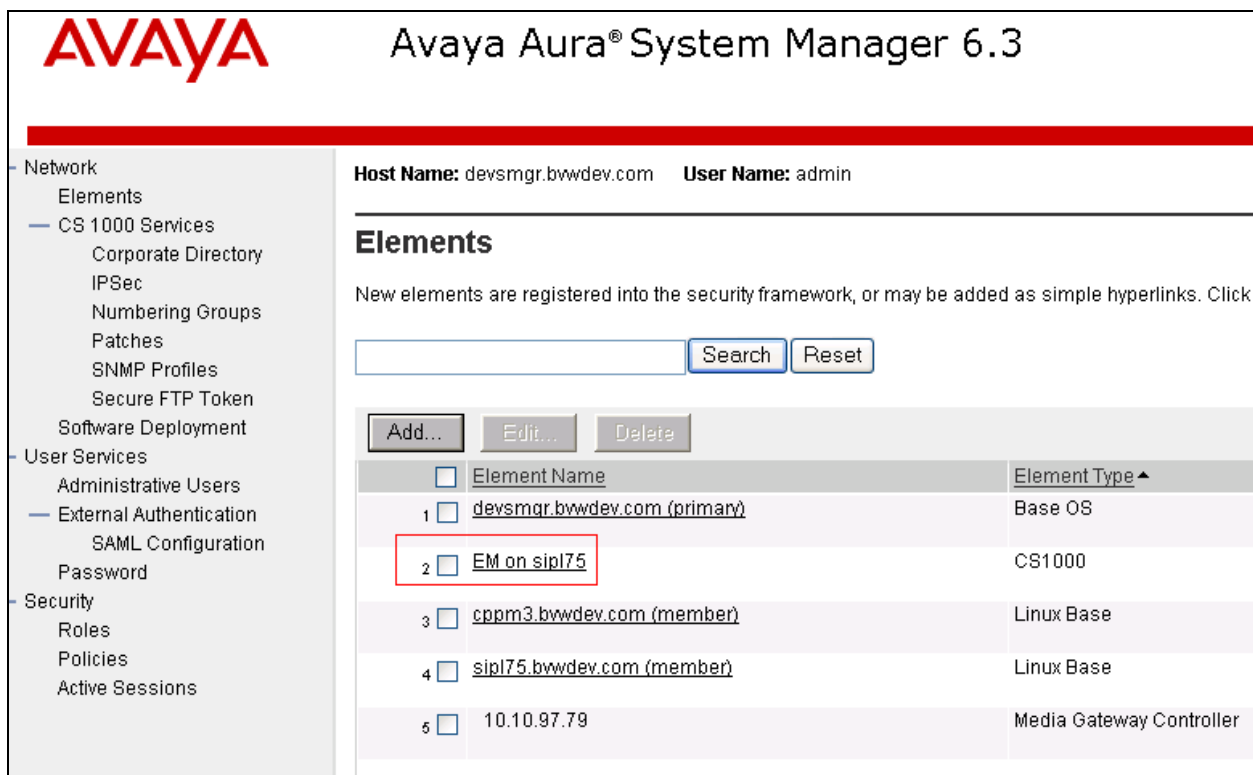
<http://www.avaya.com>.

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 Aura® System Manager (not shown) at <http://<IP Address or FQDN>> where <IP address or FQDN> is the IP address or FQDN for System Manager. Login with the appropriate username/password (not shown). From the System Manager dashboard navigate to **Elements** → **Communication Server 1000** (not shown). For more information on installing and configuring System Manager, see **Section 9**.

An **Elements** page for Communication Manager 1000 is seen as shown below. From this page, under the **Element Name** column, click the server name to navigate to Element Manager for that server.



The screenshot displays the Avaya Aura® System Manager 6.3 interface. The top header shows the Avaya logo and the title 'Avaya Aura® System Manager 6.3'. Below the header, the 'Host Name' is 'devsmgr.bwwdev.com' and the 'User Name' is 'admin'. The main content area is titled 'Elements' and contains a search bar with 'Search' and 'Reset' buttons. Below the search bar are 'Add...', 'Edit...', and 'Delete' buttons. A table lists the elements:

	Element Name	Element Type
1	<a href="#">devsmgr.bwwdev.com (primary)</a>	Base OS
2	<a href="#">EM on sip175</a>	CS1000
3	<a href="#">cppm3.bwwdev.com (member)</a>	Linux Base
4	<a href="#">sip175.bwwdev.com (member)</a>	Linux Base
5	10.10.97.79	Media Gateway Controller

The left sidebar shows the navigation menu with 'Elements' selected under 'Network'.

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

AVAYA

CS1000 Element Manager

UCM Network Services

Home

Links

Virtual Terminals

System

Alarms

Maintenance

Core Equipment

Peripheral Equipment

IP Network

Interfaces

Engineered Values

Emergency Services

Geographic Redundancy

Software

Customers

Routes and Trunks

Routes and Trunks

D-Channels

Digital Trunk Interface

Managing: 10.10.97.78 Username: admin

System Overview

System Overview

IP Address: 10.10.97.78

Type: Avaya Communication Server 1000E CPPM Linux

Version: 4121

Release: 765 P +

RS; Reviewed:  
SPOC 2/11/2014

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### 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 **26** was used.
- The rest of the values remain at default.
- Click on **Save** (not shown)

The screenshot displays the AVAYA CS1000 Element Manager web interface. On the left is a navigation menu with categories: UCM Network Services, Home, Links, System (with sub-items like Alarms, Maintenance, Core Equipment, etc.), and Customers. The main content area is titled 'SIP Line Service'. At the top of this area, it says 'Managing: 10.10.97.78 Username: admin' and shows a breadcrumb trail: 'Customers > Customer 00 > Customer Details > SIP Line Service'. The configuration section includes a checked checkbox for 'SIP Line Service', a text field for 'User agent DN prefix' containing the value '26', and a section for 'Optional features' with a checked checkbox for 'Nortel Multimedia'. A note '\*Required Value' is visible at the bottom right of the configuration area.

## 5.4. Add a new SIP Line Telephony Node

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

- **Node ID** text box: 512; this is the node ID of SIP Line server.
- **Call Server IP Address** text box: 10.10.97.78
- **Node IP Address** text box: 10.10.97.187; this is the IP address to which a SIP endpoint will register.
- **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.

**AVAYA** **CS1000 Element Manager**

Managing: 10.10.97.78 Username: admin  
System » IP Network » IP Telephony Nodes » New IP Telephony Node

### 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: 512 * (0-9999)	TLAN address type: <input checked="" type="radio"/> IPv4 only <input type="radio"/> IPv4 and IPv6
Call server IP address: 10.10.97.78 *	
<b>Embedded LAN (ELAN)</b> Gateway IP address: 10.10.97.65 * Subnet mask: 255.255.255.192 *	<b>Telephony LAN (TLAN)</b> Node IPv4 address: 10.10.97.187 * Subnet mask: 255.255.255.192 *
Node IPv6 address:	

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

\* Required Value.

**Next >** **Cancel**

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 screenshot displays the AVAYA CS1000 Element Manager web interface. The left sidebar contains a navigation tree with categories like UCM Network Services, Home, Links, System, and IP Network. The main content area is titled 'New IP Telephony Node (ID:512)' and shows 'Step 2: Associate required signaling servers for SIP Line services.' It includes a table with columns for Hostname, Type, Deployed Applications, ELAN IP, TLAN IPv4, TLAN IPv6, and Role. Above the table are buttons for 'Select to add', 'Add', 'Remove', and 'Make Leader'. Below the table are navigation buttons: '< Back', 'Next >', and 'Cancel'.

**AVAYA** CS1000 Element Manager

Managing: 10.10.97.78 Username: admin  
System » IP Network » IP Telephony Nodes » New IP Telephony Node

### New IP Telephony Node (ID:512)

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.

Select to add Add Remove Make Leader Print Refresh

<input type="checkbox"/>	Hostname▲	Type	Deployed Applications	ELAN IP	TLAN IPv4	TLAN IPv6	Role
Select from the list above and click Add to associate servers with this node. Selected servers must have identical application deployments.							

< Back Next > Cancel

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 **sipl75.com** was used.
- Enter the **SLG endpoint name**. During compliance testing **sipl75** was used.
- Enter the **SLG Group ID**. During compliance testing **512** was used.
- Retain default values for all other fields.

**Note:** that SIP Line Gateway is configured by default to allow both UDP and TCP transport protocols.

**AVAYA** **CS1000 Element Manager**

Managing: 10.10.97.78 Username: admin  
System » IP Network » IP Telephony Nodes » Node Details » SIP Line Configuration

**Node ID: 512 - SIP Line Configuration Details**

General | SIP Line Gateway Settings | SIP Line Gateway Service

SIP Line Gateway Application: ☒ Enable gateway service on this node

**General**

SIP domain name:  \*

SLG endpoint name:

SLG Group ID:

SLG Local Sip port:  (1 - 65535)

SLG Local Tls port:  (1 - 65535)

**Virtual Trunk Network Health Monitor**

☐ Monitor IP addresses (listed below)  
Information will be captured for the IP addresses listed below.

Monitor IP:  Add

Monitor addresses:  Remove

**SIP Line Gateway Settings**

Security policy:

Number of byte re-negotiation:

Options: ☐ Client authentication

\* Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved.

Save Cancel

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.

**AVAYA CS1000 Element Manager**

Managing: 10.10.97.78 Username: admin  
System » IP Network » IP Telephony Nodes » Node Details » SIP Line Configuration

**Node ID: 512 - SIP Line Configuration Details**

General | SIP Line Gateway Settings | SIP Line Gateway Service

**SIP Line Gateway Service**

Branch / GR Office Settings:

SLG role: MO  
SLG mode: S1/S2

MO SLG IPv4 address: 0.0.0.0  
The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"

MO SLG IPv6 address:

MO SLG port: 5060 (1 - 65535)

MO SLG transport: TCP

GR SLG IPv4 address: 0.0.0.0  
The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"

GR SLG IPv6 address:

GR SLG port: 5070 (1 - 65535)

\* Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved.

Save Cancel

- 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.

**AVAYA CS1000 Element Manager**

Managing: 10.10.97.78 Username: admin  
System » IP Network » IP Telephony Nodes » Node Saved

**Node Saved**

Node ID: 512 has been saved on the call server.

The new configuration must also be transferred to associated servers and media cards.

**Transfer Now...** You will be given an option to select individual servers, or transfer to all.

**Show Nodes** You may initiate a transfer manually at a later time.

- The **Synchronize Configuration Files (Node ID <512>)** 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.

**AVAYA CS1000 Element Manager**

Managing: 10.10.97.78 Username: admin  
System » IP Network » IP Telephony Nodes » Synchronize Configuration Files

**Synchronize Configuration Files (Node ID <512>)**

Note: Select components to synchronize their configuration files with call server data. This process transfers server INI files to selected components, and requires a restart\* of applications on affected server(s) when complete.

[Print](#) | [Refresh](#)

<input checked="" type="checkbox"/>	Hostname	Type	Applications	Synchronization Status
<input checked="" type="checkbox"/>	sipl75	Signaling_Server	LTPS, Gateway, PD, Presence Publisher, IP Media Services	Sync required

\* Application restart is only required for initial system configuration or if changes have been made to general LAN configurations, SNTP settings, SIP and H323 Gateway settings, network connectivity related parameters like ports and IP address, enabling or disabling services, or adding or removing application servers.

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

**AVAYA CS1000 Element Manager**

Managing: 10.10.97.78 Username: admin  
System » IP Network » IP Telephony Nodes » Synchronize Configuration Files

**Synchronize Configuration Files (Node ID <512>)**

Note: Select components to synchronize their configuration files with call server data. This process transfers server INI files to selected components, and requires a restart\* of applications on affected server(s) when complete.

[Print](#) | [Refresh](#)

<input type="checkbox"/>	Hostname	Type	Applications	Synchronization Status
<input type="checkbox"/>	sipl75	Signaling_Server	LTPS, Gateway, PD, Presence Publisher, IP Media Services	Synchronized

\* Application restart is only required for initial system configuration or if changes have been made to general LAN configurations, SNTP settings, SIP and H323 Gateway settings, network connectivity related parameters like ports and IP address, enabling or disabling services, or adding or removing application servers.

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

## 5.5. Create a D-Channel for SIP Line

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

**AVAYA**

**CS1000 Element Manager**

**- UCM Network Services**

**- Home**

**- Links**

- Virtual Terminals

**- System**

- + Alarms
- Maintenance
- + Core Equipment
- Peripheral Equipment
- + IP Network
- + Interfaces
- 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

Managing: **10.10.97.78** Username: admin  
Routes and Trunks > D-Channels

**D-Channels**

**Maintenance**

- [D-Channel Diagnostics](#) (LD 96)
- [Network and Peripheral Equipment](#) (LD 32, Virtual D-Channels)
- [MSDL Diagnostics](#) (LD 96)
- [TMDI Diagnostics](#) (LD 96)
- [D-Channel Expansion Diagnostics](#) (LD 48)

**Configuration**

Choose a D-Channel Number:  and type:

- Channel: 1	Type: DCH	Card Type: DCIP	Description: SIP	<input type="button" value="Edit"/>
- Channel: 2	Type: DCH	Card Type: TMDI	Description: ToCM	<input type="button" value="Edit"/>
- Channel: 3	Type: DCH	Card Type: DCIP	Description: SIPLine	<input type="button" value="Edit"/>

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Screen below shows the **D-Channels xx Property Configuration** page 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.

**AVAYA**  
 - UCM Network Services  
 - Home  
 - Links  
   - Virtual Terminals  
 - System  
   + Alarms  
   + Maintenance  
   + Core Equipment  
   + Peripheral Equipment  
   + IP Network  
   + Interfaces  
   - 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  
   + Logs and reports  
 - Security  
   + Passwords  
   + Policies  
   + Login Options

Managing: **10.10.97.78** Username: admin  
 Routes and Trunks > D-Channels > D-Channels 3 Property Configuration  

### D-Channels 3 Property Configuration

- Basic Configuration

Input Description	Input Value
Action Device And Number (ADAN):	DCH
D channel Card Type:	DCIP
Designator:	SIPLine
Recovery to Primary:	<input type="checkbox"/>
PRI loop number for Backup D-channel:	
User:	Integrated Services Signaling Link Dedicated (ISLD)
Interface type for D-channel:	Meridian Meridian1 (SL1)
Country:	ETS 300 =102 basic protocol (ETSI)
D-Channel PRI loop number:	
Primary Rate Interface:	<input type="text"/> <a href="#">more PRI</a>
Secondary PRI2 loops:	<input type="text"/>
Meridian 1 node type:	Slave to the controller (USR)
Release ID of the switch at the far end:	7
Central Office switch type:	100% compatible with Bellcore standard (STD)
Integrated Services Signaling Link Maximum:	4000 Range: 1 - 4000
Signalling server resource capacity:	3700 Range: 0 - 3700

+ Basic options (BSCOPT)

+ Advanced options (ADVOPT)

+ Feature Packages



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.

**AVAYA CS1000 Element Manager**

Help | Logout

**UCM Network Services**

- Home
- Links
  - Virtual Terminals
- System
  - + Alarms
  - + Maintenance
  - + Core Equipment
  - + Peripheral Equipment
  - + IP Network
  - + Interfaces
  - + 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
  - + Logs and reports
- Security
  - + Passwords
  - + Policies
  - + Login Options

Rerouting requests processed using integer value (DV2I) ☐  
 Rerouting requests processed using object identifier (DV2O) ☐  
 Diversion info. sent. rerouting requests processed (DV3I) ☐  
 EuroISDN - div. info sent. rerouting req. processed (DV3O) ☐  
 Call transfer notification and invocation to EuroISDN (ECTO) ☐  
 Malicious call identification (MCID) ☐  
 MCDN QSIG conversion (MQC) ☐  
 Remote D-channel is on a MSDL card (MSL) ☐  
**Message waiting interworking with DMS-100 (MWI) ☒**  
 Network access data (NAC) ☐  
 Network call trace supported (NCT) ☐  
 Network name display method 1 (ND1) ☐  
**Network name display method 2 (ND2) ☒**  
 Network name display method 3 (ND3) ☐  
 Name display - integer ID coding (NDI) ☐  
 Name display - object ID coding (NDO) ☐  
 Path replacement uses integer values (PRI) ☐  
 Path replacement uses object identifier (PRO) ☐  
 Release Link Trunks over IP (RLTI) ☐  
 Remote virtual queuing (RVQ) ☐  
 Trunk anti-tromboning operation (TAT) ☐  
 User to user service 1 (UUS1) ☐  
 NI-2 name display option. (NDS) ☐  
 Message waiting indication using integer values (QMWI) ☐  
 Message waiting indication using object identifier (QMWVO) ☐  
 User to user signalling (UUI) ☐

Return - Remote Capabilities Cancel

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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.

The screenshot shows the 'New Application Module Link' page in the CS1000 Element Manager. The left sidebar contains a tree view with 'Application Module Link' selected. The main area has a form with the following fields: 'Port number' (value: 32, range: 10 - 127), 'AML over ELAN' (checkbox, checked), 'Description' (value: SIPL), 'Link control system parameters' (checkbox, unchecked), and 'Maximum octets' (value: 512, range: 10 - 127). A red box highlights the 'Port number' and 'Description' fields. At the bottom, there is a 'Save' button and a 'Cancel' button.

## 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.

The screenshot shows the 'Ethernet Link' page in the CS1000 Element Manager. The left sidebar contains a tree view with 'Value Added Server' selected. The main area has a form with the following fields: 'Value added server ID' (value: 32, range: 10 - 127), 'Ethernet LAN Link' (dropdown menu, value: ELAN port configured in ADAN), 'Application security' (checkbox, unchecked), 'Interval' (value: 1, range: 1 - 10), 'Message count threshold' (value: 9999, range: 10 - 9999), and 'Time interval for checking the link for overload in five second increments' (checkbox, unchecked). A red box highlights the 'Value added server ID' and 'Ethernet LAN Link' fields. At the bottom, there is a 'Save' button and a 'Cancel' button.

## 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 as shown in the screen below.

The screenshot displays the 'Zone Basic Property and Bandwidth Management' configuration page in the CS1000 Element Manager. The interface includes a left-hand navigation pane with categories like 'UCM Network Services', 'Home', 'Links', 'System', and 'Interfaces'. The main content area features a table with two columns: 'Input Description' and 'Input Value'. The 'Zone Number (ZONE)' field is set to 1, and the 'Zone Intent (ZBRN)' dropdown is set to 'MO (MO)'. Other fields like 'Intrazone Bandwidth (INTRA\_BW)' and 'Interzone Bandwidth (INTER\_BW)' are set to 1000000. The 'Resource Type (RES\_TYPE)' is set to 'Shared (SHARED)'. At the bottom, there are 'Save' and 'Cancel' buttons, and a note indicating that the Zone Number is a required value.

Input Description	Input Value
Zone Number (ZONE):	1 (1 - 8000)
Intrazone Bandwidth (INTRA_BW):	1000000 (0 - 10000000)
Intrazone Strategy (INTRA_STGY):	Best Quality (BQ)
Interzone Bandwidth (INTER_BW):	1000000 (0 - 10000000)
Interzone Strategy (INTER_STGY):	Best Quality (BQ)
Resource Type (RES_TYPE):	Shared (SHARED)
Zone Intent (ZBRN):	MO (MO)
Description (ZDES):	

## 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 where the route will belong.

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

- **Route Number (ROUT):** 3; 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):** 8003; 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):** 512; this is the node ID of the SIP Line.
- **Protocol ID for the route (PCID):** SIP Line (SIPL).
- **Integrated services digital network option (ISDN):** Checked.
- **Mode of operation (MODE):** Route uses ISDN Signaling Link (ISLD).
- **D channel number (DCH):** 3; the D-channel number that was created in **Section 5.5**.
- **Interface type for route (IFC):** Meridian M1 (SL1).
- **Private network identifier (PNI):** 00001; 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):** Unknown Call type (UKWN).
- **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

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: 10.10.97.78 Username: admin

Routes and Trunks > Routes and Trunks > Customer 0, Route 3 Property Configuration

Customer 0, Route 3 Property Configuration

Basic Configuration

Route data block (RDB) (TYPE):

RDB

Customer number (CUST):

00

Route number (ROUT):

3

Designator field for trunk (DES):

SIPLINE

Trunk type (TKTP):

TIE

Incoming and outgoing trunk (ICOG):

Incoming and Outgoing (IAO)

Access code for the trunk route (ACOD):

8003

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):

512

(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):

3

(0 - 254)

Interface type for route (FC):

Meridian M1 (SL1)

Private network identifier (PNI):

00001

(0 - 32700)

Network calling name allowed (NCNA):

☒

Network call redirection (NCRD):

☒

Trunk route optimization (TRO):

☐

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

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Flexible Code Restriction

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Phones

Templates

Reports

Views

Lists

Properties

Migration

Tools

Backup and Restore

Date and Time

Logs and reports

Basic Route Options

Network Options

General Options

Advanced Configurations

Trunk route optimization (TRO):

☐

Recognition of DT12 ABCD FALT signal for ISL (FALT):

☐

Channel type (CHTY):

B-channel (BCH)

Call type for outgoing direct dialed TIE route (CTYP):

Unknown Call type (UKAWN)

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)

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 the route that 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.

AVAYA CS1000 Element Manager

Managing: 10.10.97.28 Username: admin  
Routes and Trunks > Routes and Trunks > Customer 0, Route 3, Trunk 1 Property Configuration

Customer 0, Route 3, Trunk 1 Property Configuration

- Basic Configuration

Multiple trunk input number: 32  
Auto increment member number: ☒  
Trunk data block: TIE trunk data block (TIE)  
Terminal number: 100 0 01 00  
Designator field for trunk: SIPL  
Extended trunk: VTRK  
Member number: 1  
Level 3 Signaling: Octal Density (8D)  
Card density: Octal Density (8D)  
Start arrangement Incoming: Immediate (IMM)  
Start arrangement Outgoing: Immediate (IMM)  
Trunk group access restriction: 1  
Channel ID for this trunk: 1  
Network music: ☐  
Class of Service: Edit

+ Advanced Trunk Configurations

\* Required value.

Save Cancel

## 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 104 0 01 04 → Terminal number on which the set is configured.
DATE
PAGE
DES
DES POLY → Description of the phone.
TN 104 0 01 04 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 54504 → SIP phone user ID.
NDID 512 → 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 RCB
      ...
CPND_LANG ENG
```

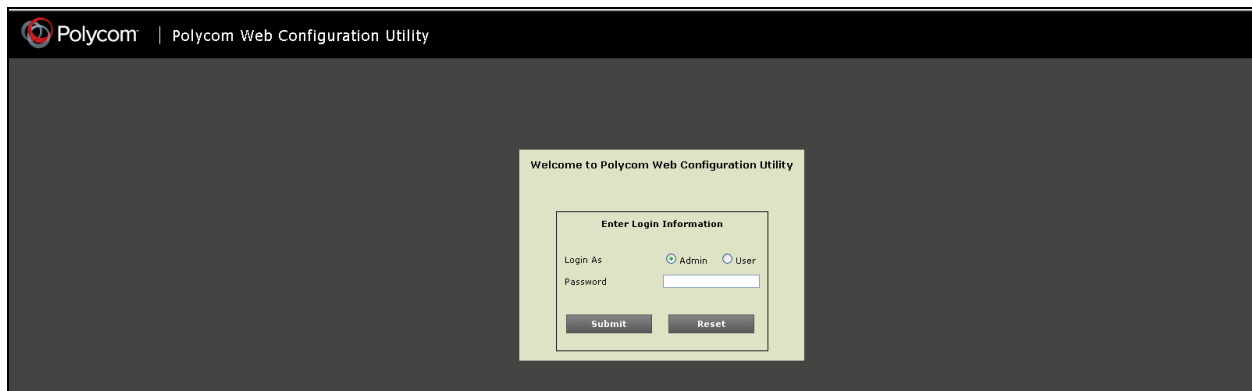
```
RCO 0
HUNT 58888 → Hunt DN
LHK 0
.
.
DNDR 0
KEY 00 SCR 54504 0      MARP → Extension number for the SIP phone
      CPND
      CPND_LANG ROMAN
      NAME Polycom, VVX → CLID information
      XPLN 13
      DISPLAY_FMT FIRST, LAST
01 HOT U 2654504 MARP 0
02
03
```



## 6. Polycom Web Configuration Utility

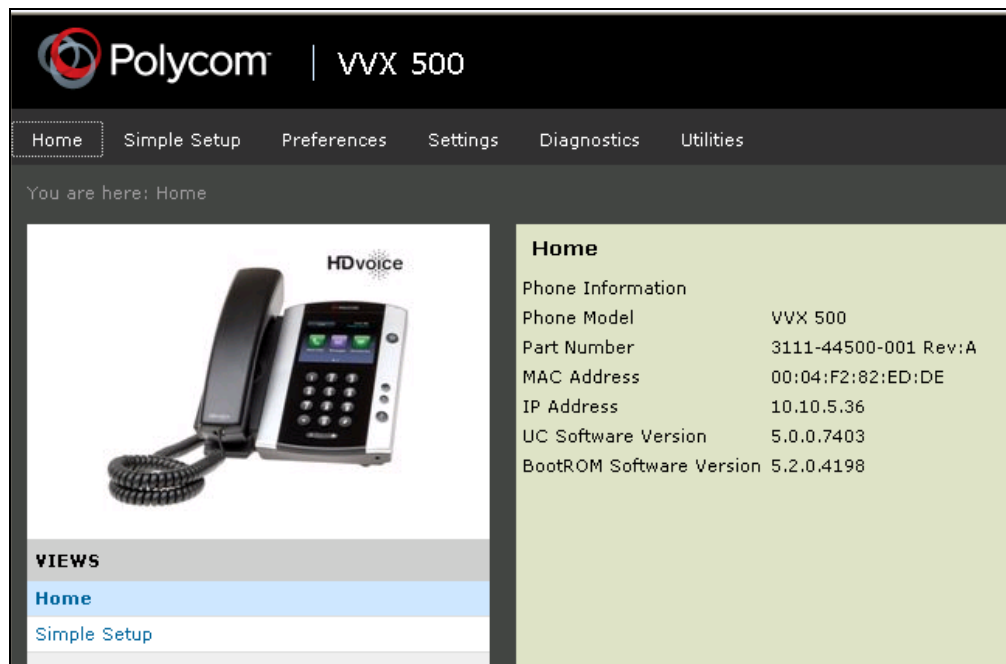
This section shows how to log in to the home page of Polycom Web Configuration Utility that is required to configure the VVX 500/600 phone.

Find the IP address assigned to the VVX 500/600 phone and type it 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**.



The image shows the Polycom Web Configuration Utility login page. At the top, there is a header with the Polycom logo and the text "Polycom Web Configuration Utility". Below the header, there is a central box titled "Welcome to Polycom Web Configuration Utility". Inside this box, there is a form titled "Enter Login Information". The form has two radio buttons: "Admin" (selected) and "User". Below the radio buttons, there is a "Password" field with a text input box. At the bottom of the form, there are two buttons: "Submit" and "Reset".

Click **Submit**, the homepage of the Polycom VVX 500 is shown below.



The image shows the Polycom VVX 500 homepage. At the top, there is a header with the Polycom logo and the text "VVX 500". Below the header, there is a navigation bar with the following links: Home, Simple Setup, Preferences, Settings, Diagnostics, and Utilities. Below the navigation bar, there is a sub-header that says "You are here: Home". The main content area is divided into two columns. The left column features a large image of a Polycom VVX 500 phone with the "HDvoice" logo. Below the image, there is a section titled "VIEWS" with two links: "Home" (highlighted in blue) and "Simple Setup". The right column is titled "Home" and contains a section titled "Phone Information" with the following details:

Phone Information	
Phone Model	VVX 500
Part Number	3111-44500-001 Rev:A
MAC Address	00:04:F2:82:ED:DE
IP Address	10.10.5.36
UC Software Version	5.0.0.7403
BootROM Software Version	5.2.0.4198

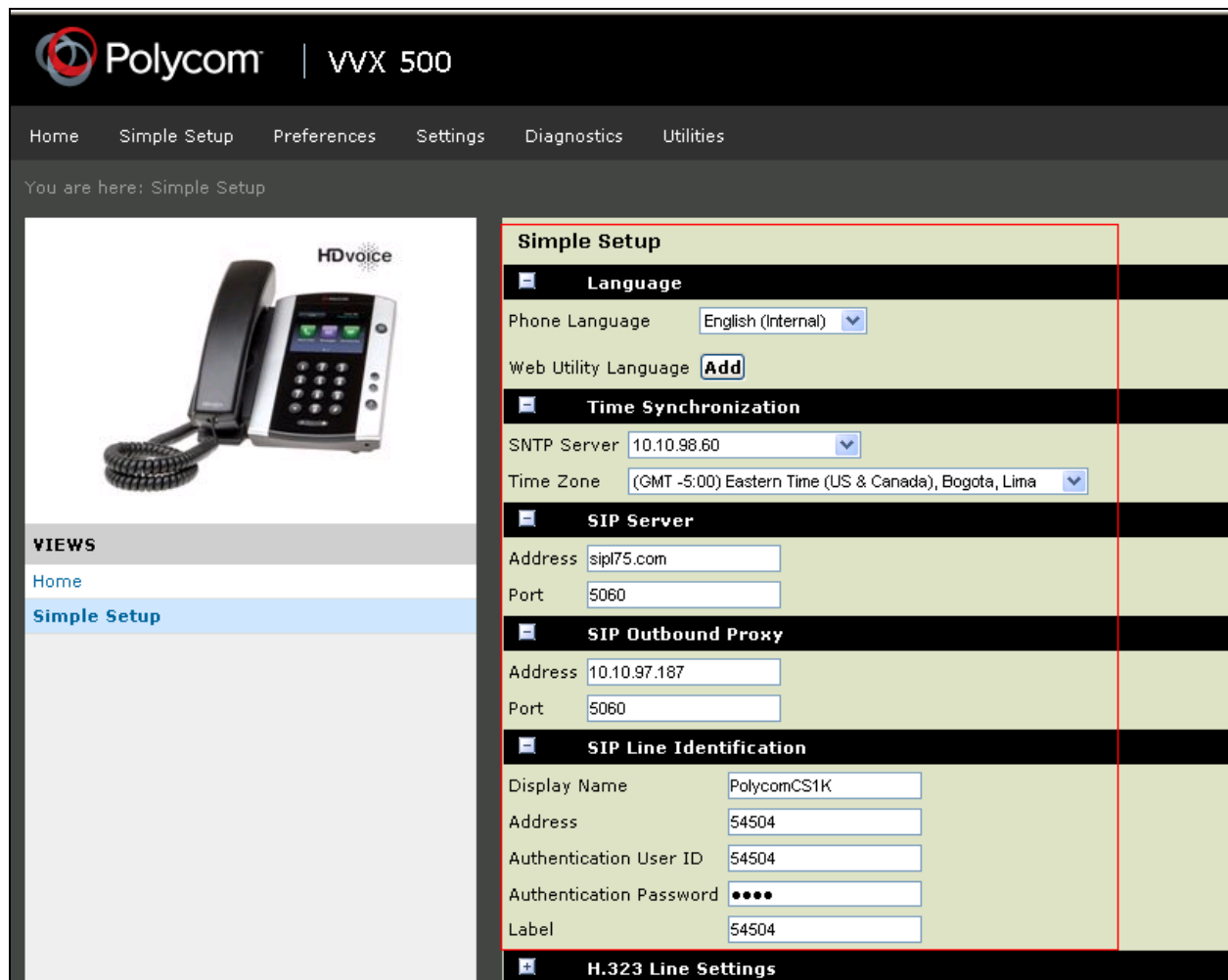
## 6.1. Configure the Lines for Polycom VVX 500/600

This section shows how to configure the Polycom VVX 500/600 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:** sip175.com and **Port:** 5060; configured in **Section 5.4**.
- Under **SIP Outbound Proxy** section, **Address:** 10.10.97.187 and **Port:** 5060; configured in **Section 5.4**.
- Under the **SIP Line Identification** section, **Display Name:** an appropriate name, **Address:** 54504, **Authentication User ID:** 54504 and **Authentication Password:** 1234; configured in **Section 5.11**.

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



The screenshot displays the Polycom VVX 500 configuration interface. The top navigation bar includes links for Home, Simple Setup, Preferences, Settings, Diagnostics, and Utilities. The 'Simple Setup' page is active, showing a list of views on the left: Home and Simple Setup. The main content area is divided into sections for configuration. A red box highlights the 'SIP Line Identification' section, which contains the following fields:

SIP Line Identification	
Display Name	PolycomCS1K
Address	54504
Authentication User ID	54504
Authentication Password	••••
Label	54504

Other visible sections include:

- Language:** Phone Language (English (Internal)), Web Utility Language (Add).
- Time Synchronization:** SNTP Server (10.10.98.60), Time Zone ((GMT -5:00) Eastern Time (US & Canada), Bogota, Lima).
- SIP Server:** Address (sip175.com), Port (5060).
- SIP Outbound Proxy:** Address (10.10.97.187), Port (5060).
- H.323 Line Settings:** (Section header visible).

## 6.2. SIP Settings

This section shows how to set SIP parameters for Polycom VVX 500/600.

On the homepage of Polycom VVX 500/600, navigate to menu **Settings** → **SIP** (not shown), **SIP** screen is shown below. Enter the following values and retain rest at default.

- Under the **Outbound Proxy** section, **Address**: 10.10.97.187 and **Port**: 5060; configured in **Section 5.4.Transport**: UDPOOnly.
- Under the **Server1** section, **Address**: sip175.com and **Port**: 5060; configured in **Section 5.4.Transport**: UDPOOnly.

Retain default values for rest of the fields. Click on **Save**.

The screenshot displays the Polycom VVX 500 web interface. The top navigation bar includes 'Home', 'Simple Setup', 'Preferences', 'Settings', 'Diagnostics', and 'Utilities'. The breadcrumb trail shows 'You are here: Settings > SIP'. On the left, a sidebar lists various settings categories, with 'SIP' selected. The main content area is titled 'SIP' and contains three sections: 'Local Settings', 'Outbound Proxy', and 'Server 1'. The 'Outbound Proxy' and 'Server 1' sections are highlighted with a red box, indicating the fields to be configured. The 'Outbound Proxy' section includes fields for Address (10.10.97.187), Port (5060), and Transport (UDPOOnly). The 'Server 1' section includes fields for Special Interop (Standard), Address (sip175.com), Port (5060), Transport (UDPOOnly), Expires (s) (3600), Register (Yes), and Retry Timeout (ms) (0). The 'Local Settings' section includes fields for SIP Protocol (Enable), Local SIP Port (0), Calls Per Line Key (24), Enable Roaming buddies for (None), New SDP Type (Disable), Live Communication Server Support (Disable), Non Standard Line Seize (Enable), Digitmap ([2-9]11|0T|011xxx.T| [0-1] [2-9]xxxxxxxx| [2-9] xxxxxxxx| [2-9] xxxT|\*\*x.T), Digitmap Timeout (3|3|3|3|3|3), Remove End-of-Dial Marker (Enable), and Digitmap Impossible Match (0).

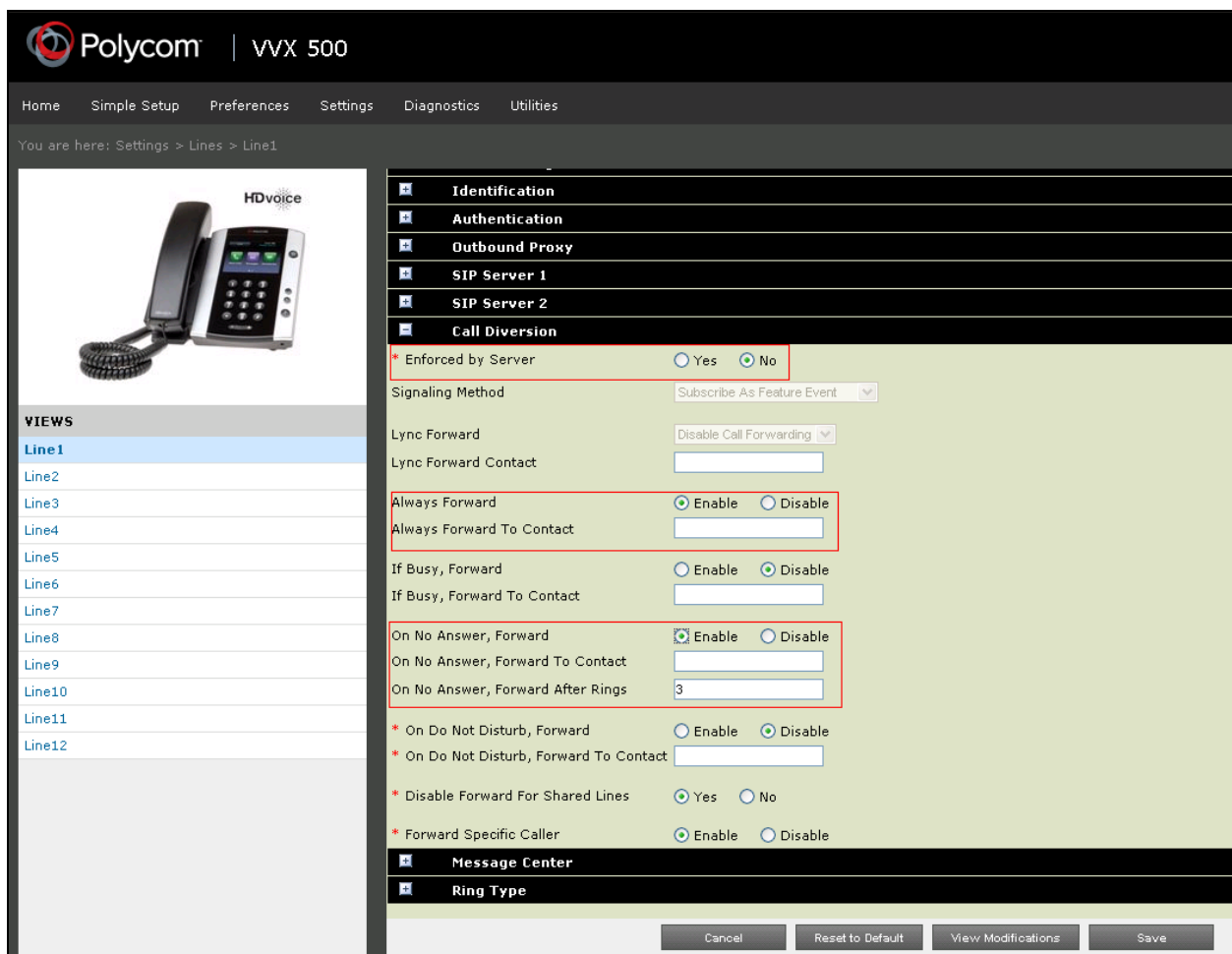
### 6.3. Local Call Forward Settings

This section shows how to set up call forward settings for Polycom VVX 500/600.

On the homepage of Polycom VVX 500/600, 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, ensure that the **Enforced by Server** radio button is **No**.
- **Always Forward**: Enable and configure an appropriate Directory Number (DN) for the **Always Forward To Contact** field.
- **On No Answer, Forward**: Enable and configure an appropriate Directory Number (DN) for the **On No Answer, Forward to Contact** field. Configure an appropriate value on the **On No Answer, forward After Rings** field.

Click on **Save** (not shown). As mentioned in **Section 2.2, If Busy, Forward** option does not function if configured here and has to be configured on the CS1000 at the set level.



## 6.4. Codec Settings

On the homepage of Polycom VVX 500/600, navigate to menu **Settings** → **Audio Codec Priority** (not shown). Select the codec list as shown below. Click **Save**.

The screenshot displays the Polycom VVX 500 web interface. The top navigation bar includes links for Home, Simple Setup, Preferences, Settings, Diagnostics, and Utilities. The breadcrumb trail indicates the current location: Settings > Codec Priorities. On the left, a sidebar lists various system views, with 'Codec Priorities' highlighted. The main content area is titled 'Codec Priorities' and features a section for 'Audio Codec Priority'. This section is divided into two columns: 'Unused' and 'In use'. The 'Unused' column contains a list of codecs, with 'Siren22 (32 kbps)' selected. The 'In use' column shows a list of codecs, with 'G.722' highlighted. A red box is drawn around the 'G.722' entry in the 'In use' column. Below the codec lists, a note states: 'Only codecs with a white background are supported on this platform.' At the bottom of the page, there are four buttons: 'Cancel', 'Reset to Default', 'View Modifications', and 'Save'.

**Polycom | VVX 500**

Home Simple Setup Preferences Settings Diagnostics Utilities

You are here: Settings > Codec Priorities

**HDvoice**

**VIEWS**

- Microbrowser
- Logging
- Applications
- Codec Priorities**
- Provisioning Server
- Syslog
- Paging/PTT Configuration
- SIP
- H.323
- Lines
- Power Saving
- Change Password
- Phone Lock

**Codec Priorities**

**Audio Codec Priority**

Unused:

- ILBC (13.33 kbps)
- ILBC (15.2 kbps)
- G.722.1 (16 kbps)
- G.722.1 (24 kbps)
- G.722.1 (32 kbps)
- G.722.1C (24 kbps)
- G.722.1C (32 kbps)
- G.722.1C (48 kbps)
- Siren14 (24 kbps)
- Siren14 (32 kbps)
- Siren22 (32 kbps)**

In use:

- Siren22 (64 kbps)
- Siren14 (48 kbps)
- G.711Mu
- G.711A
- G.729AB
- G.722**

Note:  
Only codecs with a white background are supported on this platform.

**Video Codec Priority**

Cancel Reset to Default View Modifications Save

## 7. Verification Steps

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

### Step1

Verify that the Poycom VVX 500/600 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 the SIP Line user's ID that is being checked.

```
[admin@sip175 ~]$ slgSetShowByUID 54504

=== VTRK ===
UserID          AuthId          TN                Clients  Calls  SetHandle  Pos
ID      SIPL Type
-----
---
54504          54504          104-00-01-04          1        0  0xb4440f68
SIP Lines
  StatusFlags = Registered Controlled KeyMapDwld SSD
  FeatureMask =
  CallProcStatus = 0

  Current Client = 0, Total Clients = 1

  == Client 0 ==
  IPv4:Port:Trans = 10.10.5.36:5060:udp
  Type            = Unknown
  UserAgent       = PolycomVVX-VVX_500-UA/5.0.0.7403
  x-nt-guid       = 73b8a83dce189b3ec137120debfc554d
  RegDescrip      =
  RegStatus       = 1
  PbxReason       = OK
  SipCode         = 200
  hTransc         = (nil)
  Expire          = 3600
  Nonce           = 8fb8ebb650fcdf5a8e9e26a4430b0d40
  NonceCount      = 6
  hTimer          = 0x9d862d8
  TimeRemain      = 2880
  Stale           = 0
  Outbound        = 0
  ClientGUID      = 0
  MSec CLS        = MSNV (MSEC-Never)
  Contact         = sip:54504@10.10.5.36
  KeyNum          = 255
  AutoAnswer      = NO
```

Key	Func	Lamp	Label	
	0	3	0	54504
	1	126	0	2654504
	3	3	0	54505
	4	2	0	54506
	5	22	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**

From the physical phone display of VVX 500/600 navigate to **Menu → Settings → Status → Lines** (not shown). Verify that the Lines information shows the successful registration of the VVX 500/600 phone to the SIP Line Gateway of CS1000.

Place a call from and to the VVX 500/600 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 VVX 500/600 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 VVX family of phones may be found at:

<http://support.polycom.com>

[1] *Communication Server 1000E Installation and Commissioning*, March 2013, Release 7.6, NN46041-310

[2] *SIP Line Fundamentals - Avaya Communication Server 1000*, March 2013, Release 7.6, NN43001-508.

[3] *Element Manager System Reference – Administration - Avaya Communication Server 1000*, March 2013, Release 7.6, NN43001-632.

[4] *Co-resident Call Server and Signaling Server Fundamentals - Avaya Communication Sever 1000*, March 2013, Release 7.6, NN43001-509.

[5] *Unified Communications Management Common Services Fundamentals - Avaya Communication Server 1000*, March 2013, Release 7.6, NN43001-116.

[6] *Administering Avaya Aura® System Manager*, October 2013, Release 6.3.

[7] *ISDN Primary Rate Interface Installation and Commissioning - Avaya Communication Server 1000*, March 2013, Release 7.6, NN43001-301.

[8] Polycom VVX 500/600 Documents:

<http://support.polycom.com/PolycomService/support/us/support/voice/index.html>



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